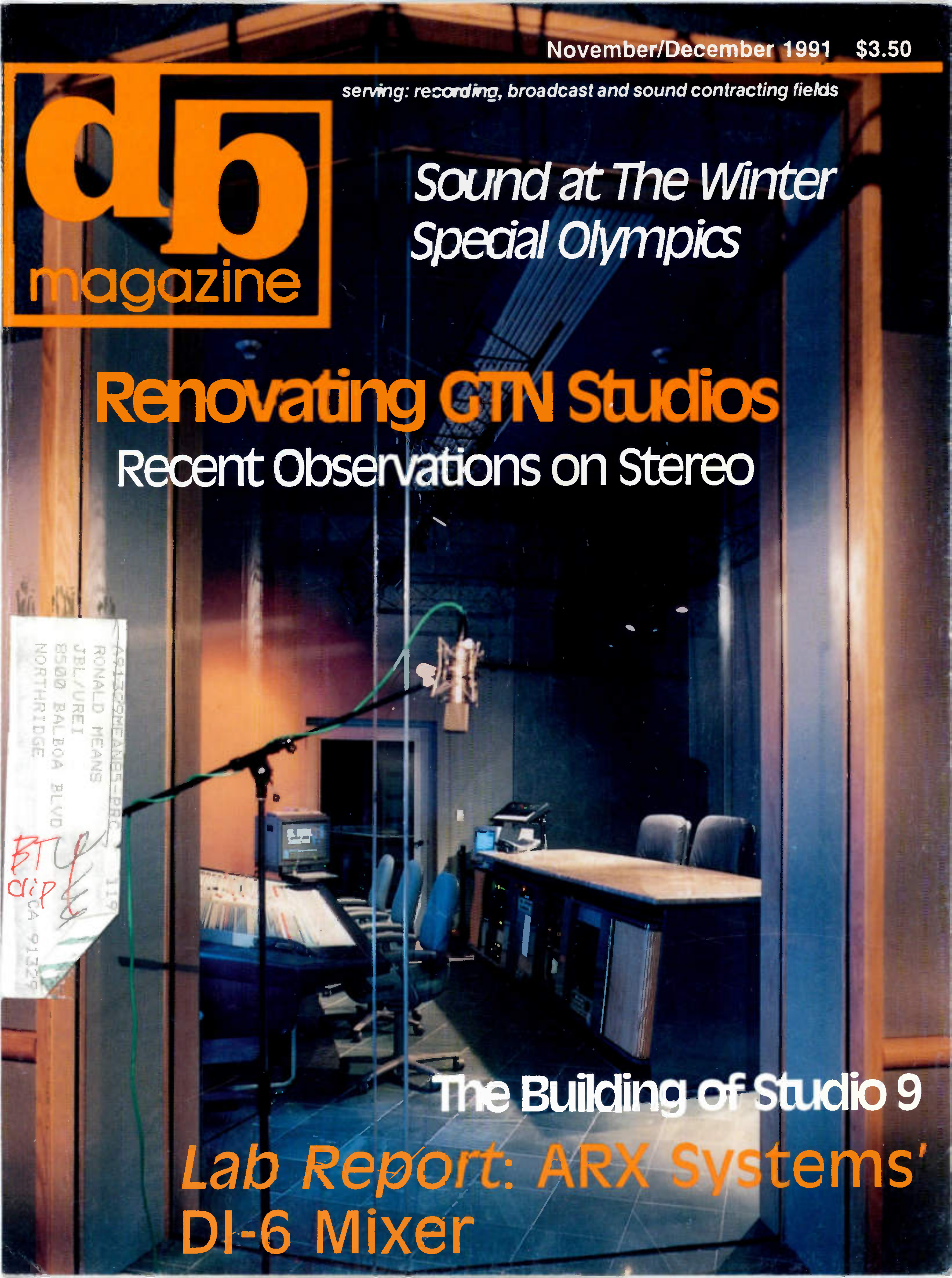
November/December 1991 $3.50

db magazine

serving: recording, broadcast and sound contracting fields



Sound at The Winter Special Olympics

Renovating GTN Studios

Recent Observations on Stereo

The Building of Studio 9

Lab Report: ARX Systems' DI-6 Mixer

BT clip

Phil Ramone photos by Michael Bloom

We asked Phil Ramone to field test our new AT4033 studio condenser microphone.

He wouldn't give it back!

AT4033
Studio Condenser
Microphone

Phil Ramone knows exactly what he wants from a studio microphone. And when he tested a sample of our new AT4033 cardioid condenser microphone, he knew it was right for him and ideal for the artists he records.

He liked being able to concentrate on getting the right music from the musicians, rather than first spending time experimenting with EQ to get the right sound.

What Phil Ramone heard was the result of new condenser technology inside the AT4033. The diaphragm is only 2 microns thick, with a vapor-deposited gold conductive coating just 50 angstroms thick. This sophisticated, very low-mass diaphragm is aged in a five-step process that insures unchanging performance for years.

The high head-room and wide dynamic range, plus low noise floor, make the AT4033 ideal for the most demanding digital recordings. And the maximum input SPL is an awesome 140 dB, so important when recording high-output instruments and very close-up vocals. In addition, transformerless design contributes to overall sonic transparency. The AT4033 also includes a switchable 10 dB pad and lo-cut filter, plus a built-in pop filter and internal shock mounting.

We're not certain we'll ever get the sample AT4033 back from Phil Ramone, but no matter. We're busy making your AT4033 right now. For more details on this impressive new microphone, ask your A-T sound specialist to schedule a test of the AT4033 today.

audio-technica®

Available in Canada and the U.S.A. from Audio-Technica U.S., Inc.
1221 Commerce Drive, Stow, Ohio 44224
(216) 686-2600 • FAX (216) 686-0719

Circle 10 on Reader Service Card

November/December 1991 Vol. 25, No.6

db MAGAZINE

serving: recording, broadcast and sound contracting fields

See page 8

Film star Ben Lyon and Howard Hughes by Howard's tapered-wing WACO which Howard was trying to sell to Bill Stancil. The picture is autographed to Bill Stancil by Lyons. See our story beginning on page 38.

ABOUT THE COVER

• When a major post-production studio does a renovation almost from the ground up, it's news. Such was the case at General Television Network in suburban Detroit, Michigan. They've included lots of glass and natural lighting, creating a place to do serious work that is also pleasing and comfortable. The heaviness of their booking also attests to the fact that "world-class" is an appellation that applies as well. *See our story beginning on page 4.*

THE RECORDING ENGINEER

THE SOUND CONTRACTING ENGINEER

THE ELECTRONIC COTTAGE

AUDIO FOR THE CHURCH

DEPARTMENTS

CALENDAR

• The third in a series of **Concert Sound Reinforcement Workshops** will be sponsored by **Synergetic Audio Concepts** from January 14-16, 1992 in Orange, CA. Major touring sound companies such as Audio Analysts, Clair Brothers, Electrotec and Showco will be represented at the workshop which will differ from the two previous concert sound workshops. For more information, please call **Synergetic Audio Concepts at (812) 995-8212 or fax at (812) 995-2110.**

• The 26th **Annual Television Conference** of the **Society of Motion Picture and Television Engineers**, whose theme is "Collision or Convergence: Digital Video/ Audio, Computers, and Telecommunications" will be held on Friday, Feb. 7 and Saturday, Feb. 8, 1992 in San Francisco, CA, at the Westin St. Francis Hotel. **For more information, please call SMPTE at (914) 761-1100.**

• Avid Technology, Inc. has announced its Fall 1991 series of **Avid Media Compser** training classes. The courses are designed to provide a hands-on introduction to digital, non-linear editing with topics covering Media Composer components and data flow, terminology and concepts. During the three-day course, students will learn basic operation of the Media Composer through lecture and hands-on exercises, examining the steps involved in going from raw footage to assembly of a master tape. Upon completion of the course, participants will receive a certificate of completion and a one year subscription to Avid's electronic bulletin board system.

Courses will be offered in New York from Nov. 13-15 and Nov. 18-20; Dallas from Dec. 3-5; Atlanta from Dec. 10-12; and Boston from Dec. 17-19. The cost of the training course is $350.00. **To register for an upcoming class, please call Kim Hajjar at (617) 221-6789, ext. 229.**

db MAGAZINE

Editor/Publisher
Larry Zide

Associate Publisher
Elaine Zide

Senior Editor
John Barilla

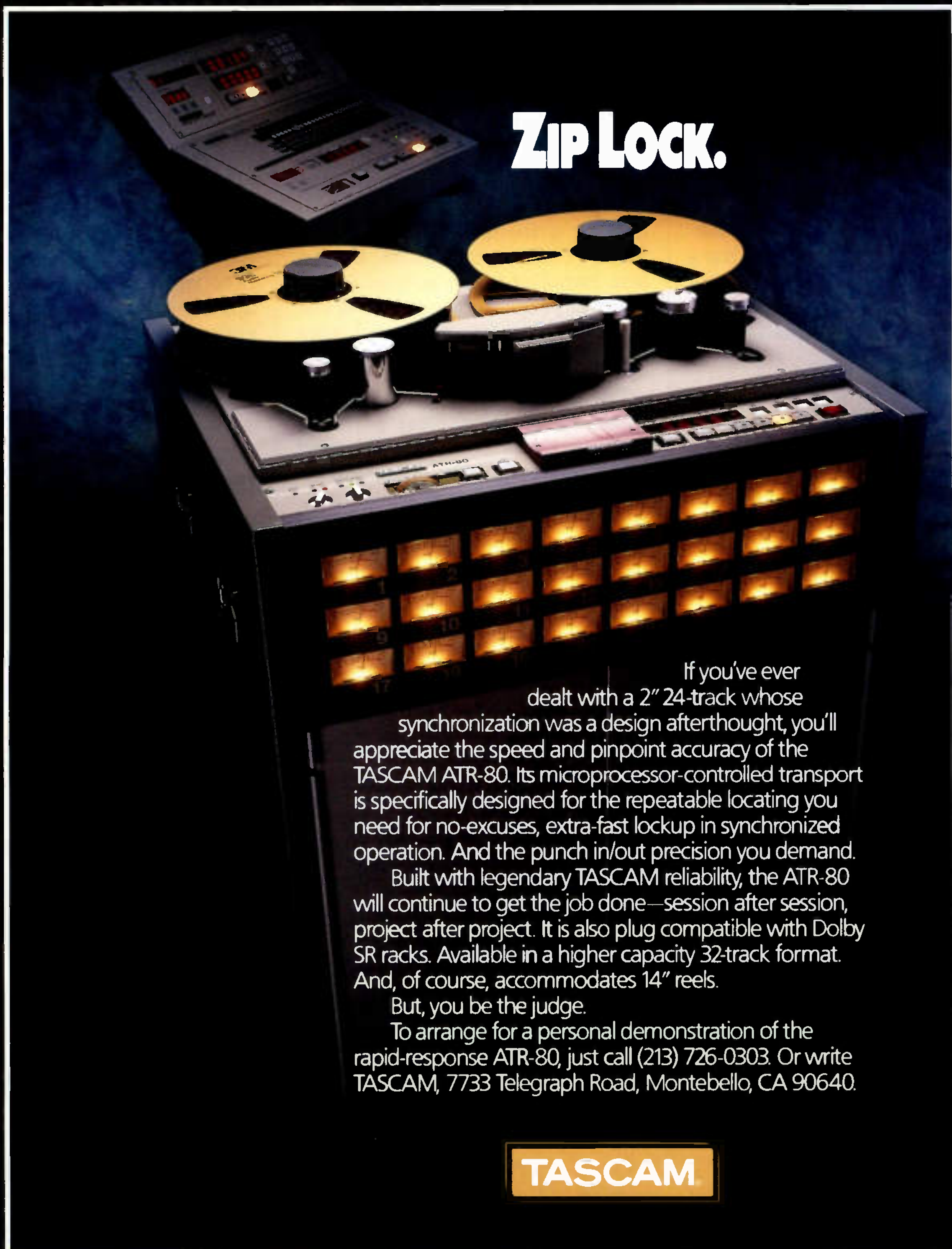
Assistant Editor
Caryn Shinske

Contributing Editors
Bruce Bartlett
Brian Battles
Drew Daniels
Robyn Gately
Len Feldman
Shelley Herman
Brent Harshbarger
Randy Hoffner

Graphics & Layout
Karen Cohn

db, The Sound Engineering Magazine(ISSN 0011-7145) is published bi-monthly by Sagamore Publishing Company Inc. Entire contents copyright 1991 by Sagamore Publishing Company Inc., 203 Commack Road, Suite 1010, Commack, NY 11725. Telephone: (516)586-6530. db Magazine is published for individuals and firms in professional audio recording, broadcast audio-visual, sound reinforcement-contracting, consultants, video recording, film sound, etc. Application for subscription should be made on the subscription form in the rear of each issue. Subscriptions are $15.00 per year ($28.00 per year outside U.S. Possessions, $16.00 per year in Canada)and payable in U.S funds. Single copies are $3.50 each. Editorial, Publishing, and Sales offices are at 203 Commack Road, Suite 1010, Commack NY 11725. Second Class postage paid at Commack, NY 11725 and an additional mailing office. **Postmaster: Form 3579 should be sent to db Magazine, 203 Commack Road, Suite 1010, Commack, NY 11725.**

Trademarked names are editorially used throughout this issue. Rather than place a trademark symbol next to each occurance, we state that these names are used only in an editorial fashion and to the benefit of the trademark owner, and that there is no intention of trademark infringement.


ZIP LOCK.
If you've ever dealt with a 2" 24-track whose synchronization was a design afterthought, you'll appreciate the speed and pinpoint accuracy of the TASCAM ATR-80. Its microprocessor-controlled transport is specifically designed for the repeatable locating you need for no-excuses, extra-fast lockup in synchronized operation. And the punch in/out precision you demand.
Built with legendary TASCAM reliability, the ATR-80 will continue to get the job done—session after session, project after project. It is also plug compatible with Dolby SR racks. Available in a higher capacity 32-track format. And, of course, accommodates 14" reels.
But, you be the judge.
To arrange for a personal demonstration of the rapid-response ATR-80, just call (213) 726-0303. Or write TASCAM, 7733 Telegraph Road, Montebello, CA 90640.
TASCAM

JOHN BARILLA

Designing Vocals: Part IV

● In the previous three installments of this series on "Designing Vocals," we have explored the various technical and performance factors in recording vocals and have begun to investigate some of the factors involved in mixing vocals: dynamics control and equalization.

We have essentially dealt with the real "nuts and bolts" issues in achieving a professional vocal sound. (If this is an area of interest to you and you haven't already read the first three parts, I would urge the reader to get the May/June, July/August and September/October 1991 issues of **db Magazine** and check them out, as they do lay the groundwork). What we are dealing with in this final installment is less foundational, but no less important: the spatial placement of vocals in the mix and the construction of a vocal ambience using reverb and delay.

The truth is, in order of importance, a smoking vocal performance that has been well-recorded will supersede any "sweetening" that you can apply to it in the mix. It's a bit like a delicious meal that has been haphazardly thrown into the serving plate; once you taste it, you will enjoy it anyway, but if it is visually attractive as well—symmetrically arranged with garnishes—it will be a total sensory experience that will gain the chef rave reviews. The same principle applies for vocals: the same diligence and intelligent technique should be applied right down to the level of garnishes in order to maximize the impact of the song.

PLACEMENT OF VOCALS IN THE MIX

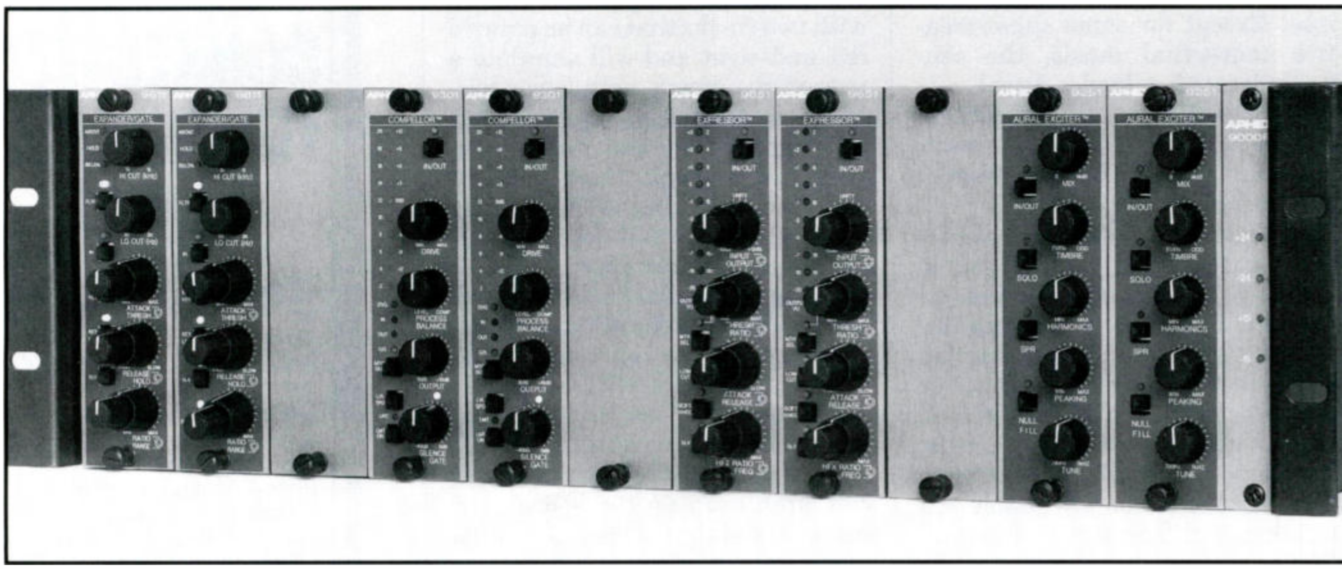
This arena has been rigidly conventionalized in the past twenty years or so. Whether it should be or not is, in my mind, a matter of debate. Think, for example, of the early Beatle albums or the great Motown hits. Vocals and instruments (even bass and drums!) might have appeared solely in the right or left channel with only a trace of ambience coming from the opposite side. By the monolithic standards of today's music, such panning options would appear naive or even crude; still, it seems that much of the charm of those records was tied to that unorthodox method of spatial organization.

Certainly, such radical panning would not be appropriate in much of today's pop music production, but if you are searching for a hint of originality in your music, it might be advantageous to first understand this unwritten standard and then see if there might be some ways to stretch it beyond the normative. The standard format is, of course, lead vocals placed dead center and background vocals—stereoized in some manner—placed hard left and hard right.

TRIED AND TRUE

This is a safe, tried and true formula that works quite well. The

Space Saver...



Aphex Modular System

Once hooked on Aphex signal processing, people have an insatiable appetite for more and more. Space then may become a problem. That is why we shrank four of our best ... the Aural Exciter®, Compellor®, Expressor™ and Expander/Gate.

These modules feature all the processing power and performance of their standalone counterparts including our servo-balanced inputs and outputs. You can fit 11 modules in our 3RU Model 9000 rack* (or nine in the compatible dbx 900 Series rack).

Here's a brief rundown on these powerful tools:

- Aural Exciter- the signal enhancer that increases intelligibility, presence, clarity, and detail.
- Compellor - the "intelligent" compressor/leveler that controls levels as if a mixer were riding faders.
- Expressor - a full featured compressor/limiter that lets you tailor the sound your way.
- Expander/Gate - simply the world's finest gate, no one ever met our $10,000 challenge to find a better one!
- And, more to come.

See your nearest professional audio dealer to rack up more processing power per inch than ever before.

APHEX
SYSTEMS 11068 Randall Street • Sun Valley, CA 91352 • (818) 767-2929 • FAX (818) 767-2641

Aphex is proudly American ... 100% owned, engineered and manufactured in the U.S.A.

Circle 18 on Reader Service Card

hard-panned background vocals provide a nice cushion for the lead and create an auditory "hole" in which the lead vocal can be localized. Except for some super-creative conceptual music, the centrally-located lead vocal is something that you should not easily depart from, but the backgrounds can certainly be treated with some latitude.

In most cases, the background vocals benefit greatly from double tracking. If you don't have time or tracks to do that though, even a single track can be artificially doubled by using an appropriate delay or chorus setting. (We'll deal with the details of that later). The point is, whether real or artificially doubled, two tracks of vocals will enable you to get a fat, non-localized sound that hovers around the lead without getting in the way. This will undoubtedly give your backgrounds some needed dimension and offset them from the lead. If, however, you'd like to take things somewhat beyond the status quo, I'll give you some ideas that you can implement without too much trouble.

Did you ever listen to the background vocals on a classic Steely Dan record? Check them out closely on headphones sometime and you'll have an example of the technique I am about to share with you.

If there is three-part harmony on a certain song, you can usually hear the three parts laid out panoramically—the lowest part being weighted towards the left side, and the highest part being weighted towards the right side. It's as though you are right there with three people standing in front of you in a semi-circle. The added dimension is subtle, but very life-like. How they achieved that sound I cannot say for sure, but they probably used stereo microphones or maybe even separate mics for each vocalist, blending and panning them later on in the mix.

ALTERNATIVES

While most of us don't have such sophisticated mics or the luxury of extra tracks, we can still achieve a very similar effect with just a standard cardioid mic and two tracks (or even just one) to record on. By simply switching the proximity of the low and high vocalist relative to the single mic (bringing one or the other closer to the mic, hence making that part louder relative to the blend), you will end up with two tracks that can be panned left and right and will simulate a true stereo image, with lower harmony predominating from the left and higher from the right. Even with just a single track you can simulate this kind of realism by creating an artificial double utilizing a delay and/or chorus effect—panning the actual track hard left and having only the affected signal emerge from the right—and carefully equalizing both channels to accentuate the low or high harmony. The equalization is usually never done, but it's a neat trick and can often open up the space even more than simple stereoization of a monaural track.

Another thing you might be radical enough to try doing sometime is to let the panning of the background vocals be asymmetrical. This works particularly well if the vocals are catchy and antiphonal (like a call and response—an answer to the lead vocal). Having the balance shifted somewhat to one side will often draw more attention to the backgrounds then if they were neatly in place—equal volumes left and right. You can still maintain stereo imaging even if it does shift towards one side or the other. If you don't want to use this technique on the whole song, try it on just the bridge or some section that is offset from the rest of the song. Go ahead and upset the apple cart—it can be very effective!

THE USE OF AMBIENCE

There is a lot of interaction between the panning and ambience in a slick mix. There is also much more to it than saying, "Yo, throw a little echo on the voice." Questions like, "Well, where is the voice located and what kind of echo and where do you want the echoes to come from?" are key factors that make a big difference in the sound.

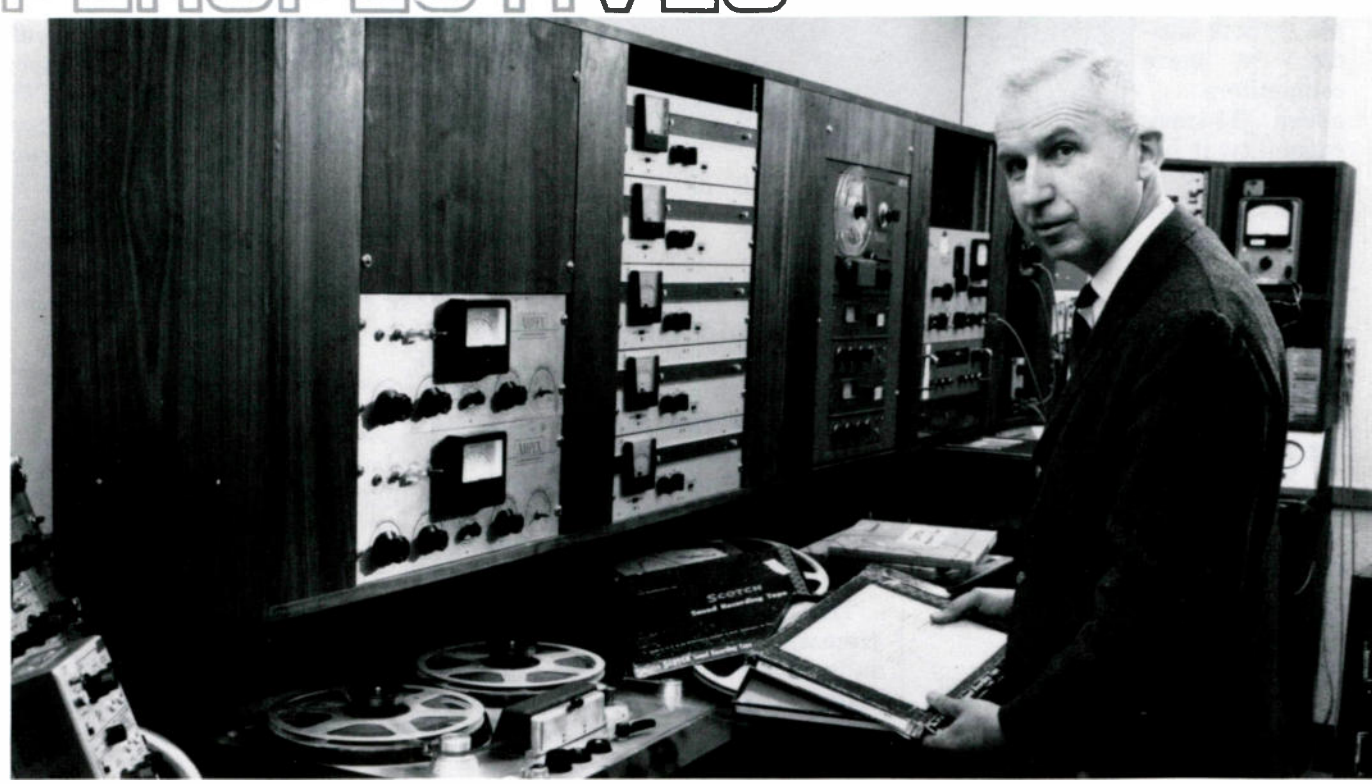
For example, in the case of a center-panned lead vocal, a chorus effect that was returned to the mix in stereo would have a decidedly different result than one that was returned in mono to the center channel. In the former case (stereo returns), the vocal would have a soft, wide, spacious sound; in the latter (mono), it would have a more pointed, localized effect. What is the bottom line here? Well, chances are that the stereo chorused lead vocal would have to be placed significantly higher in the overall mix to cut through the instruments, whereas the monaural chorused vocal—being tighter and more well-defined—could be placed further back in the mix and still be heard. Both are valid approaches, but the question one needs to ask is, "Do I want a vocal dominated pop mix or do I want more of a rock style mix where the vocal and instruments are equally important?" Deciding which way you intend to go—in advance of patching in your effects—can make mixing much easier.

'Your', of course, is not necessarily either/or. If you have two or more effects devices, there is a lot of room for cross-pollination between the various effects. Returning to our lead vocal example: if increased presence and intelligibility of lyrics is desired, a tight echo (say 30 to 50 ms) returned in mono—sitting right behind the vocal so as to bolster each syllable—will really help out a lot. (An early reflections program can be used in a similar way.) This, of course, tends to one need, but there are certainly others. In most cases, there will be a call for a "global" stereo reverb—something which can be applied almost universally in small or large doses to perhaps most of the tracks (both instrumental and vocal).

Sometimes, however, reverb alone is not dense enough to give the fluidity you might want from a vocal. Here multiple long echoes in stereo (utilizing delay times of 200-400 ms and even higher with a smattering of regeneration) can be very helpful. If the returns from these various devices can be patched back through channels on your mixer (rather than the normal effects returns), all kinds of interesting things can happen. For example, the long echoes can be fed into the send to the global reverb—adding complexity to the reverb return. The reverb return can also be sent back into itself (in controlled amounts) creating a reverb feedback loop. Sometimes I get several different types of ambience going with all of them slightly cross-feeding-back into themselves and/or the other devices. With a little fooling around, you will be able to design a unique ambience field that would be otherwise impossible to achieve.

There is certainly a lot more that could be said about designing vocals—much more than could ever be contained in four magazine articles. While the subject could not be treated exhaustively, I hope the series has exposed all the important areas you should be constantly thinking about if you really want to improve your vocal sound. The bottom line is this: while good equipment is always an asset, your attitude is most definitely the decisive factor in getting a great vocal track on tape and placing it in the mix. Experiment intrepidly; utilize your creativity and you'll have good success in designing vocals. db

Historical PERSPECTIVES



Jack Mullin returned from post-war Germany with two Magnetophon audio recorders and several reels of paper-backed audio recording tape. Mullin worked with Bing Crosby to pre-record Crosby's natiowide radio variety show on Scotch brand audio tape, and later joined 3M and worked on several audio and video recording tape products.

GARY PILLON

Renovating GTN Studios

The tremendous growth of audio-for-video in the last five years has left many facilities in a state of confusion. Market pressures, client needs, equipment purchases and operator talents have to be weighed delicately before a studio owner can arrive at a profitable solution.

THIS WAS THE QUANDRY FACED by General Television Network, a full-service teleproduction facility based in Oak Park, MI. As a major player in the Detroit and Midwest marketplaces, GTN had been concentrating on video location/studio shooting, film transfer, video post production and duplication. Audio Post was served by a simple 4-track studio. As more competitors added 24-track capability, it became difficult to keep the more sophisticated jobs from walking to the "one-stop-shop" down the street.

GTN's Owner Joan Binkow and her senior management team held a series of meetings with the following agenda:

1. Who are our present/future clients and what are their needs?

2. What equipment and physical plant will be needed to meet these present and future requirements?

3. Who can we find to both run the machines and bring in the business that has been going elsewhere?

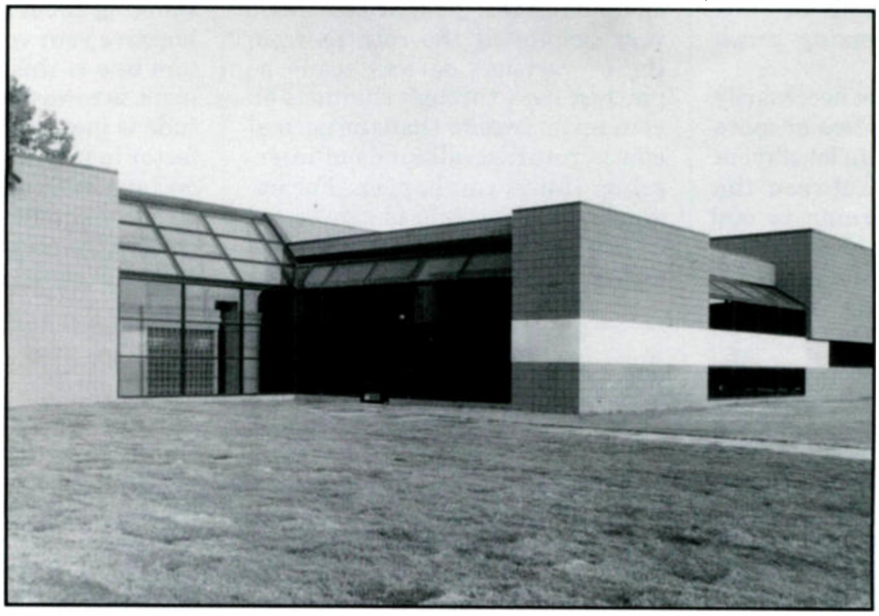


Figure 1. GTN's new multi-million dollar audio post production facility in Oak Park, Michigan.

FREE-FOR-ALLS

As you can imagine, some of these meetings turned into real free-for-alls, since this agenda calls into question the very reason for a company's existence. When the dust settled, however, a long range plan was adopted to make GTN one of the premiere facilities in the country.

The first step was to buy some time while servicing the company's industrial client base. The 4-track room was gutted, and GTN took delivery of the first Lexicon Opus Digital Audio Workstation to be placed in a video facility. With one purchase, GTN replaced the analog console and multi-track, and added the flexibility of random access editing. The familiar console layout and computer screen film log display made the workstation easy to operate and added to clients' acceptance of the new technology.

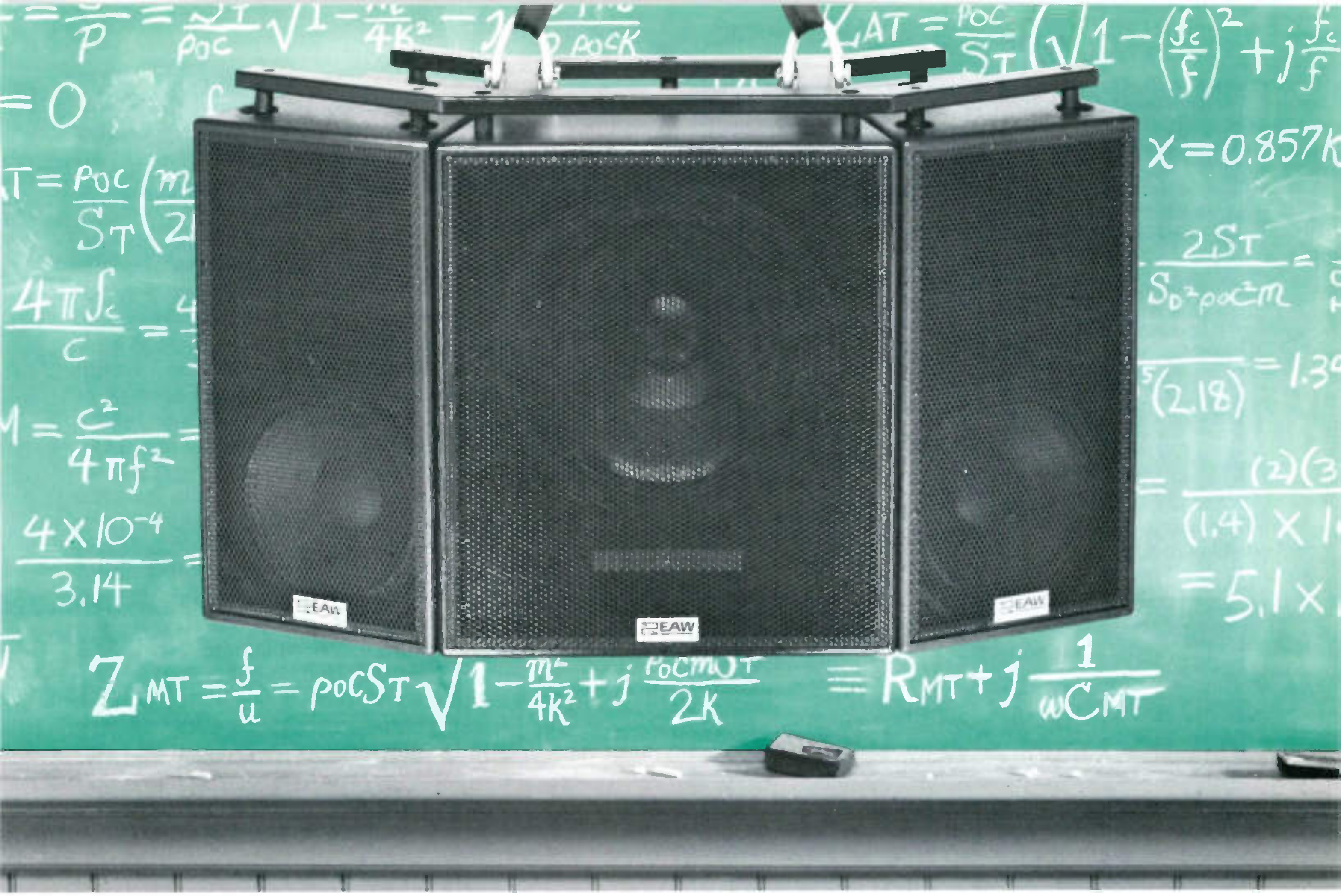
The addition of a Lexicon 480L reverb, an Eventide 3000 Ultra-Harmonizer, Aphex processing, Benchmark microphone pre-amps, Beyer 740 booth mic and Bryston bi-amped PAS monitor speakers made the former 4-track room the ultimate 8-track room. Paul Stelly, GTN's Opus operator, could now load basic tracks from any video formats, add sound effects and music from newly acquired CD libraries, and then restripe the master under time code control.

OTHER ROOMS

At the same time, the rest of the plant was also undergoing an audio upgrade. New consoles had already been put into every video suite, and each room was given its own Orban parametric equalizer, as well as a UREI compressor/limiter. All of the Betacam machines had built-in Dolby C noise reduction, so GTN focused their attention on the workhorse 1 in. decks. Six Sony 3000s were retrofitted with the new Dolby A/SR cards, and the existing 2000 editing machines were wired

---

Gary Pillon is a 21 year veteran of General Television Network. He is the recipient of his second Crown PZM Challenge Grand Prize.



We're making the laws of physics work harder than ever.

If you think it's easy to develop full performance from an enclosure this size (3.4 ft.³), just listen to any other ultra-compact system.

If you already have, you've probably decided it's impossible for anything this small to combine high output with high definition. In that case, you really should hear our new KF300 Series.

These are true 3-way designs—unheard of in this size class, but a fundamental principle in all EAW full-range systems. An advanced midbass horn and ultra-rigid carbon-fiber cone driver cover the entire midband, producing over 130 dB SPL with lower distortion than comparable two-way systems. The custom-designed woofer uses a flat wire wound voice coil and massive, optimally aligned magnet structure to achieve exceptional efficiency and surprisingly impressive bass.

True 3-way design also makes the KF300 Series a genuine Virtual Array™ system. Horizontal coverage is uniform from 400 Hz all the way to 18 kHz, and precisely matched to the enclosure angles of two KF300i's arrayed with our new SB330 subwoofer. The result is consistent, predictable coverage without comb filtering, lobing or hot spots.

The KF300 Series includes a variety of hardware configurations and powering options, such as the horizontal AS300 enclosure, designed for distributed systems. Each version delivers coherent output, controlled directivity and wide full power bandwidth in concert sound fill coverage, theatrical systems, dance clubs, corporate theater and other demanding applications.

The MX300i CCEP™ (Closely Coupled Electronic Processing™) unit provides overload protection, LF excursion control, phase compensation and idealized crossover characteristics for the KF300i.

It was no small task to make so little do so much. But at EAW, we insist that even our smaller systems embody big ideas.

EAW
EASTERN ACOUSTIC WORKS

One Main Street, Whitinsville, MA 01588
(508) 234-6158 • (800) 992-5013 • Fax (508) 234-8251
In Canada: DARPEX
(514) 521-5895 • Fax (514) 521-0232

Circle 19 on Reader Service Card

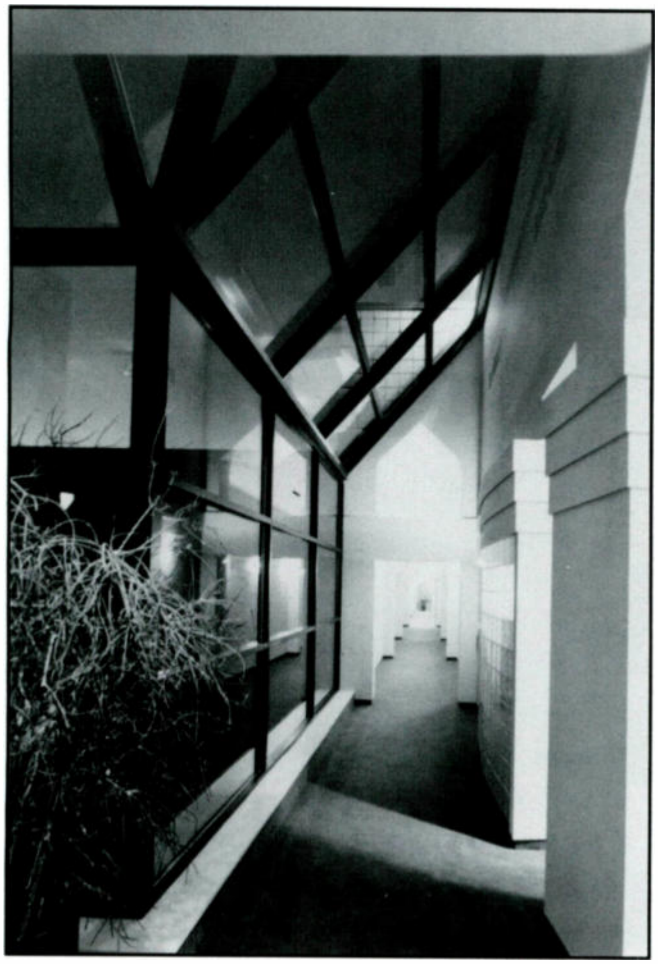


Figure 2. Natural lighting from hallways and skylights give a sense of spaciousness and comfort.

and construction, and GTN now has a showplace that works very well.

THE NEW SPACE

The 6,700 sq. feet of new space designed by Berger and interior designer Alexis Lahti allows maximum creativity, productivity and comfort. Each room, although built on a similar floor plan, has been tailored to match a specific segment of the burgeoning audio marketplace. AP 1 is aimed at the no-holds-barred national spot/program market. Mixers Scott, Stelly, and the newly acquired Bob Meloche command a forty-input version of Solid State Logic's latest 6000 G series console. Each input has its own compressor/limiter and gate section to complement the SSL Equalization modules.

Floor-to-ceiling glass panels provide good sight lines between talent, mixers and clients.

to an external Dolby rack that would follow record and playback commands. Two D-2 video machines were also added for pristine digital video and audio signals. GTN could now keep the original materials as clean as possible before going to Audio Post.

A NEW ADDITION

Meanwhile, the plans for a new addition were under way. The Detroit marketplace still revolves around the auto industry and its advertising agencies. Since GTN was already serving the training end of the business with its current plant, they had to find out what was needed to get a piece of the lucrative national advertising business. Several months of client interviews gave a clear design for a successful facility. It would need large areas devoted to clients, since the jobs at this level always attract writers, producers, directors and account people in large numbers. The rooms would need to be aesthetically pleasing and comfortable, since they would be judged against national level facilities. The gear had to be state-of-the-art and fast enough to deal with the short turnarounds and constant revisions that are part of the auto business, while operators would need a proven track record in this specialized field.

Given these requirements, the management team decided to start with a clean sheet of paper that soon turned into reams of documentation. Russ Berger Design Group Inc. of Dallas, TX, was hired to come up with an entirely new facility devoted to audio, film transfer and graphics.

This new plant would be located next to GTN's existing building, yet have its own heating and air conditioning equipment. Clients could enter from the main lobby, yet be isolated from other activities in the building. Two main audio rooms were planned, along with an off-line suite and room for further expansion. Jay Scott, a veteran mixer with years of automotive experience, was hired to design the equipment package and run the main room. Add two years of planning

Dedicated compression and EQ can also be added to the program outputs to fine tune virtually every parameter of the incoming signals, work out an automated mix that can be updated, and store the results by using the SSL Total Recall system. In case any revisions are needed, all of the board's I/O settings can be reloaded in a matter of moments. Processing includes the Lexicon 480L, Eventide 3000 and Yamaha reverbs, as well as a Dolby cat. 43 card for dialogue work. Other processors include the Aphex Aural Exciter and Orban De-esser. All of this gear was needed because of the emphasis placed on the voice during a commercial message.

To record these voices, a huge V.O. room was designed into AP 1's space. Like all of the rooms, it uses the latest floating floor, non-parallel walls, room-within-a-room and noiseless air conditioning techniques to deliver the highest quality original tracks. This "booth" is large enough to accommodate a talent group, foley and acoustic music recording. Floor-to-ceiling glass

Both methods record incredible stereo. Crown's SASS™-P makes it fast and easy.

Crown's new Stereo Ambient Sampling System™ (SASS™) and the DAT recorder combine to create a new generation of stereo recording. Unlike traditional recording methods, this powerful new combination is easy to use, more dependable and less expensive. The result is a system that records natural, beautifully imaged stereo just as your ears would hear it.

Crown's SASS-P* eliminates traditional stereo recording compromises in sound quality, ease-of-use, and cost. No longer do you have to settle for weak low-end or off-axis coloration common to Midside, X–Y and near-coincident pair mics. Assembly and positioning time is also reduced significantly compared with conventional stereo micing techniques.

*Also available as SASS-B for use with B&K 4003/4006 microphones (not supplied).

Lightweight and easy to set up, the SASS/DAT combination is a high-quality portable recording system that goes everywhere.

The SASS-P features switchable battery or phantom power and Crown's finest studio-grade PZM® capsules. You'll enjoy full ambience without coloration, excellent sum to mono, and extraordinary broad frequency response. With SASS's superb imaging capabilities, every sound is audibly reproduced in its precise position, resulting in a stereo experience of uncanny realism.

Readily adaptable to all common stands, the SASS includes a carrying case and accessories.

No matter what your stereo recording requirements—from sampling to electronic news gathering to remote recording of live events—you'll find the SASS-P the clear choice.

Circle 20 on Reader Service Card

See your Crown representative for more information or call toll-free: 1-800-535-6289.

SASS Sample CD!

Now you can hear for yourself just how great SASS sounds. This CD includes a wide selection of live and studio music, stereo sound effects and samples, including A-B comparisons with some of the best stereo mics available. The SASS CD is free, but we must charge $4.95 for shipping and handling charges. Send check or money order to: Crown SASS CD, P.O. Box 1000, Elkhart, IN 46515-1000. Please allow 3-4 weeks for delivery.

Name ______________________

Title ______________________

Company ______________________

Street ______________________

City ______________ State ______ ZIP ______

Phone ______________________

crown®

Crown International, Inc. • P.O. Box 1000
Elkhart, Indiana 46515-1000 • 219/294-8000

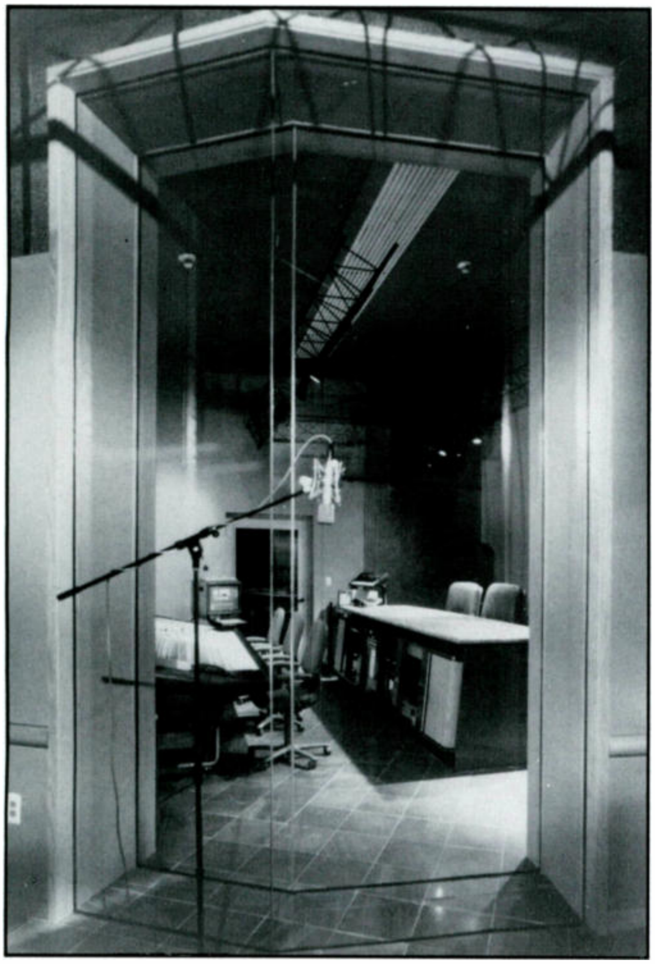


Figure 3. The entrance way to the audio facility. This photo, in color, is on our cover.

panels provide good sight lines between talent, mixers and clients. Video monitoring is handled by a Mitsubishi 35 in. video monitor, while clients and talent have their own dedicated Sony feeds.

The audio is fed by a custom surround circuit specially adapted for SSL's 6000, an SSL first. It can take a mix from mono to stereo, then go to full Shure Stereosurround in three clicks. There's enough busing and group assignments to allow full stereo music and effects stems while keeping stereo dialogue tracks isolated for possible second language revisions.

STEREO SURROUND

The Shure left/center/right/surround, using tri-amped KRK speakers, is designed for television use and allows full-bandwidth signals to be fed to the surround channel. A client's imagination isn't limited to the space between the stereo speakers, but is allowed to roam through a full three hundred sixty degree sound field. The best part of this system, of course, is that the surround matrix is imbedded in the stereo mix. As the number of home decoders (now over three million) climbs, GTN's (and their clients') work develops a greater residual value. To feed AP 1, GTN assembled some of the very best analog and digital machinery available, and snuck in some that hasn't been seen before now. For example, GTN picked up sixteen channels of ScreenSound, SSL's hard-disk digital audio system.

To make it work throughout the plant, GTN ordered the world's first SoundNet, manufactured by SSL, which allows data and channels to be accessed from multiple locations. Fed by optical and tape storage media, the ScreenSound can quickly reload projects done at GTN or elsewhere for mixing on the 6000 system. Waveform editing, track swapping, track slipping and the ability to call up sound effects from an on-board library make it the perfect tool in a deadline-oriented environment.

A full complement of the best Otari recorders provides an analog option. Over the last two years, GTN has added MTR12/15 ¼ in. time code and ½ in. 4-track machines as feeders and storage devices. To round out the lineup, GTN also obtained Otari's MTR100-A 24 track 2 in. recorder and loaded it with full Dolby SR "digital quality" noise reduction, as well as patchable dbx outboard boxes. While each suite contains a ¼ in. center track machine, the ½ in. and 2 in. tape decks can be called into a session without leaving the centrally-located machine room by using Motionworker, a sophisticated machine control system.

Designed and installed by systems integrator 21st Century Ltd. of Los Angeles, this arrangement allows control of all video and audio machines through a network of Audio Kinetics synchronizers. It will even support a write-once/read-many video disc that was added to speed up random-access editing. The central location also permits routine maintenance and air handling without disturbing any sessions.

AP 2 is equally sophisticated in its approach to good sound. A Lexicon Opus Digital Audio Workstation, previously housed in a 240 sq. foot suite, finally has the room and the KRK speakers to do it justice.

Since networking was a major design consideration, several Macintosh IIfx computer systems were also added to the complex and linked together by Ethernet. Voice, music, or sound effects can be recorded live, played from tape, or loaded from CD-ROM into Digidesign's Sound Tools computer software. They can then be positioned by Opcode's Studio Vision MIDI program or Digidesign's Qsheet AV and triggered by time code, or "played" by a Yamaha SY22

Figure 4. View across the SSL, showing outboard gear and the producer's desk.

MIDI keyboard using Digidesign's SampleCell. Each of the suites is equipped with the same package, so effects can be spotted at off-line rates and called up as files during the on-line session.

AP-2

AP 2 is equally sophisticated in its approach to good sound. A Lexicon Opus Digital Audio Workstation, previously housed in a 240 sq. foot suite, finally has the room and the KRK speakers to do it justice. Its new configuration of Sabre hard drives and multiple I/Os make it one of the best equipped Opus units in the country. Fully assignable digital equalization and the imminent addition of Lexicon automation make it the ultimate one-piece digital production system. Stelly can also access ScreenSound, the Mac programs, or any of the analog machines by using the various networks. He can then lay his work back to one of the video formats, including a digital transfer to D-2. A generously-sized voice-over room and client areas make AP 2 even more functional. Add a similar group of processing gear, combine it with our new R-DAT machines, and you have the makings of a first-rate CD pre-mastering house.

TRANSFER ROOM

A newly-expanded transfer room, operated by the author, can also be used in a number of ways. Time-coded Nagra field tracks will continue to be synced up with film transfer elements using Dolby C noise reduction on Betacam SP tapes, Dolby SR on 1 in. elements, and digital recording on the D2 format. This will give Scott, Meloche and Stelly the best possible raw materials for Audio Post. Audio Off-line will also be the pre-build room for music and sound effects searches. Close to five hundred music and sfx CDs are organized on the Gefen M&E computer program. Most of the computer networking will also start here, to offer clients the best price/performance ratio.

Another special feature of great importance to GTN's time-sensitive clients is the Landco satellite downlink that can be called into any of the rooms. It has been a tradition in Detroit to record national talent by using phone patches with New York and Los Angeles. Unfortunately, many clients cannot afford to wait for overnight delivery of the finished audio tapes. Now GTN can tie into major studios throughout the country with first generation digital transmission that will allow a client to edit, post and dub in the same day.

Both rooms are often booked far into the night, and producers are now bringing their video work to GTN so they can go down the hall to Audio Post, instead of driving across town...

Client concerns have been the driving force throughout the facility. Ample work space, beautiful surroundings, separate client lounges (with private monitoring), separate rest rooms and service bar are just part of the story. Improvements in every area were also undertaken. The Bosch "Real-Time-Steady" Film transfer has a new home, Graphics has new SuperMac workstations in addition to two Symbolics units, and Video Post now has six D-2 machines to augment its full array of recording formats. GTN even added a Cycle Sat satellite uplink to speed up the commercial release process.

Was it worth it? Both rooms are often booked far into the night, and producers are now bringing their video work to GTN so they can go down the hall to Audio Post, instead of driving across town or flying across the country. Clients are "voting with their feet" by beating a well-trod path to GTN's door. db

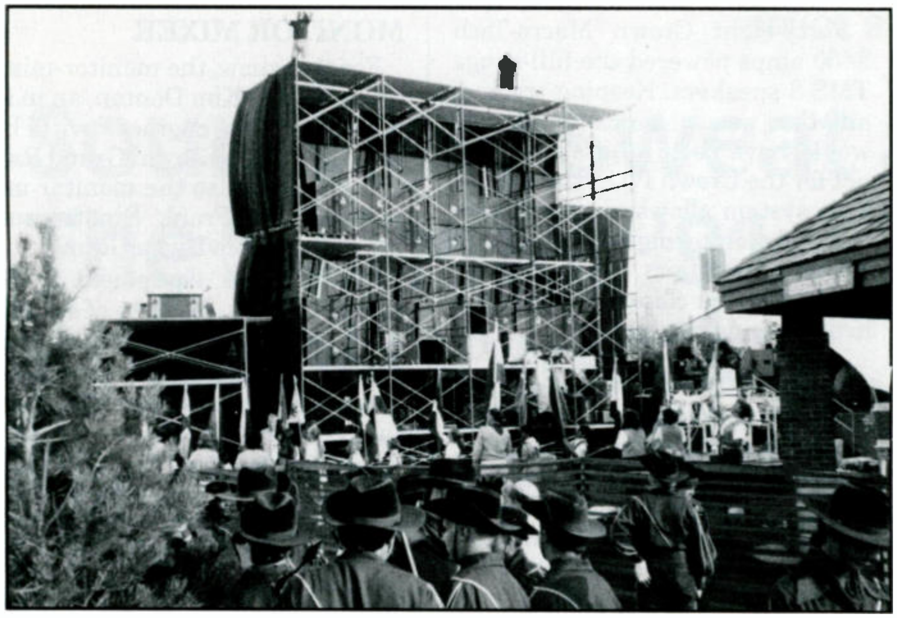


Figure 3. One of the louspeaker scaffolds at the International Prom.

the monitor system used two Ramsa monitor mixers. The Opening Ceremonies systems involved a four-way split to house, monitor, recording and video. Microphones went to a splitter in the recording truck, and from there to another splitter feeding the house and monitor mixers. Prince's mics were split to the monitor and house mixers. Stereo feeds from both house mixers went to the television audio mixer.

The $2.2 million show was videotaped for later broadcast. This vast, colorful spectacle featured the International Parade of Athletes, a most moving scene where athletes from countries all over the world gathered peacefully in a common purpose. There was the excitement of the Torch Lighting Ceremony and the concerts. Several celebrities of film and TV gave speeches, as well as Eunice Kennedy Shriver, chairperson and founder of Special Olympics International.

A glitch occurred here and there, just to keep us on our toes. When the show started, the Metrodome's overhead clusters had accidentally been shut off by a Metrodome employee, so the announcements were unintelligible until the clusters were turned back on. Another accident created a bizarre effect: an Eventide Harmonizer used for delay was knocked into pitch-shift mode, turning the announcer into Darth Vader. But Crown's Maurice Paulsen put it into context: "Hey, it was art!"

After the show, we helped strike the sound system. Some volunteers worked all night long at the task. I was awed by the number of intertwining snakes and cables, and the huge size and weight of the equipment. The roadies for the big touring sound companies really earn their money.

Figure 4. The mix position at the Entertainment Expo.

SPORTS SYSTEMS

Much smaller than the Opening Ceremonies system, but equally important, were the P.A. systems located at all the sports venues. Most of these systems included a prewired portable rack containing a mixer, cassette or CD player, and signal processors, plus a power-amp rack.

Generally, we aimed the loudspeakers across the playing field at the bleachers, making sure the athletes could hear the announcements. Often the speaker positions were a compromise between clear sound and safety: floor-mounted speakers had to be placed so the athletes wouldn't run into them in the heat of competition.

Here's an example of a sports-audio job. I was to amplify the announcers at three volleyball courts. The original request was for one mic amplified over three courts. I installed the system and taped the cables in place. But the day the Games started, the announcers told me they wanted three separate sound systems, one for each court. So I had a portable P.A. delivered for one court, panned the mics for the other two courts left and right, and re-routed and gaffed the speaker cables.

EXTRA INSTALLATIONS

Also on the day the Games started, I was told that another court needed a P.A. system. To get some sound going quickly, I scrounged up a small system that the venue owned. It lacked a mic, so I plugged in a headphone which the announcer used as a mic! Then I radioed the warehouse for a better system.

To cover a roving announcer in the large gymnasium, I used a Telex wireless system: an FMR-100 diversity receiver and HT-100 mic/transmitter. It worked great, transmitting over two hundred feet through bleachers and a metal door with no audible noise or dropouts.

One bus output of the mixer failed in the heat, so I patched into an aux out instead. By the end of the day, some of the rack equipment was almost too hot to touch, but kept on running.

When setting EQ, I aimed for a natural tonal balance on the announcers, but with a little less bass and a little more treble to aid intelligibility. The signal was limited to prevent blasting the audience and blowing the speakers.

In some areas the gain-before-feedback was marginal due to restricted speaker placement. So, near the announcer's mic I taped a note reading "Please talk close to the microphone."

Set up near a court, I got smacked by a few volleyballs and learned to duck behind the rack when I saw one coming—stuff they don't teach you in audio school! When I jumped up to pull a toddler away from a speaker plug, I snagged the headphone cable around my ankle, yanking the phones off and smashing them on the floor. They fell apart, so I gaffer-taped them to my head.

Some sports events had their own opening ceremonies. The one at my venue was frantic. While I was trying to mix an awards ceremony, the event organizers quickly taught me the opening-ceremonies sequence a few minutes before it started. We threw together a show combining three announcers, five cassettes of music, and a live trumpeter on a wireless mic.

At many sports venues, we either used the existing system or augmented it with our own. Occasionally, interfacing with an old system was a nightmare. Some of the old mixers had no phantom power which the mics required, so we added a disco mixer with phantom. Old unused speakers and inscrutable patch bays made it hard to trace some systems.

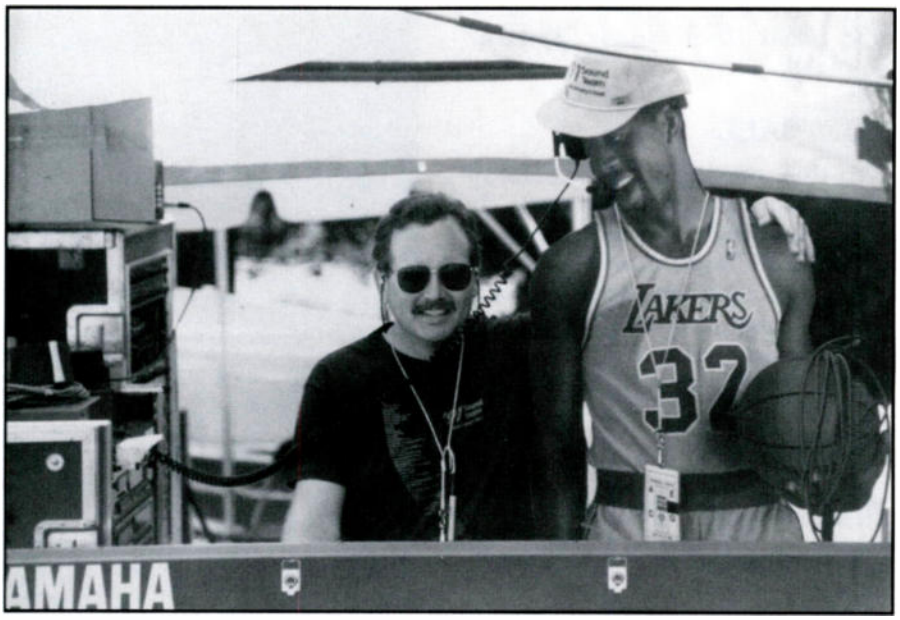


Figure 5. Two stars: Crown's Maurice Paulsen (left) and Magic Johnson during the house mix at the Yamaha stage.

Our trials were worth it. While operating the sound systems, we got to see the great performances and overwhelming joy of the special athletes, smiling jubilantly and slapping each other's hands when they won points.

NATIONAL SPORTS CENTER

This huge complex included fourteen football/soccer stadiums, each of which needed an announcing system. Running power in to the stadiums was a problem. Crown's electrician, Steve Myers, negotiated with the local power company to get one transformer moved from across the road and another one installed to accommodate the sound systems. In fact, several outdoor venues had no AC power, so the local power company installed poles and transformers.

Between every two stadiums we installed an amp/mixer rack and mic. A Klipsch KP-250 speaker fired across each stadium at the athletes and at the bleachers on the far side. I was impressed how loud

Figure 6. Speaker weatherproofing was a must outdoors.

and clear that single speaker sounded.

SOFTBALL

At this sporting complex, softball games were played simultaneously at six diamonds. The sound system used three Crown Macro-Tech 1200 amps (six channels) and two Bose speakers per diamond, mounted on the backstop. After the levels were set for each diamond locally, the amplifiers were monitored with the IQ System 2000.

RIVER FLATS PARK

Located on the banks of the Mississippi River, this park held Olympic Town and Expo Center, a group of tents and two stages which featured continuous entertainment all day long. One stage, supervised by John Schauer, a Yamaha audio engineer, used all Yamaha equipment. Poles twenty-to-thirty feet high were raised and power lines were strung there especially for Special Olympics.

We set up one concert-sound system there on very short notice. First, two scaffolds were assembled at either end of the stage and just in front of it. Under the scaffolds we placed plywood sheets to keep the subwoofers off the ground. The subs were four Community VB415s (two per side). On top of each scaffold stood two Community RS 880 speakers. All of these were powered by two Crown Macro-Tech 2400s per side.

In the audience area we erected a scaffold for the mixing platform. On top of the scaffold we placed three equipment racks, and on top of them went a Yamaha PM3000C mixing console. We placed a plastic tarp over the mixing position and over the speakers in case of rain.

Hung under the edge of the stage was a Whirlwind Medussa snake (thirty-two mic inputs, ten sends). Numbered cables were permanently plugged into the snake and were coiled near the stage edge, ready to use. As for mics, we employed Crown CM-200 cardioid condensers, Crown GLM-100 and GLM-200 mini condensers, and Whirlwind IMP2 Direct Boxes.

Figure 7. Sound system designer and technical supervisor Sam Helms (left) and Eighth Day Sound's Jack Boessneck at the right.

HUM

When the system was first powered up, it had hum. Our hum-fixing wizard, Ken Kuespert, traced the problem to an unbalanced cable in the system. Typical acts included Irish dancers, square dancers, rock bands, choirs, dancing girls and solo artists. There were only twenty minutes between acts, and ten minutes notice about the type of act. The sound crew asked the entertainers about their needs. Do they want an introduction? If they dance, do they want floor mics? Will they be using cassette tapes? Do they want stage monitors?

Audio mixer Bill Seton of William Seton and Associates in Philadelphia, PA, noted that one of the sound mixer's main jobs was to instill confidence in the performers that their sound would be excellent.

At this venue, the sweetest sound I heard was at the Yamaha stage. The band was Mary Jane Alm and Too True (from Minneapolis), expertly mixed by Pete "Herb" Butler, an independent recording engineer in Minneapolis. The sound was warm and wonderfully smooth, with no listening fatigue. "The sound is ninety-nine percent due to these musicians," said Butler. "They make my work easy. All you need are good musicians, good-sounding instruments and good mics."

The drummer for this group used brushes on most tunes and played on just two tom-toms of 8 in. and 14 in. in diameter. Yet his sound was so fantastic that one engineer thought it was sampled.

Loudspeakers at this site were Yamaha S1520S two-way systems, just four per side. These were used in conjunction with the Y20 Active Servo Processor which damps and protects the speakers, and provides fixed equalization. Also in use was the new Yamaha C20 System Controller, a digital signal processor which provides simultaneous delay, EQ and compression in separate frequency bands. I was impressed with its convenient memory presets, adjustable parameters and good sound.

In addition to the concert systems, we set up the Olympic Town Audio Control Center. This zone paging/emergency announcement system was used to communicate messages in several languages and announce the entertainment for the two stage areas.

Four Crown Com-Tech 1600s running 140-volt lines powered the paging system through eight thousand feet of speaker cable run all over the park—a constant-voltage distributed setup. Forty Atlas-Soundolier horns, which sounded impressively clean and natural, were mounted on the telephone poles. An IQ System 2000 monitored the amps in the system.

INTERNATIONAL DANCE SYSTEMS

Located at each college campus in the Twin Cities, each of these concert systems amplified a live band to provide dance music for the athletes and the public.

One dance had just begun when the light crew plugged into the power line and blew the breaker. Crown's Dr. Clay Barclay drove to the site in his tour bus, which contained a 20,000 watt generator wired with outlets for situations like this. Within ten minutes, all systems were running again.

INTERNATIONAL PROM AND PEP RALLY AT BOOM ISLAND PARK

This major event featured a 1950s/1960s concert outdoors, with acts such as the Shirelles, Bobby Vee and Peter Noone. The sound system provided by Eighth Day Sound was huge: sixty Turbosound TMS-3 four-way speakers and twelve Turbosound TMS-24 subwoofers driven by fifty-six Crown Macro-Tech 2400s. In addition, two Macro-Tech 10,000s drove eight Servodrive subs. On stage were eight custom monitor wedges.

This was only a medium-size system for Eighth Day Sound. A large system would be one hundred and twenty boxes, used in stadiums and large arenas.

For safety, both the house and monitor mixers were grounded locally to power ground. Still, there were no ground loops between them. Why? The interconnecting cables were balanced, and each shield was tied to pin 1 only at each output. According to Brnich, grounding the shield at the input picks up more interference. Brnich said they seldom have ground-loop problems unless they interface with someone else's system that has different grounding. Then, trial-and-error works out any hum problem.

Helms and Brnich mixed house while Denton mixed monitors. Brnich aimed for a balance of fifties and nineties sound production. He was careful not to add, say, a flanger to a fifties tune!

MONITOR MIXING

I asked Lennie Rosengard, Eighth Day's monitor technician, how he set up monitor mixes. He said that most musicians tell the monitor mixer what they want to hear. Almost all need to hear some kick and drums. Rhythm-section players like to hear each other to play in sync, while vocalists need to hear each other for harmony.

Rosengard gates the drum mics with a very short attack and decay. This prevents feedback because each mic is open only when the drum is hit. Since the gating eliminates the ring of the drum, some reverb is added to simulate the ring.

Denton, an independent audio engineer, worked with Eighth Day Sound by doing the monitor mix at the International Prom. He had these comments on monitor mixing and concert sound:

*The monitor mix is more important to the musicians than the house mix. Even if someone in the audience tells the band that the house sound was poor, they'll say, "Hey, it sounded great to us!"

*You eventually learn the sound of the monitor mix each musician wants. Instead of listening for a good balance of all the instruments, you listen for specific mixes of a few instruments.

*If you have only one channel of monitor mix, put in vocals and a little drums and kick. Adding more makes the sound muddy.

*The house mixer wants the monitors to be as quiet as possible for good sound, but the band wants the monitors as loud as possible so they can perform well. So the house and monitor mixers must work well together as a team.

*At one event we had to do stage changes between acts in three minutes! Take one act's equipment off the stage, put on the next act's equipment, mic it and set up a mix in three minutes. It was hard, but was great training. Really prepares you for a gig like this.

*What's needed most at concerts is good communication between all the people involved: audio, production, lighting, and so on. They must know each other's needs. Then the setup and the show will go much more smoothly.

*You're only as good as your last gig. Even if the job is a small one, you'll want to do your best because each show has your signature on it.

*Volunteering for the Special Olympics has been a great opportunity to learn more about my craft.

At the rehearsal of a local choral group, feedback and wind noise were severe problems.

So Helms DAT-recorded the choir with a Crown SASS-P stereo mic and played the recording back during the concert for the choir to sing to.

The sound from the Boom Island system was awesome, thanks in part to the subwoofers which delivered thunderous deep bass. All around the park, vocals and announcements were clearly understood.

Some twenty thousand people danced, clapped and sang along. Athletes from Barbados partied with kids from Wyoming, and folks from Indiana danced with people from Nepal. With the lights of the skyline in the distance, beautiful spotlights crisscrossed the sky. Fireworks over the Mississippi River added a knockout finale to the party.

CLOSING CEREMONIES

Held at St. Paul on the steps of the Capitol, this massive ceremony featured Helen Reddy, Lorna Luft and a twelve-hundred-voice choir with a full orchestra. The loudspeakers on the five-story scaffold threw sound over two thousand feet. According to Helms, one highlight was using the new Crown CM-200a mic for vocals.

It was an emotional time for us all, saying goodbye after the intense weeks of working together.

ABC Sports aired a special on the Olympics on August 15.

COMMENTS OVERHEARD DURING THE WEEK

On the Expo Entertainment Stage, audio engineer Read Wineland saw a person "signing" to the deaf the lyrics for a rock and roll show. He said, "Just wait till they get to the 'Ma ma do mow mow!'"

These translations of comments at the event were provided by Australian sound engineer David Moore:

An event coordinator says, "Sound system? Just a microphone or two and a couple of loudspeakers should do it." Translation: "We have the Chicago Symphony Orchestra backing the Grateful Dead live, at a place with no power, and it starts in three hours.

An event coordinator says, "This should be over by about 5 P.M." Translation: "Bring a sleeping bag."

A venue manager says, "Electricity? We've got plenty for any job." Translation: "We have a 25-amp supply shared between Sound, Lights, and three air conditioners."

An athlete says, "Thank you, I enjoyed that." Translation: "Thank you, I enjoyed that." And that, my friends, makes all of the above tolerable.

TEAMWORK DID IT

It was gratifying to experience the cooperation among all the workers. All week long, people solved problems for each other, and there was a great spirit of friendship and teamwork.

Wineland added, "Even with equipment missing or breaking down, somehow we managed to make it work. Makes you feel good at the end of the day."

Seeing the athletes' profound joy as they were cheered, and as they hugged their coaches, was what it was all about.

The sound team readily achieved their goal: to help Special Olympics athletes hear the results of their courage.

The Contributors

Much appreciated was the generosity of the companies who donated equipment for the Games. Any equipment that withstood continuous heat, volleyball smacks, dust and rain earned our greatest respect. The donors were:

Apple Computer—computers
AST Sound—power conditioners
Atlas/Soundolier—mic stands
Audio Technica U.S., Inc.—BBE Sound, Inc.—signal processors
Biamp Systems—mixers
Bose Corp.—speakers
Community Light and Sound—loudspeakers
Crown International—amplifiers and microphones
dbx, a division of AKG Acoustics, Inc.—compressors
Dobbs Stanford—blank tape for background-music recording
EAW (Eastern Acoustic Works)—loudspeakers
Eighth Day Sound—sound systems
Goldline/Loft—analyzers
GRP Records, Inc.—CDs for background music
Horizon Manufacturing—cable
International Wire and Cable—cable
Intersonics, Inc.—Servo subwoofers
JBL—loudspeakers
Klipsch and Associates, Inc.—loudspeakers
Koss—headphones
MAA (Middle Atlantic Products)—rack panels and accessories
Maglite—flashlights
Rane Corp.—signal processors
Rapco International Inc.—cables
SBP Industries—power distribution systems
Sescom, Inc.—signal processors
Shuford, Inc.—gaffer's tape
SoundTech—loudspeakers
Tascam—mixers and cassette recorders
Telex—microphones, portable P.A.s
TOA Electronics, Inc.—megaphones
Tripplite, division of Trippe Manufacturing Co.—power conditioning strips
Ultimate Support Systems, Inc.—loudspeaker stands
West Penn Wire Corp.—cable
Whirlwind—snakes, splitters, cable and direct boxes
White Instruments, a division of C. Van R., Inc.—equalizers
Yamaha (Pro Audio Division)—mixing consoles, signal processors and loudspeakers
Zoom Corp. of America—signal processors

We thank these people who contributed Community RS220 speakers: Bob Myers of Sound Com in Berea, OH; Franklin Krakowski of Sheer Sound in Lorain, OH; Jim Sweeny of Torrence Sound in Toledo, OH; and Bill Slater of WES Sound & Systems in Massillon, OH.

We also greatly appreciated the volunteers who donated their labor and skills for the project, and the Crown support staff who provided food, information, laundry service and driving.

ACKNOWLEDGEMENTS

The author thanks these people for providing information for this article: Bob Herrold, Libby Marshall, Gerry Barclay, Clay Barclay, Jo Shreiner, Don Peterson, Sam Helms, Kim Denton, John Schauer, Mark Brnich, Mark Drews, Mark Chapman, Lennie Rosengard, Bill Seton and Dave Moore.

Photos supplied by Libby Marshall, Bruce Bartlett, Frank Revfi and Read Wineland. db

DREW DANIELS

Some Recent Observations on Stereo

I have a complaint to make. You might say to yourself, "he hardly ever writes, and when he does, it's only to complain," but there's very little difference between my complaining and a newspaper reporter writing only bad news.

WELL, THE NEWS IS THAT while we were looking, equipment got better and music got worse! How did this happen? Why do I hear less and less music that turns me on these days? Admittedly, I don't have the same musical ear as the public because I'm a university-trained musician. I've been playing instruments since 1952 and recording people for pay since 1961, when I was fourteen.

My ears don't hear the same as the public when it comes to audio quality either, because I have a better playback system and more technical awareness than the average person as most of you **db Magazine** readers do. In fact, I taught loudspeaker technology and critical listening at the University of Southern California, so I know that not everyone "connects" to sound the same way. It's like tasting food—it never tastes the same to any two people.

Drew Daniels is currently the chairman of the AES Los Angeles Section and Principal Electroacoustic engineer, Walt Disney Imagineering. He was an applications engineer for JBL Professional for five years, Tascam for three years, and training manager for Fender Pro Audio. He taught Loudspeaker Technology and Critical Listening at the University of Southern California, has his own recording facilities, and was a singer and bass player in the New Christy Minstrels. Daniels also studied Opera and sang the National Anthem at a 1974 World Series game at Oakland.

We've had progress in the last decade alone that staggers the imagination. Unfortunately, it's mostly progress we've been unable to put to good use. As our machines have made us faster, they have not made us wiser.

We have digital audio workstations ranging from cards and software for your PC to half-million dollar dedicated super-systems like the Synclavier. There are sophisticated workstations such as the four-channel Adap-IV for around $20,000 and a Symetrix system for around $60,000. There are multi-track recorders of all types from a few hundred dollars to a few hundred thousand; the latest announcement is of a 64-track DASH using one-inch tape. We have vastly improved playback systems, with audio power amplifiers up to thousands of watts, and we have an incredible array of processing gear for our outboard racks, digital keyboard instruments, samplers, sequencers, and so on.

With all of this progress, however, we also have MBAs and lawyers running record companies, radio stations and MTV. We have fewer artists allowed fewer outlets of expression by record companies who are willing to risk fewer musical formats. Artists are forced to make records with fewer musicians and less studio time so they can rush back out on the road to pay off their music video advances.

The multi-million dollar studios are being sold off piece by piece as more home studios with self-taught engineers crank out foul-sounding grist for the record market mill, and, oh yes, there's the public. Let's not forget them! We, who do this music and engineering stuff, should remember that the public can't hear, and that our ears are over-trained and not at all objective. There's also the small matter of consumer playback systems. These systems used to be record consoles, hi-fi or stereo sets in the home, but they're now in the car or worn around the neck. The largest-selling audio medium is the Philips cassette, which was invented as a speech-only replacement for Dictaphone cylinders and never should have been anything else!

WHERE'S THE STEREO?

What really adds insult to the injury I feel when I momentarily tune across a pop station on the old Frequency Modulation receiver is the loss of stereo—yes, *stereo*! That technological marvel first introduced at the Paris exposition in 1890, where guests hearing the feed from the Paris Opera house on electrical headphones remarked how realistic it sounded!

I've found myself buying a lot of classical music lately. Sometimes Verve Records rereleases some of their wonderful classic 60s jazz such as Miles Davis' "Kind of Blue" or Ella Fitzgerald's "Johnny Mercer Songbook," and I run out and get a copy. The point, though, is that I've been buying fewer pop records, and it's not because I'm suffering from "old-fart" syndrome, but because pop records just don't sound good anymore. Besides being compressed and over-processed until they are as squashed as interstate road kill, pop records tend to have a flat, two-dimensional sound quality due to the lack of stereo in the mix. If you've ever sat in the sweet spot to listen to a Billy Joel record so you could hear how a master recording engineer mixes, you may have noticed that most of the sound images marvelously between your speakers and forms a phantom image in the space between. Well, I'm here to tell you that you've paid good money for stereo and got mostly mono!

WHAT'S HAPPENED TO THE RECORDING INDUSTRY?

You may hate me in the morning, but I think recording engineers have become gutless wonders. I think they're afraid of stereo, and I think they're caving in to the wishes of record producers and record companies.

Record producers are the "firemen" in the control room—you know, the guys who shovel the coal from the coal car into the boiler. Unions say you must have a fireman on every train, even though trains stopped shovelling coal years ago: it's a practice called "feather-bedding." Personally, I could always make a better record myself without the help of an extra "creative" person in the control room.

RABBITS AND SNAKES

A friend of mine told me this story after being burned in a Hollywood record-deal swindle about a certain type of record producer. To paraphrase: A blind snake bumps into a blind rabbit in the woods. After mutual apologies, they realize they are both blind and kindred spirits. The rabbit suggests "let's use our sense of touch to tell each other what we look like." The snake agrees, and begins rubbing the rabbit all over, exclaiming all the while, "um-hum, um-hum." The snake finally says, "well, you have smooth fur, a little round tail and long ears. You look like a rabbit." The rabbit then begins to rub the snake, but the snake is nervous and wants to know immediately what the rabbit thinks. He begs the rabbit to tell him what he feels. The rabbit is embarrassed, but honesty compels him to tell the poor snake, "well, you're slick and slimy and you have no ears at all, and ...." Suddenly the snake recoils and starts to cry, "oh my god, I look like a record producer."

Let me give you my little history of the decline and fall of stereo on pop recordings. It speaks eloquently for the abandonment of the current school of mixing.

The early 1960s were the glory days of stereo. In those days, soon after the mass-market introduction of stereo and X-Y lateral groove mastering, recording engineers used stereo to advantage, placing acoustic images at will. There flowered and bloomed records of all sorts made for demonstrating stereo. You might have picked up a copy of Bob Prescott's "Cartoons for Stereo" and heard Columbus sail off the edge of the world as the ocean drained from your left speaker out of your right speaker, or you might have heard "Russian Roulette," with the "loser" sliding the gun back to his opponent over a saloon bar top stretching between your speakers.

As time went on, record companies demanded quieter records—less record groove hiss in relation to the music—so they beat up on the record cutting technicians to cut hotter (greater groove amplitude) in the acetate masters.

The LP disc master cutting process produces purely lateral motion of the cutting stylus for sum signals (mono), forty-five degree lateral-vertical motion for pure left or right signals—which leaves the opposite groove wall unmodulated—and purely vertical motion of the stylus for difference signals. These two latter conditions are what constitute stereo, which, by my definition, is disparate points of view on a sound image from the two speakers.

CUTTING DISCS

Cutting engineers soon realized that in order to meet the demands of the client record companies, they would have to do a little summing of their own to prevent hot stereo (vertical component) signals from cutting "air or aluminum"—that is, to prevent the cutting stylus from moving upward, leaving the acetate and "pinching off" the grooves because it makes records skip when played, or even worse, digging down so far through the acetate lacquer that the delicate sapphire stylus would cut into the aluminum substrate and become gummed-up with aluminum, or even snap off.

Keep in mind that although cutting equipment has generally led phono cartridges in the ability to cut grooves of greater amplitude and higher velocities, phono cartridges were getting better and better all the while. Even so, the cutting process can lay down information in the grooves, too dynamic for retrieval by commercially available phono pickups.

Most disc mastering technicians started panning a lot of the existing stereo signals (which produce vertical motion) toward the center, making records more mono than stereo. By the late 1970s, this was standard practice. Of course, recording engineers soon caught on to the fact that if they used a lot of stereo in their record mix, a cutting technician would alter the mix and change the stereo balance.

Most "recording engineers" are more artist than engineer, so it's natural for them to regard the disc cutter's post-processing as "changing my mix." It's sort of the reverse of colorizing black and white movies. Over time, most recording engineers reacted by becoming conservative with stereo panning and placement, so as not to provoke the cutting houses into altering what was produced in the studio.

Well, that's a thumbnail history of the collapse of stereo (pardon the pun). As a result of all this, and in spite of the fact that today's CDs and cassettes can handle stereo (vertical and lateral are meaningless for these media), typical pop records are recorded in monaural with an occasional off-center accent instrument or background vocal. The only other stereo material present on the recording is reverberation from a reverb device. This is easily and convincingly demonstrated by passing a typical pop record signal through a simple sum-difference matrix and listening to only the difference between the left and right channels. All bass, kick drum and most voice and everything else simply disappears and only the reverb is left, clearly showing that there is no stereo on these so-called stereo recordings.

WHO IS GUILTY?

You may wonder whose records I'm criticizing this way, but rest assured I'm not talking about semi-pro garage productions. Even Grammy winners aren't immune. Names like Anita Baker, Carla Bonoff, The Doobie Brothers, Billy Joel, Lyle Lovett, Linda Ronstadt, Ricky Scaggs, Paul Simon and many others (you soft jazz guys are in there, too) are in this category. In fact, it isn't until you get to the end of the alphabet that stereo appears on records by Stevie Winwood (effects), Frank Zappa (now there's real stereo for you, just look at an oscilloscope) and ZZ-Top (honest sound from a no-nonsense group).

I went blissfully along for years, thinking only that I was losing my hearing, until I got hold of an M-S matrix decoder box from my friend Wes Dooley. The box has dual modes; a switch lets you choose M-S decode, so you can adjust the stereo width you get from an M-S (mid-side) microphone or M-S signal recording. In the case of recordings, you can adjust the stereo width after the fact. Or, you can choose the "sum-difference" switch position, allowing you, in effect, to adjust the stereo width of an already decoded stereo signal, such as the output of two mics, a stereo mic (or a 2-track recording like a record release). In this latter mode, I simply rotate the single knob slowly from the "sum" position through to the "difference" position and listeners can hear the recording virtually disappear!

Well there it is. We have amazing technological improvements today compared to 1960, but where has it all led us? A sizeable step back, in my opinion. I miss stereo. I want to hear some imagination from recording engineers, instead of the standard two-dimensional flat wash of dead sound on most pop records. You'll find that most pop material (I estimate about seventy-five percent) is really essentially monaural recording with stereo reverb.

One more amazing fact for you readers, so that you may not go gentle into that control room:

At the Audio Engineering Society convention in Los Angeles in 1988, I set up the sound demo booth for the largest studio monitor manufacturer and supplier to the majority of the world's recording studios. Due to a manufacturing error in an amp rack supplied by a vendor, the sound demo audio electronics were hard-wired in mono.

For three days, Hollywood's (and everywhere else's) top recording engineers brought their CDs to play on the demo sets of speakers. They wanted to compare what they heard in their studio control rooms to what they heard on the new models, and in most cases, on the same models in a different location.

You can guess what the result is, can't you? The reviews were all raves, everyone loved the speakers, and not one engineer ever noticed that the record they had slaved over was being played back in mono! This is terrible! We're supposed to be getting better and smarter, aren't we? db

1992 Editorial Calendar

JAN/FEB	The Sophisticated Electronic Cottage. Winter NAMM Show issue. • GUIDE: Speakers: Performance & Monitor.
MAR/APR	Broadcasting—Audio Production for Radio and TV NAB show issue. • GUIDE: Consoles and Mixers.
MAY/JUNE	Audio in Houses of Worship/Fixed Venue Sound Reinforcement NSCA show issue. • GUIDE: Power Amplifiers.
JULY/AUG	Live Sound—Touring and Stadiums. • GUIDE: Tape, Tape Recorders and Accessories, Microphones.
SEPT/OCT	The Recording Studio—Digital and Analog, Big and Small. AES in San Francisco Show issue. • GUIDE: Signal Processing Equipment, Part I, (delays, reverbs, crossovers, equalizers.)
NOV/DEC	*db Magazine is 25!* The World of Post-Production for Radio, TV and Film. SMPTE in Canada Show issue. • GUIDE: Signal Processing Equipment, Part II, (noise gates, noise reduction, limiters, compressors), Work Stations.

LARRY ZIDE

The Building of Studio 9

Everyone knows that Howard Schwartz is a major world-class recording / post production house located in New York City's historic Graybar building.

YOU GET TO HIM FROM A NUMBER of elegant entrances to the building, including one that connects directly to Grand Central Station. I was there to collect a group of pictures that I had asked him to take while his latest Suite, Studio 9, was being constructed. A group of these pictures follow—perhaps a classic group that also show why Howard Schwartz Recording is as renown as it is.

But who is Howard Schwartz the man? Certainly he is today well known in the industry. Manufacturers fall all over themselves to get their wares into his rooms.

It was music that got Howard, at the age of nine started. Fist it was the trumpet, but he soon switched to the bassoon, so that by thirteen, he was an acknowledged virtuoso. He enrolled, under a full scholarship, in the Eastman School, but left after his first year, enrolling in the pre-law program at the University, where he also majored in Entertainment.

Figure 1. Studio 9 completed.

To cover those college costs, he became a disc jockey at a local radio station. But then the U.S. Army called, drafting him, but recognizing his growing talents, assigned him to Germany and AFN (Armed Forces Network) where he spun discs for the next three years.

After discharge, Howard went to Los Angeles, where he became a recording engineer working for the legendary Mel Blanc. Then he went to work at Wally Heider Studios where he engineered for Crosby, Stills, Nash and Young, Jefferson Airplane, Wayne Newton, Elvis Presley, Perry Como, and many other major artists of the time.

In 1971, he moved to Toronto, Canada to the Production Department of CHUM Radio and the CHUM group of stations. A bit over a year later he was lured to New York as an engineer for National Recording Studios and later moved to 12 East Recording. After being turned down in 1975 for a raise, he left to form his own company, Howard M. Schwartz Recording—one studio—one engineer (himself), and one receptionist. After a year, there was enough work to construct a second room, and hire a second engineer.

Now there are nine rooms, all dedicated to post production for video/film. Business is booming and it's all commercial or post—no music. Howard states that he simply can't afford to record music anymore.

Studio 9 was constructed for engineer George Meyer. "We figured out how to make the most efficient use of the space available, and be able to accomplish the tasks George has." stated Howard, "George does major television shows and sports. We wanted to make a room that was very comfortable for the large number of people that show up for those sort of shows. We built the control room muchlike a class lecture room with three tiers of seating."

The room today has its SSL 4000 G console (with ScreenSound) close to the screen, offers D-2 1-inch, Beta, U-Matic, and time-code DAT. There's a Sony 3324 and Studer 820 24-track as well as a Studer 2-track. An Adams-Smith 6200 AV with machine control ties everything together.

John Storyk did the design work. Here is the progression of how it went together.

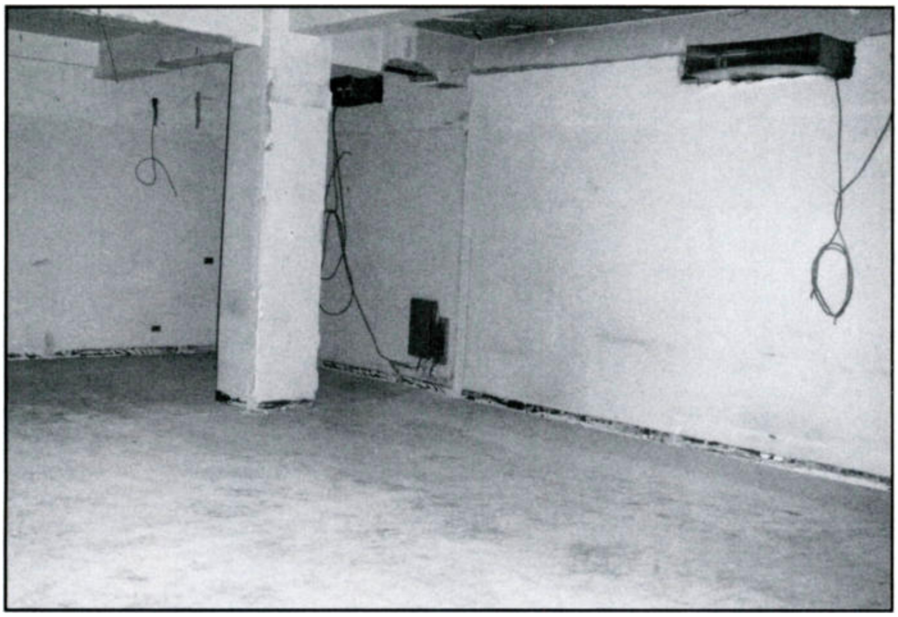
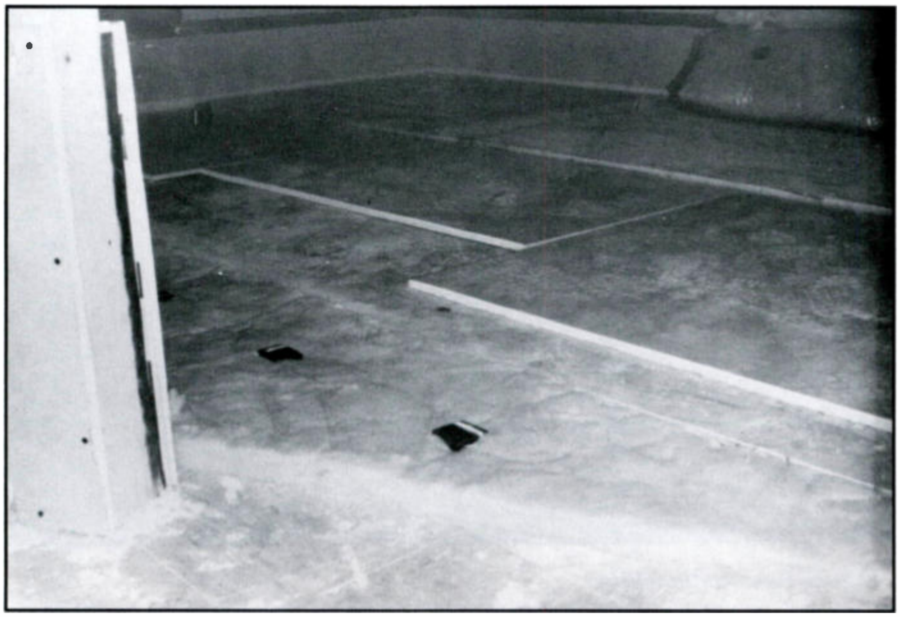


Figure 2. The bare space.

Figure 3. Makeup of the floating floors.



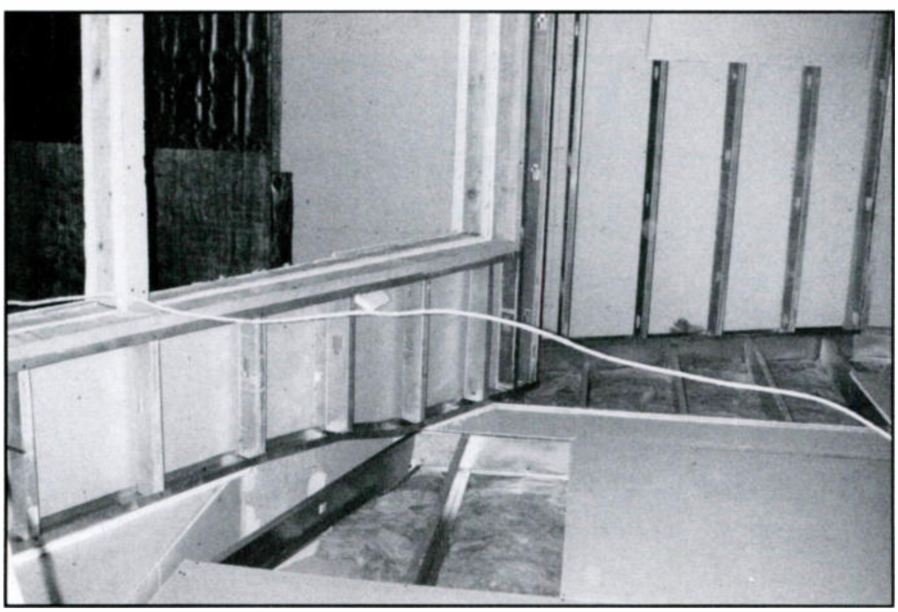


Figure 4. Detailing of floor and booth.

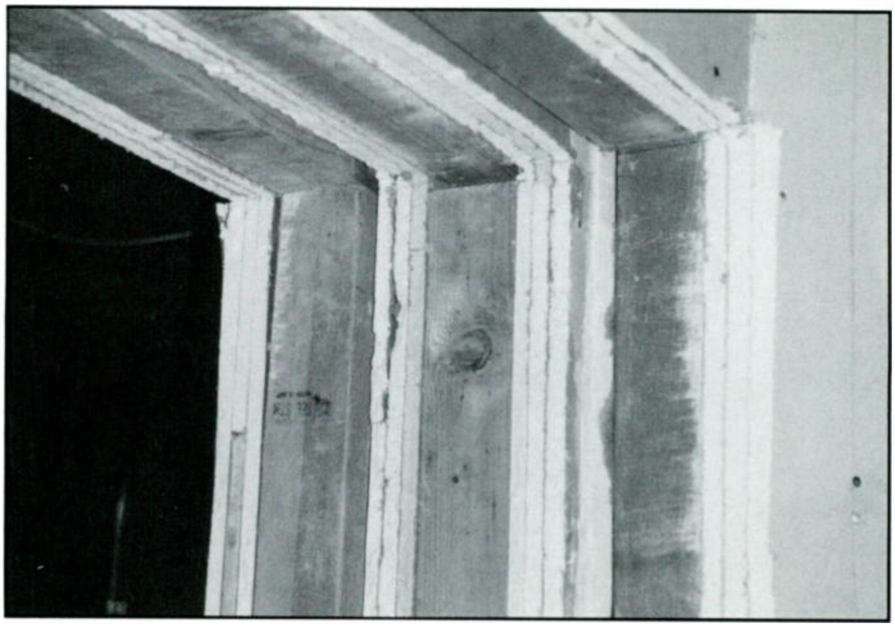


Figure 5. Detail of ceiling/wall corner.

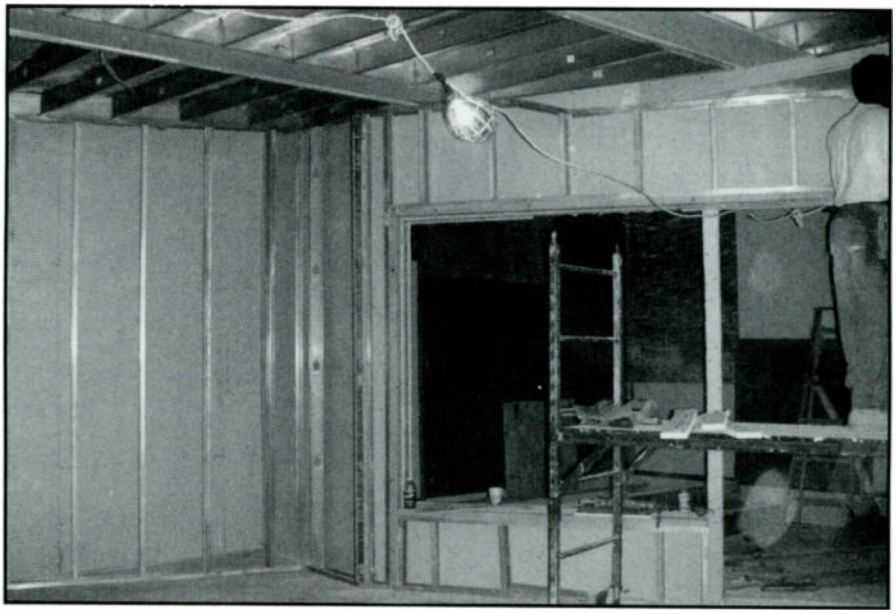
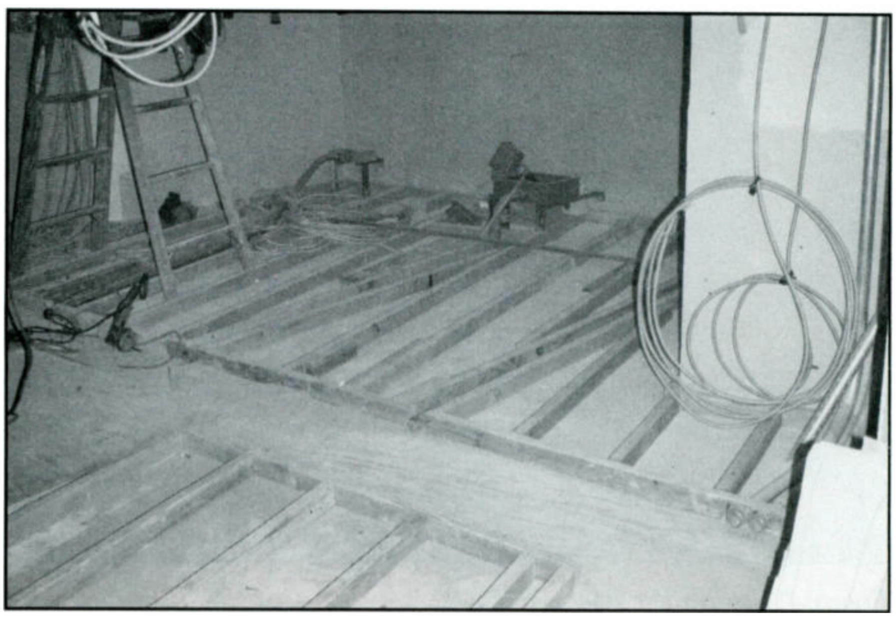


Figure 6.Acoustic walls.

Figure 7. Trothing of floor for signal flow.



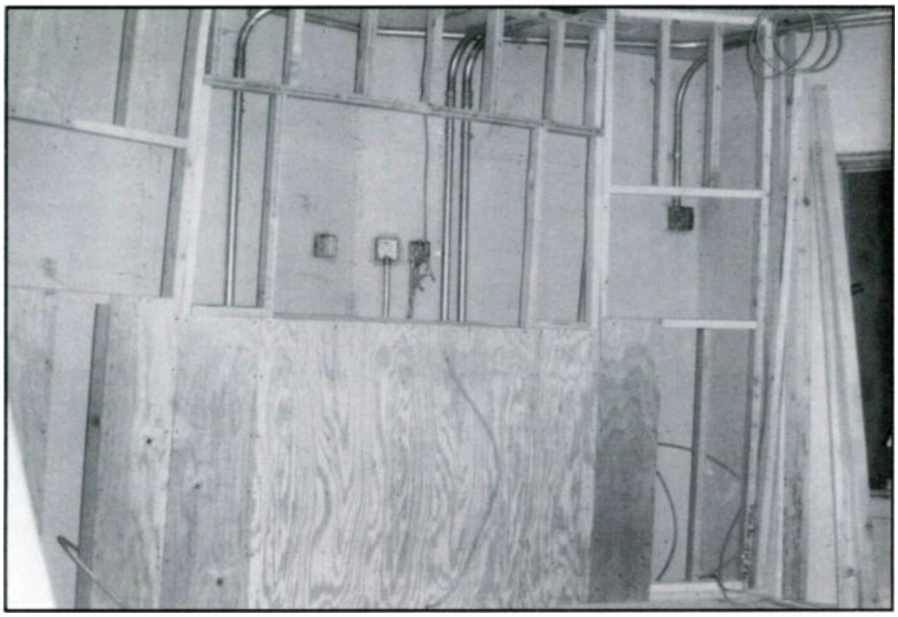


Figure 8. Front wall ready for finishing.

Figure 9. Ceiling detail and front wall are now further along.

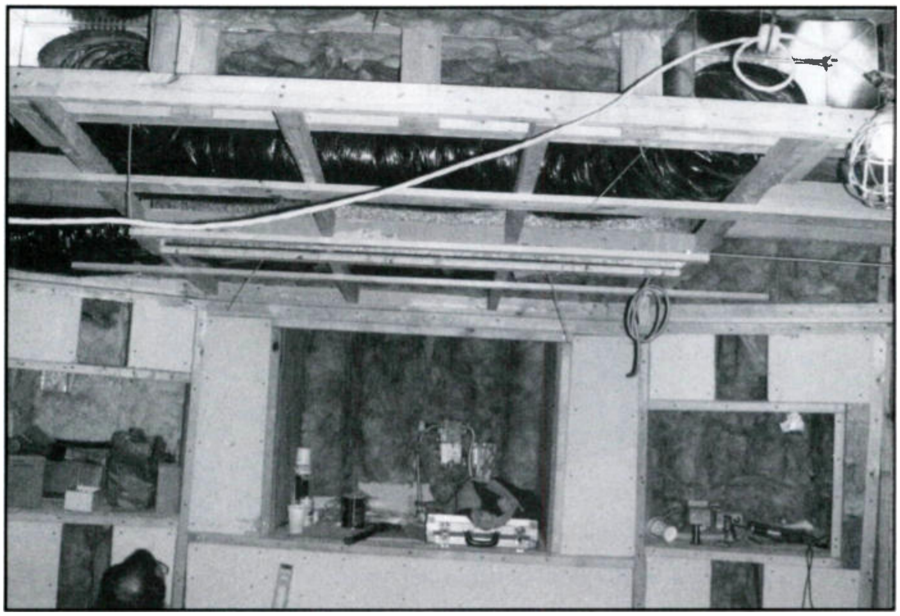
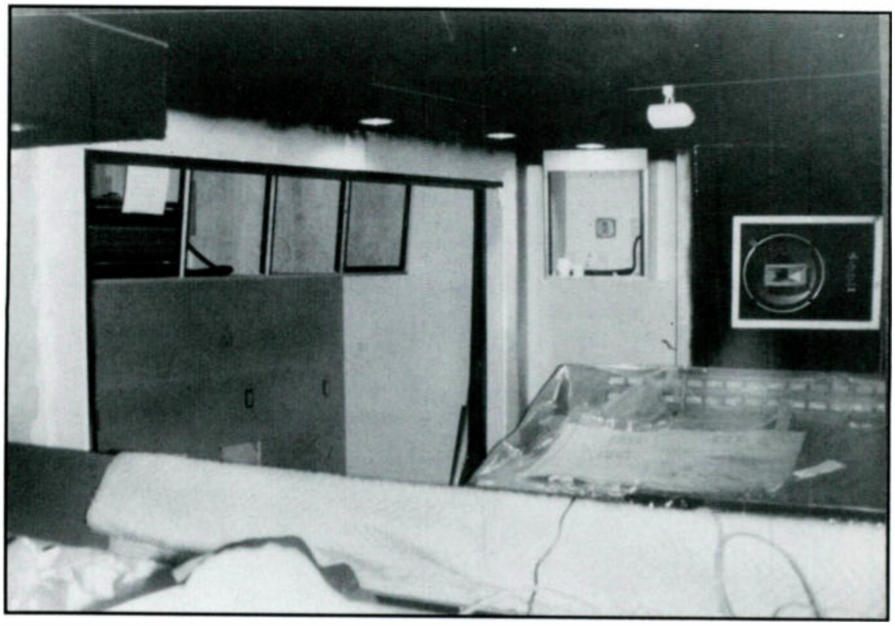


Figure 10. The acoustic ceiling nearly done.



UREI at Howard Schwartz Recording

Figure 11. Console and monitors are in.

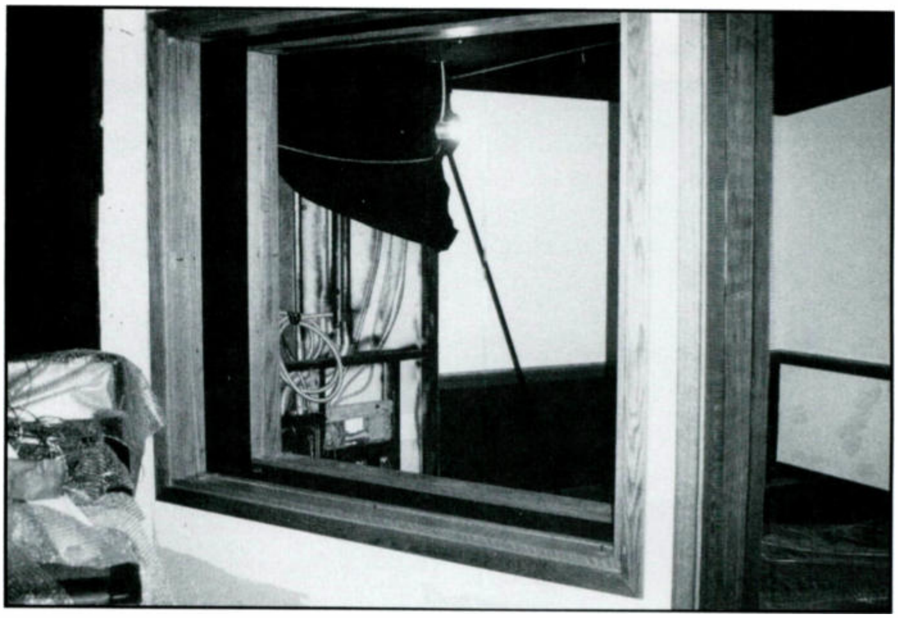


Figure 12. Detail of the nearly finished booth.

Figure 13. Master switching console.

db

BRENT HARSHBARGER

AUDIO FOR THE CHURCH

The Test and The Answers

● In the last issue, I presented a quiz, and here are the answers. Of course, there is no grading on how many questions you already know the answers to.

1. What is the purpose of an equalizer?

The purpose of an equalizer is to alter the audio signal, period. An equalizer is not capable of correcting acoustic problems.

2. What is the procedure for setting up the mixer's gain structure?

The procedure for setting up a mixing console is to put the faders at 0 (or unity gain) on the channels and the masters at 0 (unity gain) or -5 dB. Slowly bring up the gain control to the level where the program material of that particular channel is a strong, but not overpowering level, and after all levels are set, drop the masters back 5 dB from where you started.

3. Every time you double the amount of microphones that are turned on, you lose/gain ______ decibels before it feeds back.

You lose 3 dB of gain before feedback.

4. Every time you double the power, for example, when you go from 100 watts to 200 watts, you have a level change of ______ dB.

There is a 3 dB change in level.

5. How many decibels does it take to perceive double the volume?

It takes 10 dB to perceive double / half the sound pressure level.

6. Which will have a stronger level? A mic line or a quarter-inch line from a keyboard?

The keyboard will have a stronger signal level.

7. What device would you use to match a line, such as from a keyboard, to a mic level?

The device used to match a line or speaker level to a mic level signal is called a Direct Box.

8. What is the difference between a balanced line and unbalanced line?

A balanced line consists of two conductors and a shield, whereas an unbalanced line is a single conductor and a shield.

9. What is the ratio for the proper spacing between mics?

The proper spacing for mics is a 3:1 ratio, which means for every foot the mic is in distance for the sound source, the next closest mic has to be three times that distance, which in this case is three feet.

10. When wiring a mic, the hot pin is always on pin number _____.

Pin #2.

11. The shield is on what pin of an XLR?

Pin #1.

12. An echo send or effect auxiliary are usually pre- or post-fader?

The effect send is usually post so that any level adjustment made at the fader will make a proportional change in the effect.

13. A monitor send or monitor aux is usually pre- or post-fader?

The monitor send is usually pre, so therefore the monitor can be mixed independent of the house for proper queuing.

14. On a mixing console, the high and the low controls are what type of EQ filter?

The high and low filter controls are shelving-type filters. (Shelving is more common than it is actually a rule.)

15. On a mixing console the mids use what type of EQ filter?

The EQ filters on mid-band frequencies are Peak / Dip type.

16. An Omni-type mic has a 180 degree pickup pattern. (T or F)

False. An omni-type mic has a 360 degree pickup pattern.

17. A cardioid mic pattern is a good mic to use in a high decibel environment because______.

It's directional.

18. Which type of mic has more gain before feedback?

An omnidirectional mic has more gain before feedback.

19. If I have a level of 100 dB with one mic on, what will my level be with eight mics on?

The highest level of SPL with eight mics on would be 91 dB.

20. What is the international standard on the use of polarity on the XLR connectors?

The international standard is Pin #1 = Shield, Pin #2 = Hot, and Pin #3 = Return.

21. What is a good way to locate the primary source of hum in a system?

To locate hum, disconnect the audio cables starting at the mic input and work towards the power amplifier and speakers until the hum stops. The problem then is in the previous stage or the cable is improperly wired for this particular system.

22. When installing a reverb unit in a church sound system, should it be placed at the input of the main power amp or in a loop of the mixer?

The reverb unit should be in the loop of the mixer so the effects can be shut off for the announcer and turned on and easily adjusted for the musical performer.

23. What is constant Q in equalizers, and what is its advantage?

A constant-Q graphic equalizer is one whose bandwidth for a given frequency stays constant regardless of the amount of boost or cut. This type of equalizer gives a much more realistic representation of the fre-

quency response by looking at the sliders' positions than a conventional-type equalizer with sliders.

24. Where does hum come from?

Hum can come from many places, such as:

1- Ground loops (60 Hz currents in audio wires)

2- Insufficient shielding or grounding (electrostatic pickup)

3- Proximity to power supply transformers in other units (electromagnetic pickup).

25. What is the result when two channels are wired in opposite polarity?

Usually, the first noticeable result is lack of bass. Other results are localization problems and reduced intelligibility.

26. Define RT60.

RT60 is the amount of time it takes for a given power level of sound to decay to 1/1,000,000 of its original level, or 60 dB.

27. What is a preferred listening curve?

A preferred listening curve is a frequency response curve that makes a system sound most natural (usually with a 2 to 3 dB per octave roll-off above one to three kilohertz, and some bass attenuation to reduce room mode excitement, P-Pops, etc.).

28. What's the difference between a speaker's sensitivity and its efficiency?

Sensitivity is a reference level dependent on directivity, independent of efficiency. Efficiency is a percentage of electrical and acoustical power conversion independent of directivity.

29. Which produces less handling noise—an omni or cardioid mic, and why?

An omni mic produces less handling noise because a much less compliant diaphragm assembly is needed with a sealed back cavity that damps the diaphragm much better than the open cavity of a cardioid mic.

30. What is RFI, and what is one way to prevent it?

Radio Frequency Interference is prevented through: balanced lines, including transformers—an inductive coil in series with signal—a capacitor across the line—and the use of ferrite beads.

31. What is input overload, and what device is used to prevent it?

Input overload occurs when too much signal level is present at the input. The input's amplification provides so much gain that the level clips the output circuitry. Pads are used to reduce the signal at the input.

32. Do constant-voltage amplifiers put out a constant voltage, and under what circumstances can a direct-coupled amplifier drive a 70 volt line?

Constant voltage amplifiers put out a constant voltage only when a sine wave is being produced at the rated rms level. Under this condition, all amps are constant voltage. A direct-coupled amp that can produce 625 watts into an 8 ohm load can drive a 70 volt line. Also, output transformers and auto-transformers can provide a step-up voltage to smaller amplifiers' output voltages to drive a 70 volt line.

33. What is critical distance and how many values for it can exist in a room with a multiple-horn speaker cluster?

Critical distance is the distance from a speaker where the direct and reverberant levels are equal. There are as many values as there are listening positions. Every position in front of a speaker where the level from the speaker varies has a different critical distance. If this were not true, high-Q devices would not be of any use.

34. What is the standard rule-of-thumb for providing adequate damping to a woofer as it relates to cabling?

The cable resistance at a frequency to and from the speaker should be no more than ten percent of the speaker's impedance at that frequency for good damping.

35. What advantage does bi-amplification have over passive crossover networks?

Any clipping caused by the much greater power requirements of the bass section has upper harmonic components. Bi-amping prevents these harmonics from being reproduced in the treble section, thus reducing the audible distortion.

36. What is the primary reason for using High Q devices in a speaker cluster?

Uniformity of level within the coverage area; but there is not a great increase in intelligibility if a seating area requires a wide and tall coverage angle from a given cluster position. High-Q devices aimed at the rear wall give only a slight advantage in intelligibility. If this condition exists in a highly reverberant space, it's better to use delayed speakers.

37. When people in the first third of the audience from the stage complain that they can't understand what is being said, this is usually caused by what?

Echo.

38. In a fan-tailed hall, the reverberation tends to congregate in what part of the hall?

Reverberation tends to congregate in the back.

39. In a "shoe box" hall with raised seating, the reverberation tends to congregate in what part of the hall?

Reverberation tends to congregate in the front.

40. The condition of scattered sound is know as?

Reverberation.

41. The condition that describes the bouncing of sound is known as?

Reflection.

42. The scattering of sound is known as?

Diffusion.

43. The attenuation of sound (in the acoustic environment) is known as?

Absorption.

44. The bending of sound is known as?

Diffraction.

45. The bending of sound by density variations in the conducting is known as?

Refraction.

46. Room modes are known as?

Resonance.

47. Carpet on the floor of a hall will ___1___ the RT60 of the ___2___, but not the ___3___.

1=Reduce, 2=Highs, 3=Lows.

48. Define Echo.

Echo is a delayed return of sound that is perceived by the ear as a discrete sound image.

49. If a small room (1000 cu. ft.) has a RT60 of one second, and a large room (10,000 cu. ft.) has the same floor, wall and ceiling materials, the RT60 will be about?

3 seconds.

50. The proper way to set up a mix is to turn your gain all the way up with the channel slider on 0 dB (or at unity gain) and then slide your

masters up just before you get feedback. (T or F)

False.

51. Phase and polarity mean the same thing. (T or F)

False. Phase is frequency dependent whereas polarity is not.

52. When using a compressor, usually an ideal compression ratio is 2:1 for vocal work. (T or F)

True.

53. What is a compander?

A compander is a device that is both compressor and expander.

54. A noise gate is used in what situation?

A noise gate is used most commonly with the mic'ing of drum kits. With so many mics in a small area, they would violate the 3:1 rule and NOM. Use noise gates to turn on a mic channel only when a particular drum is hit, and immediately back off, based on the thresholds.

55. What are the five parameters for a good sound system design?

Coverage, Bandwidth, Level, Gain before Feedback and Intelligibility are the five parameters.

56. What is noise reduction and how does it work (very general answer is fine)?

Noise reduction systems can be described as a process or device that increases the signal-to-noise ratio. It works through an encode and decoding process.

57. Should you use noise reduction with desolver tones or timing codes?

No, because noise reduction systems work via an encoding and decoding process. The systems working in conjunction with each other can cause false cues.

58. What does the abbreviation MIDI mean?

Musical Instrument Digital Interface.

59. What is MIDI?

MIDI is a communication protocol.

I would like to thank some of the industry's leaders for taking time out of their busy schedules to contribute to the compilation of this test:

John A. Murray, Pro Sound manager of TOA Electronics

Art Noxon, president of Acoustic Sciences Corporation

Monty Ross, engineer, Rane Corporation

Larry King of Klepper, Marshall & King. db

LAB REPORT

LEN FELDMAN

ARX Systems' DI-6s Active Direct Box, Splitter and Line Mixer

GENERAL INFORMATION

● As ARX Systems explains it, most consumer-oriented audio products have unbalanced outputs, whereas balanced, low impedance is standard in pro audio. Tape decks, video audio, drum machines, synthesizers, guitars, computer; at some stage all of these need to be linked to balanced inputs, whether the application is live sound, studio or broadcast. To solve this incompatibility problem as well as for other often needed applications, ARX has come up with the DI-6s, a compact, rack-mountable unit that has six active direct injection (DI) channels plus a 1-into-6 splitter, a 6-into-1 line mixer, and a headphone amplifier.

The DI-6s allows six unbalanced audio sources to interface with balanced pro audio systems—either to individual channels or summed down to one master output. Alternatively, it can split a single incoming signal into six outputs. Each of these six outputs can be directed to a different destination and each of the six output levels can then be individually adjusted to suit the needs of the next component in its signal path. When used as a splitter in this manner, the overall master volume control still operates as well, raising or lowering the gain of all six channel outputs proportionately.

Each channel has a pair of phone jacks (one for input, the other for output), a level control for up to 15 dB of gain, plus a LED clipping indicator. The master section has a master volume control, a headphone jack and status LED indicators that show whether the unit is in its mixer/DI mode or in the splitter mode. Another feature that will, I think, be welcomed by anyone who has been plagued with hard-to-locate hum problems is each channel is equipped with a switch which, if depressed, "lifts" the earth/ground connection for that channel.

There are a couple of things the DI-6s will not do. Since it serves as a line mixer, it will not accept low-level microphone signals. For that application you will need a mic preamp or mixer ahead of the DI-6s. And, even though this would seem obvious to the professional, a special note in the ARX DI-6s owner's manual reminds us that it won't accept speaker lines either, so one should not plug the output of a power amplifier into any of its inputs!

ARX products, incidentally, are manufactured in Australia and are distributed by their United States division in Silverado, CA.

CONTROL LAYOUT

Each channel of the DI-6s is equipped with a front-panel input standard ¼ in. phone jack, front-panel output jack of the same type (both of these are unbalanced), a small gain control knob calibrated from infinite attenuation through 0 dB (unity gain), and up to +15 dB of gain and a LED clipping indicator. A numbered marker panel beneath these controls and jacks provides space for labeling DI assignments using a grease pencil or removable type. At the right end of the front panel is a standard ¼ in. headphone output jack, a master gain control and a pair of LEDs (one red, the other green) to indicate whether the unit is in the DI (mixer) mode or in its splitter mode.

Each channel on the rear panel is equipped with a balanced XLR output connector. Pin configuration is as follows: pin 3 is *hot*, pin 2 is *cold* and pin 1 is *ground*. A pushbutton switch associated with each channel is used to "lift" the channel away from ground and an indicator nearby shows the status of that switch. Viewed from the rear, at the left end of the rear panel, is an unbalanced master output in the form of a ¼ in. phone jack and a balanced master output in the form of an XLR connector with the same pin layout as that used for the channel direct outputs. Further to the left is the splitter input in the form of an XLR connector. Nearby is a pushbutton switch that selects either DI/mixer operation or splitter operation.

TEST RESULTS

With this type of product, there are not too many bench measurements that one can make in the lab. The major evaluations relate to the ease of use and the applications to which the product can be assigned. I'll have more to say about that later in this report. For the moment, however, let me review the few measure-

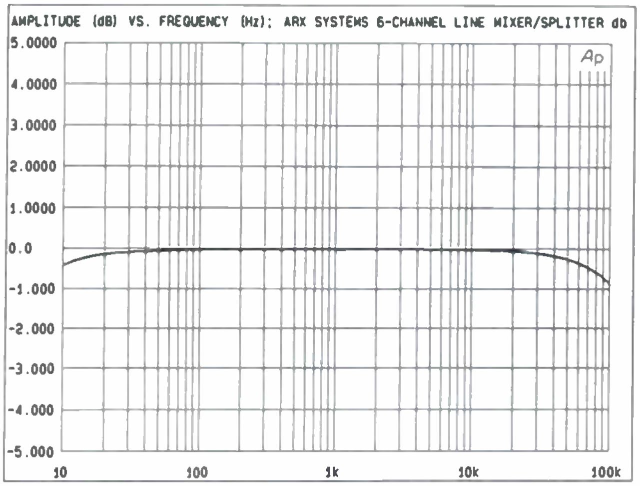
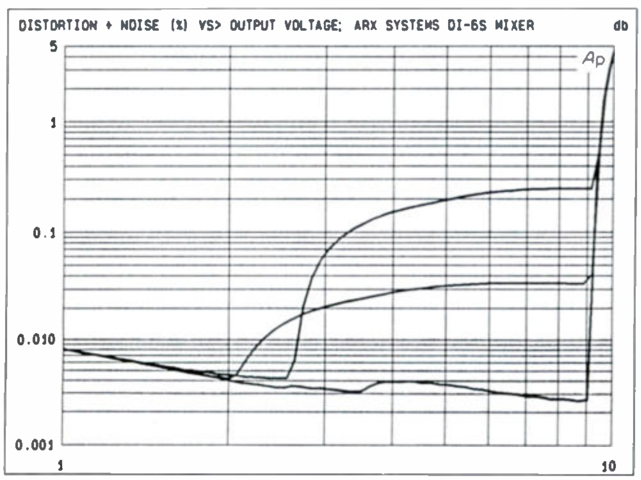


Figure 1. Frequency response at 0 VU signal level.

ments I did make. *Figure 1* shows the frequency response of the DI-6s with a signal applied to one of the input channels and the output measured at the summing output terminal. Response was virtually flat from 10 Hz to 100 kHz, with attenuation less than 1.0 dB at that top frequency. While this measurement was made with a 0 VU signal input and the channel and master gain controls set for unity gain, the tests were actually repeated at several gain settings of both controls in order to determine whether other settings of these potentiometers would affect response. They did not.

Further proof of the excellent frequency response characteristics of the DI-6s was obtained when I punched in the Splitter button on the rear panel of the unit and fed a signal into the splitter input. I then obtained outputs from each of the six channels, but adjusted the gain of each of those channels differently, based upon their calibration marks. Channel 1 was adjusted for maximum gain (+15 dB) and the response is depicted by the top trace of *Figure 2*. Successively numbered channels were adjusted for +5 dB, 0 dB

Figure 2. Frequency response in Splitter mode. A single signal was applied to the balanced splitter input, and each of the six channel outputs was adjusted to a different level.



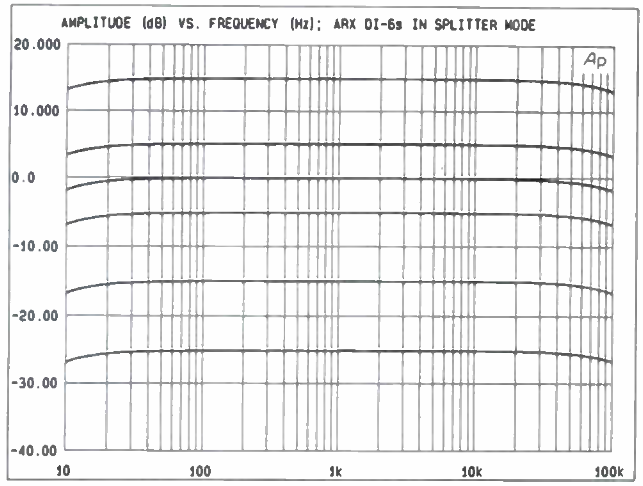
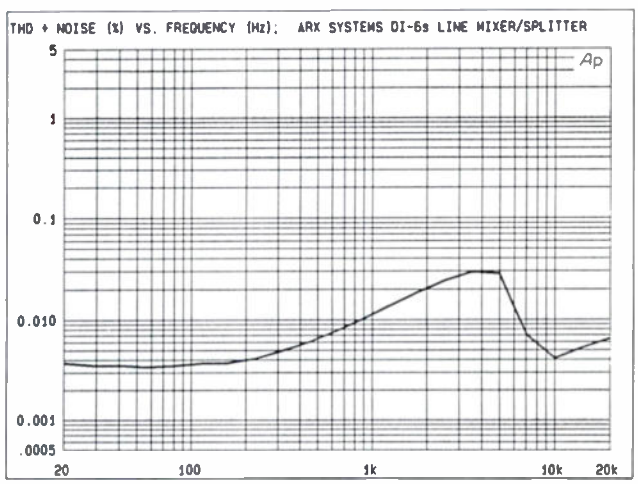


Figure 3. Harmonic distortion plus noise versus frequency for 0 VU signal in, maximum setting of master gain control.

(unity gain), –5 dB, –15 dB and –25 dB, and frequency sweeps were run for each of these outputs. As is evident from the results shown in *Figure 2*, there was no perceptible change in frequency response over that wide range of gain settings.

Next we measured total harmonic distortion plus noise as a function of frequency response, using an input signal of 0 VU, with level controls adjusted for unity gain at the mixed (summed) output. Results are shown in *Figure 3*. At 1 kHz, THD plus noise measured just over 0.01 percent. It was considerably lower at lower bass frequencies (0.0038 percent at 100 Hz), but tended to rise at higher frequencies, reaching a peak of 0.03 percent at around 4 kHz and then decreasing once again. *Figure 4* is a multiple plot of distortion versus output level for increasing levels of input, with channel and master gain adjusted to their maximum settings. This type of plot was repeated for three frequencies: 1 kHz, 20 Hz and 10 kHz. Clipping occurs at just over 9 volts rms output, equivalent to 21.4 dB

Figure 4. Harmonic distortion plus noise versus frequency for maximum setting of master volume control and input varied from 300 mV to 5 V. Best results for high levels at 20 Hz, next best at 1 kHz, poorest at 10 kHz.



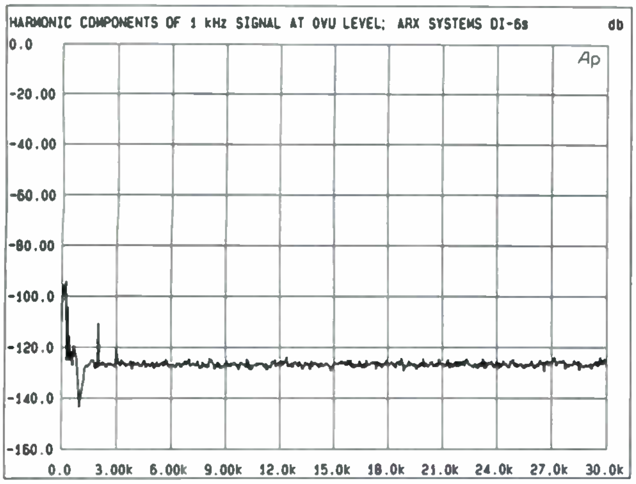


Figure 5. Spectrum analysis of harmonics of 1 kHz signal at 0 VU level. Averaged result of 16 acquisitions, to reduce displayed noise and improve display of coherent signals.

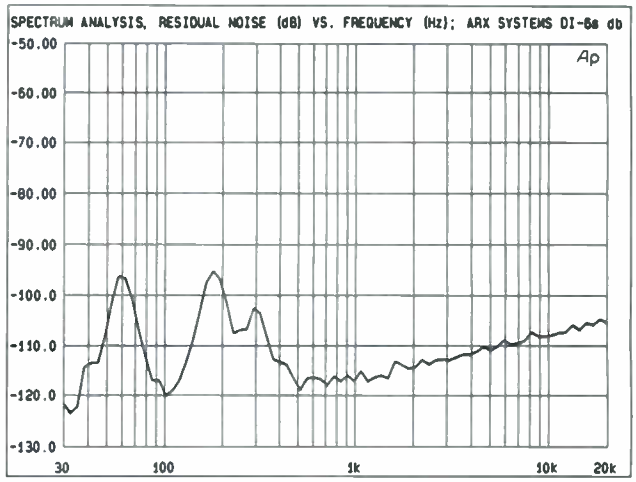


Figure 6. Spectrum analysis of residual noise. Input was 0.775 V and output was reference level.

above 0 VU. That's marginally better than the 21 dB claimed by ARX and certainly provides more than enough headroom for any application we can think of.

In order to separate the actual harmonic distortion components from the residual noise, we used the FFT function of our test equipment to run a spectrum analysis of the harmonic components generated when a 1 kHz signal was applied to a single channel and gain was adjusted to unity. The program acquires results over 16 sweeps in order to reduce displayed random noise while retaining a display of actual harmonic distortion. Results, shown in *Figure 5*, illustrate that what actual distortion existed at this level consisted primarily of second harmonic distortion, at a level some 110 dB below the reference output level. That works out to an equivalent distortion percentage of only 0.00032 percent; the rest of the earlier reading in *Figure 2* was the result of residual noise.

Besides its professional applications, I suggest that the DI-6s would be ideal as a level matching device for consumer audio products, home studio equipment and even audio for video in many applications.

As for A-weighted signal-to-noise ratio, it measured 98 dB referred to a 0 VU reference level. If you want to add the headroom to that figure, you come up with a dynamic range of about 119 dB. This measurement was made for the mixer function only. Since the headroom for an individual channel is somewhat higher than the headroom for the master section, dynamic range for that section would be even higher, or approximately 125 dB. A spectrum analysis of the residual noise of the system is plotted in *Figure 6*, and you will notice that the major contributions to the overall noise figure, however low, are related to the power line frequency (60 Hz) and its third and fifth harmonics (180 Hz and 300 Hz). At that, even these components are better than 95 dB below reference level. In this analysis, no weighting factor is used.

CONCLUSIONS

ARX Systems has managed to put many useful features and circuitry into a single height rack panel chassis measuring only 1-¾ in. high and 6 in. deep. Control functions are so obvious you'll probably be able to hook up the DI-6s without even having to refer to the six-page owner's manual supplied with the unit. It was a simple job to check all signals via the headphone jack before sending them on to other components or to my test equipment.

Besides its professional applications, I suggest that the DI-6s would be ideal as a level matching device for consumer audio products, home studio equipment and even audio for video in many applications. The unit delivers low distortion line levels of +8 dB or greater than are common to the radio and television broadcast industry, and it can do this in all three of its operating modes (direct injection, line mixer or splitter). If the audio industry were totally standardized in its levels and impedances, there might be little reason to own a unit such as the DI-6s.

However, in the real world of audio and audio-for-video as we know it, a lot of equipment that needs to be connected to professional audio systems is neither level nor impedance compatible. It is for those matching situations that this little multi-purpose unit really comes into its own and becomes worth much more than its modest price.

Vital Statistics

SPECIFICATIONS	MFR'S CLAIM	db MEASURED
Channels 1 to 6		
Frequency Response	N/A	10 Hz to 100 kHz, ±1 dB
Input Headroom	+21 dB	21.4 dB
Output Signal/Noise		
Unweighted	98 dB	N/A
A-weighted	104 dB	98 dB*
Output Level (Max.)	+26 dB	+21 dB
Dynamic Range	124 dB	119 dB*
Distortion (at unity gain)		
100 Hz	0.004%	0.0038%
1 kHz	0.003%	0.012%
10 kHz	0.005%	0.0042%
Master Section (inputs @ unity, Master at unity)		
Output Signal/Noise		
Unweighted	90 dB	92 dB
A-weighted	96 dB	98 dB
Max Output Level	+26 dB	Confirmed
Distortion @ 1 kHz	0.006%	0.004%
Headphone Jack Output	3W/8 ohms	Confirmed
Splitter Input		
Maximum Level	+20 dB	+21.5 dB
Frequency Response	20 Hz to 20 kHz, ±0.5 dB	+0, –0.9 dB
Additional Data		
Power Requirements	100/120V AC 50/60 Hz	Confirmed
Dimensions (W X H X D, inches)	19 X 1-¾ X 6	Confirmed
Weight	5 lbs (2.2 Kg)	Confirmed
Price	$549.00	

(* Measured with respect to 0 dB VU)

SHELLEY HERMAN

The Right Place At The Right Time

Bill Stancil, a pioneer in sound. And, among other things along the way: college dropout, theater usher, negative developer, portrait artist, animator, janitor, valet, personal secretary, dancer, film extra, field engineer, inventor, researcher, salesman, and once, strictly for fun, apprentice con man!

CON MAN? WELL, YES. STANCIL and his friend Ben Lyon liked flying, and would use any method they could to leave the ground. One evening Lyon told a dinner guest that Stancil was the son of a wealthy Detroit auto magnate and was interested in buying the guest's tapered wing WACO airplane. The guest took them down to Culver City Airport in Los Angeles and let them take a test flight. Three weeks and many test flights later, the plane's owner figured out that Stancil couldn't afford gas, much less the airplane, so he grounded them. Howard Hughes probably never fell for that scam again.

Stancil told me that story during an interview in December of 1990. I had approached him at a special Audio Engineering Society meeting at Disney studios with the idea of writing about his career. I remembered him from twenty-five years ago as a man who didn't suffer fools lightly, and was delighted to find him considerably mellowed, knowledgeable on many subjects with a great sense of humor and excellent recall. We shared a delightful day at his office and at his beautiful Newport Beach, CA, home.

Stancil summed up his career in one sentence: "I just happened to be in the right place at the right time."

Born in December of 1909 in Royal Oak, MI, Stancil's interest in electronics was sparked the year he was twelve when radio station WWJ went on the air. "Using oatmeal boxes, I would wind coils and make a variometer that I would sell for $10.00," Stancil said. "Eventually I helped build a radio station. I worked as an usher in a motion picture theater, and when sound came along, I just had to come to Hollywood, that's all there was to it. I got a ham license in May of 1925, and was active in what we now call electronics. I got out of high school in 1927 and went to Michigan State College for a year, studying art, and taking general courses for electrical engineering. In those days they didn't know what a milliamp was. It had to be volts or amperes, nothing smaller."

The most prominent citizen in the Detroit area at that time was Henry Ford. Mr. Stancil senior worked for the IRS, and Bill would hear of events involving Ford and met him on many occasions, including once with Charles Lindbergh. In 1929 Stancil worked for Eddie Stinson, aviator and aircraft builder. As Stancil set up a Stinson exhibit for a Detroit air show, he looked up and there was Ford all alone. Stancil had earlier visited the other booths, and met Dr. Ernst Alexanderson, inventor of the alternator. He was able to introduce Ford to Alexanderson, who had invented a device that was later used in the famed Ford Tri-Motor airplane. Once again, he was in the right place at the right time.

Stancil hitch-hiked from Michigan to Hollywood. "With the amount of money I had," he said, "I lasted for four or five months, then it was back to Royal Oak. Hollywood was a closed door. Back and forth I went until I finally got a job at MGM studios. I wound two hundred feet of film on racks, in the dark, then dipped them in the developing tanks. This was before processing machines. I advanced up to assistant negative timer on the night shift. In the daytime I was on the lot, so I haunted the sound corridor. I became friendly with Olin Dupy who got me into the sound department for $22.50 a week. After about a year and a half, a studio manager laid off everyone making less than $50.00 a week. I had done some timing for a cameraman named Gordon Avol who went to work for Jam Handy picture company in Detroit. He frantically called up and said, 'Please, please can you come back?' I said, 'I just got canned, I'll be there.'

Jamison (Jam) Handy married the daughter of Louis Chevrolet, so his company got all of the audio visual accounts for General Motors. Ben Lyon came through Detroit on a personal appearance tour and asked me to go on the road with him," Stancil said. "I told Jam Handy, 'I've got to get back to California, I can't stand these winters.' When I told them I was going back to California, the animation department people, who had known Walt Disney in Kansas City, got together and wrote a letter for me to hand-carry to Walt. I met Walt and he said, 'I understand you like art, I'll make an animator out of you.' I said, 'I don't think so, but I'll try.' I was there six weeks trying to be an animator, but it didn't work."

EARLY HOLLYWOOD YEARS

During his early years in Hollywood, Stancil lived with Bebe Daniels and Lyon, major movie stars of the 1920s and 1930s. Lee de Forest was Bebe Daniels' cousin and a frequent house guest at the Lyons/Daniels home. Over dinner, de Forest would speak of his various patents, and he later helped Stancil with multi-channel recording.

During the heyday of the MGM musicals, the big name in film sound was Douglas Shearer (1899-1971), winner of twelve Oscars. His sister, actress Norma Shearer, was married to Irving Thalberg, the wizard of MGM. When Stancil was working in the MGM sound department, Shearer was just another employee there. Stancil also worked as an extra for five dollars a day. He, like many others including my own mother, would see themselves billed as "...AND A CAST OF THOUSANDS!" Stancil was a charter member of IATSE sound local 659.

In his spare time Stancil was a portrait artist, and between takes, he drew a portrait of Marion Tally, an opera star appearing in a picture on which Stancil was working. The portrait was picked up by the studio publicity department, hit the wire services, and a photo of Stancil drawing Talley appeared in newspapers around the country.

In the early 1930s, Stancil lived rent free at a hotel in Beverly Hills by working as janitor. A fellow living there was valet to H.B. Warner (1876-1958), the actor who played Christ in Cecil B. DeMille's *King of Kings*. The valet, who planned to leave Warner and work in the stock market, asked if Stancil was interested in the valet job. Was he interested? You bet! Stancil got the job and went to Woodbury College at night to learn shorthand. He worked for Warner for two years until meeting author Don Marquis, famous for "archie and mehitabel." Marquis was writing a play and a book, and Stancil became his secretary. They went to lunch one day

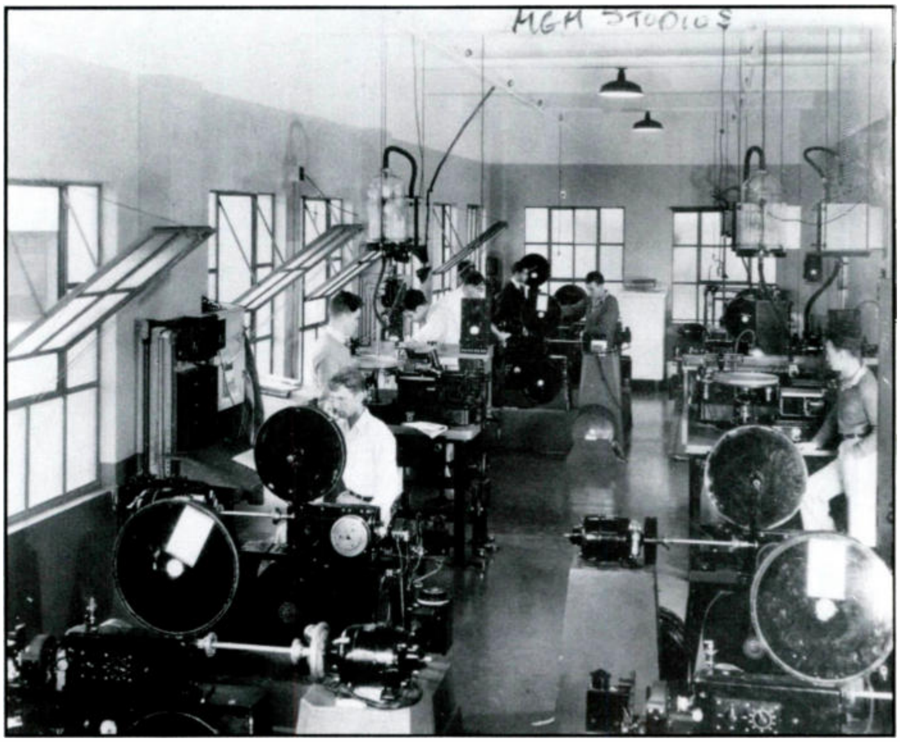


Figure1. An MGM sound recording room of the '30s. Note the film and disc recorders and also note Bill Stancil at the arrow.

with producer Kenneth McGowan (1888-1963), and Stancil did a portrait of McGowan. As payment, Stancil was given a contract as a dancer in the RKO picture *Becky Sharp* (1935), the first-ever technicolor feature.

"It didn't matter that I didn't dance; I wore a costume and moved around a little bit," Stancil said. "We just got started on the picture when the director died. They brought in Rouben Mamoulian as the new director, but we were laid

Figure 2. Bill Stancil drawing a portrait of opera star Marion Tally between takes at Republic Pictures.

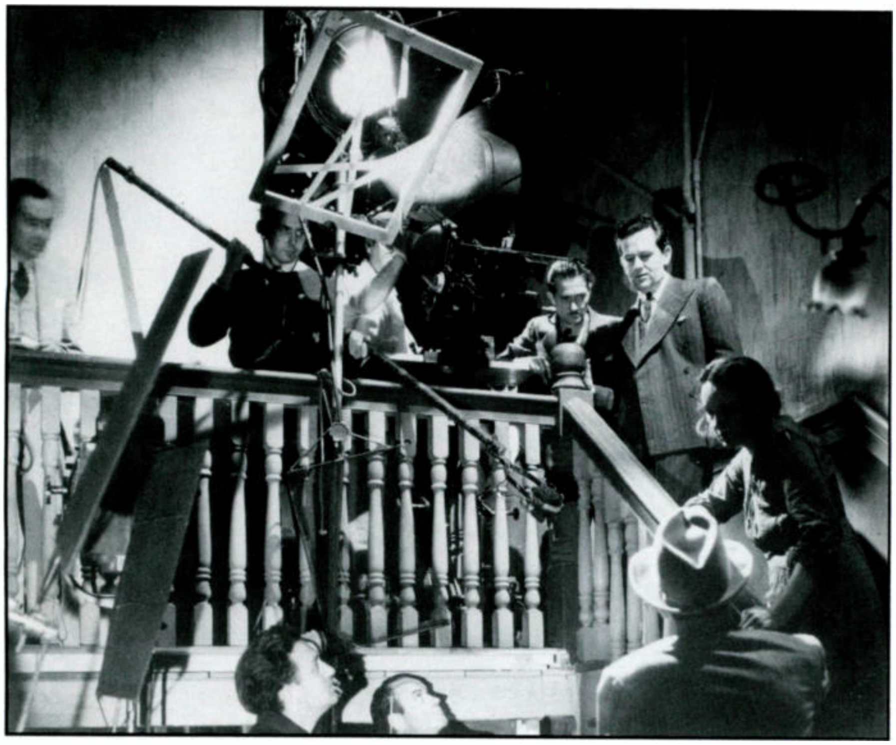


Figure 3. On the set of the 1937 Sam Goldwyn picture ***Dead End.*** *Bill Stancil holds the fishpole mic; William Wyler, director and Greg Toland, cameraman are facing the camera; Marjorie Main is on the stairs and Humphrey Bogart (with hat) has his back to the camera. Others in this historic photo are not identified.*

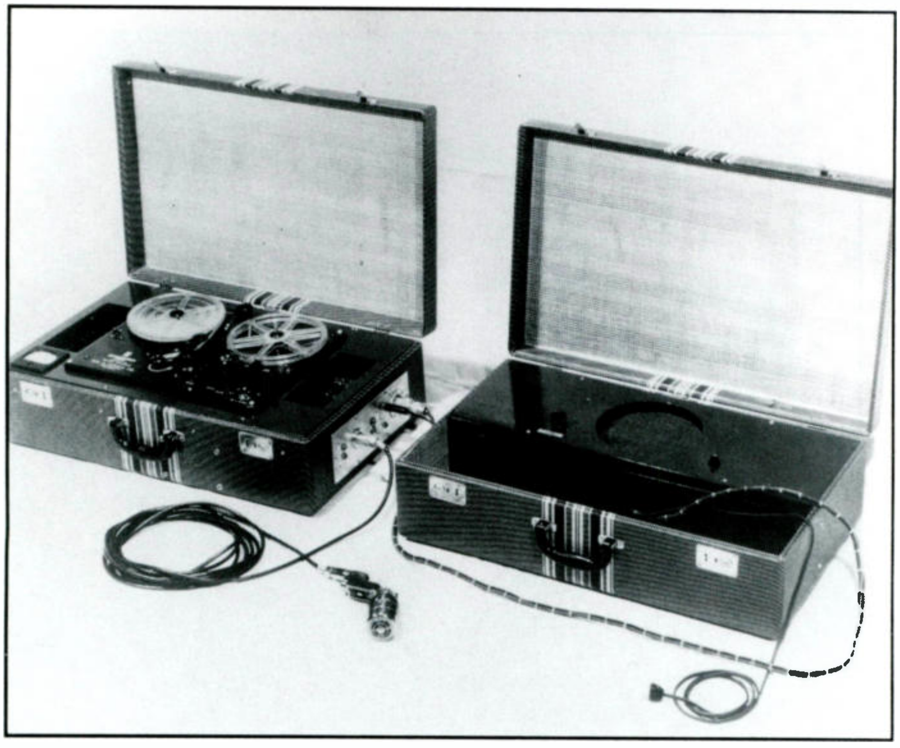
off on half pay for six months. Dollars from heaven." When Stancil saw the picture, he found that all his dance scenes ended up on the cutting room floor. Stancil's dancing partner once said, "You see that flyman up there? That's my father. And the man over there, that's his best friend Argyle Nelson, the assistant director."

Nelson got Stancil a job with the sound department at RKO. One day, a call came from Goldwyn Studio asking for an extra sound man for *Gone With The Wind.* "I went to Goldwyn Studios to sign in. As I entered, the receptionist said, 'The door's over there, when you go out, close it, please.' She practically threw me out of the office," Stancil recalled. "I was engaged to her six weeks later. We were married for fifty years."

Stancil worked for RKO, MGM, Goldwyn, 20th Century Fox, Mascot and nearly everybody else in the picture business. At some places, like MGM, he just free-lanced from picture to picture. I asked Stancil to name some of the well-known pictures he worked on besides *Gone With The Wind.*

"I got stuck with Willie (William) Wyler who did *Roman Holiday*, but he'd take every scene a hundred times. While I was at Goldwyn I did *Stella Dallas* with Barbara Stanwyk. I worked with Nat Levine of Mascot—he's the one who discovered John Wayne—as a projectionist, an assistant editor, whatever had to be done. It was a little family organization operating on a shoestring. Because I ran the dailies for Nat, when Sol Siegel was brought out from New York as a producer, I ran the film for Sol at night," Stancil said. "I did everything in the production of pictures, sweeping up, sound, and when there wasn't any sound, I'd do anything else. Of course unions weren't very strong then—you can't do that any more. I worked on some pictures with Tim McCoy, a famous silent star, from the 'one week' serials up to the major productions." Mascot eventually became Republic Studios.

Figure 4. The Stancil modified Brush Soundmirror—the first professional tape recorder made in the United States.



THE BRUSH SOUNDMIRROR

In 1938 Stancil gave a paper at a SMPE (before T) meeting at the Hollywood Roosevelt Hotel. The subject was a phonograph pickup using a D'Arsonval movement, made by George Downs, chief engineer of Lansing Speakers.

"At the SMPE meeting I met Dr. Joe Begun of Brush Development Co. I told Begun, 'Look, I've got the speakers here and I'll set it up so we can demonstrate your wire Soundmirror.' The wire Soundmirror stood about three feet high and the wire went sixty feet a minute. I didn't do Joe a favor by using good speakers, because it just made the Soundmirror sound worse."

HEWLETT PACKARD

Disney Studios hired J.N.A. (Johnny) Hawkins to design the multiple speaker installations for *Fantasia*. Hawkins, of course, involved Lansing and Downs. Simultaneously, Dr. F.E. Terman of Stanford University, author of "The Radio Handbook," put some students on special projects, one project being an audio oscillator. Downs went to a Stanford area IRE (now IEEE) meeting, where one of those students described his new oscillator. Downs saw and heard it, and met the student, Bill Hewlett. Upon his return, Downs went straight to Stancil, pulled out the schematic and said, "We've got to get our hands on this thing."

"We started discussing where it could be applied," said Stancil, "and George said, 'Johnny Hawkins, of course. He'll need it because he's going to have multiple tracks on *Fantasia*.' Stokowski was dividing the orchestra into nine sections and they wanted to have nine optical recorders on it. That meant nine oscillators. When we talked to Johnny on the telephone, he said, 'If this thing works as well as George says it does, I know Walt will let me buy them. Stancil, get your order pad and come out here.' I demonstrated an oscillator at Disney, not even knowing whether Hewlett had them in production yet. (He didn't.) They sold for $78.50. We had Dave Packard and Bill Hewlett (they hadn't yet decided whether to call it Packard Hewlett or Hewlett Packard) come down here. I rounded up all my friends in broadcast and motion pictures, hired a hall, and showed them the new oscillator. I sold one of the first oscillators to KNX radio (where) Les Bowman was chief engineer. I sold one to KFI, and of course, Dupy had Shearer buy one at MGM. Goldwyn studios bought one, (and) all the places I had worked, I was able to sell them." These sales persuaded Hewlett and Packard to go into business.

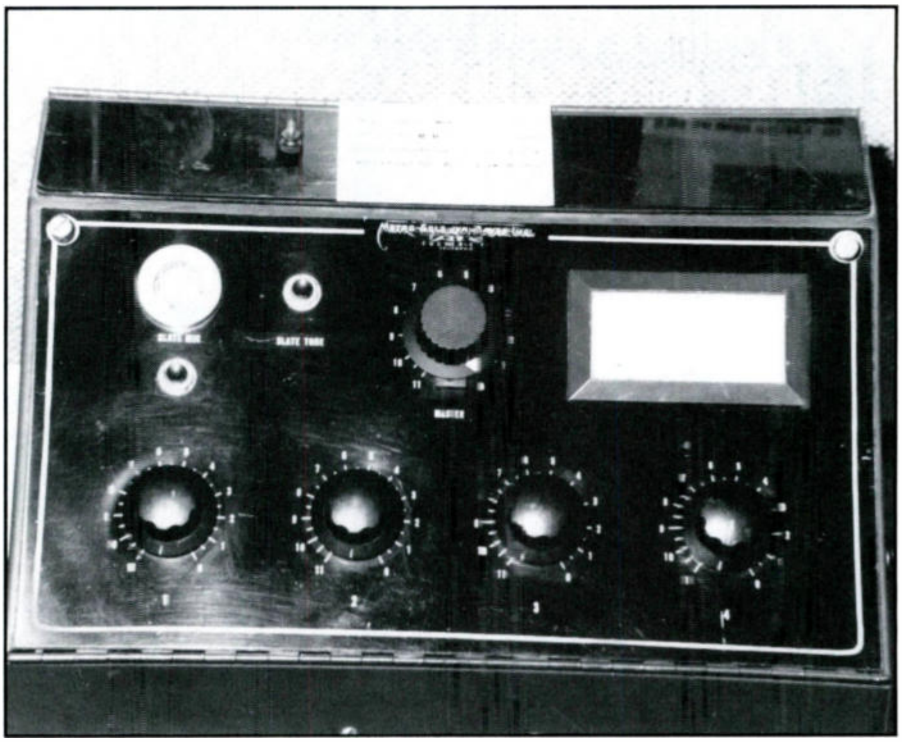


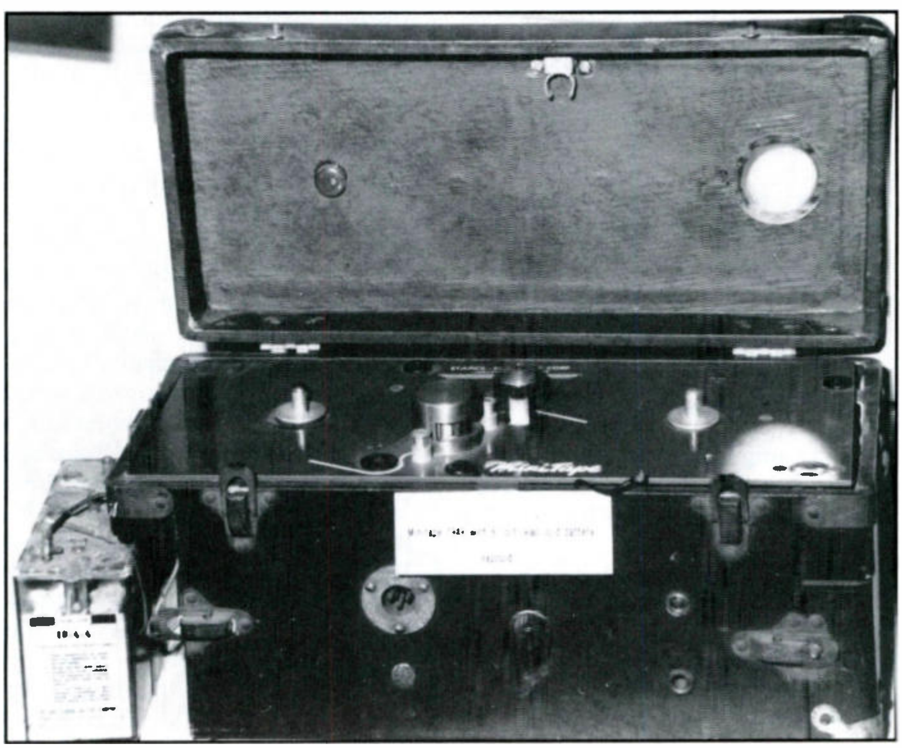
Figure 5. This is the mixer built for MGM Studios. It won a 1955 Academy Award.

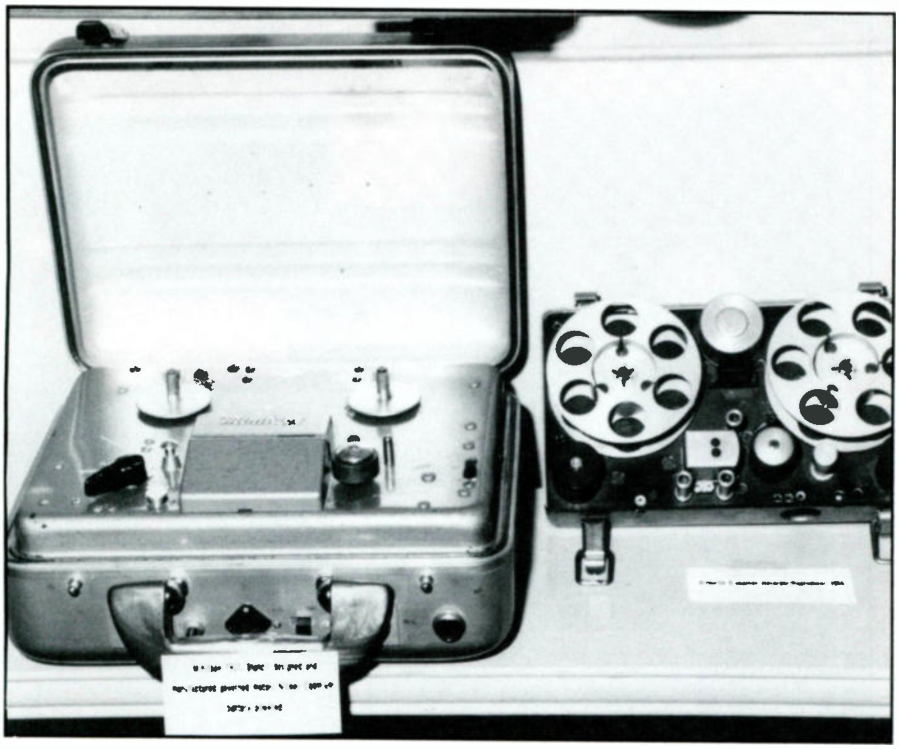
Although Hewlett Packard Corp. gave much of the credit to Stancil, Stancil felt that Downs deserved most of the recognition for the event that sparked the founding of Hewlett-Packard.

THE ATOM BOMB

In the early 1940s Stancil was a field engineer for Motorola. During World War II, while still on Motorola's payroll, he became involved with the Manhattan Project. His job was to design an absolute al-

Figure 6. The original 1949 Minitape. Note the storage battery at left.





Fiigure 7. The 1957 Minitape in an aluminum case and a 1955 8-channel flight recorder.

timeter, adjusted so that when the bomb dropped, it would explode in midair where it would do the most damage.

After the war, Dave (David O.) Selznick called Stancil for advice on the film *Duel In The Sun*. A train had to be seen coming around a mountain and crossing a trestle where it would be blown up.

There could be only one take, but the cameraman couldn't see the train coming. Stancil used two-way radios for the first time, allowing the cinematographer to remotely direct the camera shot.

When magnetic recording was new, there was a nasty patent pool fight. RCA was the policing agency, and Western Electric/Bell Labs was one of the licensors. When Brush announced the new tape Soundmirror, Stancil flew back to Cleveland and said,

"I'm going to take that Soundmirror and make a HiFi (professional) unit out of it, if you'll let me," he said.

"I'll be clear of the patent situation because you are licensed." He then repackaged the Soundmirror in two suitcases and sold it to radio stations and recording studios.

STUDEBAKER

One of Stancil's early customers was Hallock Hoffman, an audio buff and son of Paul G. Hoffman, president of the Studebaker Automobile Co. The Hoffmans, excited about this new technology, asked if they could invest in Stancil's fledgling enterprise. Still operating out of his garage, Stancil jumped at the chance and the company became Stancil-Hoffman. Stancil bought out the Hoffman interest a few years later, but retained the name until 1981.

With the Stancil/Soundmirror in production, Stancil started on the Minitape, the first battery-operated recorder. It used a Willard storage battery as there were no Gel Cells in those days. The first Minitape went to radio station KTMS in Santa Barbara in 1948. Stancil did some interviews with the Minitape recorder when he visited Lyon at the British Broadcasting Corporation in London in 1958. Lyon was so impressed with the possibilities of the Minitape that he talked the BBC into setting up a show called "Hollywood Go Round." The "BeeB" wouldn't permit third party recording from telephone lines, so Stancil ingenuity was used: using the Minitape, Stancil would record the celebrities on location in Hollywood. Lyon would ask questions by transoceanic telephone and the interviewee would respond. Stancil would then send the tape to England and Lyon's questions would be edited in. It was clumsy, but it complied with the BBC restrictions.

The Minitape was dramatized in a book called "The Great Man" which Universal Studios made into a motion picture. CBS bought Walter Cronkite a Minitape and when he recorded Prince Ranier in Monaco, the Prince was enthralled with it. Cronkite called Stancil and said, "I'll loan or sell him my Minitape, providing you can provide me another one within a week." Even President Eisenhower owned a Minitape. His vice president surely must have also been aware of the tape recorder. The Japanese made a windup imitation, and the British copied it even to the point of putting the same little window over the reels. About fifteen hundred Minitapes were made.

About the same time, Stancil was developing a multi-channel data recorder. There were no standard speeds or widths then, so he standardized on fifteen tracks on seven-tenths-inch wide tape. To do that, he used three five-track interlace recording heads and a single channel playback head that moved up and down. Stancil had a patent on the multichannel interlace heads based on soldering the lamination pieces to a brass bar. When the head wore out, the old lamination assembly was removed and a new one inserted, using the old one as a gauge. He took the recorders to Eniwitok Atoll for the Hydrogen Bomb tests where they were used to record all voice communications.

THE FIRST PRO AUDIO TAPE RECORDER

The first professional audio recorder after the modified Brush unit was the TR47 (Transcription/Recorder 1947). It was a quarter-inch professional recorder for music and speech broadcasting. The original speed was eighteen inches per second (90 fpm). Howard Chinn, chief engineer at CBS, declared, "If you can do this at 18 ips, you can do the same thing at 15 in./sec." Stancil challenged that, but eventually changed over to 15 in./sec. Having nothing on which to base his design, he had to build from scratch: heads, motors and electronics. The first TR47s used a four pole synchronous motor, with a beryllium capstan ground down to a little over an eighth inch in diameter to get 15 in./sec. Stancil started making his own two speed motors to get 7 ½ ips and 15 in./sec. Soon he was producing up to four or five thousand motors a month. He even sold motors to General Electric, one of the largest motor manufacturers in the world.

THE AMPEX 200

By the time the Ampex Model 200 tape recorder came out in 1947, Stancil-Hoffman was already in production making magnetic recorders. I asked Stancil if he had dealt with Bing Crosby.

"Before tape, Bing Crosby used an RCA film truck to record his radio shows because they could edit the film. For Crosby, editing was an absolute necessity! The show was recorded on Photophone film and edited, but the delay of the film processing made film recording unworkable. Crosby's people called me in and said that Bing's contract with ABC was predicated on the fact that they needed twenty-five tape machines. I explained that I was working in my garage. I said, 'If there is any way I can work with you and help you, I'll do it.' Crosby's man said, 'Can you handle service or something?' I declined because I was working on a tape recorder of my own. They had Alex M. Poniatoff (the founder of Ampex) come down to see me. He told me that Crosby put enough money up front to do twenty-four or twenty-five machines. I didn't have the facilities and wasn't in a position to do it. We were developing multichannel recorders, we were building the Minitape and developing the film recorders, so I stayed out of that picture.

"At an Academy of Motion Picture Arts and Sciences meeting, Jack Mullin had come down from San Francisco to discuss his (Magnetophon) recorder," continued Stancil. "Of course, Dr. Honen and Doc Frayne just pooh-pooh'd it on the basis of no sprocket holes, you can't meter it, etc.—they were against it completely."

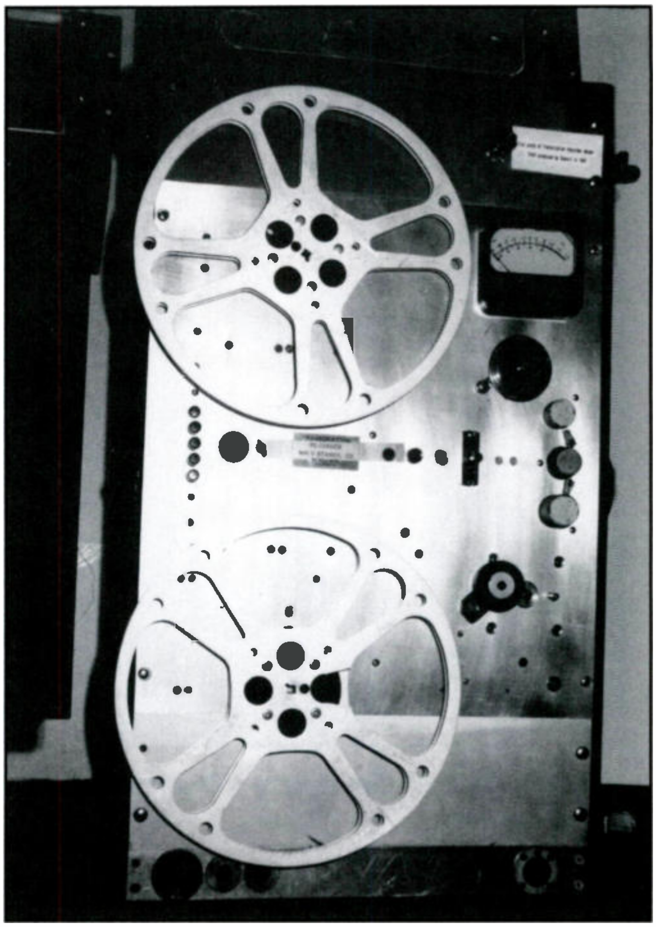


Figure 8. The first prototype of the TR-47 tape recorder.

When Olin Dupy retired from MGM, Stancil hired him as a designer. Stancil also hired C.C. Davis, of ERPI and Westrex, designer of the 45/45 stereo cutter and the patented Davis film sound transport. Because of Dupy's experience on electrical interlock motors, the company was a prime source of interlock motors for Cinemiracle, Cinerama, and other roadshows that used interlock equipment.

By this time, Stancil-Hoffman had outgrown several locations and was in the famous building at the corner of Highland and Willoughby in Hollywood. The move to Santa Ana, CA, was made in the 1980s when a Hollywood location no longer mattered. At the same time, the company name was changed to Stancil Corp.

OTHER PROJECTS

Stancil patented a film timer. It used three interlock motors—one to drive the hour drum, one to drive the minute, and one to drive seconds. The AC that fed each motor was recorded on three tracks, so regardless of the film speed, it would always show correct time. The same thing is done today electronically; it's called SMPTE time code. Another project was a special recorder for the California Institute of Technology's seismic research department. The tape moved one ten thousandth of an inch per second. It was placed on an earthquake fault and recorded for three months. The tape was then brought back and run at 15 in./sec. It sounded like gunfire, but the periodicity of the earthquakes could be charted.

RCA and Western Electric prevented film studios from using magnetic recording because they held the patents. They were not interested in replacing the optical sound-on-film method of recording, which they also owned. Stancil finally did get a Western Electric license.

After the multi-tracks and the first audio recorders, Stancil-Hoffman brought out a film recorder, the S4. Since Stancil worked in all of the studios and knew so many people, he was able to sell his equipment.

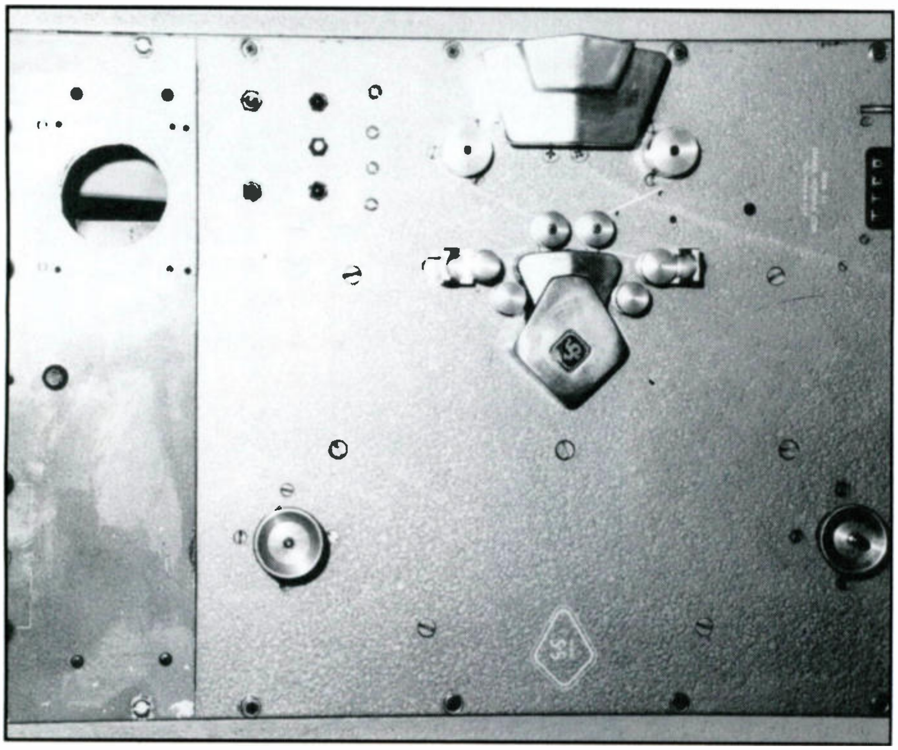


Figure 9. The S-5 film recorder. The photo is on its side, with the bottom facing left.

3D FILMS

Arch Obler, who developed 3-D movies, used Stancil-Hoffman equipment in his first 3-D production, *Bwana Devil*. (An awful picture as I remember, and Stancil agreed, even though he got screen credit.) Loren Ryder at Paramount was the only one that would take a gamble and buy any equipment before Stancil had a license agreement. When Shearer at MGM also bought equipment, Stancil was off and running in the motion picture business. Introduced in 1947, Stancil-Hoffman film recorders were used around the world. Nearly everyone attending the famed USC School of Cinema learned on Stancil-Hoffman filmsound equipment. Although they have been out of production for over a quarter century, many of the recorders are still in operation.

I found the Stancil-authored booklet explaining magnetic recording in Stancil's collection of memorabilia. The booklet was my bible when I was young and starting out in the business. I mentioned this and Stancil's response surprised me. "Sorry I did that to you. That's the most simplistic write-up in the world. It's elementary, it's grade school. I thought I'd do it in a very pedantic way. Everything I read on magnetics were books thick with formulas by nobody that ever built anything. I despise that," he said. That little pamphlet was the foundation for many people learning about magnetic recording.

Stancil has a patent for a synchronizing system between 78 RPM discs and camera. It's a toroidal coil with sweeping contacts that sense the DC at three points, one hundred and twenty degrees apart. The Navy liked the concept and gave Stancil a contract to produce a similar type of DC interlock for a radar dish. They had been using a huge mechanical differential to do it—he could do it with a 24 volt battery.

RANGERTONE

I asked about Colonel Richard Ranger. "Oh yes, I was good friends with Ranger," Stancil said. "I got him in on the pool to fight the Armour patents. We visited back and forth, he spent a lot of time at my shop. He was famous for producing the chimes for NBC and the first tape to film synchronization system. In my head displays I have some Rangertone heads. We kept in contact; I must have forty or fifty letters from him."

Stancil-Hoffman audio products became transistorized in 1955. There were only one or two types of transistors at that time and their signal-to-noise ratio was not very good. "I used to buy two or three dozen transistors, at prices I could not afford, to get enough to build a five watt amplifier," Stancil said. "If you sneezed, everything quit. Now look what you can do with a transistor and a little heat sink. They talk about five hundred watts as if it were nothing."

Early on Stancil started using the 'IF' can enclosures for modular construction. They were small cans about three inches high and about one and one eighth inch square, with octal sockets and circuit boards inside. The modules were bias oscillators, and record, play, line and power amplifiers. They were used in logging equipment, speech equipment and motion picture equipment. Often, by the time it was in a production unit, the transistor used was obsoleted in favor of a new model. This was progress, but it still left the manufacturer holding the bag. By using plug-in modules, Stancil could obsolete one circuit and not throw out the whole system. Stancil's newest equipment uses surface mount and integrated circuit technology, with multiple channels on one circuit board.

According to Stancil, "I saw the limited future of building equipment for the film business. It was too custom-oriented. The motion picture business takes a lot of time, and you don't make much money in it. In those days, in order to sell a piece of equipment, you had to teach (customers) everything. For the first time, people were able to buy a recorder, and then they were *film producers*. They had no money, everything had to be on a cash basis. They would go bankrupt right and left after you spent days and days with them. I finally gave it up. The last film unit we built was the S6."

LOGGERS

As Stancil-Hoffman Corp. moved away from the entertainment technology field, it moved more and more into the public service logging field. "Now I've got stuff under way in solid-state memory, on DAT, and also on VHS in our future recording systems. In fact, I have an instant recall recorder out now," said Stancil. "It's something you put on a telephone line, emergency messages come in and you're always recording up to sixty-four minutes. It's just like a tape loop, but you can play back any message that came in: (for example) 'My house is burning down!' but you didn't get the street number, so you back up and replay the message, meanwhile you're still recording as more messages come in. Everything is digital. We also put a beep on everything, the time of day, then we have the time display, actual time, and the time of day the message was recorded. Any message can be found in a matter of fifty or one hundred milliseconds."

After a career spanning sixty years and more, most people I know would be content to look back and review their accomplishments. Stancil was still looking ahead to new endeavors and new discoveries. His pursuit of excellence and dedication to his work was legendary. He was at work at his desk until he was taken, under protest, to the hospital where death overtook him on July 1, 1991.

Stancil said he just happened to be in the right place at the right time. But what was more important was he knew what to do with those opportunities. Stancil's daughter, Sharon Custer, along with her husband Mike, both of whom have been involved with the company for many years, will continue Stancil's work.

Here are some of Stancil's thoughts on our industry:

ON MUSEUMS and COLLECTIONS:

"Jack Mullin did a nice job on his museum. Somebody had to do what he did. I'm different—I'm working on the equipment of the future, that's why I don't get a chance to clean up my old equipment."

ABOUT WHERE WE'VE BEEN

"We started out with a Galena detector and used an Oatmeal box to wind the coils to make a variocoupler to receive a broadcast station. We wore earphones. The sound wasn't good, but to us it was better than the records of that time. About the time we became acclimated to crystal sets, we got vacuum tube circuitry. At first that didn't improve the sound much, but the standards began increasing in transmission of the audio on the telephone lines from the studio to the transmitter. In those days we didn't have volume units or volume indicators; we had miles of wire resistance, and that's what turned out to be our dB. Technology (of the telephone line and the transmission of the audio) kept improving. With the crystal set we hadn't bothered too much about the quality of audio, because for baseball games and voice it was good enough. Besides, people were used to the telephone. Signal-to-noise was an unheard-of definition in those days. As better tuning systems were developed, our tastes expanded because wider frequency bands were possible.

Once I heard that you could get amplification from a chunk of rock with a couple of electrodes in it, my ears pricked up.

When the motion picture studios came out with the big musicals, the reproduced sound, by today's standards, was awful, but it was better than most people had heard before. It created a demand for better and better audio reproduction."

ON TECHNICAL IMPROVEMENTS

"With the acoustic phonograph, the horn was used for the microphone, and the sound came out sounding like a megaphone. Once we got vacuum tubes for amplification, things began to shine a little bit. As we kept improving we got up to the point, with vacuum tubes in the late thirties, where there were eighty-two types of power rectifiers alone.

Then it really went wild with about half a dozen new tubes a year, until there were several thousand different tubes. Tubes gave us microphonics, power consumption and heat dissipation. Once I heard that you could get amplification from a chunk of rock with a couple of electrodes in it, my ears pricked up. I thought: now, maybe it's going to happen. That's why we went to transistors in 1953. I think the invention of the transistor is one of the great accomplishments of mankind. Consider what it's done for us in every field, no matter whether it's in audio, computers, instrumentation, mechanization, automobiles or whatever. Now, as we get into digital logic, transistors make possible getting four megabytes into one chip, or an IC that's the size of your thumbnail with many thousand transistors. The possibilities are endless. We've made progress, but it's not good enough yet. Who knows what's coming next. Is the sound any better? I don't know. I don't hear enough of it any more. It used to be that FM stations had to be wide band, low distortion and high signal-to-noise. Now with the modern so-called music, they might as well use two tin cans on a string."

COMPARING THE EARLY RECORDERS TO MODERN RECORDERS

"The defining factor of the fidelity of a recorded tape is the width of the playback gap. It's the rate of change of flux that determines the number of octaves you can cover. That becomes a limiting factor.

When you hear something, if you don't have anything to compare it to, it can be a tear-jerking beautiful-sounding piece. Then when you hear the real thing, the recording, in retrospect, was awful." db

ARTHUR M. NOXON

Trapping Bass In Your Project Studio

Sound is acoustic energy and rooms store this energy. Resonance is nature's most efficient way to store acoustic energy in a room. Resonant energy easily lasts two times longer than sounds that are not resonant, and this is how the coloration of sound occurs in small rooms.

AN ALL-CONCRETE REVERBERATION chamber can store sound for at least ten seconds, an empty gymnasium is good for five seconds, and an empty room in a house has a decay time of two seconds. In pro or semi-pro audio rooms, a decay time of no more than ½ second is preferred. The typical furnished but untreated residential-type room has decay times of 1 ¼ seconds. So, serious audio rooms need serious acoustic treatment. Midrange and high frequency sound is easily absorbed, but the lows are problematic. Sound absorbers that handle the lower octaves are called bass traps.

ROOM RESONANCE

Almost everyone can read about "room acoustics," which actually discusses the midrange and high frequency, the upper three octaves of the keyboard. Now, the domain of low frequency acoustics in small rooms is to be explored. This article will provide an overview of the theory, history and practice of bass trapping with an eye towards home and project studios.

Without proper decay times, mic work or listening in an audio room is hampered by excessive reverberation. Resonances color the acoustic signature because they are a group of specific tones that overhang longer than the others. Excessively sustained overtones cover over, blur and mask out the low level musical inner detail. The control of decay times in the audio room means controlling the resonances, and giving the room a neutral voice.

Resonant frequencies are not always the same; they will vary depending on speaker position. With a walking, talking person, the position of the sound source changes, stimulating different resonances. The loudspeaker however, is fixed in position. It stimulates the same group of resonances over and over again. The coloration is fixed; it penetrates and stains all recorded and playback material. Instead of capturing the "infinity" of musical variations that create evanescent luster in audio recordings, resonance forces a redundant tonal emphasis which renders music essentially boring—no matter how much talent is applied.

Electronic upgrades in the studio should develop enhanced performance. The need for any improvement springs from some dissatisfaction with the present system. The room acoustic is the first and last link in the audio chain. It is staggering to consider how many pieces of electronic gear have been purchased out of frustration with a system whose real problem was not electronic at all, but was driven by the colorations due to room resonance.

There are only two ways to get residual low frequency sound energy out of a room. The first and most common is leakage. Unlike the downtown recording studio, deep bass leaks out of most home and apartment construction. Leakage paths can be direct transmissions through the walls, ceiling, floor, doors and windows. The heavier the surface, the less leaky it becomes. Other leakage paths are through openings such as under the door.

Absorption is the second method by which acoustic energy is removed from a room. Downtown recording studios are heavy-walled and sealed airtight to keep unwanted sound out. This is called isolation. If sound is kept out, it is also kept in, and so studio builders have developed a variety of low frequency sound absorbing techniques. Hopefully, most of these will be reviewed in this article. The designer/contractor-built studios usually have bass traps built in. The rapid expansion of MIDI equipment has resulted in many serious home-based project studios that are virtually without acoustic control.

The single most important result in a properly bass-trapped room is that it has more bass, deeper punch and smoother extension. This sounds contradictory—that bass trapping a room gives more and not less bass. Actually, what you get is the bass you always had; you just could not hear it because the resonant colorations covered it.

Once the basic concepts of room resonance and bass traps are developed, the practical matter of setting up a room needs to be discussed. This is broken into two sections. Trapping the front or driven end of the room requires special considerations because of its proximity to the loudspeakers. The back of the room is more intuitively obvious and belongs to the world of deep bass traps.

Arthur M. Noxon is the president of Acoustic Sciences Corporation.

Virtually every downtown recording studio uses some type of bass trap to control distortion and coloration of the frequency response in the room due to low-end build-up. Bass traps in these studios can be found hidden above the ceiling, inside the walls, below the floor and sometimes even in adjacent rooms. The nagging problem for home and project studios is that most engineers cannot consider contractor renovations as an option for an acoustic upgrade of their living rooms.

ROOM ACOUSTIC BASICS

Before considering bass traps in detail, a review of acoustics is in order. This will develop a sense of perspective and scale. The behavior of sound waves and objects depends on the size of the wavelength, in comparison to the size of the object. Simply put, long wavelengths go *around* small things and small wavelengths get *reflected* by big things.

The wavelength of a sound is mathematically related to its frequency or tone. The higher the frequency, the shorter the wavelength. Our range of hearing officially spans ten octaves from 20 Hz to 20 kHz and we can perceive or feel sound even below 20 Hz. (1 kHz = 1,000 Hz of cycles per second.) An octave is the doubling of frequency: 20 Hz, 40 Hz, 80 Hz, and so on. For audio playback in small rooms, bass is considered to be the first four octaves (20 Hz to 320 Hz); mids comprise the next two (640 Hz to 5.12 kHz); and the highs occupy the last four octaves. Sounds of the piano keyboard are familiar to most of us; middle-C is a frequency of 256 Hz. The bass range on a piano occupies more than half of the piano keyboard, and about forty percent of the full auditory spectrum.

Bass wavelengths are similar in size to the room in which they exist. It's easy to calculate the size of a wavelength from the formula: wavelength n_\ = speed of sound (c) / frequency (f). By comparing sound wavelengths to the size of a house, the size of bass wavelengths are evident.

The shortest "bass" note—A440—has a wavelength of about 2.5 feet. The longest wavelength is 56 feet, and it belongs to 20 Hz.

Full range speakers generally produce sound extending down through most of the lower end of the piano keyboard. Subwoofers produce sound specifically in the last octave of the piano's keyboard and the one just below it, the first audible octave.

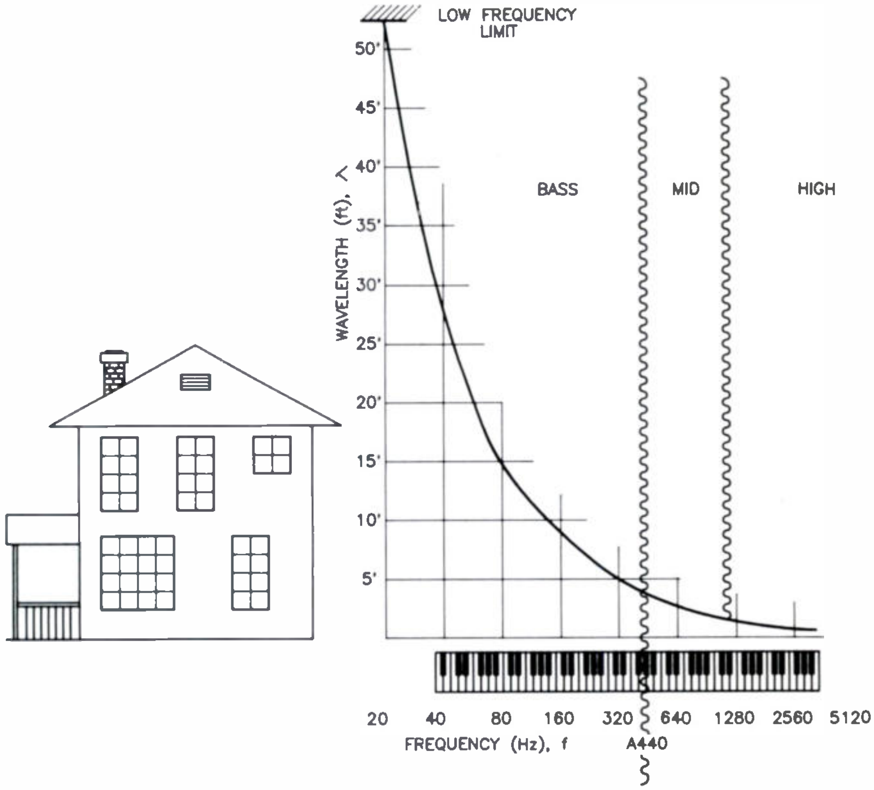


Figure 2. Wavelengths of sound.

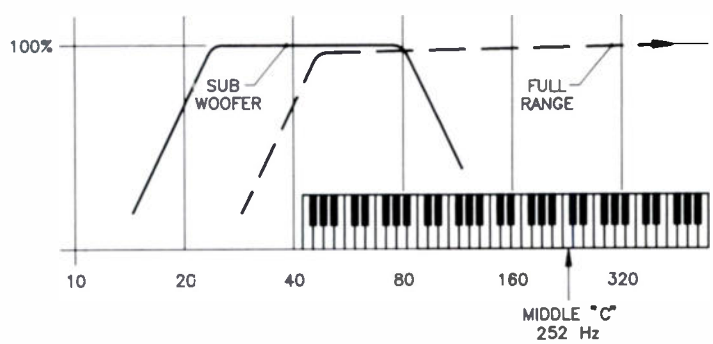


Figure 3. Loudspeakers and low-end rolloff.

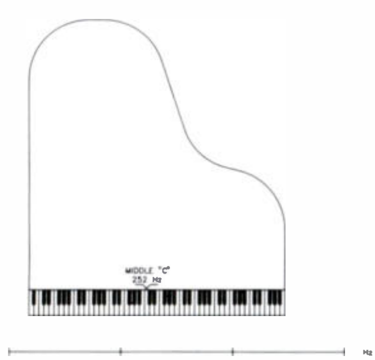


Figure 1. The keyboard and the audio spectrum.

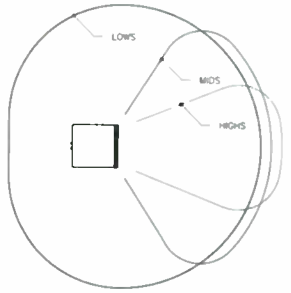


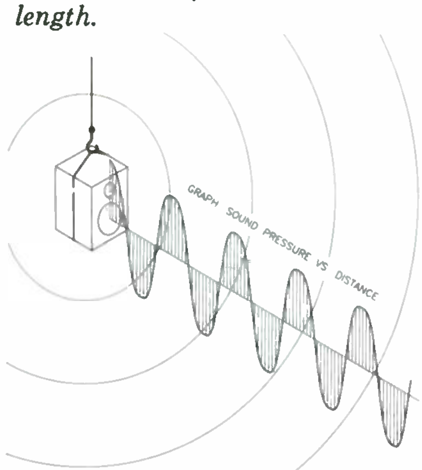
Figure 4. The directionality of speakers.

SPEAKER DIRECTIVITY

Speakers possess frequency-dependent directional qualities. For both mids and highs they produce adequate sound levels only in the forward direction towards the listener. Lower frequencies from the same speakers, however, radiate equally in all directions. This directionality means that mids and highs are efficiently beamed towards the listener, and little acoustic energy is wasted on illuminating the rest of the room.

The lows easily require six or more times the acoustic/electric power than the mids and highs to achieve the same sound level at the listener's position. Speaker efficiency is one reason for power gulping; the other is directionality. Because bass waves are bigger than the speaker, they travel with equal strength in all directions. The speaker is an "omni" pattern sound source. Often much of the bass wavefront has bounced off of the walls, floor and ceiling of the room before it even reaches the listener.

Figure 5. Wavefronts and wavelength.



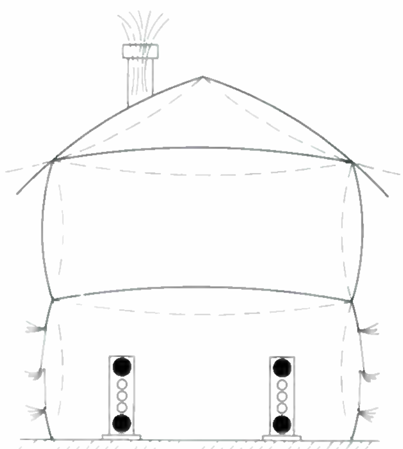
Sound is an airborne ripple or wave whose speed (c) is about 1,128 ft./second. Consider the piston of a loudspeaker that is vibrating to and fro at 100 Hz. In the exact amount of time it takes for the speaker cone to make one cycle, or complete a round trip ($1/100$ second), the sound wavefront it generated will have moved away from the speaker ($1/100 \times 1128$) some 11.28 feet. For a continuous tone, this becomes a repeating event. As you move away from the speaker, every 11.28 feet would be the same acoustic condition.

THE BREATHING MODE

This review of small-room acoustics begins with the lowest octave. Here, the wavelength is quite long as compared to the size of the playback room. The room as a whole experiences internal pressure changes. Acoustic activity in this region below the room's so-called "cut-off frequency" remains quite audible. Here the speaker is acting on the room as if it were a pneumatic plunger, alternating between pressurizing it and pulling a partial vacuum on it. The walls, floor and ceiling react to what seems to be a rapidly changing "barometric" pressure in the room. Room surfaces billow out and then cave in with each cycle.

Major structural resonances are easily stimulated by breathing mode acoustics, a common problem in playback for the larger power systems of today. The surfaces of the room simply shudder in the bottom end as the speakers stimulate, then overpower the mechanical stability of the room. The result at high sound levels is a total loss of control for low-frequency musical reproduction, as if sound in the room "crumbles" when it is overloaded. This LF breakup of the room itself is particularly evident in the concussive punch bass beat attack transient.

Figure 6. Deep bass—breathing mode.



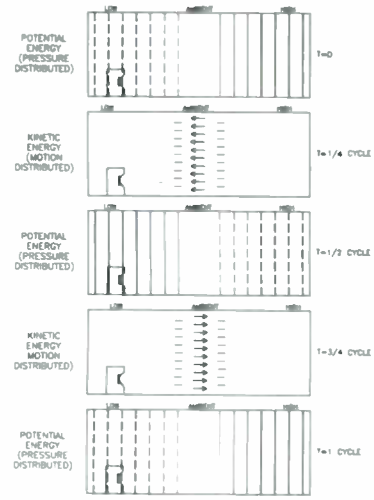
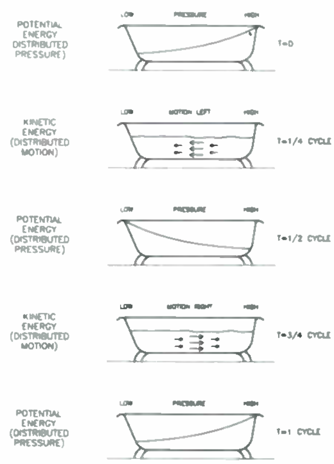


Figure 7. 1st room resonance (1,0,0).

ROOM MODES

As the tone from the speaker is raised in pitch, out of the deep bass octave and into the piano's first bass octave (40-80 Hz), a new class of room acoustics develops, called Room Resonant Modes. The lowest frequency at which this can occur is

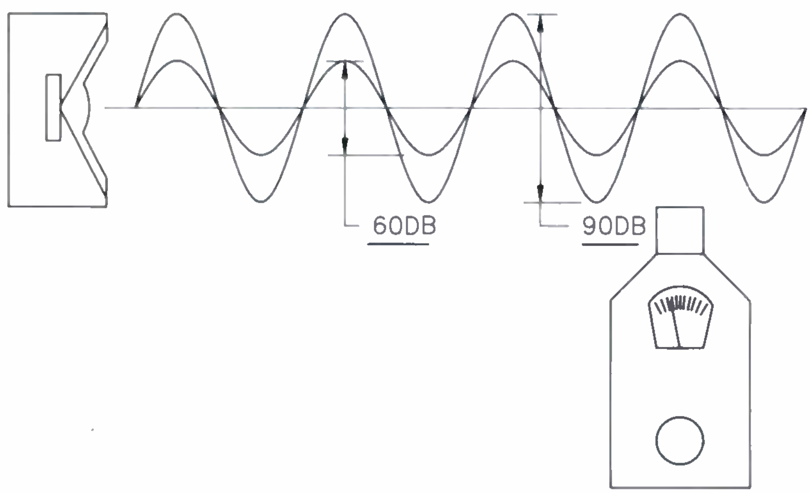
Figure 8. Bathtub mode (1,0,0).



called the long dimension axial (1,0,0) mode.

The fundamental room resonance is easily stimulated when the speaker is located at one end of the room and the wavelength of the tone played happens to be twice as long as the room. The wave from the speaker travels down the room only to bounce off the rear wall and return to the front of the room. During this time the speaker makes one full cycle of motion itself. It generates a tone exactly in step (or in phase) with its reflection. These two waves—the old reflected wave and the new one—add together exactly, without confusion. After a number of cycles the sound levels build, enveloping the room in resonance.

Figure 9. Sound and the use of an spl meter.



For a non-resonant tone, sound builds up in the room in highly disorganized manner. With resonance, however, the air is stimulated into a "sloshing" mode of behavior, not too unlike what can happen with a child in the bathtub if their to and fro movement happens to keep time with the water's natural end-to-end slosh motion, called first harmonic.

MEASURING RESONANCE

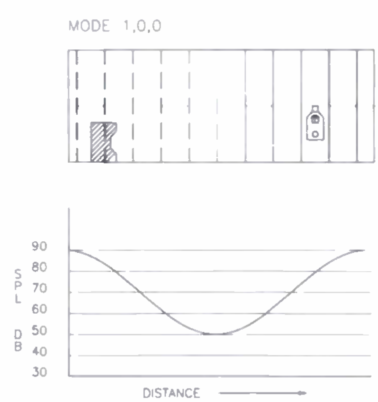
It is interesting to explore acoustic resonance with a SPL meter. Such a meter is very useful, can be found at stores like Radio Shack, and cost as little as $30.00. You can also use a mic patched into your board, keeping an eye on the VU meter. Sound meters measure the strength of "sound pressure changes." If the SPL meter reads 90 dB, that means the air pressure at the microphone is fluctuating strongly above and below ambient air pressure with a strength of 90 dB. Compare this to a 60 dB reading and notice that the fluctuations in pressure are much smaller and the sound is quieter.

By the way, dB A is not a flat response curve. It is rolled off gradually below 1 k as our own hearing response does. The dB C scale is "flat" for most purposes. A mic, patched through without equalization will be close to dB A levels, not dB A levels. The dB C or flat response weighting is best for room acoustic measurements and the mic should be an omni mic.

If the mic or SPL meter is moved from one end to the other end of a room that is in the fundamental mode of resonance, data points can be taken and plotted against position. High SPLs are detected at both ends of the room, and a low SPL in the middle. These are known in audio as "hot" and "cold" spots; the "hot spot" is where pressure changes strongly occur and the "cold spot" is a location where pressure only slightly changes.

Just because we don't hear sound in the cold spot doesn't mean the acoustic energy is gone. The sound may be "cancelled," but the kinetic part of acoustic energy is in full presence. Although we can't hear acoustic kinetic energy, a ribbon mic properly oriented can pick it up. Note that the same ribbon mic in a pressure zone will not register any sound. This is because ribbon mics pick up the air motion of sound while condenser mics pick up the air pressure of sound. For a ribbon

Figure 10. Sound level distribution.



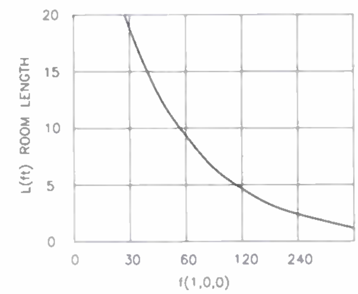
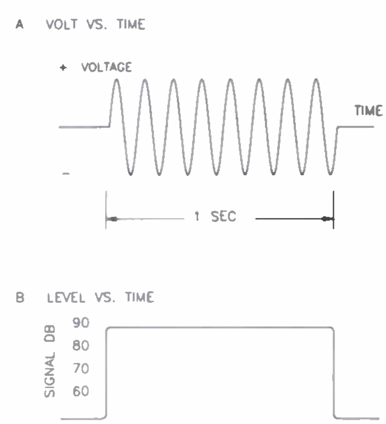


Figure 11. Lowest resonance frequency.

Figure 12. The tone burst.



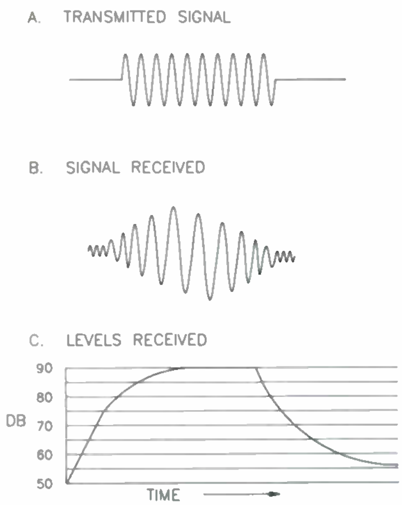


Figure 13.Burst in a room (end).

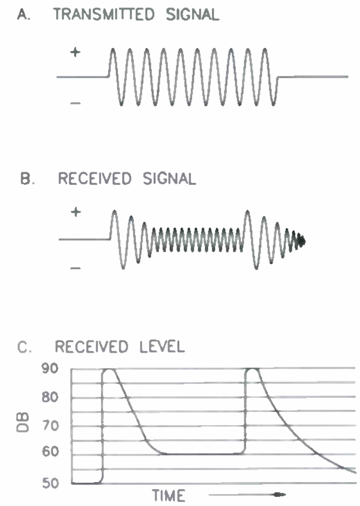


Figure 14. Burst in a room (middle).

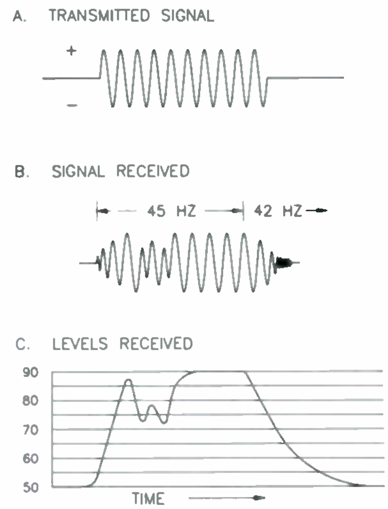


Figure 15. Resonance coloration.

mic to pick up the acoustic kinetic energy, it must be aligned per indicator to the direction of air motion. If rotated ninety degrees so the plane of the ribbon is aligned with the direction of the acoustic kinetic energy motion, the mic will not give a reading.

The frequency of the lowest room resonance (1,0,0) is easy to calculate from f_{100} = C/2L. Measure the length (L) of your room and use the equation to calculate the room's fundamental resonant frequency. The graph of the equation is also useful to use.

LISTENING TO RESONANCE

The size, shape and internal details of a simple room will affect its resonance frequencies. By using a f_{100} tone burst, lasting about one second, as a test signal and feeding it to a speaker, we can watch the SPL meter to illustrate overall frequency response of the room. By listening first to the burst over headphones and then again while using the room as an acoustic coupler, a very clear audition of room acoustic resonance effects can be heard.

This kind of test, called a MTF (Modulation Transfer Function) test, is the basis for checking the quality of any communications channel. The Studio Reference Disk by Prosonus (list $69.95) has this test on track 50. MTF testing is the more full bandwidth, musical cousin to speech intelligibility tests that sound contractors are wrestling with these days.

The "Hot" f_{100} location to illustrate the presence of excessive reverberation is at the back wall of the room. Here one hears the slow "turn on," excessively high sound levels, and a sluggish "turn off" response characteristic. The sound of the tone burst sound is not sharp, but "blooms" and "fades." This can be characterized as the difference between the test "boop" sound and the "moo" sound delivered to the listening position. The fact that a distinct, sharp signal is not really heard is clear evidence that it is the *room* we are listening to and not, as we usually presume, the *speaker*!

What we hear, in fact, is the gradual build-up of energy in the room as the speaker begins to move or slosh the air in the room. With each cycle of continuous tone, the sound level continues to build, but only until the power being pumped into the room by the speaker exactly equals that being lost and dissipated by friction and leakage. Only then can a steady-state sound level be reached.

When the speaker quits vibrating, the sound does not just simply stop. There is built-up and stored acoustic energy in the room which requires time to damp out. Acoustic friction reduces the energy of sound in the room, as does the leaking of sound out through windows, doors and the walls. It's the leaking part that neighbors will comment on.

SOUND "CANCELLING," THE COLD SPOT

When sitting about the middle of the room at the "cold spot" while the first resonance is set up, the very curious effect of "sound cancelling" occurs. Here, the sound from the speaker is exactly out of phase with that of the room resonance at that location. Sound pressure may be cancelled, but nature does not give up so easily; acoustic energy is not cancelled. If sound (acoustic pressure) is "cancelled" in one part of the room, it has only been replaced with acoustic kinetic. Conversely, sound pressure will be found substantially louder elsewhere in the room at locations that have been stripped of acoustic kinetic. Acoustic energy is an interplay of acoustic pressure and acoustic kinetic. Ocean waves have a similar action—the water wave has height (pressure) and motion (kinetic) energy.

When we audition the one second tone burst here, we first hear clearly the initial sound from the speaker. But it becomes quieted as the buildup of the resonance in the room reaches full strength and cancels the direct sound at the listening position. When the speaker is turned off, suddenly we hear the sound of the reverberant field as it decays. The response of the burst is not the clean, crisp "boop" sound. It is more like a "bow-wow."

In either case, and depending where one sits, the in phase or out of phase room resonance/speaker coupling effects dramatically rewrites musical dynamics and intonation. This illustrates why the engineer can hear magic and the producer on the talent couch still thinks it needs work—what you hear in the bottom end depends on where you sit.

Farfield playback monitors strongly couple to the room acoustic—that's why they aren't used very much except in well-designed downtown studios. It costs a lot to buy the monitors and a lot to fix the room to play them in. The move has been towards nearfield monitors that give strong direct signals and weak room resonance coupling.

The problem here is no bottom end—engineers have to just punt into the mix below 60 Hz. The next move up is to midfield monitors, a compromise, but still no bottom below 45 Hz. Another attempt is to add subwoofers into the system to get the bottom end back up.

> **A further coloration problem occurs when the speaker is shut off; the sound decays at the nearby room resonance of 42 Hz, and *not* with the sound of the musical note of 45 Hz.**

ACOUSTIC COLORATION

So far, the distortion of amplitude modulation has been shown to result from room resonance. The mic or listening position has a tough time tracking the low frequency (LF) transients in musical passages. The fast tracking of a room is one important aspect of pro room acoustics. There remains another acoustic gremlin that impacts musical accuracy: coloration. By playing a tone burst into the room at a frequency just off a nearby resonant frequency, both the attack and the sustain of the burst develop a "vibrato," a beat frequency related to the difference between the applied tone and the nearby resonant frequency.

For example, if a 45 Hz note is played into a room with a resonance mode at 42 Hz, there would be a beating effect in the attack and sustain of a vibrato at the difference frequency of 3 Hz. A further coloration problem occurs when the speaker is shut off; the sound decays at the nearby room resonance of 42 Hz, and *not* with the sound of the musical note of 45 Hz. Essentially the note sours in decay. This effect, like the other resonance-controlled playback defects, remain clearly audible by means of an A/B headphone test.

(Concluded next issue.) db

db BUYER'S GUIDE

Compressors, Limiters, Noise Gates, Noise-Reduction, Miscellaneous, Digital Workstations

On the pages that follow, we present this issue's Buyer's Guide on Compressors/Limiters, Noise Gates, Noise-Reduction equipment, Miscellaneous, Digital Workstations. The information contained is supplied by the respective manufacturers. Further, if a manufacturer that you seek is not listed, the chances are strong that, as many times as we tried, we could not get information from them.

COMPRESSORS/LIMITERS

ALESIS STUDIO ELECTRONICS

The Micro Limiter features full stereo; soft-kneed limiter/compressor with unique attack and decay characteristics; Input, Release and Output controls, as well as an in/out switch and 20 kHz bandwidth.

Dimensions: 1.75 in. x 6.3 in. x 6.25 in.

Weight: 1.5 lbs.

Price: $149.00

APHEX SYSTEMS LTD.—See our ad on page 13

The Model 320 Compellor is a compressor/leveler/peak limiter featuring dual mono/stereo operation, leveling speed and peak limiter switchable on front panel; servo balanced inputs and outputs; operates at –10 dBv, +4 and +8 dBu.

Dimensions: 1.75 in. x 19 in. x 10 in.

Price: $1,350.00

The Model 651 Expressor is a single channel compressor/limiter with a high-frequency expander. Features full function compressor with control of threshold, ratio, attack and release. Special HFX allows for dynamic "decompression" of selectable high frequency range.

Dimensions: 1.75 in. x 19 in. x 9 in.

Price: $495.00

The Model 720 Dominator II is a precision multiband peak limiter. Stereo peak limiter is used as final protection in audio chain; absolute peak ceiling with no over shoot; adjustable ceiling level; density; release time; servo balanced inputs and outputs.

Dimensions: 1.75 in. x 19 in. x 9 in.

Price: $1,350.00

ARX SYSTEMS—See our ad on page 2

The Quadcomp features four compressor/limiters with variable ratio threshold and output gain in the one package. With full LED gain reduction metering and balanced inputs and outputs.

Dimensions: 1.75 in. x 19 in. x 6 in.

Weight: 5 lbs.

Price: $799.00

The Afterburner features two channels of compressor/limiter with variable ratio and threshold and output gain. The Afterburner can be switched into "Mono" mode allowing separate Dynamics control of low and high frequencies. Also features balanced inputs and outputs on jacks and XLRs.

Dimensions: 1.75 in. x 19 in. x 6 in.

Weight: 5 lbs.

Price: $599.00

ASHLY AUDIO, INC.

CL100 is a mono compressor/limiter that includes a power supply.

Price: $229.99

CL50E is a mono compressor/limiter with detector patch points for frequency-selectiive limiting, gain reduction meter(s), in/out bypass switching.

Price: $299.99

CL52E is a dual/stereo compressor/limiter similar to the CL50E above, has output level meters and clipping indicators and front-panel stero tie switching.

Price: $499.99

CL85E is a gated compressor/limiter with independent attack, release and ratio adjustment; adjustable attck and release thresholds; and a gated release funtion that minimizes breathing.

Price: $399.99

BROOKE SIREN SYSTEMS, A DIVISION OF AKG ACOUSTICS, INC.

The DPR-402 two-channel compressor/limiter features high-frequency de-esser and wide band de-esser with peak limiting; adjustable speed; dynamics program manipulation; and full LED metering for both input and output.

Dimensions: 1.75 in. x 19 in. x 9 in.

Weight: 9 lbs.

Price: $1,339.00

The DPR-404 four-channel compressor features a de-esser with high frequency de-essing. Each channel includes threshold control; below threshold metering; ratio control; gain reduction meter; clip LED; gain control and linking.
Dimensions: 1.75 in. x 19 in. x 11 in.
Weight: 8 lbs. 2 oz.
Price: $1,399.00

dbx, A DIVISION OF AKG ACOUSTICS, INC.
The 160XT features a dual display system that monitors true RMS input or output levels with a 19 LED display while simultaneously monitoring gain reduction over a 40 dB range. Choice of OverEasy or classic hard-knee compression.
Dimensions: 1.75 in. x 19 in. x 9.25 in.
Weight: 6.5 lbs.
Price: $459.00
The 163X features OverEasy compression action with key operating parameters integrated and controlled by a single slide control. Designed for operation at nominal studio levels from –30 dBu to +10 dBu.
Dimensions: 1.75 in. x 19 in. x 9.25 in.
Weight: 2.5 lbs.
Price: $169.00
The 165A features OverEasy compression and PeakStop peak blocking. Also features automatic and fully adjustable manual attack and release rate controls with matched RMS detectors for true power summing when stereo strapped.
Dimensions: 3.5 in. x 19 in. x 9.25 in.
Weight: 6.5 lbs.
Price: $999.00
The 166 is a dual-channel compressor with an expander gate before each compressor. The combination of noise gate, OverEasy compressor and PeakStop limiting provides complete control of the input dynamics.
Dimensions: 1.75 in. x 19 in. x 8 in.
Weight: 6.5 in.
Price: $629.00

DOD/DIGITECH/AUDIO LOGIC
The Audio Logic 266 compressor/limiter/gate features program dependent attack and release times for the compressor/limiter; feed forward gain control; side chain input and outputs; stereo link; and direct input to output bypass.
Dimensions: 1.75 in. x 17 in. x 6 in.
Price: $500.00
The Audio Logic 660 compressor/limiter/gate features feed forward gain control; side chain input and outputs; stereo link; variable attack time; variable release time; and –40 dBu to +20 dBu compressor threshold control.
Dimensions: 1.75 in. x 17 in. x 6 in.
Price: $340.00

FURMAN SOUND, INC.
The Model LC-X expander/compressor/limiter also includes a de-esser and hard limiter. Controls include three threshold; two ratio; attack, release; and output. Features switchable LED meter; side chain jacks; bypass switch; de-ess button; stereo interconnect; and on/off transient muting. Dimensions: 1.75 in. x 19 in. x 8 in.
Weight: 7 lbs.
Price: $369.00
The Model LC-6 stereo limiter/compressor/gate is a two-channel unit that may be switched for stereo operation. Controls include input, output, compress, threshold, gate threshold, attack, release and ratio. Includes LED meters and side chain jacks; ground lift switch.
Dimensions: 1.75 in. x 19 in. x 8 in.
Weight: 7 lbs.
Price: $439.00
The Model LC-3A limiter/compressor includes input, output, attack, release and ratio controls. Has LED meter to indicate gain reduction; overload and power indicators; side chain jacks; de-ess button; ground lift switch; optional balanced configuration.
Dimensions: 1.75 in. x 19 in. x 8 in.
Weight: 7 lbs.
Price: $269.00

JBL PROFESSIONAL
Model 7110 is a single channel limiter/compressor for both peak and average gain reduction. Features Smart Slope compression circuitry; "Automatic Preset" feature; input/output/gain reduction metering in single rack space chassis.
Price: $495.00
Model 7112 is a dual channel limiter/compressor for both peak and average gain reduction. Features Smart Slope compression circuitry; "Automatic Preset" feature; input/output/gain reduction metering; and selectable "link" for multi-unit chaining.
Price: to be announced
Model 7122 is a dual channel frequency selectable limiter/compressor/expander that provides both peak and average gain reduction; Smart Slope compression circuitry; "Automatic Preset" feature; input/output/gain reduction metering; and selectable "link" for multi-unit chaining.
Price: to be announced
Model LA-4 Electro-Optical Attenuator compressor/limiter features selectable compression ratios; full VU metering; input overload indicator; and simple stereo coupling. Rack-mountable alone or in pairs in two rack spaces.
Price: $730.00

The 1176LN is a single channel peak limiter with selectable compression ratios; adjustable attack and release times; balanced input and transformer-balanced output; standard 19 in. rack mount in two spaces.

Price: $830.00

The 1178 is a dual channel peak limiter with selectable compression ratios; adjustable attack and release times; balanced input and transformer-balanced output; standard 19 in. rack mount in two rack spaces.

Price: $1,390.00

The Model 562 Feedback Suppressor has a single channel, with five independently adjustable notch filters; high and low cut end filters; protective peak clipper; headroom indicator; bypass option and single rack space chassis.

Price: $890.00

LT SOUND

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

Dimensions: 1.75 in. x 19 in. x 7.25 in.

Weight: 8 lbs.

Price: $995.00

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

Dimensions: 3.5 in. x 19 in. x 7.25 in.

Weight: 11 lbs.

Price: $1,250.00

Model SL-2 is a stereo limiter/expander with features including simultaneous limiting and expansion functions, de-essing and stereo or independent operation.

Dimensions: 1.75 in. x 19 in. x 7.25 in.

Weight: 7 lbs.

Price: $395.00

ORBAN, A DIVISION OF AKG ACOUSTICS, INC.

The 412A controls interact to simplify and speed setup, and to prevent errors. Peak limiting and compressor functions are cross-coupled to eliminate potential pumping and modulation effects.

Dimensions: 1.75 in. x 19 in. x 5.3 in.

Weight: 5 lbs.

Price: $525.00

The 414A is a stereo/dual version of 412A.

Dimensions: 3.5 in. x 19 in. x 5.3 in.

Price: $800.00

The 422A features adjustable attack/release time and compression ratio. Selectable linear or exponential release time characteristic; defeatable release gate with adjustable threshold causes gain to move slowly toward user-adjustable value during pauses, preventing noise rush-up, pumping and breathing.

Dimensions: 3.5 in. x 19 in. x 10 in.

Weight: 10 lbs.

Price: $680.00

The 424A is a dual/stereo version of the 422A.

Dimensions: 3.5 in. x 19 in. x 10 in.

Weight: 10 lbs.

Price: $1,150.00

The 464A rides gain and limits peaks. Provides input attenuator, gate threshold, release time and shape, pre-emphasis, output level, AGC rate and RF limiting—all can be tailored to a specific installation/application.

Dimensions: 1.75 in. x 19 in. x 9.625 in.

Weight: 8 lbs.

Price: $1,200.00

PEAVEY ELECTRONICS CORPORATION/AUDIO MEDIA RESEARCH —**See our ad on Cover III**

The CDS 2 dual channel compressor/limiter/de-esser features compression ratio control; switchable atttack/release time; "soft knee" type compression; side chain capability; stereo/mono operation.

Dimensions: 1.75 in. x 19 in. x 8 in.

Weight: 6 lbs.

Price: $199.99

RANE CORPORATION

The FPL Program Limiter from the Flex Series in the HR format provides four independent channels of servo-lock limiting. Each channel independently switchable to AutoSlave mode, which links the side chains of all selected channels.

Dimensions: 1.75 in. x 8.5 in. x 8 in.

Weight: 4 lbs.

Price: not available

ROCKTRON CORPORATION

The Model 360 features compression, peak limiting and Hush II noise reduction; stereo master; unbalanced ¼ in. jacks; and gain reduction metering.

Dimensions: 19 in. single rack space
Price: $569.00
The Model 321 features compression; peak limiting; stereo master; unbalanced ¼ in. jacks; and gain reduction metering.
Dimensions: 19 in. single rack space
Price: $499.00
Model 300A features peak limiting ratio; attack; release and threshold controls; Hush II noise reduction; gain reduction metering; side chain input and output.
Dimensions: 19 in. single space
Price: $419.00
Model 300G is foot switchable; features peak limiting ratio; attack; release and threshold controls; Hush II noise reduction; gain reduction metering and sidechain input and output.
Dimensions: 19 in. single space
Model 311 is a mono compressor/expander with input gain switch; slave/master switch; is foot switchable; and has gain reduction metering.
Dimensions: 19 in. single space
The Model CE2 is a compressor/expander with mono; stereo strappable gain reduction metering and a clip indicator.
Dimensions: ½ rack
Price: $219.00

RSP TECHNOLOGIES
The Model 2200 features multiband compression; leveling; peak limiting; Hush noise reduction; ¼ balanced/unbalanced and XLR I/O; stereo master; and crossover point of 500 Hz or 2 kHz.
Dimensions: 19 in. single rack space
Price: $899.00

SOUNDTECH
The ST200CL compressor/limiter features stereo/mono; compressor on/off switches; gain reduction meter; variable gate; threshold; compression ratio; attack; release; input/output levels; balanced XLR/unbalanced ¼ in. inputs/outputs.
Dimensions: 1.75 in. x 19 in. x 7 in.
Weight: 6 lbs.
Price: $349.90

SYMETRIX
Model 501 Peak/RMS compressor/limiter includes separate processors for simultaneous compression and Infinity:1 peak limiting. It provides absolute overload protection. Balanced and unbalanced ins/outs make interfacing easy.
Dimensions: 1.75 in. x 19 in. x 4.5 in.
Weight: 7 lbs.
Price: $349.00
Model 501-01 is the same as the 501, but with transformer-coupled outputs.
Dimensions: 1.75 in. x 19 in. x 4.5 in.
Weight: 7 lbs.
Price: $349.00
Model 525 dual gated compressor/limiter is a two-channel or true stereo device with program-controlled attack and release times. The compressor/limiter governs levels, while the expander/gate eliminates "breathing" and extraneous noise. Has side chain accessibility.
Dimensions: 1.75 in. x 19 in. x 4.5 in.
Weight: 7 lbs.
Price: $539.00
Model SX208 stereo compressor/limiter is easy to use with straightforward controls. Program-driven attack and release times help produce wide dynamic range and low distortion. Balanced and unbalanced signal connections make setup fast.
Dimensions: 1.5 in. x 8.2 in. x 6 in.
Weight: 5 lbs.
Price: $299.00

NOISE GATES

ALESIS STUDIO ELECTRONICS
The Micro Gate features keyable stereo in, stereo out, noise gate with Threshold, Delay and Rate controls; smooth, quiet operation; 20 kHz bandwidth; In/Out switch.
Dimensions: 1.75 in. x 6.3 in. x 6.25 in.
Weight: 1.5 lbs.
Price: $149.00

ASHLY AUDIO, INC.
SG33E is a dual/stereo noise gate with a fast10 msec rise time, 60 dB threshold range, balanced or unbalanced inputs; and extremely low noise and distortion.
Price: $419.99
SG35E is the same as the SG33E but is a quad noise gate.
Price: $669.99

APHEX SYSTEMS LTD. —See our ad on page 13
Model 612 Expander/Gate is dual channel with switchable high/low cut filters in side chain; full function with controls for attack, hold and release times; key listen, duck and stereo line switches; servo balanced in and out.
Price: $795.00

ARX SYSTEMS —See our ad on page 2
The Sixgate offers six channels of independent noise gating with variable threshold; attenuation and release times; balanced jack in and out; and LED indication of operating status as well as a hardwire bypass. Other features also available.
Dimensions: 1.75 in. x 19 in. x 6 in.
Weight: 5 lbs.
Price: $649.00

AUDIO MEDIA RESEARCH —See our ad on Cover III
The NGT 2 dual channel noise gate features side chain input and insert capability; synch trigger outputs; complete parameter control; bypass switches; variable attack/release time.
Dimensions: 1.75 in. x 11 in. x 8 in.
Weight: 6 lbs.
Price: $199.99
The NGT 4000 is a four-channel VCA-based noise gate/downward expander with complete control of threshold; attack; release; hold-off time; attenuation; gain trim; and side chain frequency contour; side chain signal shaping or processing; and electronically balanced inputs/outputs.
Price: to be announced

BROOKE SIREN SYSTEMS, A DIVISON OF AKG ACOUSTICS, INC.
The DPR-502 features two channels with key filters; an internal/external key source; key listening; gating or ducking; peak and average active window metering; threshold control; range control; gate profile section; auto attack mode switch; and auto dynamic enhancement.
Dimensions: 1.75 in. x 19 in. x 9 in.
Weight: 10 lbs.
Price: $1,359.00
The DPR-504 four-channel noise gate features a parametric key filter; key filter listening; simultaneous key level; threshold metering with average and peak metering; gate status LED; release/hold switch (hold tracks proportionally with release times); attack; switchable auto/fast.
Dimensions: 1.75 in. x 19 in. x 9 in.
Weight: 11 lbs. 10 oz.
Price: $1,359.00

dbx, A DIVISION OF AKG ACOUSTICS, INC.
The 363X features two channels with separate threshold; hold and release controls plus key monitor; key engage; stereo couple and bypass for stereo or dual independent operation. Allows removal of unwanted background sounds.
Dimensions: 1.75 in. x 8.5 in. x 7.25 in.
Weight: 2.5 lbs.
Price: $269.00
The 904 uses dbx's OverEasy action for smooth onset of gating. Attenuation limit, attack and release rates, and threshold are all adjustable. Also features programmed latch mode.
Dimensions: 5.25 in. x 1.5 in. x 9.5 in.
Weight: 0.75 lbs.
Price: $499.00

DOD/DIGITECH/AUDIO LOGIC
The Audio Logic 440 Quad Noise Gate features –60 dBu to +20 dBu threshold; 0 dB to 90 dB attenuation; 50 microsecond to 50 millisecond attack time; 50 millisecond to 5 second release time; feed-forward gain control; and separate key input for each channel.
Dimensions: 1.75 in. x 17 in. x 6 in.
Price: $400.00

FURMAN SOUND, INC.
The Model QN-44 quad noise gate features threshold; attack; release; and depth controls with "channel on" indicator. Key input jacks are provided for special effects. Features extremely low noise and distortion.
Dimensions: 1.75 in. x 19 in. x 8 in.
Weight: 7 lbs.
Price: $429.00

PEAVEY ELECTRONICS CORPORATION —See our ad on Cover III
The GateKeeper has five channels with automatic dedicated gate. Each channel has adjustable gate threshold and adjustable release time; threshold adjustable 10 dBv to constant on; electronically differential input, S.E. output.
Dimensions: 1.75 in. x 19 in. x 9.25 in.
Weight: 7 lbs.
Price: $299.99

SYMETRIX INC.

Model 564E is a four-channel expander/gate with professional features like "frequency-conscious" operation. A unique rotary control turns each channel into a gate or downward expander with a twist of the knob.

Dimensions: 1.75 in. x 19 in. x 10 in.

Weight: 11 lbs.

Price: $989.00

NOISE REDUCTION EQUIPMENT

AUDIO MEDIA RESEARCH —See our ad on Cover III

The Q Factor features two bypass switches; three dual concentric controls per module for precision adjustment of threshold and slope, plus attenuation of dynamic low pass filters and downward expanders. Electronically balanced inputs/outputs; two noise reduction systems in one chassis.

Price: $349.99

dbx, A DIVISION OF AKG ACOUSTICS, INC.

The 140X has two-channel Type II Noise Reduction. Patented RMS detection makes the system virtually immune to phase shift-related tracking problems. Tailoring of detector bandwidth prevents mistracking on broadcast quality media with limited high and low-end frequency response.

Dimensions: 1.75 in. x 8.5 in. x 7.25 in.

Weight: 6.5 lbs.

Price: $319.00

The 150X features two channels each of encode and decode electronics in a single package. Industry standard Type-I NR, compatible with all earlier Type-I systems.

Dimensions: 1.75 in. x 8.5 in. x 7.25 in.

Weight: 2.5 lbs.

Price: $319.00

The 911 incorporates one channel of encode and one channel of decode circuitry in the dbx Type-I format. Type-I provides as much as 40 dB of noise reduction for typical wide bandwidth media operating at 15 ips or faster.

Dimensions: 5.25 in. x 1.5 in. x 9.5 in.

Weight: 0.75 lbs.

Price: $239.00

The 941A has two channels of Type-II encoding. Tailored for greater than 40 dB of noise reduction with most broadcast media.

Dimensions: 5.25 in. x 1.5 in. x 9.5 in.

Weight: 0.75 lbs.

Price: $259.00

The 942A is the same as 941A, except it provides two channels of Type-II decoding.

Dimensions: 5.25 in. x 1.5 in. x 9.5 in.

Weight: 0.75 lbs.

Price: $269.00

The 563X reduces the steady-state hiss created by analog tape, guitar signal processors, samplers, digital keyboards and sound effect tapes discriminates between unwanted hiss and desired high frequency signals on sound effect tapes. Stereo strappable.

Dimensions: 1.75 in. x 8.5 in. x 7.25 in.

Weight: 2.5 lbs.

Price: $229.00

The 929 is designed for use in the F900A powered frame systems. Provides two channels of effective single-ended reduction of constant hiss from the output of analog tape recorders; multiple signal processors; noise digital samplers and storage devices.

Dimensions: 5.25 in. x 1.5 in. x 9.5 in.

Weight: 0.75 lbs.

Price: $399.00

DOLBY LABORATORIES INC.

The Model 422 Reference Encoder/Decoder provides four channels of Dolby B-, C- and S-type noise reduction in a 1-U high frame. Contains signal generator providing calibration tones corresponding to selected NR type. Features include overall frequency response of 20 Hz to 15 kHz 1 dB, encode-decode at any level. Electronically balanced input circuits and electronically balanced and floating output circuits.

Dimensions: 1.75 in. x 19 in. x 10.2 in.

Weight: 13 lbs.

Price: not available

The XP SR Series features up to 24 channels of Dolby SR (Cat. No. 431 modules); individual channel bypass; uncal controls and Auto Compare circuitry. Interchange with Cat. No. 331 modules for Dolby A-type noise reduction. Also features overall frequency response of 20 Hz to 20 kHz, 1 dB, encode-decode at any level.

Dimensions: the card frame is 8.75 in. x 19 in. x 18.25 in.; the PS3 power supply is 3.5 in. x 19 in. x 18.75 in.

Weight: the XP 8 is 28 lbs; the XP 16 is 40 lbs.; and the PS3 is 30 lbs.

Prices: the XP 8 is $11,790.00; the XP 16 is $17,800.00; the XP 24 is $22,500.00 the No. 431 Module is $925.00; and the No. 280 Module is $900.00.

The MT Series features up to 24 channels of switchable Dolby SR and A-type noise reduction; software-controlled automatic alignment; flexible assignment of any number of channels to separate groups for multi-machine use; electronically balanced/floating input and output stages and Auto Compare circuitry; overall frequency response of 20 Hz to 20 kHz, 1 dB; and encode-decode at any level.

Dimensions: the card frame, which accommodates up to 24 Cat. No. 445 modules, is 8.75 in. x 19 in. x 19 in.; the PS4 power supply/control unit is 3.5 in. x 19 in. x 19 in.

Weight: 31 lbs.

Prices: $29.775.00; the MT 8 is $14,225.00; the MT 16 is $22,015.00; and the Cat. No. 445 is $1,240.00.

Model 363 features two-channel switchable Dolby SR and a Dolby A-type noise reduction unit with two channels in a 1-U high frame; automatic record/play changeover; built-in Dolby noise/tone generators; auto-compare test facility and transformerless balanced input and output circuits. Basic specifications include overall frequency response of 20 Hz to 20 kHz 1 dB; encode-decode at any level. Available in three versions: Model 363—SR/A (Cat. No. 300) with switchable Dolby SR and Dolby A-type; Model 363—SR (Cat. No. 350) SR only and Model 363—A (Cat. No. 450) A-type only.

Dimensions: 1.75 in. x 19 in. x 10.2 in.

Weight: 14 lbs.

Price: $2,995.00

Model DP501/DP502 Audio Coding Units provide audio at 128 kbits/sec. per channel. Ideal for transmission systems requiring both high audio signal transparency and low, spectrum-efficient data rates. Includes Data rate of 128 kbits/sec. per channel; frequency response of 20 Hz to 15 kHz 0.2 dB; dynamic range greater than 90 dB.

Dimensions: 1.75 in. x 19 in. x 10 in.

Price: $2,990.00

Model SDU4 is designed for reference monitoring of Dolby Stereo or Dolby Surround program material. Accepts two-track matrix-encoded signal as its input and generates four output signals: left, center, right and surround. Overall frequency response is 20 to 20 kHz 1 dB (L, C and R); 100 Hz to 7 kHz 3 dB (surround output).

Dimensions: 1.75 in. x 0.875 in. x 10.25 in.

Weight: 11 lbs.

Price: $2,200.00

PACKBURN ELECTRONICS INC.

The Model 323A Audio Noise Processor has three processors plus other features for optimum noise reduction from all types of disk recordings.

Dimensions: 7 in. x 19 in. x 10 in.

Weight: 18 lbs.

Price: $2,650.00

ROCKTRON CORPORATION

The Hush 8x features eight separate channels of single-ended noise reduction; single rack space; -10 or +4 operation; fast/slow release; balanced or unbalanced four stereo master/slave switches.

Dimensions: 19 in. single space

Price: $799.00

The Pro Hush is MIDI programmable single-ended noise reduction; two channels; 60 dB of noise reduction; complete MIDI control.

Dimensions: 19 in. single space

Price: $749.00

The Hush IICX is stereo and features 60 dB of noise reduction; stereo master; slow/fast release; threshold and sensitivity adjustments; gain reduction and bandwidth filter metering.

Dimensions: 19 in. single space

Price: $439.00

The Hush IIBX is mono, featuring 60 dB of noise reduction; gain reduction and bandwidth filter metering.

Dimensions: 19 in. single space

Price: $329.00

The Hush IIX has 50 dB of noise reduction; slow/fast release; –10, +4 reference switch; gain reduction and bandwidth filter metering.

Dimensions: ½ rack single space

Price: $219.00

The 180A encode/decode tape noise reduction unit features an eight channel encode/decode system; operates at 15 ips to 30 ips; has headroom of +20 dB for use with either +4 dB or –10 dB tape machines.

Dimensions: 19 in. single space

Price: $799.00

RSP TECHNOLOGIES

The Hush 2000 features multiband single-ended noise reduction with variable expander, release and ratio; filter has variable sensitivity and release; has stereo link; stereo link balanced ¼ in. and XLR.

Dimensions: 19 in. single space

Price: $799.00

SYMETRIX INC.

Model 511A is a two-channel or true stereo single-ended noise reduction system. Its dynamic high-frequency filter and downward expander reduce hum, hiss, RF buzz and other noise by up to 30 dB, anywhere in the signal chain.

Dimensions: 1.75 in. x 19 in. x 7 in.

Weight: 9 lbs.

Price: $599.00

MISCELLANEOUS

APHEX SYSTEMS LTD. —See our ad on page 13
Model 250 Aural Exciter Type III is a professional signal enhancer with adjustable harmonics mixing, timbre. Features servo-balanced in/out; relay bypass; and dual NR modes.
Dimensions: 1.75 in. x 19 in. x 9 in.
Price: $995.00
The 9000 Series Modular Signal Processing System features: the 9000R unpowered rack with eleven slots; the 9251 Aural Exciter Module with single channel; 9301 Compellor Module with single channel; the 9611 Expander/Gate Module with single channel; the 9651 Expressor Module with a single channel compressor; the 9901 Parametric Equalizer with three bands and single channel; and the 9000PS power supply.
Dimensions: the 9000R is 5.25 in. x 19 in. x 8.5 in.
Prices: the 9000R is $379.00; the 9251 is $449.00; the 9301 is $549.00; the 9611 is $449.00; the 9651 is $449.00; the 9901 is $449.00; and the 9000PS is $499.00

ARX SYSTEMS —See our ad on page 2
The DDP-1 Dynamics Processor features two channels of total gain control; noise gate; compressor/limiter with variable ratio, threshold, output gain and a peak limiter; full LED indication of input; output; and gain reduction levels. Other features available.
Dimensions: 1.75 in. x 19 in. x 6 in.
Weight: 5 lbs.
Price: to be announced

AUDIO MEDIA RESEARCH —See our ad on Cover III
The CDS 2000 is a two-channel VCA-based compressor/de-esser/limiter/expander. Features independent control of compressor, limiter, de-esser and downward expander threshold; also side chain signal shaping or processing. Electronically balanced inputs/outputs.
Price: to be announced

BROOKE SIREN SYSTEMS, A DIVISION OF AKG ACOUSTICS, INC.
The FDS-310 Sweepable Frequency Dividing System is a two-way stereo or three-way mono, sweepable crossover system built around 24 dB/octave Linkwitz-Riley filters. Internal jumpers change all frequency ranges down in divisions of 10.
Dimensions: 1.75 in. x 19 in. x 8.5 in.
Weight: 7 lbs.
Price: $659.00
The FDS-360 Integrated Frequency Dividing and Limiting System features separate MID filter frequency band limiting; polarity switching; up to 360 degrees of phase correction; auto muting circuit; mono low linking; LEDs for limiting, signal, mutes and modes status; and interchangeable frequency cards.
Dimensions: 1.75 in. x 19 in. x 9 in.
Weight: 9 lbs.
Price: $1,529.00
The TCS-803 Multi Tap Time Corrector is a mono triple-tap digital delay line designed for applications in which full range audio programs have to be delayed for multiple speaker installations.
Dimensions: 1.75 in. x 19 in. x 9 in.
Weight: 10 lbs.
Price: $1,695.00
The TCS-804 Dual Time Corrector is a two-channel dual-tap or single-channel quad-tap digital delay line designed for critical speaker systems displacement and "Delay towers" distance correction in large arenas.
Dimensions: 1.75 in. x 19 in. x 9 in.
Weight: 10 lbs. 5.2 oz.
Price: $2,995.00
The AR-130 Phase Check System frequency range can be selected to match that of the equipment being tested. Detector unit can be connected directly to microphone cable or power amplifier outputs.
Dimensions: 3.94 in. x 2.95 in. 1.5 in.
Weight: 10.6 lbs. excluding batteries
Price: $635.00

dbx, A DIVISION OF AKG ACOUSTICS, INC.
The 263X De-Esser is designed to provide control of problem "ess-filled" vocals. Single-slider action sets the exact amount of sibilance reduction by ear, with visual confirmation from LEDs.
Dimensions: 1.75 in. x 8.5 in. x 7.25 in.
Weight: 2.5 lbs.
Price: $169.00
The 902 De-Esser uses patented dbx sibilance detection circuitry. By comparing the RMS energy of signals above and below a user-selected crossover point, the 902 detects undesirable sibilance regardless of level.
Dimensions: 5.25 in. x 1.5 in. x 9.5 in.
Weight: 0.75 lbs.
Price: $449.00

JBL PROFESSIONAL

The 7942 features 1 in/2 out Digital Delay line; ten microsecond to four second resolution; eighteen bit sigma-delta technology with 64X oversampled convertors; precision calibrated attenuators; and "lockout" protection circuitry.

Price: to be announced

The 7944 is a 2 in/4 out Digital Delay line with ten microsecond to four second resolution; eighteen bit sigma-delta technology with 64X oversampled converters; 422 based remote capability; digital output bitstream (AES/EBU) expansion; precision calibrated attenuators; and "lockout" protection circuitry.

Price: to be announced

ORBAN, A DIVISION OF AKG ACOUSTICS, INC.

The 222A Stereo Enhancer detects/enhances psychoacoustic directional cues present in stereo program material. Increases brightness, impact and definition of music, with no increase in sensitivity to vertical tracing distortion during disc playback.

Dimensions: 1.75 in. x 19 in. x 10.5 in.

Weight: 6.75 lbs.

Price: $975.00

The 245F Stereo Synthesizer creates a pseudo-stereo effect from any mono source. Total mono/stereo compatibility. Saves tracks in multi-track recording situations. Allows for stereo cart transfers with no phasing problems.

Dimensions: 1.75 in. x 19 in. x 9.625 in.

Weight: 7 lbs.

Price: $445.00

The 275A automatic stereo synthesizer works by detecting absence of audio on one channel, or presence of mono in both channels. Smooth cross-fading between true and synthesized stereo; automatic detection and correction of polarity-reversed stereo inputs.

Dimensions: 1.75 in. x 19 in. x 9.625 in.

Weight: 12 lbs.

Price: $2,400.00

The 536A De-Esser features two channels of effective, inaudible de-essing over a 15 dB input range; dual-LED gain reduction metering; dynamic range typically 105 dB; very low distortion, effective RF suppression.

Dimensions: 1.75 in. x 19 in. x 5.75 in.

Weight: 5 lbs.

Price: $650.00

RANE CORPORATION

The DC 24 Dynamic Controller consists of a stereo servo-lock limiter; stereo compressor and stereo expander/noise gate. Also included are a 24 dB/octave crossover, three-pin balanced inputs and outputs, side chain access, slave switch, bypass and gain reduction metering.

Dimensions: 1.75 in. x 19 in. x 5.3 in.

Weight: 5 lbs.

Price: $549.00

ROCKTRON CORPORATION

The Intellifex features 24 bit 164X's oversampling; 100 dB dynamic range; complete MIDI control; 1.5 sec. delay; 8-tap chorus; reverb; 4-voice pitch shift; ducking; mixing functions; and digital Hush.

Dimensions: 19 in. single space

Price: $1,149.00

The RX20 is a stereo exciter/imager; Hush II; frequency/phase mode; mix; ¼ in. unbalanced I/O; input process metering.

Dimensions: 19 in. single rack space

Price: not available

The S212 line mixer has twelve channels. Each channel has level; pan; bass; and treble.

Dimensions: 19 in. single space

Price: not available

RSP TECHNOLOGIES

The 2400 features a multiband enhancer; Hush II; unique sum/difference mode; high and low mix; phase/frequency mode; balanced/unbalanced ¼ in. and XLR I/O; and input process metering.

Dimensions: 19 in. single rack space

Price: not available

SYMETRIX INC.

Model 528 Voice Processor will enhance mic and line levels. Includes a preamp; de-esser; compressor/limiter; downward expander; 3-band parametric EQ and +48 V phantom power for condenser mics.

Dimensions: 1.75 in. x 19 in. x 7.5 in.

Weight: 9 lbs.

Price: $679.00

Model SX206 Multi-Dynamics Processor can operate in compressor/limiter; gate; downward expander; ducker or slave mode. Manual attack and release controls are dynamically sensitive thanks to unique active integrators.

Dimensions: 1.5 in. x 8.2 in. x 6 in.

Weight: 5 lbs.

Price: $329.00

DIGITAL WORKSTATIONS

AKAI PROFESSIONAL

The DD1000 records directly to a 650 Megabyte removable, rewriteable magneto-optical disk cartridge, and multiple drives can be chained to extend recording time. The DD1000 records in stereo or mono, and can output up to two stereo pairs simultaneously. Editing features include: SMPTE-based Q-list; real-time digital crossfades; sequencing of cuts in "Song mode;" and more. Connections include SCSI, SMPTE, RS422, MIDI, digital word-sync, digital audio I/O, and analog audio I/O. The DD1000 is a compact, rack-mountable stand-alone unit, or it can be used with the DL1000 remote unit, as well as an optional Macintosh-based software front end.

Dimensions: 8.75 in. x 19 in. x 16.75 in.

Price: $13,500.00

AKG ACOUSTICS, INC.

The DSE 7000 Digital Sound Editor includes an 8-track digital recorder; ten-input mixer; and two-track digital mixdown deck. Yet when mounted in its optional workstand, it occupies no more space than a standard analog two-track tape machine. DSE 7000 software uses familiar plain English terms such as Cut, Erase and Leader. In combination with the one-button onscreen Help function, the software interface makes the DSE 7000 easy to learn. Also included is a hard disk which automatically "shadows" operations in the background. The hard disk can also be used to store a selection of music beds, tags, sound effects, etc. for rapid retrieval.

Dimensions: the System Unit is 26 in. x 8 in. x 17 in.; the Display Unit is 14 in. x 14 in. x 15 in.; and the Controller is 3 in. x 25 in. x 15 in.

Weight: 129 lbs.

Price: $27,950.00

AMS INDUSTRIES, INC.

The AMS LOGIC 1 is a dynamically automated mixing and editing console integrated with the AudioFile PLUS hard disk editor. The system has fourteen channel strips providing inputs to the AudioFile PLUS during recording, and playback from the AudioFile PLUS during mixing and dubbing. The optional two-layer configuration doubles the number of available inputs. Logic 1 also incorporates four-band EQ filters, dynamics and auxiliaries.

Dimensions: 37 in. x 50.5 in. x 37 in.

Price: available upon request

The 16-output AudioFile PLUS has analog and digital inputs and outputs; ADR software; remote machine control and powerful editing functions. Software to process and auto-conform video-edit decision lists, combined with machine control functions, allow the system to offer an unparalleled degree of automation when conforming original source material and final edits. AudioFile PLUS has the latest DSP processors to provide scrub-editing and "Timeflex" real-time compression and expansion. The system comes with a minimum of four hours of audio storage, expandable in four-hour blocks.

Price: available upon request

AQ DESIGN, INC.

The AQ1 Digital Audio Editorial System allows the editing and SMPTE timecode synchronization of stereo sound files recorded to hard disk. The digital I/O option allows DAT material to be loaded, edited, sequenced, and transferred back or archived, without leaving the digital domain. Selectable sampling rates, along with time displays in timecode and footage/frame, tailor the system for any application. Timecode and MIDI interfaces are built in. An IBM-compatible computer, the AQ1 can also run other DOS applications such as sequencers and databases. Color graphics waveform display and real-time recording to removable magneto-optical cartridges round out the capabilities of the AQ1.

Price: Systems start at $7,999.00 for one hour of stereo recording to hard disk.

DIGIDESIGN

The Pro Tools Multi-track Digital Recording and Editing System for the Macintosh II features graphic non-destructive editing of multiple tracks of audio and MIDI with track slipping, region trimming and crossfades; from four to sixteen independent channels of recording/playback; balanced XLR analog I/O, AES/EBU and S/PDIF digital I/O; virtual tracks; real time EQ and digital effects (completely automatable); dynamic and state-based automation with instant update; MIDI recording, playback and editing; and SMPTE synchronization.

Price: complete systems start at $5,995.00

The Sound Tools Digital Recording and Editing System for the Macintosh II consists of the Analog Interface analog-to-digital converter; the Sound Accelerator digital signal processing card and the Sound Designer II audio editing Software. Options include the Digital Interface; Pro I/O Professional Analog Interface; and the Pro Store series of 660 megabyte, 1 gigabyte or magneto optical hard disk drives. Features include 16 bit, 44.1 kHz stereo direct-to-disk digital record and playback; non-destructive playlist editing; real-time dynamics compression/expansion/noise gate; stereo time compression/expansion; pitch shift with time correction; 2:1 or 4:1 data compression options; real-time parametric/graphic EQ; continuous SMPTE resynchronization; and sample editing/transfer.

Price: $3,285.00

The Yamaha PDS CD-Recorder is for Pro Tools and Sound Tools. Controlled by Digidesign's Master List PDS software, you can record CD-quality digital audio directly from the AES/EBU digital outputs or your Digidesign system to the Yamaha PDS encoding and recording units—with complete control over song sequence, start times, and CD subcode parameters. Features include writes Red Book standard CDs; front-loading; supports track sequencing; track spacing and track volume; supports index points, markers, ISRC codes and catalog codes; supports 3 in. and 5 in. write-once media; and maximum recording time of sixty-five minutes.

Price: $24,995.00

DIGITAL AUDIO LABS

The CardD System, a hard-disk recording and editing system, offers sound quality and many editing features. The CardD plugs into an IBM AT-compatible and has stereo analog inputs and outputs. The EdDitor program does on-destructive stereo waveform editing. Features include cut; paste; mix; fade; crossfade; and multiple undo. For direct digital transfers between hard disk and DAT, add the optional I/O CardD. Among the many uses of CardD System are stereo mastering; creating remixes; and editing Digital Audio Tapes.

Prices: The CardD is $795.00; EdDitor is $250.00; I/O CardD is $295.00; and The Developer's Software Toolkit is $500.00.

DIGITAL AUDIO RESEARCH LIMITED

The SoundStation SIGMA is a completely integrated digital audio production environment which offers Segment Based Processing and DSP functions along with a host of features. In addition to DSP power giving four-band parametric EQ on every segment in real-time, SIGMA's comprehensive facilities include eight or sixteen channels of simultaneous analog or digital recording/playback with full varispeed operation; built-in rewriteable Optical Disk storage; all SoundStation edit features; and an advanced CPU for fast touch-screen response; parameter change and edit execution. WordFit, DAR's unique automatic dialogue replacement system, is optionally available for all SoundStation SIGMA Systems.

Dimensions: Control console is 25 in. x 8.75 in. x 16.625 in.; Processor and storage is 19 in. x 15.625 in. x 24 in.

Price: from around $100,000.00

The SoundStation II multi-channel digital audio editing and production system is available in two models. The entry-level four or eight channel SoundStation II system is the workstation which provides a professional alternative to PC or mouse-based machines. This pre-configured SoundStation II offers comprehensive facilities for fast and easy tape-like recording/editing utilizing DAR's touch-screen and dedicated controls. As working requirements grow, this basic system can be fully upgraded with DAR's wide range of more advanced editing features.

Dimensions: Control console is 25 in. x 8.75 in. x 16.625 in.; Processor and storage is 19 in. x 15.625 in. x 24 in.

Price: from around $40,000.00

The mid-range eight and sixteen channel SoundStation II systems have been newly engineered with enhanced processing power. Ideally suited for most video and film post production applications, this flexibly configured SoundStation II incorporates full-featured recording/editing, up to eight track hours of storage, with such edit functions as Time Warp time compression/expansion and full machine control capabilities. DAR's rewriteable Optical Disk sub-system and WordFit automatic dialogue synchronization are among the options available for system upgrade.

Dimensions: Control console is 25 in. x 8.75 in. x 16.625 in.; Processor and storage is 19 in. x 15.625 in. x 24 in.

Price: from around $80,000.00

DOREMI LABORATORIES, INC.

The Macintosh-based DAWN (Digital Audio Workstation Nucleus) offers eight simultaneous analog inputs, eight outputs and eight tracks of hard disk recording; a 16/24-track version is also available. The DAWN unit resolves to SMPTE time code or House Sync and is capable of operating with both Longitudinal or vertical interval time code (LTC or VITC). It also has a stereo digital input and output (AES/EBU or SPDIF). CMX-format EDLs can be imported or generated by the system, and automatically executed to conform audio to timecode events. In addition, direct remote control is now available from many systems.

Dimensions: 5.25 in. x 19 in. x 14 in.

Price: from $15,000.00 to $18,000.00

E-MU SYSTEMS, INC.

The Emulator III Digital Sound Production System is designed to be a musical instrument, audio post production workstation, and a digital effects processor in one system. Features stereo sampling; sixteen voices; sixteen bit linear data format (30 kHz and 44.1 kHz sample rates); variable sample rate pitch-shifting with 2x oversampling (for virtually distortion free transposition); up to 8 Mbytes internal RAM; and up to 135 seconds sampling time. A 40 Mbyte internal hard disk drive is standard; three envelope generators and one multi-wave LFO per channel; advanced 16-track sequencer with cut and paste; SMPTE cue list sequencing; and parametric quantization for a very human feel. SCSI, MIDI and RS-422 interfaces. System expansion options also included.

Price: $9,995.00

The Emax II 16-Bit Digital Sound System has 32 channels; is configured as 16 stereo channels or 16 stereo chorused channels; 16-bit linear quantization; up to 8 Mb sample memory (209 sec @ 20 kHz); five sample rates from 20 kHz-39 kHz; compatible with Emax 1 library; truncation; splicing; crossfade looping; advanced digital signal processing; full function additive synthesizer; backwards mode; 32 dynamic digital filters w/resonance; 32 DCAs; (32) 5-stage envelopes; 16 LFOs; programmable panning; 61 velocity sensitive keys (Keyboard version); 122 samples per preset; 8 programmable polyphonic outputs (4 stereo); preset stack mode for up to four presets on one key; 16-track sequencer; arpeggiator; 3.5 in. disk storage; CD-ROM and R/W optical storage compatible; RS422; MIDI and SCSI interfaces; rack mount and hard disk versions available.

Price: with 1 Mb RAM is $3,495.00

FAIRLIGHT

The MFX2 features continuous 16-track operation from one to six 1.2 Gbyte disk drives using a new high speed "Turbo SCSI" bus. File backup/restore uses a new double-speed 5 Gbyte 8mm tape drive. The new MFX upgrade also includes double resolution screen graphics featuring clip and track names as well as waveforms, fades, track level and status indication. Disk drives can be backed up at a speed five times faster than real-time with a 5 Gbyte 8 mm data tape drive using data cartridges. A rewriteable optical disk drive is also available which enables stereo recording directly to its removable disk while several tracks play back. It can also be used for instant access sound archiving.

Dimensions: 363 mm x 355 mm x 406 mm

Weight: 104 kg

Price: not available

HYBRID ARTS, INC.

The ADAP II Digital Audio Recorder and Editor can handle tasks including DAT editing; CD pre-mastering; sound effects design and spotting; dialog editing and conforming; ADR and music editing.

Price: from $10,000.00 to $17,000.00

The ADAP II Portable has all the functions of the ADAP II Turnkey system, all condensed into one small unit to go from the recording studio to the Film Dubbing Stage.

Price: from $12,000.00 to $17,000.00

The ADAP IV has four discrete inputs and outputs and a minimum of thirty-two track files of unlimited mixing of tracks. Our custom sound file system features waveform display and non-waveform display with non-destructive editing. Other features include management of sound databases, including search functions; multi-device volumes; magnetic or removable optical; four channel digital i/o; Chase/lock to all kinds of SMPTE; Jog/shuttle scrubbing; instant-locate; punch in/out; DAT back-up; import/export files in other formats; Sound Designer; real-time and off-line digital filtering and mixing; user-customizable crossfades; multi-level undo/re-do; unlimited number of editing functions.

Price: to be announced

LEXICON, INC.

The OPUS Digital Audio Workstation is the only fully integrated random access digital audio production system. It provides the functions and controls of a multi-track tape machine, a 12-channel mixing console with digital EQ, and a random-access editing system in an environment designed to be comfortable for engineers familiar with traditional analog consoles. Program material is recorded on up to four hard disks for 4.8 Gbytes (790 track minutes) or audio storage. An 8-mm helical-scan tape transport provides archival storage.

Price: $185,000.00

The OPUS/e Digital Audio Editing System is designed to bring its editing capabilities to existing rooms, or to act as a remote editor in conjunction with an OPUS system. OPUS/e provides the advantages of random access digital editing, retrieval and storage. It interfaces with any combination of digital and analog signals simultaneously and includes an 8 x 2 stereo digital mixer. Up to four hard disks can be used to store 4.8 Gbytes (790 track minutes) or audio information, and archived storage is maintained on 8-mm tape.

Price: $120,000.00

MICRO TECHNOLOGY UNLIMITED

The Windows 3 based MicroEditor provides direct-disk recording of 2/4 tracks from 8 k - 48 kHz rates at –110 dB typ. noise floor; random access in all files instantly; create Mix files three stereo hours long with 1000+ audio segments (individual fade/gain) drawn from over twenty recorded files; up to thirty-eight internal tracks can overlap to play simultaneously at any time; position segments to magnetic gridlines to where they play in the mix; copy and all delete edits are done (clickless) and undone instantly any time; zoom editing to sample precision; SMPTE/MIDI interfacing and chase-lock.

Prices: MicroSound for your computer from $2,690.00 to $3,890.00, or eighteen ready-to-use models including computers from $7,600.00 to $11,450.00.

NEW ENGLAND DIGITAL

Based on a multiple 68020/56000 processors and incorporating multiple 32-bit data buses, the DSP Option offers on-board mixing capabilities including 5-band parametric EQ with switchable shelving; panning; summing; gain control; and digitai crossfades. The 32, 44.1, 48 and 50 kHz and drop frame sampling rates are supported. I/Os options include eight channel 18-bit D/As; SDIF-2 and SDIF-M modules Mac-based AudiMation mixing software which may be controlled from any MIDI fader module; provides eight or sixteen channel strips with processing available on all channels simultaneously; plus four mono sends and returns configurable as stereo; and four mix buses.

Price: starts at $35,000.00

SoundDroid Off-line allows spotting of effects, dialogue and foley from any Apple Macintosh II series computer as well as the creation and printing of cuesheets, ADR scripts, and "to be recorded" lists. Powerful project management features and on-line report generating capabilities help control costs. Cuesheets created with SoundDroid Off-line can be uploaded onto PostPro SD multi-track, running SoundDroid On-line software, for editing of production sound, recording of Foley and ADR, and placement of library effects.

Price: $1,500.00

The MultiArc Audio Editing Software family provides a Macintosh-based interface for the Synclavier, PostPro and DSP Option systems.

A new Record Panel which allows the user to switch between random access and conventional multi-track style recording will be featured, as well as EditView, a tape-style EDL editor for RAM samples and disk cues, and TransferMation, a librarian which catalogues on-line and off-line sounds.

Price: $4,000.00

New sound libraries are available in WORM and Magneto-Optical formats: The Hollywood Edge provides over 1,500 digitally recorded foreground, background and foley sound effects. The Bob Clearmountain Library is a comprehensive drum, percussion and electric bass library, and The Works, from Sonic Boon, is an encyclopedic collection of machinery sound effects.

Price: Hollywood Edge is $2,300.00; the Bob Clearmountain Library is $3,550.00; The Works not priced as of press time.

OPCODE SYSTEMS INC.

The Studio Vision Integrated MIDI and Digital Audio Recording and Editing Software System combines Opcode's professional Macintosh MIDI sequencing software program, Vision, with the ability to record and edit digital audio. This capability allows, for example, the user to add vocals, guitar, saxophone, or any live instrument or voice to MIDI sequences. Studio Vision works with Digidesign's Pro Tools, Sound Tools or Audiomedia card and allows playback of four digitai audio channels simultaneously with Pro Tools (expandable to sixteen) or two with Sound Tools or Audiomedia. The program incorporates extensive, non-destructive editing of the digital audio in the same environment as MIDI.

Prices: Studio Vision software is $995.00; Pro Tools is $5,995.00; Sound Tools is $3,295.00; Audiomedia is $995.00; and the Studio Vision/Audiomedia bundle is $1,995.00.

OTARI CORPORATION

The DDR-10 is a Macintosh-based, two-channel hard disk recorder that comes standard with a IIci/25 MHz CPU; 345 Mbyte hard-disk drive; an integrated hard-disk frame with accommodations for three additional drives; integrated audio monitoring; extensive editing capabilities; 19 in. high resolution monitor and Otari's control panel surface. This panel gives an operator familiar with audio tape recorders instant familiarity with the DDR-10. Optional hard disks provide increments of thirty, sixty or one hundred minutes of storage. Standard DSP functions include sample rate conversion; graphic EQ; parametric EQ; time comp. exp; pitch shift; mixing; merging and soundfile FFT analysis.

Price: $19,990.00 complete

The PD-464 multi-track hard disk recorder system may be configured in four-track increments to sixty-four simultaneous inputs and outputs mimicking the operation of analog reel to reel recorders. Unlimited crossfade on edits, track slipping, auto punch in/out, individual track alignments, graphical waveform editing and unlimited non-destructive edits provide powerful control over your audio files. Locking to time code is sample accurate even across hard disks. Backups and restorations are completed at 3X real time with the standard 8 mm tape drive. Other options include digital I/O in a variety of formats, DSP, storage time extension and "field" system expansion.

Prices: the 4-track is $31,950.00; 12-track is $62,295.00; 24-track is $115,350.00; and 64-track is $299,950.00.

ROLAND PRO AUDIO/VIDEO GROUP

The DM-80 Multi-track Hard Disk Recorder comes in four and eight track configurations, and is completely self-contained including a 24 bit digital mixer (featuring internally programmable snapshots and external dynamic control), analog and AES/EBU-standard digital inputs and outputs, 48, 44.1 and 32 kHz sample rates and can act as either a master or slave using SMPTE, MTC or MIDI timing formats. The DM-80 may be controlled by the DM-80-R Remote Controller or Roland's Track Manager software for the Macintosh. Both feature non-destructive random access, cut and paste editing. Track Manager also allows multiple units to be controlled as one and offers advanced editing and display features, with an emphasis on post-production applications.

Dimensions: 7.375 in. x 18.875 in. x 15.75 in.

Prices: the DM-80-4 4-track version is $6,995.00; the DM-80-8 8-track version is $9,995.00; the DM-80-R Remote Controller is $1,895.00; the DM-80-F optional Fader unit is $1,295.00; the DM-80-S Track Manager Software is $650.00; and the IB-1 MIDI/DM Buss Interface Box is $99.95.

SOLID STATE LOGIC

Screensound is a fully-integrated digital audio-for-video editor/mixer that provides full digital soundtrack assembly. Up to seven ScreenSounds can be linked, offering central mass storage of audio, off-line back-up, restore and more. New features include a public domain interface that allows third-party users to create custom ScreenSound extensions; EDL Scan which takes CMX EDL files and imports them into the ScreenSound Desk; autoconform, which controls external machines to load program material with reference to source timecode; time compression and expansion of audio clips, while maintaining pitch; variable control of crossfade edits and rehearsal of In and Out points and crossfade rates; and new Magneto-Optical drive storage.

Price: depends upon system configuration

SONIC SOLUTIONS

The SonicStation is a Mac-based random-access workstation with two input and output channels, playback, editing and mixing of multiple independent channels from disk. Audio editing uses waveform displays with high-speed zoom out to any level. All edits and crossfades are done in real-time and are totally nondestructive. Includes mixing, real-time shelving and presence filters, and dynamics processing. Background transfer saves time by letting the user edit audio even while the sound is being loaded. Expandable to four, eight, sixteen or twenty-four tracks.

Price: starts around $6,000.00, exclusive of computer, converters and sound disks.

The Sonic System 4-Channel Editor/Mixer is a complete workstation, including high-speed graphic editing, mixing and signal processing, 24-bit resolution. All edits are non-destructive, and crossfades, EQ and mixing are all done in real-time. Edit changes and multiple versions can be made, and all changes can be auditioned immediately. As many as eight filter sections on an audio channel, with over twenty filter types, and filter resolution down to one-half hertz. Built-in 4-to-2 mixing desk features real-time dynamic automation of all parameters, including EQ. All dump and load functions are done in background, with machine control, while work continues in the foreground. System is expandable.

Price: starts at $26,000.00, exclusive of computer, converters and sound disks.

SOUNDTRACKER PTY, LTD.

The portable Soundtracker unit has four or eight inputs/outputs and is not expandable. The multi-user allows up to six different operators with different sync references to share disk drives, databases and up to twenty-four inputs and outputs. Recording and editing can be done on the same screen or separate screens. Work is organized into jobs, subjects and reels and there is an internal 8 into 2 mixer. Each cue on any of the 8-tracks can have automated level and pan. Project management can be organized by the off-line spotting system provided.

Price: not available

STUDER REVOX

The Studer Dyaxis' multi-format digital I/O supports virtually any input source and includes digital effects loop. The System Synchronizer offers LTC and VITC reader/generators, MIDI time code output, film tach and video sync reference. Just released is the new MacMix Version 3.2 software upgrade and removable Magneto Optical Drive. Other backup storage options include the 1.2 Gigabyte integrated R-DAT. Other features include digital EQ; an unlimited number of internal tracks; "snapshot" automation; mixing and editing capabilities; and superior sound quality. With the Dyaxis workstation, complete individual control of each sound file is maintained in a totally nondestructive editing environment leaving master recordings fully intact.

Dimensions: standard 19 in. rack mount

Price: starts at $12,995.00

TURTLE BEACH SYSTEMS

The 56K digital recording system features EDL-type playlist editing; cut/copy/paste editing; stereo mix; sample rate conversion; SMPTE chase/lock; four band parametric equalizer; and configurable crossfades. These are just some of the tools included. The systems requires an IBM 286 12 Mhz or faster, a large hard disk, and a DAT machine.

Price: $1,995.00

MultiSound/Wave for Windows is a 16 bit card offering two tracks of compact disc-quality digital audio, as well as the MIDI output section of an E-mu Proteus for all of the MIDI instruments. This is a professional quality audio presentation all on one card. Wave for Windows is a device independent editing software for Microsoft Windows 3.0 and Multimedia Extensions 1/0. Both products are MPC compatible.

Prices: the MultiSound is $995.00; Wave for Windows is $149.00

WAVEFRAME CORPORATION

The AudioFrame can be specifically configured for a variety of professional audio applications such as post-production, mastering, recording, editing and composing music.

Dimensions: 17.5 in. x 19 in. x 22 in.

Price: from $45,000.00 to $150,000.00

The CyberFrame is available in two renditions, (1) CyberFrame Editorial, a comprehensive digital audio workstation specifically designed for professional sound editors working on feature films and episodic television, and (2) CyberFrame Multi-track, an eight-channel disk recorder and editor for more generic applications.

Dimensions: 9 in. x 19 in. x 16 in.

Price: from $40,000.00 to $100,000.00

GET THE FACTS!

Please use this card to receive information from the manufacturers.

CIRCLE ONLY NUMBERS CORRESPONDING TO THOSE IN THIS ISSUE!

If you circle every number on the card, or put one large circle around the whole card, the processing company automatically discards your card.

Please also remember to put your name and address on the card.

Get db the Sound Engineering Magazine.

The one magazine for ALL your needs, large or small.

NOTICE

Please enclose check or money order (US funds, only) with the subscription card **WE WILL NOT BILL YOU** For credit card orders, please fill in the account number, expiration date and your signature Thank you

Free Information from db

NOVEMBER/DECEMBER 1991 EXPIRES 2/28/92

To receive manufacturer's information about products advertised or editorially mentioned in this issue Circle the numbers on the card to correspond with the Reader Service number on the bottom of the advertisement or editorial that interests you

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
16	17	18	19	20	21	22	23	24	25	26	27	28	29	30
31	32	33	34	35	36	37	38	39	40	41	42	43	44	45
46	47	48	49	50	51	52	53	54	55	56	57	58	59	60
61	62	63	64	65	66	67	68	69	70	71	72	73	74	75

Please Print Carefully

Name ______________________

Company Name ______________________

Address ______________________

City ______________ State ______________ Zip Code ______________

Valid for four months after cover date

Free Information from db

NOVEMBER/DECEMBER 1991 EXPIRES 2/28/92

To receive manufacturer's information about products advertised or editorially mentioned in this issue Circle the numbers on the card to correspond with the Reader Service number on the bottom of the advertisement or editorial that interests you

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
16	17	18	19	20	21	22	23	24	25	26	27	28	29	30
31	32	33	34	35	36	37	38	39	40	41	42	43	44	45
46	47	48	49	50	51	52	53	54	55	56	57	58	59	60
61	62	63	64	65	66	67	68	69	70	71	72	73	74	75

Please Print Carefully

Name ______________________

Company Name ______________________

Address ______________________

City ______________ State ______________ Zip Code ______________

Valid for four months after cover date

NOVEMBER/DECEMBER 1991

Subscribe to db Magazine—*Six Big Issues Each Year*

CHARGE TO ☐ **MASTERCARD** ☐ **VISA**

My account number is ________/________/________/________

Exp date ______________

Signature ______________________

Name ______________________ Title ______________

Company Name ______________________

Street Address ______________________

City ______________________ State ______ Zip ______________

I'm in: (Please check off your professional status.)

☐ I'm an owner. ☐ Engineer ☐ Manufacturers R&D/Sales support

☐ Other ______________

and I'm in (check all those below that apply.)

☐ Studio Recording ☐ Film or TV Post Production ☐ Broadcast Audio

☐ Touring Sound ☐ Fixed Sound Reinforcement ☐ Church Sound

☐ Duplication ☐ Other (please explain) ______________________

CHECK ONE:

☐ **RENEWAL**

☐ **NEW SUBSCRIBER**

RATES FOR U.S.

☐ $15.00 for 1 year

☐ $24.00 for 2 years

☐ $30.00 for 3 years

CANADA/MEXICO

☐ $16.00 for 1 year

☐ $25.00 for 2 years

☐ $31.00 for 3 years

OTHER FOREIGN

☐ $28.00 for 1 year

☐ $46.00 for 2 years

☐ $55.00 for 3 years

PLEASE NOTE: WE CAN NOT BILL

NO POSTAGE
NECESSARY
IF MAILED
IN THE
UNITED STATES

BUSINESS REPLY MAIL

FIRST CLASS MAIL PERMIT NO. 47 DEER PARK, NY

POSTAGE WILL BE PAID BY ADDRESSEE

MARDATA
PO BOX 821
DEER PARK NY 11729-9862

NO POSTAGE
NECESSARY
IF MAILED
IN THE
UNITED STATES

BUSINESS REPLY MAIL

FIRST CLASS MAIL PERMIT NO. 47 DEER PARK, NY

POSTAGE WILL BE PAID BY ADDRESSEE

MARDATA
PO BOX 821
DEER PARK NY 11729-9862

NO POSTAGE
NECESSARY
IF MAILED
IN THE
UNITED STATES

BUSINESS REPLY MAIL

FIRST CLASS MAIL PERMIT NO. 48 COMMACK, NY

POSTAGE WILL BE PAID BY ADDRESSEE

SAGAMORE PUBLISHING CO., INC.
203 Commack Road, Suite 1010
Commack, NY 11725-9855

ADDRESSES

Akai Professional
1316 East Lancaster
Fort Worth, TX 76116

AKG Acoustics, Inc.
1525 Alvarado Street
San Leandro, CA 94577

Alesis Studio Electronics
3630 Holdrege Avenue
Los Angeles, CA 90016

AMS Industries, Inc.
7 Parklawn Drive
Bethel, CT 06801

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

AQ Design, Inc.
902 Broadway
Suite 1400
New York, NY 10010

ARX Systems (USA)
P.O. Box 842
Silverado, CA 92676

Ashly Audio, Inc.
100 Fernwood Avenue
Rochester, NY 14621

Audio Media Research
711 A Street
Meridian, MS 39301

Brooke Siren Systems, A Division of AKG Acoustics, Inc.
1525 Alvarado Street
San Leandro, CA 94577

dbx, A Division of AKG Acoustics, Inc.
1525 Alvarado Street
San Leandro, CA 94577

Digidesign
1360 Willow Road #101
Menlo Park, CA 94025

Digital Audio Labs, Inc.
6311 Wayzata Boulevard
Suite 330
St. Louis Park, MN 55416

Digital Audio Research Ltd.
2 Silverglade Business Park
Leatherhead Road
Chessington, Surrey KT9 2QL
England

DOD/DigiTech/Audio Logic
5639 South Riley Lane
Salt Lake City, UT 84107

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

Doremi Laboratories, Inc.
4927 North Glen Arden Avenue
Covina, CA 91724

E-mu Systems, Inc.
1600 Green Hills Road
Scotts Valley, CA 95066

Fairlight ESP Pty. Limited
30 Bay Street Broadway
NSW 2007 Australia

Furman Sound, Inc.
30 Rich Street
Greenbrae, CA 94904

Hybrid Arts, Inc.
8522 National Boulevard
Los Angeles, CA 90232

JBL Professional
P.O. Box 2200
8500 Balboa Boulevard
Northridge, CA 91329

Lexicon, Inc.
100 Beaver Street
Waltham, MA 02154

LT Sound
7980 LT Parkway
Lithonia, GA 30058

Micro Technology Unlimited
156 Wind Chime Court
P.O. Box 21061
Raleigh, NC 27619

New England Digital
Rivermill Commercial Center
Lebanon, NH 03766

Opcode Systems Inc.
3641 Haven Drive
Suite A
Menlo Park, CA 94025

Orban, A Division of AKG Acoustics, Inc.
1525 Alvarado Street
San Leandro, CA 94577

Otari Corporation
378 Vintage Park Drive
Foster City, CA 94404

Packburn Electronics, Inc.
P.O. Box 335
Dewitt, NY 13214

Peavey Electronics Corporation
711 A Street
Meridian, MS 39301

Rane Corporation
10802 47th Avenue West
Mukilteo, WA 98275-5098

Rocktron Corporation
2870 Technology Drive
Rochester Hills, MI 48309

Roland Pro Audio/Video Group
Roland Corporation US
7200 Dominion Circle
Los Angeles, CA 90040

Solid State Logic
320 West 46th Street
Second Floor
New York, NY 10036

Sonic Solutions
1902 Van Ness Avenue
Suite 300
San Francisco, CA 94109

SoundTech
230 Lexington Drive
Buffalo Grove, IL 60089

Soundtracker Pty, Ltd.
169 Bank Street
South Melbourne, Victoria, 3205
Australia

Studer Revox America, Inc.
1425 Elm Hill Pike
Nashville, TN 37210

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

Turtle Beach Systems
P.O. Box 5074
York, PA 17405

WaveFrame Corporation
4730 Woodman Avenue, Suite 405
Sherman Oaks, CA 91423

MARK MAULUCCI

Producer's Notebook Part I

● In this and the next issue, I will detail the anatomy of a production project from beginning to end. Hopefully, it will be of interest to all of you budding producers out there. Even if you don't aspire to produce others, perhaps it will inspire you to produce your own best work.

A producer is to music what a director is to film. Both of these individuals oversee a project, drawing the best performances possible from the artists while coordinating the technical work of others to make the finished project a cohesive whole.

The artist is the actor, interpreting the song or script. All films start from a strong script and music is no different—the song is the thing. Actors and musicians are always seeking strong material. How can they express themselves without it? To take this film analogy a bit further: the engineer of a recording session is like a cinematographer (director of photography). This individual actually "photographs" the sound. The experienced engineer knows about different microphones (cameras) and placement techniques and understands EQ and signal processing (lighting and color).

In smaller productions, the music producer will often engineer just as the film director will shoot the picture. Music production is a great way to bone up on our engineering chops. Most of us aspiring producers own recording studios. Yes, even four-track recorders count! In trying to capture our music on tape over the years, we've actually acquired some pretty good engineering techniques. This is important when we advance to the next level so we can communicate intelligently and effectively with the other technicians in larger productions.

Often, in order to remain objective in the overview, the music producer wants someone else turning the knobs so he or she can focus on performance and content the way a director can focus on the actors. Famed director and screenwriter John Huston was legendary in how he could frame and set up a shot. And yet, when he called "action," he usually listened to the actors with his eyes closed! This surprised newcomers to the set, especially after he yelled "cut and print" having not "seen" the performance. But of course he did, by listening. He already knew what it looked like.

Now that we understand the role of the producer, let's take a look at the project I'll be discussing. My friend Becky Williams is a very talented singer and songwriter. I've heard her perform several times over the last few years and was moved by her songs. Consequently, I made a mental note that I would like to work with her some day. This brings up an important point: you should really like the music of the artist you'll be working with since you're going to be spending a lot of time with this person. Initially, an artist's music should impassion or excite you; you should want to produce it. It should feel right and suggest ideas to you in an obvious and natural way. It helps to like the individual as well as their music, although you may not really get to know the performer until hours into their project. I'm one of those people who has to believe in what I'm doing to be good at it—my heart has to be in it. That's why I suggest you work with artists you respect or admire.

Anyway, Becky, a folk singer (for lack of a better term) in the greater-Boston area, was in a bit of a quandary. She had been involved with an album project that was never completed and had talked to a couple of established producers, but scheduling and money were obstacles. She didn't know if she should finance and distribute her own album. I'd heard some of her demos and felt she had never really been captured on tape, so I suggested she record a few songs in my studio. Just have a positive experience—have fun. At the very least, she'd have a good demo for procuring more coffeehouse work. She'd have a tape to shop at various labels or we'd have twenty-five percent of an album in the can.

A few weeks later, Becky came over to my place to check out my studio. I played for her a demo I did for a vocalist/guitarist duo and she liked it. It obviously helped to have something in a similar style to play for her. Most of my work is in the pop, rock or blues category, although my last production for someone else was an instrumental, light jazz thing. I played a couple of my own acoustic guitar pieces. Becky is very concerned with the recording of her acoustic guitar and hasn't been satisfied with the past results. I then played an elaborate dance production which featured the aforementioned woman vocalist, just to show what the studio was capable of, and finished with a couple of my band's songs. She liked the vibes in my studio.

Becky was going to prepare a simple vocal/guitar tape of six to eight songs so I could "live" with them awhile and choose the three I thought were strongest. I suggested we repair to my kitchen. During a dinner of linguine with white clam sauce, she played and sang about seven songs. I had my boombox recording on the counter and amidst the chopping of garlic, we talked about each song after she performed it. I was familiar with several of the songs, but there were three that leapt out at me, and I was excited as we discussed some of my ideas.

There are some producers who have a "stamp" so that every artist they produce has their sound. Phil Spector is the first and most famous of these producers. All of his productions bore his "wall of sound." Todd Rundgren is a contemporary producer who seems to color all of his projects with his sound. I like to let the artist shine through. My job is to *transfer* the artist's vision to tape, not *transform.* I may have strong opinions about which songs to use, their arrangement and a lyric or chord change, but in the end, it is the artist's intent that I seek to capture—the essence that grabbed me in the first place. I believe it's the producer's job to make the artist comfortable enough to feel that everything else is taken care of. Just ply your art—sing and play your heart out.

Next time: Producer's Notebook Part II—The Anatomy of the Recording Session. db

New Products

HYPERCARDIOID

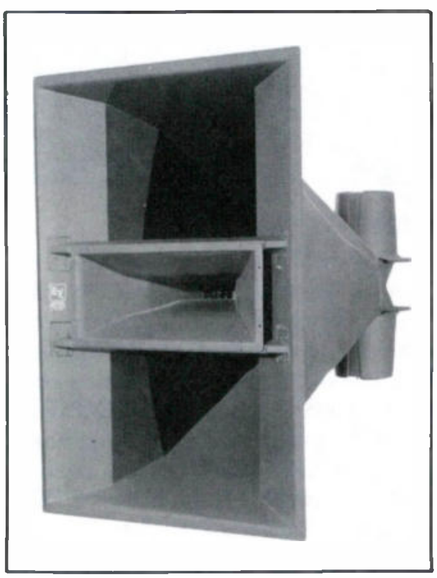
• The PRO 10HE features a neodymium magnet for high-energy output, a double-dome diaphragm for extended high frequency response, floating diaphragm (patent pending) to minimize handling noise, and a voice coil wound with Copper Clad Aluminum Wire for low mass with high output. The hypercardioid polar pattern of the PRO 10HE ensures optimum gain before feedback, and provides isolation between artists during performances and while recording. It offers a 50 to 15,000 Hz frequency response and a sensitivity of –56.4 dBm.

Manufacturer: Audio-Technica U.S., Inc.
Price: $209.00

Circle 60 on Reader Service Card

LOW-FREQUENCY SYSTEM

• The MH6040 wide-range horn-and-driver system is designed to cover a frequency range of 100-4,000 Hz, with a sixty degree by 40 degree coverage pattern. To extend output to 20,000 Hz, a HP640 horn may be coaxially mounted within the MH6040's 39 in. x 59 in. mouth. The MH6040 is equipped with two DL10XWP drivers mounted in a manifold configuration, with each driver having a long-term power capacity of 300 watts. Manifolding allows the output of two or more drivers to be summed without the interference problems that commonly plague "Y" throats or multiple sources aims in the same direction. In addition, manifolding builds redundancy into the system. In the unlikely event of failure of one driver, the other is still available.



Manufacturer: Electro-Voice
Price: $3,165.00

Circle 62 on Reader Service Card

HANGING MIC

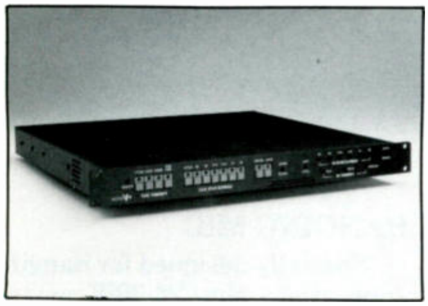
• Specially designed for hanging applications, the CM-30W mic is a miniature, supercardioid electret condenser unit. The CM-30W is operable with 12-48V of phantom power, and incorporates electronics which attach to a rectangular or circular electrical plate for quick installation into any ceiling electrical box. To eliminate the need for connectors, screw terminals have been provided at the input and output of the electronics. Featuring a sensitivity of 13.5 mV/Pa and an impedance of 150 ohms (balanced), the mic is also equipped with a strain relief which can be used to securely clamp the unit's cable at the desired length.

Manufacturer: Crown International, Inc.
Price: $215.00

Circle 61 on Reader Service Card

DIGITAL EFFECTS

• The RSP-550 contains effects including reverb, multi-tap delay, enhancer, equalization, flanging, and phasing as well as two new kinds of chorus effects, multi-band pitch shifting, rotating speaker effects and vocoder. The RSP-550 delivers a dynamic range of 95 dB, a frequency response of 10 Hz to 21 kHz



and a THD of 0.02 percent or less. All signal processing is conducted at a sampling rate of 48 kHz, with fully independent 16-bit A/D and D/A converters for each channel to provide true stereo processing capabilities as well as the ability to process independent multi-effects for each channel. Also featured are 39 effects algorithms with a comprehensive selection of editable parameters, enabling you to create or modify effects. A wide variety of reverb effects are available, as are other features and effects.
Manufacturer: Roland Corporation US
Price: $1,295.00

Circle 63 on Reader Service Card

PRO R-DAT

• The SV-3700 Pro-DAT recorder features a front-panel shuttle wheel, with 0.5 to 15 times speed range. In addition, it features a horizontal cassette tray for easier tape loading; Program, Absolute and time-remaining displays; push-button selection of 44.1/48 kHz sampling rates, via either analog or digital inputs; push-button fade-in and fade-out functions, for automatic level-change ramps at the start and end of a recording; balanced inputs and outputs via XL-type connectors, with a choice of –10 dBu or +4 dBm output levels, for interfacing with virtually all studio systems; and up to four hundred times fast-forward/rewind and search speeds, to provide high-speed access to any point of a two-hour DAT tape within twenty-seven seconds.
Manufacturer: Panasonic
Price: $1,599.00

Circle 64 on Reader Service Card

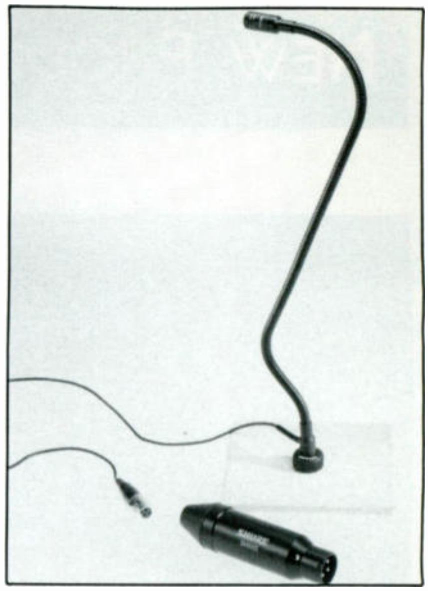
SURROUND SOUND CHIPS

• The SSM-2125 is the industry's first Dolby Pro-logic Surround Sound Decoder to fully integrate an auto-balance function. In fact, the SSM-2125 combines all the core functions of a complete Dolby Pro-Logic system on a single chip, including active decoding matrix, center mode control, noise generator and auto balance. Auto-balance provides dynamic correction of left-right input signal-level imbalances, eliminating the need for manual user adjustments and improving center-channel dialogue separation. The complete system integrates up to thirty operational amplifiers, ten voltage-controlled amplifiers (VCAs), a proprietary operational conveyor amplifier, two dual-output rectifiers, two log-difference amplifiers, comparators, random logic and a digital noise source.
Manufacturer: Analog Devices
Price: $15.00 in quantities of 100

Circle 65 on Reader Service Card

MINI GOOSENECK

• The SM99SE, an alternate version of the SM99 miniature gooseneck-mounted condenser microphone, is a member of the new MicroFlex line of miniature condenser mics. The SM99SE differs from the original in that the preamplifier is separate from the base to allow for remote mounting, and the gooseneck is designed to thread onto the supplied black 5/8 in.—27 flange. The flange can be attached to the desired mounting surface with three screws. The SM99SE is therefore designed for installations in which the 1-1/8 in. hole required to



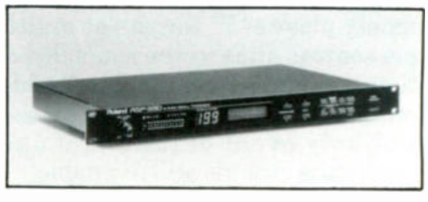
mount the original SM99 is undesirable. In addition, the SM99SE can be retrofitted to an existing flange from a previously installed gooseneck. The mic is compatible with the new A99SM Shock Mount, and its audio performance is identical to the original SM99.
Manufacturer: Shure Brothers Inc.
Price: $260.00

Circle 66 on Reader Service Card

SAMPLE RATE CONVERTER

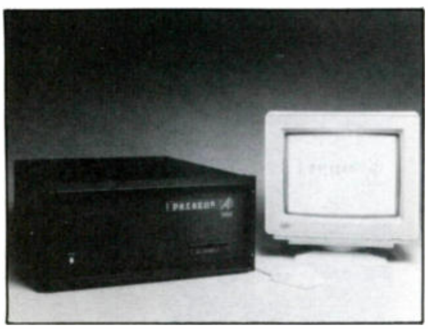
• The NV4448 Digital Audio Sample Rate Converter solves the problem of interchanging audio data between systems of different standards. Some features include all digital format and rate conversion between AES/EBU, SDIF II, and SPDIF Machines with flat response and group delay, and 24-Bit precision. The NV4448 makes it simple for any operator to convert between different sample rates and formats. Simply connect the necessary cables and select the desired output rate and format. Input rate and format are automatically determined. There is no signal degradation as all functions are performed digitally.
Manufacturer: NVision
Price: $7,500.00

Circle 67 on Reader Service Card



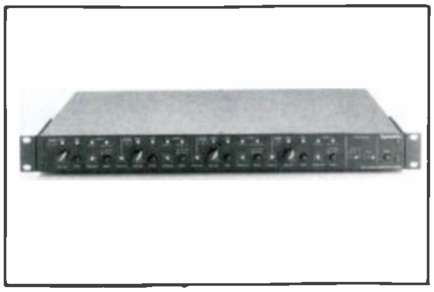
WORKSTATION

• The paragon-studio offers four independent I/O channels of signal processing in an AES/EBU digital format, and will support both 44.1 kHz and 48 kHz sampling rates. Each I/O channel can be user-configured with up to two signal processing functions which will work simultaneously and independently of processors in other channels, regardless of each processor's configuration. Each I/O channel and processors within that channel are set up, manipulated and controlled by a mouse-pointing device, and the system is viewed through a stand-alone VGA color video monitor. Each individual processor setups and all I/O channel route path configurations can be saved and recalled from the unit's 40 megabyte hard disk drive. Furthermore, each of these setups can be recalled and controlled through MIDI.
Manufacturer: Audio Animation Inc.
Price: $9,450.00
Circle 68 on Reader Service Card



ACTIVE CROSSOVER

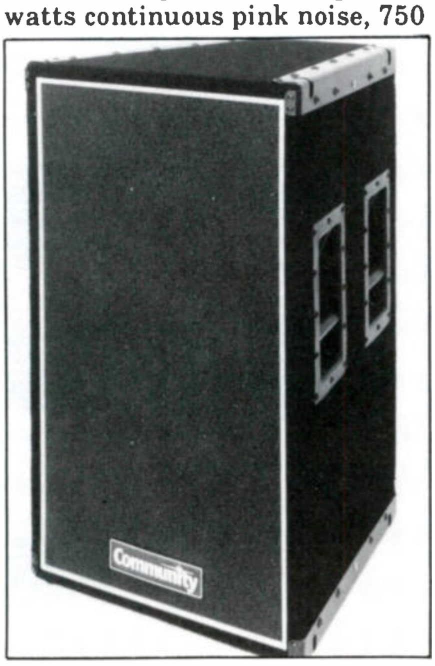
• The new 524E Multi-Mode Crossover has four crossover bands and can be configured as a mono two-, three-, or four-way or as a stereo two-way crossover. Crossover points and filter slopes (6, 12 or 24 dB/octave) can be set for each band. In addition, each band can be individually processed with a driver protection limiter and phase alignment compensation delay. High frequency EQ compensation for constant-directivity horns is also available, along with a subsonic filter for woofer excursion control. The front panel includes recessed limiter Threshold controls for each band, along with LEDs indicating limit threshold and +6 dB into limiting, respectively. For fast system setup and testing, the front panel also includes individual Gain controls plus Mute and Phase Reverse buttons on each band. Three recessed Phase Adjust controls set the driver alignment compensation. Other LEDs show Signal and Mute status for each band and the unit's crossover mode.
Manufacturer: Symetrix, Inc.
Price: $679.00



Circle 69 on Reader Service Card

COHERENT SPEAKER SYSTEM

• Each of the RS660's drivers is mechanically signal-aligned with the aid of a fiber-glass molding which serves as the backbone of the unit's trademarked Wavefront Coherent design. At the low end in the three-way system, twin 10 in. ferrofluid cooled woofers handle signals up to 650 Hz. Within the critical midrange bandwidths, frequencies are routed by the crossover to a 2 in., M200 compression driver. Above 3 kHz, signals are directed to the high end transducer, which features a titanium diaphragm and 1 in. exit area. General specifications include an operating range of 70 Hz to 18 kHz (±3 dB), power handling of 300 watts continuous pink noise, 750 watts program, a maximum continuous output of 129 dB SPL (typical), a nominal dispersion angle of 65 degrees horizontal (1 to 16 kHz), and 50 degrees vertical (1 to 16 kHz), and a sensitivity of 108 dB SPL (400 Hz to 4 kHz bandlimited pink noise) at 1 watt, 1 meter. The 660 System Controller provides optimal equalization and electronic filtering via its exclusive InteliSense Circuitry, is equipped with balanced inputs and outputs, and drives both the full-range and subwoofer systems.



Manufacturer: Community Light & Sound, Inc.
Prices: the RS660 is $1,895.00 and the 660 System Controller is $749.00.
Circle 70 on Reader Service Card

COMPUTER-CONTROLLED SYNCHRONIZER

• The PPS-2 synchronizer, an updated version of the PPS-1 synchronizer, reads and generates SMPTE time code, and converts SMPTE into MIDI Time Code or Direct Time Lock. It also reads and generates the "Smart" FSK sync, and converts it into MIDI clock with song position pointer. New features on the PPS-2 include Jam Sync, which provides SMPTE regeneration capability for tape duplication and time code repair; flywheeling, which protects

against tape dropouts; and Auto-Merge. Optional features include the PPS-2 Remote Software, which lets the user enter a start time for striping SMPTE, and select from 24, 25, 30 or 30 drop frame formats.
Manufacturer: JLCooper Electronics
Prices: the PPS-2 is $169.95, and the PPS-2 Remote Software is available for Macintosh and Atari computers at $14.95.

Circle 71 on Reader Service Card

TIMECODE DAT

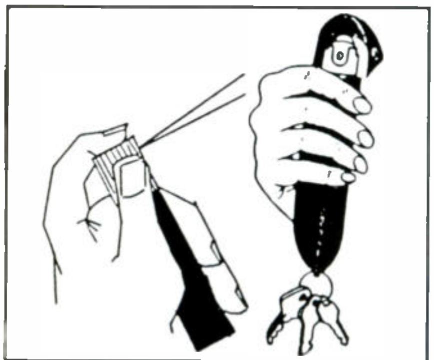
• PD-2 introduced to the world at APRS in June is now available in the U.S. PD-2 is a portable 4-head DAT with switchable 48 kHz, 44.1 kHz and 44.056 sampling frequencies, built-in timecode generation, full off-tape monitoring, and synchronization via a variety of external references. Timecode recording is possible in both IEC and Fostex formats. The internal IEC format generator can jam-sync to an external input or run from the unit's onboard real-time clock. A two-channel input stage provides channel-independent mic/lin switching, two phantom-power options, three-stage low-cut filters, mic-input limiter, and right channel phase reversal. Tape indexing includes slate tone or mic, auto/manual take numbering, manual start ID and error mark, and a system to identify and log PCM errors, input overloads, or timecode dropouts. There are three power supply options: rechargeable Nicads, "C" type batteries, or a four-pin XLR for external 12 V.
Manufacturer: Fostex Corporation of America
Price: $10,950.00

Circle 73 on Reader Service Card

CLASSIFIED

FOR SALE

Recording Studios and Broadcast stations work around the clock, so more than most need a safe personal protection device to ward off those who present a threat to safety.



Instantly-Effective BODYGUARD-The Magnum Powered Pepper Spray (5% oleoresin capsicum) has proven superior to Mace, CS and CN tear gas in personal protection. The convenient purse/pocket unit comes in a key ring holster providing you with 25 one-second blasts at a range of 3-5 feet that will render your adversary/attacker helpless from 30-60 minutes with *no* after effects.

Price: $15.00 + $1.50 postage/handling.

Checks and money orders only —no C.O.D./credit cards

The Right Response Company
P.O. Box 679
Medford, NY 11763

Allow 3-4 weeks for delivery

Note: *This offer is void where prohibited by law. This offer is not intended for sale to minors.*

SERVICES

HEAD RELAPPING AND REPLACEMENT

All tape and film formats over 30 years design experience **IEM - 350 N. Eric Dr., Palatine, IL 60067 800/227-4323**

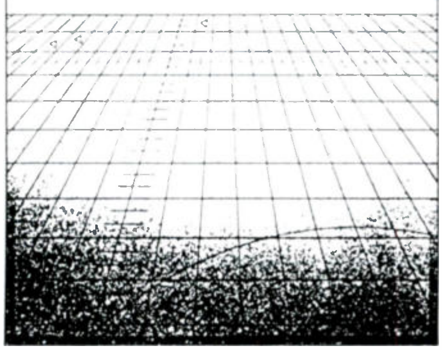
DAVIDSON ELECTRONICS

516 753-0197

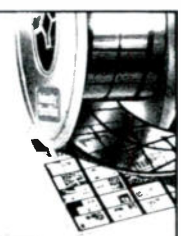
Authorized Service For

Yamaha, Crown, Tascam, Fostex, QSC, Carver, A&HB, Soundcraft, JBL and more!

BY APPOINTMENT ONLY



This publication is available in microform from University Microfilms International.



Call toll-free 800-521-3044. In Michigan, Alaska and Hawaii call collect 313-761-4700. Or mail inquiry to: University Microfilms International, 300 North Zeeb Road, Ann Arbor, MI 48106.

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

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

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

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

Send copy to:

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

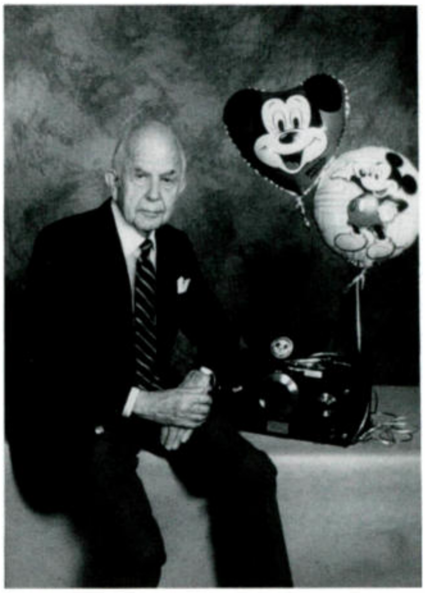
People, Places, Happenings

In Memoriam

William V. Stancil 1910-1991

On July 1, 1991, the audio industry lost one of its most prolific engineers. William V. Stancil passed away from the complications of pneumonia at age 81. In his career, Bill associated with such luminaries as Bebe Daniels, Ben Lyon, Lee de Forest, Eddie Stinson, Henry Ford, Dr. Alexanderson, Howard Hughes, William Wyler, Bill Hewlett, David Packard, Col. Richard Ranger, Jim Lansing, Sherman Fairchild, Gen. Curtis LeMay, Gen. Claire Chennault, Robert Oppenheimer, Lowell Thomas, Walter Cronkite, Walt Disney, Alexis M. Poniatoff, David O. Selznick, Douglas Shearer, and many others.

As a motion picture soundman, Bill worked on pictures such as *Wings*, the first Academy Award winner, *Stella Dallas, Gone With The Wind*, and the first three-dimensional picture *Bwana Devil*. His company, Stancil Hoffman, had the first professional tape recorder on the market in the United States, and later the first battery-operated portable tape recorder. He, along with Olin Dupy, was instrumental in developing film-sound recording and synchronization systems, and he sold the first Hewlett Packard oscillators ever made. Bill was part of the Manhattan Project that developed the atomic bomb, and was involved at Eniwetok for the



first hydrogen bomb tests. He had a 'ham' license (W6PF) in 1925, was an accomplished portrait artist, worked for Disney as an animator, and even appeared in the picture *Beck Sharpe* as a dancer. Bill was a charter member of IATSE local 695.

Bill started using solid state devices and modular construction in the 1950s. He was a pioneer in the development of multi-track recorders for logging and data recovery. He started concentrating primarily in that field in the 1960s and was responsible for many major safety and law enforcement data recovery systems throughout the world.

Bill's pursuit of excellence and dedication to his work was legendary. He was at work at his desk until he was taken, under protest, to the hospital from where he never returned.

In a recent interview for a soon-to-be-published biography, Bill summed up his career in a few short words.

"I was just in the right place at the right time." Those who knew him understood that Bill knew what to do with those opportunities.

Bill is survived by his two daughters, Sharon Custer and Judith Coolidge, and five grandchildren.

Shelley Herman, the author of these words about Bill Stancil has completed a detailed interview series with Mr. Stancil. It is in this issue

Norman H. Crowhurst 1913-1991

I've been saddened to learn of the recent death of **Norman H.Crowhurst**. Norman was for many of the early years of **db Magazine** the author that created the column *Theory and Practice*. Norman wrote that column for fifteen years. We got to know him well, and shall miss his talent and knowledge.

He died on March 7, 1991, apparently the culmination of a bicycle/car accident the previous autumn. Even though there were no apparent serious injuries, he did not recover completely, becoming partly bedridden by February of this year. Death was attributed to a sudden heart failure. LZ


"I did a great deal of shopping around while equipping my home studio. When I found the Peavey AMR 2400 console, I was thrilled: It had all the features I needed, was easily understood, looked great, and cost half the price of any other comparable board. Now I have so much production freedom — the 2400's logical layout makes using it so easy. My Peavey AMR 2400 console is the centerpiece of my home studio."
Jerry Goldsmith
Movie Score Composer/Producer