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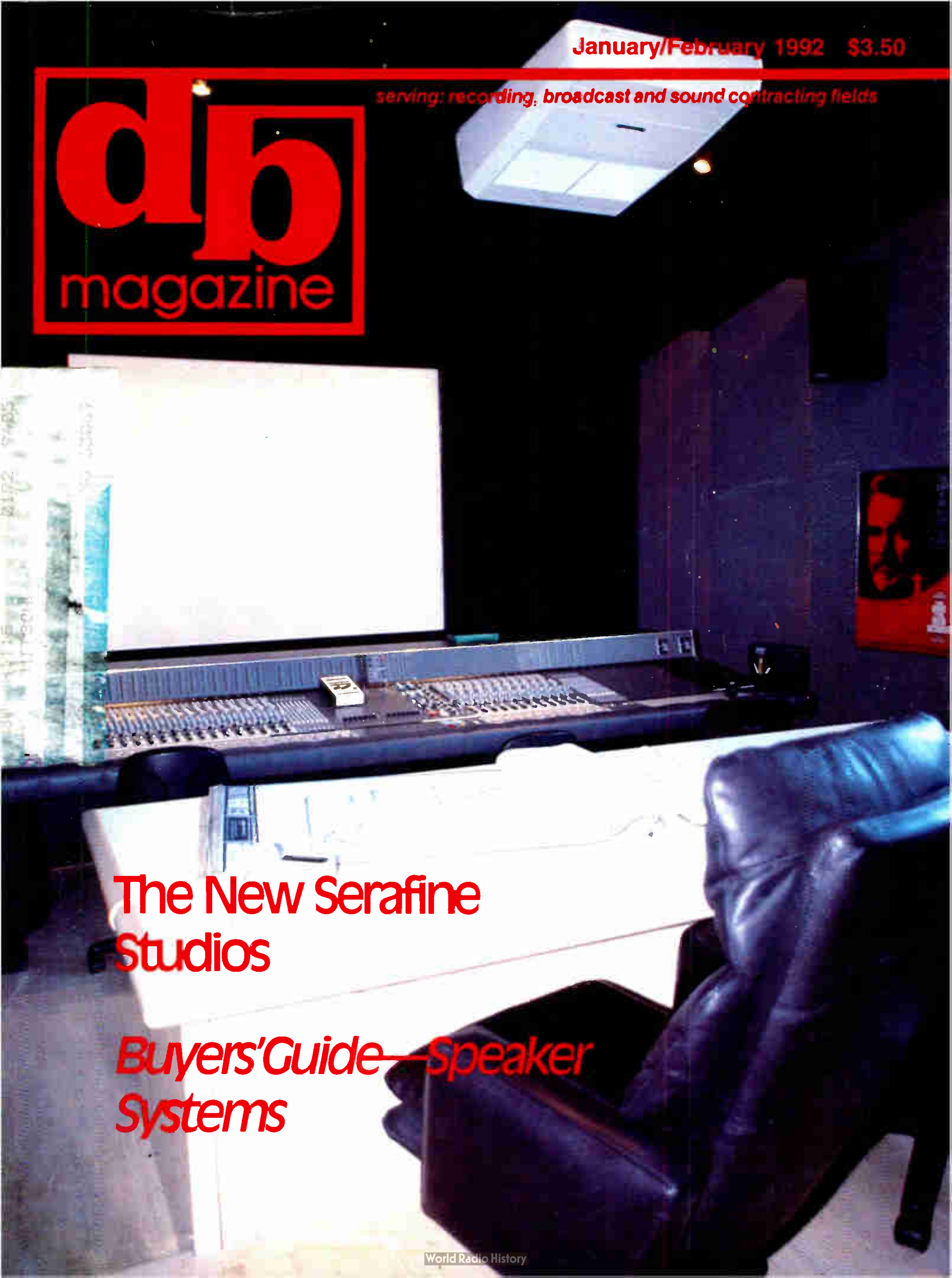

The New Serafine Studios

Buyers' Guide—Speaker Systems

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.

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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.

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January/February 1992 Vol. 26, No. 1

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The Recording Engineer

The Electronic Cottage

See page 10.

Audio for the Church

The Sound Contracting Engineer

The Broadcast Engineer

About the Cover

● Serafine Studios are but a few blocks from the beach in Venice, California. But inside, as can be seen from this view can be found a new modern console, video projection, and much more. Frank Serafine's new home and studio contains a veritable storehouse of synthesizers, computers, synchronizers, recording gear (digital and analog), and MIDI/SMPTE etc. See Brad Leigh Benjamin's story beginning on page 10.

Departments

Tek Texts

Calendar

• **SPARS** (The Society of Professional Audio Recording Services) will host a weekend business conference at the University of California, Los Angeles, in association with UCLA Extension. The two-day seminar will be held on January 25 and 26 and addresses the requirement necessary to establish, operate, and maintain a financially sound, professionally run recording studio. Topics to be discussed include: Controlling the Bottom Line, Pricing, Personnel, Legal Issues, Marketing, and Personal Use Studios. Entitled: *The Business of Operating A Recording Studio: Realities and Opportunities in the 90s*, the conference is organized by Guy Costa, CEO of Quadim Corp. and former President of SPARS.

For more information, **contact Shirley Kaye, SPARS Executive Director at (800) 771-7727.**

• **UCLA Extension's Certificate Program in Electronic Music** presents winter quarter schedule of courses,designed to acquaint participants with the techniques and language of the field. The schedule of courses include: *Electronic Music II: Introduction to Midi (Music Instrument Digital Interface)*, a thorough initiation into the function of MIDI, which meets Wednesdays, February 19-March 25, 7-10 p.m., at UCLA, 1344 Schoenberg Hall; and *Music Applications for the Macintosh Computer,* a course for composers, producers and recording engineers to make informed decisions regarding the Macintosh music computer system and to determine those applications that work best for their purposes, which meets Tuesdays, February 24-March 10, 7-10 p.m., at UCLA, 1344 Schoenberg Hall; the fee is $250.00 and the credits amount to two units in Music.

Those who successfully complete a sequence of courses can earn the UCLA Extension Certificate in Electronic Music, a program coordinated by Jeffrey Rona, president, MIDI Manufacturers Association, author, *MIDI—The Ins, Outs & Thrus*, and a synthesist and composer. **For more information on the program or to enroll in winter quarter courses, contact the Performing Arts Program, UCLA Extension, 10995 Le Conte Ave., Room 437, Los Angeles, CA 90024, or call (310) 825-9064.**

• **NAB** (National Association of Broadcasters) is sponsoring its **27th Annual Management Development Seminars for Broadcast Engineers**, February 9-14 at the University of Notre Dame in South Bend, IN. The seminars offered are Management I: *Fundamentals of Leadership*; and Management II: *Toward Leadership Effectiveness*. Management I topics include interpersonal communication, leadership style and

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Editor/Publisher
Larry Zide

Associate Publisher
Elaine Zide

Senior Editor
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Contributing Editors
Bruce Bartlett
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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.

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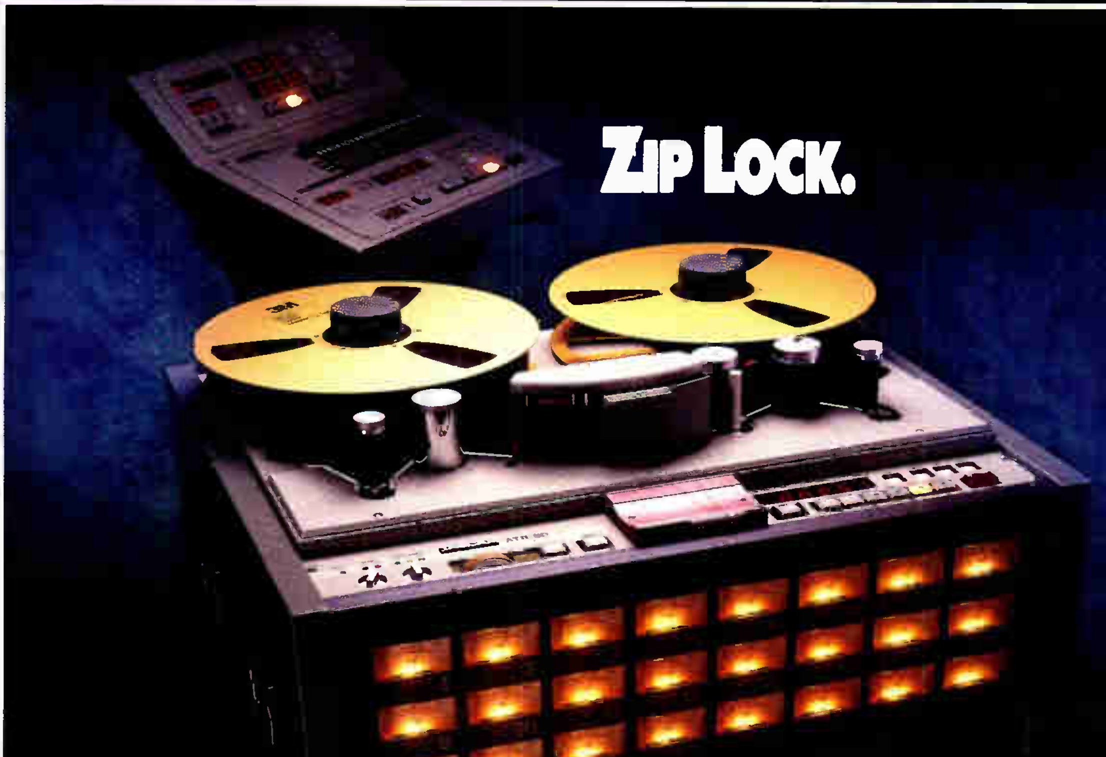

If you've ever dealt with a 2" 24-track whose synchronization was a design afterthought, you'll appreciate the speed and pinpoint accuracy of the TASCAM ATR-80. Its microprocessor-controlled transport is specifically designed for the repeatable locating you need for no-excuses, extra-fast lockup in synchronized operation. And the punch in/out precision you demand.

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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

projecting a professional image. Management II topics include motivation and personality assessment, performance improvement and a review of Management I. Courses are sequential in design to incorporate progressive training techniques. Participants successfully completing either seminar will receive 3.4 CEU points.

The seminars will be directed by Richard D. Cupka, president of the Cupka Corporation.

Included in the registration fee is tuition, housing for six nights on campus at Notre Dame's Morris Inn, all instructional materials, Sunday evening reception and dinner, Monday luncheon and Friday evening graduation banquet. Cost is $1,675 for members and $1,850 for nonmembers. Call NAB Science and Technology at (202) 429-5346 for further information and a registration brochure outlining the program in greater detail.

NAB will hold its annual Radio Group Head Fly-In February 10-11, 1992, at the Grand Hyatt in Washington, DC.

Participants will engage in roundtable discussions targeted to the specific interests and concerns of group operators and explore proven management and cost-saving ideas. In addition, they will receive updates on legislative and regulatory issues.

Topics include: Digital Audio Broadcasting, LMAs, Marketing Strategies, Sales Problems/Opportunities, Stretching Your Operational Dollars, Income-Producing Plans, Recruiting and Retention. **Cost is $225 for NAB members, $350 for nonmembers. The event kicks off with an evening reception on February 10 and sessions begin with continental breakfast on February 11. Tuesday's lunch is sponsored by Birch Scarborough Research Corporation. For additional information or to register, call NAB Radio, (202) 429-5402.**

- **Stage Art**, a trade fair for theater, music and dance, will be held for the first time in Frankfurt from March 11-15 1992, to run parallel with the Music Fair. The trade fair is organized mainly by Messe Frankfurt. Director, August Everding, has been a member of the advisory council on artistic matters for Stage Art since 1990. **More detailed information is available from the Stage Art representative, Anja Diete, at: Messe Frankfurt, Ludwig-Erhard-Anlage 1, D-6000 Frankfurt 1, tel.: +49 69 7575-6228.**

- **SBE** (Society of Broadcast Engineers) and **NPR** (National Public Radio) is offering a **Technical Certification Course in Broadcast Technology**. The NPR/SBE Course will be carried by NPR via closed-circuit satellite to its member stations in the winter of 1992. It will cover the SBE Broadcast Technologist certification level, as well as a review of the Broadcast and Senior Broadcast Engineer certification levels. Each course will include a workbook, accompanying audio tapes, review exams, and live satellite interconnects with expert instructors. The course is divided into two phases, the self-study phase which includes the workbook, tapes and exams; these will be sent out at the end of January to those interested; and the second phase, live satellite interconnect, will be in March. The project is partially funded through a grant from the Corporation for Public Broadcasting's system Development Fund. (The price for SBE chapters is $150 per course, used to offset expenses incurred in preparing the program.)

The course will cover five important elements of radio station operation: Electronic Theory, Audio Theory and Practices, AM/FM Radio Frequency Theory, Satellites and Microwave, and FCC Rules and Regulations. **For information on the certification course, contact Donna Fox at National Public Radio at 1-800-235-1212, extension 2737, or write to: NPR Training—SBE Certification Course, 2025 M Street, N.W., Washington, DC 20036.**

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Letters

The Editor:

I always enjoy reading what other people are doing in the audio industry, especially with church audio. In reading the September/October 1991 issue, Hal Swinhart wrote an article on *Creating A Church Recording Studio* which I enjoyed as well, and was glad to see that churches are putting in high-quality recording studios. However, Hal stated "The control room is set up on the live-end-dead-end (LEDE) arrangement. A thick carpet covers the rear wall to eliminate bounce." Although in reading the article and looking at the photo one would conclude the front end is live (LE) and the back end is dead (DE), this is technically incorrect. The term LEDE gets thrown around a lot in the professional audio industry and is often misrepresented. I would like to clear up what LEDE represents. I have heard many times that a LEDE room was the same as described by Hal, and others state that the loudspeaker end of the room is dead and the back half is live, which is more correct but not the whole story and therefore not correct either.

The originator of the live-end-dead-end control room concept is Don Davis of Synergetic Audio Concepts (Syn-Aud-Con). LEDE is a registered trademark of Syn-Aud-Con. A certified LEDE room can be obtained by meeting the following criteria:

1-Proof by means of energy-time-curve (ETC) measurements—that the initial time-delay gap (ITD) of sound from the monitor loudspeakers in the control room at the operator's position behind the console is greater by at least 3 msec than ITD of the associated recording studio.

2-Proof by means of ETC measurements that there are no reflections from the face of the console within 20 dB of the direct sound energy from the monitor loudspeakers.

3-Proof by means of ETC measurements that there is no early-early sound (EES) generated by unexpected flanking paths possessing higher transmission velocities than air.

4-A full set of control-room drawings indication where absorptive material, Haas reflectors, and diffusers are placed as well as mounting geometry of the monitor loudspeakers.

5-Diffusion of the quadratic residue (QRD) or primitive root diffuser (PRD) type fully qualifies a rear wall. When other types of diffusion are used, a TEF 3D measurement of the diffuse field only, is required.

6-The inner shell of the control room must be symmetrical and should exhibit a crossover to low-frequency absorption at a frequency indicated by the following equation: fx= 3(velocity of sound) /smallest room dimension.

An unsymmetrical outer shell is preferred, but as long as all other LEDE criteria are satisfied, certification will not be refused.

7-Proof by energy frequency curves (EFC), phase frequency (PFC), and ETC measurements that the monitor loudspeakers associated with the control room are not grossly out of signal alignment, i.e., absence of comb filtering capable of masking the inherent benefits of an LEDE control room.

There you have it, an LEDE room is not made by putting sonex at one end of the studio, and leaving the other end bare. So if someone tells me that they have a LEDE room, I'll want to see their certification.

If you would like more information on LEDE control rooms or certification, I'm sure Don or Carolyn Davis would love to hear from you.

Brent Harshbarger

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ERRATA

In our November/December isnsue in the story *Sound at the Summer Special Olympics*, a photo error occured. The photo of *Figure 5* on page 19 showing Maurice Paulsen with basketball star Magic Johnson is actually Maurice Paulsen and a *cardboard cutout* of Magic Johnson. To the best of our current knowledge Mr. Johnson was not actually at the Olympics.

Historic Savet Audio Artifact Collection Acquired by NARAS Foundation for Exhibition

The NARAS Foundation has acquired the SAVET Collection of entertainment technology artifacts—a historic collection of several hundred pieces of equipment—that document the development of audio recording and playback between 1900 and 1960, from wax cylinder to magnetic tape.

The acquisition announcement was jointly made by **Michael Greene**, President, National Academy of Recording Arts and Sciences, Inc. (NARAS) and audio engineer **Shelley Herman**, President, SAVET (SAVE Technology, Inc.) of Hollywood, CA.

The NARAS Foundation plans to curate and display the artifacts in a special "Living Technology Exhibition." The Exhibition is most likely to be displayed in the Los Angeles area.

"The archiving and preservation initiatives of the Recording Academy are among the most important activities of our organization. The acquisition of the SAVET Collection underlines the Recording Academy's desire to preserve and document a physical record of recording's technical history for generations to come," noted Greene.

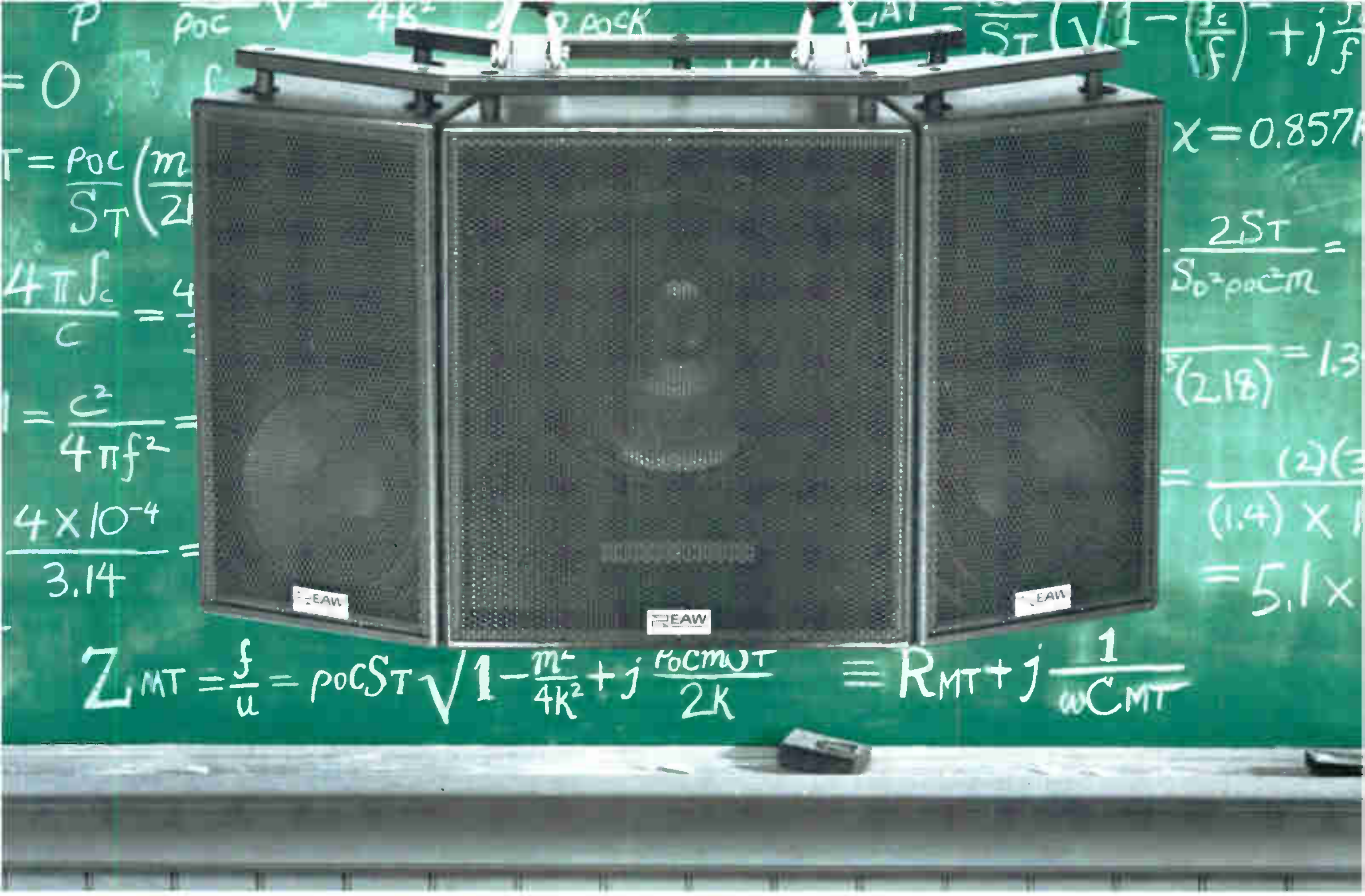
Holding a 1920s Westinhouse radio, part of the SAVET Collection of entertainment technology artifacts are (right) Michael Greene of NARAS and (left) Shelley Herman of SAVET. Shelley is also a Contributing Editor to this magazine.

Of special interest in the Collection are an Ampex Model 200, the first American professional tape recorder and a Stancil Hoffman, the first battery-operated tape recorder.

Also included are a Victor Orthophonic phonograph, the first acoustic "Hi Fi" phonograph, early acoustic cylinder and disc reproducers, a 1920s Westinghouse radio, early Western Electric amplifiers and Vitaphone disc cutting lathes, examples of magnetic recorders from Ampex, Stancil-Hoffman, Scully, Telectro, 3M. The Collection also contains documentation and recorded material.

SAVET was formed by audio pioneer **John T. (Jack) Mullin**, **Peter D. Hammer**, former curator and builder for the Ampex Museum of Magnetic Recording, and Herman.

"The transfer of the Collection to NARAS complied with SAVET's charter, which calls for the collection of entertainment technology artifacts to be saved from destruction and the transfer of these artifacts to facilities, such as the Recording Academy, where they will be preserved and displayed," said Herman.



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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.

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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.

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BRAD LEIGH BENJAMIN

Cover Story: The New Serafine Studios

The new complex is located in Venice, California and is the LA area's newest film mix, sound design and post-production facility.

As a young and insatiably curious child reared in a New York City apartment, I remember being awestruck with the homes of friends visited by my family—actual houses with an upstairs and a downstairs and all sorts of endless everywheres going somewhere for what seemed like forever. Mesmerized, I'd waddle down one step at a time to the rec room for the ensuing chase of nurf this or that—lost in the spellbound delights of childhood.

With the passage of time, tag gave way to ping-pong. Billiards and girls replaced hide 'n' seek, and yet my youthful fascination for recreational space in such close proximity to one's eating, sleeping, and living areas remained untarnished. Years later, having become a musician and studio junkie to the utmost extreme, I secretly vowed to one day own and operate a state-of-the-art recording facility from within the comfortable confines of my own home.

That fantasy has been vicariously realized through the vision and magnificent effort of sound designer/composer, Frank Serafine, owner of newly opened Serafine Studios in Venice, CA.

Underneath Serafine's oceanside penthouse digs are two floors of studios, offices and plenty of parking for guests and clientele. When Frank Serafine wakes up in the morning, he can leisurely breakfast, tub, read the paper or relax amidst the sweet bounty of a private rooftop garden—then eventually make his way downstairs to greet clients in Serafine Studios' new sound design suite or film/video mix stage. He has successfully created a commercial recording facility replete with huge penthouse apartment and panoramic landscaped rooftop.

Figure 1. The completed Serafine Studio building.

Serafine built the facility because he wanted the freedom to work at home, unencumbered by city ordinances and residential zoning regulations.

"I just wanted to be zoned commercially," comments Serafine. "I've been based out of my home for years— going back and forth from working in commercial facilities, to working back home—to commercial facilities, and home again. It was like a never ending circle. With the way that Los Angeles is getting as far as traffic and the hassle of getting around, it's much better to work at home. You don't have the commute and the wear and tear on your system for those couple of hours everyday—fighting for your life on the freeways. I'd rather spend the time actually working than getting to work."

He adds, "The last facility I had in Santa Monica was in my home, but it was a thousand square foot studio—probably one of the more elaborate home studios in L.A." Having seen Frank's former facility, this writer can attest to its formidable array of synthesizer and sampling technologies—not to mention Serafine's enormous sound library. That library is now equipped with the complete Sound Ideas, Gefen Systems, Omnifx and Northstar/Wizard libraries.

"I was concerned that the zoning administration would screw me in Santa Monica, and they eventually

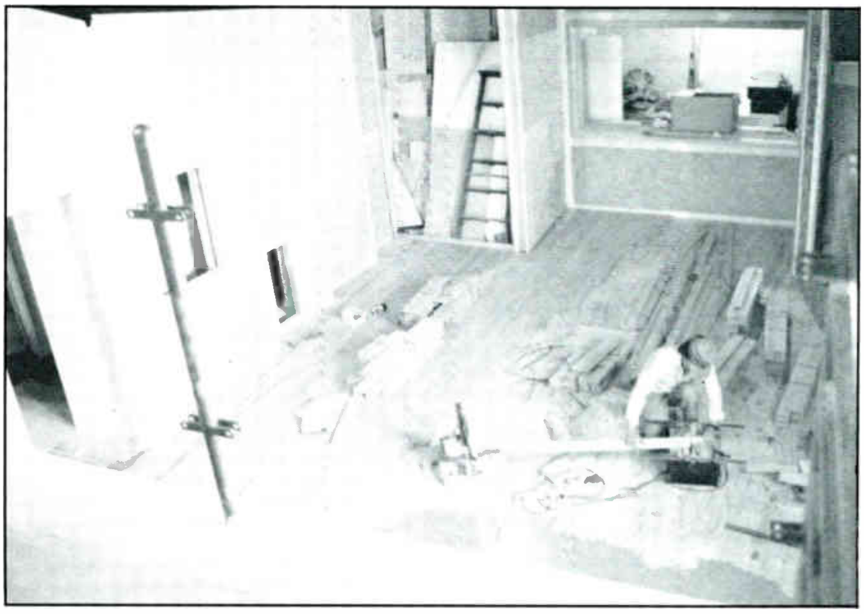
Figure 2. Basic building framing completed. Frank is standing in what will become the main studio room.

did, but I was already building this place, so it didn't matter," adds Frank. "I also wanted to expand my business, I was just getting too busy. I needed more room and commercial zoning, which allows me to have more employees and a smoother running, more professional facility."

THE STUDIO LAYOUT

The ground floor of Serafine Studios consists of 2,500 square feet of office space (leasable to outside interests such as production companies and entertainment related businesses) and plenty of covered parking. On the second floor, one encounters the studios themselves, totalling 4,400 square feet. It all begins in Serafine's sound design suite—very deep in E-mu Systems technology. A pair of new E3Xs handle sample playback from the afore-mentioned library. The E3Xs provide a total of sixteen polyphonic outputs between them. The samplers are complimented by a diverse assortment of both analogue and digitally based synthesizer modules—all interfaced and accessible from six Macintosh computers located throughout the facility.

Figure 3. A detail of the interior framing in the studio room.



Serafine works with his clients initially in the sound design suite where they can provide feedback on various stages of effects design and production—monitoring visuals on a Sharp Data Projection System utilizing a 4-foot x 4-foot view screen. Serafine generally prefers to work with all Mac-based data such as timecode, command functions, and parameters projected in an isolated window on the video screen. Clients will ultimately listen to his final product on the 28-foot wide X 35-foot long X 22-foot high mix stage, monitoring video on a 6-foot X 8-foot screen, sourced by a Panasonic 200P video projection system equipped with a digital double line scanner, "a component which makes the image look more like high definition TV or film," says Serafine.

When the facility first opened, sound effects and samples were originally dumped to individual tracks on a Sony APR 24 in the sound design suite for subsequent mixing on the main stage. Serafine's intention, however, is to interface all sound sources in the design suite directly to the new Otari Series 54-P console on the main stage, enabling him to route and process all synth-based and sampled audio signals directly from their sources (with no generation loss) to the final mix—mounted on six tracks of an Otari MX 80 24-track tape machine with Dolby SR available. Final multi-track mixes are to be sent to lab facilities for print mastering.

MAIN STAGE EQUIPMENT

On the main stage, Serafine will be using an E3 Remote Front Panel interfaced to a Mac front end utilizing Opcode's Vision and Galaxy software programs, to control the sound modules in the design suite and access all sound banks and libraries. Appropriate selection of MIDI channels will allow sound designers to use the E3Xs in the design suite while Serafine accesses those same modules from the mix stage—a cost-effective approach to time management.

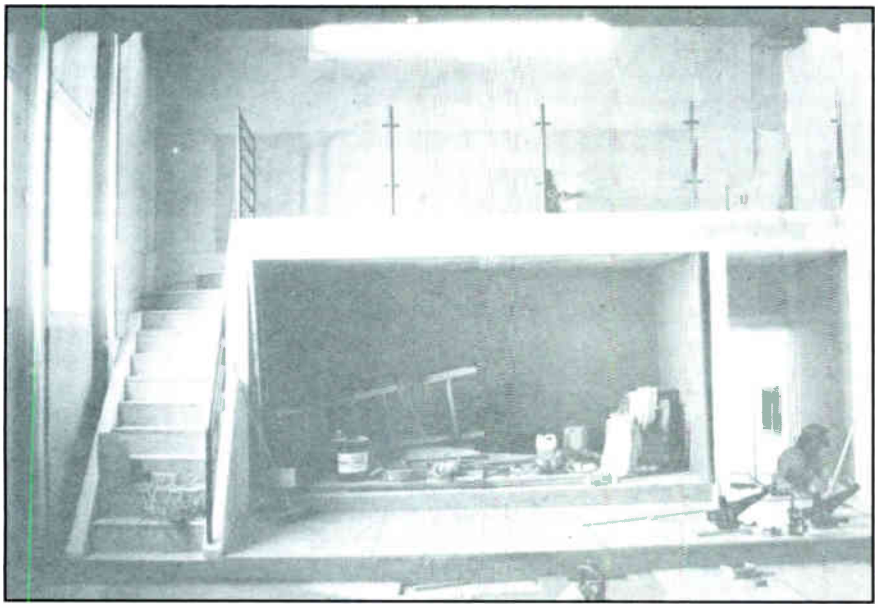


Figure 4. Studio front framing.

The new mix stage (system designed by Randy Honaker) is equipped with an E-V Mark IV cinema monitoring system and the previously mentioned Otari Series 54-P console. The Series 54-P is based on a standard Otari Series 54 console design, extensively modified to meet the needs of *Left-Center-Right-Surround* mixing for film and video formats. Serafine's new 54-P is configured as a two-position console with 36 dual-path input modules and 16 group reassign modules. The genuine dual-path modules actually provide Serafine with 72 individual input/returns—each independently capable of accessing its module's EQ and auxiliary send functions. Additionally, the console is outfitted with a 40-channel DISKMIX 3 Moving Fader Automation System, two quad joystick panners, and two mono 7-band EQ's.

"We had to have the right console and we didn't want to have to make elaborate modifications to it after it got here," comments Serafine. "The Series 54-P is the only console we liked with LCRS architecture built right into it."

WORKING AT HOME

Frank Serafine has been spending a great deal of time both preparing and working in the new facility. "It's great," says Serafine. "The thing is, I work long hours whether I'm at a 'commercial' facility or not. When I'm here, I have the option of showering on a break or relaxing before clients arrive—taking a nap if I'm working double shifts. You can't really do that in a commercial environment. You're stuck sixteen hours a day and that's it, and it's hard on you. It's nice to have some sort of release in your environment—a means of relaxation."

For relaxation, many studios serve up a selection of home video games in their respective lounges, providing a diversion during session time-outs.

This writer generally returns from such breaks, twice as stressed as before the break—usually from savage beatings sustained at the hands of pizza-devouring, six-foot turtles skilled in the art of self-defense, or from perilous journeys vicariously experienced through the action-packed adventures of those Super Mario Brothers—retired wallpaper hangers with bionic vertical leaping skills—the illegitimate twins of Geppetto and Spud Webb.

Every time I play, I exit the lounge irritated and uptight—cursing the moment I cast my fate with two guys who probably should have stayed in South Brooklyn making pizza for those turtles.

NO STRESS-INDUCING GAMES HERE

The Serafine complex provides genuine ambiance and relaxation including hot and cold tubs, sunning terraces, and of course, the panoramic rooftop garden—amenities which can be made available to clientele. Serafine regularly entertains clients in his 2,500 square foot penthouse apartment.

"It's a nice, personal touch," says Frank. "They know that I'm bringing them into my home. We have dinners here and then go back downstairs to work. There's more of a personalized feeling here. In a strictly commercial environment, it's a little more impersonal. You always go out to restaurants or have take-out food delivered. Here, we just go upstairs, have dinner, relax and then go back to work. There's a real nice feeling about that—and you always have the option of not bringing clients upstairs if you don't want to.

THE BEACH AMBIANCE

If clients prefer to head out, Serafine Studios is only two blocks from the Sidewalk Cafe and Venice Boardwalk where one can observe bikini-clad Amazons and bare-chested terminators on roller-skates.

The beach and ocean are ever present. This is a boon for Frank Serafine whose conceptual vision of the studio has consistently embraced the architectural design influences of Southern Europe and the Mediterranean. Prior to having the plans drawn, Serafine travelled to Europe and took photos of buildings throughout southern Italy, from Portofino and Florence, to Monte Carlo—garnering ideas and inspiration for his building's overall design.

PROFESSIONAL HELP

He enlisted the aid of European design architect, Majid Farzanefar to translate his vision into an architectural program and working plans. Farzanefar and American architect George Fulks worked together on what were to become the blueprints for Serafine Studios.

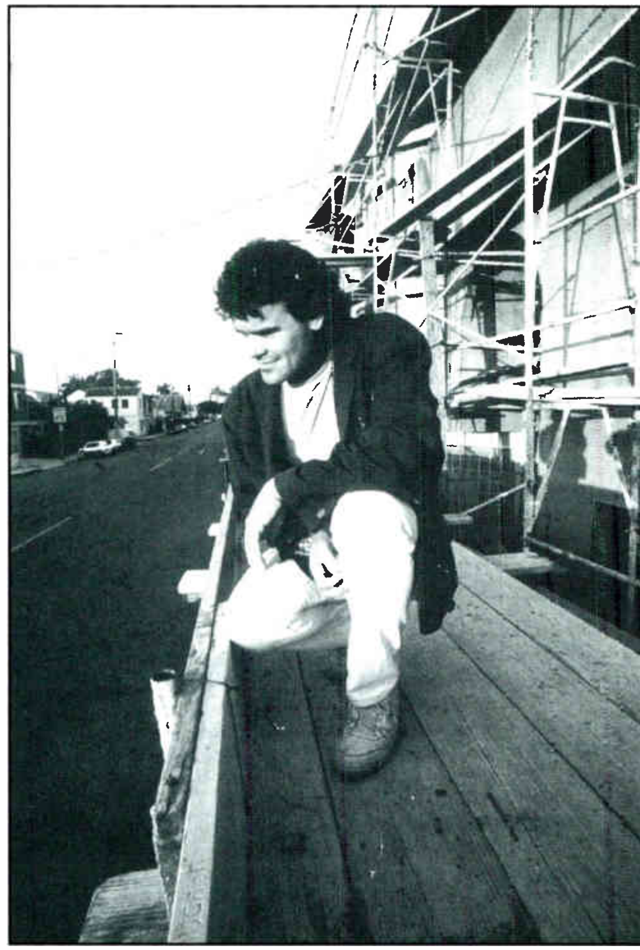
"I wanted to take ideas from Michelangelo and incorporate them into the nineties," says Serafine. "I gave all of my ideas and photos to the architects and told them what I wanted. I chose Majid for his European sensibilities. That was important to me." He adds, "The studio exterior—the two-tone colors—the beiges—the sandy whites and almost peach colors were taken directly from buildings in southern Italy. The shape and texture of the building—the rooftop and hanging gardens over the sides of the building—all of these facets were derived from buildings in Florence and Monte Carlo."

Serafine continues, "It's exciting to see a building go up but it's not always fun dealing with people who don't know what you're thinking. Most building contractors don't know how to build recording studios. Carpenters have a natural tendency to brace and reinforce everything. They don't understand the concept of *floating* rooms and de-coupled walls to isolate low frequencies and prevent them from travelling throughout the structure. They need constant supervision because the very structural principles which make for great recording studios sometimes go against their natural tendencies. It was an absolute necessity to have acoustic consultants present at every step of the way, advising me on what was and wasn't being done correctly. Acoustic engineer, Carl Yanchar of Lakeside Associates was with us every step of the way. He anticipated problems and oversights before they became insurmountable."

DESIGN DETAILS

He adds,"Carl realized during the early stages, that we wouldn't have sufficient isolation between the floor of our second story studio and the covered parking directly below at ground level. We certainly didn't want to hear people revving up their Harleys downstairs while we were trying to mix, so Carl decided that we had to have two 4-inch concrete floors poured in the mix stage to create mass and effective isolation. This also meant that K. K. Kwong, our structural engineer, had to come back in and re-design the whole support structure of the building. He had to restructure the steel—the foundation—

Figure 5. Frank is standing on a short second floor (studio complex) looking satisfied with the progress of construction.



everything in order to hold that kind of weight in the studio. Instead of 2 X 12s we had to go with 4 X 12s to support the concrete floors.

Needless to say, there were a lot of expenses incurred that we had not anticipated in the original drawings. Also, when you make those kinds of changes, you have to go back and get them OK'd by the city."

He continues, "Another unexpected surprise previous to actual construction, was the realization that we'd have to lower the studio floors by two feet, otherwise the elevation of the screen wouldn't have been high enough, Originally, we were supposed to be able to crawl underneath the floor, but we had to lower it in order to get the right sight lines between the Otari console and the screen, otherwise the mixers would have had to look down at the screen. Can you imagine if we'd have caught that *after* we'd poured those concrete floors?"

"Also," says Serafine, "The city plays a major role in your plans and schedule. They made us put in six street lights—way after the fact. I'd already poured my sidewalks, paved a street and *then* they told us about the lights. Now I had to pay a civil engineer for plans, pay the city for a permit, and a contractor to put the lights in. Then I had to pay another contractor to put the sidewalks back in because we'd torn out the original sidewalks we poured to make way for the lights. Civil engineer, Art Letores supervised most of this work for me because you can't mess with the city. They do whatever they want. The city thinks it is God and you have to listen to them if you want to build. That's the bottom line."

CHANGES

"There weren't too many other problems," recalls Serafine. "In the original plans though, the air conditioning for the sound design suite room wasn't calculated for all the heat generating equipment that was going in there so when we got in and realized we didn't have adequate HVAC, we had to tear through three stories of structure including concrete, expand the opening through its entire length,

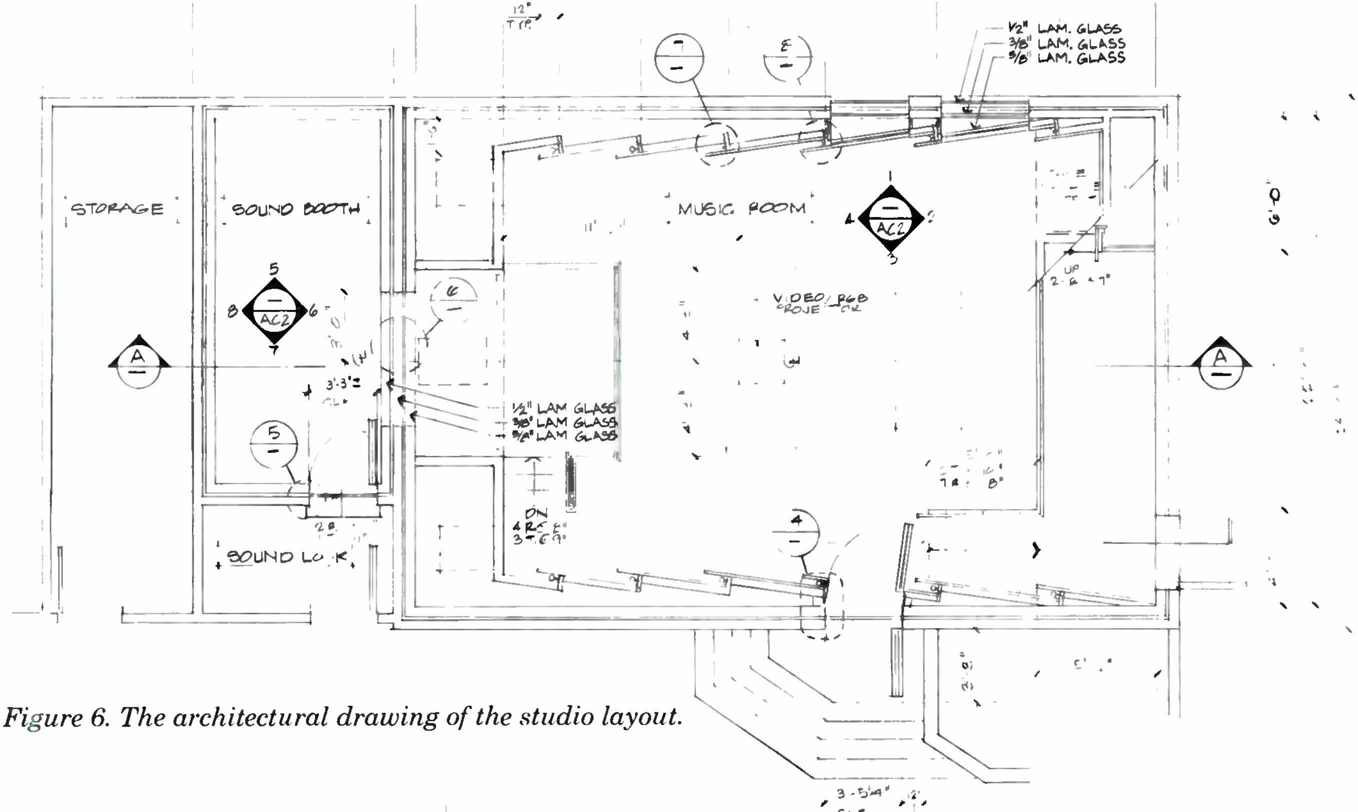


Figure 6. The architectural drawing of the studio layout.

and bring ducts down through to the studio.

Nobody had foreseen the heat from the instruments. Majid Imeri, our mechanical engineer worked it all out."

He pauses. "You know, in a specialized building like this—combining studios, offices, and living space, it's real tricky getting everything to work together. We actually had to have K. K. Kwong restructure the supports a second time to accommodate the concrete boxes housing the gardens on the roof.

No one had thought of that and it meant additional expense and another trip to the city for a permit."

Figure 7. This glass-block enclosed stair goes from ground level to the studio.

THE BUILDING IS READY!

Now that the structure is complete, Serafine sleeps more soundly (secure in its structural integrity, nomadic beauty and ability to deliver product). A waterfall cascades gently from the rooftop gardens down through three floors of the building's interior, pausing at each level to swirl and eddy through landscaped, indoor ponds—hydroponic gardens surrounded by rock and stone. From behind and underneath water irises, lilies and assorted aquatic plants, tropical fish peer out at offices and traffic areas. A natural open air shaft created by the falls, ventilates the offices without much need for air conditioning. The shaded shaft and cool water serve as a natural funnelling conduit for cool ocean breezes.

The interior walls bespeak a modern, high-tech look—red railings, grey carpet, white walls, black trim, stucco, drywall, a spiral staircase. While Serafine embraces the influences of Southern Italy and the Mediterranean, he shies away from the ornate and potentially garish possibilities inherent in the neo-Roman influences of Northern Italy. The interior walls at Serafine Studios are not cluttered with artifacts nor filled with antiques. They remain more sparse and simple—giving way to the natural lines of the structure. *Les objets d'art* gently reflect the traditional undertones of a Japanese temple or Shinto shrine. Small, laughing Buddhas impose no doctrine, yet smile upon those who encounter them—joyous in the simplicity of the moment. Not to worry, Serafine is not attempting to make a theosophical statement. There are no brown rice/tofu dispensers next to the Coke machine. He is simply trying to capture and provide very Zen-like, peaceful qualities of a Japanese garden for his clientele.

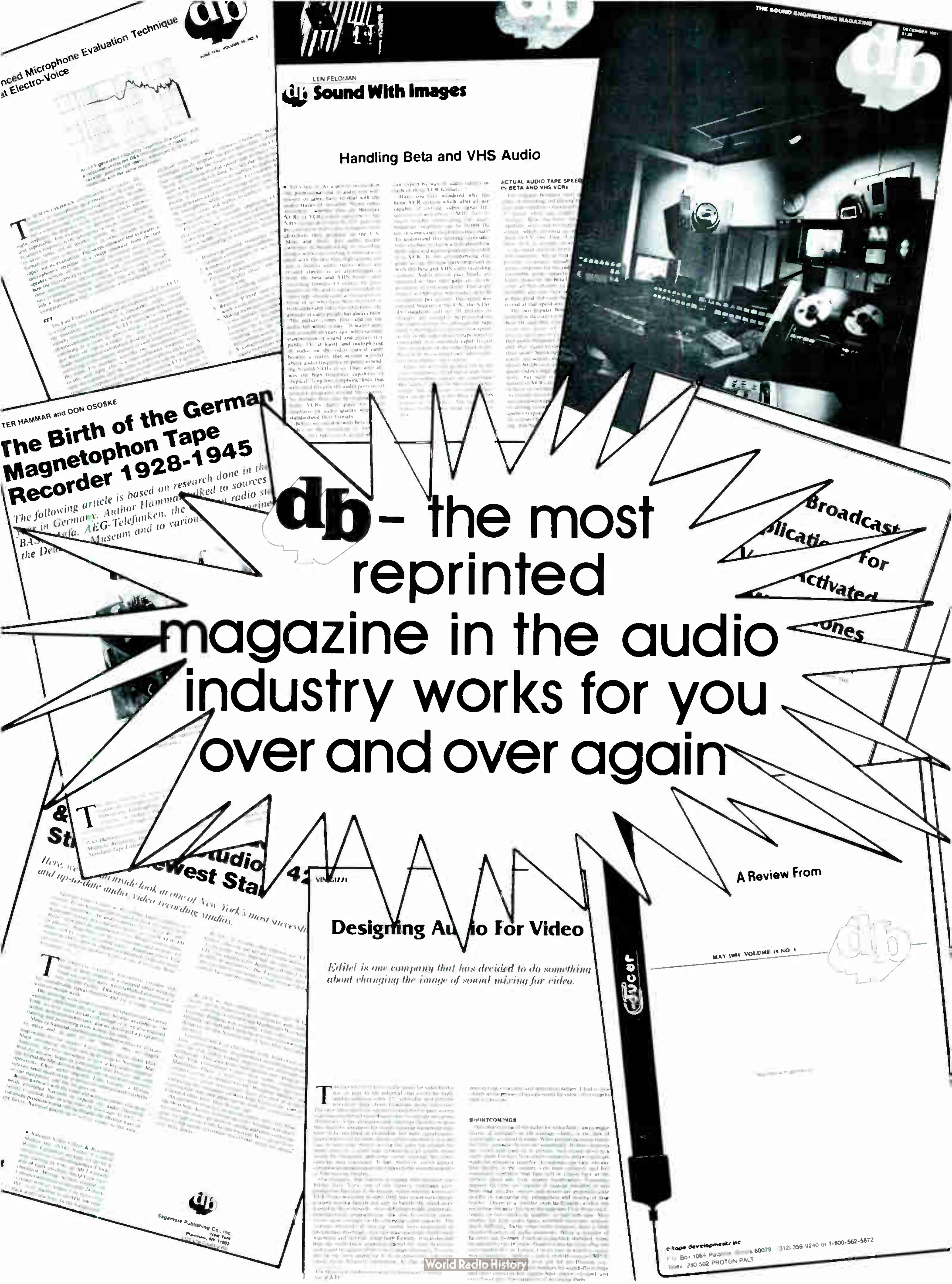
RECENT WORK

Serafine recently completed a portion of the sound effects design for Cecilia Hall, Supervising Sound Editor on the new *Addams Family* feature release. Serafine also worked with Hall on the sound effects for *Hunt For Red October*. He is presently working on *Lawnmower Man*, a New Line Cinema release, based on the novel by Stephen King. Serafine will be taking *Lawnmower Man* from the concept and audio design stages, through Foley, ADR, and mounting of the final mix—all at Serafine Studios. *Lawnmower Man* is being shot with high quality, hi-tech computer animation, calling for some interesting audio effects. Dialogue editing will be accomplished on a Studer Dyaxis using Dynatek optical media with 650 megabyte storage and a 5 gigabyte DAT backup. The musical score will be supplied by composer Dan Wyman of San Francisco. Wyman will bring his score to Serafine Studios on four tracks of Digidesign's Pro Tools direct-to-disk recording software which will drive the numerous sound sources available in the sound design suite. Having locked to SMPTE on the work print, Serafine and Wyman will mix the musical score on the Serafine stage.

Frank Serafine is also busy scoring commercials for Compuserve, Montana Tourists Association, and Boom Ball. Additional feature film packages are reported to be in the works. "We're looking to take on more full packages from start to finish—ADR, Foley, final mix—similar to what we're doing on *Lawnmower Man*," comments Serafine. With credits like *Hunt For Red October, Star Trek, Poltergeist, Tron*, and a brand new seaside studio to boot, it shouldn't take long for clients to start lining up for the peaceful, Mediterranean, Zen-like ambiance of Serafine Studios—a high-tech Japanese garden featuring exotic audio design, complete film mix capability, and Southern European intrigue—kind of like Marco Polo's summer place in Kyoto with a full service audio facility.

Cioppino!
Tempura!
Mangiamo!

db



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THE ELECTRONIC COTTAGE

Mastering Tips For The Home Studio, Part I

● The unsung hero, in the process of making records, tapes and CDs, is the mastering engineer. While the producer and the mixing engineer receive the lion's share of credit for the sound of a given product, it is frequently the mastering engineer who makes the difference between a recording which sounds mediocre and one which sounds awesome. Often, what it takes is a minimalist treatment—a bit of EQ here, a little compression there, but knowing exactly what to do is not as simple as it sounds. Precisely where to EQ, or what type or how much compression or limiting that might be required is an art and science that takes many years to gain proficiency in. That's why these people are in demand and the opinions of the *top dogs* in the field hold an aura of infallibility.

WHY IS MASTERING IMPORTANT?

Doesn't my mix represent exactly what I wanted to hear? Well, the answer is maybe yes, maybe no. Let's take a self-evident case as an illustration. Let's say a client wants to record a ten song collection to be released on cassette.

(This is not at all unusual today, even in a small studio, since lots of people are now into doing custom albums and putting them out on their own.)

The album is likely to have been recorded in pieces—tracking, overdubs, vocal, etc., over a span of several weeks, or even months. Recorded sounds are likely to vary from cut to cut, even with the utmost care. Mixing is also likely to be done over several days or weeks, and once again, even with the most stringent quality control, there will undoubtedly be some rather noticeable variations in apparent volume, EQ, and dynamics from cut to cut. If you just play the songs individually, they might all sound like acceptable mixes, but when you string them all together and play them in proper sequence, the differences may be very noticeable. The questions then become, which song will serve as the model for EQ, dynamics, and volume, and how can we get all the songs to sound reasonably similar in overall response? Here's where the mastering engineer can work some incredible magic.

Even individual songs can often be radically improved or even saved through some clever mastering. As frequently happens in long mixing sessions, the human ear accommodates to high levels of bass and highs. After a protracted period, the powerful midrange frequencies may seem to be missing and a mixer may erroneously keep boosting the mids in order to achieve clarity. When the arduous session is over, playback will reveal a powerful mix—just what you thought you wanted to hear.

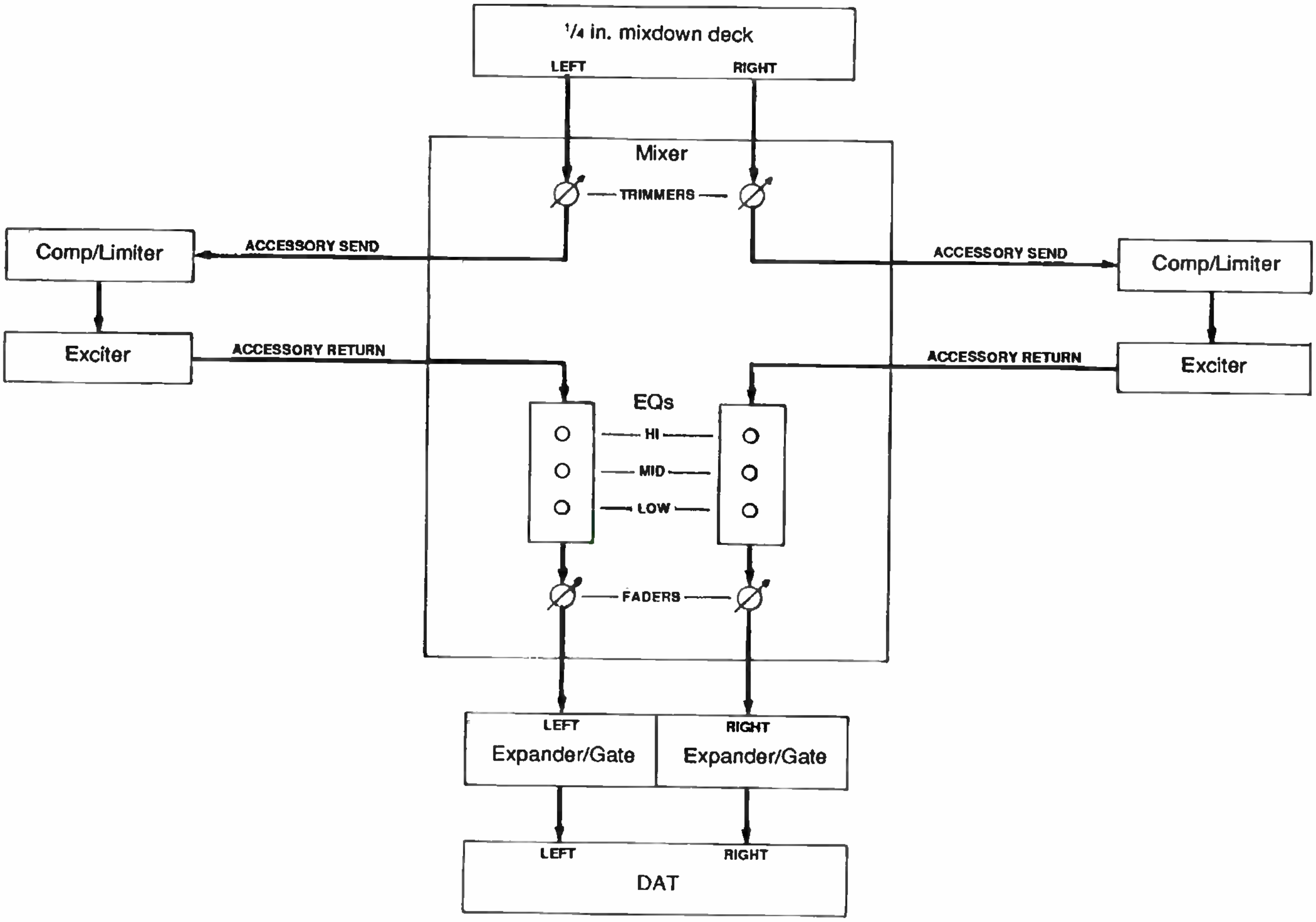


Figure 1. John's basic setup for mastering to a digital audio tape recorder. The same setup would be used to master to open reel or cassette recorders. In this case, noise-reduction would be added at the last stage before the recorder.

However, a couple of days later a more objective listening reveals a mix that may be balanced, yet very unnerving to listen to more than once. Your immediate reaction is, "Oh no, how could I have let this slip by me!", but the thought of remixing this complicated tune from scratch positively nauseates you. Can anything be done to save the mix? Enter the mastering engineer.

One final scenario will suffice to point out the importance of creative mastering. What if your final mix is tonally acceptable, but the level of a vocal or solo instrument is a bit too loud or too soft overall? It would seem that there is little that could be done for the ratios of tracks once they are fixed in stereo mix. While some cases will undoubtedly require a remix, many times this condition can also be favorably resolved through mastering.

Now the reader might be thinking, "This is all wonderful news, but who can afford hundreds of dollars in mastering fees just for a four song demo?" Well, be of good cheer, because many of the same techniques used by mastering engineers can also be applied in the home studio. No one is expecting you to have pricey tube equalizers, leveling amplifiers, frequency sensitive limiters—or any other item in the arsenal of a mastering suite. The home studio engineer can work some incredible miracles on mixes, getting them to sound smoother, punchier or whatever, with some very basic tools: a mixer with 3 band eq (hi, low and frequency selectable midrange), a stereo compressor/limiter, a stereo expander/gate, a stereo exciter, and a stereo effects unit.

(You may be wondering what the *effects* unit is for, since it is not usually considered a mastering tool, but as shall be seen, you can do some things in your home studio that are even beyond the normal realm of mastering.)

Not mentioned in the above list is the obvious: you will need your usual mixdown deck to play back your masters (which will usually be on reel-to-reel analog tape, preferably with noise reduction), and of course one other deck—whether it be another reel-to-reel, or (preferably) a DAT machine. However, even a good quality cassette deck will do.

THE TYPICAL SETUP

For the remainder of this article, I will refer to a setup that works well for me. If you have different gear or lack some of the pieces, just utilize whatever aspect of the chain you can. It will be beneficial anyway. (My typical setup can be seen in *Figure 1.*) Usually, I will mix onto a ¼-in. half-track (reel-to-reel) with dbx Type I noise reduction. My preference here over DAT is simply that it allows the flexibility to edit individual songs and also to re-order the selections at any time. (Of course, if you will ultimately be mastering to another deck, the order of the songs can be changed during the mastering process, so really any deck will do.)

I also like ¼-in. analog tape for the warmth that small degrees of tape saturation can bring to a mix. In any case, while the source is analog, the target deck for mastering is preferably a DAT machine. DAT has become the *de facto* standard for musical tape and CD duplication, and of course, the digital medium prevents any further generation loss.

I feed the source deck into two channels of my mixer panned left and right (so I can do some eq), with each channel's accessory send (the internal patch point) goes in to a channel of a stereo compressor (so I can affect the dynamics), then into an exciter (to psycho-acoustically spice-up the mix), returning to the channel through the accessory receive.

Finally, the entire stereo mix goes into an expander (to keep the whole chain super quiet). Meanwhile, two effects sends on the mixer are fed into an effects unit and a stereo delay (just in case I want to add some more dimension to certain sections of the overall mix).

Ultimately, the whole thing ends up on DAT. Now let's take a further look at each stage of the chain and see what improvements can be made on the overall sound.

WHAT YOU CAN DO WITH THE MIXER.

Mixers are designed primarily to control the level of a signal and to adjust the frequency content, and this is precisely what we use it for. How we use it is what makes the difference. First, with regard to level, we know that the ultimate output from song to song must be adjusted to have the loudest passage be at approximately the same level. This is especially important on DAT machines where maximum level must not be exceeded. Don't judge totally by the meters, though. While digital maximum level must be respected, songs should be adjusted on their sonic level, since equal meter readings may not always sound equally loud. The listener's perception, of course, is the bottom line.

This might be unorthodox, but it works. Pro studios running balanced lines throughout the system and +4 nominal levels don't need to worry about this as much, so they play it by the book.

Another area where meters should be considered helpful but not definitive, is the level of signal coming into the mixer from the source deck. Now this may fly against conventional engineering practice, but the truth is, the maximum level you push into the board without crashing, the quieter the output signal. Forget nice happy 0 or even +3 levels. Open the trim, if necessary and get the hottest signal into the board you can get, short of any audible distortion. Most boards have headroom that exceeds the metering range by 6 dB or more. While I would hardly advise this practice if you were recording a solo tambourine track, the fact is that the program material (completed mixes) are by nature somewhat compressed and certainly less peaky than individual tracks. So if you can judiciously squeeze a few dB more signal into your board, you can effectively save yourself the same few dB of noise by keeping your master faders down lower. This might be unorthodox, but it works. Pro studios running balanced lines throughout the system and +4 nominal levels don't need to worry about this as much, so they play it by the book. But the –10 unbalanced equipment most of us use can really benefit from such "seat-of-the-pants" techniques.

WHY USE A COMPRESSOR/LIMITER?

That's a legitimate question to ask. After all, you used it on the vocals and the bass because they needed it, so why should you put everything through it again? Well, you don't always *need* to treat the dynamics of a song, but most frequently you will find it beneficial to do so. There are a few reasons why this should be so. One, is that for most pop music formats, we can say that the listener reasonably expects it to be that way. For years of listening to LPs, we have gotten used to a compressed sound. It was probably done originally for technical reasons: there was only so much energy that could be captured in a record groove without having the cutter blow a hole in the disc. The depth and width of the grooves were proportional to the:

a) volume of the song and

b) to the amount of bass energy.

As a result compression and frequency sensitive limiting were often applied. Later, I surmise, people started fixing on the steadiness of the sound, noticing that it kept the listener engaged a lot more than if it had too many fluctuations.

Needless to say, the threshold should be set fairly high—so that only the peaks exceed it—or else, everything will sound squashed.

So the smoothness factor became an inherent consideration and an integral part of pop music. While the limiting of dynamics is no longer so important on CDs, the flavor of the sound still persists. Likewise, for cassette duplication purposes, a compressed or limited tape has every advantage; a much hotter tape copy can be made (minimizing tape noise) and when such a tape is played on car stereos, it helps keep the tape audible on low level passages, even when road noise is high. The Bottom line is that compression/limiting can be very beneficial when applied in the final mastering process.

HOW MUCH AND WHAT TYPE?

That's a question that really can't be answered across the board; it has to be found out experimentally for any given program. Once again, never mind formulas, and let your ears be your guide. Still, a few general guidelines can be mentioned. Peak and RMS type compression or limiting differ widely in their effects. The peak type is most sensitive to transients—that is, the instantaneous changes in dynamics. The RMS type responds to a more average measure of the dynamics and so is smoother and more natural, but then again less absolute in control of the peaks. Both have their uses.

As to when to limit and when to compress, the answer is not pat either. In general when the aim is simply to lop off a few dangerous peaks while preserving most of the dynamics, limiting is the choice. To do this, simply set the ratio up pretty high—maybe 10:1 or higher. This simply means that for every 10 dB in excess of the threshold, the output of the device will only show a 1 dB increase. Needless to say, the threshold should be set fairly high—so that only the peaks exceed it—or else, everything will sound squashed.

Compression differs only in degree. The ratios are usually a lot lower 2:1 or 3:1, and the threshold then can be set a lot lower, also allowing for a more persistent but gentle gain reduction.

A typical use of compression is the following: Assume that we have a song where the vocalist seems a little too predominant. Since he/she is the loudest element when singing, the compressor will be most active, gently pushing the overall gain down a few dB; but when not singing, it allows the background to quickly surge upward.

The illusion is therefore that the vocalist is not so distant from the music. The overall impact of the song is also increased because there is a constant intensity that was not there prior to compression. Knowing this technique, if there is ever a doubt where a vocal should "sit" in the mix, you can feel confident in placing the vocal at the higher level and pulling it under, if necessary, during the mastering stage.

In the next installment of this series, we will deal with *eq*, *exciting* and *gating* as part of the mastering process. db

LARRY ZIDE

A Digital Playroom

Can a home studio compete—technically and financially—in a crowded commercial market? Boston radio producer Jay Rose thinks it's the only way to go.

TWENTY YEARS AGO ROSE WAS an unemployed film editor, staring at a pair of consumer two-tracks in his living room. Today, Rose works out of a brand-new digital audio post room—ten feet from his living room. He says that both technical and business developments over the past few years have combined to make the 'electronic cottage studio' fully competitive. Rose's most recent clients (including Abbott Pharmaceuticals, A T & T, Blue Cross and Group W Broadcasting) seem to agree.

Rose's new room is both a culmination of his career and a reminder of his roots. "I worked at radio stations and film companies for a couple of years, but the first studio I could call my own was a bedroom in my apartment. Tascam, Fostex and similar 'personal use' studio equipment didn't yet exist, but 4-channel home sound had just run its course and I was able to find a used Teac 'quad' recorder and modify it for sync. I traded with a hardware store—they got some commercials and I got cork and acoustic tiles—and I was in business."

A decade later, his bedroom studio had become an advertising-oriented multi-studio complex in a downtown Boston office building. When the building was torn down to make room for a skyscraper, Rose shut the operation down and became principal sound designer at mega-facility Century III (now Editel/Boston). There, he created program openings for NBC and ads for major national clients. But he was also working fourteen-hour days.

"The Clios, Emmy, and New England 'Best of Show' awards were nice, but my kids were growing up and I was missing the 'Daddy' awards," he said. So he left Century III and built a half-inch eight-track facility in his attic (profiled in **db Magazine's** September/October 1989 *issue*).

By last winter, Rose had added a digital workstation, four keyboards and a large mix/pix monitor.

"There wasn't room for clients, and stereo imaging had gone out the window. In fact, even the window had gone out the window: I had blocked over the opening for soundproofing, and the HVAC wasn't adequate for all the equipment. I knew I needed first-class space again."

Fortunately, Rose's home still had rooms to spare. He appropriated a child's former bedroom, and this summer built a comfortable edit room...downstairs.

"My 1972 studio became a nursery when my first kid was born. This 1991 room was my second kid's nursery. Either I've come full circle, or I'm starting the darned thing all over again."

THE FACILITY

Jay Rose's Digital Playroom is modeled after a video post-edit suite rather than a traditional recording studio, because advertising sessions tend to spend more time editing voices and effects than recording them. While the new space is acoustically fine for recording, Rose prefers to book outside rooms for his basic tracks.

"There are some excellent studios in Boston for voice and dramatic recording. Since I'm not tied down to any one facility, I can book the best room—large or small, live or dead—for the project at hand. I do my own engineering, so I can be sure of a technically consistent product." He's been known to use two or three different rooms for the same project, to take advantage of different acoustics.

Rose records direct to DAT, using either the studio's recorder or his Panasonic SV-255 portable. Location interviews and effects are also recorded on the SV-255. He lays down tracks without any limiting or equalization, relying on the DAT's wide frequency response and dynamic range. All processing is done in the monitoring environment of his own studio.

CONSTRUCTION AND LAYOUT

Accurate monitoring was a primary design goal for the new space. The room couldn't be made symmetric—a staircase to an upstairs office cuts into a rear corner—so Rose uses nearfield monitors and plenty of absorption to control room reverb.

"Nearfield monitoring is an excellent idea that's seldom done well," he said. "A pair of NS-10s or Auratones are stuck on the con-

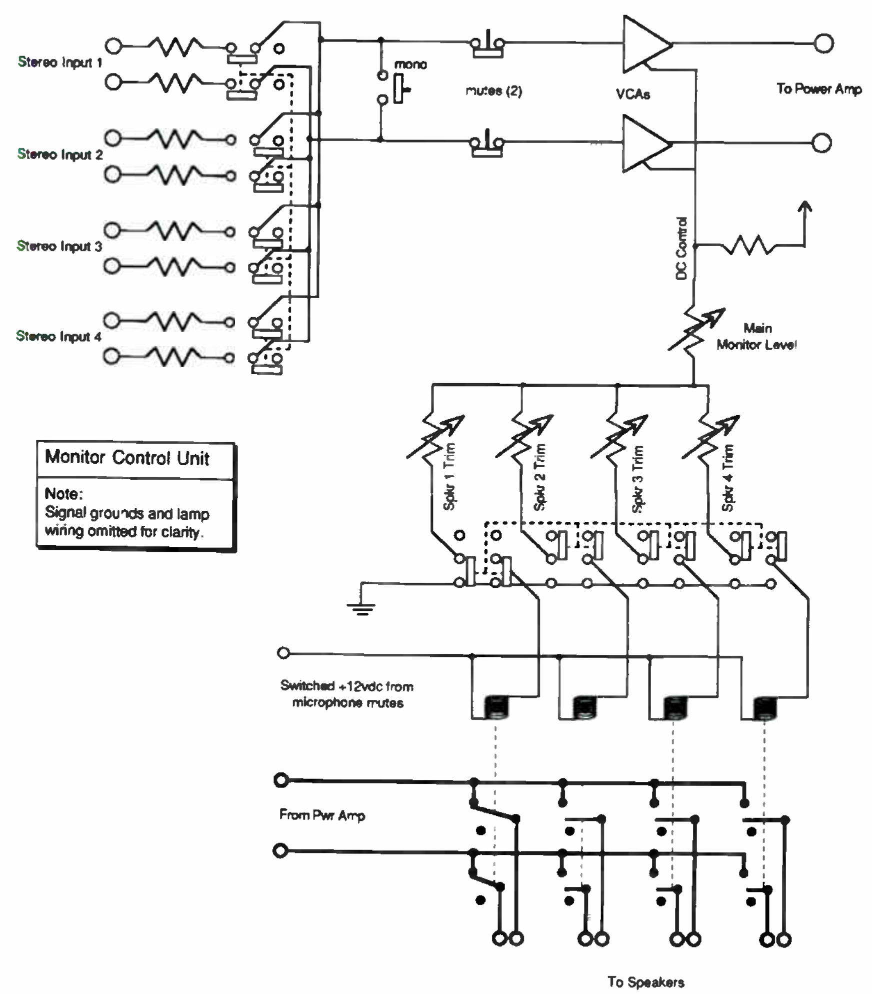


Figure 1. Monitor switching.

sole, and that's it. Sure, you don't hear room nodes any more, but you also don't hear mistakes and problems. You just can't pump full bandwidth without distortion through a seven-inch speaker."

Rose uses JBL 4410s rigidly mounted four feet from the prime mixing position. These speakers, designed for digital mastering, usually aren't considered nearfield monitors, since they weigh fifty pounds each. A custom speaker switcher (see *Figure 1*) switches between them, two home hi-fi speakers, and the ubiquitous Auratones. The switcher uses VCAs and trimmers to compensate for sensitivity differences.

"I can bounce between any pair, including the three-inchers in the video monitor, and the only differences I hear are detail and bandwidth. The stereo field and overall level remains the same."

Room nodes were controlled by slightly dropping the ceiling and adding eighty square feet of strategically-placed two-inch acoustical fiberglass faced with three-inch wedge tiles. The client can get a good idea of the stereo mix almost anywhere along a wide center line.

More space also meant the room could be more comfortable than the attic. Unobtrusive air conditioning, glare-free lighting, and a producer nook with a separate telephone is provided.

EQUIPMENT SETUP

The centerpiece of the Digital Playroom is an AKG DSE-7000 multi-track workstation. This system includes an eight-track hard disk recorder, a 10X4 digital mixer, and exceptionally fast software.

"Advertising clients have little patience for slow workstations. The DSE manipulates a whole production in RAM, so it can scrub and edit much faster than other systems I've worked with." Rose estimates the unit lets him work about three times faster than he did with analog.

A Mackie Designs CR-1604 mixer acts as a pre-selector for the workstation. While the mixer's low cost entered into the purchase decision, size and quality were more important to Rose. "The Mackie packs sixteen inputs into twelve inches of rack space, but still sounds good enough for digital.

Two microphones, two DAT recorders, a CD player, a keyboard mixer, and the DSE's stereo mixed output are permanently assigned to twelve mixer inputs; four other inputs are switchable between analog tape, an Eventide H-3000B, and the DSE's effects outputs.

"The only problem with the Mackie's size is they had to use half-sized faders. But I mix digitally on the full-size faders in the DSE, so this doesn't bother me," said Rose. The output of the mixer is distributed through a bridging network to the two DAT recorders, three cassette decks, and two analog decks (for station dubs). Other effects and processors range from such classics as Gain Brains and an Ursa Major Space Station, to a brand-new Orban limiter and Aphex, Orban, and BBE enhancers. These are connected through the patchbay to sends and returns on the Mackie.

Equipment placement was optimized for advertising-post sessions (see *Figure 2*). A custom-built cabinet surrounds the operator, with DSE and video monitor screens on the acoustic centerline. From the main mix/edit position, the right hand falls on the cueing wheel of a Panasonic SV-3700. The input mixer, monitor control and primary effects are in a rack at the left hand. Further to the left are a patchbay, Macintosh, and synths; on the right are CD, sampler, analog decks and a large CD and DAT library. Since Rose uses the sampler more for design than for music, it was logical to keep it close to the CD player and workstation. However, a MIDI switcher connects it to the multiple synths and Mac sequencer.

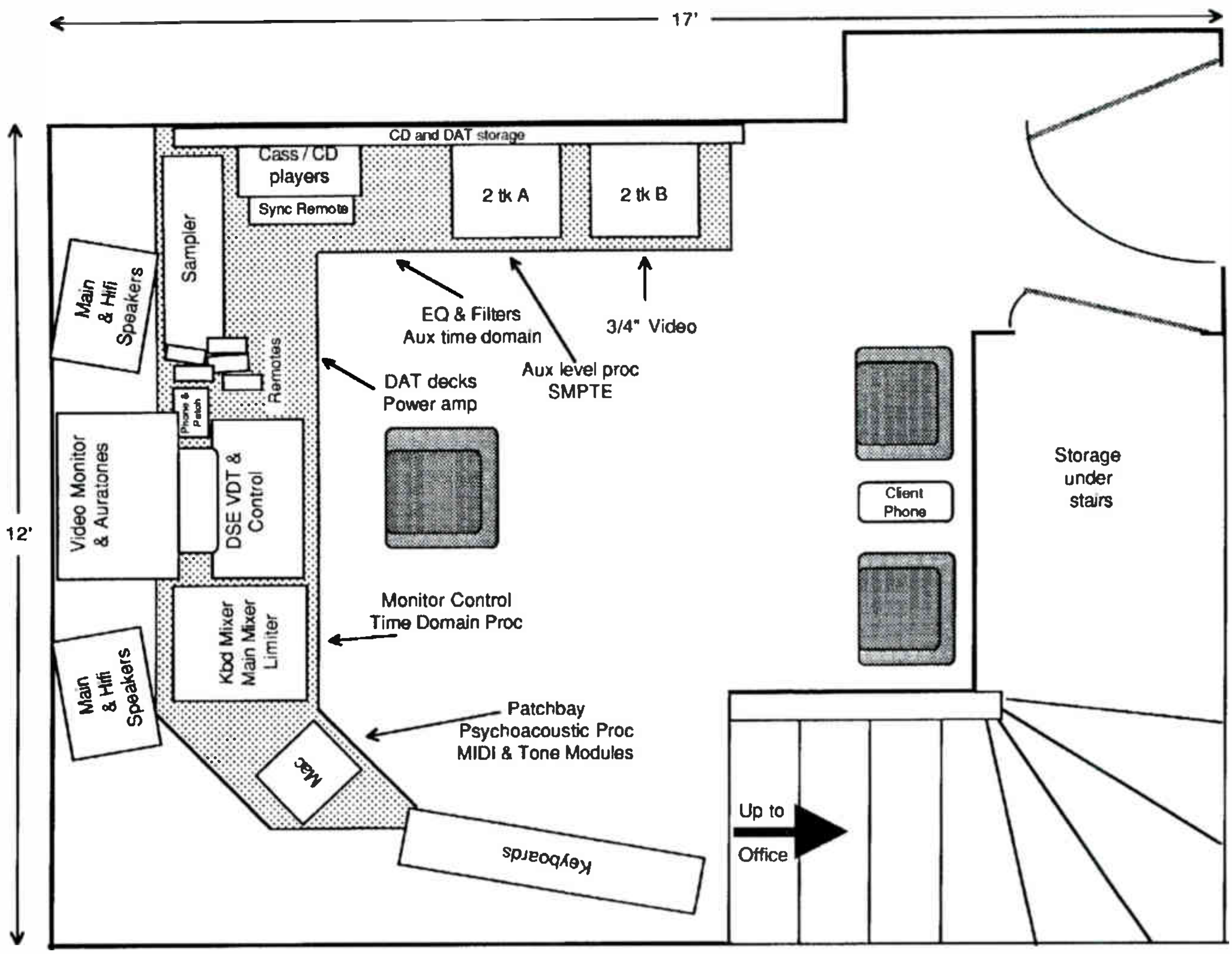


Figure 2. The room layout.

Figure 3. Overall view of the Rose Playroom.

NETWORKING AND CONTROL

Synthesizers are controlled with Dr. T's *Beyond* software. The Mac also handles the sound effects catalog, job tracking and tape labelling chores. It is networked to two additional Macs and a laser printer Rose and his wife, a writer, share upstairs (see sidebar). But aside from MIDI, finding a convenient way to control new digital and classic analog equipment has been a problem. Rose admits the solution still eludes him.

"One DAT has hard-wired control with proprietary serial interface. The other DAT and CD have wireless controllers. The analog and the videotape decks have parallel ports with tach to drive the synchronizer. The DSE has a custom sync connector, while the Mac

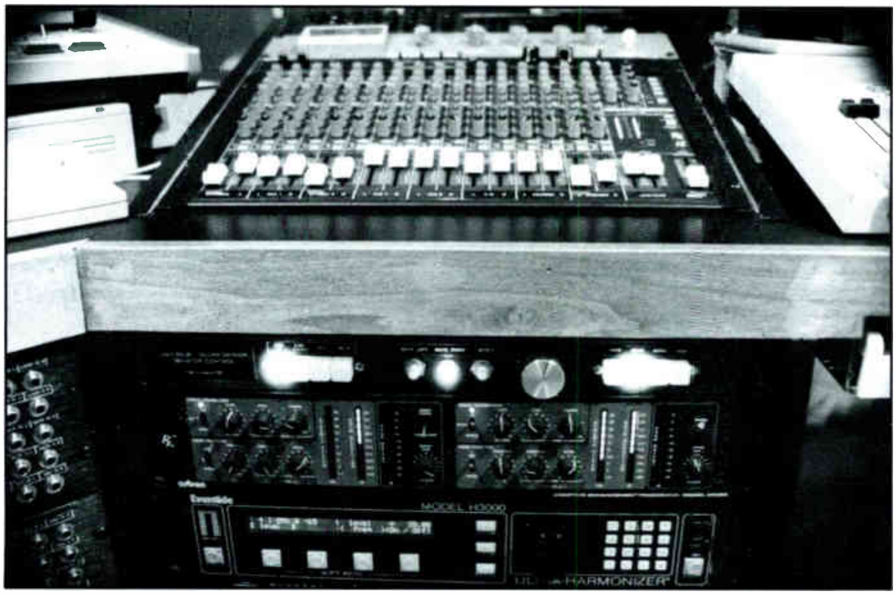


Figure 4. Rose's custom monitor switcher, effects and workstation pre-mixer.

speaks MIDI to the synths and some of the effects. I gave up; I terminated as many lines as I could at a punchdown block. Sooner or later I'll punch together a way to control it all." Rose hopes manufacturers will eventually implement a common RS-232 control language for mid-priced audio gear, much as they've done for high-end video, and intelligent interfaces will be available for classic units.

PRE-PRO AND PHONE-PRO

Unlike some sound designers who tolerate advertising work while waiting for film or record jobs, Rose has built his career around commercials. He says he enjoys the creative partnership involved.

"Some years ago I heard a world-class art director at a photo shoot. She turned to the photographer and said 'If all I hear from you is the camera clicking, next time I'll rent just the camera.' Good agency people hire suppliers who can actively contribute to making their ideas work, rather than just operate the equipment," Rose said.

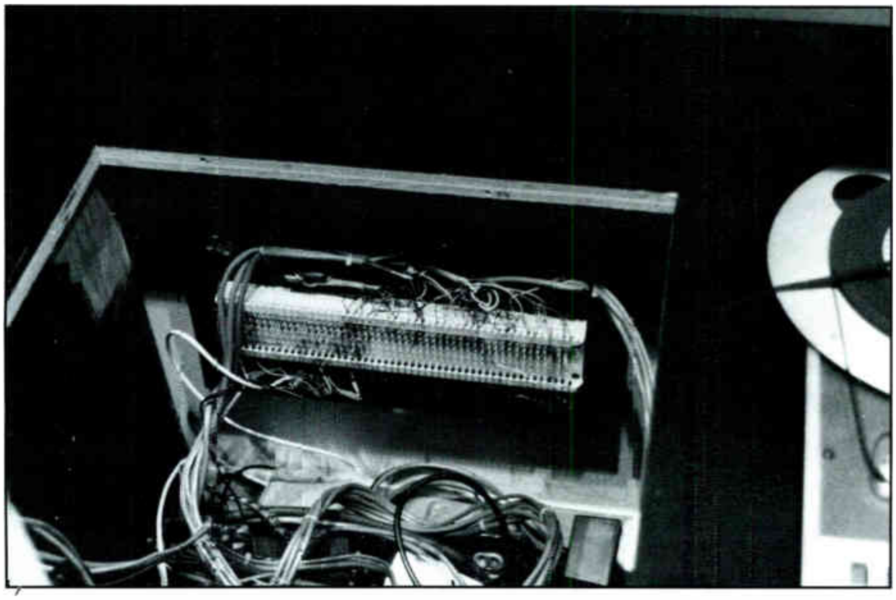
Extensive pre-production and planning is a key to success in the commercial world.

"I talk to the writer about the script long before a session, frequently even before the client sees it. It's amazing how a one- or two-word change can influence sound effects and music, or how the proper effects and music can eliminate the need for some of the words."

With this extensive pre-production, writers are often willing to let Rose do much of the work unsupervised.

"It's more than just trust: it's also having a good phone patch. Agency people can work productively in their offices while I edit, knowing that if I run into a problem or have an inspiration, they'll hear all the options immediately. I frequently do phone-level editing with writers who are across town. Then when it's time to mix, they grab their clients and come over." The speed of the AKG workstation, which lets Rose make final edits while the client is listening over the phone, contributes a lot to this working style. Phone-patch editing in sync with picture is slightly trickier, but Rose claims to work with at least one producer willing to do that...as long as things can be slid during the mix.

Figure 5. Remote lines at a punch-down block (still incomplete at the taking of this photo).



FUTURE OF THE ELECTRONIC COTTAGE

This year is probably not a good one for building a studio in Boston. During the past twelve months, three major local facilities have either down-sized or shut down entirely. Despite this fact, developments of the 1980s have made first-class home operations possible. Computerized audio has reached the point where moderate-priced workstations can do more than a 24-track and console of a few years ago. DAT recorders sound far better than analog decks costing four times as much. Rose believes the trend has as much to do with the music industry as with advances in technology.

"During most of the 1980s, companies like Yamaha and Roland were doing everything they could to put new keyboards under the fingers of part-time musicians. Other manufacturers saw a need for low-cost, good-sounding effects, mixers and recorders. This created a lively and competitive retail scene. If you've got technical know-how and decent capitalization, vendors will fall over themselves helping you build a studio." He acknowledges the phenomenon might be short-lived, as cash and client budgets dry up. "But, hey, that's the beauty of a cottage industry—when things are hectic, I can work fourteen-hour days and still see my kids. When (the days are) quiet, I don't have to look busy for anyone." db

Sidebar: MEET THE LOYAL STAFF

The entire business operation consists of Rose as chief editor, technician, and janitor; and a pair of Macintosh SEs—one in the studio, and one upstairs in the office. Scheduling, accounts payable and receivable, and talent and music reports are handled through custom HyperCard stacks; a commercial checkbook program (Dollars & $ense) serves as the actual ledger. He also catalogs his sound effects in a 7,000-card stack that lets him conduct multiple searches and print out references for each project. Farallon's MacRecorder is sometimes used to create special effects, and T/Maker's Write Now and Silicon Beach's SuperPaint handle general text and promotion duties. (Promotion consists of a quasi-annual newsletter and occasional directory ads.) Carla Rose, Jay's wife and former studio manager, is a novelist and computer writer. She has a Mac IIsi on an adjacent desk in the same office. They have collaborated via AppleTalk on two books (published by Macmillan), numerous ad camaigns, and a sitcom pilot.

TOYS IN THE PLAYROOM

Workstation
AKG DSE-7000

Mixers
Mackie Designs CR-1604
Kawai MX-8R (keyboards)

Recorders
Panasonic SV-3700 digital
Tascam DA-30 digital
Panasonic SV-255 digital
Revox A-700 (2)
Teac V-270 Cassette (3)
JVC CR-6600 (video)

Monitor
JBL 4410
JBL 62
Auratone 5C
Adcom GFA 535

Level Control
Orban 418A Limiter
Orban 424A Compressor
Valley People Gain Brain (2)
Ashly SC-33 Gate
Valley People Kepex (2)
dbx 263 De-esser (2)
dbx 150X Noise Reduction (2)

Equalizer
Orban 621B Parametric
Urei 546 Parametric
Crown VFX-2 Filter

Enhancer
Orban 290Rx
BBE 422
Aphex Type E

General Purpose Audio Manipulator
Eventide H3000B/SE

Time Domain
Alesis Midiverb II
ART 1500 DDL
Ibanex HD1000 DDL
Sound Workshop 220 DDL

Synthesis
Kurzweil K1000 Enhanced
Kurzweil AX+
Roland JX8P
Roland MKS 50
Casio FZ-1 Enhanced

Control
Macintosh SE 4/40 (2)
BTX 4600
Roland MSQ 700
Southworth Jambox 4+
Kawai MAV-8

Wishlist
Reasonable cost time-code DAT

LEN FELDMAN

Stereo TV—Real or Fake?

The title of this article was also the subject of a press conference held by dbx Technology Licensing, one of the splintered sections of the once prestigious dbx Corporation.

LAST OCTOBER 31ST, THAT COMPANY held a well publicized press event in which they demonstrated the fact that "certain models of 'Stereo' TV receivers deliver only pseudo-stereo TV sound." Win Craft, Engineering Manager of dbx Licensing Technology, assisted by Frank Barr, of the Advanced Product Evaluation Labs in Bethel, Connecticut, demonstrated that TV sets labeled as being capable of receiving and decoding "Broadcast Stereo TV" sound, provided little or no measurable stereo separation.

On the very same day, a few hours after dbx launched its public information campaign, Thomson Consumer Electronics, the firm that manufactures and markets the RCA and G.E. brands of TV sets, held their own press conference. The subject of their conference was the introduction and demonstration of their new XS Stereo Sound system. This "new" stereo TV system, they maintained, was so novel as to have been recently approved for the issuance of a patent by the U.S. patent office. The representatives of Thomson admitted that they used the dbx technology in their more expensive sets, but found it more expedient (and more cost effective) to resort to their own "stereo" circuitry in lower price, smaller-screen sets.

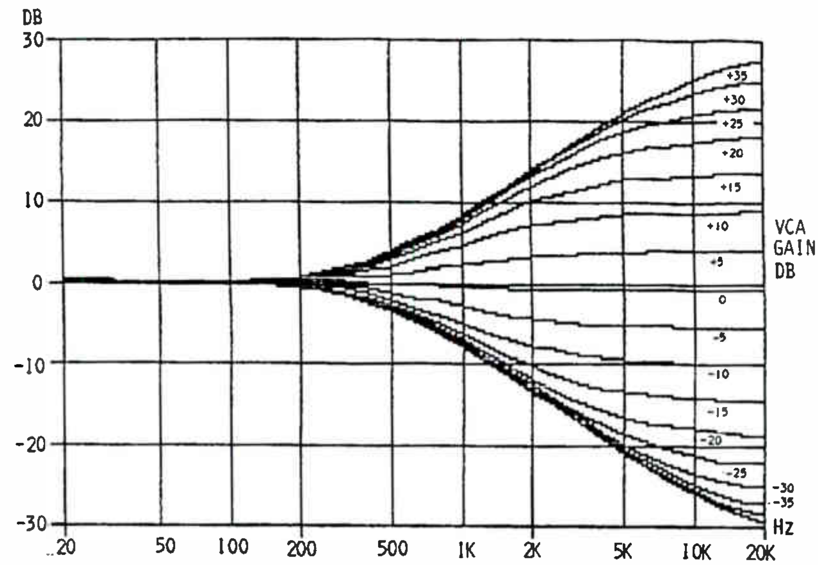
BROADCASTERS' CONCERNS

Broadcasters should be aware of this conflict, and for that awareness it's necessary to look back to the early 1980's, when stereo TV was first sanctioned by the FCC after a series of tests involving several proponents had been conducted under the auspices of the BTSC (Broadcast Television Systems Committee) and the Electronic Industries Association. The system ultimately chosen, sometimes referred to as MTS (for Multi-channel Television Sound), "broadcast TV stereo," "stereo TV," and other such terms. The transmission system selected is one proposed by Zenith Corporation. It is in many ways similar to the way in which stereo FM is transmitted. Specifically, the sum of the two stereo channels is transmitted as the "main" audio channel, while the difference signal (L-R) is transmitted by AM modulation of a subcarrier located at 31.468 kHz (as opposed to the 38 kHz subcarrier used in stereo FM). As is true of FM stereo, this subcarrier is suppressed and only the sidebands are transmitted. Unlike stereo FM, however, the BTSC committee realized that if only the subcarrier approach alone was used to transmit the "difference" signal, viewers/listeners at some distance from the TV transmitter would hear greatly increased noise levels when switching from mono to stereo programming—a condition which has plagued stereo FM listeners who are subject to fringe area reception.

dbx COMPANDING SYSTEM CHOSEN

Accordingly, several companding systems of noise reduction were tested at the same time as the basic transmission systems were being evaluated. The committee unanimously chose the companding

Figure 1. Variable preemphasis/deemphasis. Frequency response for constant gain settings.



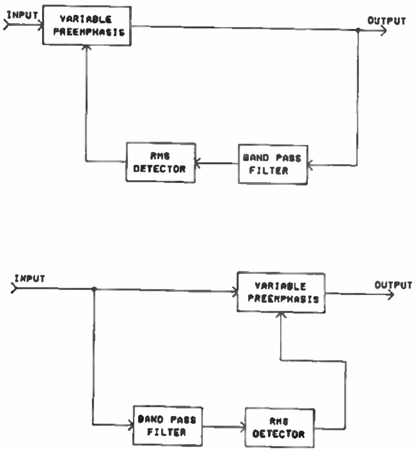


Figure 2. (Above) spectral compression while (below) a spectral expander.

system proposed by dbx, Inc. Essentially, the dbx noise reduction system (which is an integral part of the finally approved MTS broadcast standard) combines the traditional approaches of fixed pre- and de-emphasis and wideband companding with a newer concept of spectral companding. The spectral compressor in the encoder of the dbx system looks at the spectral balance of the input signal ("how much high frequency material is there relative to low-frequency material?") and varies the high-frequency pre-emphasis accordingly so as to increase the potential for masking.

The argument put forth by dbx Technology Licensing is that with such a complex encoding system in place at all TV broadcast stations transmitting stereo, only a complementary decoding circuit in a TV receiver can properly provide the kind of stereo that was present at the input to the system.

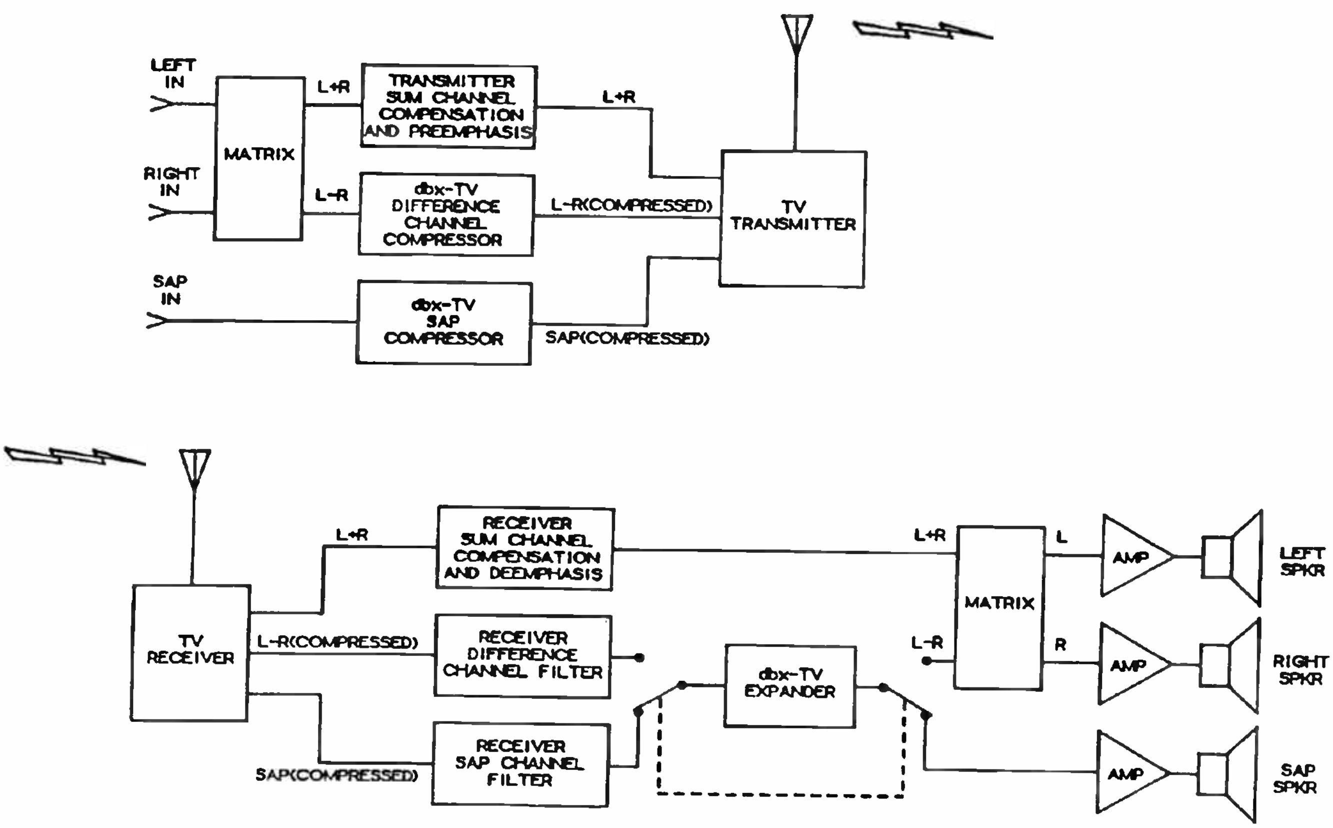
Masking, as defined in this context, can be stated in simple terms, as follows:

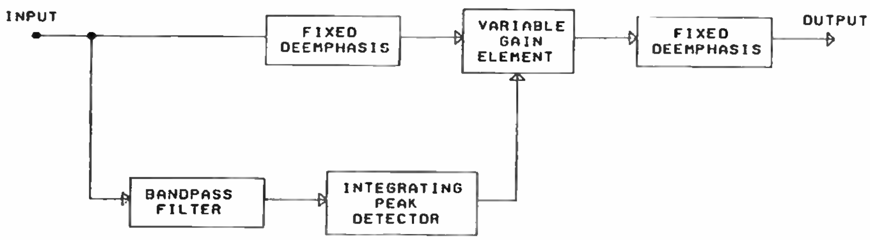
If a desired program signal (music or speech) is loud enough and broad enough in its spectral content, then the ear's attention will be captured by the signal rather than by the noise of the transmission system.

The resulting encoded signal is therefore dynamically adjusted to consistently contain a substantial proportion of high frequencies before transmission, providing masking of channel noise.

During reception, the spectral expander (in the decoder) restores the high frequencies to their proper amplitude. If the original input signal contained predominantly low frequencies, the decoder will attenuate the high-frequency background noise, which will be masked by the low frequency signal itself. If the original signal contains predominantly high frequencies, the decoder will not need to attenuate high frequencies to restore correct frequency response. In this case, the

Figure 3. A block diagram of the complete broadcast system for TV stereo sound.





signal itself masks the noise. By using a spectral compressor at the transmission end of the system, spectral shaping of the input signal is adjusted according to the needs of the input signal, to provide high noise masking at all times. Furthermore, headroom is maintained throughout the frequency range because extreme pre-emphasis is used only when it is really needed. *Figures 1* and *2* show the principles involved in variable de-emphasis and spectral compression/expansion, while *Figure 3* shows block diagrams of the complete audio system involved in MTS stereo transmission and reception.

The argument put forth by dbx Technology Licensing is that with such a complex encoding system in place at all TV broadcast stations transmitting stereo, only a complementary decoding circuit in a TV receiver can properly provide the kind of stereo that was present at the input to the system. Even a slight amplitude error of the "difference" (L-R) signal compared with the unprocessed main channel (L+R) signal will result in severe decrease in measurable separation.

In fact, at the demonstration conducted during the aforementioned press conference, an RCA set (Model 26040 WN) and a G.E. set (Model 25G2534) were shown to have virtually no separation at critical mid-frequencies.

Interestingly, a few days before the dbx conference, Philips Consumer Electronics Company (whose Magnavox Brand Model RS2560) had also been shown to have poor stereo performance) announced that they "will be adding dbx circuitry to those television sets in the line that currently do not have dbx. This will take effect in the second quarter of 1992."

While their demonstration did provide an illusion of stereo, the question remained in the minds of many attendees as to how an incompatible decoding system could possibly restore full stereo to the received signal, or at least a replica of the original left and right signals as supplied to the input lines at the TV station.

Whether this statement was prompted by the advance publicity associated with the dbx press conference is a matter for conjecture.

THOMSON ANNOUNCES THEIR OWN "STEREO TV" SYSTEM

By contrast, the Thomson organization maintained at their own press conference that "There is no requirement for the use of dbx noise reduction on the part of TV receiver manufacturers." Instead, they demonstrated a comparison between one of their sets that was equipped with dbx noise reduction circuitry and one of the same screen size that was equipped with their newly developed "XS Stereo" sound system. Like the dbx system, XS Stereo apparently uses some form of dynamic expansion, as illustrated in the rather sketchy block diagram of *Figure 4*. In addition, there was a suggestion that a system of enhanced stereo illusion used in these sets resulted from a rather old trick of feeding some of the out-of-phase left recovered signal into the right channel and some out-of-phase right signal into the left channel. Thomson maintained that this was a particularly effective technique to be used when speakers are close together, as they would inevitably be with smaller screen TV sets (such as under 25-inch).

While their demonstration did provide an illusion of stereo, the question remained in the minds of many attendees as to how an incompatible decoding system could possibly restore full stereo to the received signal, or at least a replica of the original left and right signals as supplied to the input lines at the TV station. dbx insists that only the complementary dbx decoding circuitry can do the job correctly. Thomson insists that if the listener perceives the resulting sounds as "stereo", than it is stereo, since stereo in the first place has never been carefully or fully defined. So, one question that remains is: if a system measures poorly as far as stereo separation (in dB) is concerned, can that system still claim to be "good stereo" if listeners hear different things happening in the left and right channels. dbx Technology Licensing says "no," while Thomson says "yes!" We would be interested to hear what readers in the broadcast industry feel about this question. Are broadcasters concerned that certain receivers are not reproducing their carefully transmitted stereo audio signals as they were meant to be heard, or are they primarily concerned only with the video signal, caring little about the quality or perceived quality of their audio signals? db

BRENT HARSHBARGER

AUDIO FOR THE CHURCH

● After seeing all the new product releases and line catalogs, and reports from the AES, there is no doubt in my mind that the analog era is soon to be over. Some of you reading this so far would disagree with my previous statement because analog is a mature technology, and for example, in most cases, analog has a better dynamic range than its digital counterpart. Yes, currently there are limitations on affordable digital technology compared to analog, as well as technical limitations; but as we move forward in digital, the price of converters and mass storage is dropping drastically and quickly making R&D advancement unstoppable.

So what does this mean to the worship technician? Well, I just recently looked over the new Yamaha catalog and was hard pressed to find any analog products, not that there were any, but it was a drastic change from the last. This change is wonderful for the church community, as well as the total audio community, but it is change. *Change is difficult*! So we would like to think. It is just a new way of doing things. The hardest part of change is mainly in our thinking. For example, any one who has worked with me knows I love to play with knobs and faders; these two devices are my connection with the audio world. I still, like most people who have been in audio for sometime, have a problem with not having the same interface to audio control (knobs and sliders) in using digital equipment, even though the system is far more flexible. I am currently writing this on a computer. I love my word processor, but recording on the computer for me is still uncomfortable. It isn't that it is harder, in many cases it is easier, but it is different to the way I originally learned and have practiced recording over the years. Therefore, in this segment and several to follow let's pretend we are learning audio for the first time and learn it the digital way. Starting with the basics of digital electronics, covering the digital audio chain, and how that applies to the church today and a few hours from now.

First let me take a minute to review what prompted me to do this segment on digital technology. If you read this column regularly you

Figure 1. Decimal and binary equivalents.

10^3	10^2	10^1	10^0
1000	100	10	1

DECIMAL

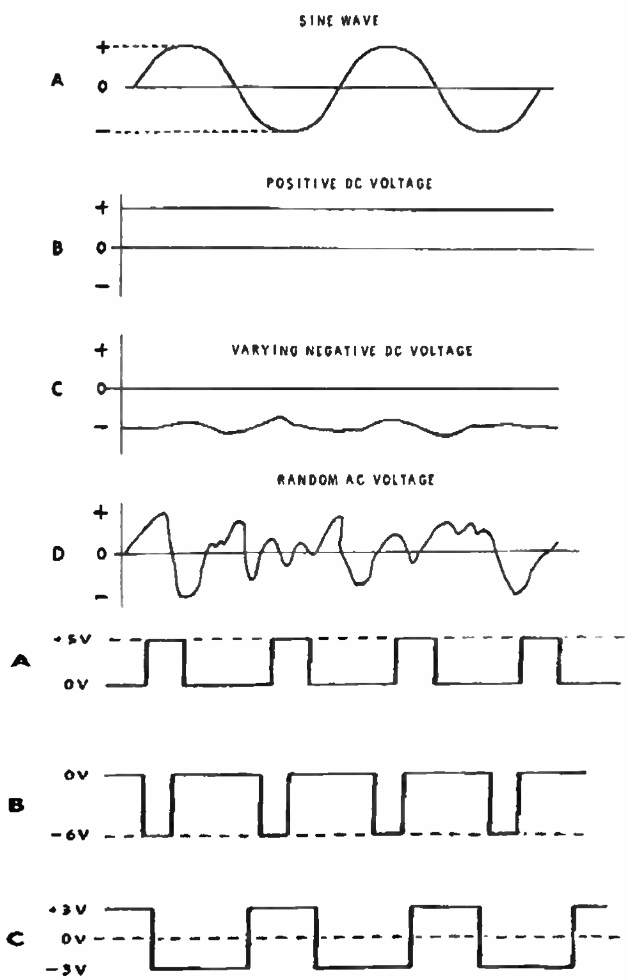
2^4	2^3	2^2	2^0
16	8	4	1

BINARY

2^4	2^3	2^2	2^0
1	0	1	0

16 + 0 + 4 + 0 =20

Figure 2. Types of analog signals (above); types of digital signals (below).



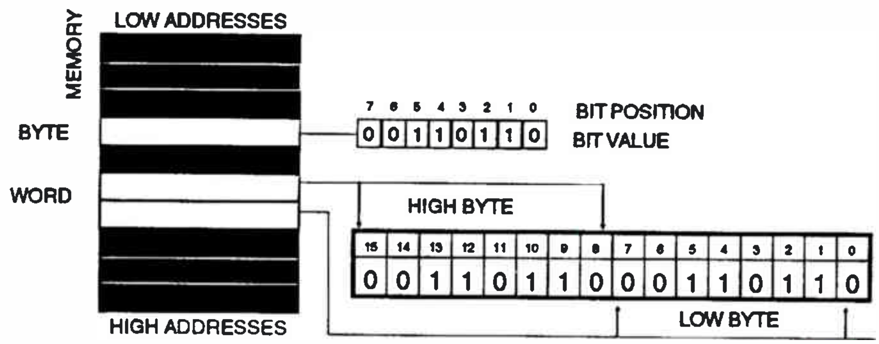


Figure 3. Each box is a single byte, 8 bits=1 byte and two or more bytes make a word.

will remember I very briefly touched on the issue of digital audio; also, there were comments and questions on who, what, why, and how, which lead me to believe that there are many looking for more detail on the subject. The next was seeing all the new product releases, and how many were digital. The ones that come most quickly to mind is the full line of new digital products as I mentioned from Yamaha's newest catalog, the Roland DM80 digital audio workstation, and the New Alesis multi-track DAT, using S-VHS tape and which can be "locked" together for a total of 128 digital tracks. Yamaha had something similar, however using standard DAT tape. Roland and Yamaha had a complete digital recording and mixing system, Roland's being disk based and Yamaha being DAT based. These products are going to change the way we do recording and our music in worship. This not only changes the production process but these systems will also evolve the way we design sound systems for churches, which we will discuss in detail as we cover this topic.

I would like to state before moving on that this is an overview and not a complete course (if you will) on the topics we will be covering, but giving you enough for a working knowledge of terms, processes, and techniques.

The first step, since we are starting to learn audio all over, is the difference between analog technology and digital technology. Therefore, we could define analog as any random but continuously varying voltage waveform. Digital is considered to be a series of pulses or rapidly changing voltage levels that vary in discrete steps or increments, which are between two distinct voltage levels. Now let's define analog and digital methods in terms of devices and ideas, in a way that will simplify the subject using an older style television that had fixed position for the VHF and a variable position UHF selection dial. The VHF is digital in nature because it can only be in one discrete position and could be changed into 13 unique states. The UHF channel selector in operation is analog because you have to rotate the dial back and forth to get the best reception. To further illustrate a house hold lamp is digital because it is either on or off and the volume control on the television is analog because it is variable to an undetermined amount of levels ranging from off to extremely loud.

Because we are to use the decimal number system, the binary number system is difficult for us to use.

All digital circuits, instruments, and systems work with numbers that represent specific quantities. For example, the analog voltage measured by a digital voltmeter is converted into digital form and displayed as a specific decimal number. Digital devices accept input numbers, process them and generate them with number outputs. The decimal number system is the system with which we are most familiar, which is a number system base of 10. Digital systems use a binary number system with a base of two, since only the digits or bits 0 and 1 are used in forming numbers (or data). Because we are to use the decimal number system, the binary number system is difficult for us to use. For the machine, binary is the simplest, most economical, and fastest, because it only has two states: *on* or *off*.

MORE ABOUT NUMBERS

The decimal and binary number systems are positional or weighted number systems. This means that each digit or bit position in a number carries a particular weight in determining the size of that number. The decimal system has positional weights of units (tens, hundred, thousand, etc.) which are powers related to the base of 10 (10 to the first power is 1 or unit, 10 to the first power is =10, ten to the second power is =100, and ten to the third power is=1000 etc.). Binary numbers work the same way. Each bit position carries a specific weight. These weights from right to left are 2 to the 0 power=1 *, 2 to the first power=2, 2 to the second power=4, and 2 to the third power= 8.

*(Note: *Any number with an exponent of zero is equal to one.)*

See *Figure 2* for more detail.

Thus far we have only discussed whole numbers or integer quantities. Although fractional numbers are possible, we will not discuss it at this time. Converting decimal numbers to binary and vice versa is relatively easy, but why twist your mind when you can buy a $20.00 calculator that automatically does it for you at a touch of a button? So I won't go into that either.

Binary numbers are also referred to as binary words. A 16 bit binary number is also a 16 bit word, in other words, it takes 16 bits (or digits) in a group to make a 16 bit word. The size of this word determines the maximum magnitude and resolution with which numbers can be represented. (See *Figure 3.*) More next issue. db

TEK

TEXT

Talking About Microphones—A Guide To Microphones For Project Studios and Home Studios

This article is offered as a basic tutorial on mic usage. It is based on a booklet published by the Sennheiser microphone people and will explain why the examples given are from that company.

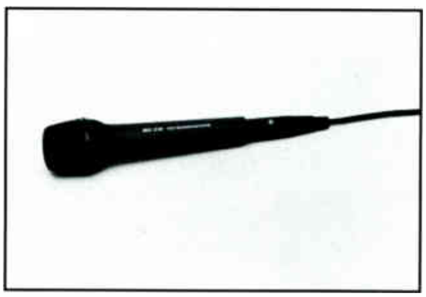
MICROPHONES ARE AS critical an interface between sound and media as any other component in your studio, and have as much to do with establishing your sound as the multitrack recorders, consoles, outboard processors, samplers and other gear you've put together for your project or personal recording studio. There's a lot to know about microphones, and we can't make you an expert in a matter of a few pages, but this article will help you understand the fundamentals of microphones, more about the kinds available and how best to choose them and use them.

MICROPHONES AND THE PROJECT STUDIO

Project studios have become a powerful force in the modern recording industry. The project studio by definition is a specialized facility; it does not have to be all things to all clients. It could be sound effects in New York, country music demos in Nashville, industrial video scores in Chicago and film sound design in Hollywood.

Project studios don't need as many microphones as traditional recording facilities, but their need for an inventory of appropriate microphones for their specific needs is no less intense. Project studios doing jingle work need to be able to capture group vocals perfectly; country demo producers want crisp, well-defined acoustic guitars, audio-for-video composers want a bright, brash electric guitar to complement synthesizer pads; sound effects creators need highly accurate microphones for sampling. Your microphone choices are as much a reflection of your style and talents as the rest of your studio and your client list are.

Figure 1. Photo of a dynamic microphone.



THE BASICS

There are two fundamental microphone types: *dynamic* and *condenser*. The dynamic (*Figure 1*) microphone's operation is based on a conductor, in the form of a coil, moving in a magnetic field. That movement is imparted by a diaphragm, which responds to incoming sound. These physical movements generate analogous electrical signals.

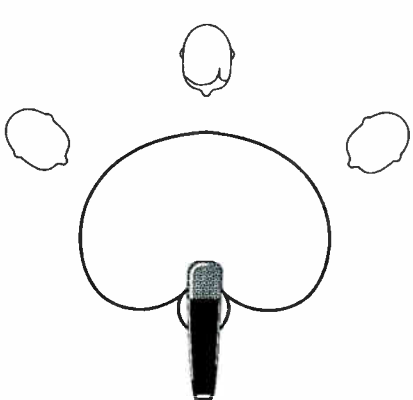
The condenser (*Figure 2*) microphone usually has a much lighter, thinner diaphragm which is one of two plates in a capacitor, to which

Figure 2. Photo of a condenser microphone.

an electrical charge is applied. (The second plate is known as the back plate.) The change in distance between the two plates as sound waves move the diaphragm is reflected as a change in capacitance and produces an electronic signal equivalent to the sound it's recording. This type of microphone responds to transient signals more rapidly. Condenser microphones require an outside power supply—either a battery or 12-to-48-volt phantom power from a console or mixer.

For both types of microphones, their usefulness in given applications is partially based on what's called their *polar pattern* (*Figure 3*), their response to sounds coming from different angles or directions. Tight patterns are called *cardioid* or uni-directional. Cardioid microphones reject or attenuate sound from the sides and behind, concentrating on sounds from directly in front of them. Progressively tighter patterns are referred to as hyper-cardioid and super-cardioid. Cardioid patterns are especially useful when recording more than one instrument at a time, allowing each microphone to concentrate on a single sound source while rejecting other sounds. (See *Understanding Polar Pattern and Frequency Response Diagrams* later in this article.)

Figure 3. Polar pattern illustration; an example of a cardioid pattern.



Omni-directional microphones pick up sound equally from all directions. These types of microphones are good for picking up room ambience when recording live drums or guitar amps, and are also useful for group vocals.

Bi-directional or figure-8 patterns are useful for situations such as recording two vocalists face to face, and are often used in stereo microphone techniques.

There is more interest lately in *stereo microphones*, which have two separate elements mounted in a single assembly. These types of microphones are increasingly used in sampling stereo sound effects.

THE RIGHT TOOL FOR THE JOB

Now that we know the types of microphones available, the next step is choosing the appropriate ones for each application and maximizing their potential in each application.

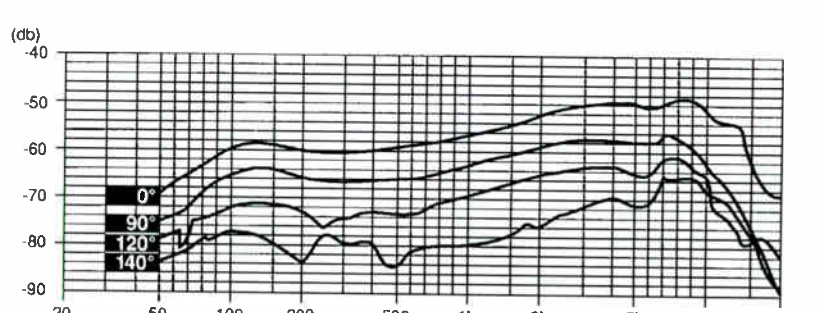
There are a few basic points to be aware of about *choice* and *placement* of microphones, regardless of the specific application. A microphone's frequency response will affect its coloration of the sound source. Certain microphones are weighted around a specific frequency range (called a presence peak, measured in decibels and usually located at about 3 kHz to 5 kHz in a dynamic microphone, sometimes slightly higher in a condenser microphone) and will enhance the sounds at those frequencies, (see *Figure 4*). Thus, microphones that enhance in upper frequencies are more suited for sources like vocals and acoustic guitars; microphones that enhance mid-range frequencies work well with guitar amps and drums. Conversely, other microphones have no such peak and are referred to as *flat-response* microphones, with a variation in frequency response of less than 3 dB.

Where a microphone is placed in relation to the sound source goes a long way towards how it sounds on tape. The further away it is, the more room sound coloration ("ambience") it will add. The angle at which it's placed relative to the sound source will affect how loudly certain frequencies get to the microphone. Many producers and engineers will tell you that placement is everything with a microphone. There are no real rules, only some guidelines based on experience. Experimentation is the essence of the art of microphone placement.

For lead vocals, uni-directional microphones with good high-frequency response are the best overall choice. A good example of this is Sennheiser's super-cardioid condenser MKE 4032-P3, which features a low-roll-off switch allowing you to use it very close to the sound source. Use a windscreen to mitigate pops from the singer's lips. For positioning, keep the microphone level with the mouth and start about five inches away. For vocals in general, find a spot in the room away from hard reflecting surfaces that will bounce back into the microphone.

Group vocals benefit from dynamic microphones, particularly very powerful vocal parts. The MD 518 is an excellent example of this type of microphone. Position groups of three or four singers in a

Figure 4. Frequency "presence peak" diagram.



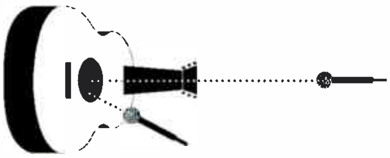


Figure 5. A guitar microphone placement illustration.

quarter-circle with the microphone at about eye-level and pointed slightly down, about a foot away from the center of the group. High sensitivity in the microphone isn't a necessity since there'll be three or four pairs of headphones leaking into the room. Quieter "oohs" and "ahhs" can be handled by condenser microphones.

Acoustic guitars are good candidates for condenser microphones. While an acoustic guitar can be micéd with a single microphone, you'll get a better result if you use two instead. Place the first about six inches away from the front of the guitar, aiming it at a point between the sound hole and the bottom of the neck. You can place it a bit closer in if the part is to be picked; back it off a bit for a hard strumming part. Keep the microphone slightly off-axis from the sound board of the guitar; a 90-degree angle between the two may cause a buildup of standing waves which can distort the sound. Place a second condenser microphone about three feet away pointed at the guitar to pick up the overall sound of the guitar as well as some of the room ambience (see *Figure 5.*) Now mix these two signals together to a single track while at the same time listening for any phase problems between the two, which you would hear as a decrease in bass frequencies.

Guitar Amps are also good candidates for multiple microphone approaches, but the high sound pressure levels (SPL) involved here make dynamic microphones preferable. Place one microphone about six inches from the speaker slightly off-axis to catch the punch from the amp. The Sennheiser MD 409 is perfect for this application because its large membrane reproduces high SPLs without distortion or coloration. Place a second dynamic microphone about three feet back aimed squarely at the amp to catch coloration and room ambience. A third microphone can be placed up around ear level; this not only provides a second ambience perspective but also mimics what it would sound like to the guitar player standing on stage.

Drums: For *snares*, try a condenser microphone about three inches above the rim and pointed across the drum and a dynamic microphone, like the Sennheiser cardioid MD 421, below pointed at the snares. For *toms*, the Sennheiser MD 421 dynamic microphone has become pretty much a studio standard. Place it about three inches above and slightly inside the rim (see *Figure 6*). The higher frequencies of *cymbals* call for condenser microphones. For *high-hats*, aim them from about six inches away at the spot where the stick strikes them. For *overheads*, set up a pair of condensers (preferably of the same model) about four inches above to capture the stereo image.

Brass: Try to face the bell towards a hard reflective wall. Such reflections are welcome with brass instruments and are an integral part of the sound. Place a dynamic microphone, like a Sennheiser 421 or 441, a couple of inches above the bell and a condenser microphone about three feet away at ear level to catch the reflections.

Acoustic Keyboards: Pianos like to be micéd in pairs and reproduced in stereo. Try two condenser microphones angled downward into the piano housing about three inches above the sound board.

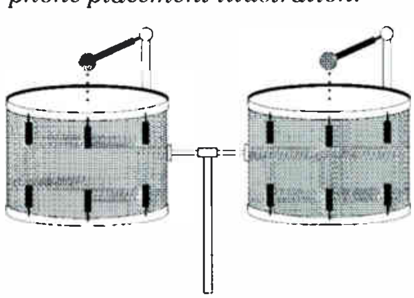
A FEW TIPS WHEN BUYING MICROPHONES

• Good, solid construction is the first cue. Studio microphones don't experience the handling shocks of live performance, but they can get knocked around and a well-put-together microphone can take it without failure.

• Check the pattern graphs. These are graphic illustrations of the microphone's pickup pattern. Find out which frequencies are best attenuated from the sides and back. For good isolation of the desired sound, you'll want at least 10 dB of attenuation of any unwanted frequencies coming from the back or sides of a uni-directional microphone.

• If possible, record a few sources like vocals and an acoustic guitar to tape without any equalization and listen for how well the microphone reproduces the original sound source.

Figure 6. A tom-tom microphone placement illustration.



HOW TO READ A MICROPHONE SPEC SHEET

Frequency Response—The frequencies that a microphone reproduces. For instance, the MD 518's range is from 50 Hz 16 kHz, perfect for guitar amplifiers and group vocals.

Directional Characteristic—Indicates the pattern of the microphone: cardioid, super-cardioid, omni, etc.

Rejection at 1 kHz—Indicates how much the sound is attenuated at that frequency compared to the pick-up along the center axis of the microphone. This is a central function of the polar pattern. The rejection value should be at least 18 dB at 1800 for cardioid microphones and at 1200 for super-cardioid microphones.

Sensitivity—The rating of a microphone's voltage output for a particular SPL input. Higher sensitivity generally provides a better signal-to-noise ratio.

Impedance (Nominal)—The lower the better. You want an impedance of at least five times lower than that of the console you're connecting the microphone to. The term "nominal" refers to an average since impedance changes slightly over the frequency range. A lower impedance value also means you can run a longer cable without loss of signal integrity.

Connector—Type of cable required. Virtually all professional microphones use a 3-pin XLR con-

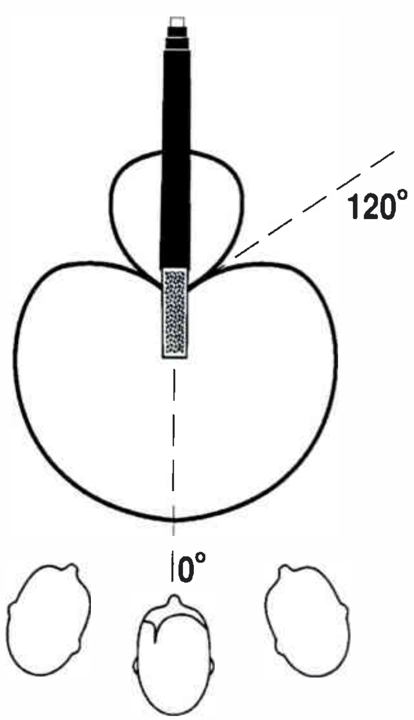


Figure 7. MD 441 Polar pattern.

nector, known as a balanced configuration.

Insensitivity to Magnetic Fields—The ability to reject electrical interference, mainly from power lines. You want to see a maximum value of 5 µV (pronounced "mu- or micro-volts") per 5µT (for "Teslas").

UNDERSTANDING POLAR PATTERN AND FREQUENCY RESPONSE DIAGRAMS

The polar pattern diagram can tell a lot at a quick glance. Here's an explanation of the information it contains. The different thickness lines each represent a frequency, and you'll notice they're spaced an octave apart. The circles are divided into degrees away from the front of the microphone at different decibel levels. Where the frequency lines move towards the center of the chart indicates the level of attenuation (in dB) of those frequencies at certain points from the front of the microphone.

For instance, the pattern above shows the attenuation characteristics of the super-cardioid dynamic MD 441 (see *Figure 7*). Notice that attenuation of all frequencies is sharpest at 1200 away from the front. This is called the "null point." If you were micing two drums, for instance, you would want to position the 441 on the first drum so that sound from the second would come at it from 1200 from the 441's front, thus minimizing the amount of leakage from the second drum into the 441.

The frequency response graph of the MD 441 (*Figure 8*) is the perfect illustration of a "flat-response" microphone which adds no coloration at all to the sound. Compare that to the frequency response of the MD 518 (*Figure 9*) whose "presence peak" begins just before 2 kHz, adding emphasis to the upper midrange frequencies.

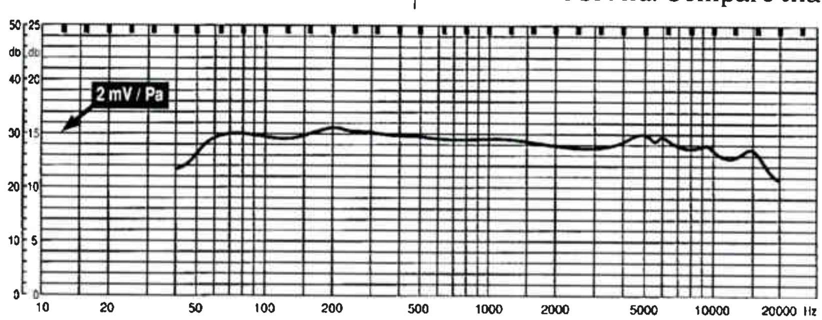


Figure 8. MD 441 frequency response graph.

A FEW USEFUL DEFINITIONS

Balanced Connection—Mic cable connection in which both conductors are electrically symmetrical with the case and cable shield (ground). This type of connection offers a high degree of freedom from interference and hum, even in long cable runs, because unwanted signals have almost identical effects on both conductors and cancel each other out at the balanced input. (Compare with *Unbalanced Connection*)

Cardioid Pick-up Pattern—The usual shape of the pick-up pattern of uni-directional microphones. Best attenuation occurs at signals arriving 180 off axis.

Impedance—A measurement that specifies the resistance of an electrical device using alternating current, measured at its terminals. It's important that items like microphones and headphones have relatively matched impedances, to produce what is known as a "no-load condition." For instance, the input impedance of the amplifier should be significantly higher than the impedance of the microphone being used with it.

Proximity Effect—A low-frequency boost that occurs in uni-directional microphones as the sound source gets closer.

Signal-To-Noise Ratio—A measurement referenced to a fixed value that equates to 94 dB. This measurement provides an indication of the inherent noise level of a microphone or other piece of audio equipment.

SPL—*Sound Pressure Level*—A measurement of the force of air moved by sound.

Unbalanced Connection—In this type of microphone connection, only one conductor wire carries the signal voltage with the cable shields serving as the return line. Its advantage is a higher resistance to interference from the shield and the option to be able to connect the microphone to balanced inputs, as well as unbalanced ones. (Compare with *Balanced Connection*)

CARE & MAINTENANCE

Care for microphones as you would any other valuable piece of studio equipment. Keep them as dust-free as possible, especially their elements. Wipe them off with a soft cloth slightly dampened if necessary; never use solvents or harsh detergents. Use a nylon pop shield or foam windscreen when recording—it not only mitigates pops but also helps keep corrosive saliva off the elements. Store them in their cases and mark the outside with the make and model number on a piece of masking tape for

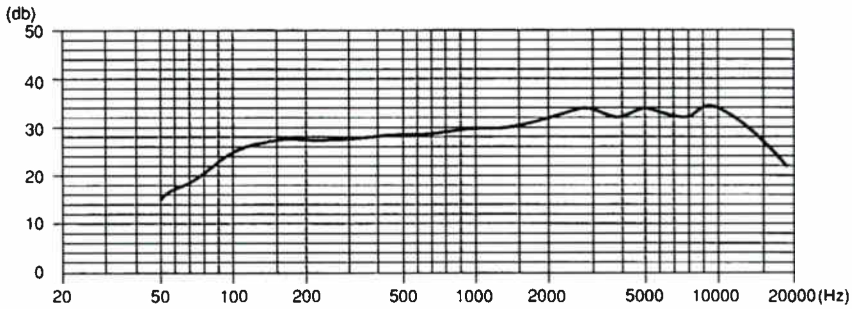


Figure 9. MD 518 frequency response graph.

quickly picking the right microphone during a session.

ACCESSORIES

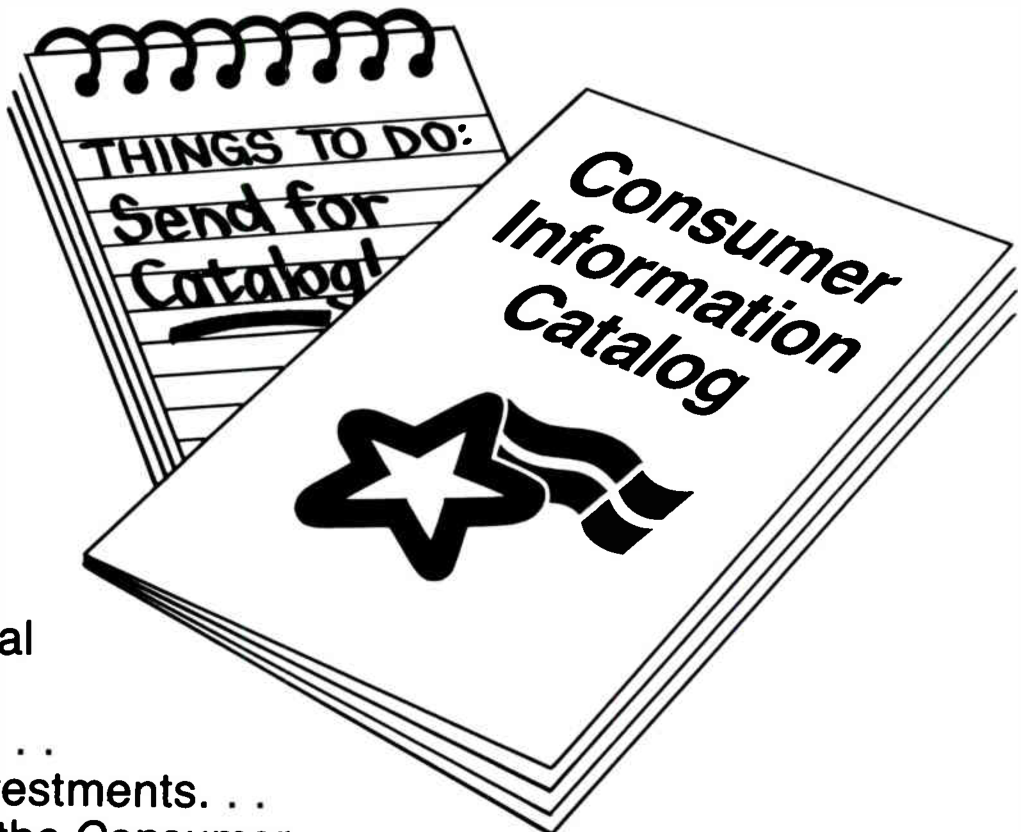
Adaptors—They are usually supplied with the microphone. One end will adapt to a standard microphone boom or stand, the other will fit the specific microphone. There are also shock mounts, tightly strung elastic bands which isolate the microphone from vibrations that travel through the stand.

Windscreens or pop filters—Important for keeping pops off tracks and saliva off microphone elements. Foam types are the standard and can be conformed to custom-fit the microphone head. Foam windscreens can cut out certain high frequencies, however, and for vocals many engineers prefer sheer mesh pop screens.

Cables—Microphones come in two impedance configurations: high and low. High-impedance (unbalanced) cables have quarter-inch jacks and carry a line level similar to that of a synthesizer. Low-impedance (balanced) microphones are the preferred choice for professional recording. These three-pronged cables are quieter and allow for a longer cable run.

Stands—Heavier bases on microphone stands help keep mechanical vibrations to a minimum during recording; tripod bases are lighter and easier to store. Choose what works best for your studio. Booms are essential; there are a number of instruments, such as violin and flute, that require the microphone stand to accommodate the musician, not the other way around. A retractable boom with good reach should be able to steadily position a microphone looking straight down at a height of six feet. db

Put Our List On Your List



Our list can help you do the other things you have on your list. Such as buy a car. . . estimate social security. . . start the diet. . . check out investments. . .

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ZACHARY JAQUETT

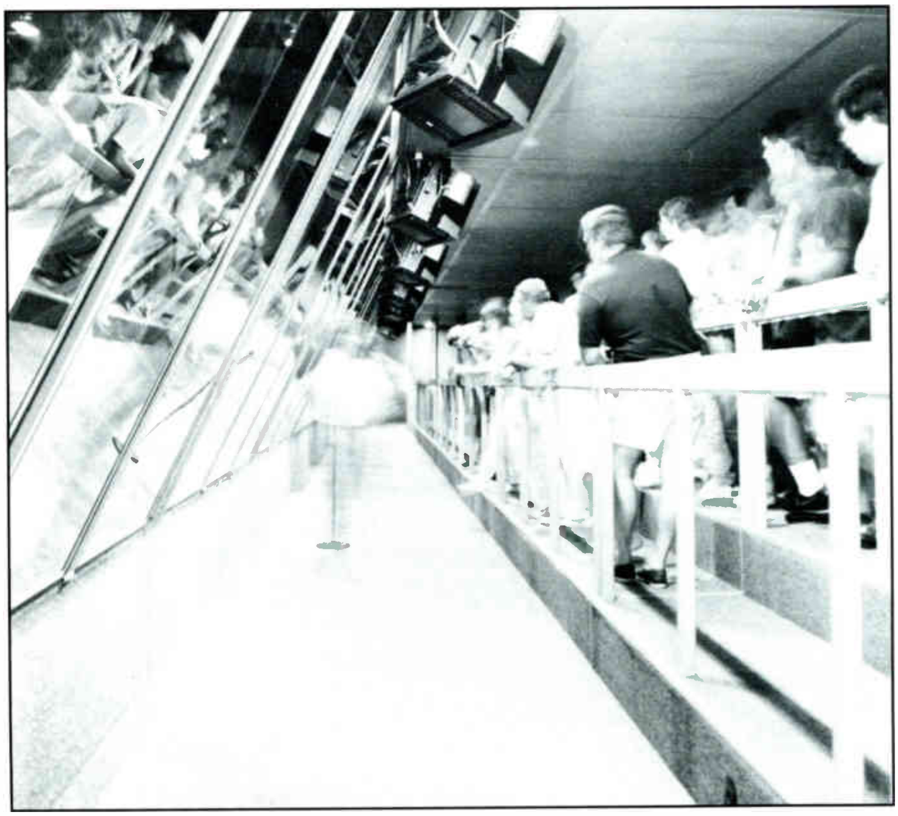
Acoustical Viewing Tube at Universal Studios in Florida

Universal Studios began as a pipe dream—literally. Seems as though pioneer filmmaker Carl Laemmie came to California and glimpsed a sign on a truck reading "Universal Piping Company." Since Laemmie had every intention of launching the biggest filmmaking studio in the world, he appropriated the "Universal" title for the studio he opened in 1915.

SEVENTY-FIVE YEARS LATER, THE company marked a three-quarter century milestone with the opening (near Orlando) of Universal Studios Florida. Ranked the largest and most technically-advanced motion picture studio outside of Hollywood, the installation occupies 444 acres and cost in excess of $600 million. Among its facilities are six mammoth sound stages, five covering 16,500 square feet and the sixth 22,000 square feet with expansion capabilities for up to five additional sound stages of this scope.

Designed from the ground up, these structures providing producers, directors, writers and actors with state-of-the-art environments for creating entertainment are today a major attraction also for the visiting public.

Figure 1. The VisionWall windows are made with one-way glass. The visitors get a full view of the production, while the production teams do not see those watching them. This eliminates any distraction for cast or crew.



VIEWING TUBE ACOUSTICAL PEOPLE MOVER

Not only can people tour the complex, but with the opportunity to view close up the inner workings of film and television production, they can get the feeling of actually participating in the magic of moviemaking. To give visitors this idea of involvement as they pass through a sound stage, soundproof "Viewing Tubes" enable hundreds to move in and out of the sets during shooting of a film under production without the noise of their passage reaching the set to disturb cast and crew.

The Viewing Tube concept originated within Universal's Studio Operations Group whose staff designed the unique structure. Industrial Acoustics Company of New York implemented the design with acoustically-rated components engineered to meet the project's requirements. A viewing tube is 112-feet long, and includes fourteen interconnected but movable sections, each measuring 8 feet x 14 feet x 10 feet and featuring IAC Noise-Lock VisionWall construction. An overall tube assembly is commodious enough to comfortably accommodate up to two hundred and fifty people, but visitor groups are usually limited to a manageable fifty shepherded by a Universal tour guide.

Segmented construction with acoustical components enables relocation of Viewing Tubes from sound stage to sound stage; this modularity gives production teams the option of having or not having visitors. Design modularity is implemented by the Moduline System consisting of modular components including wall and roof panels, doors, widows, and built-in silenced ventilation systems that integrate readily with each other to constitute a versatile building system whose balance, integration, and durability enable each Viewing Tube to be taken apart, moved, and reassembled with no loss of acoustical or structural integrity.

Viewing Tubes' controlled acoustical environment (utilizing the Moduline system) was created taking into account a group of basic sound-conditioning factors. These are: noise propagation and flanking paths, transmission loss characteristics of walls and ceilings, compatibility of windows and doors, and the noise control of air conditioning systems, trim and hardware.

Thus do Viewing Tubes at Universal Studios Florida accomplish two objectives simultaneously—protecting the sound-stage environment from the intrusion of visitors' noise while providing individuals with optimal opportunities to enjoy personally the excitement and glamour characterizing film and tv production. db

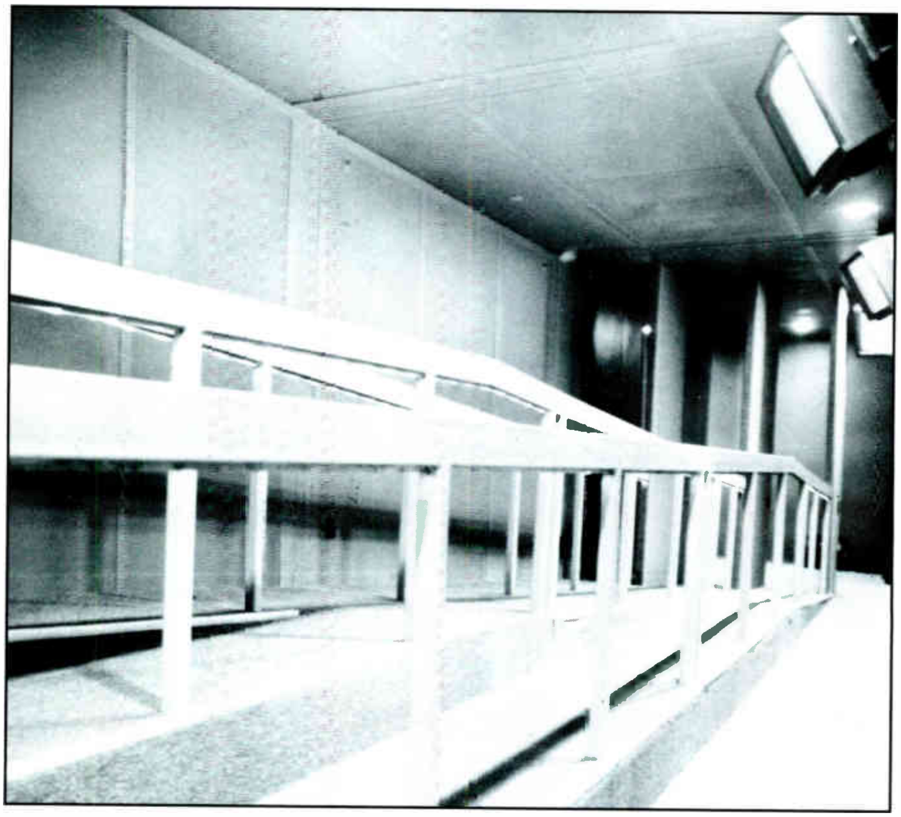


Figure 2. IAC Moduline System components (note back wall) are only 4-inches thick yet provide the sound conditioning required. The light weight facilitates relocation of the Viewing Tube.

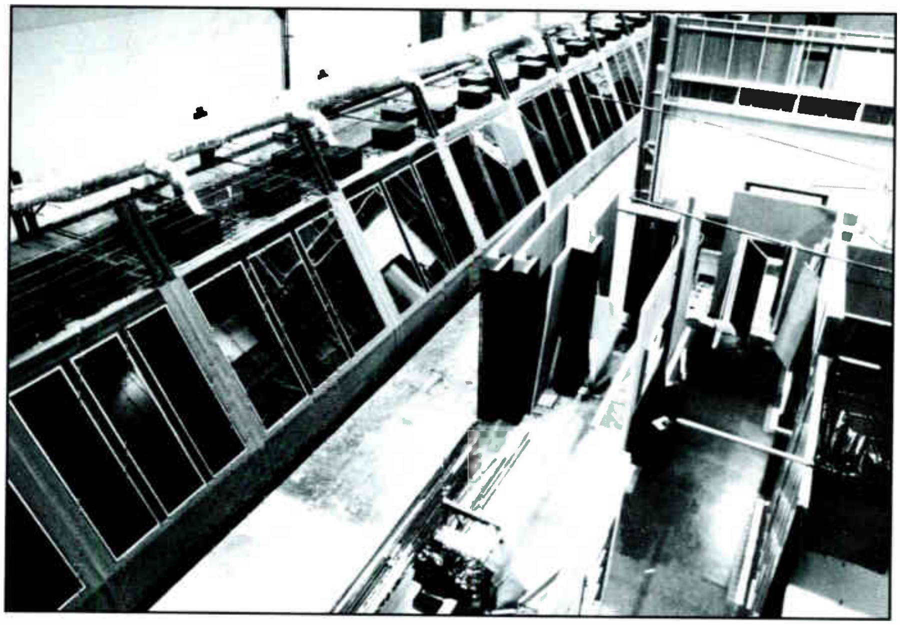


Figure 3. The Viewing Tube as seen from the production floor.

Acoustic Performance—Viewing Tube Components

Transmission Loss, dB

Center Frequency, Hz 1/3 Octave Band	63	125	250	500	1000	2000	4000	8000	STC
Panel	27	30	32	41	50	59	67	71	45
Window	24	33	36	48	50	56	58	64	49
Door	24	28	39	46	50	53	55	64	49

ARTHUR M. NOXON

Trapping Bass in Your Project Studio, Part II

Boom Busters

There seems to be a popular misconception about the role of bass traps. The uninitiated often say, "I want to kill my resonances with some bass traps". When absorption is added to any resonant circuit, be it electronic or acoustic, only the rate of energy drain from the system is increased. It must be stressed, that from a practical basis, absorption can never eliminate resonance: resonance exists *because* the room exists. Absorption can only reduce the strength and sharpness of the resonance, (its "Q") but not eliminate it.

Sound will build in intensity until there is a balance between the power delivered into the room and the power absorbed or leaked out of it. Increased absorption means the room reaches its peak sound level more quickly. Why? Because the equilibrium sound level attained in the room is lower and not because the energy rise rate is any more abrupt. Adding absorption, however, increases the sound decay rate in the room.

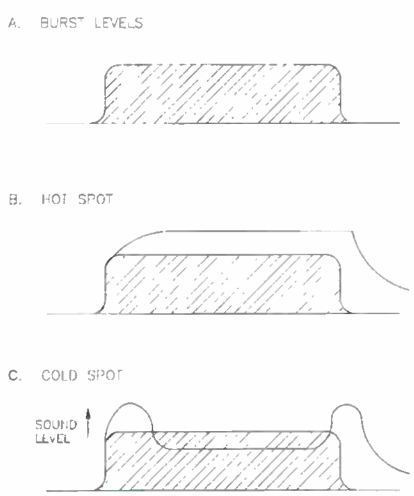


Figure 1. Absorption corrects response.

Other benefits are noted at the cold spot. The resonant field strength is weaker overall due to the added bass absorption. The reverb field's reverse phase cancelling effect of the direct wave from the speaker is less strong. As a result, the cold spot "warms" up and the pulses at turn-on and off are accordingly diminished.

As to the coloration effects, added absorption reduces the "Q" of room resonance, the sharpness of its response. Low "Q" rooms lose attack transient and sustain distortion. The beating effects have disappeared and the tone in the decay is the same as that of the driven frequency.

Absorptive damping of room resonances, as we have seen, will improve the dynamic response characteristics of the room. It is quite clear by now that it is the *room* that we listen to in the lower registers. Accordingly, the better behaved the room, the better the track and mix will sound.

A caution needs to be noted at this point. Nearly all recording engineers have access to an RTA, typically ⅓ octave bands. Their experience with electronic equalization, particularly parametric, leads to the desire to see a flat room acoustic response curve. Good luck! It is always a surprise to realize that dynamic transient stability in the room can be developed to satisfaction, and yet the ⅓ octave RTA shows less than 1 dB improvement. Just as it is impossible to fix room acoustics with an equalizer, it is likewise impossible to read room acoustics with an equalizer meter, the ⅓ octave RTA. The narrow band Modulation Transfer Function (MTF) type of test is how room acoustics must be evaluated in the low end.

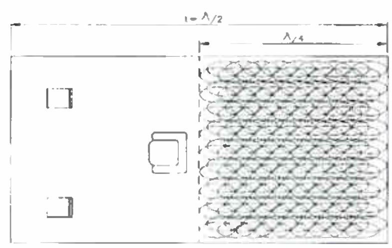


Figure 2. ¼ wavelength trap.

BASS TRAPS

Many ingenious designs have been developed to provide low-frequency absorption. In the beginning, no doubt a bass trap probably was little more than "great balls of fuzz," fiber-glass insulation or batting stacked to the ceiling in the back of the room. Such a system was so ugly that it was covered over with "scrim cloth." It did, however, provide absorption for frequencies whose wavelength is up to four times the fill depth. A 3 foot deep fuzz trap is effective to the 12 foot wavelength, about 94 Hz.

It is instructive to calculate how deep this trap would need to be to

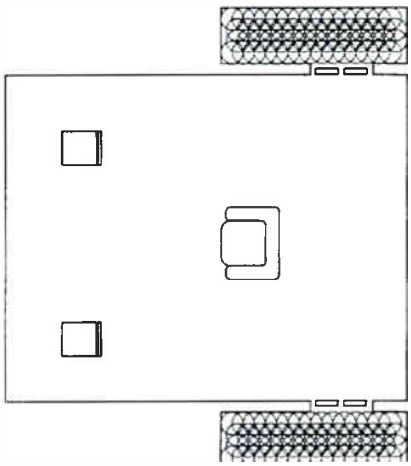


Figure 3. Bass traps located next door.

dampen the fundamental room mode now that digital tape can store such low frequencies. Calculate:

1st Mode Depth = $\frac{1}{4}\lambda = \frac{1}{4}(2L) = \frac{1}{2}L = \frac{1}{2}$ Length

A 24 foot room would need a bass

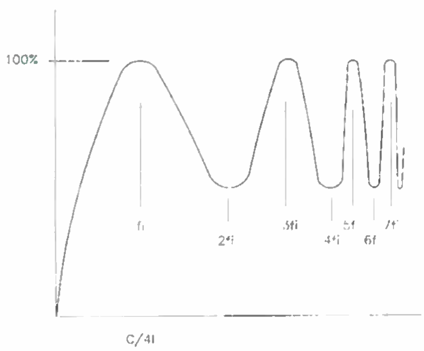


Figure 4. ¼λ trap response.

trap about 12 feet deep. Obviously, converting half the room into a bass trap is *not* an option for most people!

An alternative to filling the back of the room with fuzz is to remove the closet doors at the back of the room and fill them with fiber-glass. The frequency response curve of the ¼ wavelength trap system shows strong absorption on the first, third and fifth harmonics, because the air friction occurs at the position of "sound cancellation" or maximum air motion, typically ¼ λ and ¾ λ from the trap's wall.

SLAT BASS TRAPS

The basic mechanism for sound absorption is the friction of air as it moves across a surface. The more surface and the more air motion, the better the absorption. But

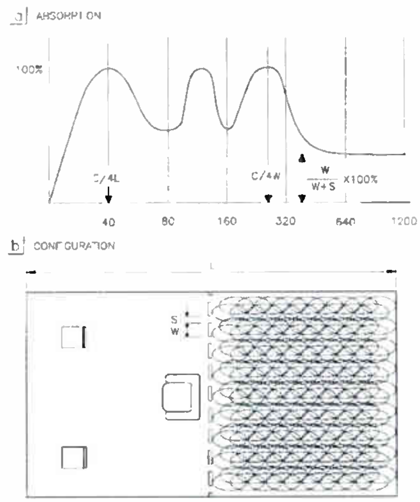


Figure 5. Traps with backscattering.

large scale bass traps are physically unacceptable in the smaller home recording studio. Another problem with giant absorption is that it makes for an uncomfortable and distracting listening environment, because it is anechoic or too dead sounding.

Consequently, wooden slats are added to most traps, somewhat like a fence. The frequency response for such a system is much more acceptable, since the mids and highs remain lively, yet the bass becomes damped. Larger wavelengths pass easily through the openings between the slats. But when the wavelength is less than four times the slat width, the sound is back scattered.

MEMBRANE TRAPS

The need for low-frequency absorption, combined with the back scattering of mids and highs, has been around for a long time. A different solution was developed early on and became a standard in studio design for forty years. "Membrane traps" utilize thin sheets of plywood, ⅛ inch typically, that are bent into a sequence of curved surfaces around the perimeter of the room. The airspace

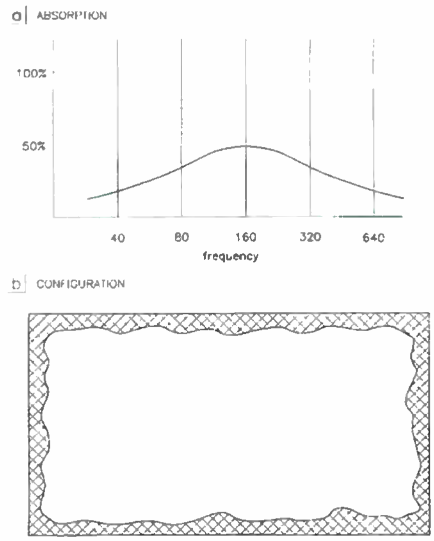


Figure 6. Membrane traps.

between the membrane and wall ranges from inches to feet and is packed with building insulation batt.

This technique provides low frequency absorption with the important benefit of continuously curved surfaces creating lots of mid and high frequency diffusion. Rooms with membrane traps are lively, diffuse and well-damped. The efficiency of this technique is only fifty percent at best. This means that twice as much surface area is needed, but we end up with twice as much sound-scattering power. All in all, it's a reasonable trade-off. These rooms are expensive, but not too different than building a giant acoustic guitar. Their concave curve sections produce local sound focus effects, a problem for mic set-ups especially in a smaller studio.

PERIMETER TRAPS

Another style of big room acoustics that has been used in control rooms is to lay up row after row of lightweight building insulation along the walls, but angled out from the walls. The hanging batt curtains occupy the outer two-foot to three-foot perimeter of the room. This technique is acoustically comfortable and stable. As the entire room surface has been converted into a great ball of fuzz, there will always be erosion of even the deepest bass energy. The depth of these fuzzy walls can vary depending on the location of the kinetic energy zones for certain problematic modes. The actual volume of room is about twice that of the apparent room. It is somewhat like a welter-weight anechoic chamber. This room can be successful in a down-town designer/contractor studio,

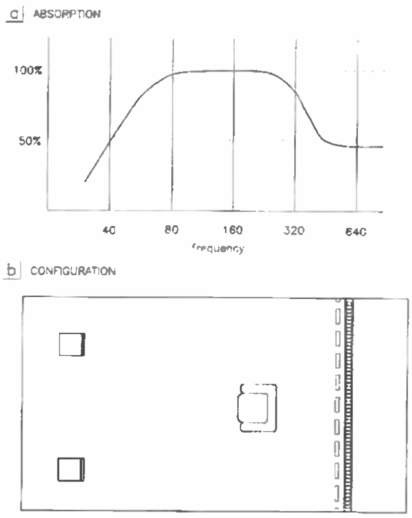


Figure 7. Pressure zone trap.

but is not an option in the limited floor space of the home or project studio.

PRESSURE ZONE TRAPS

Yet another version of deep bass absorption utilizes the sound pressure-zone concept. The fiber-glass batt used in a ¼ wavelength trap is compressed by ten to twenty times into a medium density fiber-glass board (commonly referred to as 703). This board is then "firred out" a number of inches from the wall to produce a very effective sound trap. The major difficulty with this technique is keeping the fiber-glass from vibrating as air moves in and out. When the flat sheet of fiber-glass moves, it shorts out the bass trap. Its response curve is spotty, and some frequencies are absorbed while others are not.

The trap design can also be outfitted with spaced slats to back scatter the mids and highs, and if properly made can develop high acoustic efficiency while staying close to the wall. The most common mistake in slat/pressure zone traps is that the slats are set flush against the fiber-glass. This chokes off the bass breathing ability of the trap. There needs to be at least a ½ inch air gap between slats and the face of the fiber-glass.

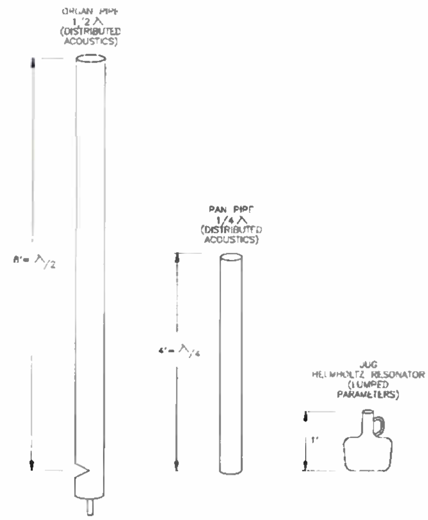


Figure 8. Lumped and distributed parameters.

The pressure zone trap is a different type of sound trap than those mentioned. It uses lumped parameter acoustics while typical fuzz type absorption uses distributed parameter acoustics. Lumped parameter devices are designed

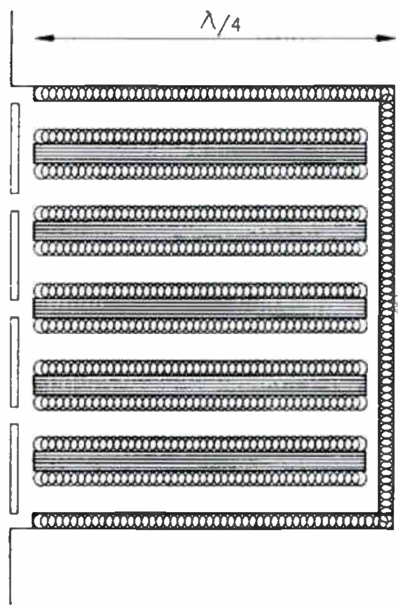


Figure 9. A classic ¼λ trap.

like an electronic circuit with discrete items such as resistors, capacitors and inductors, and can be quite small. The distributed acoustic devices use the wave-guide approach to design and are sized directly to the wavelength of the note. For example: the pan pipe (¼ wavelength) and a soda bottle (lumped parameter) can both sound out the same note and equally loud, but the pan pipe will be many times longer than the soda bottle.

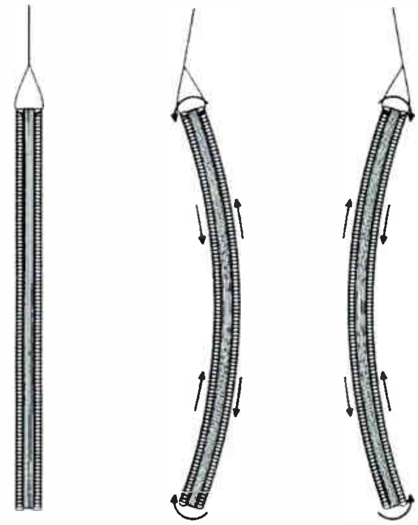


Figure 10. Sympathetic resonance traps.

IMPROVED QUARTER-WAVELENGTH TRAPS

Rather than a loosely packed fiber-glass batt, which always settles, we can glue it to sheets of sound board which can be suspended by wires inside the closet. Nothing much new here; the same response curve as for the "ball-of-fuzz" ¼ wavelength trap. The fiber-glass does not settle out and so the trap keeps working for years.

SYMPATHETIC RESONANCE TRAPS

The sympathetic resonance or panel trap is a creative cousin to the sound board and fiber-glass trap. Often suspended in, supposedly, random overhead positions, these panels are each tuned by trimming to size and adding weights. Particular frequencies set these panels into sympathetic vibration motion, and the incident acoustic energy is converted to vibrating panel energy.

Dissipation of the energy occurs with the air moving back and forth across the face of the panel as it "twangs." Its own internal friction also dampens its motion. These panels have to be ¼ n_\ in size, otherwise they would not be able to interact with the sound wave. An 8-by-8-foot panel would function at 40 Hz, if it was correctly tuned. Panel traps work best if aligned to meet the sound wave face on (like a ribbon mic) to engage action. The flat of the panel needs to face the wave front. Too often it is physically impossible to set up a real room with these panels because of size constraints.

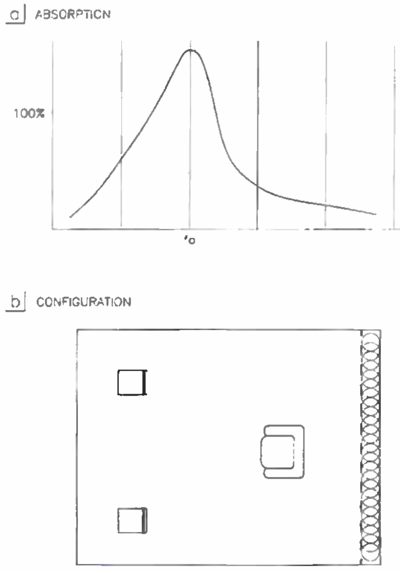


Figure 11. The Helmholtz trap.

HELMHOLTZ TRAP

A classic never-to-be-forgotten sound trap is the Helmholtz trap, which carries the name of a great, old-time German acoustical scientist. Conceptually, the Helmholtz is little more than a jug, tuned with loose batt stuffed inside. However, it usually looks like a panel of ¼ in. pegboard behind which is a 1-3 in. air space fluffed with light building insulation.

The absorption curve illustrates the strong frequency selective property of this type of absorber. Two difficulties exist with using such a trap:

1. It is a single-frequency type, and must be tuned to a known room mode, and

2. The trap's performance is strongly dependent on the amount of batting placed in the cavity and the rigidity of its wall, especially the perf panel. It is difficult to tune.

FUNCTIONAL TRAPS

In the early 1950s, Dr. Harry Olsen, director of RCA Labs and a prolific masterful contributor to audio practice and theory, presented his "functional sound absorber." It was especially unique because of its unprecedented one hundred and sixty percent efficient handling of low frequency sound. He envisioned its use overhead in large rooms and halls. But elsewhere in his literature he advises that low-frequency sound absorbers are best located in the corners of smaller rooms.

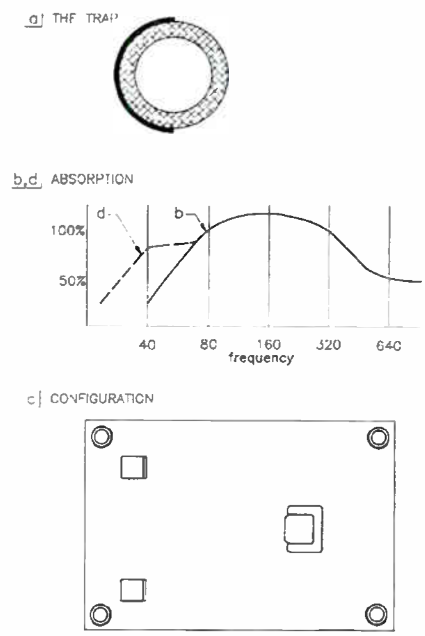


Figure 12. The "functional" trap.

The "functional sound absorber" is a close cousin to the flat pressure zone trap. The density of the fiberglass for this type system is impedance matched to the radiation impedance of free sound waves in air. Essentially, if the fiber-glass is too dense, sound bounces off; if it is too loose, sound goes right through. The resistance of the surface combines with the volume of the airspace inside to provide a very low frequency response curve for the trap, similar to an electronic RC circuit. By adjusting the value of R and C, the desired RC time constant can be picked for the trap's roll-off characteristic.

Sound absorption is always a function of two factors: the surface of acoustic material exposed to the sound field and the efficiency frequency response of the surface. Dr. Olsen's cylinder bass trap has just over three times the apparent frontal surface area. Secondly, it is very efficient into the lower frequencies because it is an acoustic circuit of RC time constant design, rather than the more traditional ¼ wavelength "fuzz ball" approach to acoustics.

As with all traps, midrange and high frequency partial reflectivity remains of value. Accordingly, today's pro style functional-type bass trap is usually outfitted with a membrane section to back scatter mid-range frequencies (usually above 400 Hz). These traps are extremely efficient, and particularly when located in the corners of a room. To increase absorption in a selected frequency band or to extend the low frequency response curve, the interior volume can be fitted with a low Q Helmholtz resonator. It is particularly suited as a corner-loaded bass trap in small audio rooms because it is small, efficient, modular and easy to set up, more like studio equipment than a remodel construction project.

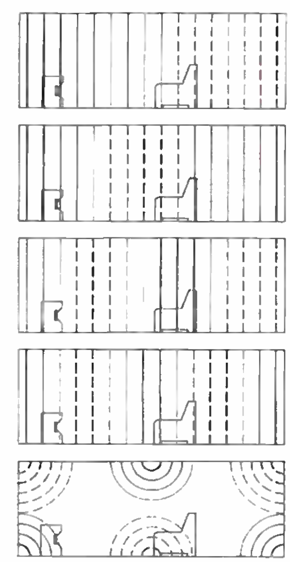


Figure 13. Room resonance modes.

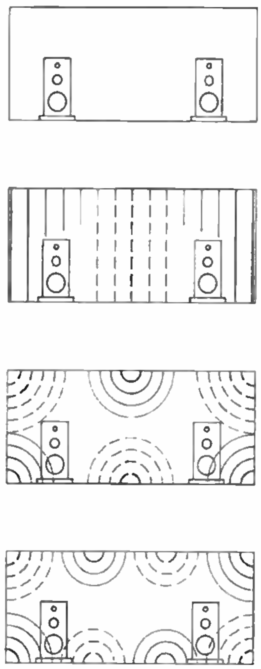


Figure 14. Head end ringing.

RECTANGULAR ROOM DISEASE—HEAD END RINGING

Home/project studios in rectangular rooms suffer from a malady that most designer studios do not have—head-end ringing. Speakers are usually located near the front of the room. From this location they easily stimulate room resonances along the length of the room. It takes about ten exchanges of sound between the front and back of the room to build up the condition of resonance, typically ¼ second.

Speakers may be far from the back wall, but they are very close to the side walls and floor/ceiling walls in the front of the room. Because of these short lateral dimensions, side to side and vertical resonances can build very quickly (within 1⁄20 second in the front end of the room), long before the entire room can be engulfed in the resonance. This fleeting, quick resonance is called "head-end ringing" and because of the time scale, dramatically affects imaging and the color of attack transients.

Head end ringing is not a deep bass problem—it is a mid bass coloration effect due to a lack of bass traps in the front end of the room. Designer studios with the Reflection Free Zone (RFZ) cup shaped front end don't have this problem. The raked walls and ceiling eliminate any opportunity for reflections to stay and build up in the front of the room. But with home and project studios set up in rectangular rooms, head end ringing is a major problem that near-field or mid-field monitors cannot even avoid. Typically, playback monitors are located about half way between floor and ceiling, and about one-third in from the side walls. The classic head end ringing problem occurs at about 140 Hz. A substantial distribution of mid-bass traps on the walls and in corners of the front end of the audio room is the only way to control head end ringing.

EPILOGUE

Over the years bass trapping has matured unique to the recording industry. We don't usually see them in press release photos because they have always been built in behind the walls of the designer/contractor studio. Nevertheless, bass traps are a tradition that is integral to the definition of a recording studio or control room. They are the primary acoustic consideration that separates recording rooms from regular rooms. Although many versions have evolved, one thing is for sure: bass traps have been, are now, and will most probably continue to be the cornerstones for the pro room acoustic.

Since bass traps won't be built in behind the walls of any home/project studio, they will have to be set up in front of the walls and corners of the room.

But these are modern times and the availability of personally affordable studio grade equipment is changing the face of the recording industry. Home and project studios are being set up at a ratio of ten to one compared to the traditional designer/contractor-built studio. This new and rapidly developing division of the recording industry may be wired like downtown studios, but their room acoustic is all too often set up with no more than a couple of pieces of foam tiles and particularly depleted of bass traps. Consistency is always important, and the first rule in studio design is that it must "look like a studio." In this sense the topic of bass traps in the designer/contractor-built studio and the home/project studios do have one thing in common—no bass traps are visible.

There is only one reason that studios have to look like studios— to help establish client confidence. But this requirement for designer/contractor studios does not apply in the home/project studio.

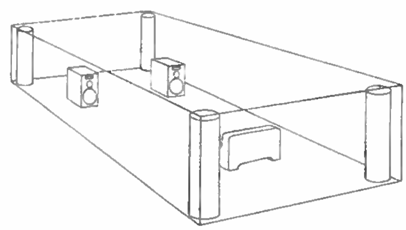


Figure 15. Bass traps for a listening room.

To a large degree, the owner of the home/project studio *is* the client of the studio. The home/project studio may not have to look like a designer/contractor studio in order to do its business, but it certainly has to act like one. Since bass traps won't be built in behind the walls of any home/project studio, they will have to be set up in front of the walls and corners of the room. For the first time, engineers will simply have to look at bass traps.

Essentially, bass traps are "coming out of the closet" in order to get back to work in the home/project studio. After all, any chain, even the home/project studio audio chain, is no stronger than its weakest link, and bass traps are critical to the last link of the audio chain—the room acoustic. db

JAYANT DATTA

The 10th AES International Conference

THE 10TH AUDIO ENGINEERING Society (AES) International Conference, Images of Audio, was designed to allow a forum of discussion of some of the most important topics facing audio engineers and broadcasters today. Quite appropriately, London, one of the world's major recording and broadcasting centers, was chosen to be the first venue in Europe to host this conference from Sept. 7-9, 1991.

The first day was planned as a tutorial to familiarize the attendees with the principles and technology of digital audio. It was designed both as an isolated event for students as well as a 'refresher' course for those who wished to attend the four sessions over the following two days that papers were to be presented in.

SESSION A: RECORDING AND POST PRODUCTION

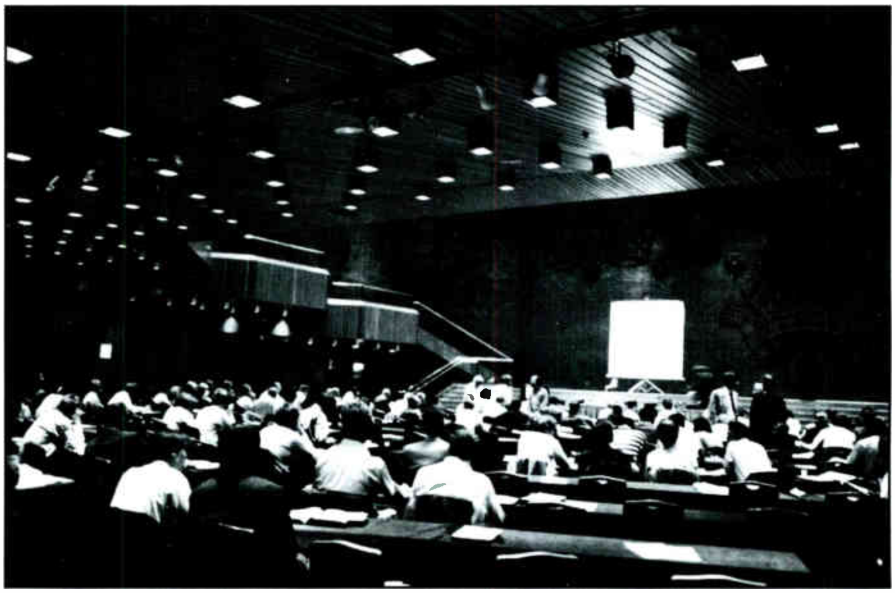
Although hard-disk recording and editing have been involved in production, many foresee the digital video recorder as a medium that will be used in a growing range of programs in the future. John Watkinson, author of such books as *R-DAT*, *The Art of Digital Audio*, and *The Art of Digital Video*, pointed out that most digital VTRs only offer the 48 kHz rate, making it incompatible with the more common 44.1 kHz rate. This oversight makes it difficult to master CD-video discs and obtain sound track masters in joint television and music productions. The sampling rate of 48 kHz was originally used to allow varispeed operation over a large speed range without aliasing, but oversampling filters can be designed to automatically track the sampling rate and avoid aliasing problems. Editing of digital audio is complicated by the fact that data is recorded using product codes and interleaving. Clearly, the data must be de-interleaved and replayed into memory to affect the edit, then re-interleaved and rerecorded. The whole process is generally known as read-modify-write (RMW).

The acceptance of DAT as a professional format requires modifications to be made to the consumer models it was based on. Operational features such as confidence monitoring, memory start, time-code recording, video sync, RMW and advanced editing capabilities (assemble, insert and cluster edits) are necessary. The sequence of the four heads that allows confidence monitoring could be reversed to allow RMW. The problem of startup time delay associated with rotary head formats can be overcome by pre-loading a memory buffer with sound around the start point. Then, on cue, replay can begin instantly from the memorized audio, while the tape transport goes into forward mode and locks to the memory output.

The SoundStation Sigma is a new generation of a disk-recorder from Digital Audio Research, where the concept of segment-based parameters is extended to include EQ and pan. The segment parameters are bound within the segment itself, and will therefore remain unchanged during moving or copying. Although the Sigma offers only sixteen channels of simple replay and record, it is capable of routing one hundred and twenty-eight channels of audio, and extremely long crossfades on all sixteen channels. During this, it is effectively handling thirty-two channels, all with possibly different EQs.

Solid State Logic described SoundNet, a workstation networking system complementing the ScreenSound random access editing and mixing system. SoundNet supports a central resource of ScreenSound processors, hard disks, optical disk libraries and backup tape streamers. Forty-eight hours of audio may be stored centrally with up to fifty-six tracks

Figure 1. The Great Hall of Kensington Town Hall, where the papers were presented.



Jayant Datta has a B.Tech degree in EE from the Indian Institute of Technology (Bombay,India) and is working on two MS degrees at the University of Miami, Florida—Audio Engineering and Computer Engineering.

of simultaneous playback. Each network user can select any unassigned disk as the Working Disk for full random access editing. The modular, open expandable architecture, along with industry standard interfaces, helps ensure forward compatibility and a natural expansion path as new devices and subsystems become commercially available.

SESSION B: BIT-RATE REDUCTION

Welcome to the '13 dB miracle.' Dr. Brandenburg, one of the principal contributors to ASPEC coding, played two pieces of music with added noise—one with white noise, and the second with noise shaped such that it lay below the masking threshold of the signal. Although both pieces of music had identical signal-to-noise ratios of 13.6 dB, the first piece was punctuated by annoyingly audible clicks, while in the latter there was absolutely no hint of noise. This demonstrates the uselessness of traditional measurement techniques such as SNR in the realm of low bit-rate coding. The true test is subjective listening.

In order to achieve the high level of data compression required in certain applications (i.e. DAB, ISDN), the general procedure is to split the signal into a number of narrow frequency bands either by filtering, or by a mathematical transform. The masking model of the ear is then used to requantize the signal in the various bands to varying accuracy depending on the level of the signal in the band, and knowledge of which signals mask others.

Eureka has adopted Coded Orthogonal Frequency-Division Multiplexing (COFDM) which was developed by the French CCETT to transmit compressed signals either from satellite or terrestrial stations. This involves the modulation of a large number of low bit-rate input channels onto a very large number of closely-spaced carriers. Interleaving and convolutional error correction is introduced in both the frequency and time domains resulting in its ability to ignore multipath reflections, which has been a major problem with FM transmission.

Figure 2. Session A. From left, John Ives (Sony Broadcast); Mike Parker (DAR); John Watkinson (Watkinson Int.); Mark Yonge (SSL); and Richard Shiller (Sony Broadcast).

Several companies are upgrading their first generation coders by lower bit-rate versions. Audio Processing Technology, one of the first companies to come forth with a commercial bit-rate reduction system when they introduced the apt-X100, is working on the apt-X64 featuring a 12:1 compression ratio with a data rate of 64 kbps. Dolby Labs has developed two transform-based coders of the AC-2 family providing 4:1 and 6:1 bit-rate compression at low and moderate time delays respectively. The low delay version is optimized for music material contribution applications, whereas the moderate delay version is appropriate for DAB and HDTV applications. Presently, AC-3 is being developed to handle five channels + subwoofer at 320 kbps using directional cues.

The Swedish Broadcasting Corporation (SR) was assigned the task of evaluating the performance of low bit-rate codecs by meaning of subjective tests on two separate occasions—1990 and 1991. The 1990 test was carried out in real-time hardware on four codecs—MUSICAM (32 band, sub-band), ASPEC (transform), ATAC (transform) and SB/ADPCM (8 band, sub-band). ASPEC got a slightly higher score on the subjective part, while MUSICAM was superior in the objective part (delay, complexity, etc.). SR concluded that none of the codecs could be accepted at that stage of development for two reasons:

(1) there is potential for further refinement of the algorithms, and

(2) artifacts that may be difficult to detect at a first listening may become more obvious as time goes by. The decision, therefore, was that MUSICAM and ASPEC should merge their two algorithms into a common one. This work was finished and tested by SR in 1991. The new codecs showed an improvement in subjective sound quality compared to both MUSICAM and ASPEC.

The MUSICAM coder achieves a six to twelve fold reduction over conventional PCM by using a sub-band compression technique assisted by a perceptual model, whereas the decoder requires very low computational power. ASPEC combines transform coding and entropy coding to achieve good coding efficiency together with high flexibility for audio coding at low bit-rates. The quantization and coding are controlled by two nested iteration loops. The outer loop controls the quantization noise, adjusts scale factors, and does the psychoacoustic weighting. The inner loop ensures that the amount of data needed for coding does not exceed the number of bits available.

SESSION C: DIGITAL AUDIO SIGNAL PROCESSING

There is no doubt that the performance targets of digital audio systems are becoming easier and more economic to achieve as technology advances. Floating point DSP chips offering high precision, dynamic range, speed and flexibility are an obvious choice, but it is possible that an Application Specific IC (ASIC) solution may offer advantages in certain well-defined applications. There are advantages and disadvantages associated with both DSP and ASICs, making one or the other more appropriate for a particular application.

Commercially available DSP chips are not capable of handling the frequency resolution needed. For example, the DSP 56000 can barely handle a 256 tap filter which has a frequency resolution of 172 Hz; therefore, the solution might be to use ASICs. Manufacturers should consider ASICs, whereas the users should not be impressed by the technology of the DSP—they should demand performance.

A highly modified FIR structure with emphasis on coefficient design/modification and parallel/distributed processing has been developed by Audio Animation. It has been called the FIReEngine and can do up to eight independent filters. It is a virtually self-contained IC that runs at worst case speeds of 50 MHz and best case speeds of over 75 MHz. Software runs in the background to determine the number of taps for each filter.

In the early days of digital audio, those who ran into the need to convert between sampling rates could go two ways. Either they could perform a D/A conversion of the digital audio signal and then re-sample it at the desired rate, or they could use a fixed-rate, bulky and costly digital sampling rate converter because of the state of the technology at that time.

Today's sampling rate converter should be small, inexpensive, be able to handle most sampling rates from 30 kHz, and be able to accommodate time varying input rates, as when using the pitch control on a CD player while putting out a stable output sampling rate. Tomas Ahrne of Swedish National Radio reached these goals by choosing the technique of oversampling the input signal followed by decimation to arrive at the new sampling rate. A multi-step conversion of 2x, 64x and 1024x was employed to subject the input signal to 2^{17}x oversampling. The resultant distortion after decimation for this system was estimated to be better than –98.8 dB.

Figure 3. Dr. Karlheinz Brandenburg presenting his paper on ASPEC Coding.

SESSION D: AUDIO FOR HIGH DEFINITION TELEVISION

In an exciting HDTV listening/viewing session, the British Broadcasting Corporation featured five excerpts of program material (classical music, opera, parade, sports and drama) during a listening/viewing session. Speaker configurations were switched in the midst of the programs to enable the audience to compare between either five and three, or five and two channels of sound. Changing from five to either three or two channels definitely created a collapse in the acoustic space that had just been created all around. The best surround effects were obtained in the sporting event (Wimbledon finals), especially during the applause. During the drama (which was set in a restaurant), the ambient sounds from the surround speakers (the continuous clanking of cutlery) had a tendency to drown out the dialogue that was fighting to be audible through the front speakers—this was very distracting. This problem arises as a result of the increased number of variables that have to be dealt with, and is representative of the various issues facing audio for HDTV.

The sound system for HDTV aims to meet a number of criteria laid down in a CCIR report: better sensation of reality, wider listening and viewing area, better correspondence between the direction of the image and the sound, downward compatible, compatible with movies, optimum economy in all respects, including transmission efficiency and real-time mixing should be easily operated.

While there is agreement that more than two replay channels are

Figure 4. Session B. From left, Francis Rumsey (Univ. of Surrey); Yves-Francois Dehery (CCETT)); Neil Gilcrist (BBC Research); Stephen Smyth (Audio Precision Tech.); Louis Fiedler (Dolby Labs); Karlheinz Brandenburg (Univ. of Erlangen); and Christer Grewin (Swedish Nat. Radio).

needed, the exact number is still open to debate. Obviously, the larger the number of channels, the better the overall effect, but at some point, practical considerations must step in. After carrying out exhaustive experiments (on-axis and off-axis location, aural and visual image differences, microphone arrangements, etc.), NHK concluded that reproduction of a minimum of three discrete signals from the front loudspeakers (Center, Left and Right) is desirable to maintain better correspondence between the picture direction and the sound localization for off-axis observers. Reproduction of an additional signal by at least two loudspeakers placed symmetrically to the rear is ideal for the reproduction of sounds from behind, and the production of a more effective acoustic environment.

The introduction of multi-channel sound will require changes in the mixing room. New sound panning facilities will be needed on the sound desk, and space has to be made for the surround loudspeakers in the often cluttered rear of the room. With so many channels involved, automated mixing becomes essential, and multi-format monitor switching is needed for compatible reproduction of the surround sound production in stereo, three channel, and any other format that may be considered to be representative of a reasonable proportion of the audience. Downward conversion to fewer channels of sound is accomplished by using matrices to combine the original multi-channel signal into fewer channels. The amount of sound energy in the surround channels (i.e., crowd noise in sporting events) has to be tempered, on occasions, because it would be overpowering in the stereo mix.

The provision for broadcasting an extra language is also being considered. For sports, one extra channel per language suffices, but in most other programs, the production is attempting to place the spoken work into the acoustic environment. This might require five channels per language, making it expensive for many programs.

SUMMARY

The industry is approaching a stage at which audio signals will remain in the digital domain throughout the majority of the production, post production and distribution phases of their lives, requiring that signal modifications be performed digitally. The potential now exists for achieving effects that were either very difficult or impossible in the analog domain.

There are some issues relating to low bit-rate coding that need to be considered. For example, during post-production work in the studio, the signals may go through a number of stages of processing which may modify the masking effects. There seems to be consensus on the fact that different codecs have different impairments, and tandeming them add the respective impairments. These are areas that need further study. Future work will concentrate on advanced joint stereo modes and improved psycho-acoustical models. Standardization of low bit-rate audio coding will soon open the door for widespread applications.

It is becoming apparent that techniques in DSP that are too much processing power to be done in real-time today will be tomorrow's common practices. In the future, one can expect to see hybrid filter structures that will use FIR techniques for the upper frequencies, IIR techniques for the very low frequencies, and a more widespread use of ASICs for certain computation-intensive, but specific tasks.

Developments in HDTV sound are proceeding with international committees studying some of the more theoretical aspects of the subject. Although existing facilities can be adapted to the needs of multi-channel sound production, purpose-designed facilities would be better. Multiple language working might, in principle, be a worthy cause, but it could be expensive for many programs.

This conference was very timely, attempting as it did to discuss emerging topics of great relevance to the future applications of digital audio. Full credit is due to the Conference Chairman (Jeff Baker) and other members of the committee (Heather Lane in particular) for their efforts to make this a success. The papers presented were relevant, the discussions (often punctuated by subtle British 'humor') were stimulating, lively, productive and highly informative, and last but not least, the food was also good. What more can anybody ask for? One thing—more such conferences—definitely more! db

MUSICIANS' NOTEBOOK

Producer's Notebook Part II

● In the last issue, I discussed an overview of the producer and his or her role in a recording project. In this issue, I'd like to recount some specific experiences that I've had producing others.

Last time, I spoke of a project I had begun with a folksinger. After she played me seven or so songs, I picked the three that I thought were strongest. I listened to a rough tape of all seven while riding in my car the next week and was still certain about my choices. The plan was that one song would be recorded simply with acoustic guitar and vocal, another with additional guitars and vocals and the last would be a full Pop production with rhythm section and keyboards and lush harmonies.

Because life is fraught with change, we only completed the first two songs. For those I used an AKG 414 for her voice and an Audio Technica AT813 for her guitar. Both are condenser mics which I prefer to use in the studio for voice and acoustic guitar. They lend a naturalness, a purity of tone and an accuracy. I recorded the voice with a slight compression but without EQ; I added that later in the mix. I noticed that the large diaphragm in the AKG 414 picked up a lot of guitar. I wasn't able to get the separation I would have liked.

I had to come to terms with this problem with another artist I'm producing named Mike DeLuca. Mike is a talented singer-songwriter I've known for years. He is an ideal candidate to produce: he's a great guy, he's talented and he pays his bills! We talked about working together for a long time. I used one microphone—the AT813 and placed it equidistant from his voice and guitar. He used my Martin D-35 acoustic which has a deep, rich sound. The mic picked up both voice and guitar in a balanced mix. Again, I recorded with a gentle compression.

I never record with EQ as I don't want to be committed. It's much quieter without it and I can add it judiciously in the mix, if necessary. We ran into a snag when there was a glitch in the guitar part. It would be hard to overdub in a natural sounding way as his voice was on the same track. It would be a noticeable overdub to just correct the guitar part. I suggested just recording the whole guitar part first and then the entire vocal on another track. In which case, we'd have total separation and he could concentrate on each part for a *perfect* performance.

Guess what? It didn't work. Something was lost when both parts were done separately. The feel was different. Both parts were dependant on the other for a more natural, organic rendering. You play differently while you're singing. The parts were meant to be performed together; a true illustration of the whole being greater than the sum of its parts. I had found this to be true when I recorded my own vocal and acoustic guitar pieces.

As a producer I had to make a decision. I opted for the "feel" of the performance rather than having optimum sonic separation. I was willing to compromise technically so that I would get a warmer, more human performance. We just recorded until we got a take we liked. The keeper usually comes within the first three takes. If you do too many, the feeling is lost, even if there aren't any mistakes. There's often an urgency, a plaintiveness in the first few takes that's hard to capture in take 13. A producer has to know when to keep pushing and when to take a break and come back to it later. Patience is one of the most important attributes a producer must possess.

Mike and I had come back a week later and had gotten a track we liked (within three takes of course!). It worked out better another way as well, because he made some lyric changes that made the song even stronger. We then set about the task of recording additional guitar parts.

Mike had some great ideas but wanted me to play because I'm more of a lead guitarist...I deferred, saying that it would sound more natural if he played it. They were his ideas and he knew how to play the part already. I told him not to underestimate his own ability. A producer must encourage and bolster the artist's confidence while not settling for less than he knows the artist can deliver. Again, I prefer feeling over technique; whatever it takes to get the song across.

The Beatles were great examples of this and of course, George Martin, their producer was largely responsible for their success in the studio.

Mike laid down the single string acoustic track and it served to weave in and out of the melody, creating a nice movement, a tapestry. Next, he played my old Epi-phone Riviera 12 string. It's an electric semi-hollowbody guitar much like a Gibson ES-335. We plugged it right into the mixing console and it added a shimmering Byrds-like sound which accentuated the first guitar part. We'll probably add a chorus effect to it in the mix.

Mike and I discussed the arrangement of the song, but we just have to hear whether all of our ideas would work or not. (It's easier to take away than it is to add, though MIDI people often overdo because "it's there!") Mike "heard" strings on the piece so we spent some time auditioning string samples from my large library and settled on a string quartet sample. To this very realistic sound, we added a synth string pad to smooth it out and thicken it. It was crucial to find the right mix so that the sampler and the synths would work as one sound. I backed way off on the synths and just a hint of them added greatly to the sampled strings.

Mike wants to bring in a mutual friend to do harmony vocals and another one to play Celtic harp. We'll wait until we have basic tracks down for another song so they can come in and do both songs at once. A producer should work efficiently and economically.

One time, I was working on the production of a dance tune. I had written the song a few years previously and had finally gotten around to recording it. (Hang onto your songs, fragments and sketches—you never know when you'll be able to use them.)

I had painstakingly programmed the drum machine and sequenced the bass part from my guitar controller, combining a sampled bass with an analog synth for a monstrous bass sound. I programmed horn blasts and breathy sounds stabbing from a sampler and synth for texture. I recorded a sync track so all the MIDI stuff was going to be recorded hot and first-generation directly to two-track during the mix. I had recorded the guitars to the multi-track. They and the vocals would be the only tracks previously recorded for the mix. All else would be "virtually recorded" with MIDI. A friend of mine whom I've worked with over the years came by to do background vocals. She listened to the song and liked it a lot. She didn't think the lead vocal was working however.

Now, in all Pop music, especially dance music, the vocal is the most prominent track (along with the snare drum!). Most people key in on the vocal. Massive production and a great guitar solo will not save the day if the lead vocal is weak. In this instance, the lead vocal had to be changed, and my friend offered to sing it. I thought about it and promptly fired the lead vocalist-me! Producers must be flexible and open to other people's ideas. They might just have a better idea than we do! A producer mustn't let him or herself get too attached to anything. The artists will do enough of that. It's their work. Their very subjectivity is what we're seeking to capture on tape. We producers must remain objective and open. When it works, we'll know it.

We can work in our local areas or aspire to become a big-time producer in one of the three major music centers: New York, Los Angeles or Nashville. Minneapolis, Atlanta, Austin, Memphis, Miami, Chicago...

Finally, I produced a saxophone player awhile back. He brought in his core rhythm section which consisted of a keyboard player, bassist and drummer. They were well rehearsed. I can't emphasize preparation enough. It saves our clients money. Studio owners might love it when artists rehearse on the clock, but often it ends up eating into actual recording time; both time-wise and budget-wise. Producers must pace the entire project and bring it in on budget. I have a direct studio. The drummer brought in his own Octapad (a MIDI drum controller) and sound sources. The bassist plugged directly into the board. The keyboardist brought his controller but used some of my gear. We recorded them live in the studio. That is, they performed together, recorded to tape and didn't sequence.

It's amazing what live musicians come up with together. The feel was real. I've heard subsequent projects by this sax player which had been computer-sequenced and they don't match the excitement of the live ensemble.

After we laid down basic tracks and fixed mistakes, we started overdubbing more keyboard, percussion and guitar parts. Finally, we recorded the sax parts and worked with several mics and placements to get a sound. We ended up using an E-V PL20 or a Sennheiser 441.

Every project is different but the more artists you work with, the more experience you garner. Many of us start out by producing ourselves in our own multi-track settings. We gain engineering skills and begin to learn how to get a good sound. If you are a songwriter, you can acquire the skills of arranging and "hearing"a full production as well as make constructive suggestions regarding music and lyrics. The main thrust of the producer is to help the artist capture his or her work on tape. We can go out and recruit talent in our local area or work with friends and acquaintances. We can work in our local areas or aspire to become a big-time producer in one of the three major music centers: New York, Los Angeles or Nashville. Minneapolis, Atlanta, Austin, Memphis, Miami, Chicago—most major cities have very active recording scenes. Whatever your goal, producers serve as a bridge in helping manifest music to a listening medium. We wear many hats not the least of which is coach and psychologist. It's fascinating and hard work but it should also be fun.

Good luck and keep tweakin'! db

BRUCE BARTLETT with JENNY BARTLETT

Phantom Power Primer

● If your studio uses condenser microphones, they need power to operate their internal circuitry. Some use a battery, but batteries wear out. If this happens on the air, you'll hear a weak or distorted mic signal.

It's better to remote-power each condenser mic with a phantom-power supply. This article explains phantom powering, tells how to use it, and explains how to add it to your mixer.

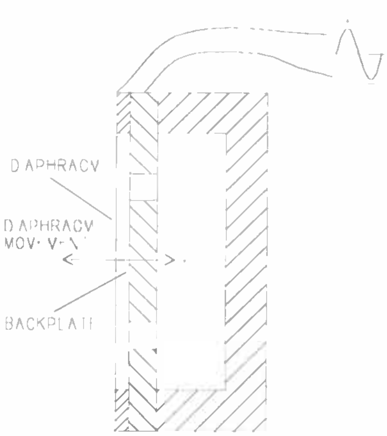
Phantom power is supplied to the mic through its 2-conductor shielded cable. The power can be supplied either from an outboard box or from your mixing console (at each mic connector). The microphone receives power from, and sends audio to, the mixer along the same cable conductors.

According to DIN standard 45596, phantom powering is a positive voltage (12-48 V) on XLR pins 2 and 3 with respect to pin 1. The cable shield is the supply return. There is no voltage between pins 2 and 3. Pin 1 is ground; pin 2 is audio in-phase, and pin 3 is audio return.

WHY CONDENSER MICS NEED POWERING

Let's explain why condenser mics need a voltage in order to operate. In the condenser microphone (*Figure 1*), a conductive diaphragm and an adjacent metallic disk (backplate) are charged with static electricity to form two plates of a capacitor. When sound waves strike the diaphragm, they vary the spacing between the plates. This varies the capacitance and generates an electrical signal similar to the incoming sound wave.

Figure 1. A condenser microphone transducer.



The diaphragm and backplate can be charged in two ways:

1. By an externally applied voltage (from phantom power). This arrangement is also called *external bias*.

2. By a permanently charged *electret* material in the diaphragm or on the backplate. This is also called *internal bias*.

The output of the condenser mic capsule is extremely high impedance so it is very hum-sensitive. To bring that impedance down to a usable value, an impedance-converter circuit is typically connected to the capsule output. This circuit is necessary whether the capsule is electret or non-electret. The converter needs a DC voltage to power it, and this voltage is supplied by phantom power.

Sometimes other transistors are added to give the mic a balanced output, and these components work off phantom power too.

In contrast, a *dynamic* microphone needs no power because it has no active electronics. It generates its own electricity like a loudspeaker in reverse. In a *moving-coil* dynamic microphone (*Figure 2*), a coil of wire is attached to a diaphragm. This voice coil is suspended in a magnetic field. When sound waves vibrate the diaphragm and its attached coil, the coil vibrates in the magnetic field and generates an electrical signal similar to the incoming sound wave.

In a *ribbon* type of dynamic microphone, the diaphragm is a thin metal foil or ribbon. Sound waves vibrate the ribbon in a magnetic field and generate corresponding electrical signals.

You can plug a dynamic microphone into a phantom supply without damaging the mic. That's because the voice-coil leads are not connected to pin 1, so no current from the phantom supply can flow through them.

However, if there's any imbalance in the phantom voltage applied to pins 2 and 3, a current will flow through the microphone voice coil (which is connected to pins 2 and 3). For this reason, it's best to switch off phantom power for dynamic mics.

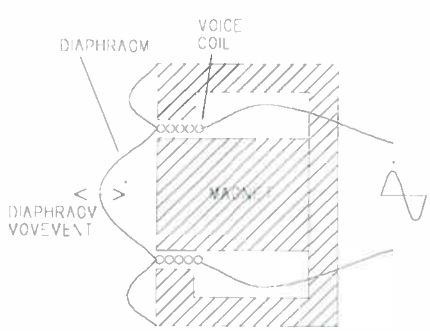
CIRCUIT DETAILS

In a typical phantom-power supply (*Figure 3*), a DC voltage (say, 48 volts) is applied through two equal resistors to pins 2 and 3. Inside the microphone, two equal resistors in series are across pins 2 and 3. Power for the microphone circuitry is taken off the center tap of the two equal resistors. (These are built into the mic; you don't need to add them).

The power-supply resistors must be high enough in value so they don't load down the microphone, and high enough to isolate several microphones from each other in case one mic cable shorts the supply.

These resistors also must be low enough in value so that, when the mic drains current through them, they don't drop the phantom-sup-

Figure 2. A dynamic moving-coil microphone transducer.



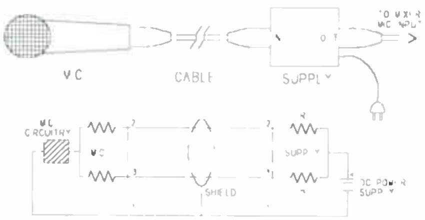


Figure 3. Two-resistor phantom powering.

ply voltage excessively. If the resistors are too high, the phantom voltage will sag when a mic is plugged in. The higher the current drain of the mic, the more the supply will sag.

USING A STAND-ALONE SUPPLY

You can buy a phantom power supply (*Figure 4*) from your microphone dealer. Some supplies are AC powered; some are battery powered. Some power a single microphone; others power several at once.

In any case, you plug the supply in series with the mic line. The supply has XLR-type input and output connectors, one pair per channel. Connect a mic cable between your microphone and the supply's input connector. Plug another mic cable between the supply's output connector and your mixer mic input.

ADDING PHANTOM TO YOUR MIXER

If your mixer doesn't have phantom power and you want to add it, here's how (refer to *Figure 3*):

First, find a well-filtered DC voltage in your mixer. Around 48 V is ideal; 24 V works with most microphones, but 15 V may be inadequate. If that's all you have, try to find a power-transformer winding at about 48 V, full-wave rectify it, and filter it as shown in *Figure 5*.

Apply the DC voltage through two equal resistors to pins 2 and 3. These must be matched within 1 percent so that the line stays balanced. The table below, based on DIN spec 45596, shows what resistors to use:

12 V supply: 680 ohms
24 V supply: 1.2 kilohms
48 V supply: 6.8 kilohms

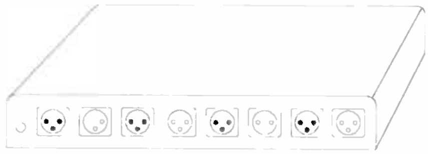


Figure 4. A phantom power supply.

Warning: the 680 ohm resistors may load down a microphone excessively.

Solder a matched pair of these resistors to each mic input connector, pins 2 and 3. Supply all these resistors from a common voltage source.

Here's another method. Apply phantom power through one resistor to the ungrounded center tap of the mixer mic input transformer (*Figure 6*). Use these resistor values:

12 V supply: 340 ohms
24 V supply: 600 ohms
48 V supply: 3.4 kilohms

This method is not recommended. Any imbalance in the windings on either side of the center tap will cause phantom-supply current to flow through the transformer, possibly magnetizing and saturating the transformer, which causes distortion. For best results, use a transformer with bifilar windings, or use the two-resistor method.

CAUTIONS FOR USE

Don't plug a mic into an input with phantom already switched on, or you'll hear a loud pop. If you have no choice (as during a live concert), try to have the mic's fader down when you plug it in.

Have a spare 48 V supply in case the main supply goes down.

Avoid having phantom in a patch bay because someone is likely to patch in and cause a pop. If you must patch into a jack with phantom on it, mute the input module that the mic is connected to, or turn down its fader. Mic-level patches should be avoided anyway.

Some phantom supplies cause a hum when you plug in a connector that ties the shell to ground. Float the shell.

Since the cable shield carries the DC return, be sure the shield and its soldered connections are secure. Otherwise, you can expect crackling noises—especially when the cable is moved.

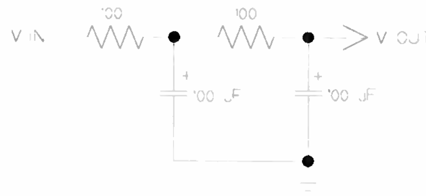


Figure 5. A supply-voltage ripple filter.

Power supplies are rated in the total number of milliamps they can supply. Make sure that the total current drain of all the mics plugged into the supply doesn't exceed the supply's current rating.

Some microphones work on either internal batteries or external phantom power. In most designs, connecting the mic to phan-

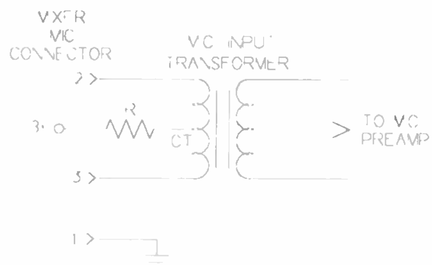


Figure 6. Phantom powering with a center-tapped transformer.

tom automatically removes the battery from the circuit. Otherwise, the battery would severely load down the phantom supply. If this appears to be happening, remove the battery.

If a condenser microphone doesn't work due to low phantom-supply voltage after the mic is plugged in, try these suggestions:

- Supply phantom from a better-regulated console.
- Use a mic with less current drain.
- Redesign the phantom supply as described in this article.
- Add a voltage regulator to the supply voltage.

By using an external phantom-power supply, or by adding phantom power to your mixer, you can depend on reliable powering for your condenser microphones.

Portions of this article were excerpted from the author's article on phantom powering in ***Radio World Newspaper****.* db

Buyer's Guide—Speaker Systems, Performance and Studio Monitor

• On the pages that follow, you will find our Guide to speakers, both studio monitors and performance/stage types, each treated separately. The Guide is in chart form. A repeated heading of product specifications is on each page to aid in clarity.

• As usual, be aware that we attempt to contact every manufacturer but not all are prompt or cooperative enough for our necessary deadlines.

• On page 64 we provide a listing of each manufacturer's mailing address. Write to them directly for further information on their products listed —and tell them you saw their listing in **db Magazine**.

MONITOR SPEAKERS

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, In.	Weight, lbs.	Price, $	Features
ALTEC LANSING															
M600	26 22 17	black	black	8	60-20k	16	cone					1.5	66	$1.714.00	Full range sound; highly accurate sound reproduction contains famous Altec 604.
M300	22 12 9	black vinyl	black fabric	5.4	50-20k 3	8	cone	3.5	cone	1	dome	350/ 3500	28	$580.00	
M200	17 9.5 8.5	black vinyl	black fabric	6	65-20k 3	6.5	cone			0.75	dome	3500	19	$500.00 pair	
APOGEE SOUND															
SSM		black epoxy	black foam	16	280-21.5k 4	4.5	cone			1					Designed for linear, high power output, These monitors conquer the most demanding applications with smooth frequency response.
AE-3M	16 12.5 12	black epoxy	black foam	8	70k-18k	1-10	cone			1	horn		44	same	
AE-4M	14 14 23.25	black epoxy	black foam	8	55Hz-16.5k		1-12	cone		1	horn		78		same
AE-6	14 23 15.5	black epoxy	black steel	8/8	53Hz-17k	1-12	cone			1	horn		78		same
AE-6B	14 14 23.25	black epoxy	black foam	8/8	53Hz-17k	1.12	cone			1	horn		78		Designed for high power output. the epoxy monitors conquer demanding applications with smooth phase and frequency response.
AUDRA INTERNATIONAL															
Custom															Custom designed and fabricated units.
BOND ELECTROACOUSTICS c/o EQUITY SOUND INVESTMENTS															
DF-12 Granite	17.5 17.5 14	Granite marble	foam	8	80-17k	12.	cone			2.8	horn	1.6k	62	$2495.00	Actively cooled, 128 dB long-term spl.
CELESTION INDUSTRIES															
C 3	12 8 9	black ash	black cloth	8	75-20k 3	8	cone			1	dome	5k	8.4	$300.00 pair	
C5	13.8 8.1	black ash	black cloth	8	70-20k 3	8	cone			1	dome	4k	10.4	$450.00 pair	
ELECTRO-VOICE															
FR12-2	25.5 41.9 22.2	veneer		8	80-18k 3	12	cone			1.5	dome	1.5k	28		All Sentry series monitors
PI100	61 38.1 21.6	vinyl		8	80-18k 3	15	cone			1.5	dome	1.5k	23.7		
FR15-2	72.1 80 42.2	veneer		8	50-15k 3	12	cone			90x 30	horn	1.5k	43.5		
FR200	25.5 41.9 22.2	veneer		8	50-18k 3	12	cone			1.5	dome	2.	51		Sentry series
FM-1502ER	28.7 13.8 24.4		black grille	8	65-20k 3	15	cone			90x 40	horn	1500	72.2		
GAUSS															
3288				8	40-18k	12	cone			1	comp drvr	1.8k	24	$925.00	Raw coaxial, works well in 2 cu.feet, aligned to provide a phase correct relaitionship.
3588				8	40-18k	15	cone			1	comp drvr	1.2k	25	$990.00	As above, 15-inch coax.

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, In.	Weight, lbs.	Price, $	Features
JBL PROFESSIONAL															
Con- trol 1 Plus	9.25 6.25 5.63	black	black metal	4	120-20 3	5.25	cone			.75	dome	6k	4	$295.00	
Con- trol 5	15.25 9.88 9	black	black metal	4	75-20k	6.5	cone			1	dome	3k	10	$197.00	
4435	35.75 38 17.13	oiled walnut	blue fabric	8 3	30-16k (2)	15	cone				horn	1k	240	$2,395.00	
4406	15.38 9.38 8.5	oiled walnut	blue fabric	8	55-20k	6	cone			1	Ti dome	3k	17	$250.00	
4408	17.25 12 11.63	oiled walnut	blue fabric	8	50-20k	8	cone			1	Ti dome	2.5	26	$325.00	
4410	23.5 14.25 11.25	oiled walnut	blue fabric	8	45-20k	10	cone	5	cone		Ti dome	800 4.5	43	$495.00	
4312 A	14.25 23.5 11.25	oiled walnut	blue fabric	8	45-20k	12	cone	5	cone	1	dome	800 4.5	47	$575.00	
KLEIN & HUMMEL (GOTHAM AUDIO CORP.)															
098	15 10 7.25	gray	black cloth	4.7	50-16k 2.5	8	cone	1.5	dome	.75	dome	800 6k	26.4	$1,400.00	Active, tri-amplified, 2 electronic X-overs with location dependent equalizer. (amps:1-100W, 2-50W).
096	20.7 12.4 11.4	gray	black cloth	4.7	50-20k 2.5	10	cone	2	dome	.75	dome	600 4k	48.4	$2,360.00	Active, tri-amplified (3-60W), 2 electronic X-overs with location dependent equalizer.
092	31.5 17.3 11.8	gray	black cloth	4.7	50-17k 2.5	10 (2)	cone metal	3.5	cone	1	dome 3k	500	66	$4,500.00	Active, tri-amplified (1-120W, 2-60 W), 2 electronic X-overs with lo-cation dependent equalizer.
MEYER SOUND LABORATORIES INC.															
833	32 20 14.75	black satin/ walnut	cloth	8	35-18k 3	15	cone				rad horn		115	$6,930.00	Ultra-low distortion with high continuous peak output. System includes 2 S833 cabinets and control electronics unit.
834	38.5 24.13 20.13	black satin/ walnut	cloth	Syst Load	30-18k 4	18	cone						127	$1,845.00	Subwoofer for 833 system.
HD-1	16 12	Black satin			32-22K 40-20	8			dome	1	silk		51	$2.275.00	Self-powered near field reference monitor, corrected for amplitude and phase accuracy.
N.E.A.R.															
10M	11 7.17 9.75	black vinyl	cloth	8	48-20k 2	5.25	conk metal			1	metal		16	$175.00	Magnetic shielded fo use near video monitors.
30M	17 10.5	black vinyl	cloth	8	42-23k 2	8	metal conk			1	dome metal		25	$379.50	As above.
PANASONIC RAMSA															
WS- A10	8 12.5 8	black resin	black metal	8	80-16k	6	full- range						6	$120.00	Compact full-range, near-field monitor. Magnetically shielded. Accessories opt.
WS- A70	10.4 16.5 9.3	white black mold	white black wood	8	50- 18k	4.7	comp drvr		cone horn			2500	14.3	$240.00	
PEAVEY ELECTRONICS CORP. **See our ad on Cover III**															
PRM 312A	25 16 13	black vinyl	black fabric	8	42-20k 3	12	cone	6.5	cone	1	soft dome	300- 3000	58	$429.00	Phase reference monitor with acoustic foam baffle, linear phase system response.
PRM 2085	9.75 15 8.25	black vinyl	black fabric	8	68-20k	8	cone			1	soft dome	2.5k	16	$220.00	Phase reference monitor with acoustic foam to reduce diffraction, switch selectable response.
PRM	12.25 17.5 11.5	black vinyl	black fabric	4	45-20k 3	8	cone	5.25	cone	1	soft dome	300- 3000	26	$320.00	As above.
PROFESSIONAL AUDIO SYSTEMS															
SM-1	25	oak	black	8	37-20k	15	cone			7.25	horn	1.2	100	$4,800.00	The SM1 is a biamplified monitor designed for

	Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, In.	Weight, lbs.	Price, $	Features
	19.5 18.5	black	cloth		3					4.25				pair	midfield applications, offering the advantages to play at high levels without distortion.
SM-3	36 36 24	black paint		8&4 ohms	27-20k	15	cone	15	cone 4.25	7.25	horn 1.2k	100	210	$10,500. pair	The SM3 is a 3-way triamplified studio monitor system for very large control rooms wanting high levels, extremely low bass.
PROFESSIONAL TECHNOLOGIES															
PT 652D	17 12 14	oak	brown cloth	4- 8	40-20k 3	6.5	cone			1.25	dome	2.5k	40	$679.00 pair	D'ppolito configuration to maintain polar integrity, optional finishes available.
PT821	35.5 10.5 13.25	oak	brown cloth	8	31-20k 3	8	cone			1.25	dome	2.5k	60	$849.00 pair	
SHURE HTS See the Shure ad on Cover IV															
650SW	19.5 23 14	black oak	black cloth	8	33-80 4	12	cone						65	$499.00	Subwoofer with magnetic shielding.
640LRS	13.5 10 8.7	black oak	black cloth	5.6	80-18k 3	6.5	cone					2400	24.2	$299.00	Magnetic shielding, built-in inserts for Omnimount 75W brackets.
STAGE ACCOMPANY, USA															
SA 4549	38 35.8 22.4				30-32k	2X 15		15			ribbon			$22,558.00	This is the top studio monitor from the master series, includes amplifiers and cabling.
STUDER REVOX AMERICA INC.															
2706	12 15.1 13.5	dark gray nextel	black anod alum	4	42-20k 3	12	cone	2	dome	1	dome	720 5k	48	$795.00	Compact. Can be installed on floor stands or suspension brackets. High power dome transducers.
A723	22.9 12.7 16	wal- nut venr	black cloth		40-20k 3	12	cone	5	cone	1	dome	300 27k	68	$1,995.00	Each crossover has dedicated 100w amp. Useful in small to medium sized listening environments.
TANNOY NORTH AMERICA INC.															
PBM 6.5	11.88 8 8.5	pewter gray	black material	8	57-20k 3	6.5	poly	cone		.75	dome	2.6	10	$375.00	High sensitivity and power handling, rear pair firing tuned. Portable, compact-sized enclosure, ferrofluid cooled dome.
PBM 8	15 10.13 10.63	pewter gray	black material	8	47-20k 3	8	poly cone			1	dome	2.4	18.5 pair	$545.00	High sensitivity and power handling, rear firing extended low frequency response, cooled polyamide dome, five-year warranty.
System 8	18.1 11.8 9.1	lami- nated	black wood	8	48-25k 3	8	poly cone			1	horn	2.3k	26.4 each	$1,095.00	Eight in. point source phase coherent dual concentric transducer, high sensitivity high power design, bi-wired gold-plated terminals,.
System 10	22 14.3 11.4	lami nated	black wood	8	46-25k 3	10	poly cone			1	horn	2.3k	41.9 each	$1.595.00	10 in.point source phase coherent DMT. Gold-plated terminals, hard wired crossover network, DMT energy controlling enclosure.
TECHNICAL AUDIO DEVICES															
TSM-1	43.31 35.44 28.06	maple	black	4	29-20k	2 16	cone			2	90 40	650	319	$7,500.00	Designed for large studio control rooms.
TSM-2	26 31.44 24.19	maple	black	8	29-20k	16	cone			2	90 40	650	205	$5,750.00	For small control or editing rooms.
TOA ELECTRONICS															
265- ME-AV	13.8 8.1 9.6	gray poly	black jersey	8	60-20k 3	6.3	cone	1.2	dome			3k	11.5	$398.00	Magnetically-shielded components, smooth crossover, wide dispersion pattern.
280- ME	15.7 9.3 9.3	gray poly	black jersey	8	60-20k	7.9	cone	1.2	dome	0.79	dome	1.5k 14k	15.4	$518.00	same
312-	22.88	gray	black	8	50-20k	11	die-	4.7	cone	1.2	dome	500	35.7	$869.00	same

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
ME-AV	13.2 11.6	poly	jersey				cast frame					5k			

YAMAHA CORPORATION OF AMERICA

NS10 MS	8.5 15 7.75	black wood	black cloth	8	60-20k	7	sheet formed			2.4 dome	soft	2k	13.2	$237.50	Industry-standard studio close-field monitors. Available for commercial version.
NS40 M	23.5 11.5 12	black wood	black cloth	8	30-20k	7 (2)	sheet formed	2.4 dome	soft dome	1.2 dome	soft 5k	1.2k	37.4	$465.00	Bigger version of the NS10MS with greater low-end response.
S10X	6.125 9.5 6.5	black	black metal	8	65-20k	4	carb fiber						6.2	$145.00	Very compact wide range system. Handles up to 150 watts of program material.
S20X	7.5 11.625 7.75	black	black metal	8	65-20k	4 (2)	carb fiber						4.6	$210.00	Compact dual-driver wide range system. material.
MS101	8.5 7.7	black	black	10	30-20k	4	full 5.8		metal				4.9	$125.00	Has mic and two line inputs, volume and tone controls.
MS202	8.5 11.5 7.7	black	black metal	20		4	full range						8.8	195.00	Wide range with mic and three line inputs, Volume and tone controls.

YORKVILLE SOUND, INC.

YSM-1	16 10 9	black wood	black cloth	6	40-20k 3	6.5	cone			1	dome	2.5	18	$280.00 pair	5-way binding posts, front porting, time aligned, 90 dB sensitivity.
YSM-2	13 7 8	black wood	black cloth	6	80-20k	5.25	done			0.75	dome	2.2k	9	$200.00 pair	As above.
YSM-4	9 6 6	black plstc	black metal	8	50-18k	4	cone			0.75	dome		2k		Compact size, ideal for tight spaces,

PERFORMANCE SPEAKERS

ALTEC LANSING

937	24 18 16.3		black metal		70-15k	12	cone				const direc		49	$950.00	Two-way loudspeaker system—rugged, reinforced construction, 150 watts.
M200	17 9.5 8.5	black	black nylon		65-20k	6.5	cone					3500	17	$400.00	Easy installation in 8 ohm or 70V distributed application. Excellent for near field monitoring.
554A	9.6 7 5.3	black	black nylon		90-20k	4	cone					3500	4.6	$398.00	Compact, weather-resistant system has versatile omnimount brackets included.
M500	33 26.5 17.5	black	black	8	46-20k	10	dir. rad.					630	4	$1.398.00	Front mounted components—2-way vented system.
M400	23 17.75 17.5	black	black	8	80-20k	12	dir. rad.					2k	43	$799.00	Compact size with "big" source, 1 in throat, 150 watt power handling.
M300	22 12 9	black	black	8	50-20k	8	cone					3500	35	$550.00	Accurate sound reproduction for studio playback and nearfill applications, with 75 watts.

APOGEE SOUND

3X3	45 29 22	black epoxy	black steel	8L 8M	53-19k	15	cone	2	horn	1	horn		265		Designed for high power output, tight pattern control, warm musical response.
AE-1	10.25	black	black	8	63-19k	8	cone			1	horn		18		Designed as a foreground music

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
	16 8	epoxy coat.	foam coat.												system and for theatrical fill applications.
AE-3	16 10	black coat.	black	8	70-18k coat.	10	cone			1	horn		40		A powerful, compact vocal reinforcement system for front fill
AE-5	23 14 16	black epoxy coat.	black foam	8	53-17k	12	cone			1	horn		78		Performs as stand alone unit. Also installed to form large arrays.
AE-10	22.5 32 24	black epoxy coat.	black foam	8	38-120k	15 (2)	cone					120	138		Designed for convenient use and easy "truck pack." Provides high acoustic output.
AE-12	30 44.75 22.5	black epoxy coat.	black epoxy coat.	8	35-120k	18 (2)	cone					120	160		Engineered and designed for large scale sound reinforcement, providing very high acoustic output with linear low distortion response.

ATLAS/SOUNDOLIER

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
W150T	9 6.31 6.56	black metal	black metal	8	75-20k	5.25	cone			1	ferro fluid cone		10	$275.44	Fifty watt, indoor/outdoor loudspeaker system with Integral 70V Tx.
SEQ-844	20.44 13.125 9.125	oak/ black vinyl	black cloth	8	45-18k	8	cone			4	piezo cone	3k	22	$270.83	Tuned port extended bass enclosure; 5o watt system.
SEQ-1232	29 18.31 12.5	black vinyl veneer	black cloth	8	45-20k	12	cone			2x 5	piezo horn	3.5k	45	$520.83	Tuned port bass reflex enclosure; 65 watt coaxial system.

BOND ELECTROACOUSTICS c/o EQUITY SOUND INVESTMENTS

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
DF-12 Granite	17.5 17.5 14	Granite marble	foam	8	80-17k	12.	cone			2.8	horn	1.6k	62	$2495.00	Actively cooled, 128 dB long-term spl.

BOSE CORPORATION

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
402	23.25 8.12 7.25	mica foam	black	8	90-16k	4.5 drivers							15	$598.00	Designed for high-quality reinforcement of voice and music.
802II	13.5 20.5 13	mica foam	black	8	50-16k								31	$998.00	same
Acousti- mass	16 22.25 23.25	poly foam	black steel		10-18/ -3dB	4.5	12					150	72	$3.600.00	same

CARVIN

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
973	30 22 15.25	gray/ black ozite	black metal	8	50-19k 3	15	cone	6.5	cone	3.5x 4.4x 3.5	horn	400 4k	81	$369.00	Three-way speaker system with biamp capability; 400 watts.
993	47 25 19.25	grey/ black ozite	black metal	4	45-19k 3	15 (2)	cone	6.5 (2)	cone	3.5x 4.5x 3.5	horn	400 4k	146	$649.00	Three-way system with biamp capability (2) woofers, (2) middrivers, (2) tweeters; 800 watts.
1331	11 28 19	gray black ozite	black metal	8	1.5-18k (horn)		15	cone			horn	2k	46	$589.00	Two-piece system with biamp capability; 100 watt radial horn; 400 watt woofer.

CELESTION INDUSTRIES,INC.

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
SR Com	7 8.5 7	black polymer	black	8	100-20k +3/-5dB	5	cone						71	$199.00	For recording, stage or sound contracting applications. Has cone/dome radiator and edge wound voice coil.
SR-1	12 22 14.5	black polymer	black	8	60-20k +/-2	8	cone						34	$599.00	Excellent sound quality and projection for demanding sound reinforcement.
SR-2	28 23.5 20	gray paint		8	40-200k	18	cone						108	$950.00	The subwoofer adds massive low end punch without sacrificing portability.-
SR-3	10.25 13 9.5	black polymer	black	8	70-20k +2/-4	8	cone						201	$320.00	For use where moderate sound levels required. Handles 250 watts with SRC-1, 150 watts with SRC-3.

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, In.	Weight, lbs.	Price, $	Features
CERWIN-VEGA															
CVX-253	55.5 24 23.875	black paint	black metal	8	40-18k	15 (2)	cone	12x 20	horn	10x 3	horn	800 3.2k	173	$1.800.00	A full-range 3-way system with built-in passive crossovers for high performance.
CVX/H183	49.5 24 24	black paint		q8	40-18k	18	cone	12x 20	horn	10x 3	horn	350 3.2k	160	$2.000.00	Fully portable using compact single throat folded horn/full range with passive crossover built in.
V-37D	36 24	gray carpet	black cloth	8	40-15k	18	cone			10x 18	hom	1.2k	98	$800.00	Good for on the road or in small and medium clubs. Includes 2 position response equalizer.
V-30D	32 24 16	grey carpet	black cloth	8	40-15k	15	cone			10x 15	horn	1.2k	75	$600.00	Front baffle woofer for greater midbass efficiency and throw/controlled coverage.
V-15B	29.25 18.25 17.375	grey carpet	black cloth	8	35-20k	15	cone 14	9x	horn	2.25	horn	2.5k 5k	47	$500.00	A versatile triple action system; has 15 in. woofer with symmetrical magnetic field.
MV-15	29.75 19 17	grey carpet	black cloth	8	32-16k	15	cone	10	cone	3x 3	horn	150 3k	55	$500.00	Triple application system with vented 10 in. cone midrange features high output and natural sound quality.
L-36PE	36 24 36	gray carpet		8	30-30k	18	cone						160	$1,000.00	Has folded horn which packs high performance into a compact cabinet; 30Hz bass in a free-standing horn.
M-1	14.5 21 7.5	gray carpet	black cloth	8	70-16k			12	cone	3x3	horn	2.5k	36	$375.00	Features an ultra-compact vented enclosure with a 12 in. driver for non-fatiguing natural midrange.
COMMUNITY LIGHT AND SOUND															
RSJR	26 11.25 11.25	black carpet	grey	8	100-18k	2X 6.5	cone				horn	1800	38		
FB	26.25 18.25 14.25	black carpet	grey	8	45-18k	12	cone	6.5	cone		horn	200- 1800	75		3-way passiive of active tri-amped.
RS220	26.75 14.25 14.5	black carpet	grey	8	100-18k 3	2X 8	cone	2	horn		horn	800- 3000	60		
RS-660	33..6 20.5 20.5	black carpet	grey	8	70-18k 3	2X 10	horn	2	horn	1	horn	650- 3000	125	$1,160.00	
CERWIN-VEGA															
SM-12B	17 15 18	carpet	black metal	8	70-16k	12	cone			1	horn	2500	30	$375.00	
SM-15B	19 17 20	carpet	black metal	8	50-16k	15	cone			1	horn	3000	37	$500.00	
L-36PE	36 24 36	carpet		8	30-300	18	cone						166	$1,000.00	
B-36A	36 24 24	carpet		8	40-4k	18	cone						115	$800.00	
EAW See our ad on page 9															
FR153 HR	24.7 19.75 19.75	poly	metal	8	48-19k 3	15	cone	7	cone	1.75			88	$1.020.00	
FR253 HR	41.5 24.63	poly	metal	4	45-18k 3	2X 15	cne	2X 7	cone	1.75	hom		175	$1.750.00	
ELECTRO-VOICE															
FM-1202ER	24.3 19.1 11.7		black grille	8	75-20k 3	12	cone			90x 40	horn	1500	65	$1,895.00	
SH-1512ER	31.9		black	8	50-20k	15	cone			90x	horn	1600	75		

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, In.	Weight, lbs.	Price, $	Features
	24.7 16		grille		3					40					
FM1202ER	24.7 19.1 11.7		screen	8	75-20k 3	12									
FM1502ER	28.7 13.8 24.4		screen	8	65-20k 3	15									
SH1512ER	35.5 19.4 28		screen	8											
E3MC/EMILAR															
EM10	15 18.5 15.12	grey carpet	black cloth	8	40-20k 3	10	cone				horn	1500	39.25	$690.00	
2040	18 19.5	fiber- glass	black plstc	8	20-20k 3	15	cone				horn	1000	72	$2,950.00	
EM15	20 23 16.6	grey carpet	black metal	8	20-20k 3	15	cone				horn	900	57.25	$910.00	
EM12	18 21 15.25	grey carpet	black metal	8	20-20k 3	12	cone				horn	1200	44.75	$780.00	
FOSTEX															
SPA32	13 20 7.9	black plstc	black metal	4	55-18k	11.8	cone				horn		32	$899.00	Contains 250W RMS amp auto-eq'd for flat response
SPA11	13.8 7.1 8.3	black plstc	black metal	8	80-18k								16.75	$399.00	Contains 100W RMS 8-ohms amp.
GANE LOUDSPEAKERS															
SH12P	19 15 11	black carpet	black metal	8	80-17k 5	12	cone			1	horn	2400	28	$399.95	Fluid-cooled, 250W music program.
SH8P	13.25 10.5 8	black carpet	black metal	8	100-17k 5	8	cone			1	horn	4000	16	$229.95	Fluid cooled, 200W music pragram.
10TCA	16 12.5 9	oak grain	black metal	8	90-17k 5	10	cone			1	horn	3000	30		Fluid cooled, 25W music program.
GAUSS															
3588				8	40-18k	15	cone			1	horn	1.2k	25	$990.00	Coaxial rated at 200 watts RMS and a sensitivity of 96dB.
3288				8	40-18k	12	cone			1	horn	1.4k	24	$925.00	Coaxial rated at 200 watts RMS. Sensitivity of 91dB.
3285				8	70-15k	12	cone			1	horn	1.8k	24	$925.00	Coaxial rated at 200 watts RMSI for compact, high power floor monitor.
INTERNATIONAL MUSIC COMPANY															
TX122	21 16 16	gray carpet	black metal	8	60-18.5k +/-4dB	12	cone			4x 10	comp. driver	3k	45	$249.95	Performance speakers: metal stacking corners, reinforced OSB/plywood construction.
TX152	24 18 17	gray carpet	black metal	8	55-18.5k +/-5dB	15	cone			4x 10	comp. driver	3k	65	$399.95	same
TX153	28 23 16	gray carpet	black metal	8	55-18.5k +/-4dB	15	cone	8	cone	4x 10	comp. driver	300Hz 3k	88	$549.95	same
TX252	32 25 17	gray carpet	black metal	8	55-18.5k +/-4dB	15	cone			4x 10	comp. driver	1.5k	65	$499.95	Performance speakers: direct radiating metal stacking comers. OSB/plywood construction.
TX452HL	48 25 17	gray carpet	black metal	4	45-18.5k +/-4dB	15	cone		10	4x driver	comp.	1.5k	110	$649.95	Performance speakers: horn loaded low frequency drivers. metal stacking corners.
H118EV	43 26 18	gray carpet	black metal	4	40-18.5k +/-4dB	18	cone	8	cone	4x 10	comp. driver	300Hz 3k	125	$899.95	Performance speakers: Bi-ampable, horn loaded mid, metal stacking corners.
M152EV	17	black	black	8	55-18.5k	15	cone			4x	comp.	3k	56	$499.95	Stage monitor: metal stackng cor-

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	28 14		metal		+/-5dB					10	driver				ners, titanium compression driver.
M122HS	21 16 12	black	black metal	8	60-18.5k +/-4dB	12	cone			4x 10	comp. driver	3k	30	$249.95	Stage monitor: metal stacking corner, constant directivity horn.
JBL PROFESSIONAL															
SR4732	43 25 18	fabric	black metal	4	40-20k	12	cone				horn	1.2k 6k	135	$1,850.00	Vented Gap Cooling low frequency transducers.
SR4735	36.25 25 18	fabric	black metal	8	35-17k	15	cone	8			horn	600Hz 2.8k	92	$1,050.00	Pure titanium diaphragm compression drivers mated to patented JBL Bi-Radial horns.
SR4738	43 25 18	fabric	black metal	4	30-17k	18	cone	10			horn	600Hz 2.2k	111.5	$1,550.00	Trapezoidal enclosure design allows tight clustering of multiple enclosure systems.
SR4718	43 25 18	fabric	black metal	4	30-3.3k	18	cone						87	$895.00	Vented Gap Cooling low frequency transducers.
SR4722	28 20 13.5	fabric	black metal	8	53-17k	12	cone				horn	1.2k	57.5	$850.00	Pure titanium diaphragm compression drivers mated to patented JBL Bi-Radial horns.
SR4725	29.5 25 18	fabric	black metal	8	36-18k	15	cone				horn	1.1k	78	$1,050.00	Trapezoidal enclosure design allows tight clustering of multiple enclosure systems.
MEYER SOUND LABORATORIES															
MSL-10A	85 41 35	black	black metal		40-12k 4	4x 12	cone			3x2 comp.	horn		700	$38,000.00	Provides controlled coherent coverage. High acoustical output with low distortion. Rugged construction
MSL-3A	56.75 21.25 30	black/ gray carpet	black/ vinyl metal	4/8	70-20k 4	2x 12	cone			comp. w/ VHF array	horn		265	$4,950.00	Rugged, arrayable system capable of high power with high clarity and coherence.
UPA-1B	22.37 14.5 14.5	black	black steel	8	60-16k 4	1x 12	cone			1.4 comp.	rad. horn		66	$2,530.00	Rugged, arrayable system that is compact and versatile. Efficient operation.
UPM-1	18.125 6.75 7.125	black	black metal	16	70-20k 4	2x 5	cone			piezo electric	hom		17	$992.00	Ultra compact and lightweight with uncompromised sound quality.
UM-1A	14 14 22.5	black	black steel	8	60-16k 4	1x 12	cone			comp.	const. directivity		66	$2,390.00	Ultra low distortion and efficient high power with ultra flat frequency response. Floor monitor.
USW-1	21.56 31 21.31	black	black steel	4	40-100 4	2x 15	cone						115	$1,796.00	Subwoofer, high power, low distortion, compact.
MODULAR TECHNOLOGIES															
RL-153-P	16 24 21	ozite	black metal	8	30-15k 3	15	cone	1	horn		piezo	2.5k	45	$479.00	Rugged handles.
410	24 26 14	ozite	black	8	40-4k	10	cone						50	$499.00	
PANASONIC/RAMSA															
WS-A10	6.12 9.875 6.12	black/ white	perf. metal	8	70-18k		full range			4.75	cone		5.70	$120.00	Features moded resin enclosures for portablee usage and limited outdoor outdoor exposure.
WS-A80	11.12 17.06 9.31	black/ white	perf. metal	8	65-20k	8		cone		comp driver	horn	2500	16.5	$280.00	same
WS-A200	21.93 15.54 10.75	black/ white	perf. steel	8	70-20k	8	12	cone		comp. driver	horn	2500	35	$590.00	same
WS-A240	21.93 15.54 10.75	black/ white	perf. steel	8	30-Xover		12	cone					35	$510.00	same
WS-A500	21.93 15.54 10.75	gray/ white	perf. steel	8	100-20k		12	cone		comp. driver		1600	35.2	$1,000.00	same
WS-A550	21.93	gray/	perf.	8	30-Xover	12	cone						38.5	$650.00	same, subwoofer

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	15.54 10.75	white	steel												
PASO SOUND PRODUCTS, INC.															
C-1000	17 6.25 4.25	black	black	16	120-20,000	5	cone				3			232	Directional sound column for critical critical reinforcement.
C324HP	33 6.25 4.25	beige	beige	8	100-20,000	5	cone				3			16	$330.00Applications-rugged design, high output and uniform dispersion.
PEAVEY ELECTRONICS CORPORATION See our ad on Cover III															
115 HC	30.5 21.75 16.75	black carpet	black nylon	4	40-18k	15	cone	6.5	cone		horn	1.5k 8k	85	$899.99	Three-way enclosure; full bandwidth response; back widow and 22A driver equipped.
SP-2Ti	31.875 23.75 17.125	black	black nylon	8	60-16k	15	black			1	horn	800	87	$499.99	Two-way enclosure; high level passive crossover; biamp capability; integral stand adapter.
SP-4Ti	52.5 25.75 19.25	black carpet	black nylon	4	48-16k 3	15 (2)	black			1	horn	1200	135	$749.99	Wide range; 2-way enclosure; 22A driver and dual 15 in. black widow drivers; high level passive cross-
1545Ti	15.5 24.5 16.	black carpet	black metal	8	88-14k 3	15	cone				horn	1200	53	$499.99	Stage monitor with dual baffle angles
PANASONIC RAMSA															
WS- A70	13 21 12	black or wht paint	black or wht cloth	8	50-18k	8	cone			120x 120	horn	2k	14	$240.00	Power capacity is 80 watts. Magnetically shielded, variety of mounting options.
WS- A80	14 21.5 12	black or wht resin	black or wht metal	8	65-18k	8	cone			60x 40	horn	2.5	16	$280.00	High SPL, compact, stackable, variety of mounting options.
WS- A200	28 20 16	black or wht resin	black or wht metal	8	70-20k	12	cone			60x 40	horn	2.5	35	$590.00	125 watt power handling, compact, high SPL, stackable, variety of mounting options.
WS- A240	28 20 16	black resin	black grille	8	35- X-over	12	cone						35	$510.00	Subwoofer system requires model WS-SP2A electronic crossover.
WS- A500	22 10.7 16	gray or white	metal grille	8	100- 20k			12	cone		horn	1.5k	40	$1200.00	A mid/high system for use with two WS-A550 low-frequency systems and WS-SP2A crossover.
WS- A550	22 10.7 16	gray or white	metal grille	8	35- X-over	12	cone						35	$680.00	
PROFESSIONAL AUDIO SYSTEMS															
RS-2	24 17 19	black paint	black steel	8	50-15k	15	cone	15	cone	7.25x 4.25	horn	1k	82	$2,400.00	A full-range coaxial speaker system designed for the professional. stallation.
SW-2	20.5 17 23	black paint	black steel	8	60-17k	15	cone	15	cone	7.25x 4.25	horn	1k	68	$2,400.00	A full-range 2-way compact stage monitor.
RENKUS-HEINZ															
FRS 1S1C	20.25 30 16	black carpet	perf. metal	8	50-17k	15	cone			1	comp. driver	1600	75	$1,010.00	Incorporates constant directivity horns and hardwood ply construction.
FRS152CD	48 20 18	black carpet	perf. steel	4	50-17k	15 (2)	cone			2	comp. driver	1200	123	$1,595.00	same
SR-1A	30 19.75 16.25	black carpet	black metal	8	50-17k 3	15	cone				horn	1200	73	$1,342.00	
SR-2A	51 23.75 29	black carpet	black metal	8	50-17k 3	2X 15	cone				horn	1200	128	$2,400.00	
ROSS SYSTEMS															
RCS252	30 25	black carpet	black metal	8	50-18k 4	15	cone				horn	1600	65	$499.95	

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RCS452	17.5 44 25 17.5	black carpet	black metal	4	14-18.5k 4	2X 15	cone				horn	1600	110	$649.95	
TOA ELECTRONICS, INC.															
380-SE	29.8 19.6 16.1	charc. gray	black mesh jersey	8	50-20k	15	cast frame	3.7	CD horn	3.7	horn	800 8k	79.2	$998.00	
480-SE	32.3 22 17.7	charc. gray	black mesh	8	45-20k	18	cast frame	3.7	CD horn	3.7	horn	600 8k	99.2	$1,125.00	
TURBOSOUND															
TXD- 520	16 11 10	blue	perf steel	8	100-18k 4	10	cone			1	soft dome		26	$665.00	Low frequency enclosure for TXD series.
TXD- 530	12 25 13	blue	perf steel	8	90-20k 4	10	cone				slot tweet		45	$1.064.00	Wide dispersion.
TXD-560	34 19 15	blue	perf. steel	8	60-18k	15	cone	10	cone		slot tweet	250- 4000	931		Compact, wide dispersion. available
TXD-518	29 21 17	blue	perf. steel	8	45-250Hz	18	cone						82	$999.00	Low frequency enclosure for the TXD series.
WHARFEDALE/OPTIM AUDIO															
FORCE 9	22 15 15	blue gray	blue metal	8	70-20k	12	cone			1	horn	2500	38	$990.00	High intensity sound field, 122 dB spl with 250W, coaxial.
FORCE 9SB	22 15 15	blue gray	blue metal	8	40-600	12	cone						36	$990.00	Add deep bass to other FORCE speakers, same power handling as above.
FORCE 5	17 12 12	blue gray	blue metal	8	70-20k	8	cone			1	horn	2500	20	$590.00	Delivers 115 dB SPL with 125W, coaxial.
FORCE 12SB	43 18 25	blue gray	blue metal	8	30-1k	2-	cone						160	$2250.00	Clean powerful bass, 131 dB SPL /170W.
YAMAHA CORPORATION															
S4115 HIII	30.7 25.2 18.2	black	black metal	8	45-16k	15	carb fiber			1.7	comp drvr horn	1.6k	97	$695.00	New version of the 2-way stage monitor.Rugged cabinet
S2115 HIII	21 23.3 26.2	ply w/blk	black metal	8	50-16k	15	carb fiber			1.7	comp drvr	1.6k	77	$695.00	High power-handling and excellent low frequency response. Rugged 2-way system for stage monitoring.
SW118II	30.7 25.2 18.2	ply w/black	black metal	8	40-3k	18	cone					150Hz	90	$575.00	High power subwoofer system. Rugged cabinet with recessed handles.g
S3112MT	15.7 12.4 25.2	ply w/black	black metal	8	50-20k	12	cone	6.5	cone	3.2	bullet comp. driver	1k/ 8k		$495.00	
S3115HT	30.7 25.2 18.2	ply w/black	black metal	8	40-20k	15	carbon fiber	1.7	comp. driver horn		ring rad comp.	1.6k/ 8k		$775.00	High power-handling 3-way system with wide bandwidth.
YORKVILLE SOUND															
M-160	17.9 14.6 13.6	black ozite	black metal	8	65-19k 3	10	cone			9x5	horn	4k	33	$479.00	Dual purpose electronic processor to linearize bass response or as stereo crossover with subs.
M-600	19.5 23.2 13.4	black ozite	black metal	4	50-16k	10 (2)	cone			9x 5	horn	1.8	77	$989.00	same
MX- 401	27.2 21.7 16.1	black ozite	black metal	4	45-19k	15	cone			6x 13	horn	2k	66	$899.00	same
MX-2000	45 26 19	black ozite	black metal	4	50-17k	15 (x2)	cone	11.5x 18.5	horn		slot	1-7k	175	$1,949.00	Plywood construction, electronic processor, biampable, transistor protection circuit..

Addresses

Altec Lansing Corporation
P.O. Box 26105
Oklahoma City, OK 73126

Apogee Sound Inc.
1150 Industrial Ave., Suite C
Petaluma, CA 94952

Atlas Soundolier
1859 Intertech Drive
Fenton, MO 63026

Audra International
P.O. Box 38
Silverado, CA 92676

Bond Electro Acoustics c/o Equity Sound Investments
P.O. Box 752
Sturtevant, WI 53177

Bose Corporation
The Mountain
Framingham, MA 01701

Carvin Corporation
1155 Industrial Ave.
Escondido, CA 92025

Celestion Industries Inc.
89 Doug Brown Way
Holliston, MA 01746

Cerwin-Vega
555 E. Easy St.
Simi Valley, CA 93065

Community Light & Sound
333 East Fifth St.
Chester, PA 19013

Dynacord
200 Sea Lane
Farmingdale, NY 11735

Eastern Acoustic Works, Inc.
One Main Street
Whitinsville, MA 01588

E3MC/Emilar
1620 N. Missile Way
Anaheim, CA 92801

Electro-Voice
600 Cecil St.
Buchanan, MI 49107

Fostex Corp. of America
15431 Blackburn Ave.
Norwalk, CA 90650

Gane Loudspeakers c/o Equity Sound Investments
PO Box 752
Sturtevant, WI 53177

Gauss
9130 Glen Oaks Blvd.
Sun Valley, CA 91352

JBL Professional (UREI)
8500 Balboa Blvd.
Northridge, CA 91329

Klein & Hummel (Gotham Audio Corp.)
1790 Broadway
New York, NY 10019-1412

Martin America
22930 Miller Rd.
Chicago Heights, IL 60411

Meyer Sound Laboratories Inc.
2832 San Pablo Ave.
Berkeley, CA 94702

Modular Technologies
1380-C South Pennsylvania Ave.
Morrisville, PA 19067

N.E.A.R., New England Audio Resource
569 Lisbon Road
Lisbon Falls, ME 04252

Panasonic/Ramsa
6550 Katella Ave.
Cypress, CA 90049

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Pelham, NY 10803

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San Marcos, CA 92069

Professional Technologies
Box 282A, Rd. #1
Rome, NY 13440

Renkus-Heinz
17191 Armstrong Ave.
Irvine, CA 92714

IMC (Ross Systems)
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Fort Worth, TX 76102

Shure Brothers Inc.
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Evanston, IL 60202-3696

Stage Accompany USA
65-60 Booth St. #3J
Rego Park, NY 11374

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

Sunn (Fender Music)
7975 N. Hayden Rd.
Scotsdale, AZ 85358

Technical Audio Devices
2265 East 220 St.
Long Beach, CA 90810

TGI North America (Tannoy)
300 Gage Ave., Unit 1
Kitchener, ON.N2M 2C8, Canada

TOA Electronics
601 Gateway Blvd., Suite 300
South San Francisco, CA 94080

Turbosound, Div. AKG Acoustics
PO Box 1383
Pleasant Valley, NY 12569

Wharfedale/Optim Audio
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Stamford, CT 06902

Yamaha Corporation of America
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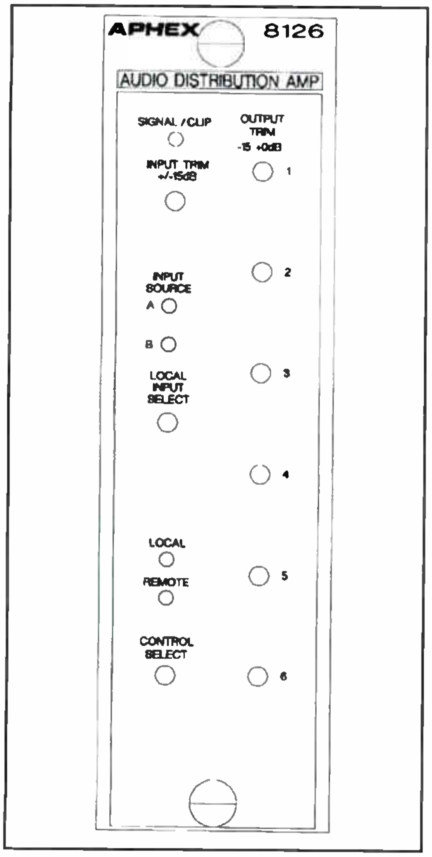

• The versatile 8126 Amplifier and Rack, equipped with a single, high-impedance input and six low-impedance outputs, are ideal for a variety of applications requiring signal splitting and routing of several destinations. Unique to the 8126, the servo-balanced, RF suppressed input is switchable via an A/B selector that can be accessed from the unit itself, or by remote operation. In either case, a visual indication of the selected source is provided by an LED. The six outputs offer the same transformerless, servo-balanced circuitry, plus an added feature providing tremendous benefit to users: complete protection from damage to the power supply, achieved by the installation of a fuse and diode on each leg of the outputs. As a result, maximum interface safety and freedom from breakdown are insured, even in the most complex applications. As a further convenience feature, each output has its own amplifier and level control which is recessed to prevent accidental alteration of the settings. The 8000 Series Rack, includes a power sensing circuit that automatically switches to a backup supply whenever a rack's primary power source is interrupted. This allows the system to remain operational even when the main power supply is removed for maintenance or repairs. Upon re-installation of the main power supply, the sensing circuit automatically switches back to the main power supply and operation is returned to normal. A visual alarm on the front panel will illuminate when the system is working on backup power.

Manufacturer: Aphex Systems
Prices: 8000, $420.00; 8126, $400.00
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NEW STEREO CONDENSER MIC

• The AT822 is equipped with a pair of wide-range, closely-matched cardioid (unidirectional) condenser elements, optimally-positioned and so uniform in response that the AT822 fully reproduces the spatial impact and realism of a live sound field while consistently delivering natural response across an arc of 170 degrees. The stereo elements' ultra-low-mass diaphragms ensure superb transient response and wide 101 dB dynamic range. Frequency response is flat from 30 Hz to 20 kHz, with maximum input SPL rated at 125 dB. The high output, stereo QT822 terminates its standard cord with two mini plugs threaded inside a pair of ¼-inch phone plug adapters. Also included is a mic cable terminating in a single stereo mini plug—compatible with a common input format for portable semi-pro and consumer stereo DAT recorders. It includes a switchable low-cut filter, windscreen, and camera shoe mount adapter. It operates in a standard 1.5 volt AA battery. Current demands are so low that battery life exceeds 1,000 hours with normal intermittent use.

Manufacturer: Audio Technica
Price: $299.00
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CUSTOM ENCLOSURES

• Modular System V provides the flexibility to design a coordinated custom enclosure configuration with the advantages and cost efficiency of standard components. This series of metal enclosures is based on a structural framework of 14 gauge MIG-welded steel with 11 gauge corner caster gussets and 14 gauge mounting rails. Series V catalog offering includes choice of 7 heights, 3 widths, and 2 depths of modular vertical cabinets, 5 factory-assembled consoles and 3 models of electronic pedestal desks. The multi-form channel system offers superior interlocking support, convenience and durability as required in industrial, institutional, commercial and military applications for computer-control equipment, telecommunications, switching, studio electronics, building management, industrial controls, communications and security systems.

Manufacturer: Atlas / Soundolier
Price: to be announced
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ACTIVE CROSSOVER SYSTEM

• The EX-24 offers twelve selectable crossover frequencies per channel ranging from 80 Hz to 6.3 kHz, and frequency setting may be locked at a given setting. Frequency settings are 80, 125, 160, 250, 500, 630, 800, 1,250, 1,600, 2,500, 5,000 and 6,300 Hz. A choice of ¼-inch unbalanced or balanced, as well as XLR-type three-pin balanced inputs and outputs provides flexibility in interconnection. Each channel features low and high output level controls, on/off switches for each output to aid in system setup, and polarity switches. Infrasonic filtering (–3 dB at 30 Hz) is built in to make certain that woofers are not overworked and amplifier power is not wasted on very low frequencies. Stereo or mono low-frequency output is also offered, providing the option of a single-channel subwoofer system. It also has a switchable horn equalization circuit to achieve flat frequency response with constant-directivity horns. The power supply is internal, and may be configured for operation anywhere. An IEC connector with detachable ac line cord is included. The main PC board uses surface mount technology (SMT), allowing for larger traces on the PC board as well as more shielding. As a result, distortion and noise are minimal.
Manufacturer: Electro-Voice
Price: $598.00
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D/A CONVERTER

• Developed in collaboration with Goldmund, a Swiss sister company specialized in ultra-sophisticated domestic audio products, the Stellavox Stellamode uses a new technology developed by a common laboratory. If the companies refuse to disclose in detail the converter technology they use, they announce some performance far away from usual figures:

—Non-linearity: dB level typically <1 dB at –100 dB.

—Noise: typically <100 dB.

—Phase error: typically <1.5 degree between channels.

The unit can convert from a digital AES/EBU input or S/PDIF input. However it includes an unusual but very convenient new feature: An AES/EBU loop without breaking or changing any other connection. The analogue outputs exist in symmetrical or asymmetrical and can be adjusted both in level (0, +4 dB, +6 dB), and in polarity (front panel phase inverter). To make the unit able to evolve with technology, the 2 key parts of circuitry (digital interface and converter-filter) are molded in pluggable modules which can be exchanged easily to be replaced by more accurate ones as soon as the company will produce them.
Manufacturer: Stellavox
Price: not available
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SMPTE/MIDI SYNC CONTROLLER

• The basic Micro Lynx supports two transports plus MIDI and features: SMPTE timecode generator, MIDI timecode generator, two transport synchronizer/revolvers with wide band readers, MIDI-to-SMPTE synchronizer, computer control port plus direct Macintosh interface. Micro Lynx is also expandable. Its options include: third timecode reader and synchronizer/resolver, Digital Audio Workstation interface providing synchronized digital audio sampling clocks, VITC timecode reader and Video Sync Generator in NTSC or PAL. TimeLine is known worldwide for the Lynx Time Code Module, a pioneering product which has become a defacto standard for professional machine control. Additional products in the line integrate multi-machine system synchronization and interface to console automation systems. The new Micro Lynx marks TimeLine's entry into the rapidly growing project studio market. "The technology we've developed in machine control for the world's most advanced recording and post-production facilities has been channeled directly into this new streamlined, self-contained system.
Manufacturer: TimeLine
Price: $2,495.00 (including keyboard)
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AUTOMATIC HISS REDUCTION

• The front panel slider of the 563X Hiss Reducer, which can be strapped for stereo operation, can be adjusted for the hiss characteristics of a particular system. This activates a sliding filter network which evaluates the spectral content of a signal. As "useful" high frequencies occur, the filter instantaneously opens and the useful high frequencies pass through unaffected. After the transient high frequencies pass, the filter closes again to the preset point. Once the 563X has been programmed, operation is automatic. Since the sliding filter closes to remove hiss, equalization may be employed after the unit without increasing overall hiss.
Manufacturer: dbx, a division of AKG Acoustics, Inc.
Price: $229.00
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New Products Information

The information on these pages is from the respective manufactures. The reader service numbers you circle are processed by computer and information will come later directly from those manufacturers.

Note to Manufacturers

To have your product shown in our New Products pages, you need only send us the release, a clean b/w photo and a suggested user price. Of course, this is on a space-availability basis.

db Magazine 1991 Index

INDEX BY TITLE

THE ELECTRONIC COTTAGE

THE BROADCAST ENGINEER

AUDIO FOR THE CHURCH

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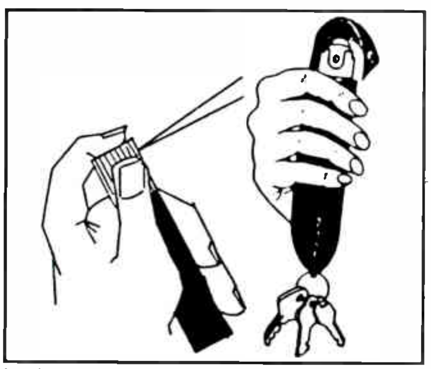

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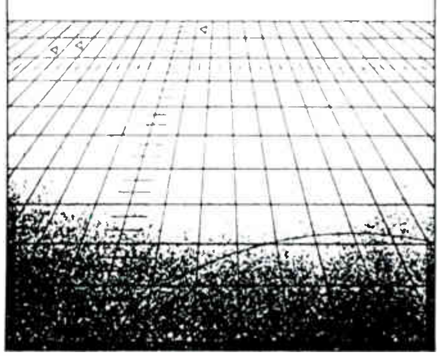

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PEOPLE, PLACES & HAPPENINGS

• **Frank Serafine**, composer/ sound designer of **Serafine Inc., Venice, CA**, has finished composing a music score to accompany the new **CBS Entertainment Production ID**, replacing music that has been with the original logo for the past fifteen years. Serafine created a new, high-tech image, but with a traditional orchestral quality associated with CBS' original music, according to **Andy Hill**, executive producer of CBS. Serafine's credits include *The Hunt for Red October*, as well as numerous other feature films and commercials.

• **Tannoy/TGI North America, Inc.**, has become the new, exclusive United States distributor of **Bruel & Kjaer** professional audio products.

• In a move geared toward increased distribution within the European and North American broadcast markets, **Studer Revox AG, a subsidiary of the Motor Columbus Group (Baden, Switzerland)**, has announced the completion of the acquisition of a majority interest in **Digitec S.A.** The Digitec organization has changed its name to **Studer Digitec S.A.** and operates as a full family member under the Studer Division of SRAG. Studer Revox has also delivered a Studer Dyaxis workstation to KWMX-FM, "Mix 101." The radio station is the first in Seattle, WA, to install and use a digital audio workstation for their day-today production tasks.

• **The Toy Specialists** has taken delivery of the **Roland Sound Space Processing System**, which enables a three-dimensional aural stereo system. No additional equipment is required for playback. The Toy Specialists is the first rental company in the United States offering the RSS.

• **Post FX**, of Boston, MA, is the first video post production boutique to offer the editing capabilities and technology of Sony D-2 tape. The facility, which opened in December, 1990, counts Hewlett Packard, Insight Producers, Lechmere, Production Values, Raytheon and Smash Advertising among its clients.

• **Skywalker Sound**, a division of **LucasArts Entertainment Company**, has installed the first production models of the new Console Control Unit (CCU) from **TimeLine**. The CCU operates the TimeLine System Supervisor multiple machine controller which interfaces to standard console automation software. The system interfaces to Skywalker's console automation, allowing any machine to be designated the "master," with "slaves" and time offsets automatically calculated through the new CCUs. The facility also houses a full-size THX screening theater.

• **Musicon**, a full-service audio recording and duplicating facility located in suburban Portland, has expanded its duplication capacity with the addition of four more slave recorders to the plant's existing eight-slave Versadyne 1500 system.

• **Adams-Smith** has been granted a European patent for high-speed recovery of Vertical Interval Time Code. Adams-Smith already holds United States and Canadian patents for high-speed recovery of VITC, and has applied for an equivalent patent in Japan.

• **E-mu Systems, Inc.**, has appointed **Peter A. Hayes** to vice president of marketing...Other promotions and appointments include **Bob Green** and **Richard Fusco** as electro-acoustic designers at **Jaffe Acoustics Sound System Department**...**Ferdinand Boyce** has been appointed vice president, marketing, at **DOD Electronics**...**Brian M. Maloney** will succeed **Thomas E. Yingst** at **Harris Corp**. as vice president—general manager of the **Harris Broadcast Division**. Yingst is retiring from full-time management...**Audio Plus Video International** promoted **Anthony Beswick** from chief engineer to director of engineering and post operations. Beswick's promotion coincides with the growth of the company through the addition of **International Post** and APVI's forthcoming expansion into New York City...**Curt A. Rawley**, currently president and chief operating officer of **Avid Technology, Inc.**, has been appointed president and chief executive officer. **William Kaiser** of **Greylock Management** was also appointed as acting chairman of the board...and **Robert "Bob" Mahoney** has been appointed director of marketing at **NVision, Inc.**

• **Digital Designs International** has moved its entire facility to Oklahoma City, OK. The new address is 100 North Quapah, Suite K, Oklahoma City, OK, 73107.

• **Ed Learned,** contributing editor to **db Magazine**, will be on tour with the **Dirty Dozen Brass Band** from Nov. 8 to Dec. 11, and with the **Pharoah Sanders Quartet** from Jan. 19 to Feb. 19. His approximate tour schedule is as follows:

Ed Learned with the Dirty Dozen Brass Band

Nov. 8-10 Port Moresby, Papua, New Guinea; Nov. 12-14 Bangkok, Thailand; Nov. 15 Chiang Mai, Thailand; Nov. 16 Songkhla, Thailand; Nov. 18-19 Indonesia at the Jakharta Jazz Festival; Nov. 21 Bandung, Indonesia; Nov. 22 Yogyakarta, Indonesia; Nov. 24-28 Malaysia (at) Kuala Lumpur and Penang; Nov. 29-Dec. 3 Singapore; Dec. 4 Cebu, Philippines; Dec. 5-8 Manila, Philippines; Dec. 9-10 Taiwan, Taipei; Dec. 11 Kaohsiung, Taiwan; Dec. 12-13 Taiwan, Taipei.

Pharoah Sanders Quartet (African tour)

Jan. 19-22 Abidjan, Ivory Coast; Jan. 23-27 Accra, Ghana; Jan. 28-Feb. 1 Lagos, Nigeria; Feb. 2-5 Libreville, Gabon; Feb. 6-10 Brazzaville, Congo; Feb. 11-13 Yaounde, Cameroon; Feb. 14 Douala, Cameroon; Feb. 15-19 Dakar, Senegal.

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