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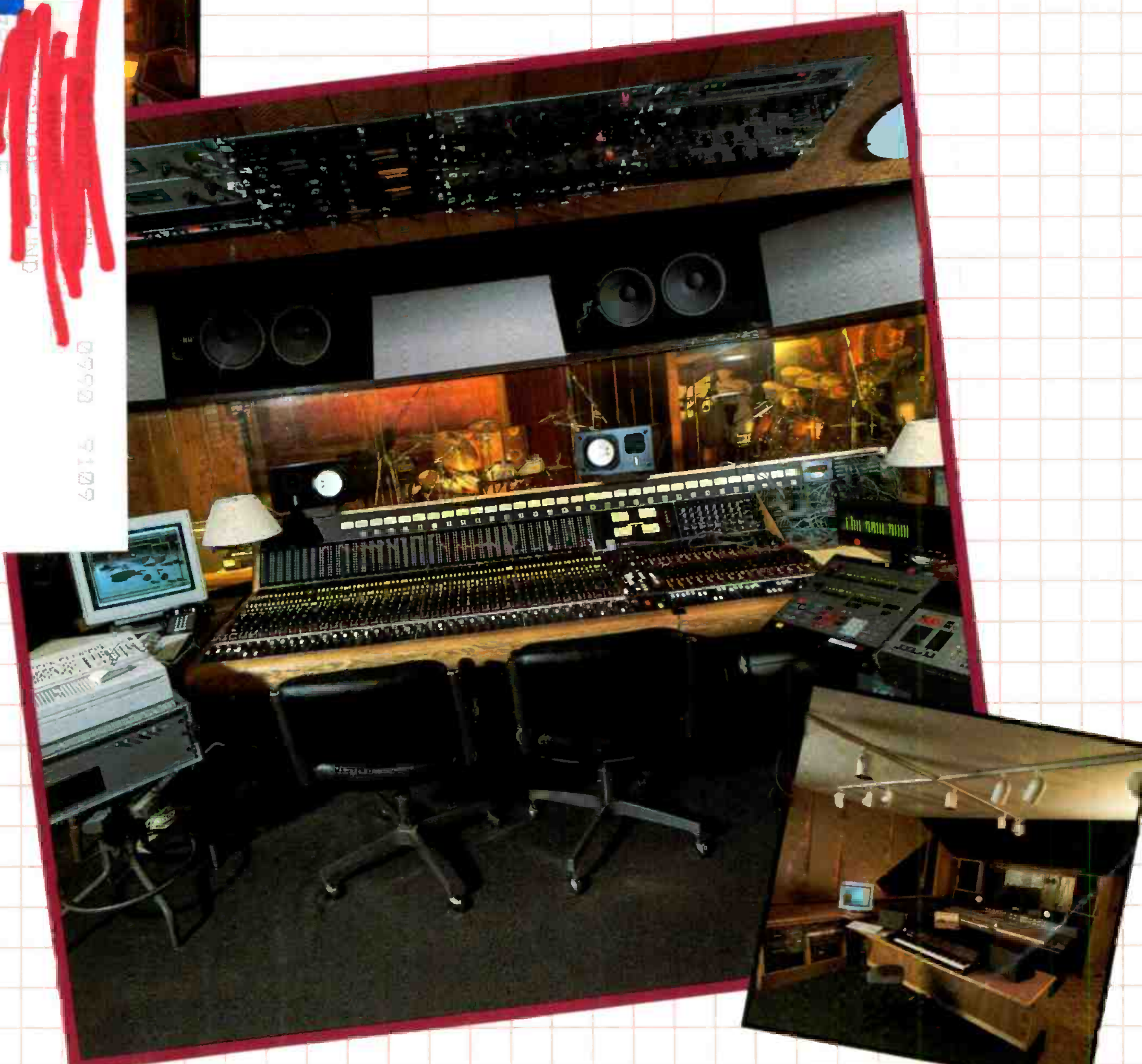
The Hunt for Red October

Church Audio

and Barilla and Hoffner



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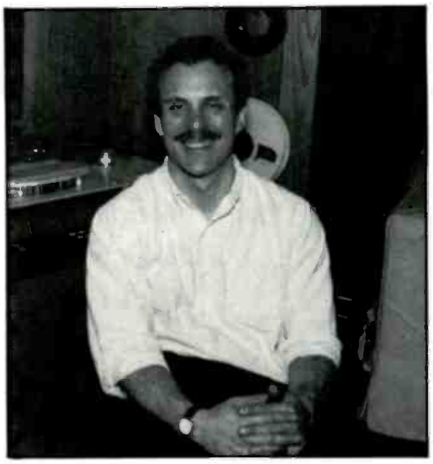
See page 41

About the Cover

• These are views of The Plant, located in Sausalito, California. If you cross the Golden Gate Bridge heading north from San Francisco, Sausalito is the first major exit you get to on the other side. The town is nestled along the bay side and has long been known for its marina ambiance and easy access from the big city. The hills move up sharply from the bay, offering spectacular vistas across that bay. In such a setting, the ultra-modern studio complex offers the best of today's technology in surroundings that offer privacy and natural beauty. See page 19 for the full story.

the sound contracting engineer

the recording engineer

THE ELECTRONIC COTTAGE

the broadcast engineer

Calendar

• The fifth Programming and Production Showcase, a mini-fair of products and services for the radio industry, will be presented at the RADIO 1990 convention of the **National Association of Broadcasters** Sept. 12-15 in Boston, MA. The Engineering Conference begins Sept. 11. The showcase will be held in the Special Events area of the exhibit hall Friday, Sept. 14 from 4:30 to 6:00 p.m. Exhibitors will present their products and services, including syndicated programs, programming software, jingles, entertainment graphics, voice-overs and production libraries. To register for RADIO 1990, call **(800) 342-2460, or for more information, contact Aimee Jennings at (202) 429-5402 or FAX at (202) 775-2146.**

• A call for papers for the **25th Annual Television Conference** of the **Society of Motion Picture and Television Engineers** has been issued by Frank J. Haney, SMPTE Editorial vice president. The conference will be held in Detroit, MI, site of the first SMPTE television conference 25 years ago. The meeting will be held Feb. 1-2, 1991. The conference's theme is "A Television Continuum—1967 to 2017." Those wishing to present papers may submit a 500-word synopsis and a completed author's form by Sept. 14, 1990, to **SMPTE Headquarters, 595 W. Hartsdale Ave., White Plains, NY 10607. Forms are available from Marilyn Waldman.** Authors will be notified of acceptance before Oct. 15; final manuscripts of accepted papers are due back at Headquarters by Jan. 7, 1991. Authors who submitted papers for SMPTE's 132nd Technical Conference and Equipment Exhibit will be notified before Oct. 31. Completed forms and synopses are due Sept. 14.

• **Synergetic Audio Concepts** of Norman, IN, announces new dates for the Intelligibility Workshop II which will be held at Indiana University Oct. 7-9, and will deal with the measurement of speech intelligibility and its uses in planning sound reinforcement systems with acceptable intelligibility characteristics. **Intelligibility II** will employ new technical tools, unavailable at the time of the earlier workshop, to advance the understanding of the participants with respect to the proper measurement of speech intelligibility of a sound system. Special emphasis will be placed on understanding the role of the pinna and the ear canal's directional influence in intelligibility. A DAT recorder will be used to make recordings of %ALcon losses using live listeners, TEF analysis and in-the-ear measurements. DAT recordings will be made of 5, 10, 15, 20 and 25 percent Articulation Loss of Consonants. The

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role of Signal-to-Noise and late reflections will also be studied. For further information, please contact **Synergetic Audio Concepts at Rt. #1, Box 267, Norman, IN 47264. Phone (812) 995-8212 or FAX (812) 995-2110.**

• **The Society of Broadcast Engineers** has announced it will hold its Fall affiliate engineering meeting at the 1990 SBE convention in St. Louis, MO. The SBE convention will provide more than 3 1/2 days of intensive training in television and radio broadcast technology. Coupled with the largest exhibit floor ever assembled by the SBE, engineers will be able to see and test the latest in broadcast equipment and technology. A special series of day-long seminars are held in conjunction with the convention. The Ennes workshops will be held Wednesday, Oct. 3. For more information, please call **(317) 842-0836 or (317) 842-0394.**

• **SPARS, the Society of Professional Audio Recording Services,** will conduct several meetings at the annual **Audio Engineering Society Convention** in Los Angeles, CA Sept. 21-25. An Educational Conference to discuss SPARS programs and review the National Studio exam will be held Friday, Sept. 21, from 3-6 p.m. The election of the 1990-91 officers will take place at the Annual General Membership Meeting, which will be held at Sound-Works West on Saturday, Sept. 22 at 8:30 a.m. A SPARS/AES Seminar will be conducted Saturday, Sept. 22 from 2-5 p.m. Topics include diversification, rate structures and successful marketing in the 90s. **An informal meeting with Advisory and Associate Advisory members will be held at the SPARS Manufacturers Interface Sunday, Sept. 23 at 8:00 a.m. at the Los Angeles Hilton and Towers.**

• A series of workshops and seminars for music ministers, musicians, audio, video and lighting technicians, volunteers, pastors and building committees has been announced by **Soundcheck Workshops**. In addition to their **Sound Reinforcement for Churches** seminar, Soundcheck Workshops is scheduling five more seminars and three new workshops across the United States.

Several of the seminars include:

PA Loudspeaker Evaluation Clinic, Sept. 8, Atlanta, GA;

Effective Use of Synthesizers in Worship, Sept. 8, Atlanta, GA and Oct. 20, St. Louis, MO;

Where and How to Buy a Sound System, Sept. 14, Mt. Vernon, IL, Sept. 21 at the AES Show in Los Angeles, CA, Oct. 5, Atlanta, GA and Oct. 12 in Milwaukee, WI;

Sound Reinforcement for Churches, Sept. 15, Mt. Vernon, IL, Sept. 22 at the AES Show in Los Angeles, CA and Oct. 13 in Milwaukee, WI;

Critical Listening and Sound Mixing Class, Oct. 6, Atlanta, GA;

Five Day Sound Reinforcement Retreat, starts Oct. 15, St. Louis, MO;

Where and How to Buy a Church TV System, Oct. 19, St. Louis, MO;

Videotaping the Church Worship Service, Oct. 20, St. Louis, MO;

Five Day Intensive Study in Sound Recording, starts Oct. 29 in Atlanta, GA;

Five Day Sound Recording Retreat, starts Oct. 29 in Atlanta, GA.

Summer and Fall classes are being scheduled in Texas, Georgia, Missouri, Illinois, Ohio, Kentucky, California and New York, with more to come. Contact **Soundcheck Workshops at 1471 Colgate Drive, St. Charles, MO 63303 or call (314) 946-4360 for more information.** db

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The Multiverb LT gives you the power of the Multiverb with the simplicity of 1-touch control. The LT has 192 of the finest studio multi-effect combinations ever created! For those who don't have time for the complexity of programming, the LT gives you all the power you can use at a great price! Midi addressable.

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ALAN P. KEFAUVER

Down Mexico Way

As I was peacefully sitting in my office one fine afternoon, the phone rang and my friend and mentor Harold Boxer said, "How would you like to help me build a recording/broadcast studio in Mexico?" Without a moment's hesitation, I accepted. After further conversation, it turned out that through a conductor friend, Boxer had been asked to submit a proposal to the University of Mexico in Mexico City for a two-track recording/broadcast facility to record the UNAM (Universidad Nacional Autonoma de Mexico) Orchestra. The University Orchestra is a paid professional union orchestra and has a 44-week season. They play in a hall on the UNAM campus that seats about 2600 people. Audience seating is principally in front of the orchestra, but there is also a large balcony and additional raised seating behind the orchestra. These rear seats wrap around to the sides of the stage area, and are often used for a chorus when works such as the Beethoven *Symphony Number 9* are performed.

To this point, recording for delayed broadcast and some live broadcast programs from the hall had been done in a hit-or-miss fashion. The requirements were pretty simple. "Provide us with a facility that will generate high-quality stereo recordings of the orchestra for duplication and distribution to radio facilities throughout the country. Include the capability for live broadcasts." Since Boxer and I have collaborated on similar facilities and broadcasts earlier, including the Philadelphia Orchestra broadcasts, we felt it would be a relatively simple task to meet these specifications.

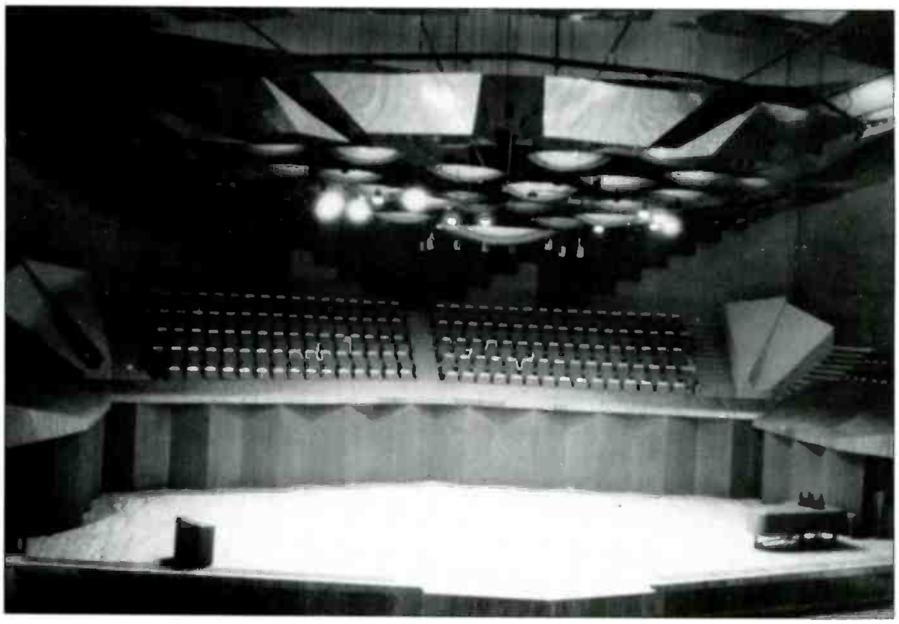

Figure 1. Sala Nezahualcoyotl.

The types of music performed there would require a system with an extremely low noise level, and it seemed to me that the key would be to select top-flight equipment and combine that with a flexible interconnect system and a good solid electrical and signal ground. The clients had requested digital recording capabilities, even though the Mexican radio stations mostly used half-track reel-to-reel playback equipment. Some have Dolby A noise reduction, but many do not. A few stations are cassette and turntable playback only.

FIRST SITE SURVEY

In January of 1989, we flew to Mexico City for a site survey. Upon arriving in Mexico City, we were met by a young woman from UNAM who assisted us through Customs and drove us directly to the Sala Nezahualcoyotl, the concert hall on the UNAM campus. We arrived just in time to hear the last half of the Stravinsky *Le Sacre du Printemps*, better known as *The Rite of Spring*, being performed by the orchestra with Lukas Foss conducting.

The Hall was crowded, and standing in the back, I had a good sense of the acoustics of the place. It is a lovely fan-shaped hall with mostly wood side-walls, seating rising rather steeply from the foot of the stage, parabolic dish reflectors hanging over the orchestra and as mentioned earlier, a large balcony and seating behind the orchestra. The reverberation time seemed to be around the two-second range, and high-frequency attenuation in the hall appeared to be average even though we were at high altitude and very low humidity. (Mexico City is situated at an altitude of 8500 feet.) It is about 65 feet from the stage to the overhead reflectors and an additional 20 feet above that to the ceiling. Above the ceiling are some very narrow catwalks suitable for ex-

Alan P. Kefauver is the Director of Recording Arts and Sciences at the Peabody Institute of Music in Baltimore, MD.

EAW INSTALLER PROFILE

Max Tech's Jim Pici
Club Design & Installation

Williamsville, NY - Jim Pici, owner of Max Tech here, purchased his first EAW products about one year ago. He says his phone has not stopped ringing since. "It's a flawless product line," enthused Pici. Once I came across EAW, I stopped having speaker problems. I've had zero defects. I don't have to carry spare units on the service truck like I used to. In the past, I've tried just about everything else... you name it. EAW systems are simple, easy to install, and they offer high output. They sound smooth . . . no harshness. EAW products have changed the way I do business. The delivery and service aspects of dealing with the product line have been very impressive."

Max Tech has been doing sound and light installations for nearly two decades, and has extensive expertise in designing and building systems for entertainment venues. A variety of nightclubs and skating rinks in the northeast have been equipped with Max Tech / EAW systems. In fact, 18 systems based on EAW speaker systems have been installed in the past nine months.

One of the firm's recent jobs is Sharky's, a 3-bar 7,500 ft^3 complex. The club's 25'x16' high-tech dance floor system features 4 SB52B subs and 10 DS123 high definition systems. Additionally, (8) MS30 monitors were installed to cover the bar areas. "We chose EAW for its outstanding sonic accuracy, wide coverage, compact size and impressive hi-fi like sound," advised Pici.

"The MS30s enabled us to put program material in more social, non-dance areas of the club, yet still have bass output and high power handling. I really enjoy the fact that the units are compact, because today's club owners like things hidden. You have to place the components in a very sleek, high-tech visual environment. EAW gives us the tools we need, systems that install easily, look good, sound powerful and are both reliable and affordable. I can't ask for more!"

A recent roller-skating rink installation in the firm's home base of Williamsville features two DS123s mounted near the ceiling in each corner of the rink, with SB528 subs recessed in the walls. "The expectation of the American public for quality sound has never been higher," noted Jim Pici. "I've been doing sound installations for many years ... and have probably seen every type of sound system and technology that there is to work with. EAW products are the best thing we've ever had to offer."

Pici designs a unistrut-type frame to suspend multiple cabinets. An experienced labor force with sound, lighting, electrical and rigging skills put the company's design concepts into action. Typically, up to five technicians work non-stop onsite to get systems up and running. "The thing that has been taking the least amount of time is tuning the sound system," Pici advised. "The speaker systems are so inherently smooth that we don't have to do a lot of equalizing, even in difficult rooms."

Pici appreciates the high quality crossover networks in EAW's full range systems. "The company has gone to great efforts to design and construct a complex passive network featuring asymmetrical slopes, exceptional heat dissipation, and good power handling capabilities. It lets us build high powered systems without using complex 4-way electronic crossovers and separate types of amplifiers. People who feel that passive systems are outdated really need to hear these new EAW systems."

Max Tech's sound and lighting system installations are attracting attention in the nightclub industry. The firm's reputation is spreading internationally. Jobs have recently been completed on Long Island, in Westchester, and a new job will go in soon in Florida. Inquiries are coming in from as far away as Jamaica, the Orient, and England. "Our philosophy is very simple," notes Pici, "Offer the club owners the best sound we can find for them at the most affordable price, with fail-safe reliability and great attention to appearance and detail." Max Tech can be reached at (716) 633 - 2289.

Max-Tech's Installations In The Past Year Include:

Club	Location	Size (ft^3)	System
Bull Dog Lil's	Buffalo, NY	3,000	SB528, DS123HI, MS30
Cafe Rumors	Buffalo, NY	6,000	SB528, DS123HI
Cafe Sports	Buffalo, NY	6,000	MS30, DS102HI
Cathode Ray	Buffalo, NY	2,500	DS153
Club Heat	Buffalo, NY	4,000	SB528, DS123HI
Crash Club	Buffalo, NY	2,000	SB528, DS123HI
DJ Spinner	W. Seneca, NY	14,000	SB528, DS123HI
Jam Club	Buffalo, NY	6,000	SB528, DS123HI
LaBoom Night Club	Buffalo, NY	7,000	SB528, DS123HI
Late Show	Niagara Falls, NY	14,000	SB528, DS123HI
Montauk Yacht Club	Montauk, NY	3,000	SB528, DS123HI
Mulligan's	Buffalo, NY	3,000	SB528, DS123HI
Razzberry's	Olean, NY	5,000	SB528, DS123HI
Sharky's	Amherst, NY	5,000	SB528, DS123HI, FR102
Stones	New Rochelle, NY	3,000	SB520, DS123HI
Village Inn	Bradford, Ontario	8,000	SB528, DS123HI
Wise Guy's	Buffalo, NY	7,500	SB528, DS123HI

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tremely agile recording engineers or acrobats. *Figure 1* shows a picture of the empty hall prior to microphone placement.

We determined that the best approach for recording the orchestra in this hall would be to use a central pick-up microphone array or stereo microphone, with flanking microphones on the same lateral plane and spaced halfway toward the rear violins and cellos respectively. Provisions also needed to be made regarding soloists' microphones, chorus microphones and possible woodwind microphones. Since the front array would be needed whenever the orchestra was in residence, it was decided to hang the front line microphones semi-permanently from the catwalks and use movable stands for auxiliary pickups.

To this end, three quad boxes with XLR connectors would be permanently mounted above the ceiling next to the forward catwalk that ran roughly parallel to the front edge of the stage. Microphone cable cleats would be mounted on the catwalk itself at the projected microphone drop position. This would allow vertical adjustments to be made from above. It was envisioned the microphones would then be guyed, with clear fish line, to the rear wall of the stage, thus allowing for lateral and in/out adjustments. Six additional quad boxes with microphone connectors were to be flush mounted at various points around the performance/stage area to handle auxiliary microphone requirements.

PROBLEM 1

The first major problem we encountered was a lack of space suitable for a control-room site. After much discussion, it was determined that, by commandeering an existing secretarial space and combining that with new construction, a room 18 feet by 20 feet and with 12-foot ceilings could be built without totally blocking the corridor at the stage-left side of the hall. Since there would be no sight lines to the concert hall, a surveillance-grade video system with a wide-angle lens was specified. With the camera mounted on the stage wing, the engineer in the control room would have a clear view of the conductor and soloists.

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

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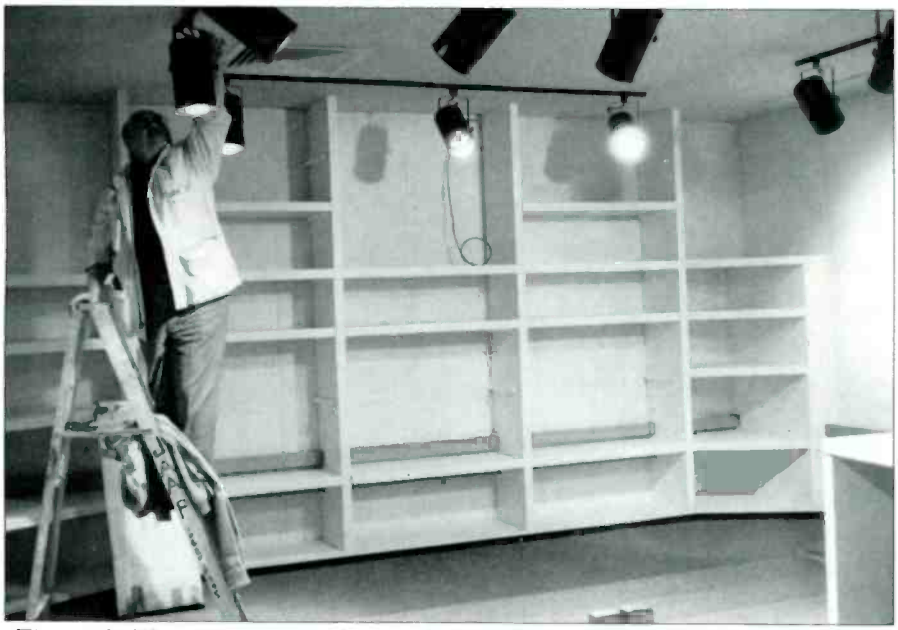

Figure 2. The empty control room before the equipment was moved. Harold Boxer adjusting the lights.

IS THERE A GROUND IN THE HOUSE?

That afternoon, we met with the school's architects, electricians and building planners. It was one of the most interesting meetings I have been to in a long time. Their English was about equal to my Spanish—nearly nonexistent. However, we spent many hours discussing floor, ceiling and wall construction by drawing rudimentary diagrams combined with sign language and pidgin Spanish/English. For instance, I was making little headway discussing grounded power receptacles with the electrician until I went over to the secretary's lamp, unplugged it and showed them the two-conductor A.C. plug and indicated with three fingers that there should be one more blade on the plug. After a collective "Aaaah," the meeting progressed. I did find out that a good ground was a difficult thing to come by in Mexico City. About the best we could hope for would be a resistance of five or so ohms to ground. It seemed that it would be best to float the whole system from the house ground and to only use the power company's electrical ground for the whole system to avoid differing ground potentials. This would also be taken into consideration when the equipment for the job was specified.

We left Mexico a day later with a better understanding of the problems involved in the project, and with the promise that the architects' and builders' plans would be sent to us for comments and modifications before work started. At this point, my partner and I decided that given the complexity of the logistics, the grounding and power problems and so on, it would be best for us to design a modular system. That is to say, a system that could be purchased and completely assembled and tested in the United States, then broken down and packed for shipment to Mexico. This would include wiring harnesses, patchbays and microphone cables as well as the equipment itself.

I felt that if I could set up the entire system and have it work to my satisfaction here, the potential for problems in Mexico would be greatly diminished. Since we had been told how difficult it would be to get any kind of miscellaneous audio hardware in Mexico, it was incumbent on us to make sure everything we needed was sent to Mexico prior to the actual installation. However, even the best laid plans can go awry.

THE EQUIPMENT

Selecting the equipment was pretty straightforward. We had budget limitations and needed equipment that would provide high-quality audio for an extended period with a minimum of repair and maintenance problems. Since I have purchased and worked with a lot of equipment for many years, and since requirements for ruggedness and RF protection were important, certain manufacturers came immediately to mind. The equipment selected was as follows:

- Sony MXP2000 16 input-console
- 2 Sony APR5002 Tape Recorders
- 1 Otari MTR5050II Tape Recorder
- 1 Sony PCM2500 DAT Recorder
- 2 Nakamichi MR2 Cassette Recorders
- 6 Channels of Dolby SR/A Noise Reduction
- 2 Biamped JBL 4430 Monitors
- 2 Tannoy NFM-8 near field Monitors
- Bryston Power Amplifiers
- UREI ⅓-octave graphic equalizers
- 2 UREI 1178 Stereo Peak Limiters
- A Dynamics module for the MXP2000
- AKG ADR68K Digital Reverberation System
- 2 AKG 422 Stereo Microphones
- 6 AKG 414ULS Microphones
- 4 Neumann KM84i Microphones
- 3 Sennheiser MKH20 Microphones
- 2 5 foot Equipment Racks
- Custom 192 point Patchbay
- Plenty of Canare Star-Quad cable
- Miscellaneous Hardware, connectors, stands, patch cords, etc.

Sixteen microphones would probably be the most ever used even with a work as large as the *Berlioz Requiem*. The Otari was provided as a remote recorder as well as a duplication deck since it contained microphone preamplifiers. The reverberation unit was supplied to add low-frequency reverberation time length to the existing acoustic. This could be used to warm up the sound of the natural ambience, and also, when close mic'ing soloists, it is often advisable to supplement the natural ambience to provide a better blend and to keep the soloist from appearing too far forward of the orchestra. Stereo microphones were selected since it would then be possible to remotely control the polar patterns of the left- and right-microphone capsules from the control room.

INITIAL APPROVAL

The facts and figures were submitted and approved and the monies were allocated for the initial phase of

the project, the equipment purchases and the room construction. Washington Professional Systems in Wheaton, MD was chosen as the equipment supplier, and they agreed to hook up the system in their showroom as it would be set up in Mexico.

We decided to build a raised floor in the control room, and to have wiring troughs located around the perimeter of the room with a leg of the trough extending into the center where the console would be located. The area in front of the console was to be at the original floor level for possible visitor seating. From here, the visitors could see the video monitor and hear, but would not be in the acoustical path between the engineer and the monitor loudspeakers. After determining the cable lengths required for all interconnections, a custom patchbay was ordered from Pro Co of Kalamazoo, MI. It was specified that all standard connections were half-normalled. That is to say, the console two-mix outputs were normalled to the recorders and the recorder outs were normalled to the console line-ins. This meant that all standard signal paths did not require patching, yet any deviation from that path could be easily rerouted. The patchbay tails were to be cut to length, terminated with the respective connectors, and labeled. This included wiring the Hiroshi connectors for the Sony MXP2000. XLR wiring for line-level equipment was specified as pin-3 hot and microphone lines were specified pin-2 positive. This follows the so-called Westrex convention for line-level interconnects and the standard RCA/European convention for microphone sources. If things were to go as planned, the patchbay would be shipped, with its wiring, in one of the equipment racks. Then, upon arrival, the equipment could be bolted into the racks, the connections made and everything would be interfaced properly and available at the patch points.

GOOD SOUND IN WHEATON

The equipment was ordered shortly before Christmas 1988, and all had arrived at the shop in Wheaton by March 1989. At this time, all equipment was hooked up through the patchbay and system performance was checked. The only ground used was through the A.C. mains, yet the system was as quiet as could be, just as we expected. As long as there was good ground for the A.C. in Mexico, things would be fine. To this end, we called Mexico and specified separate neutrals on all A.C. receptacles with all A.C. third pins tied together and run to the A.C. mains ground. I didn't really trust, nor do I ever trust, the conduit ground. This would prevent ground loops and put the system at the same ground potential as the rest of the building. The only things not tested operationally at WPS was the biamped monitor system, and the auto-switching for the Dolby SR/A units. This would come back to haunt us later.

The crossover cards for the electronic frequency-dividing network came separately from the unit, and it was felt it would be better to ship them in their original packing.

The crossover was checked electrically, and the amplifiers and monitors were run on the test bench. The system sounded superb, and the noise floor from microphone to tape was inaudible.

Figure 3. The Sala with the Orchestra on stage and the microphones in position.

ON TO MEXICO CITY

The equipment was then unassembled, repacked and put on two pallets. These pallets were trucked to Houston where UNAM has an office. Here, the equipment cleared Customs, was flown to Mexico City and delivered to the concert hall. The equipment was secured to await our arrival for the installation after the completion of the control room.

During this time, there was a Presidential election in Mexico. The party then in power lost the election, and since the head of the University is a political appointee, budget priorities changed quickly and the recording project stalled.

Nearly a year later, we were informed the project was once again "on," and that preliminary construction on the control room had begun. My partner flew to Mexico to check on the control room and to ensure that construction was proceeding properly. Budget and availability of materials severely limited the design parameters of the control room. Because of this, a simple reflection-free room was thought to be the best approach. Obviously, all reflections could not be tamed, but they could be greatly attenuated. We had specified double-wall construction throughout with splayed walls and an "acoustic" ceiling hung on resilient clips. The walls were to be covered with absorbent material and the raised floor was to be carpeted.

CONSTRUCTION CONTINUES

After some on-site modifications recommended by Boxer, construction continued with a targeted completion date of December 1989. At this time, Boxer met a young man employed by UNAM, who would assist us with our installation and be responsible for the studio and the recordings after our departure. The young man, Marcos Deli, is a gradu-

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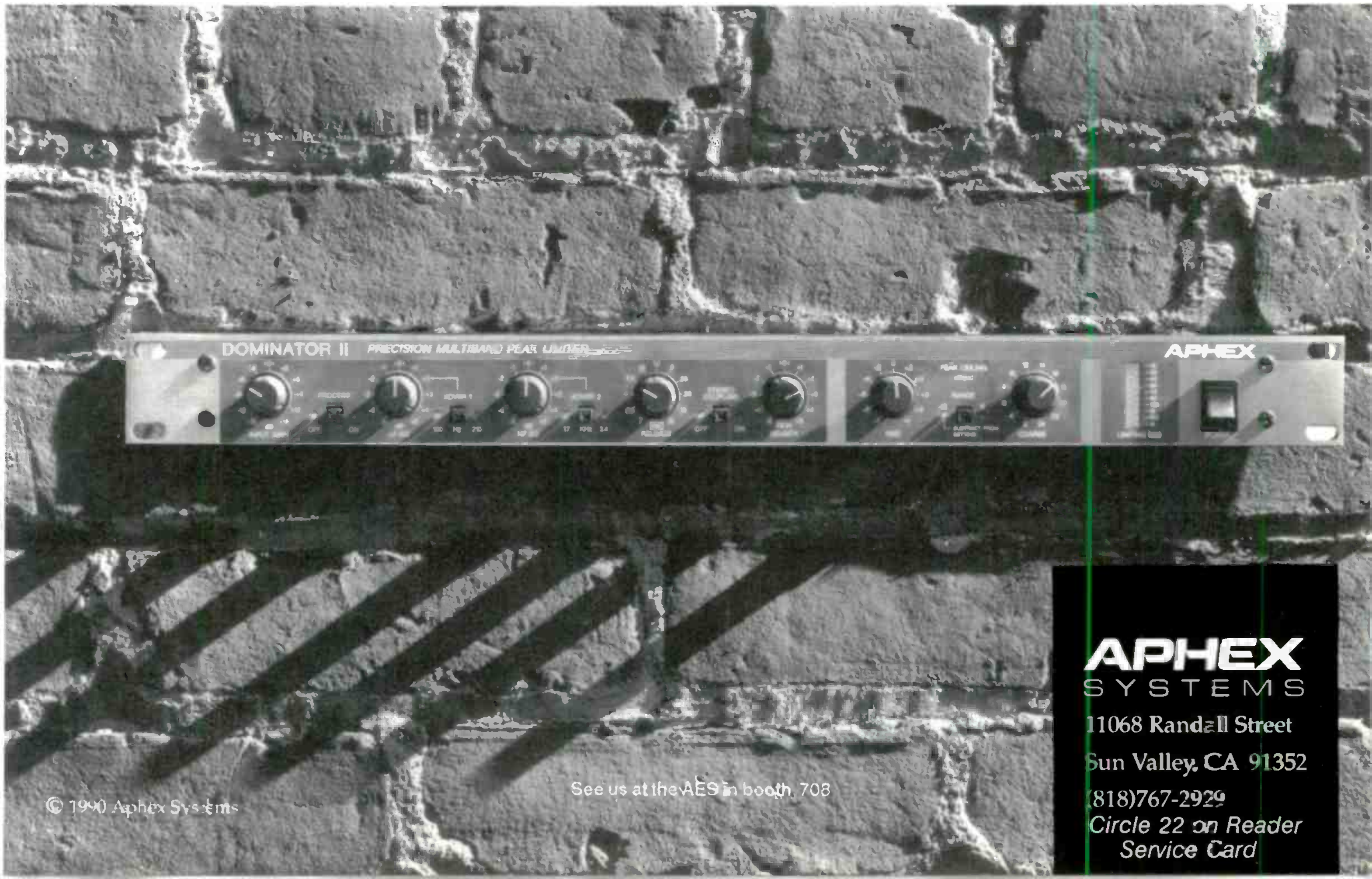


Figure 4. The completed control room with equipment installed.

ate of the Institute of Audio Recording in New York and speaks excellent English and Spanish. In December, we were told the studio would be ready for equipment installation and alignment in January.

We were also told that at this time, the contractor would be finished and that all cable would be pulled in conduit and connected to the appropriate stage boxes on the stage and in the fly.

We asked to have the two pallets of equipment moved to the control-room location.

In mid December, Boxer and I shipped a package by Federal Express containing our tools and miscellaneous hardware, along with my spectrum analyzer and digital multimeter to UNAM. We felt that shipping the materials there would be better than taking them on the plane with us both for convenience and to minimize Customs hassles. Little did we know at the time we would never see the equipment and tools again.

We flew to Mexico City on Sunday, Jan. 17th, 1990,and were once again met by a University representative. We planned to drop our luggage at the hotel and go immediately to the theater to meet Deli and start preliminary equipment setup.

Microphones are light, but Sony consoles and Audio tape recorders are not. We did find an electrician and his helper to assist us.

Upon arrival at the University, the gates were closed and no one seemed to know why, except that maybe there was a bicycle race occurring within the huge University road system. The young woman driving us, however, insisted that she knew another way in (Here it should be noted that Mexico City is the largest and most populous city in the world, and that the University itself is the size of a small city). For the next two hours, we drove to every conceivable entrance only to find them all closed. Undaunted, we parked the car and walked through the campus to the concert hall. It had already been a long day, what with a 6 a.m. Mexican flight, and by the time we got to the hall, we were tired. But, with only five days available to install and proof the system, we felt we should at least get the equipment properly placed so we could begin hookup early Monday morning.

The orchestra's last rehearsals for the week were to be on Wednesday and Thursday mornings, and it was imperative we start listening through the microphones to determine final position as soon as possible. We finally arrived at the studio door, but Deli was nowhere to be found. Our guide made some calls and determined that Deli was the only one who had a key. A very frustrating day at best. Oh well, back to the hotel, dinner and bed. We finally did reach Deli from the hotel, and he promised to be at the studio by 8 a.m. We didn't know it at the time, but Murphy's Law was to be in full force for the duration of the trip.

Monday morning at 7:45 a.m. we were in front of the hotel where we were supposed to be picked up. No one showed, so we hailed a cab. After negotiating the fare of 40,000 pesos to 20,000 pesos for the ride to the Sala Nezahualcoyotl at the University, we were off. At 8:15 a.m. we were outside the studio door waiting for Deli. He arrived at 9:30 a.m. Car

Figure 5. Alan Kefauver (right) and Marcos Deli behind the console during the Thursday morning rehearsal.

trouble we were told. After all of the delays, we wanted to get to work.

WHERE IS THE EQUIPMENT?

The door opened and, "Uh, Marcos, where is the equipment?" we said. "Oh," Deli said, "we didn't get a chance to move it yet." "Fine, let's get the key and move it to the studio," I said. The man who had the key to where the equipment was stored didn't show up until 10:30 a.m. When we located the key, got into where the equipment was stored (two flights down on the other side of the stage near the loading dock) and started to move it, we discovered the elevator was broken and there were no stage hands to help move the equipment. "Gee Whiz (in Spanish), it worked fine yesterday," we were told. Microphones are light, but Sony consoles and Audio tape recorders are not. We did find an electrician and his helper to assist us. By the time we got the equipment to the studio, unpacked it and placed it in the room, it was 6 p.m., the building was closing and Boxer and I were beat. We also learned during the day that the equipment we had shipped had not arrived yet, but that it was at the airport and would be picked up first thing in the morning and brought to the studio. Hotel, dinner, bed.

Tuesday morning: no ride, same cab driver. "To La Sala? 40,000 pesos!" he said. "Yesterday it was 25,000!" I said. "That was yesterday!" the cab driver said. Ten minutes of bargaining later, we were on our way for 25,000 pesos. After arriving at the hall, we found out why the two electricians who had helped us move the equipment had been there. They were still pulling microphone cable in conduit and no microphone boxes or connectors had yet been installed.

I must say, however, that the construction people did a very nice job on the control room. The room sounded pretty good. The A.C. was correctly installed with isolated ground receptacles and separate neutrals, the wiring troughs were in the right places and the equipment tables were strongly constructed.

The only problem was the door. It was hollow core, and with no real sound seal. We were told this would be fixed. *Figure 2* shows the control room basically empty, but after some preliminary equipment was placed.

WHITHER THE TEST GEAR

In the meantime, our package of test equipment was not at the airport. It seems Federal Express does not have landing rights in Mexico City but has to land in neighboring Toluca, an hour-and-a-half away. The equipment had not been sent to the Federal Express Office in Mexico City because no one in Toluca could translate the invoice, and therefore, the package could not clear Customs. We were also told there is a 10 percent usage tax for any materials not purchased in Mexico, even if they are going to be returned to the United States. The package was valued at $3,000, so we had to come up with some additional money for the tax. Tuesday was spent wiring equipment. Hotel, dinner, bed.

We had a copy of the invoice translated into Spanish and on Wednesday morning, while I was working on the equipment installation, Boxer and Deli drove to Toluca with the money to retrieve the package. At the hotel: no van, same cab, same conversation, same result. I had said everything but the biamped crossover had been checked out in Washington, but one of the crossover cards, naturally the one that had not been opened in D.C., was a dummy. No resistors, capacitors, or other such parts on the card. There was a chart showing the reactive component values required for the various frequencies, but that doesn't help much in Mexico.

By Wednesday evening, using Deli's tools, most of the wiring, including the microphone lines and drops, were completed. However, Boxer had not fared as well. On his arrival at Customs, he was told that the paper work needed to release the package would take several days to complete, and that the tax had gone up to $750. One of the big sticking points was an item labeled miscellaneous screwdrivers. It seems that every item has to be listed separately, with a price, in order for Mexican Customs to figure the appropriate tax.

At this point, it seemed that since we would not get the equipment in time to equalize the control room before the Thursday (last chance) rehearsal, it would be best to return the equipment to the United States. However, Mexican Customs would not release the equipment back into the United States. They wouldn't even tell us why!

After we returned to the States, the entire problem would have to be turned over to the Federal Express representatives for resolution. Federal Express in Mexico was no help whatsoever, and nothing could or would be done by the Mexican authorities either at Customs or at the University. Thank goodness we insured the package for its replacement value.

Also on Wednesday, workmen came to install a new sound seal for the studio door. This included new molding with rubber gaskets on both sides of the door. To make the door fit, they cut a semi-circle in the inside molding for the door knob. This was accomplished with great sawing, pounding and extended conversation in Spanish. This procedure helped cut down on the sound leakage, but the hollow core door still acted as a diaphragmatic resonator. I still recommended that this door be replaced. A side note to this is that the door shut so tightly at first that it could not open from the inside. You could not get a grip on the doorknob because the cut out semi-circle just barely cleared the knob. I was locked in the room for about two-and-a-half hours! No phone, and no one around outside. I had plenty of work to do, though.

THURSDAY MORNING

No van, different cab driver, same conversation, same result, but by a more scenic route (not my choice) which took an additional half hour. Anyway, we got the microphones hung properly and the system powered up. Clean, quiet, no hums or buzzes. Perhaps Murphy had decided to take a Mexican holiday.

The Orchestra started rehearsing at 8:30 a.m., and by 11 a.m., we had a good orchestral sound. *Figure 3* shows the hall with the orchestra, and if you look closely, you can see the hanging microphones. Even though we weren't biamped yet, the sound in the control room was good. The conductor was extremely pleased with the quality and amount of reverberation, the balance and the general "feel" of the recordings. The remainder of Thursday was spent cleaning up some odds and ends, closing the troughs, dressing cable and so on. *Figures 4* and *5* show the finished control room. Deli said he could take care of equalizing the room if I would loan him my spectrum analyzer. I said that would be okay if we got it released from Customs. Well, back to the hotel, a large margarita, dinner and bed.

Friday was a free day for me, while Boxer finished up the financial arrangements with the University. I went sightseeing to the Pyramids of Teotihuacan. As we passed Toluca, I thought fondly of my equipment. Juan, my driver, then got lost. I wondered whether or not he was related to the Toluca Customs people or the young woman who picked us up at the airport.

At this writing, the story is still incomplete. When Boxer went to receive the check for the final installment which was due on completion of the work, he was told that since the room had not been equalized, and that since one crossover card was missing, the job wasn't finished. Therefore, payment could not be made. No amount of discussion changed this. Actually, our contract specifically stated we were not responsible for construction errors or the equalization of the room. I had brought my analyzer along, or so I thought, to equalize the room as a courtesy to UNAM. (Ed. note: see the Postcript below.)

We found out later that since the project had been instigated by the previous orchestra conductor, and contracted by the previous administration, the current head of the music department (also politically appointed) had lost great face. He was trying to recover his political standing by doing whatever he could to snafu the former administrators' project. I still wonder whether the Customs problem had anything to do with this. Currently, lawyers are involved, and Federal Express says the package of equipment will be returned sometime.

All in all, it was an interesting and engaging project that was complicated beyond reason by bureaucracy. If asked, I would probably take on a similar project again because good music and musicians know no international boundaries and the bureaucratic problems pale when confronted by Stravinsky and his peers.

However, I think I would stay away from Toluca. db

POSTSCRIPT

Shortly after this story was written, Federal Express sent a special agent to Mexico to retrieve the test equipment from limboland. They would not say what transpired, but the equipment returned with the agent to the U.S. and it was duly delivered to Harold in Washington.

And, a week later and with the help of a Mexican attorney, final payments were received after the crossover parts discrepancies were corrected.

The Plant—Post-Production for the 90s

It's a complicated world, full of choices and options. Diversification is the key to success in many businesses these days. Offer a full range of services or products to your prospective clients that span the entire spectrum of their needs, and they'll be back. The world of music is no exception. Studio owners with vision are jumping into the swiftly moving currents of sound and music post-production for film, video and advertising. These services are being offered by studio owners who must continue to update and maintain state-of-the-art facilities for their present client base, recording artists.

Some studio owners have opted for producing music only for visual media, while abandoning their client base (recording artists) and losing everything. Other studio owners may watch the world go by, cowering at the thought of expansion and diversification. The initial costs of such a change can be exorbitant. Success at this venture requires a supreme act of balance, grace and wits. One of the best examples of this formula at work is at The Plant Studios in Sausalito, CA.

Brad Leigh Benjamin is a freelance writer based on the West Coast.

Figure 1. The Plant: Arne Frager, standing; Bob Skye, seated.

ACQUIRING THE PLANT

In 1986, Bob Skye, founder of Skyelabs, Inc., purchased The Plant Studios. Three years prior, he converted a General Motors diesel coach into a dual 24-track recording facility on wheels. The bus was designed to live end-dead end specifications, and to this day, is the only LEDE-certified mobile facility in existence. Affectionately called "Rover," this set of reels on wheels, artfully utilized by Skye to record live performances at arenas and major venues all over the East Coast, eventually brought him to Marin County, CA where his own wheels began to turn at The Plant. In August of that year, Skye bought The Plant.

Over the next two years, he upgraded the equipment while retaining the sense of tradition that came to be associated with The Plant. Studer tape machines were installed. Video lock-up soon followed, and The Plant was again ready to be recognized as a strong force in the recording industry. However, with the increasing popularity of MIDI home pre-production, major studios were booking less time and losing revenue. Skye knew a piece was still missing to his puzzle. During September, 1988, he found that piece in Arne Frager, present co-owner, digital audio pioneer and marketing mastermind.

Frager, former owner of Spectrum Studios, a.k.a. Hollywood Central, had been freelancing since 1986, engineering album projects for Robert Palmer, Prince and Paul McCartney. Together, concluding the futility in competing for a share of the MIDI pre-production market, Skye and Frager set their sights on far greater horizons. Their new objective was to provide technical proficiency and excellent service in music and audio post-production. A joint venture agreement was developed between themselves and Synclavier Artist Greg Shaw of San Francisco, whose previous production credits included projects for ABC Sports, HBO and Audi.

Renovations on Studio C started. Soon after, Studi 01 was born, a Synclavier room, with full lock-to-picture capabilities, including tapeless direct-to-disk digital multi-track recording.

Figure 2. Studi 01 control. Effects racks, Synclavier, and console—face into the studio.

INDEPENDENT SOUND/WEST

At about this time, Peter and Mary Buffett, owners of Independent Sound, a Bay Area music production house, decided to move to the Midwest. Independent Sound was sold to new principles Frager, Shaw and Skye, and Independent Sound/West was born. Its new home became Studi 01 at The Plant.

Although Peter Buffett's compositional skills are still available via Independent Sound, Frager and Shaw make up the core of this new Bay Area production team. In collaboration with other composers, Frager and Shaw are intent on establishing Independent Sound/West as a highly professional and well-recognized name in the West coast ad music and post-production scene.

"We're continually establishing and building on relationships with ad agencies and filmmakers so that Independent Sound/West can continue to expand and grow both regionally and nationally," said Yvonne Champion, Independent Sound/West's marketing representative and client liaison. "If you notice, some music houses have a particular *sound*, one trademark style of music production. Our diverse composer base allows us to do it all—rock, techno-pop, ethnic, thematic, whatever suits the clients' needs."

Independent Sound/West have recently completed a 30-second television spot for McCann-Erickson and The California Milk Advisory Board. It aired April 1990, and was produced by Frager and Shaw.

Regarding his working relationship with Independent Sound/West, Scott Deal, McCann-Erickson producer, said, "When Peter and Mary Buffett left town, I was nervous about whether I could find someone with whom I could work as effectively. I was delighted to find that Independent Sound/West was able to offer me creative input and excellent production, effectively putting my fears to rest," Deal said. "I absolutely will be working with them again in the future, and another thing," he adds after a moment's consideration, "Rose Greenway, the studio manager, makes incredible banana bread."

While Studi 01 and Independent Sound/West are significant examples of the strides being made in the music post-production area at The Plant, they are singular aspects, individual brush strokes in a much larger portrait.

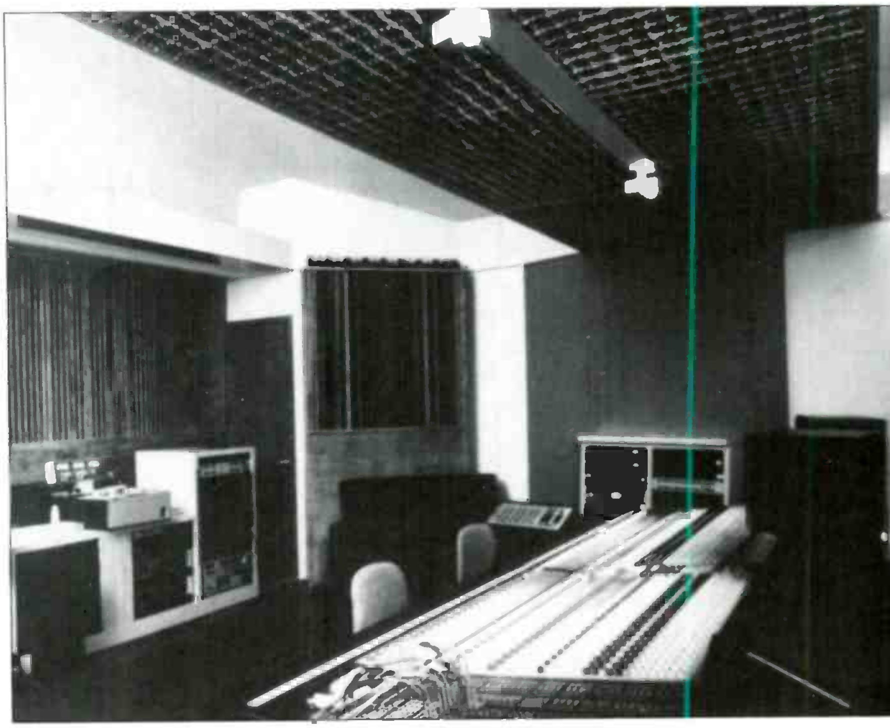


Figure 3. Boomtown control.

BOOMTOWN STUDIOS

Late 1988 also marked a significant crossroad in the careers of Mark Keller and Jeff Cohen, principles in their jingle company, and steady clients at The Plant. Having had great success producing ad spots for Levi Strauss, California Raisins and other clients, Keller and Cohen considered building their own facility in lieu of the costs incurred purchasing hourly studio time at The Plant. Taking the initiative, Skye and Frager suggested Bob design and construct from the ground up, a room specifically aimed at Keller's and Cohen's needs.

Figure 4. A view into Studio B.

They found it expedient to have The Plant construct a room built in accordance with their design parameters, and leasable on a steady basis. Soon after construction began, Boomtown Studios, home to Mark & Jeff's Jingle Company, was born.

One significant advantage to building Boomtown was its relatively small spatial requirements. Keller and Cohen did not require a lot of space for live studio recording. Studios A and B, the rooms available prior to Boomtown's construction, were designed primarily for live recording, commanding higher fees due in part to their greater available square footage. Keller and Cohen did not wish to pay for space they would not use. They relied largely on drum machines and synthesizers situated in their control room, and required only a small isolation/vocal area. It would not have been cost-effective for Keller and Cohen to lock out A or B at the going rate, nor for The Plant to dedicate either Studios A or B to them. The construction of Boomtown, therefore, was the ideal solution and most efficient use of space at The Plant.

The design and layout of Boomtown, a free-floating room within a room, is based on a unique and functional concept. Imagine a long conference table similar to those found in a large corporate boardroom. At one end of the table is the engineer sitting at a mixing console.

At the other end of the table, facing the engineer, is a keyboard player or MIDI workstation programmer with a controller on the table and sound modules at arms' length. Between the musician and the engineer, along the sides of the table, are chairs for producers, ad agency personnel or whomever might be at that particular session. This configuration not only creates a very immediate, face-to-face rapport between the musician, engineer and producers (free of encumbering talkback systems), but also allows other creative participants to confer and interject ideas from their positions around the table. There is ample room behind the chairs for people to walk completely around the table and stretch their legs without getting underfoot, truly an innovation in control room design. The circular and free-flowing nature of the room creates a constant sense of movement, with any potentially stale vibrations being swept away in a creative jet stream.

WORKING IN THE ROOM

"The layout of the room is wonderful for the way we work, and the space is such that you can move around the room easily," Keller said. "Quite often, we have a lot of people at our sessions, but the traffic flow is really manageable. The agency producers show up, and sometimes bring their clients, which can result in large numbers of people. The layout of Boomtown gives people a way to move so you don't feel like you're closed in as in a traditional control room," Keller said. "Aesthetically, it's nice, too," he said, glancing around the room and sizing up the space. "The ceiling goes up. We wanted a higher ceiling just to give ourselves the feeling of air in the room."

Many advertising VIPs and their clients relax here, monitoring the control room activity via headphones

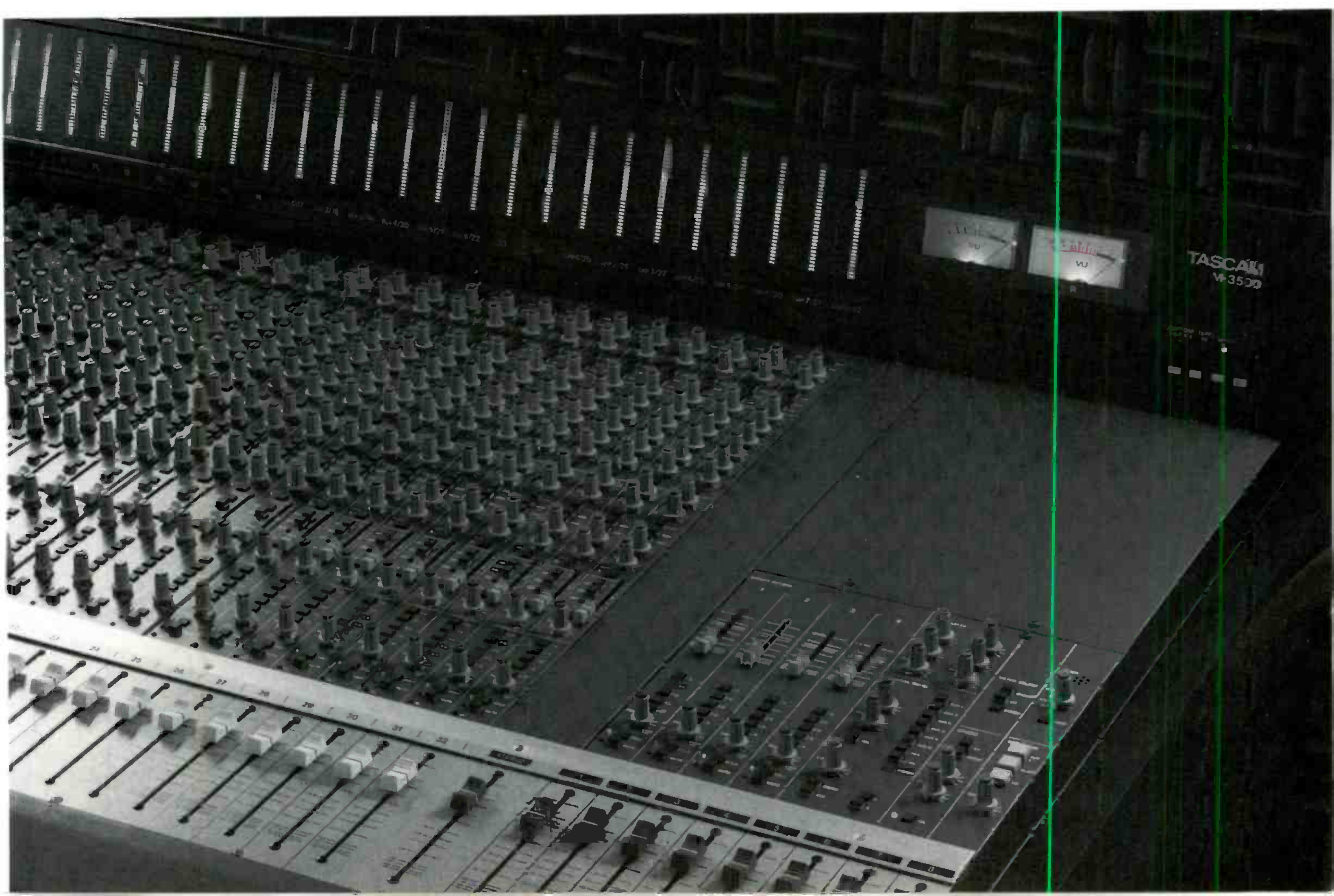
In response to Boomtown's acoustical value, Keller said, "the listening is easy from all around the room. Boomtown is on the *live* side, which is nice for us because you can talk to someone all the way across the room. Things aren't muffled in here. It's a nice, natural sounding room."

Boomtown's *liveness* is not only desirable from the perspective of ease-in-communications between Keller, Cohen and associates, but presents several unique acoustical advantages in regards to accurate monitoring of their program material at the mix position.

Skye was able to develop an accurate mix position with excellent stereo imaging by constructing purposefully-built acoustic panels and wall shapes over the console and around the mix position, designed to address and balance Boomtown's unique acoustic signature.

"Bob Skye designed the acoustic bridge which goes over the console and engineer's listening area," Keller said. "That breaks up the sound a lot, and along with the RPG diffusers, breaks up any unwanted reflections."

Adjacent to the control room is an area doubling as a live-mic'ing room for vocals, acoustic guitars and a kitchenette/lounge. Many advertising VIPs and their clients relax here, monitoring the control room activity via headphones. Although the room is available to other clients of The Plant, Keller and Cohen are its most frequent residents.



RECORDING MUSIC

While mix-to-pix and post-production are of great interest to Skye and Frager, recording artists and their producers command a very high priority at The Plant. "This year has been really a big year to redesign, reevaluate and readdress the markets we're going after," Skye said. "Both my partner and I are musicians,"

Skye added, "We're both very pro music and want to continue working with recording artists. After all, music is what built this place (and) record albums.

"Going exclusively into music production for commercials, industrials, or all video is just not in the cards, but to offer those services, and include those aspects of music production within our clientele, is really the key," he said.

Frager added, "What was important to us was to go after film and TV work, but to maintain our relationship with our rock 'n' roll, blues, jazz, R & B and country people, because they are the heart and soul of the industry," Frager said. "To go after new markets and lose the already established markets is not gaining anything. If you can build into new markets, the way we built the media-oriented Synclavier Room (Studi 01) and Boomtown, and have those rooms situated in such a way where their potentially conservative clientele are not infringed on by the sometimes playful and outrageous vibes of the rock 'n' roll or Record People, then you have effectively taken care of your clients and added to your business," he said.

> **Studio A is the best-sounding monitoring room, because it is the first room that we built totally for mixing and mastering.**

STUDIOS A AND B

While Boomtown and Studi 01 service the bulk of The Plant's commercial/post-production/mix-to-pix clients, Studios A and B are dedicated to the production of recording artists. The rooms are distinctly different, each one possessing its own acoustic signature and characteristics. Between the two rooms, a wide range of technology is available to recording artists and producers working at The Plant.

Studio A may well be the premier SSL room in the Bay Area. It is equipped with a 56-input SSL 4064G Series Console with Total Recall. Otari MTR 100 and MTR 90 24-track recorders (with optional Dolby SR) are available. A Sony 3348 48-track Digital recorder is also available upon request. The new 400 square foot control room, designed by Skye and Lakeside Associates' Carl Yanchar, features a custom two-way monitor system utilizing TAD Drivers. The studio is 1200 square feet and has an isolated vocal booth dubbed *Iso Alley*, which has been host to artists such as Huey Lewis and The News, who did tracking overdubs for their multi-platinum album, *Sports*.

The sound quality in Studio A's control room is excellent. "Studio A is the best-sounding monitoring room, because it is the first room that we built totally for mixing and mastering," Frager said. "Obviously, sound quality and precise stereo imaging were important."

Studio A's reputation is growing so quickly that singer Michael Bolton is working there with producer Walter Afanasieff on his next album. Afanasieff, longtime keyboard specialist utilized by producer Narada Michael Walden, and frequent president of Studio A, says, "When I'm sitting in Studio A listening to a track that I did, I can hear every single instrument, every single sound that's coming out of the speakers with perfect clarity. In many of the rooms I've worked in, the sound of the mains doesn't accurately translate what we've got up into the nearfields. In Studio A, however, the big speakers always give me a crystal-clear representation of the mix with added power and presence" he said. Reflecting a moment he adds, "Studio A kicks a serious butt."

The Studio A control room boasts ample square footage, allowing for a MIDI keyboard/workstation above and behind the engineer's actual mix position. The workstation itself is an extremely accurate listening area. "We designed the room this way so that if the producer is also programming keyboards, he or she can get an accurate representation of the mix right from that keyboard/workstation position," Frager said. "It eliminates the need for the producer/programmer to continually get up from the workstation, walk around and ask the engineer to move aside so they can hear what's really going on," he said.

"Studio B is still known as the classic," said Skye. "It remains unchanged from the original design built by Gary Kellgren and Tom Hidley. The original Trident TSM console is still intact," he said. "There have been repairs. Gear has been upgraded where technology permits, but for all intents and purposes, this room is exactly as it was when built 15 years ago. Nevertheless, it is now replaced by an SSL.

Due to the contrast in appearance between Studios A and B, some musicians may be skeptical about recording in Studio B, based on its non-high-tech appearance. "I used to get a kick out of when Jim Gaines, one of our leading producers, would have a new group come in, and invariably, the drummer or drum tech would be setting up the kit, take a look around, and based solely on visual perception, grumble about not being able to get a big drum sound in a room too dead and too small," Skye said.

"Of course, Jim's expertise, combined with the room and the way he likes to work, produced great results, giant, hugely ambient drum tones. It's not a magical thing. It's experience and it's having the right space and tools to work with," he added.

EYE BEFORE EAR JUDGEMENTS

"What I find from a lot of people who don't have real engineering experience is that they tend to make *eye* judgements before they make *ear* judgements, and I think in this business, that's a little dangerous," Skye said. "Show me a CD or a piece of vinyl. Put it in front of my face. Do you really expect me to make a judgement about the music based on what the physical medium looks like, or what the packaging consists of?

No. The bottom line is listening. How does it sound? The product, the results. That's the bottom line," he said.

One encounters an extraordinary blend of new and old technologies at The Plant. For example, an SSL console is side-by-side with a Pultec tube equalizer. Weathering a climate in which everyone is clamoring for the newest and latest in digital technology, it is interesting to look at The Plant's approach.

Stepping outside into a beautiful Fall morning, it is impressive how quiet, secluded and close to San Francisco Bay The Plant is.

"Over the years," said Frager, "there's been a lot of technology that's changed. Supposedly, all the new technology is the best, but there's also a lot of old technology that's the *best*, for example, tube microphones, tube equalizers and EMTs. There's a lot of things that were around 15 years ago that are considered in today's market, vintage stuff," he said. "This studio probably has one of the largest collections of well-maintained vintage microphones and signal processing gear. In these Pultec equalizers, the primary circuit components are tubes, so when you put them on line, they have a tendency to be warmer than the digital EQ's. Each of our control rooms has several of them," Frager said.

Elaborating on his recent experience with a particular group of recording artists and their preferences in technology, Frager added, "We just recorded this group called *Mother Love Bone* from Polygram. They worked in both rooms (Studios A & B) and they actually preferred doing their tracking in Studio B because of the ambience of the room. They liked the sound of the Trident board, and they liked using the tube equalizers and the tube microphones. I think Carlos Santana is another example of a guy who, when he cuts his tracks, wants to cut his tracks in Studio B. Studio A has become highly desirable for its mixing applications, but Studio B, for many of our clients, is the desirable room for cutting tracks."

Stepping outside into a beautiful Fall morning, it is impressive how quiet, secluded and close to San Francisco Bay The Plant is. Skye refers to The Plant as "a place to call home for a lot of the artists and the anonymity to go with it, so they're not on display and not on show when they come in and work here, and that's important to us." db

BRAD LEIGH BENJAMIN

Sound Whiz Frank Serafine on The Hunt for Red October

"It's a keyboard." "It's a tank." "It's a keyboard! "It's a tank!"

It's the M-1, and it is a keyboard...to most musicians, unless of course you're contemporary composer and sound effects wizard, Frank Serafine, in which case it could be a keyboard, but it might also be a tank. Commissioned by the Pentagon in 1985 to create sounds for the Army's M1 tank operations simulator, Serafine, whose current library contains well over 30,000 sounds, was able to use many of his M1 tank sounds on his most recent feature project, Paramount Picture's *Hunt For Red October*.

"It was back in '85 when I rode around in one of those M1's. They're turbo-controlled tanks capable of speeds of up to 60 m.p.h. They can even pop wheelies," Serafine said. "Well at the time, I used an Electro-Voice RE20 mic and a Sony PCM F-1 digital recorder and got all these sounds, exterior, interior and turret rotation sounds.

"The tank had a small interior, spatially similar to the interior of the DSRV, the little rescue sub in *Hunt for Red October*. They had a similar environment and ambiance, and so many of the M1 sounds worked just perfectly for the DSRV.," he said.

"Luckily, I had all these sounds already sampled into my Emulator III and stored on a Sony 650 megabyte Pinnacle drive when Cecelia Hall and George Watters II came over to preview sounds from my library," he said.

Hall and Watters, the supervising sound editors on the film, were sufficiently impressed with the sounds enough to offer Serafine a position working on sound effects for the entire film.

Figure 1. Frank Serafine at his keyboard with racks of needed equipment behind him. Not seen, the racks of synthesizers that are in front of him.

"Cecelia and George, with whom I've worked since 1979, gave me my first film, *Star Trek The Motion Picture*," Serafine said. "I did the first and third *Star Treks* with them. They called and said they had another big picture for me. This time though, instead of going into space, we were going into water, deep water.

"First we started out trying to figure out what we were going to do for every single effect," he said. "We read the script and went through a rough sketch. Then they had a VHS, real rough temporary dub that they wanted to put sound on so that the Director, John McTiernan, could kind of get an idea of what was going on, so, basically, the movie was made first, the whole thing, and then we had a couple of weeks to come up with all the sound effects," Serafine said.

During that period, most of the concepts came together, and Serafine obtained permission from Disneyland to tape anywhere in their park. "I went over (to Disneyland) and recorded all of their turbine systems," he said. "They have these big hydraulic air-conditioning systems that power the whole park. They're huge. When the turbines turn on, it takes about three minutes for them to reach their peak, so it's like this giant wind-up, this build-up of power and pitch.

"The opposite occurs when they shut down, a very gradual decrease until the system comes to rest. The decreasing sounds were used to simulate the nuclear reactor breakdown on *Red October*," he said.

ON TO DISNEYLAND

"I went to every location at Disneyland that might have potential

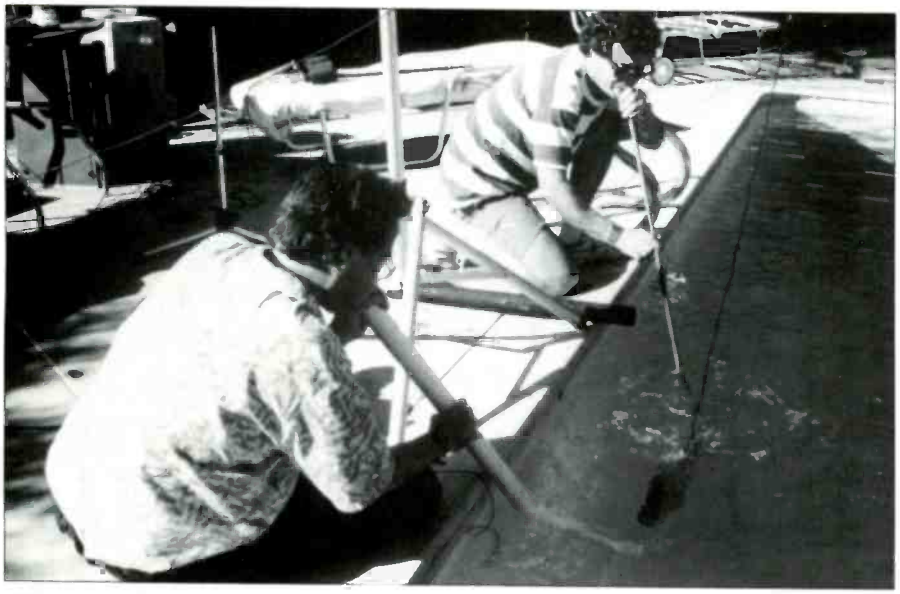


Figure 2. Serafine and assistant, Drew Neauman blowing bubbles in Cecelia Hall's pool.

sounds that we could use for submarine technology," Serafine said. "I even recorded the submarines, the ones from the ride over there that actually go underwater. I went underneath the pavilion at Star Tours, Lucasfilm's ride simulator, and recorded all the hydraulics and machinery that moves the cabin around. Mission to Mars had some interesting mechanical sounds and this is all being recorded from underneath Disneyland," he said.

Serafine previously worked for Disneyland on Space Mountain's grand opening in 1976, and on *Tron*, a Disney movie. He credits the Disney people for letting him into the park to do his work, despite the work being for a Disney competitor.

"I got lots of sound effects doing that—all kinds of hydraulics and metal latching, turbine wind-ups, and these grinding, metallic hatch sounds," he said.

"The steel structure of rides such as Space Mountain withstand thousands of pounds of stress from the centrifugal force of the cars as they pass through sharp turns. That ride yielded lots of great squeals and groans really suitable for the sound elements used to convey stress on the submarines' hulls as they dove deeper and deeper in the film," he said.

Figure 3. Left to right, John-Paul Fasal, Paramount sound designer; Cecelia Hall Paramount sound supervisor; and Frank Serafine.

INTO A SWIMMING POOL

Serafine and John-Paul Fasal, of Paramount Pictures, did quite a bit of work in Hall's swimming pool, knowing they would need many underwater ambiances, such as different water motions and torpedoes. "I brought out dozens of microphones and experimented. I took a film can, filled it with 40-weight oil, put a Crown PZM inside it, sealed it shut with epoxy to make it completely waterproof and then we threw it in the water and it floated just underneath the surface," Serafine said. "We used it to record all these different kinds of sound effects and movements that we did underwater," he said.

A Crown SASS mic was used on the surface and Schoeps mics were also used. "What we did was put the Schoeps' inside condoms," Serafine said. "We put Vaseline on the inside edge to make a good seal, and sealed them shut with rubber bands and pipe cleaners. We dropped those in the water as well. So we had Nagras going," he said.

Serafine performed *cannonball* dives, swimming movements and other similar acts with underwater air tools while a DAT machine was on. "The great thing about it is, underwater, we'd get the recording through the film can mic," he said. "For any sound, it worked just like an underwater sounding board. It was fantastic. It picked up a lot of low frequency vibrations. We also created another film can mic without any oil in it, and attached it to a sinker which pulled it down a little deeper. It was brighter sounding, picking up more of the highs. For every movement or effect done in the pool, we always recorded in two channels (stereo) with two different mics, giving us a wide variety of recorded textures to choose from for any given sound," he said. "Back at the studio we could then play with two different aspects, in sync, of any given sound. We used all sorts of mic'ing combinations. John was running the Nagra with the underwater Schoeps', and Sennheisers on the surface. I had the film can mics underwater and E-V 309's on the top, as well as the Crown SASS PZM on the surface," he said.

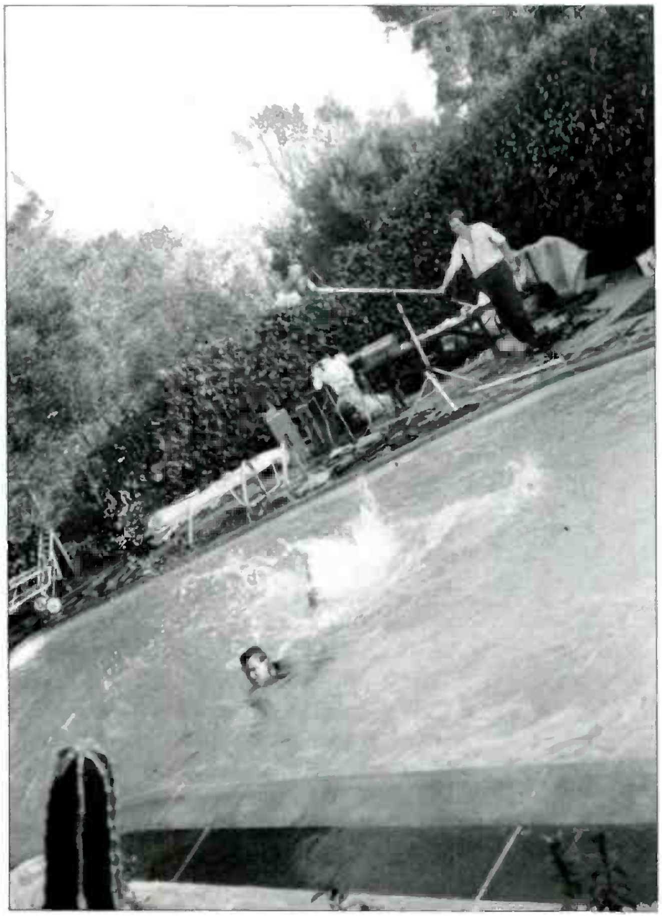


Figure 4. Frank Serafine swims around while his sound assistant Drew Neauman records his splashing sounds.

"In the studio, we were able to take the high end off of the surface-recorded material to make it sound more *underwatery*. We were also able to boost the highs on the underwater recordings and give them more presence," he explained. "We used these sounds for the sub's movement underwater, propeller wash, depth charge impacts, everything."

The prop noises themselves were a combination of several elements including an old steam paddleboat and an empty tanker recorded with its prop slightly out of the water as it entered Los Angeles harbor. Additional prop elements were taken from underwater recordings made by Alan Howarth, an independent sound contractor, of ships and tankers traveling between Long Beach and Catalina Island.

SONAR WAS DIFFICULT

The sonar sounds proved difficult to capture or create on tape. There were none available in anyone's library that were satisfactory to the sound editors or the military consultants on the film, Ron Patton and Bob Smith of Sonalysts. Patton and Smith were hired to provide a sounding board for credibility and authenticity of the film's effects. They interfaced with Hall and Watters, who in turn, gave feedback to Serafine, Fasal and Howarth in regards to authenticity.

"One of the things we were challenged with was that the subs were supposed to be kind of silent," Serafine said. "They couldn't be those loud World War II submarines. They had to be quiet, you know, so you couldn't easily detect them, so we had to use a lot of finessing and make the sounds really discreet, not overt, not real subby and bassy—more subtle, quiet but ominous, so that when they came by they really scared you. It's not like *Star Trek* where we could shake and rumble the whole theater when space ships passed over. On that project we were playing totally in the realm of fantasy. For *Red October*, we had to address a degree of realism and authenticity. It is fantasy to a degree, like any movie, so we couldn't go totally quiet on engine sounds," he said.

A staff of nuclear submarine consultants knew how the submarines should sound, and they gave Serafine and his crew feedback on all of the sounds. "Oddly enough, several of the turbines at Disney sounded exactly like the turbines they have on these submarines," Serafine said.

"But many of the sounds we made they dumped, and so we'd get these reality checks from time to time and have to make decisions. In most instances we took their advice. Other times, we went with more fantasy-oriented sounds. I mean, if you went accurate on everything, you'd have a silent movie because some things just don't make sound, so you have to kind of make things sound like they'd appear to sound," he said.

Using the Star Trek movies as an example, Serafine explained that although there is no sound in space, sounds were designed for the movie in the way people perceive them to sound, as well as for entertainment purposes. "There's no real sound for a warp drive acceleration, but we've set the standard for what that should sound like through sound effects design," he said. "Then there's lasers—lasers don't make any sounds, but in your mind you think they're supposed to make this hot, piercing electronic sound because that's what we've made you think from the way in which we've continually created these effects. So now we're playing with submarines and making you think that this is how submarines sound, while trying at the same time to provide a high degree of realism, because submarines, unlike starships and laser pistols, really do exist and really do have their own unique sound," he said.

The sonar pings had to have a high degree of realism; there was no compromise on that issue. Several versions of sonar were amplified underwater in Hall's pool and recorded at the other end 42 feet away. The natural reverb yielded the sonar sounds for the little DSRV sub, but the sonar for the large American and Russian ships took a little more work and perseverance.

A highly authentic sonar element contributed by Fasal was processed through an Eventide 949 Harmonizer along with many closely spaced delays. That signal was sampled into a Synclavier and played back along with an interval just under a fifth below the original tone. These sounds eventually served as the sonar sounds throughout the film. Howarth sequenced the *pings* to picture, increasing the rate of repetition as other ships or torpedoes approached the *Red October*. Making

Figure 5. Supervising Sound Editors Cecelia Hall and George Watters at Paramount's editing-viewing equipment.

this work dramatically required more than just speeding up the rate of the *pings*;

Howarth used a Synclavier to achieve the faster repetitions through the use of a manual arpeggiator while varying the sounds' parameters and degrees of intensity. The result is very real.

TORPEDO SOUNDS

Getting the right torpedo sound proved to be just as challenging as the elusive sonar ping. This is due in part to the fact that an actual torpedo's sound is classified information. Since the sound itself is crucial to the defense of a submarine, tapes of the sound exist only in the Navy's vaults. Patton, who had heard the real thing, guided Serafine and staff toward (and away from) the ultimate result which took four months to produce. Fasal recorded motorboats passing on Lake Castaic with two underwater mics. The sounds were very similar to a torpedo because motorboats and torpedoes both have piston-driven engines. The final torpedo "sound" consisted of these motorboat sounds layered with a Ferrari, various animal screeches and growls, bubbles coming off of a boat prop, a motorscooter and a screeching screen door spring. This resultant sound was combined with the underwater sound of water rushing from Hall's garden hose into the pool. The rushing water element was processed to simulate the explosive element of the initial, compressed-air release of the torpedo when launched from the tube. The overall sound effect is not just your ordinary everyday torpedo, but, according to Serafine, "the howling vengeance of a weapon anxious to hit its target," extremely menacing and frightening, as if it had a mind, a will and volition of its own.

The "silent" Caterpillar Drive, required to be non-overt and virtually undetectable by sonar, also presented some interesting challenges to Serafine and crew.

"I did communicate quite a bit with Basil Poledouris, the composer on the film, on the sound of the silent drive, the Caterpillar Drive," Serafine said. "He had played some sounds for the Director, John McTiernan, that the director liked, from a Korg M-1. He played the M-1 patch for me and I went off and made something within that feeling because it sounded like metal kind of driving, so I developed something out of some of the things we'd done at the pool," he said. "It was a splash from a *cannonball* dive which I took down a couple of octaves, layered with this pile driver from my library, and looped on the E-III, making it repeat like a giant caterpillar pushing along the ocean bed. It worked really well.

"For the Caterpillar failure, an oil pump, some weird tire skids and a boat banging and skidding against a dock were all combined in the final mix to create an alarming but realistic malfunction," Serafine said. "McTiernan, concerned that the subtle effect might go unnoticed, requested that an anvil sound be inserted in order to create the rhythmic pounding of metal that audiences would associate with a damaged engine," he said.

While the task of ultimately laying the sounds to film was in Hall's and Watters' hands, Serafine, Fasal and Howarth were more concerned on a day-to-day basis with the acquisition of sounds and the design of sound elements.

"I used the Emulator III with a Sony 650 megabyte read/write/erasable optical drive to do the whole film. I'd take all my DAT recordings from out in the field, bring them back to the studio, sample them all into the E-III and store them in my read/write optical," Serafine said. "The disk comes with removable cartridges which hold up to 650 megabytes apiece. I'm currently transferring my entire library over to the Emulator III, and I can access a huge database of sounds through the Sony drive. I can run both of my E-III's off the Sony. I'm constantly sampling stuff. I have over 30,000 digitally recorded sound effects now, catalogued in the Mac, using Microsoft Works," he said. "I've got them listed in categories and cross-referenced in several sub-groups."

In my new facility we'll be able to post in-house and also offer some of those services but in a way, I like going outside to post and mix...

COMMERCIAL ADS

Most of Serafine's work for commercial ads is original music with a sprinkling of effects. He does mostly sound effects for features, but he's beginning to score feature films. His diverse and well-established clients include Nintendo, Chevron, Mercedes-Benz, Chrysler, Nissan, Eveready, CBS, Fox Broadcasting, "Entertainment Tonight," Universal, Paramount and MGM.

"I do all of my tracking right here for commercials. Then we take it outside for posting," Serafine said. "I like to work a lot at Mix Magic right here in L.A. In my new facility we'll be able to post in-house and also offer some of those services but in a way, I like going outside to post and mix, you know, just be the creative guy, put the arrangements together and get them down on tape really super-clean, and then let someone else deal with the grind of mixing, mastering and preparing it for broadcast, he said.

"I really feel strongly though, about standing by my projects throughout the entire production process, especially at the critical points like mixdown and post-production.

"If I do an entire project here, there are many instances where I'll take the whole thing outside for mixdown," Serafine said. "For instance, I just finished doing a thing up at Skywalker Ranch for (Filmmaker George) Lucas. It was a simulator ride through space and I did all the music, voice and effects right here on multi-tracks. Then I brought the masters up there and mixed them on Lucas' big theater system.

"For sound effects, I master all of them down to DAT, and get them just the way I want them because sound effects you have to really fine tune," Serafine said. "For commercials, I record a sync pop to tape and then just record each individual sound element in sequence onto that same DAT recording," he said. "The elements are transferred to mag or multi-track tape, according to the format, and posted along with the other elements."

Serafine does all of his musical compositions on sequencers using Opcode *Vision* and Digidesign *Cue Sheet A/V* for effects. "I have 25 synths which I can mix directly down to DAT. After I've got my effects arranged in the computer," he said, "I edit and transfer them to multi-track tape which is brought to the dubbing stage and locked to an editing system for mixdown. On *Red October*, though, we transferred each sound individually to analog, two-track, with Dolby SR, on quarter-inch tape at 15 in./sec., and gave everything to the sound editors in stereo. They cut everything in themselves," he said.

IN THE NEW BUILDING

In Serafine's new facility, he'll be able to do all of the music and effects for films. He says that no one has ever really scored and done all of the sound effects for a feature, and that movie effects have always been traditionally done by sound effects editors who are not composers.

"For my scores, I'd like to blend sampled and synthesized sounds with live orchestral players," Serafine said. "In my new place I'll be able to bring in about ten players at a time, but truthfully, for say strings, I'm getting orchestra-quality strings out of my synthesizers and samplers. I play my string pads for people and they can't tell the difference. I'd much rather work with synths because I like the instant gratification of being able to compose and hear something right now with the picture," he said. "I might add ten real string players afterward just to make it sound super-real. I love the real orchestral stuff but I also really love exploring the technology."

Serafine approaches effects and music together with equal focus. For him, they play off of one another and should balance and compliment each other. He follows his instincts and works on those facets of an audio track which compel his inspiration at any given time, working with two different engineers, one for music, and one for effects.

The rough mixes a computerized sequence of music down to tape, and locks tape to his effects sequence, so he can preview them together and really get a sense of the blend.

COMMMUNICATION

In Serafine's opinion, it appears that sometimes, the separate individuals contributing to a feature's soundtrack, specifically, the composer and sound editors, can unintentionally step on each others' toes by not communicating their intentions, resulting in music which masks effects, and/or effects which mask music.

"I've worked with some of the greatest composers in Hollywood, doing effects for Jerry Goldsmith, Alan Silvestri, Jamie Horner and I know how effects work with music," Serafine said.

"I'd worked with Jerry Goldsmith on *Star Trek*, but I'd never met him. I'd spent weeks working on some effects, getting them just right, and then go in and Jerry's music would just knock it all down, just overwhelm it with volume. Conversely, he'd put all this energy into some music cue and I'd just clobber him with sound effects. Well, I just didn't ever want to let that happen again so when it was time to work on *Poltergeist*, I called him and requested that we meet and talk from time to time about what we were doing, and as it turned out, it was one of the most homogenized soundtracks you'll ever hear, just beautifully engineered between music and sound effects. That's because we really communicated," Serafine said.

"Most sound effects guys and composers don't communicate. There's reasons, I mean, a lot of it is economic. People want to use their time effectively and efficiently without putting too much time into any one given project. It's business, but that little bit of extra communication between Jerry and myself made all the difference."

In producing both the sound effects and the music for a feature, Serafine believes that even more cohesion can be reached in blending music and effects. In between recording projects, Serafine is spending his time preparing for the grand opening of his brand new multimedia facility in Venice, CA. Serafine Studios will offer complete musical and post-production services including symphonic, orchestral and progressive electronic music composition and production.

Jingle production will also be in full swing. Sound services will include sound effects and design, ADR, Foley, mixdown and post-production.

With the success of *Hunt for Red October*, Serafine has had several offers to work on more feature films. Unless he decides to hijack a submarine and defect to the Soviet Union, you'll probably find him relaxing in his new, exotic digs atop Serafine Studios, basking in the Southern California sun.

Evian anyone?

Editorial

A Guest Editorial by Shelley Herman

Over the past 40 years, as an employee of consumer and professional audio supply houses and manufacturers, I have expressed the thoughts that follow verbally. However, the legal departments of those firms told me that as an employee of a firm engaged in commerce, if I were to write a letter to this effect, or publish it in a magazine, that action could be construed as restraint of trade.

Now that I am an independent consultant, I can say what I want. Just one of the perks of being "self-unemployed"! What I have to say is sensible advice to both dealers and customers in the pro-audio business who would like to remain and thrive. Fast-buck artists and bean counters couldn't care less about the future of this business.

For several years, customers with aggressive purchasing departments have been conducting telephone auctions with dealers to see how little they can pay for each piece of equipment. They send their technical staff to examine a piece of equipment on the local showroom floor. Those with even more arrogance will ask a nearby dealer to send a piece of equipment over for examination, with the dealer spending his money to have it in stock and for shipping. When the customers have decided what they want, they use their 800 number to call everyone in the country and then buy from the mail-order house that has the lowest price. The mail-order house is able to maintain these lower prices because they don't have to stock or display any merchandise, they don't have to hire competent people to service customers, they don't have to maintain a showroom in the "high-rent district" and they don't have to have a service department to repair the merchandise if it becomes defective.

On the other hand, some dealers lower their price, trying to buy the loyalty of customers and state that they will "beat any deal in town," only to discover they're indeed making more sales and losing money on every sale!

In my former life as a pro-audio salesman, I discovered it was impossible to get the price low enough. I was once asked to bid on an item for a film studio. Our business with that studio had fallen off to nothing in previous months, and I was determined to find out why. The studio wanted only one, rather inexpensive but esoteric item, whose path I knew I could trace. I bid the item at cost plus five percent, just to see what would happen.

Sure enough, I found out who underbid me, just because this particular company couldn't stand to lose even one sale. At five percent over cost, the seller lost about 18 percent on the deal! Someone had to prove that they were the cheapest house in town. This is stupidity! Everyone's in business to make money. If a dealer stops making a profit, his business will cease to exist.

Each time the price drops a little more, it cheapens the product in the customers' eyes. Every manufacturer spends a lot of time, effort, energy and money to build a reputation of quality, reliability and class. He doesn't want that reputation to be torn to bits by allowing his product to become the industry football.

Stop and recall those product brands that have the best reputations, the products that have the highest resale value. Those companies have sales policies that are careful about the dealers chosen, taking care to see that there are not too many dealers in any one area, and that those that have been selected hire competent sales personnel, stock sufficient merchandise, and aren't known primarily as a lowest-prices-in-town establishment.

Many businesses that buy or sell pro-audio equipment are being operated by managers who are completely bottom-line oriented, shortsighted and perhaps only in this business as a stepping stone to another career. They don't know, or perhaps don't care, what happens to this business in the next decade, only what the profit and loss statement says about the last quarter! They will probably be in some other business as a "professional manager" by then.

Today, consumer stereo equipment is sold in discount houses by order takers (I won't malign the sales profession by calling them salesmen) who haven't the vaguest notion of how the equipment works, or even how to hook it together. It wasn't always that way. Twenty-five years ago, hi-fi dealers were small entrepreneurs, just as pro-audio dealers are today. They hired competent salesmen who knew the equipment, could explain how to install it and even go to the customers' home and hook it up. Now, the manufacturers can just sit back and allow the dealers to beat each other to death, because another dealer is right around the corner.

However, in the pro-sound industry that option is not available. The business is too small to support the mass marketing techniques of hi-fi. The manufacturers must keep their dealers in business as it takes a lot of time, effort and money to establish a new dealer. Each time a dealer goes out of business, the manufacturer is sure to suffer a financial loss. By the way, don't think that one dealer can drive everyone else out of business and be the "only game in town." That's been tried and doesn't work. Single dealerships certainly are not to the customer's advantage. If the dealer is successful and has the market all to himself, he can charge whatever price he wants. If he wants to take a month to deliver or service a piece, where else is the customer going to go?

For purchasers who conduct those low-ball auctions, imagine what will be lost if they continue to do business this way. In the short run, they may save a few percent on the purchase price of the item, but where will they go when the item needs repair? Who will they call when they need help in connection or operation? Have they got the effrontery to ask the dealer they just bypassed for half a percent to tell them how to hook it up? They probably do! The next time these purchasers call that same dealer and ask him to send over the latest toy, if he's polite, he'll decline. Most will just tell them where to go.

Pro-audio is a small industry, probably fewer than 30,000 people in the whole country, and anyone who wants to make it their lifetime career will always have to deal with the same people. These relationships are not the same as buying or selling home appliances, but are professional relationships that businesses depend upon for survival. Bridges cannot be burned in an industry this small.

It's time for us to start treating each other like ladies and gentlemen and conduct our business as a profession, not a swap meet!

JOHN BARILLA

Understanding Time Code Synchronization

• Over the past few years, the use of SMPTE time code in the small-format recording studio has greatly increased. With declining price tags on synchronization equipment, what was once strictly the domain of the major studio is now commonly found nestled in the home studio.

Since SMPTE time code is a universal standard, all kinds of inter-format production become possible. Projects started in a smaller studio (with limited tracking capabilities) can be augmented in a more sophisticated studio without additional recording or generation loss. *Off-line* video dubs of TV or even film productions can be scored in the comfortable (low cost/low pressure) environment of the modern electronic cottage. Later on, they can be mixed and laid back utilizing the full arsenal of an *on-line* facility. The possibilities open to a studio that can handle time-coded products have been greatly expanded—especially at the level of the home studio industry. So let's examine some of the basic concepts of SMPTE time code synchronization, the equipment used and the way this technology is bringing the smaller off-line studio into a new and productive relationship with the larger on-line facility.

HOW IT ALL BEGAN

The original concept of synchronization was pioneered in the film industry. Film is generally shot in what is known as *double system*: the visuals are captured by a camera and the dialogue recorded on standard audio tape. Since these are separate devices, a method was needed to keep them running together on playback. While the camera's film transport is inherently frame-locked (in a very secure way) by the nature of the mechanical sprocket drive, this is not the case for audio transports. Without some constant means of correction, tape slippage alone would spoil all hopes of long-term lip sync.

A master reference was needed whereby minor fluctuations in tape speed could be neutralized or *resolved* on playback. The solution was to record a continuous, precise audio frequency alongside the dialogue tracks. Later, the variations in this frequency would serve as an absolute reference for the speed of the playback deck. Through this *resolver*, the tape transport would receive messages to speed up or slow down—whatever it takes to keep this previously recorded tone at its original frequency.

This system of resolver synchronization still works marvelously well for the film industry—where scenes are pieced together from many shorter takes. The sync sound from the location recorder is quickly *resolved* to magnetic film where sprockets once again give frame-accurate control to the sound editor and render a one-to-one correspondence to the visuals. Resolver sync works quite well for transferring audio tape to mag film. ("Mag film" is simply film coated with an oxide formulation like standard audio tape, but having no picture. It is a convenient way to link up reels of picture and sound mechanically, by the same sprocket drive.) The process is done once and then the sound is manipulated in the sprocket format. For strictly audio purposes, however, it leaves a lot to be desired. While two machines can be kept synchronized utilizing this level of technology, the drawbacks are pretty sizeable. Unless you always start precisely at the beginning of the tape, the two transports will never be locked. Try doing that every time you want to overdub a hit three minutes into a song! Guaranteed, it would make things very frustrating.

Fortunately for us in the world of audio, the video industry had their own share of frustrations trying to label a frame of video, and this led directly to the development of SMPTE code technology. Until the late 60s, all that could be done to locate a particular video frame was to count tachometer or control track pulses. This provided some increasing numbers to go by (from a given fixed point on the video tape); it was a nice rule-of-thumb, but hardly accurate.

Drop-outs and tape slippage once again resulted in accumulated inaccuracies and no one was yet able to give the video frame an *address* and get the machine to go there twice in a row. The best that could be expected with this type of video editing system—even today—is a consistency of two or three frames between the preview and the actual edit. It was hit or miss at best, and would never do for any kind of audio purposes where time durations equal to only a fraction of a single frame can make a noticeable difference. If we consider, for example, the monochrome video standard of 30 frames per second—as related to an audio tape spinning at 15 inches per second, then each frame equals about 1/2 inch of audio. In most situations, that frame of audio could be subdivided into several slivers of sound with quite an audible change at each point.

Suffice it to say, whether it be for editing purposes or synchronization of multiple tape transports, a means of positively identifying the location of the tape in a precise and repeatable way is needed. Thanks to the

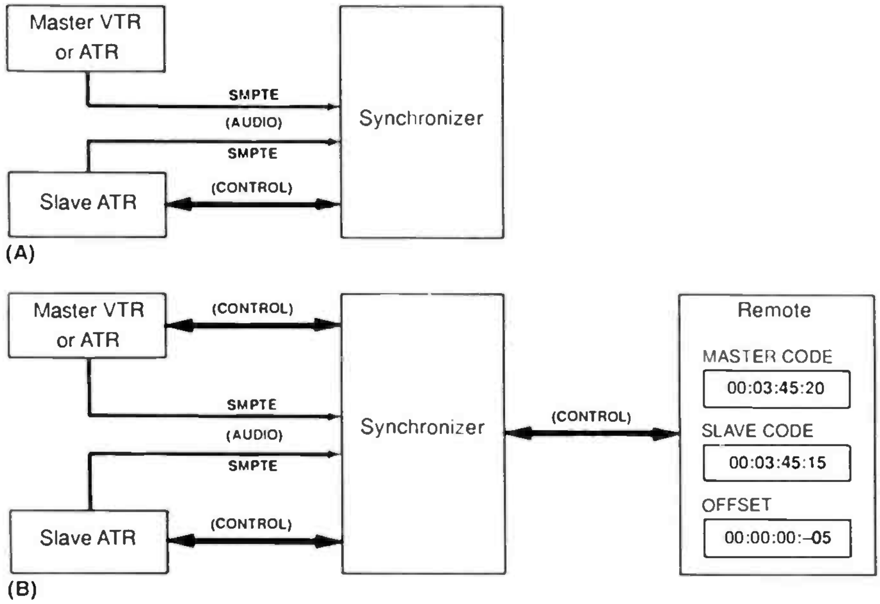


Figure 1. A two-machine lock-up. At (A) Without remote–slave chases location of master code (all control is from master transport), (B) is with remote–all control is from the remote.

work begun by the Society of Motion Picture and Television Engineers (SMPTE) in 1969 and standardized in 1971, that need has been fulfilled and increasingly implemented by manufacturers with each passing year.

THE ESSENTIALS OF SMPTE TIME CODE

Reduced to absolute basics, SMPTE time code is a frame-accurate digital clock. The passage of time that it marks need not correspond to actual local time, but can be arbitrarily set by the user. When any audio or video event is marked by SMPTE time code, it establishes a permanent identity usually referred to as an *address*. Measured in hours, minutes, seconds and frames, the events become temporally and spatially related to discrete points along the path of the tape.

Actually, there are two kinds of SMPTE code. Both are composed of continuous streams of digital words created by a SMPTE generator and stored on tape. The most commonly used variety is longitudinal time code (LTC), which is stored as an audio signal along the length of a spare track. (This track of recorded SMPTE is frequently referred to as a *stripe*.) The second variety, called vertical interval time code (VITC), is stored as video information on the video track during the blanking interval in each frame of video. (The blanking interval is the time during which there is no video information, in order to allow the CRT electron gun scanner to retrace its path to the top of another frame.) While VITC has many advantages, LTC is most commonly used in a multi-track audio facility—even one that does audio-for-video work. Hence, for the remainder of this article, when we speak of SMPTE time code, we will be referring to the common longitudinal variety.

HOW DOES SMPTE TELL TIME?

SMPTE tells time simply by increasing a digital counter by one with every passing frame. Technically, SMPTE time code is not really SMPTE time code unless the generator's counter uses video sync as its timing reference. In other words, the SMPTE generator ought to agree with the video sync generator on where a frame actually begins.

If you are shooting or editing video, this can be of utmost importance. If you operate a smaller audio facility, you don't need to be too concerned.

For most audio applications, it's simply not relevant. Simply locking up two audio machines does not require a video reference, and even if you are doing post production sound to a video dub, usually the tape will have been striped with SMPTE at a facility utilizing a house sync reference. So for most purposes in the smaller studio, you won't need to deal with the added expense of a video sync generator.

Here is what you will need for basic synchronization of two tape transports: a SMPTE generator, a synchronizer and the appropriate interface cables. A remote controller is a useful and time-saving option, but if you are on a tight budget, get just the basic gear and you can get to work right away. Granted that the remote will allow you to preprogram a lot of repetitive moves, but the synchronizer itself controls all the essential operations (See *Figure 1*).

Simply stated, the synchronizer is a device that "listens" to the SMPTE time code previously recorded on two or more tape machines (audio/audio or audio/video), compares the code and adjusts tape speed based on the results of that comparison.

One of the time code inputs to the synchronizer is designated as the *master* reference, and hence, sets the standard. The other time code reference is aptly called the *slave* reference. According to how we implement the synchronizer, it will continuously adjust the speed of the slave transport so that the time code numbers of the master and slave maintain the desired relationship.

For example, assuming we want the slave to follow the exact time code sequence of the master (which is commonly the case), we would enter a zero *offset* on the synchronizer—meaning we want the time code numbers of both master and slave to be *locked*, exhibiting identical numbers when both transports are playing. The synchronizer will compare the two time codes and control slave speed so there is no difference between the numbers. At this point, the machines are said to be *frame-locked*.

Note, however, that a zero offset might not always be desirable. If you are working with video, and dialogue is on screen, there is always a possibility that the lip sync might need adjusting. As little as two or three frames of lead or lag in the audio can begin to look a little strained. By experimentally determining the exact amount of difference, the number of frames can be programmed in as an offset, to better coordinate lip movement and voice. Another possible use of offsets is for signal processing when doing music on two multi-tracks. Small offsets may, in certain cases, add a spacious dimension to tracks that seem a little sterile when perfectly locked. This is similar to the *chorusing* effect which results from splitting a signal, delaying one path very slightly and recombining the two.

SOME PRACTICAL QUESTIONS ANSWERED

With this brief overview in mind, let's cover some applications notes about specific areas of concern. For openers, what about record level? It seems that no two sources on the subject of time code ever seem to agree on this. My own experience tells me it is best to derive it experimentally for the system you are working with. This is especially important for narrow gauge recorders (1/4-inch 8-track machines, 4-track cassettes, etc.) where headroom is limited and noise reduction often cannot be switched off. Much of this flies in the face of conventional wisdom which recommends putting healthy levels on tape and switching

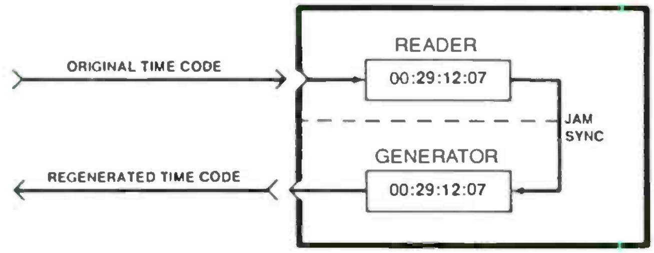


Figure 2. The SMPTE Time Code Reader/Generator. During the Jam/Sync process, original time code (perhaps during a discontinuity or damage, or just in the need of copying) enters reader and emerges freshly generated.

off noise reduction. Nevertheless, SMPTE code and its related equipment seem to be more forgiving than people initially thought. For such equipment, I find that conservative levels (somewhere around –3 VU) seem to work best; this also reduces the tendency to bleed through on adjacent audio tracks.

As to which track (on a multi-track recorder) is best for holding time code, a common practice is to utilize one of the edge tracks, leaving the track next to it empty as a guard band. However, the outside edges of a tape are susceptible to damage by improper handling or poorly aligned tape transports, so some users are most cautious and record the code one track shy of the outside track (for example, track seven on an 8-track recorder). The reason for using a guard band is threefold: to protect the SMPTE from intermodulating

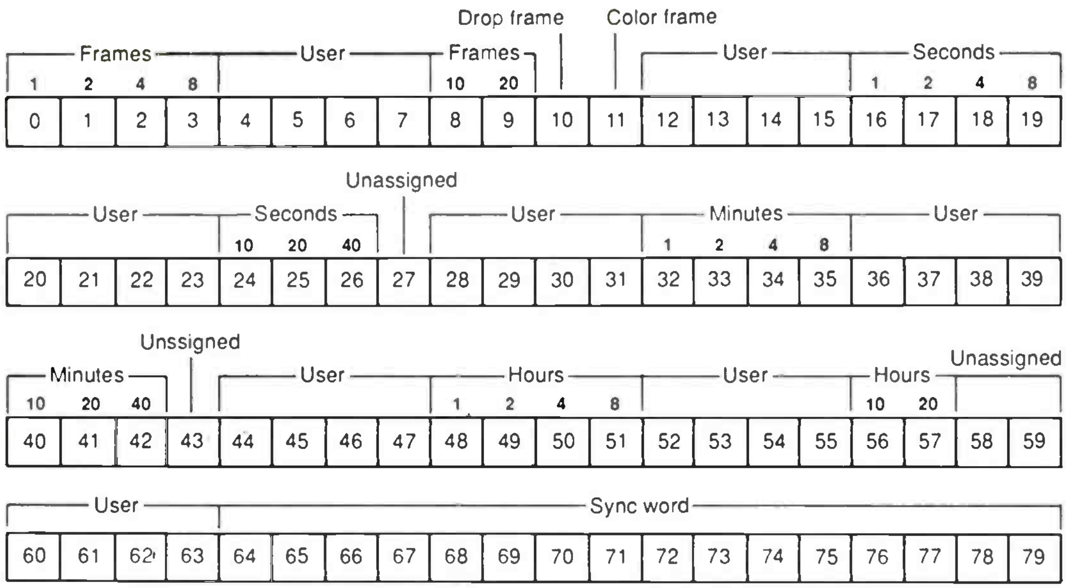


Figure 3. SMPTE time code bit map (longitudinal time code–80 bits). Note that counting proceeds by a hybrid of true binary and some arbitrarily chosen decimal numbers. A digital "one" in a bit cell indicates that quantity has been added to the count.

with the neighboring track, to protect the audio tracks from the sound of SMPTE (which is truly hideous) and also for protecting the SMPTE from nervous punchers (by sticking it on a remotely located track).

If your multi-track tape is going to remain in-house for an entire project, using an edge track with a single guard band is all you need to do. If, however, the project is going to be shuttled to various studios, it might be wise to use an inner track for that extra measure of protection against edge damage. While it's never a good policy to record on your guard tracks, if you really need to squeak out another track, you can try (carefully) recording some low-end, intermittent program material on those tracks (such as bass or percussion) and gate out any SMPTE crosstalk on mixdown. If you do end up utilizing a guard band, make sure the level is not so loud that it interacts with the SMPTE, thereby making the synchronization unstable.

How about frame rate? Since SMPTE was designed primarily to meet the needs of the television and film industries, the subdivision for seconds of time is always related to the frame. SMPTE time code generators give us a choice between several frame rates: 30 frames/second (the black and white TV standard), 29.97 frames/second (the NTSC color TV standard, called *drop frame*), 25 frames/second (the PAL European TV standard) and 24 frames/second (the frame rate for film).

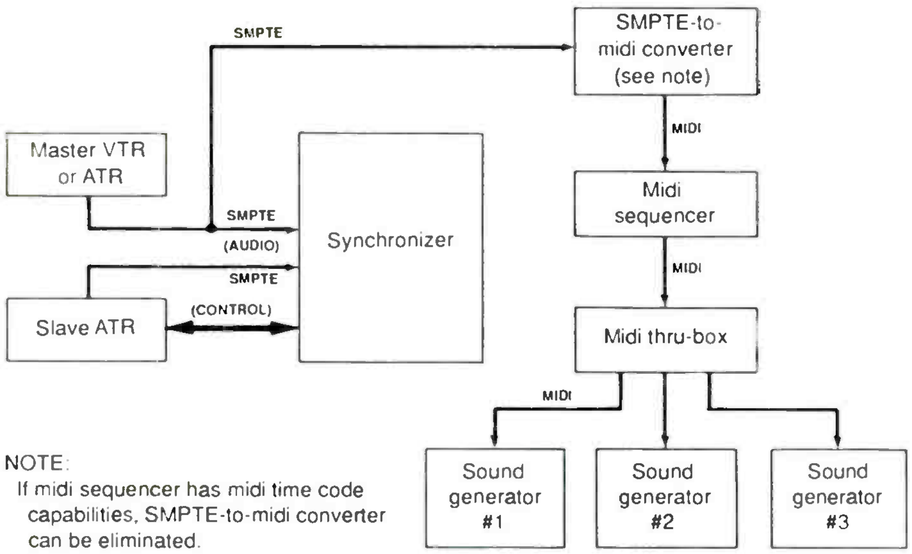
For audio purposes, the 30 frames/second rate will be used almost all the time, but sometimes when a video dub comes in from an outside facility, its time code may be at one of the other rates—particularly if the visuals are not final cuts, but are earmarked for further editing after you track the sound. Synchronizers act a little strange when master and slave code have different rates, so don't assume anything when synchronizing a product from another studio. Be sure before laying code on your slave.

Another important point is to record sufficient amounts of extra time code both before and after the program (about 20 to 30 seconds of extra SMPTE at the head and tail should suffice). While your synchronizer may usually be able to stop on a dime, occasionally it will mysteriously refuse. If your little buffer of extra code is only five or ten seconds, the machine may careen into the next song or go totally off the reel before the synchronizer can regain its poise. This usually occurs if you instruct the machines to go close to the beginning or end of the program. If the tape runs out of code, the synchronizer will shuttle the tape transport all around trying to find something it can read. So don't be too frugal with your tape, even if you have to crack open another reel in order to insure a sufficient pad of extra time code.

If your system is locking up quite well for a while and then things start to go slightly berserk, try some first aid before panicking. First, it may be time to clean the tape heads again. When your track width is narrow, a little dirt can do a lot of damage. If that doesn't do it, the next thing to do is make sure all connections are still sound. Jiggle them all around; maybe even apply a little contact cleaner. Usually a little tender loving care will take care of it.

WORST-CASE SCENARIOS

There are worst-case scenarios, though. What if your machines won't lock up because you accidentally erased a big chunk of time code? And you were almost finished with a week's worth of recording! A nightmare and embarrassing, too; but fortunately, in most cases this problem is easily solved—assuming you have access to the proper equipment. I didn't mention this equipment in the basic list, because it is not an absolute requirement for the smaller studio. But serious professionals in the large audio-for-video houses would not be without a versatile time code *reader/generator* module as a part of their system. (This is not because they accidentally erase time code every day, but rather because of all the other nifty things they can do with it.)



NOTE:
If midi sequencer has midi time code capabilities, SMPTE-to-midi converter can be eliminated.

Figure 4. A two-machine plus MIDI lock-up

A reader/generator differs from a plain old generator in that it can read existing SMPTE code from tape and spew a refreshed version of the incoming code from its generator output. It will do this even if the code is somewhat degraded or partly missing. This process is generally referred to as jam sync (See *Figure 2*) and is properly used in any case where SMPTE time code needs to be copied. Analog audio always gets somewhat degraded with each generation, and while program material might not seem noticeably different, SMPTE code is particularly sensitive to any loss in the integrity of its waveform. While you could probably get away with one generation of direct copying without losing sync, there would undoubtedly be a noticeable difference in the way the transports handled at high wind speeds, where degradation is more noticeable to the synchronizer.

In any case, to fix a missing code problem and still maintain the original time-code numbers, a reader/-generator is the only way to go. (To the small studio owner, it is comforting to know that such devices can be rented as needed.) The actual procedure is pretty simple. Find a spare track on which to record the regenerated code. Route the damaged code into the reader, and route the generator output into a spare track. Switch the reader/generator into the *continuous jam* mode and start recording.

The jamming procedure will generate fresh time code that takes its numbers from those still present on the damaged time code.

When the erased section comes up, the generator will just keep on running at the same rate (temporarily taking its cue from the tachometer pulses)—even though no code is coming into the reader. In this way, it will fill in the blanks with the code that theoretically should be there. It's easy as pie when you have the right gear.

There is actually another way to remedy this particular problem without benefit of a reader/generator. It will take a little longer, and the original numbers will not be precisely in tact, but it will work! Prepare to record another track of SMPTE and set your no-frills generator box (or time code CD) to start rolling at the same numbers that were on the original. Cue the tape up to the top, and start the generator at the appropriate time as you go into record.

The resulting new code will probably be ten or more frames off, depending on your reflexes. Later, upon synchronization with the other transport, the exact offset can be experimentally determined and stored in the synchronizer. The results will be satisfactory even though the numbers may be a little different between the master and slave.

WHEN YOU NEED A READER/GENERATOR

There are, however some things that only a reader/generator can do. While the smaller, more specialized studios may not use these functions as often, they are really a staple item in the large multi-service facility. These functions are also worth mentioning to illustrate the flexible and farsighted nature of the SMPTE time-code specification.

Referring to *Figure 3*, the longitudinal variety of SMPTE time code is basically a stream of 80-bit digital words increased by one for each passing frame. There is, however, much more information stored in those 80 bits than just a frame count. Actually, only 26 of those 80 bits are occupied with the usual numbers we associate with SMPTE. Some of them are occupied with housekeeping functions (like the 16 bits dedicated to defining the end of each digital word, hence defining the beginning of the next word as well). All this is necessary just so we may know in which direction a tape transport is moving. Some bits are *flags* which are mostly useful in editing NTSC color video, and others remain undefined and reserved for future implementation.

The most helpful of all, though, are the 32 appropriately named *user bits*. These chunks of data, dormant until accessed by the user, can be used to label sequences, trigger remote equipment and/or run alternative time code sequences (simultaneously with the main reader output). One use is to run time code backwards so elapsed time can also be displayed. Another common use is to run alternative frame rates so a choice would always be available to the next user.

These rather specialized applications of SMPTE time code will probably not be utilized much by the small studio operator, their preference being for a low-cost turn-key system that allows two tape transports to *chase* and *lock*.

Still, as the technology continues to get less expensive to design and manufacture, many of the very sophisticated features of high-end synchronizers will, undoubtedly, continue to find their way into the small studio.

THE SMPTE TO MIDI CONNECTION

Since 1983, when the specifications for MIDI were first laid down, a major change has been seen in production technique. A good deal of the process is done in a tapeless world of synthesizers, samplers and sequencers that communicate via the *musical instrument digital interface*. (A recently published article on MIDI revised this well-known acronym to read *multi instrument digital interface*—which I think is a fairer description of its contemporary applications, since it has expanded to include control of signal processing, console automation, lighting control and a host of other activities not even vaguely related to music.) Both SMPTE and MIDI are powerful tools, and when they come together synergistically, the production possibilities are virtually limitless.

When tape is not required, MIDI is a self-sufficient means for electronic musical production. Even rigorous pieces of music with complex tempo changes can be programmed into computers, edited and played back on a variety of sound generating modules. But when such a production needs to be synchronized with tape, two sovereign domains must somehow find a common ground. In the course of its short history, several approaches have been developed to synchronize MIDI with tape, all of them having some major drawback—until very recently.

If MIDI data were capable of being recorded in real-time within the bandwidth of a standard analog recorder, the matter would have been much simpler, but such is not the case. SMPTE is transmitted at a maximum frequency of 2400 Hz (80 bits/frame X 30 frames/second)—which is easily recorded; MIDI (being loaded down with much more than just timing information) is transmitted at 31.25 kilobaud—far above the recordable range. Various means of extracting timing information—apart from all the other data—and representing it on tape have been devised over the years:

FSK

First came FSK (Frequency Shift Keying). FSK simply shifts between two easily recordable tones (one low and one high) 48 times per quarter note, adequately representing the MIDI clock rate of 24 PPQ (pulses per quarter note). While this works rather well for locking MIDI to tape, it does have its limitations. One drawback is that it is always tied-in to the tempo of the musical composition at the time of recording the tones. The only way to change tempo later on is speed up or slow down the tape transport—which may not be acceptable. Another inconvenience is that FSK contains no location information—it just ticks away, one tick being indistinguishable from the next—the user being required to start the synchronization process at the top of the program every time. This makes one very inclined to print all synthesized tracks to tape before mixing, in order to avoid a lot of extra tape shuttling and interminable waiting. Given these limitations, it does work quite well.

A recent improvement in FSK, sometimes called *Smart FSK*, has become very popular in recent months. Smart FSK, in addition to the MIDI clock information, also contains the additional MIDI information of Song Position Pointer (SPP), a means of counting and thereby identifying each sixteenth note in a sequence. The big advantage here is that a MIDI sequencer can lock up to a tape even if the tape is started in the middle of the composition.

With Smart FSK, the producer is now encouraged to keep the synthesized tracks as *virtual* tracks, thereby preserving more tape tracks for live instruments.

Still, what is glaringly missing from both of these systems is any kind of real-time reference linking a particular location on tape with a particular moment in time. Only SMPTE time code is able to positively identify space and time in this manner, and for this reason, there have been various attempts to get SMPTE and MIDI to talk to each other in a comfortable and sophisticated way. Until recently, it took a dedicated piece of outboard gear—a "SMPTE to MIDI converter" such as the classic Roland SBX-80 (See *Figure 4*). Such devices are easiest to use on simple pieces of music with an invariant tempo; with pieces that have tempo changes, however, the programming can get quite tedious.

For such devices to transmit such information to a MIDI sequencer requires that the duration of the piece and each point of tempo change be programmed in at the converter itself.

Correlating the absolute time of SMPTE with the relative time of the MIDI sequencer is called *tempo mapping* and must be performed in order to translate real-time into MIDI clocks, but the programming required by outboard SMPTE-to-MIDI converters is time consuming—especially when dealing with accelerating or decelerating passages.

CREATIVE FREEDOM, AT LAST

The search for a totally transparent SMPTE to MIDI interface may well be at hand. Several ways have been developed to eliminate the hardware-based SMPTE-to-MIDI-converter with all its complexities, the most notable of which is MIDI time code (MTC). From the sound of the phrase, it might seem, at first, that someone finally found a way to put the MIDI signal on tape; but nothing could be farther from the truth. MIDI time code still relies on a stripe of conventional SMPTE time code as the absolute timing reference.

What is new about MTC is that by utilizing some of the previously unassigned bytes and also some universal system exclusive commands (the provisions made in the MIDI specification for unforeseen applications), engineers have devised a way to send essential SMPTE timing information down a MIDI line.

Sequencers able to utilize MTC can now superimpose their own tempo maps on top of the real-time reference. Any changes in arrangement or tempo can now be quickly dealt with by the sequencer itself—without ever needing to restripe the tape or do additional tempo mapping with some external box.

Likewise, when scoring or adding sound effects to video, *hits* can be tagged by SMPTE number or MIDI event—whichever works best. SMPTE and MIDI have become interchangeable currencies, each having unique strengths which can now be tapped without restriction. The possibilities, most of which are just beginning to be implemented, are rather exciting. db

JOHN BARILLA

Perspectives on Sound Design: An Interview with John Alberts

What is sound design anyway? This contemporary buzzword often means different things to different people, but John Alberts' personal definition is all-encompassing.

John Alberts says "My concept of sound design for television shows is to see the audio from start to finish—from pre-production to post-production, which may involve consultations with producers, advising on microphone choices or techniques, interfacing equipment, etc."

Much of Alberts' work involves this kind of holistic approach, and the net result bears an integrity that has made him one of the industry's most popular sound designers.

Much of his work also involves music-oriented television projects. A typical Alberts sound design was the Spike Lee *A Cappella* show (recently aired as a PBS *Great Performance*). Alberts was involved with this show six months before taping, consulting with producers and designing all sound effects and ambiences for the non-concert portions of the show.

Figure 1. John Alberts at the studio console.

Another aspect of Alberts' work is recording, editing and post-production of live musical performances for TV and video release. His recent credits include concerts by The Who, The Rolling Stones and Paul Simon. Since audio-for-video requires a different set of skills than mixing a hit record, on such venues Alberts often works with Producer/Engineer Bob Clearmountain. The Alberts-Clearmountain collaboration has been quite fruitful, since their skills are so complementary.

"We work together and have different areas of expertise," Albert says. "On a show like that (the Rolling Stones concert), I consider myself the equivalent of a mastering engineer in the record process. It's kind of my job to get his mixes on the air in the truest possible fashion."

While Alberts is constantly involved in special projects of this nature, he is probably most well-known for his post-production mixing. Most of it goes on in a 48-track digital suite at Howard Schwartz Recording in New York City. An independent contractor, Alberts has an exclusive arrangement virtually locking out the suite for his personal use.

A major part of Alberts' week is spent doing post-production for his number one client, Broadway Video, which produces NBC's *Night Music*. Having once been a studio musician, Alberts has all the right sensitivities for portraying Jazz Artist David Sanborn and his ensemble of great studio musicians (and guest artists, as well) in the best possible light. Alberts was preparing a series of previously aired *Night Music* shows for international distribution when I arrived to interview him.

We first spent some time exploring the new tools of post-production, and then talked about what goes on at the various stages of the post-production process.

BULLISH ON D-2

A typical Alberts project involves mixing a 48-track digital recorder (a Sony 3348) to picture. At one time, all digitally recorded sound was eventually laid back to some analog audio tracks on a one-inch video layback deck. By the time the product hit the airwaves, the audio was destined to go through several analog generations—with losses in fidelity. In recent years however, some post-production facilities have installed the relatively new D-2 format DTTR (digital television tape recorder). In an effort to maintain the pure quality of digital audio for as long as possible in the broadcast chain, Howard Schwartz Recording purchased a SONY DVR-10 D-2 recorder which features digital composite video plus four channels of 16 bit digital audio (with extra dedicated analog tracks for cue, control and time code. *See Figure 2).* Naturally, considering the potential improvement in the quality of the end product, Alberts is quite pleased with the D-2 format.

Figure 2. The Sony DVR-10 D-2 system at Howard Schwartz Studios.

The show is digitally recorded on two 24-track machines, and the tracks are always edited to picture prior to the mix.

"I think D-2 is the biggest innovation in audio post-production in quite a while," he said. "For example, this show *Night Music* we've been repackaging for international distribution and we need both stereo and mono mixes; We've gone back to the original digital audio masters reconfiguring them on D-2 with stereo mix on tracks 1 and 2, and a mono mix on 3 and 4," he said. "D-2 is pretty amazing. The shows I've done recently...

"The *Tommy* show (The Who's 1989 Los Angeles performance)—that was shot in D-2 and recorded on 48-track digital and edited in D-2. It was the first all-digital TV program—both audio and video," Alberts said. The show aired live as a pay-per-view, rebroadcast for the Fox Network as a 90-minute special, and finally packaged for home video as a 2-hour and 15-minute program. The Fox and home-video versions were remixed by Alberts and Clearmountain on the original 48-track digital. Alberts then "re-conformed" (basically putting the songs in the right place as per the edited picture) the mixes to the edited video on D-2. While re-conforming to the new song order—with possible deletions, insertions and edits—does not require a remix of the tracks, it does require building new transitions, "whether it be applause or whatever is happening between the songs to make it seem like it actually happened that way,"

Alberts said. "That's technically called a reconform and sweetening—the addition of other elements to build the transition. The home video of Tommy was actually mastered to VHS Hi-Fi right off D-2, so therefore, there are no analog generations in the taping of the show until the final product," he said.

The D-2 format has gained rapid acceptance in audio post houses over the past two years. According to Alberts, the D-2 is now seen as a cost effective unit, not exceeding the price of a one-inch video deck (about $80,000). As historically proven, TV stations—having such a mass of equipment and the need for uniform standards—have not been rushing to convert to D-2.

One-inch and M-2 machines are still the *de facto* broadcast standard—by virtue of sheer numbers in use. For post-production purposes, however, many stations are making use of D-2.

Figure 3. Alberts at the console mixing ***Night Music.***

Figure 4. The foley room at Howard Schwartz.

WORKING ON "NIGHT MUSIC"

This show is an example of Alberts' innovative post-production techniques. Undoubtedly, his past training as a musician serves him well in this task, because it requires a degree of musical discernment that mere technical training could not provide. The show is digitally recorded on two 24-track machines, and the tracks are always edited to picture prior to the mix.

"We used to do shows like this where we just mix the songs and then edit the songs to the picture," Alberts said. "My feeling is it's better to edit the tracks. If you can mix to the final picture—actually, if they change a shot, like a close-up of David Sanborn playing a sax fill—you're going to want to push his track a little bit. This way you have the raw tracks available to you looking at the final picture. Also, if you have edited tracks, you don't have to deal with offsets; you just match (timecode) to picture," he said.

For *Night Music*, the two 24-tracks are edited to a 48-track machine (digital-to-digital), resulting in an edited audio master. What differentiates the edited audio master is that it may contain a solo performance from another take integrated into the tracks; also, its time-code numbers have been conformed to the edited picture. When blending tracks from two or more takes, having access to the raw tracks allows Alberts to make the insertion of the new material at the optimum point for each instrument. Drums may all be from a single take, but a saxophone phrase may be inserted while the musician has actually paused for a breath! Other tracks may be edited at different points, making the transition completely undetectable.

"They usually edit the picture first, so I'm kind of locked-in to where I'm coming from and where I have to end up," Alberts said. "How I get there is usually up to me. Sometimes, however, when circumstances dictate that the musicality of the edit may be a critical call, I would make the audio edit first, and let them conform the picture to the audio edit."

I do the run-through on Saturday, and get all my tracks together and we tape dress rehearsals and the air show.

Figure 5. One of the Studio's Sony 3348s alongside a 3324 permit an entire production to be built up in the digital domain.

TRACKING *SATURDAY NIGHT LIVE*

The sound design for NBC's *Saturday Night Live* is done by Technical Producer Stacy Foster, who has acted as somewhat of a mentor to Alberts. Here, Alberts acts as part of Foster's team, taking control of the live mixing and tracking for the show.

"We record *Saturday Night Live* 48-track digital and we re-edit and remix every show for the repeat broadcasts (This 48-track recording is done on two ganged 24-track digital recorders)," Alberts said. "One 24-track (manned by Music Engineer Jay Vicari) is dedicated to music—both the house band and the guest band—using a pretty traditional music tracking situation: 6 or 8 tracks for drums, everyone on a separate track, very little mixing to tracks. That's what's handled by the music control room. (Two separate teams of engineers record music and production tracks separately, synching them together later on, for post-production.) I record the production tracks which consists of stereo pairs: stereo audience, stereo sound effects, stereo underscored music and 8-tracks dedicated to dialogue—that's a big part of my job," Alberts said. "I get all the microphones in the studio off a splitter. Generally, we average between 18 and 24 microphones—that includes wireless mics, mics that are mounted in sets, etc., and I mix those to tracks. I only work on Saturday, so I start rehearsals with them on Saturday at noon. I do the run-through on Saturday, and get all my tracks together and we tape dress rehearsals and the air show. The post-production for (future broadcasts) I do at NBC," he said.

Figure 6. The SSL fully-automated console.

It should be noted that there are different requirements for synching analog and digital multi-track machines.

DEFINING THE POST-PRODUCTION PROCESS

While the terminology and concepts of post-production may be obvious to those who work in the field, it is, no doubt, an audio specialty. Some competent recording engineers outside the field might want some definitions, so I asked Alberts to walk us through the various stages of the post-production process. (While there is clearly no such thing as a "standard post-production job," the attempt here was to generalize as much as possible, but where a digression is profitable—to explain a piece of equipment or a given technique—that has also been included.)

There are basically four stages in audio-for-video post production: The first stage is the *strip* (an East-Coast term) or *lay-off* (as they call it on the West Coast), which textbooks refer to as the *layover*. (With so many terms describing the same phenomenon, it is easy to see how an outsider could find the field a bit perplexing.) This is followed by the *pre-lay* (the second stage), then the *mix* (the third stage) and finally the *lay-back* (also known as *re-lay*). This maze of expressions becomes easier to grasp when Alberts explains their meaning.

1. *The strip* involves "taking the video master and transferring the two tracks of audio and time-code to an audio multi-track tape. The two audio tracks may be a live mix of the show, or in the case of a documentary, narration and natural sounds," Alberts says. "Whatever the sound is that accompanies the edited picture, you transfer those tracks along with matching (regenerated) SMPTE time-code to a multi-track tape," he says. (It should be noted that some shows have tracks recorded simultaneously on multi-track, in which case the task here may also involve editing the tracks to match the previously edited picture. This process of machine-to-machine digital editing will be discussed shortly.—ED.)

2. *Pre-lay* is "adding elements to your multi-track tape prior to the mix session. It is the equivalent of sound editing in filmmaking, where ADR, Foley, sound effects editors and music editors are cutting mag-film to picture,"

(For those unfamiliar with film terminology, ADR is short for automatic dialogue replacement, where an unsatisfactory passage is later dubbed-in; Foley is where technicians create footsteps, punches and other such sounds while watching the on-screen action and. Mag-film is the medium on which film editors handle sound elements: it is film that has been coated with a magnetic oxide, hence, capable of recording sound in a conventional way. –ED.)

Alberts says, "For example, let's take a documentary as a generalized case. You go through the entire show (with a multi-track audio machine synched to picture) and build your tracks. You might, for example, put two tracks from your video tape on tracks 23 and 24. On tracks 17 through 22 you could build your background ambiences—traffic, birds, wind, whatever. Tracks 9 through 16, you might build your spot effects—door closings, car horns—whatever's happening with the action," he says.

It should be noted that there are different requirements for synching analog and digital multi-track machines. When doing the strip (layover) to analog, not only regenerated SMPTE should be recorded, but also a 59.94 Hz NTSC sync pulse—as an absolute time-accurate reference and also as a safety factor. In case the SMPTE time code was accidentally damaged or destroyed, restoring the code would be a totally accurate process.

"With digital, however, it's not necessary, because digital has a control track built into the format, which is more or less the same thing," explained Alberts. "If something happened to your time-code track on a digital 24, you'd still be okay. The digital multi-tracks don't resolve to time-code, they resolve via control-track. It's like a video control track; it makes the tape play back at the right speed.

The time-code simply enables you to do editing with it," he said. "Digital playback is flawless, but if you don't lock the multi-track to incoming video (during the recording process), then (the digital multi-track)

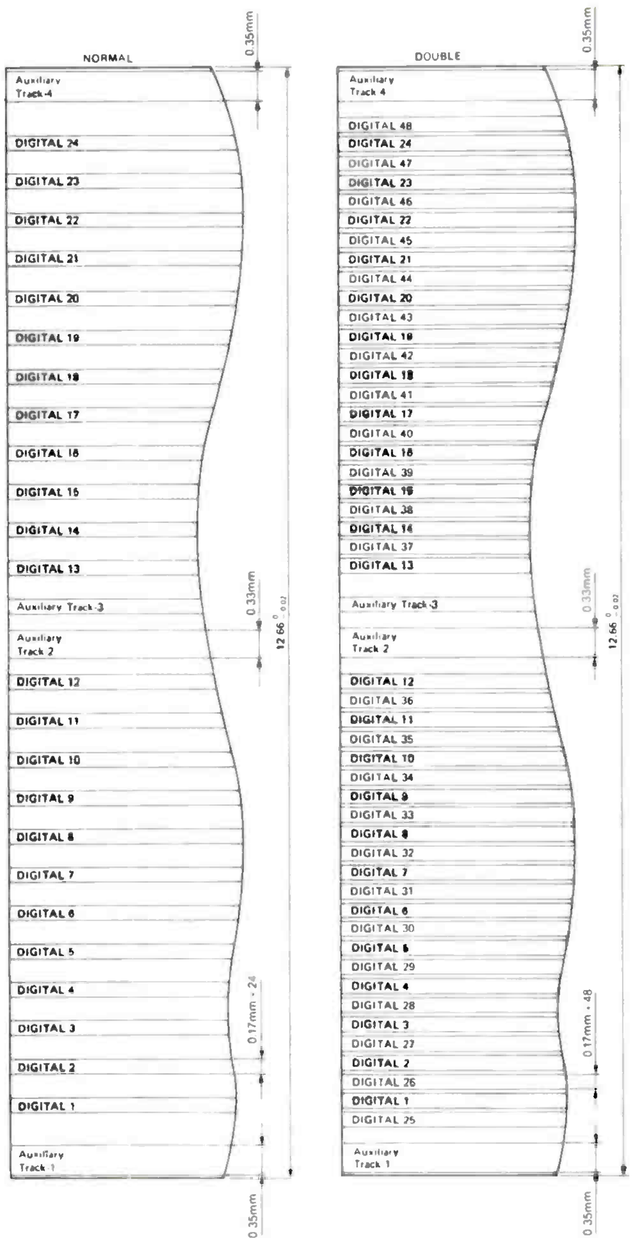


Figure 7. The DASH-system normal (24-track, left) and double density (48-track, right) formats. This clearly shows how tapes can be moved from one format to the other with compatible ease.

will run on its own internal crystal and there will be a discrepancy: 60 Hz versus 59.94. In post-production then, the audio and video will not be synchronous. But even that is fixable using the precise varispeed function of the digital 48-track," he said.

While we were speaking of 48-track digital, Alberts took a moment to list some of the features of the SONY 3348 that really aided the post-production process. "It has an on-board sampler with 20 seconds of RAM, which means if you have something on a track and you want to move it, you can basically put it on a chip and then put it some place else," Alberts said.

"It comes in real handy. It also has two-channel digital bouncing capability, so if you want to just move tracks for one reason—for ease of mixing, you can do it.

"I've been working in the digital 48-track realm for three or four years (prior to that working with two 24-track machines). It's only with the advent of the 3348 that we're down to one machine, half-inch tape; so it doubles the speed you can work at," he said.

"You're not always waiting for two machines to lock up. Additionally, on the 3348 you can take a tape that was recorded on the *3324*, put it on the 48-track and get 24 more tracks. And 1 through 24 on the 48-track format play back on the 24-track," he added. (For a visual representation of the "double density" format used in SONY 3348 DASH recorders, see *Figure 7*).

THE MIX

Once all the additional elements—dialogue replacement, sound effects, or musical *sweetening*—have been added to the existing raw tracks, it is time to mix down to a standard format, which makes creating variations an easier job.

"I generally mix to tracks 1 through 6. Basically, when you're mixing on a multi-track, it's pretty universal for audio post-production. You're mixing within the machine. You build your tracks with your strip and pre-lay, and then you basically do a bounce," Alberts said. "So you're rerecording your mix onto tracks of your multi-track, and this way, you don't have to spin another mix machine.

There is a piece of equipment that I'm pretty excited about, and that is *ScreenSound* ...

Your elements and your mix are on the same reel. I generally put my mix on tracks 1 through 6: 1 and 2 being stereo dialogue, 3 and 4 being stereo music, 5 and 6 being stereo effects. Using the SONY 3324 and 3348, you can play your mix back on either the 48- or the 24-track machine," he said.

Layback is simply the transfer of the final mix back to the video master. "When you do the layback of a 90-minute show, you do it as an A/B roll," Alberts said. "In other words, you roll the second machine later in the show and you just do a switch, during a commercial or whatever."

Alberts makes the switching routine simple by cascading the digital output of the second machine through the first: tracks 1 through 24 on the second machine feeds 1 through 24 on the first, so the switch is effected by going into input mode on the first machine. "If there is no commercial, the process is a little more involved," Alberts said. "A manual switch might result in an audible glitch in a continuous program, so the reel change has to be done as an edit on the video master," he said.

BULLISH ON SCREENSOUND

I asked Alberts if there was any new piece of post-production gear that captured his imagination. Without hesitation he replied, "There is a piece of equipment that I'm pretty excited about, and that is *ScreenSound* from Solid State Logic. It's a hard-disk based editing and mixing system which uses a recordable laser disc for video and various hard drives for the actual sound," he said.

"It's an all-inclusive mixing/editing system. It's very exciting. ScreenSound is a compact workstation-type system using a light pen and tablet—similar to paintbox video. While it would not replace a large digital multi-track facility (it's only 8 tracks), it would be most appropriate for jobs with a limited number of tracks. For such jobs, ScreenSound is amazingly fast," Alberts said.

As to applications, Alberts says he sees using it as a tool for doing fast *pre-lays*—off-line—and then bringing it into a larger facility and doing a dump onto digital multi-track to finish the job (SSL loaned Alberts a ScreenSound system for trial and evaluation. His opinions appear in the sidebar below–ED.).

From the time I spent with Alberts, it became very clear he enjoys getting up in the morning to go to work. He forsook the glamour of rock-n-roll mixing a long time ago for the less visible, but more stable world of audio post-production. Not that it doesn't have its long nights now and then. Sometimes Alberts gets so busy he can't even make it from the Manhattan studios to his home in nearby Brooklyn.

Alberts likes to keep his work to somewhat normal hours as much as possible. But whatever the hours, Alberts has a job that totally satisfies his creative needs.

"I love what I do," he says. "I really love what I do. I enjoy coming to work. I've been very lucky, I have to say. I've been blessed with some really good clients and jobs that are lots of fun." db

Solid State Logic ScreenSound

JOHN ALBERTS

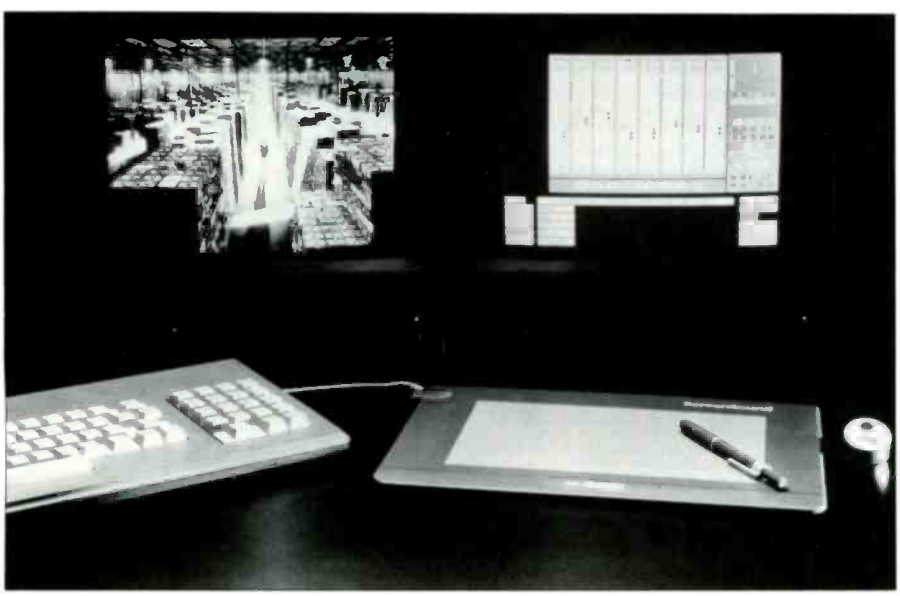
• I used the SSL ScreenSound in a demo situation at Howard Schwartz Recording for one week. My goal with the demo was to do some relatively simple jobs with the unit. I did the assembly, sweetening, mixing and re-lay to D-2 for two half-hour specials of *Saturday Night Live* (European syndication), as well as the assembly and placement of sound effects for NBC's *How To Be Famous*. (Spy Magazine)

I was extremely impressed with most aspects of the device. The human interface is designed in a way enabling one to really start working after only a two- or three-hour training period. The intuitive prompts guided me through the more complicated functions, and there is a kind of real-time feel about things which enabled me to work in a "stream of consciousness" manner. The *Store* (hard-disk storage for sound clips) is available at all times to preview any and all sound files on the various disk drives. The speed at which you can access this is amazing.

Using the companion TEAC recordable video disc as my video source, I noticed a hard-lock to video that I had never before experienced. Placement of sounds was effortless and always exactly on the desired frame. Also, while using the automation, I was able to do static fader moves on the exact frame of the scene change.

The pen and tablet system, in my opinion, has its limitations, but to me, is preferable to an intensive keystroke situation. The automation works nicely, but of course is limited to one fader or one group of faders at a time. You can go back and write other faders, however. This automation does provide some interesting possibilities for panning (i.e., following the action with the pen across a stereo effect on two grouped faders).

Figure 8. SSL's ScreenSound system.



Finally, I found the computer very fast, with no waiting time between windows. Also, there was never any loss of data (as I have found with other systems). Machine control is another plus, with ScreenSound's on-board synchronization system. I used it with a Sony DVR-10 and it was flawless.

I completed the second *Saturday Night Live* show in less time than I would have by conventional means. For fairly simple mixes, and for preparation of elements for bigger mixes (transferring to multi-track), I feel there is a real place for this device. As SSL further develops both software and hardware (there is plenty of room for growth in the system), it will become even more attractive. db

LEN FELDMAN

Lab Report

Sony Model MU-R201 2-Channel Digital Reverberator

GENERAL INFORMATION

• To understand Sony's design philosophy of the MU-R201, it is necessary first to consider just what a reverberator does. As Sony explains in their well-written owner's manual supplied with this product, the sound field experienced by listeners in a concert hall, for example, is made up of three basic components: the direct sound arrival from the sound source, the early reflections from the walls and ceiling and the later reflections, often referred to as reverberation. These three elements are always present, but their quality and ratio depends upon several factors, such as hall size, walls and ceiling shape, sound absorption characteristics of the building, decorating materials, and so on.

According to Sony, to achieve natural sounding reverberation and sound fields requires a reverberator with stereo inputs, stereo signal processing and stereo outputs. This goal is achieved in the MU-R201 by use of a two-channel digital audio processing LSI. All algorithms were developed specifically for the MU-R201 and form the basis of 10 different effect modes numbered 0 through 9. Table I summarizes the parameters available for each of these modes. The modes themselves are configured to recreate the following sound fields:

Mode 0 A large concert hall or an open air stage.
Mode 1 A small room, such as a club.
Mode 2 "Plate" reverberation, used in recording studios.
Mode 3 Multi Gate Reverb (for drum and percussion).
Mode 4 Multi Reverse Reverb (for cymbals, etc.).
Mode 5 Dual Multi Delay (for use as two independent delay devices).
Mode 6 Multi Auto Pan.

Mode 7 Dual Reverb (for use as two independent reverberators).
Mode 8 Reverb and Gate (used for vocals in Channel 1 and drums and percussion in channel 2).
Mode 9 Reverb and Delay (used, for example, for vocals in Channel 1 and keyboards in Channel 2).

Within these 10 modes, 100 presets have been created and stored, based on the suggestions of musicians, recording engineers and sound technicians from all over the world. Each preset can be selected at the touch of a couple of buttons. "Edit" and "EQ" functions can be used to change preset reverberation effects and to produce original settings. Besides the 100 factory presets, there is user memory capacity for another 100 settings created by the user. The 10 basic modes serve as a reference, augmented by such versatile effects as hall ambience, acoustically-live room effects, tightly-controlled plate reverberation, percussive multi-gate reverb and multi-reverse reverb. The dual multi delay setting offers up to 20 repeat outputs for each channel separately.

Parameter \ Mode	0	1	2	3	4	5	6	7	8	9	Range
REV. T.	0		0								0.3—99.0sec
		0									0.07—24.75sec
				0	0				2		0.1—99.0sec
								1, 2	1	1	0.90—9.90sec
						1, 2				2	1—1001msec
							1, 2				0—1001msec
PRE. DT	1, 2	1, 2	1, 2								1—733msec
				1, 2	1, 2				2		1—850msec
						1, 2	1, 2			2	0—255msec
								1, 2	1	1	1—572msec
E. RFL. T.	1, 2	1, 2	1,2,3,4								1—733msec
				1,2,3,4	1,2,3,4				2		1—850msec
						1, 2				2	0—255msec
								1,2,3,4	1	1	1—572msec
E. RFL. L.	1, 2	1, 2	1,2,3,4	1,2,3,4	1,2,3,4	1, 2		1,2,3,4	1, 2	1, 2	±.000—±.992
RT HIGH	0	0	0			1, 2	1, 2	1, 2	1	1, 2	0.01—0.99
SPREAD	0	0	0	3, 4	3, 4			0	1	1	1—16
				1, 2	1, 2				2		1—12
GATE T				0	0				2		1—300
FBL						1, 2	1, 2				±.000—±.992
REPEAT						1, 2					0—20
DEPTH							0				.000—1.000
SPEED							0				0.10—20.00Hz
EQ LOW	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	−12—12 dB
L. MID	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	−12—12 dB
H. MID	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	−12—12 dB
HIGH	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	−12—12 dB
EFF L	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	1, 2	.000—1.000
CASCADE									0	0	OFF, 1—2, 2—1

0	common for both channels
1	channel 1 only
2	channel 2 only
1, 2	separate for channels 1 and 2
3, 4	separate for channels 3 and 4
1, 2, 3, 4	separate for channels 1, 2, 3, 4

Table 1. The parameter reference chart.

In modes 3, 4, 5, 7, 8 and 9 the two channels of the MU-R201 operate separately. This permits use with two different instruments during recording or during a live performance. Up to 26 parameters can be set to design the sound field the user wants and to create special effects not available in ordinary acoustic environments. The MU-R201 also incorporates a programmable two-channel four-band equalizer. EQ data can be stored along with effects settings to provide the sound shaping characteristics required for each application.

Although our unit did not come with them, an optional remote control and an optional foot switch can be used to permit operation of the unit from any convenient location, and to increase or decrease memory numbers,

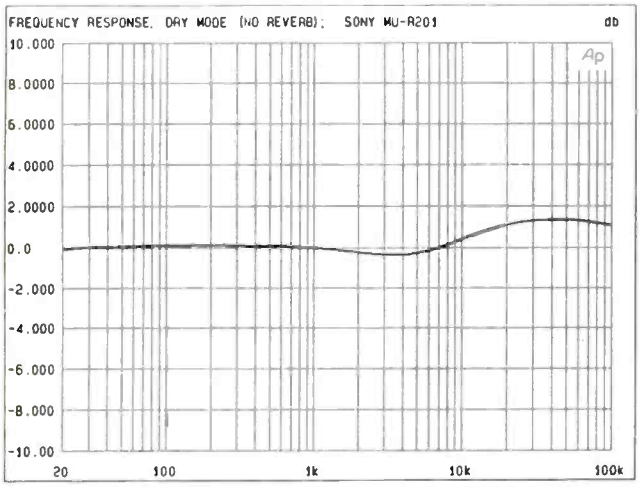


Figure 1. Frequency response. Effects control at minimum and the dry control at maximum.

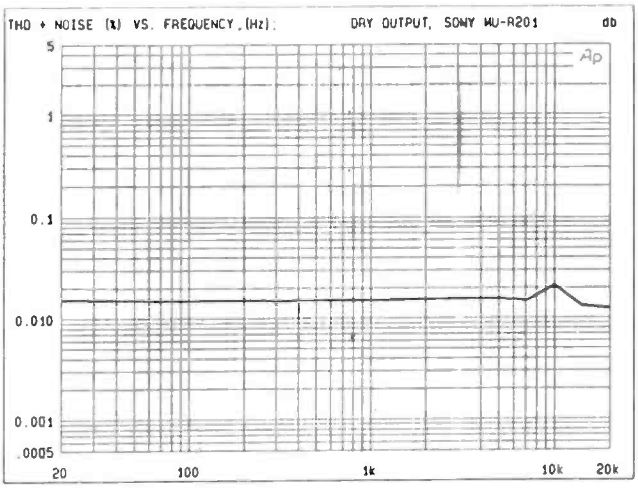


Figure 2. Harmonic distortion-plus-noise versus frequency, again with effects control at minimum and dry control at maximum.

switch effect processing off and on, and suspend effects. Two sets of input/output jacks are provided to accommodate either standard phone plugs or RCA-type phono plugs. A +4/–20 dB level selector matches the unit to professional or consumer equipment or to any musical instrument.

The integrated MIDI interface on the MU-R201 unit accepts program change signals for external selection of memory numbers during a performance. The effect level can also be controlled via the key-touch and control change signals, all of which enable the MU-R201 to be used as an effect creator for digital as well as analog musical instruments.

CONTROL LAYOUT

A combination power switch and indicator light is at the left end of the panel. A dual concentric input level control and individual "dry" and "effect" level controls come next. The settings of the dry (unprocessed signal) level and effect (processed signal) controls determine the reverberation ratio at the outputs. Above these rotary controls is an LED input level meter calibrated from –20 dB to +6 dB in seven steps. A display to the right of the level meter shows necessary operating information such as memory number, mode number, channel number and data (parameter) settings. "+" and "–" keys beneath the display are used to change parameter values or the first digit of preset memory numbers.

Six buttons perform memory functions (storage and call up) and editing functions (*Edit*, *EQ* and *MIDI* programming). Two rows of six pushbuttons each are further to the right. The action of these buttons depends on which function is selected. They can be used to call up the second digit of a preset memory number, select one of the four EQ bands for adjustment, adjust certain MIDI parameters, select mono or stereo and edit other parameters. At the extreme right of the panel is a *Bypass* button which, when depressed, feeds input signals directly to the outputs.

On the rear panel, in addition to the pairs of RCA-type and phone-plug inputs and outputs, there are MIDI in and out connectors, memory *Up* and *Down* jacks (intended for use with a foot switch), a remote connector, a *Hold On/Off* jack and a level selector switch with settings for matching the input and output of the MU-R201

Figure 3. Spectrum analysis of a 1 kHz signal passed through the Sony MU-R201. It shows residual noise and harmonic components.

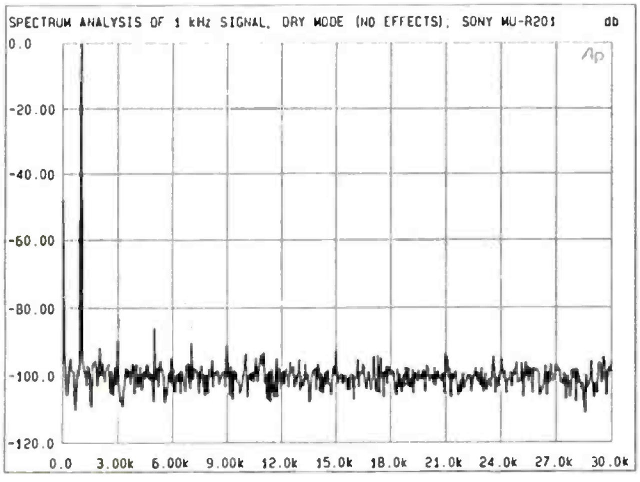
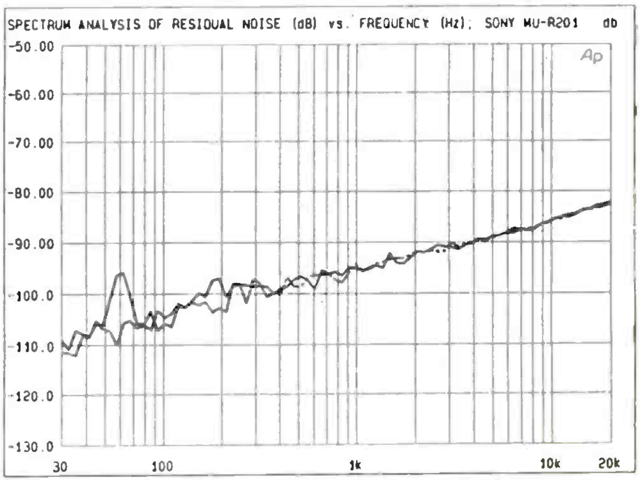


Figure 4. A spectrum analysis of the residual noise, referred to 0.5 volts input, effects control set at minimum.



The contents of memory numbers (chosen by steps 3 and 4 on the preceding page) are as follows.

- **Memory numbers 00.–09.**
  The standard settings of the ten basic modes 0–9 (described on pages 14 through 17) are stored in these numbers. The numbers correspond to the respective modes, for easy recall when creating your own effects.
- **Memory numbers 10.–64.**
  Effects mainly intended for studio recording or instrument performances.
- **Memory numbers 65.–74.**
  Effects for public-address applications and live events.
- **Memory numbers 75.–84.**
  Special effects for radio shows or live performances.
- **Memory numbers 85.–99.**
  Effects for hall simulation and creating ambience during audio playback.

Application	Memory number	Program
Standard patterns	00.	Hall
	01.	Room
	02.	Plate
	03.	Gate Reverb
	04.	Reverse Reverb
	05.	Delay
	06.	Flash Panning
	07.	Dual Reverb
	08.	Reverb & Gate
	09.	Reverb & Delay
Recording, instrument use	10.	Large Hall
	11.	Medieval Hall
	12.	Concert Hall
	13.	Piano Hall
	14.	Chamber
	15.	Pipe Organ
	16.	Brass Solo
	17.	Reggae
	18.	Shadow
	19.	Pinball
	20.	Space Echo
	21.	Compact Room
	22.	Powerful Drum
	23.	Woodwinds
	24.	Live House (1)
	25.	Live House (2)
	26.	Vivid Snare
	27.	Piano Bar
	28.	Baroque
	29.	Trumpet
	30.	Beat Sound
	31.	Snare
	32.	Clear Reverb
	33.	Electric Guitar Distortion
Recording, instrument use	34.	Flute
	35.	Lyric Piano
	36.	Mystic Strings
	37.	Cathedral
	38.	Guitar Reverb
	39.	Polyrhythm
	40.	Rock Vocal
	41.	Vibrato
	42.	Sharp Snare
	43.	Tom Beat
	44.	Kick Gate
	45.	Fancy Kick
	46.	Slip Gate
	47.	Heavy Reverse
	48.	Reverse Slip
	49.	Scratch Reverse
	50.	Single Tapped Delay
	51.	Twin Delay
	52.	Tapped Delay
	53.	Flying Delay
	54.	Panning Delay
	55.	Shadow Rhythm
	56.	Rose Piano
	57.	Auto Panning
	58.	Tremolo
	59.	Twin Plate
	60.	Tom Tom
	61.	Ethnic Drum
	62.	Cliffverb
	63.	Solo Guitar & Metal Gate
	64.	Special Beat
PA (SR) use	65.	Standard Vocal
	66.	Rock Vocal
	67.	Ballad (1)
PA (SR) use	68.	Ballad (2)
	69.	Male Vocal (1)
	70.	Male Vocal (2)
	71.	Female Vocal (1)
	72.	Female Vocal (2)
	73.	Female Vocal Ballad
	74.	Solo Synthesizer
Special effects	75.	Tunnel
	76.	Stadium
	77.	Gymnasium
	78.	Indoor Pool
	79.	Radio Play
	80.	Mecha-Voice
	81.	Cave
	82.	Mountain Echo
	83.	Karaoke (1)
	84.	Karaoke (2)
Audio use	85.	American Large Hall (1)
	86.	American Large Hall (2)
	87.	European Large Hall (1)
	88.	European Large Hall (2)
	89.	European Recital Hall
	90.	Orchestra (1)
	91.	Orchestra (2)
	92.	Chamber Music (1)
	93.	Chamber Music (2)
	94.	Opera (1)
	95.	Opera (2)
	96.	Jazz (1)
	97.	Jazz (2)
	98.	Rock (1)
	99.	Rock (2)

Table II. The 100 preset effects.

to the level of connected equipment (+4 dB for pro equipment, -20 dB for consumer equipment and electronic instruments).

LABORATORY MEASUREMENTS

In attempting to measure the electrical performance of the MU-R201, it soon became evident that this product cannot be measured the same way we would measure a preamplifier or an amplifier. If a frequency response sweep was attempted while any degree of "effects" was on, the time delay introduced between input and output resulted in an inability of the test equipment to produce a smooth sweep. Accordingly, the frequency response curve of *Figure 1* represents the response of the product with the *Dry* control fully clockwise (unprocessed signals), and the *Effect* control fully counterclockwise (to eliminate any time-delayed reverb effects). On that basis, response was reasonably flat from 20 Hz to well above 20 kHz, with a slight rise of approximately 1 dB noted at 20 kHz.

I cannot begin to calculate how many permutations and combinations of effects are possible, but they must surely number in the hundreds of thousands...

Under those same test conditions, a test of harmonic distortion-plus-noise was made for an input signal that was just below the clipping point of the unit and the frequency of the signal was varied over the audio spectrum. Results are shown in *Figure 2*. THD plus noise measured approximately 0.015 percent with a slight but insignificant rise at 10 kHz, at which frequency the reading was 0.02 percent. To isolate the noise components from the actual harmonic distortion, we conducted another test, using an FFT spectrum analysis program available on our test equipment and a fixed, 1 kHz test signal at the same level as in the previous test. Results are shown in *Figure 3* and are expressed in dB relative to the desired 1 kHz output signal which is represented by the tall spike at the left of the graph. The greatest harmonic component visible is the 5th harmonic of the fundamental, at 5 kHz, and it is some 86 dB below the reference level. Translated to a percentage, that works out to be 0.005 percent, well below Sony's claim of 0.008 percent.

When a 0.5 volt signal was applied to the inputs and the input level controls were adjusted for unity gain, A-weighted signal-to-noise ratio for either channel measured 77.3 dB below 0.5 volts. In most professional applications, inputs and outputs are likely to be greater than 0.5 volts (+4 dB equals approximately 1.25 volts which, in turn, is nearly 8 dB greater than 0.5 volts, so effectively, the S/N ratio of the unit referenced to +4 dB input and output levels would be over 85 dB). A spectrum analysis of the residual noise content was made at the same reference levels as the A-weighted overall S/N measurement and results are shown in *Figure 4*. The left channel showed a slight peak in noise at the power line frequency compared with the right channel—caused no doubt by the layout of parts for that channel. Even at that, primary noise contribution came from the higher frequencies and was not great enough to be audible under actual operating conditions.

As explained earlier, if you try to measure any of the editing or effects capabilities such as EQ bands by turning up the *Effect* knob and conducting a frequency response test, what you get, initially, for a graph of the results, are a series of spikes and dips as the tracking bandpass filter of the test equipment tries, in vain, to follow the delayed composite output signals. Our test equipment, however, has a "smoothing" function that can be introduced to replot the curve so it looks more like what you would expect. The lower plot was made with effects turned up, in Mode 2, with the EQ bands set to their midpoint. The upper two curves were made with the EQ setting at maximum boost and at maximum cut. It would have been nice if we could have "centered" all three curves at midfrequencies to be on the "zero" dB line, but having done that after conducting the first sweep (with EQ neither at boost or cut settings), we had to accept the next two curves wherever they fell. You'll have to use your imagination to "shift" those two upper curves so the lower ends of the sweeps lie atop the first sweep to get an idea of what the EQ midfrequency band can do.

CONCLUSIONS

The real test of the Sony MU-R201 came when I hooked it up in my "amateur" recording studio and began toying with the many presets and altering some of them to see what I could come up with in my own listening environment. To put it simply, the number of possible permutations possible is mind-boggling. Table II lists all 100 built-in presets and suggests what each might best be used for. Sony devotes some ten pages in their owner's manual to a more detailed description of what each of the 100 presets does. For example, they describe Preset 11 as a "Medieval Hall," explaining this setting would produce the effect of a medium size hall with fairly live characteristics such as produced by stone or glass walls. They suggest, too, that this preset setting would be suitable for wind instruments.

Preset 40, listed as "Rock Vocal," is described simply as "suitable for dazzling male or female vocal performances." And so the descriptions go—on and on—giving the user a good idea of what to use for almost any possible application and effect. While I enjoyed checking out most of the factory presets, the fun really started when I began creating my own acoustic environments, deriving them from the available presets. I cannot begin to calculate how many permutations and combinations of effects are possible, but they must surely number in the hundreds of thousands, if not the millions. Yet, for all the flexibility and versatility of this two-channel reverberator, I found it quite easy to use and adjust for any combination of effects and EQ wanted. All of these effects are, of course, done digitally, which is why they can be trimmed and adjusted to such a fine degree. Trying to do what the MU-R201 can do using analog reverberation devices would be virtually impossible. The audio world is certainly going digital, and Sony can rightly claim that they are at the forefront of digital technology, based on the performance of this and other digital products the company has developed. db

VITAL STATISTICS

SPECIFICATION	MFR'S CLAIM	db MEASURED
Quantization	16-bit Linear	Confirmed
Sampling Frequency	26 kHz	Confirmed
Standard Input Level	–20/+4 dbs	Confirmed
Frequency Response		
Direct	20Hz-20kHz,±1dB	+1,-0.3 dB
Effect	20Hz-11kHz,±1.5 dB	See Text
Dynamic Range	90 dB	85 dB
Memory Capacity		
Preset Memory	100	Confirmed
User Memory	100	Confirmed
THD	0.008%	0.005%
Power Requirements	120 V 60 Hz 28 W	Confirmed
Dimensions (WxHxD, inches)	19x1-3/4x12-58	Confirmed
Weight	9 lbs. 15 oz.	Confirmed
Price: $1050.00		

BRUCE BARTLETT

Lab Report

Digital Designs Model LS161 Studio Monitor Loudspeaker, (Upgraded for 1990)

• A recent line of small monitor speakers has attracted a lot of attention because of their accurate sound reproduction in a small package. Digital Designs makes and sells them, along with a wide variety of other pro audio gear. Designer Jassa Langford showed me the model LS161 monitor at a recent NAMM show, and I was very impressed with its sound.

The LS161 is a small, portable two-way unit of sealed-box design. It is intended to be an accurate reference monitor, wide-range except for the deep bass. To sharpen stereo imaging, the speaker is sold in mirror-image pairs, and the woofer and tweeter are mounted close together.

Digital Designs is offering an improved woofer in its 1990 model. The claimed benefits are extended low-frequency range, smoother and better-defined midrange, greater power handling and higher output. The woofer uses a 6½-in. polypropylene cone of low mass for improved transient response. Its textured, semi-hyperbolic cone shape is said to yield excellent strength and damping. A rubber surround and precise edge termination reduce nodal resonances in the cone. The voice coil is a four-layer design made of high-temperature wire wound on kapton and nomex formers.

At 3.5 kHz, the woofer crosses over to the tweeter with a 12 dB/oct. slope. The woofer's natural high-frequency roll-off is used for its crossover to minimize phase distortion in this range.

The 20 mm polyimide dome tweeter is a low-mass design with ferro-fluid cooling. A phase-correcting plate over the tweeter is meant to provide adequate acoustical loading and uniform dispersion. Supplied with the speaker is a foam diffraction ring which you can press into place around the tweeter. The foam is intended to absorb the tweeter's side radiation. This prevents re-radiation of the tweeter sound waves by baffle obstructions, and should improve the tweeter's impulse and frequency responses.

The woofer is not fused, Langford says, because fuses tend to deteriorate over time. The tweeter has a fuse which resets itself 15 minutes after opening. Langford also notes that the woofer will not bottom out. So, if you want to know when you're applying too much power to the speaker, watch for excessive cone excursions rather than listening for knocks.

On the back of the cabinet is a two-position switch labeled *accurate* and *mid-boost*. The *accurate* setting gives the flattest response, while the *mid-boost* setting gives a broad 4 dB boost centered around 1 kHz to simulate the response of the popular Yamaha NS-10M monitor speaker.

Also on the back are two gold-plated binding posts that accept bare wires or dual banana plugs. At the base of each post is a black or red ring to indicate polarity, but these rings are hard to see unless you know to look for them.

The well-crafted cabinet is made of multi-density PBC to reduce panel resonances, and has curved edges to minimize diffraction. The oak veneer finish is available in natural or black. A snap-on black grille-cloth enhances the handsome appearance of this speaker. Internal magnetic shielding permits the LS161 to be placed near video and computer monitors.

Rated frequency response is 50 Hz to 20 kHz 3 dB. Impedance is spec'd at 4 ohms, while sensitivity is rated

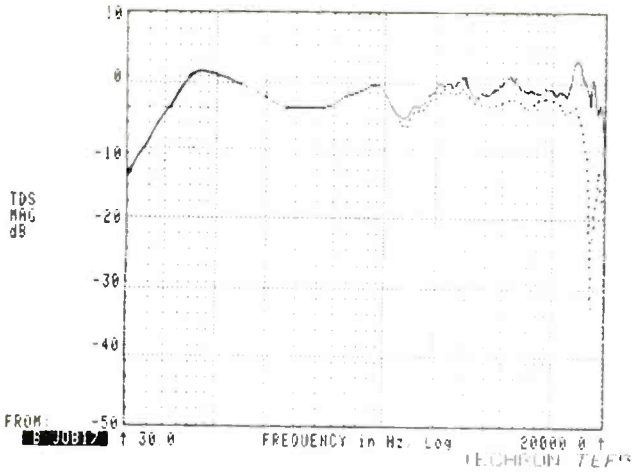


Figure 1. Frequency response, no diffraction ring. Solid line—on axis, dotted line—30°off axis.

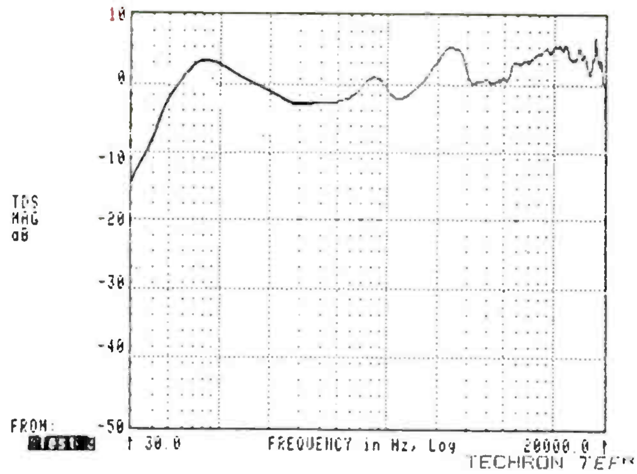


Figure 2. Frequency response of the previous model LS161.

high at 90 dB/watt/meter. Power handling is claimed to be 60 watts continuous average power and 100 watts music power. Dimensions are only 13.5-in. H x 9-in. W x 10-in. D (34.3 cm H x 22.9 cm W x 25.4 cm D), and the weight is 35 lbs./pair (15.9 kg/pair).

The manufacturer has plans for future products that will be compatible with all existing units. In the works is a passive dual 12-in. subwoofer, to be followed by an active system. Also on the drawing board are monitors with built-in bi-amplification and active signal processing.

I would not hesitate to use the LS161 for a location monitor or close-field monitor. Designer Langford is to be commended for a job well done.

HOW DO THEY SOUND?

I auditioned a pair of LS161s on a variety of program material (all compact discs and master tapes) before making any measurements. I placed them in a close-field arrangement on top of a console, about one meter apart and one meter from me, without the grille. When I angled the LS161s inward to aim at me, they sounded too bright. When aimed straight ahead, they sounded more natural, so I did most of the listening tests that way. This aiming also made the sound more spacious, since it reduced the direct-sound level and increased the reflected-sound level at my ears.

Since the *mid-boost* setting sounded a little honky or nasal on most material, I used the *accurate* setting for the listening tests.

The LS161 sounds much bigger than it looks! It's loud, with plenty of bass. For example, on the first cut on Tracy Chapman's *Crossroads* album, the weight of the bass reproduced by these tiny speakers is amazing.

The overall tonal balance seems just right. Acoustic guitars, such as those heard on *Never Go Home Again* by Fleetwood Mac, are neither edgy nor dull—just delicate. They seem to float in front of the speakers. On some recordings with lots of high-frequency content, the speakers sound bright or forward. A good example is *Let It Roll* by Little Feat.

Do the diffraction rings help? To me, they make recordings sound more like speakers and less like music. It's a subtle effect; others might have a different opinion. Since the rings roll off the highs off axis, they can be used to tame the high end if you think it's too bright.

Vocal quality depends on the recording. On Fleetwood Mac's *Songbird*, or on folk recordings by Kate Wolf, the voice is smooth and integrated. It's neither tubby nor sibilant. But on Tracy Chapman's *Crossroads*, sibilants are harsh if you listen at a high volume. That's okay: a good monitor reveals the differences among recordings, rather than imparting the same coloration to all of them. Similarly, Phil Collins' vocal on *Rain Down on Me* sounds pinched, but sounds more natural on *Another Day in Paradise*.

Details and sonic flaws are clearly revealed by the LS161. On the recording just mentioned, you can easily hear gated reverb on the snare, pick out harmony vocal lines, etc. Different instruments are clearly delineated.

I was surprised how accurately this speaker reproduces classical-music recordings. It never sounds boxy or bloated. On Telarc's recording of Stravinsky's *Firebird*, all timbres are natural—not thin as I heard on the previous model. Strings sound smooth and sweet, never edgy. Pizzicatos are palpable. Triangle and cymbals sound crisp and airy. The fullness of the bass belies the size of these little monitors. Brass sounds just right, neither tizzy nor muffled. Instruments are reproduced with great detail.

The bass-drum roll at the beginning of the *Firebird* is nearly absent, as you'd expect with speakers this small. They strain or distort on heavy bass-drum beats. Still, they have impressive impact for their size.

Imaging is very sharply focused, such as heard on the album *Second Stage*, a compilation by recording engineer John Eargle. On pop recordings, I heard imaging tricks I never heard before, such as woodblocks recorded in opposite polarity on Ricardo Silveira's *Sky Light* album. Imaging is sharper when the speakers are oriented vertically.

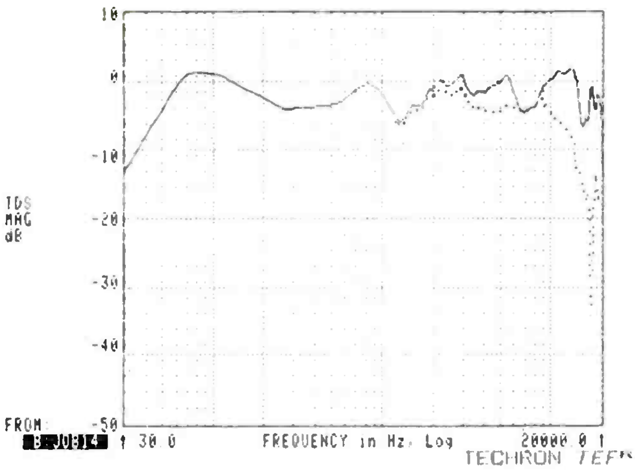


Figure 3. Frequency response with the diffraction ring. Solid line—on axis, dotted line—30°off axis.

Tight bass is another attribute of these speakers. For example, on *Big Notes* by Flim and the BBs, electric bass is full, but tight, with no overhang. All instruments are detailed without being edgy. On this and other recordings, kick drum has impressive punch. You may very well be satisfied with these speakers without adding a subwoofer.

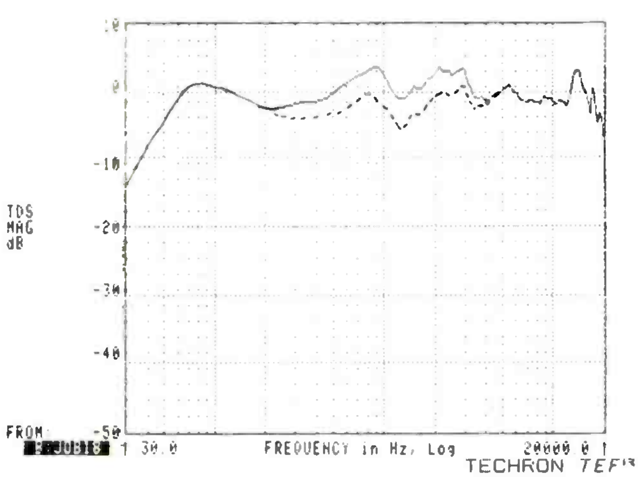
On *Chorus*, by Eberhard Weber, the bass and kick drum sound articulate and tight, never muddy. When trombones join in, however, the sound becomes slightly bloated at high levels; the LS161 has some trouble handling high-level bass, but this is normal for such a small speaker.

I listened to several of my own recordings over the LS161. The balance is similar to what I originally set up over wide-range large studio monitors, except the deep bass is weaker. The LS161s also reveal mic'ing distance. I heard some things I didn't like about my recordings that I hadn't heard over larger monitors, but this shows the LS161 does its job well.

HOW DOES IT MEASURE?

I measured the LS161 with a Techron TEF System 12 audio analyzer/computer, Bruel & Kjaer 2235 sound

Figure 4. On-axis frequency response. Solid line—mid-boost setting, dashed line—flat setting.



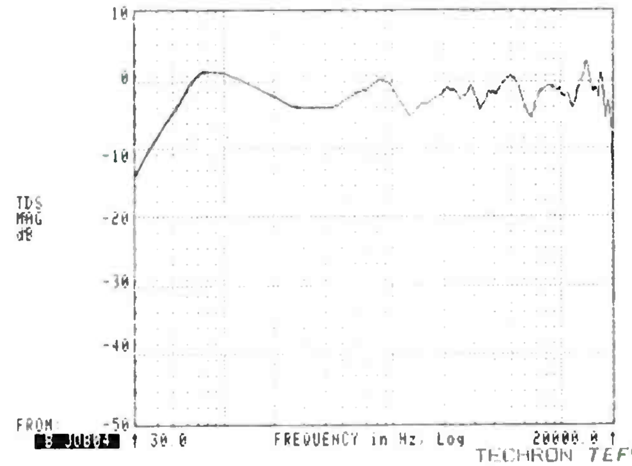


Figure 5. On-axis frequency response with the grill, but no diffraction ring. Compare to Figure 1.

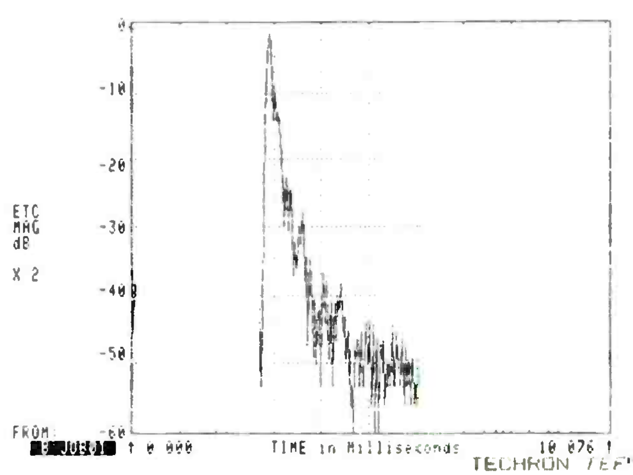
level meter, Bruel & Kjaer 4165 ½-in. laboratory-calibrated condenser microphone and Crown D-75 power amplifier.

The measurement microphone was placed one meter away with the microphone midway between the tweeter and the woofer edge. To avoid room reflections, the TEF was set for a one-meter space window which provided anechoic measurements accurate down to 333 Hz. I measured low-frequency response with the mic very close to the woofer, which resulted in the anechoic half-space response. Finally, I spliced the two curves together to get the full-range frequency response.

For all these measurements, the speaker's tone switch was set to "accurate" and the grille was removed, except as noted.

Figure 1 shows the frequency response without the diffraction ring. Neglecting the 4 dB bump at 13 kHz, the on-axis response is an excellent 2.5 dB from 52 Hz to 18 kHz. This accounts for the speaker's neutral sound character. At 30 degrees off-axis, the response drops 3 dB at 5 kHz, but is still quite smooth. You can easily angle the speaker to provide the desired high-frequency balance at the mix position.

Figure 6. The energy time curve, no grille or diffraction ring.



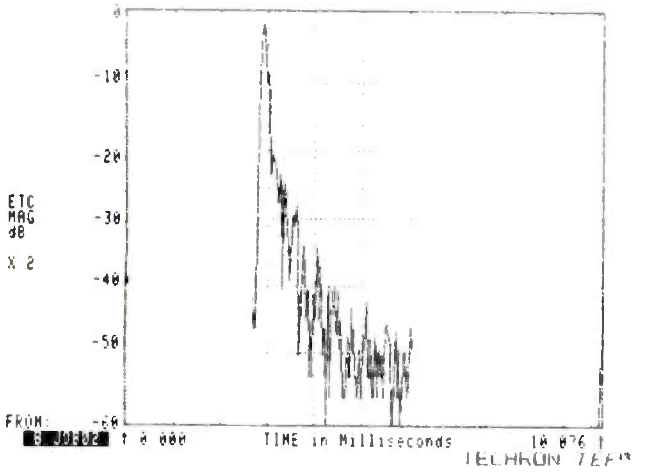


Figure 7. Energy time curve, no grille, but with the diffraction ring. Compare to Figure 6.

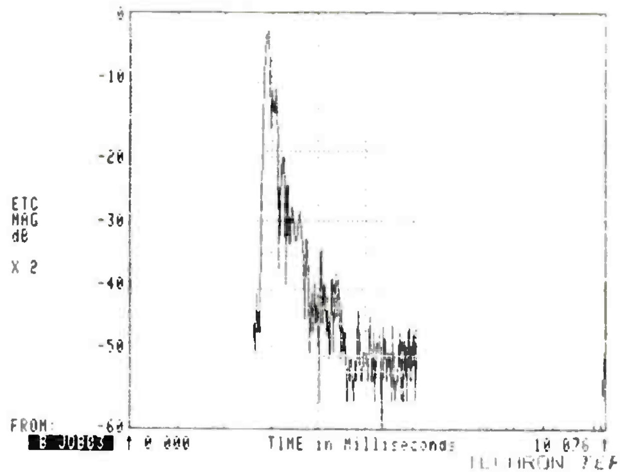


Figure 8. Energy time curve, with the grille, but no diffraction ring. Again, compare to Figure 6.

Note the low-frequency response was measured in half-space (2 π steradians), which approximates a mixing-console surface. The low end would be flatter if the speaker were moved into open space (4 π steradians).

The response of this new model is better than that of the previous model LS161. As shown in *Figure 2*, previous units had a 5 dB bump at 2.3 kHz, a rising characteristic, and an overall response of 50 Hz to 18 kHz 3 dB. Thus the newer model has been improved significantly.

Figure 3 is the frequency response of the 1990 model with the foam diffraction ring added around the tweeter. The ring slightly roughens the response and tightens the high-frequency dispersion. A different foam porosity or tapered cutout might provide a smoother response.

As shown in *Figure 4*, the mid-boost switch setting provides a broad 4 dB boost centered at about 1 kHz. This measured response correlates well with the nasal or honky effect heard in the listening tests.

However, I doubt I've heard such an accurate, well-balanced, close-field monitor in such a compact package, and at a bargain price.

Finally, in *Figure 5*, we see the response with the grille in place. As expected, the response is rougher with the grille, so it is better left off for studio use. Still, the overall trend is uniform.

Turning to the time domain, we see an excellent Energy Time Curve in *Figure 6*. The grille and diffraction ring are removed. In *Figure 7*, the diffraction ring is installed, and it makes the direct-sound spike even sharper. Finally, *Figure 8* shows that the grille degrades the ETC slightly.

The group delay (not shown) was very uniform, remaining under 0.5 msec from 333 Hz up. This excellent time performance correlates well with the tight, clear sound heard in the listening tests.

SUMMARY

The Digital Designs LS161 monitor is remarkably clear and articulate in all frequency ranges. Bass is tight, never boomy or tubby. So if your mix sounds muddy through these speakers, it's probably the recording, and not the speakers.

Since the LS161 sounds like many audiophile bookshelf speakers, mixes made on it should hold up well on upscale home speakers.

This speaker is very revealing of mic techniques and stereo placement. If your stereo recordings have vague imaging when heard over a pair of LS161s, the problem is in your mic'ing technique and not the speakers.

As for drawbacks, some listeners might complain of a slightly forward or bright tonal balance, but this is partly a matter of taste. The LS161 has weak deep bass like other small speakers. When mixing with the LS161, you should expect this effect and NOT boost the deep bass until it sounds right. You might want to add a subwoofer, but the speaker sounds warm rather than thin, so most of the important bass information is there. Like other small speakers, the LS161 distorts with high-level low frequencies, so it is not recommended for extremely loud monitoring.

However, I doubt I've heard such an accurate, well-balanced, close-field monitor in such a compact package, and at a bargain price. It looks great and is ruggedly built. The overall character is neutral. My unit came without an instruction manual, but it did have a spec sheet.

I would not hesitate to use the LS161 for a location monitor or close-field monitor. Designer Langford is to be commended for a job well done.

If you have a pre-1990 pair of LS161 monitors, you can send in your woofers and get an upgraded pair for $43. I highly recommend the woofer upgrade. You can either install them yourself, or have Digital Designs (or another dealer) install them for a nominal fee. The limited warranty extends for one year on labor and three years on parts. The LS161 costs $349.00-$367.00 a pair. db

RANDY HOFFNER

Broadcast Audio

Placing Sound in the Television Picture—The Old Fashioned Way

• A recent clean-up drive around the NBC Engineering Development offices turned up some interesting files. They deal with the inclusion of an emergency audio signal that formerly accompanied the video signal sent out on the inter-city network circuits that carried NBC's programming to its affiliates. That emergency audio system was developed in 1960, and in addition to having historical interest, it illustrates once again that there is nothing new under the sun. It also reminds us how far television network audio has come since then.

In the days before the networks delivered programming to their affiliates via satellite, this function was performed by the telephone company (and for most of those years, I do mean *the* telephone company). From about 1978 forward, the telephone companies have used diplexed video and audio (VANDA) circuits: the audio signals are carried as frequency modulated subcarriers on the same transmission path as the video.

This state of affairs has not always existed. Prior to diplexing, television picture and sound components were carried by the telephone company over separate facilities; different in the type of physical apparatus employed, in their characteristics, and even in their routings. An example of such differences was that long inter-city video circuits were one-way paths, while the audio circuits were reversible by means of a series of relays operating in tandem at the repeaters along the way.

These differences in equipment and routings gave rise to regular interruptions of the audio signals from the network control center to the affiliate broadcasting stations. Because of the nature of the circuits and the equipment used for them, audio failures were far more frequent than video failures. It can be seen that one failure mode of the relay-reversible audio circuits was that spurious voltages could appear along the various lines producing unwanted relay actuation, and thus a break in audio continuity.

In response to the audio interruption problems, NBC Engineering considered various schemes of varying degrees of complexity to provide an emergency audio signal that would be available when the normal inter-city network audio feed failed.

They selected a system they called "interleaved sound," which sandwiched a sound signal into the picture. The interleaved sound process was described as being similar to sliding a sheet of paper between two pages of a book while the book remains intact. The video signal containing the emergency audio signal is carried on the same inter-city path as before, with no additional bandwidth being required and no special attention needed.

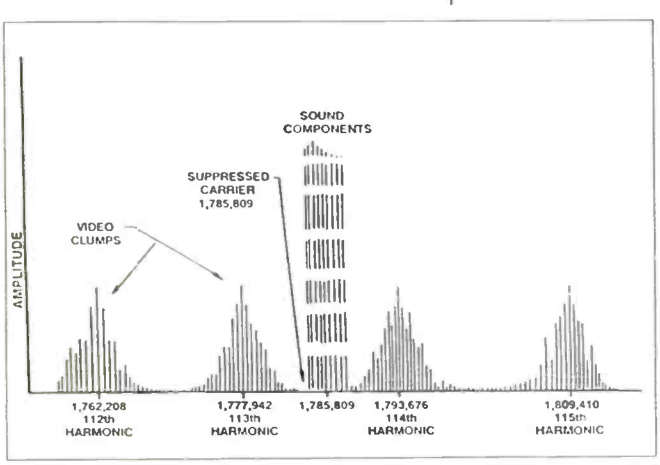


Figure 1. Video energy concentrations and interleaved sound.

It has been known for some time that television scanning of picture material produces concentrations, or *clumps* of energy distributed throughout the video spectrum at harmonics of the horizontal line frequency (15,734 Hz for NTSC video), each having sideband components at multiples of the field scanning frequency (59.94 Hz for NTSC video).

At the points midway between clumps, there is little video energy, and it was thought that there would be sufficient space to fit an interleaved sound channel into one of these gaps. *Figure 1* is taken from the description of interleaved sound, and diagrammatically illustrates some of the video energy bunches, including the area selected for placement of the interleaved audio signal. The selected location was the midpoint between the 113th and 114th harmonics of the horizontal scanning frequency. The interleaved sound signal was thus centered at 1.7858 MHz. Although any other gap in the same general area of the video spectrum would have served, the general area itself was quite carefully selected.

If an area appreciably lower in frequency is considered, such as around the 80th harmonic or around 1.2 MHz, it is likely that considerably more video energy would be encountered, producing greater interference to the recovered sound. If a higher-frequency area, such as around the 140th harmonic or around 2.2 MHz were chosen, chrominance components would interfere with the audio signal.

Because the presence of a large amount of high-contrast fine detail can narrow the gap between two energy clumps, it was desired to make the interleaved audio signal occupy as narrow a portion of spectrum as possible. For this reason, the audio signal was carried as a single sideband, suppressed-carrier transmission. To avoid visual interference with the television picture, the amplitude of the interleaved sound signal was kept as low as possible. The normal level of the signal was about one percent of the peak picture amplitude, which was considered adequate for filling short audio interruptions.

It was proposed that when prolonged use was made of the emergency audio signal, its amplitude would be increased fourfold, improving the signal-to-noise ratio of the recovered sound signal by 12 dB, still without causing visible picture interference.

NBC asked the FCC for permission to test the interleaved sound system in normal operation, and permission was given to conduct such tests on the circuit between New York and Washington for the month of September, 1960. Successful completion of these tests eventually resulted in the use of interleaved sound between NBC New York and NBC Burbank beginning in December, 1961.

A report of test observations on the use of interleaved sound between New York and Burbank for the period December 19, 1961, to June 19, 1962, reported that during that six-month period, there were twelve audio outages, with the interleaved sound signal being used to restore audio continuity each time.

The success of those tests led to NBC implementing the interleaved sound system on its network feed. Interleaved sound remained on the NBC feed until 1978, when telephone company video/audio circuits became diplexed, obviating the need for it.

The reported quality of the interleaved sound was "adequate for emergency use." Frequency response was reported as 100 Hz to 4.3 kHz, harmonic distortion below 3 percent, and signal-to-noise ratio 27 to 46 dB below program level depending upon the nature of the video signals. This performance is definitely poor by today's standards, but "listening comparisons between the derived sound and the sound transmitted by the common carrier reveal little difference in fidelity." This observation points up the fact that in that era, longline television network audio was really not very good.

We have a far better situation today, with wideband, high-quality stereo television audio being routinely transmitted over both satellite links and common carriers. It is interesting to note what problems the television engineers of thirty years ago were addressing, and how they solved them.

Now it is time to return to the television audio problems of today, such as how to incorporate eight channels of digital audio, frequency range 20 Hz to 20 kHz, into the digital video bitstream! db

BRENT HARSHBARGER

Audio for the Church

• In the July/August issue, I discussed the five parameters for good sound system design. They are:

1) Coverage
2) Bandwidth
3) Level
4) Gain Before Feedback

5) Intelligibility (the last and most important ingredient in church audio).

If you remember from my previous article, we wanted the coverage of the sound system to provide no more or less than 3 dB of SPL to any seat in the house. The bandwidth was the second parameter that had to be determined; it would be related to the type of service you participate in. If you were used to a large music program, you would require a wider bandwidth or frequency response than a worship service that was primarily voice. The desired level would also be determined primarily around the type of worship. Using the parameter above, we then need to know if our system will have enough gain before it feeds back. Finally, intelligibility is how well we will be able to understand what is being said.

Thus far I have discussed electroacoustics, but for the remainder of this article, I will discuss the hidden aspects of a good sound system— the acoustics.

The acoustic environment is the most misused parameter in a good sound system design. Many technicians try electronics to fix acoustic problems, which only makes matters worse. The acoustic environment should first be optimized, and then use electronics to enhance the system, instead of destroying it.

To start our discussion on acoustics, I will use the analogy of taking a raw speaker and playing music through it, which would not yield a pleasing result. If we took the same speaker and put it into a speaker cabinet, it would have a pleasing fidelity. The change is due to resonance frequencies created by the speaker cabinet's dimensions. The same thing happens when we put a speaker system into a church, or a bigger box.

As the sound waves move back and forth in the church, this creates additions and cancellations of sound waves. These additions and cancellations are known as room modes. A classic case of someone using electronics to fix the acoustic environment is in working with room modes. If a technician is trying to equalize the sound system with a Real Time Analyzer, he may see a lower frequency that is drastically down, compared to all the frequencies around it. He would boost that frequency to an extreme before seeing a difference, if any, on the RTA. What he is really experiencing is a room mode cancellation and not a true response of the sound system's speaker. If the technician moved the RTAs mic several feet, this frequency would be equal with the frequencies surrounding it, providing he had the equalizer in its original position. You can reduce the number of room modes by reducing the amount of parallel surfaces. Therefore, square-, rectangular- and octagon-shaped rooms are not recommended for a church's audio rooms.

As I stated in the July/August issue, church sound systems are quite complicated because they really require two different sound systems, and for many of the same reasons, the acoustic trearment is no different. When dealing with acoustics and comparing good acoustics for speech and music, you could take a piece of paper, for example, and divide it in half, labeling the top of one side as Speech, while writing Music as the heading on the other side. Then list everything that would make good acoustics for speech, and write the complete opposite on the side representing good acoustics for music, and you would be correct. For example, speech requires reverberation time (RT60) of 1.5 seconds or less, and music requires RT60 of greater than 1.5 seconds. To create a good acoustic environment for churches takes a solid understanding of acoustics, because putting a good sound system in a poor acoustic environment will not gain you much of an advantage.

SPEECH INTELLIGIBILITY

The primary concern in working with acoustics in the church is speech intelligibility, because it is the most difficult to achieve. Reverberation can deteriorate speech intelligibility. As someone speaks, the sound travels through the room and hits a surface, such as a wall, and bounces back in the room, or hits another surface. The sound bounces (or reflects) at the same angle it struck a surface. Reflections that arrive within the first 40 milliseconds at the listener's ears are called early reflections, but they are too close for the brain to detect a difference. Reflections start to interfere with speech when they arrive later than 40 milliseconds. The brain realizes there is a delay from the direct sound coming from the speaker, and this delay can cause syllables to run together, which will start to reduce intelligibility. As the reflections get farther in time, they can end up blurring words together.

Although reverberation can reduce intelligibility, no reverberation is not optimum either. Therefore, reverberation for speech should be between 0.5 and 1.5 seconds. Music sounding full and robust likes to have a reverberation time starting at 1.5 seconds on up to 3.0 seconds. Many late reflections are a desirable effect with symphonies and organ music. However, today's electronic music could be less reverberant because each instrument, or voice, can be electronically enhanced for special effect.

Many variables make up reverberation. One of them, such as room modes, are the dimensions of the sanctuary. The other is the type of surfaces in the room. Every type of material is either reflective or abortive, and each type of material varies that state. For example, acoustic foam is more abortive than most carpet without foam, and these will have a great effect on the reverb time. While we are discussing acoustics and acoustic foam, I would like to clear up the myth that many people use acoustic foam to "sound proof" a room. Acoustic foam only works on mid to higher frequencies. Lower frequencies will still pass through, which can make a room muddy-sounding. A dense wall or materials such as sand- filled cinder blocks are certain for stopping the lower frequencies.

REVERBERATION

Reverberation time can be complicated as well, because it varies at different frequencies, and figuring them "long hand" is a time-consuming mathematic headache. Fortunately, several of the sound system design programs available today can do RT60 calculations in a fraction of the time. I strongly suggest that anyone constructing a new building, renovating the existing, or just upgrading their church sound system, should consult with a qualified sound contractor, engineer, or acoustical consultant who uses an acoustical CAD or sound system design program—it will save you much more than you spend (see my July/August article for more details).

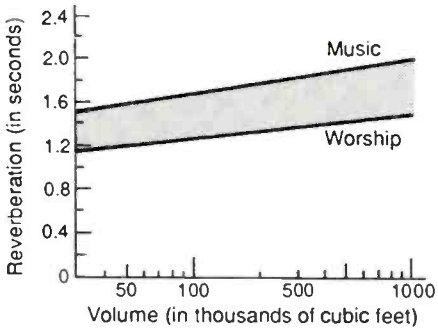


Figure 1. This shows dimensions in total cubic feet, and reverb for calculating the RT60 time in seconds.

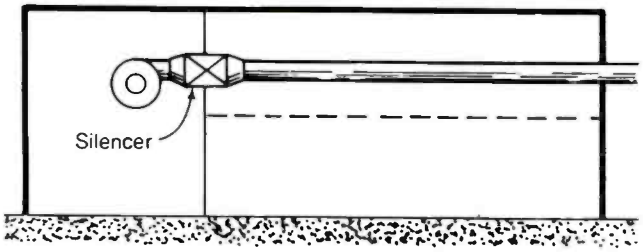
THE CORRECT RT60 FOR A CHURCH

Now the big question—what is a good RT60 for a church? The easiest way to determine that is by the room's size. *Figure 1* shows dimensions in total cubic feet, and reverb time in seconds. The bottom line is optimum for a church that relies greatly on speech, and the top line is optimum for music. Again, you determine the amount of RT60 by the type of worship you are used to. For example, a church with 400,000 cubic feet that relies on a good, strong music program would have its RT60 at about 1.8 seconds. As you can see from the chart, it lands slightly higher than where a middle line would be drawn. This reverb time would be adequate for speech without losing the spaciousness needed for orchestral-style music.

The biggest enemy to good acoustics and speech intelligibility is *noise*, or ambient noise. Firetruck sirens, screeching tires, rain and thunder are all types of noise, but the worst offender to speech intelligibility, in my opinion, is air handling noise from your Heating, Ventilation and Air Conditioning system. These noises push air in the frequency range that is important to speech intelligibility. To eliminate these noises, get the air handling system as quiet as economically possible. This may be done by putting the heating and air conditioning unit on the ground instead of on top of the building. This will help lower frequency rumble. Use an isolated room away from the sanctuary for the air handling unit, and use distributed duct work with branches. Aim the vents so they dump air instead of throwing it. Also, use opposed blade damper instead of the less expensive butterfly type, and place the dampers considerably upstream from the vents. Lastly, you can put silencers in the duct work by placing them in the wall of your isolation room to the duct work that feeds the sanctuary (See *Figure 2*).

When using a sound system, you need to have a minimum of 25 dB of signal-to-noise ratio for good speech intelligibility. For example, if your ambient noise is 40 dB, your sound system should be putting out a sound pressure level of 65 dB. A word of caution—this works until you reach 90 dB. After reaching that level, your intelligibility will start to drop anyway. The level of noise in a good acoustic environment for a church should be no greater than 42 dB, using an A weight scale on a SPL meter, and if it is, then you can almost be guaranteed that your HVAC system needs some adjusting, although other factors can apply, such as not enough isolation from the outside walls.

Figure 2. An HVAC silencer is shown in its proper position for reducing air-handling noise.



SO DID WE LEARN ANYTHING?

Sure we did. We learned that all rooms have resonances and we can reduce them by reducing parallel surfaces. We learned what is the optimum RT60 for a church, by size and type of worship. We found out how to reduce HVAC noise, and what the maximum noise level for churches should be for good acoustics.

Next time we will take a trip to digital land and see the very near future of, not tomorrow's technology, but *today's*. db

New Products

TRUE PIANO SOUNDS

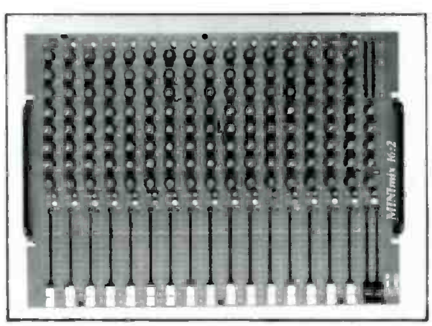
• Proformance/1 and Proformance/1+, are a family of 16-bit true stereo piano modules. Proformance represents the first and only sampled piano module to offer true stereo samples. Proformance offers a greater-than 90 dB signal-to-noise ratio and true, phase coherent stereo samples processed and transferred directly from the Emulator III. Not simply mono samples panned left and right, Proformance contains actual stereo recordings that precisely capture the expereince of sitting at the keyboard of a concert grand piano. All the sonic complexity and rich spatial resonance that defines a piano's sound is accurately reproduced. Proformance/1 gives a variety of piano sounds to choose from. The Proformance/1+ adds electric pianos, organs, vibes and acoustic and electric basses.
Manufacturer: E-mu Systems, Inc.
Price: $499.00
Circle 60 on Reader Service Card

TRIPOINT MICROPHONE

• The "TriPoint" microphone (Model RD303), offers a new look to the old problem of clustered mics for multiple audio feeds. The TriPoint has the smooth, extended response of a condenser microphone, with a symmetrical unidirectional polar pattern and incorporates three miniature permanently-polarized condensers in a single, inconspicuous case. Separate feeds, such as broadcast, house PA and back-up, are easily accomplished in a design that virtually disappears on-camera.

Operable with three 9 V alkaline batteries or 9-52 V DC phantom power, the TriPoint features 30-20,000 Hz frequency response, 130 dB maximum input sound pressure level, 100 ohm balanced output (with power module) and durable turned brass housing with low-reflectance finish.
Manufacturer: Audio-Technica
Price: $1,660.00
Circle 61 on Reader Service Card

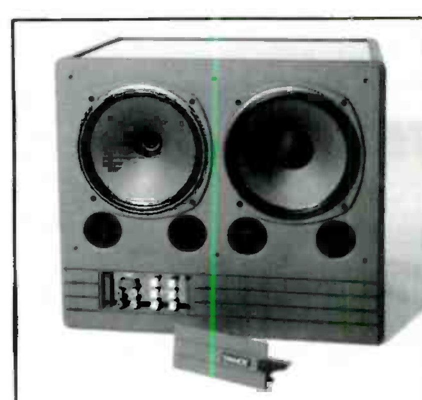
SMALL CONSOLE



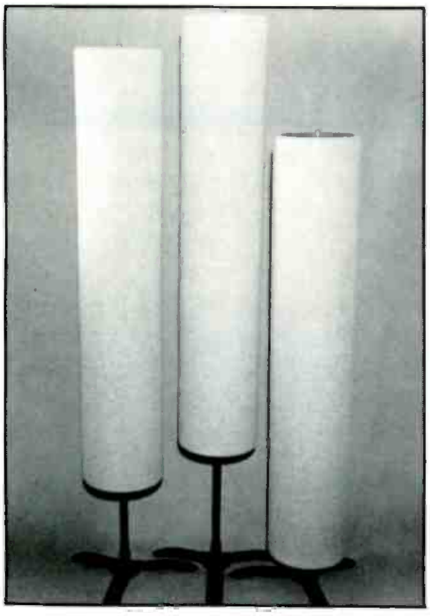
• The MINImix 16:2 satisfies the demand for a console providing 16 mic and 16 line level inputs with four auxiliary sends in a 19-inch rack mount format and 8U space. Applications include stage keyboard or electronic drum monitoring, live sound reinforcement, confrerence use, installations and location recording/broadcasting.
Manufacturer: Hill Audio, Inc.
Price: $1,699.00
Circle 62 on Reader Service Card

STUDIO MONITORS

• This new line of studio monitors consists of the System 2 NFM, System 8 NFM, System 10 DMT, System 12 DMT, System 15 DMT and the System 215 DMT. All feature biwired terminal panels, hard-wired crossovers, polymide wire insulation and magnetic gap coolant for improved thermal conductivity and power handling. The Dual Concentric models feature gold-plated contacts, a cast or moulded waveguide, and sculpted, twin-ducted ports. System 10 DMT through System 215 DMT feature vented chassis', a specially-designed diffraction ring (ensures undisturbed HF wavefront and prevents HF diffraction at junction of LF surround and cabinet edge), adjustable HF level and superior crossover topography and components wired with Vanden Hul cable.
Manufacturer: Tannoy
Price: from $550.00/pair to $5,500/pair
Circle 63 on Reader Service Card



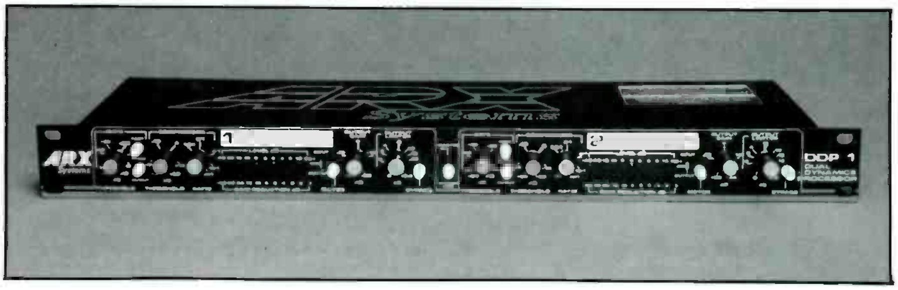
NEW TUBE TRAP



• The Studio Trap is a portable, self-contained freestanding 9-inch diameter Tube Trap, tripod-mounted with floating suspension for easy height and rotational adjustments. Studio Traps are tuneable, featuring adjustable diffusion. A built-in, ASC-patented Sound Diffusion Panel diffuses sound above 440 Hz. Half of the Studio Trap surface is midrange reflective, the other half absorptive. Rotating the Trap turns the panel into or away from the sound field changing its brightness. Studio Traps may be lowered to 6 inches from the floor and as high as 6-1/2 feet to treat standup or sitdown talent.

Manufacturer: Acoustic Sciences Corporation
Price: $255.00/unit
Circle 64 on Reader Service Card

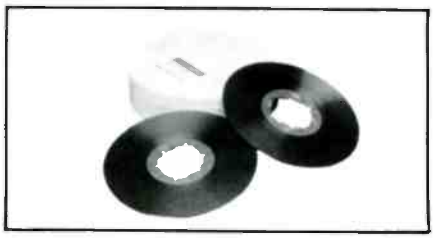
DUAL-CHANNEL PROCESSOR



• The DDP 1 is a dual channel multi-mode dynamics processor, with three independent functions per channel. Housed in an all-steel one rack unit chassis, features include an easy-to-use noise gate with switchable fast or slow release, a variable ratio compressor from 1:1 to infinity:1 and an independent peak limiter following after the output gain. A highlight of the DDP 1 is its comprehensive metering, with 27 LEDs per channel covering input and output levels, gain reduction metering and status indicators. Each channel has its own hardwire bypass switch, and a stereo link switch for master/slave operation. Other features include true differential balanced inputs and outputs with very high CMRR, sliding bias VCA and detector loop insert for frequency sensitive compression. The DDP 1 has been designed for use in all professional audio applications. It is equally at home in studio and sound reinforcement, with specifications to suit the exacting requirements of Digital Audio.

Manufacturer: ARX Systems
Price: to be announced
Circle 65 on Reader Service Card

DUPLICATING TAPE



• Ampex 617 C-60 and Ampex 618 C-90 Audio Cassette Duplicator Tape have been introduced. Both are Type I High, Performance-Extended Frequency Response tapes. Ampex 617 and 618 utilize high performance magnetic particles for frequency response. In addition, the oxide and binder system provides electromagnet characteristics. The oxide is processed to assure consistent and uniform dispersion of magnetic particles for high quality duplication. Both products feature interlocking, symmetrical hubs which are color coded for C-60 and C-90 tape thicknesses. Hubs feature a larger flange.

Manufacturer: Ampex Recording Media Corporation
Price: N/A
Circle 66 on Reader Service Card

BACKGROUND MUSIC CONTROLER

• Designed for background music installations, the Pro-PT5R Preceiver, Control Center/Preamplifier/Tuner combines performance and features of Soundcraftsmen separate tuner and preamplifers. The approach allows the cost saving of combined preamplifier and tuner stages on a single chassis. It has Feather-Touch Digital CMOS Switching, FET preamp, source selections including CD/DAT, Phono, Tuner (built-in), Audio/Video, plus two Tape Monitors with dubbing. Unique Spectral Gradient circuit eliminates high frequency harshness found on many compact discs. Two pairs of line outputs are provided for Surround Sound applications or Subwoofer/Satellite speaker systems. A Variable Contour Loudness Control enables precise and easy selection of frequency balance at any volume level. A wireless Remote Control is included and allows remote selection of volume up/down, tuner memories, input selectors, audio mute and power. The Digital PLL Tuner Section features a Micro-Computer memory system, coupled with Automatic Scanning, to allow extremely simple programming of 16 stations.. The rear panel provides two switched and one unswitched AC convenience outlets.

Manufacturer: Soundcraftsmen
Price: $599.00
Circle 67 on Reader Service Card

On the pages that follow, we present this issue's Buyer's Guide on equalizers, crossovers, delays, and reverbs. Other signal processing equipment will appear in the next issue. The information contained is supplied by the respective manufacturers. Further, if a manufacturer that you seek is not listed, the chances are strong that, as many times as we tried, we could not get information from them.

*Editorial Note: Only units whose sole function is creating reverb have been classified as "Reverbs." Units which utilize algorithmic (software) representations of several signal processing functions (which usually include reverb), have more accurately been labeled as "Multi-Effects Processors."

MULTI-EFFECTS PROCESSORS

Alesis Studio Electronics

It has fully programmable simultaneous digital effects processor capable of creating up to four effects simultaneously, all at 20 kHz bandwidth. Effects include stereo reverb, stereo delay, stereo chorusing, stereo flanging, phase shifting, pitch detuning, up to 5 bands of parametric EQ and 11 bands of graphic EQ. Also, 1.5 seconds of sampling, programmable multi-tap delays, programmable panning, resonators and ring modulators. MIDI implementation includes real-time modulation of up to 8 parameters simultaneously. 100 programs. 90 factory presets.
Dimensions: 1.75-in. x 19-in. x 7-in.
Weight: 4.5 lbs.
Price: $499.00
MIDIVERB III has preset/programmable simultaneous 16 bit digital effects processor capable of generating four effects including Reverb, Delay, EQ and Chorus or Flange. 15 kHz bandwidth, 200 memory locations: 100 non-volatile factory presets, 100 user-programmable memory locations. Real-Time MIDI parameter control. MIDI mapping and system exclusive program dump capability. Bypass footswitch jack.
Dimensions: 1.75-in. x 19-in. x 7-in.
Weight: 4.5 lbs.
Price: $349.00
MICROVERB III is a preset digital effects processor featuring 256 all new 16 bit reverb, delay, gated/reverse reverb, multi-tap delay and special effects programs. Each program can be fine-tuned by pre-emphasizing or filtering out frequencies with the High and Low EQ controls before signal is effected. These shelving filters are at 100 Hz and 10 kHz with 10 dB of boost/cut for wide ranging tonal variations of the effects. The unit also features stereo input on ¼-in. jacks, and 15 kHz bandwidth.
Dimensions: 1.75-in. x 19-in. x 4-in.
Price: $249.00

Applied Research and Technology —See our ad on page 5

The SGE MACH II is a stereo multi-effects processor that will do 12 simultaneous audio functions with up to 20 KHz bandwidth. It features over 70 different effects.
Dimensions: 1.75 x 19-in. x 9-in.
Weight: 11 lbs.
Price:$749.00
The DR-X is a new stereo studio processor that does 10 simultaneous effects including an exciter, compressor, limiter, noise gate and more than 50 other effects.
Dimensions: 1.75 x 19-in. x 9-in.
Weight: 11 lbs.
Price: $629.00
The MULTIVERB III will do 4 simultaneous effects from a choice of over 50 and like the SGE and DR-X offers a MIDI DATA MONITOR for actual readout of the MIDI data stream.
Dimensions: 1.75 x 19-in. x 9-in.
Weight: 11 lbs.
Price: $529.00
The LT will offer the power of the MULTIVERB III with a greater complexity of programs. It will do 3 simultaneous effects and has 192 studio created patches.
Dimensions: 1.75 x 19-in. x 9-in.
Weight: 11 lbs.
Price: $299.00

BGW Systems

SPA-3 Signal Processing Amplifier is a complete built-to-order Signal Processing System with three amplifiers, built-in electronic crossovers, horn equalization, parametric EQ, all-pass filter delays, precision digitized attenuators, active balanced input, high-frequency propagation loss compensation and system configuration via plug-in jumper network.
Dimensions: 5.25-in. x 19-in. x 13.1-in.
Weight: 48 lbs.
Price: $2,599.00

DOD Electronics Corpooration

The IPS-33B Super Harmony Machine generates 1 and 2 note chromatic and intelligent scalar harmonies from a single note played. Choose from 59 scales. Includes 7 different effects.
Dimensions: 1.75-in. x 19-in. x 8.50-in.
Weight: 5.50 lbs.
Price: $899.95
Price: $782.00

Peavey Electronics Corporation —See our ad on page 3
QFX 4x4 offers four channel multi-effects processor featuring 4 independent, full-featured, digital, multi-effects processors in one I.U. rack space; 16-bit sampling; up to 2.75 seconds of digital delay available; full MIDI or front panel access to all 128 presets and 128 patches.
Dimensions: 1.75-in. x 19-in. x 8.875-in.
Weight: 7 lbs.
Price: $1,299.00

Yamaha Pro Audio
SPX900 Digital Multi-Effects Processor is second generation DSP technology; 50 factory preset effects including compression and ADR noise-gate; 49 RAM user memory locations. Sampling frequency of 44.1 kHz. Comprehensive parameter control for each effect. Multiple and dual effects programs. Optional full function remote control. Full MIDI compatibility.
Dimensions: 1-¾-in. x 18-⅞-in. x 12-½-in.
Weight: 9.7 lbs.
Price: $995.00
SXP1000 Digital Multi-Effects Processor is also second generation DSP technology and offers a wide selection of professional-quality digital effects; 40 factory preset effects including compression and ADR noise-gate, 59 RAM user memory locations. Sampling frequency of 44.1 kHz. Comprehensive parameter control for each effect. Multiple effects programs and dual effects programs. Full MIDI compatibility.
Dimensions: 1-¾-in. x 18-⅞-in. x 12-½-in.
Weight: 8.2 lbs.
Price: $1,995.00
FX500 Simul-Effect Processor provides up to 5 different effects, simultaneously, and has extensive programming capability for "personalization" of effects sounds. The effects are all created utilizing leading-edge Yamaha DSPII chip technology, which gives full 20kHz bandwidth, extensive programming capability, and superb sound quality; 60 preset effects programs, each with changeable parameters, and has 30 additional RAM (Random Access Memory) locations which can store user-modified effects for recall. Full MIDI compatibility.
Dimensions: 8-⅝-in. x 1-¾-in. x 9-⅞-in.
Weight: 3 lbs.
Price: $495.00

EQUALIZERS

Alesis Studio Electronics
MICRO EQ is a monophonic, three band parametric equalizer featuring continuous frequency control, boost/cut controls with a range of 15dB, and two position "Q" switch that selects 2-octave and ½-octave bandwidth curves.
Dimensions: 1.5-in. x 5.75-in. x 7-in.
Weight: 1.75 lbs.
Price: $149.00
MEQ-230 Precision Equalizer proovides dual 30 band, ⅓ octave EQ in a single 19-in. rack space. Interface provided by means of quarter-inch and RCA jacks. Center frequencies range from 25 Hz to 20kHz and are set to ANSI/ISO standards. Each band provides +12dB cut/boost. In/Out switch.
Dimensions: (HxWxD) 1.75-in. x 19-in. x 4-in.
Weight: 2.5 lbs.
Price: $249.00

Altec Lansing Corporation
8558B Programmable Microaudio Equalizer offers eight memories, only one rack space, no front panel controls, 28 one-third octave filters with 12 dB of cut/boost, fixed HP/LP filters, elect. balanced in/out, xfmr in/out optional barrier strip only.
Dimensions: 1.75-in. x 19-in. x 7-in.
Height: 5.9 lbs.
Price: $1,250.00
1750A Cut-Only ⅓ Octave Mono Equalizer has 28 constant-Q filters from 31.5 Hz to 16 KHz, 15 dB of attenuation per filter, 20 dB of broadband gain, variable HP/LP filters, elect. balanced in/out with optional xfmr, XLR and barrier strip.
Dimensions: 3.5-in. x 19-in. x 9.75-in.
Weight: 10.7 lbs.
Price: $1,150.00
1753A Boost-Cut⅓ Octave Mono Equalizer has 28 constant-Q filters from 31.5 Hz to 16 kHz, 12 dB cut/boost per filter, 20 dB broadband gain, variable HP/LP filters, elect. balanced in/out with optional xfmr, XLR and barrier strip.
Dimensions: 3.5-in. x 19-in. x 9.75-in.
Weight: 10.7 lbs.
Price: $1,150.00

Applied Research and Technology —See our ad on page 5
IEQ Intelligent Equalizers are a 31 band controller that may be matched with up to 15 satellite units. The IEQs offer a video output so the acutal frequency curve is on a monitor screen. The IEQs are available in one-third and two-thirds octave versions.
Dimensions: 1.75 x 19-in. x 11-in.
Weight: 11 lbs.
Price: $829.00
HD Series Graphic Equalizers are new equalizers that feature ART's exclusive High Definition filter technology circuits that dramatically reduce phase shift and centerpoint drifts. Ultra-Low-Noise design yields a 115 dB signal-to-noise ratio.
Dimensions: 3.5 x 19-in. x 7-in.
Weight: 11 lbs.
Price: $425.00

ARX Systems —See our ad on page 2
EQ 60 is a dual channel 30 band EQ in a 3 rack units package. Features Low Noise (98dB A/W) Constant Q circuitry, balanced Inputs and Outputs, switchable 15 dB/6 dB cut and boost, with 30 well-damped, center grounding sliders on standard ISO frequencies, from 25 Hz to 20 kHz.
Dimensions: 5.25-in. x 19-in. x 9-in.
Weight: 15 lbs.
Price: $1,349.00
Multi Q is a six channel parametric equalizer, with exclusive internal patching system. User can access up to 6 channels of parametric EQ individually, or in multiples without the need for patch leads. A unique creative tool for any EQ situation, in one rack unit.
Dimensions: 1.75-in. x 19-in. x 6-in.
Weight: 5 lbs.
Price: $688.00

Ashly Audio, Inc.
The PQ-16 and PQ-26 are parametric equalizers and feature tunable low and high frequency shelving filters; center frequencies tunable over a 5.6 octave range; adjustable badwidths from 3-1/3 to 1/20 of an octave; full 20 dB headroom; true reciprocal curves; optional 12-band mono operation (on the PQ-26) and; master and individual band in/out bypass switching.
Dimensions: 1.75-in. x 19-in. x 6-in. (PQ-16)
3.5-in. x 19-in. x 6-in. (PQ-26)
Weight: 8 lbs. (PQ-16)
12 lbs. (PQ-26)
Prices: $399.99 (PQ-16)
$699.99 (PQ-26)
Models GQ-215, GQ-131, GQ-231 are a graphic equalizer series that feature precision Wein-bridge filters for accurate response and low distortion; detented metal-shaft fader with saddle knob; constant "Q" design with low ripple and accurate response near "flat" setting; selectable 15 dB or 6 dB range (on 31-band models); switchable-tunable low-cut filter (switchable 40 Hz fixed on GQ-215); 9-position, 3-color LED Level Meter (on 31-band models), plus peak LED indicators and balanced XLR and Unbalanced 1/4-in. inputs and outputs.
Dimensions: 3.5-in. x 19-in. x 6-in. (GQ-215, GQ-131)
5.25-in. x 19-in. x 6-in. (GQ-231)
Weight: 9.5 lbs. (GQ-215, GQ-131)
13 lbs. (GQ-231)
Prices: $499.99 (GQ-215)
$569.99 (GQ-131)
$1,059.99 (GQ-231)

dbx Professional Products, a division of AKG Acoustics, Inc.
1531X Graphic Equalizer has selectable 15 band stereo (2/3 octave) or 31 band mono (1/3 octave) equalizer on ISO centers. With constant-Q and symmetrical peak/dip curves with selectable 7.5 or 15 boost or cut, and switchable HP filtering @ 20 Hz, 60 Hz or 120 Hz.
Price:
$399.00

Electro-Voice, Inc.
Model 2710/1/3-Octave Graphic Equalizer features 27-band, 1/3-octave equalizer, constant range variable-Q filters, minimal interference between adjacent filters, user-selectable high- and low-pass filters, built-in pink-noise generator for noise masking, system equalization and other applications.
Dimensions: 3.5-in. x 19-in. x 10.25-in.
Weight: 11.5 lbs.
Price: $889.00

Orban, a division of AKG Acoustics, Inc.
642B Parametric Equalizer is a stereo 4 band or mono 8 band parametric equalizer with each band tunable over a 20:1 frequency range. +16dB boost or – 40 dB cut for each frequency and continuously variable "Q" for true infinite depth notching. Continuously-tunable 18 dB/octave HPF and "Automatic Sliding" 12 dB/octave LPF. Other variations for tuning ranges include the 642B/SP and 642B/SPX.
Price:
$1,045.00
672A Graphic Parametric Equalizer is an 8 band mono, Graphic/Parametric equalizer with tuning and bandwidth controls and "fader-style" boost/cut controls. Each band tunable over a 3:1 frequency range and "Q" variable between 0.3 and 20 with tuning control centered. Narrowband notching possible where the "Q"=10. Switchable, 12dB/octave, continuously tunable HP and LP filters, with separate outputs for use as a crossover. The stereo version is the Model number 674A.
Price:
$689.00 for the 672A
$1,299.00 for the 674A
The 764A Programmable Parametric incorporates all of the audio advances as the 642B. However, Storage, Recall and Programmability now possible for up to 99 presets of EQ for the single channel unit and an additional 99 presets for two channel unit. Instant Recall, Grouping, Comparing and Naming are also standard features. An additional 99 presets are available with the 764A/SL Slave unit. We also offer MIDI, RS-232 and RS-422 as optional interface modules.
Price: $2,495.00 for 2 channel
$1,995.00 for single channel
$1,895.00 for slave unit

Oxmoor Corporation
DEQ-29 Programmable ⅓ Octave Equalizer is an octave programmable equalizer without a physical control panel. Equalization is adjusted by using the Oxmoor TWEEQ software program and the Apple MacIntosh computer. Sixteen equalizers can be controlled by a single computer. Each filter and the overall gain is software programmable in ½ dB steps over a range of 12 dB. The corner frequency of the 18 dB/octave high pass filter is also software programmable. The PC is required only during the equalization process. You can store up to eight curves in the DEQ's non-volatile memory. Through the control port you can recall any one of the eight curves remotely using a simple dry-contact closure.

Peavey Electronics Corporation —See our ad on page 3
PME 4000 is a four-channel parametric equalizer featuring full control of frequency, cut/boost, and Q (variable from ⅙ to 2 octaves); top and bottom bands switchable (peak to shelving); +4 dBu balanced in and out.
Dimensions: 1.75-in. x 19-in. x 7-in.
Weight: 6 lbs.
Price: $349.00
AEQ 2800 is a MIDI controlled automated equalizer featuring 28-band EQ on 3rd octave centers; 40x2 character liquid crystal display; MIDI controllable "sliders"; 6 dB in 0.5 steps; 128 complete EQ program memories.
Dimensions: 1.75-in. x 19-in. x 8.125-in.
Weight: 7 lbs.
Price: $549.00

Roland Corporation US
E-660 Digital Parametric Equalizer is a fully programmable, digital parametric EQ. Stores 99 settings. Operates as two-channel, four-band EQ or single-channel, eight-band EQ. Analog, AES/EBU digital I/O connections. Accommodates 48 kHz, 44.1 kHz signals.
Dimensions: 3-9/16-in. x 19-3/16-in. x 16-9/16-in.
Weight: 17 lbs., 10 oz.
Price: $1,995.00

Soundcraftsmen —See our ad on page 11
PRO-EQ 22 C-MOS 0.1dB Differential/Comparator Octave Equalizer is a two-channel device with 10 octave-wide bands of adjustment for each channel featuring C-MOS Digital Switching, Differential/Comparator 0.1 dB True Unity Gain controls, LED True Unity Gain indicators, EQ defeat totally bypasses equalizer, Pre/post EQ processor loops.
Dimensions: 3.50-in. x 19-in. x 11-in.
Weight: 15 lbs.
Price: $349.00
PRO-EQ 44 is a C-MOS 0.1dB Differential/Comparator Third Octave that features C-MOS Digital Switching, Two independent Channels of EQ, ⅓ -Octave 40Hz/1kHz, Alternate ⅓ -Octave 1kHz/16kHz, Exclusive Differential/Comparator Unity-Gain Circuits, balancing LEDs for instant adjustment to Unity-Gain, Pre-Post EQ Loops and EQ Defeat Switch.
Dimensions: 3.50-in. x 19-in. x 11-in.
Weight: 15 lbs.

Symetrix
SX201 Parametric EQ/Preamp features three bands with +15dB boost and– 30dB cut, 16Hz to 20kHz frequency control, .05 octave to 3.3 octave variable bandwidth, separate line and preamp inputs, balanced and unbalanced inputs and outputs.
Dimensions: 1.75-in. x 8.5-in. x 6-in.
Weight: 5 lbs.
Price: $259.00

Tascam Professional Division (Teac Corporation of America) —See our ad on page 20-23
The GE-20B 2-Channel Graphic Equalizer has 10 linear equalization controls per channel, each providing up to 12 dB of boost or cut in the corresponding band. Each channel also has separate 12dB/Octave high pass and low pass filters.
Price: $325.00
The PE-40 4-Channel parametric equalizer has 4 EQ Channels each with 4 EQ bands 800 Hz to 16 kHz, 500 Hz to 10 kHz, 200 Hz to 4 kHz and 40 Hz to 800 Hz. Each EQ has frequency, Q (Bandwidth) and gain control.
Price: $650.00

White Instruments
Model 4320 One-Third Octave Passive Equalizer is twenty-eight one-third octave L-C filters on I.S.O. centers. Continuously variable, rotary controls, 10dB cut only. Used with 0/10 kilohms termination. Plug-in 2-way or 3-way crossovers. Security cover included.
Dimensions: 3.5-in. x 19-in. x 7-in.
Weight: 5 lbs.
Price: $800.00
Model 4700-2 Digitally Controlled One-Third Octave Active Equalizer offers two channels of one-third filters like the 4700. Remote preset select interface, RS-232 interface, EIA-422 interface and preset switch kit options available for both 4700-2 and 4700.
Dimensions: 1.75-in. x 19-in. x 2-in.
Weight: 7 lbs.
Price: $1,375.00
Model 4700 Digitally Controlled One-Third Octave Active Equalizer provides the flexibility of digital control with the performance of analog circuitry, 10 dB boost/cut, 0.5 dB resolution, 10 memories and presets, 2 level password protection, non-volatile memories. Servo-balanced differential I/O.
Dimensions: 1.75-in. x 19-in. x 12-in.
Weight: 6 lbs.
Price: $875.00

Model 4500 One-Third Octave R-C Active Equalizer has twenty-eight one-third octave R-C filters on I.S.O. centers. Continuously variable, rotary controls, 10 dB boost/cut, 12 dB/octave variable high-pass and low-pass filters. Active buffered input, 3 outputs. Plug-in 2-way or 3-way crossovers and I/O transformers options.

Dimensions: 3.5-in. x 19-in. x 5-in.

Weight: 7 lbs.

Price: $750.00

Model 4650 One-Third Octave R-C Active Equalizer offers twenty-eight one-third octave R-C filters on I.S.O. centers. Continuously variable, 60mm linear controls, 12dB boost/cut, 12 dB/octave fixed high-pass and low-pass filters. Optional input/output transformers.

Dimensions: 3.5-in. x 19-in. x 5-in.

Weight: 7 lbs.

Price: $649.00

Model 4675 Two-Thirds Octave Two Channel R-C Active Equalizer has fourteen two-thirds octave R-C filters on I.S.O. centers. Continuously variable, 60mm linear controls, 12 dB boost/cut, 12 dB/octave variable high-pass and low-pass filters. Servo-balanced input/output circuitry.

Dimensions: 3.5-in. x 19-in. x 5-in.

Weight: 7 lbs.

Price: $779.00

Model 4660 One-Third Octave R-C Active Equalizer has twenty-eight one-third octave R-C filters on I.S.O. centers. Continuously variable, 60mm linear controls, 12 dB boost/cut, 12 dB/octave variable high-pass and low-pass filters. Optional input/output transformers.

Dimensions: 3.5-in. x 19-in. x 5-in.

Weight: 7 lbs.

Price: $699.00

Model 4400-One-Third Octave L-C Active Equalizer offers twenty-eight one-third octave L-C filters on I.S.O. centers. Continuously variable, rotary controls, 10 dB boost/cut, 12 dB/octave variable high-pass and low-pass filters. Active balanced input, 3 outputs. Plug-in 2-way or 3-way crossovers, I/O transformers available.

Dimensions: 3.5-in. x 19-in. x 7.875-in.

Weight: 15 lbs.

Price: $1,050.00

Yamaha Pro Products

DEQ7 Digital Equalizer is a 16-bit digital programmable equalizer with a range of EQ formats available: 1-octave, ⅔ octave, ½-octave or ⅓-octave. All except ⅓-octave are full stereo configurations—either simultaneous of independent programming.

Dimensions: 18-⅞-in. x 1-¾-in. x 11-¼-in.

Weight: 8.2 lbs.

Price: $1,395.00

GQ1031BII Graphic Equalizer is a single channel, ⅓ octave EQ. Only one EIA rack space high. 31 bands of EQ, each 12 dB boost/cut. Input level control. Peak level LED set at +17 dB with 3dB additional headroom. ¼-in. Jacks and RCA pin jack for Input and Output. 22 k ohm input impedance for easy interface. Output source impedance of 600 ohms.

Dimensions: 1-¾-in. x 18-⅞-in. x 8-¾-in.

Weight: 6.4 lbs.

Price: $375.00

Q1027 Professional ⅓ Octave Graphic Equalizer is a single channel, ⅓ octave EQ, 27 bands at ISO frequencies, each 12dB boost/cut. 27 separate active peaking filters and summing network for minimum phase shift. Calibrated, 1 dB-stepped input level control. High Pass filter—18 dB/octave, selectable 40 Hz or 80 Hz. Front Panel LED indicates 3 dB below clipping. XLRs and ¼-in. jacks for both Input and Output.

Dimensions: 3-⅔-in. x 18-⅞-in. x 12-in.

Weight: 17.6 lbs.

Price: $1,095.00

Q2031A Dual ⅓ Octave Equalizer is a dual channel, ⅓ octave EQ. 31 bands of EQ on ISO frequencies from 20 Hz to 20 kHz. Selectable 12dB or 6dB boost/cut for each channel. Octal socket for optional BRT-15K transformers. Peak level indicator set for 3 dB below clipping. XLR electronically balanced and ¼-in. unbalanced inputs.

Dimensions: 3-½-in. x 18-⅞-in. x 11-¾-in.

Weight: 11.2 lbs.

Price: $695.00

CROSSOVERS

Altec Lansing Corporation

Model 1631A Electronic Crossover is a two-way electronic crossover using plug-in modules to select crossover frequency, and configure specific equalization to provide flat power response for various horn/driver combinations. The high-pass output has a level control and the low-pass output has a delay adjustment of 0 to 25 ms.

Dimensions: 1-¾-in. x 19-in. x 4-⅞-in.

Weight: 4.74 lbs.

Price: $625.00

1632A Electronic Dividing Network is a daul channel two-way or single channel three-way active crossover, 24 dB/octave, selectable from 50Hz to 10KHz, elect. balanced in/out with xfmr in/out optional, 30/60 Hz HP inputs, hard limiters on all 4 outputs, sub-modules to customize response.

Dimensions: 1.75-in. x 19-in. x 9.75-in.

Weight: 8 lbs.

Price: $1,100.00

15594A Low Pass Crossover/Equalizer Module is a plug-in module for 9400 series power amplifier. 18 dB/octave roll-off pre-programmed at 125 Hz, 500 Hz, 800 Hz, 1250 Hz, customer programmable for other frequencies, programmable 12 dB HP roll-off with pre-sets at 16 Hz or 32 Hz.

Dimensions: 1.6-in. x 2-in.

Weight: 1.6 oz.

Price: $88.00

15595A High Pass Crossover/Equalizer Module is a plug-in module for 9400 series power amplifier. 18 dB/octave roll-off pre-programmed at 125, 315, 500, 800, 1250 Hz, customer programmable for other frequencies, sub-modules available to customize frequency response to horn/driver.

Dimensions: 1.6-in. x 2-in.

Weight: 1.6 oz.

Price: $88.00

ARX Systems —See our ad on page 2

EC 1 is a stereo 2 way, mono 3 way Electronic Crossover. The single rack unit package has true Linkwitz Riley 24 dB per octave filters, 2 or 3 way mode switch, and 'user friendly' crossover point changing.

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

Weight: 4 lbs.

Price: $299.00

Ashly Audio, Inc.

12dB Per Octave Electronic Crossovers

Models XR-22E, XR-77E, XR-80E, XR-88E. All crossovers use state-variable filter circuits to perform the frequency divisions. Features include balanced or unbalanced inputs/outputs; variable filter response allowing tuning-in; damping control to flatten frequency response and; peak overload warning lights. Layout provides "easy-operation" feel.

Dimensions: 1.75-in. x 19-in. x 6-in. (XR-22E, XR-80E)
3.5-in. x 19-in. x 6-in. (XR-77E, XR-88E)

Weight: 8 lbs. (XR-22E, XR-80E)
10 lbs. (XR-77E, XR-88E)

Prices: $359.99 (XR-22E)
$499.99 (XR-77E)
$419.99 (XR-80E)
$629.99 (XR-88E)

24 dB Per Octave Electronic Crossovers, Models XR-1000, XR-2000, XR-3000, XR-4000. Some features include variable filter response allowing tuning in Linkwitz-Riley, Butterworth or other filter performances; balanced XLR or 1/4-in. Phone,TRS inputs/outputs; 20 Hz hi-pass filter; peak overload warning lights and tuning ranges of 40 Hz-8 kHz for the XR-1000/2000, and 40 Hz-24 kHz for the XR-3000/4000.

Dimensions: 1.75-in. x 19-in. x 6-in. (XR-1000, XR-3000)
3.5-in. x 19-in. x 6-in. (XR-2000, XR-4000)

Weight: 8 lbs. (XR-1000, XR-3000)
10 lbs. (XR-2000, XR-4000)

Prices: $399.99 (XR-1000)
$599.99 (XR-2000)
$499.99 (XR-3000)
$699.99 (XR-4000)

BrystonVermont Ltd.

10B Active Crossover features independently selectable crossover points for high-pass and low-pass, also independently selectable crossover slopes from 6, 12 or 18 dB/Octave, and incorporates Bryston's exceedingly linear and superbly quiet discrete op-amp circuitry. Two-way stereo, three-way mono, No integrated circuits in the signal path, uses only triple gold-plated contacts and all modular construction.

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

Weight: 12 lbs.

Price:$1,195.00

Electro-Voice, Inc.

Model XEQ-3/Electronic Crossover features 3-way configurations, allows low-frequency signal delay for source alignment, low-frequency boost for extended bass, step-down operation of TL bass system. Has simple, easy-to-install modules for compression-driver high-frequency equalization.

Dimensions: 1.73-in. x 19-in. x 7.28-in.

Weight: 6.8 lbs.

Yamaha Pro Products

F1030 Frequency Dividing Network is a two-way or three-way active frequency dividing network with three outputs. Precision 26-position Log-Linear detented and dB-calibrated attenuators. Three high-pass filters, two low-pass filters. First high-pass is at 40Hz; 12dB/octave for sub-sonic. Each filter ahs selectable crossover point. Filter slopes are selectable –12dB/octave or 18 dB/octave. XLRs for input and output.

Dimensions: 3-3/4-in. x 19-7/8-in. x 9-3/8-in.

Weight: 16.5 lbs.

Price: $745.00

F1040 Professional Frequency Dividing Network is a two-way, three-way or four-way active frequency dividing network with precision 26-position Log-Linear detented and dB-calibrated attenuators, and four outputs. Three high-pass filters and three low-pass filters, each filter has selectable crossover point. Filter slopes are selectable –12 dB/octave or 18 dB/octave. XLRs for input and output.

Dimensions: 3-3/4-in. x 18-7/8-in. x 12-in.

Weight: 17.6 lbs.

Price: $920.00

DIGITAL DELAYS

Applied Research and Technology —See our ad on page 5
The PD3 is a 1 in/3out speaker alignment delay offering up to 256 ms per tap in 1 ms increments. Its 64 kHz simple rate yields flawless signal reproduction.
Dimensions: 1.75 x 19-in. x 9-in.
Weight: 11 lbs.
Price: $749.00

Roland Corporation US
SDE-3000A Programmable Digital Delay. Maximum delay time 4500 ms. 10Hz - 17kHz (+0.5/–3.0 dB) frequency response from O to 1500 ms. Eight memory locations. 88 dB S/N ratio. 0.05 percent THD at 1 kHz. 100 dB dynamic range. Modulation CV input and CV In- output jacks.
Dimensions: 1-13⁄16-in. x 19-in. x 12-13⁄16-in.
Weight: 11 lbs.
Price: $1,095.00

Sound Concepts
Model SS550 Delay and Surround Processor is a companded analog delay system with two channels of 5 to 50 milliseconds delay that may be switched between stereo and sequential mono operation. Also has film surround mode.
Dimensions: 3.50-in. x 15-.25-in. x 9-in.
Weight: 8 lbs.
Price: $879.00
Price: $549.00

Yamaha Pro Products
DDL3 Digital Delay Line offers high-performance digital delay with on-board equalization. 1-in/3-out configuration with independently programmable delays up to 1300 milliseconds. Independent digital equalization for each delayed output with four modes: 1-3-band parametric EQ, 2-3-way frequency divider, 3-2-way frequency divider pluslow-pass filter, and 4-2-way frequency divider plus band-pass filter. 15 memory locations for EQ settings, recallable via front panel, MIDI or customer controls. Delay settable in milliseconds, feet or meters. 20-20 kHz response with greater than 100 dB dynamic range and THD less than 0.03 percent.
Dimensions: 18-7⁄8-in. x 1-3⁄4-in. x 12-3⁄4-in.
Weight: 8.4 lbs.
Price: $1,295.00

REVERBS

Alesis Studio Electronics
MICROVERB II 16 Bit digital reverberation system offers a comprehensive range of reverb programs. The heart of MICROVERB II is the custon-designed Alesis VLSI microchip. INPUT, MIX and OUTPUT controls. Stereo in, stereo out, 15 kHz bandwidth.
Dimensions: 1.5-in. x 5.75-in. x 7-in.
Weight: 1.75 lbs.
Price: $199.00

Orban, a division of AKG Acoustics, Inc.
111B/1 Dual Channel Reverb is a 2 channel reverb featuring 6 springs per channel for applications where natural, open sound is preferred. An adjustable Floating Threshold Limiter to minimize spring twang on transients also provided. Bass and Mid EQ also adjustable, while Mid EQ is bandwidth adjustable.
Price:
$959.00

Roland Corporation US
R-880 Digital Reverb is four independent DSPs. Wide variety of reverb types. 90 dB dynamic range. 20 Hz-20 kHz (+0.2/-3.0 dB) frequency response. 48 kHz and 44.1 kHz sampling rates. Chorus, delay, EQ, compressor. Analog, AES/EBU digital I/O connections.
Dimensions: 3-9⁄16-in. x 19-in. x 16-13⁄16-in.
Weight: 22 lbs., 1 oz.
Price: $3,995.00
GC-8 Graphic Controller for R-880 is a graphic remote control unit for the R-880 Digital Reverb. Large, 256 x 64 dot LCD. Five rotary knobs, numeric keypad. Memory card slot for storing and loading programs.
Dimensions: 2-in. x 13-1⁄8-in. x 6-15⁄16-in.
Weight: 2 lb., 10 oz.
Price: $850.00

Peavey Electronics Corporation —See our ad on page 3
The DSR 1000 digital stereo reverb featuring adjustable Echo, Pre-Delay, Early Reflections, Room Size, Tonal Color, Reverb Time, Left and Right Stereo Channel Delay, Left and Right Stereo Echo Feedback, Chorus Rate, Depth, Delay Time and Feedback; full MIDI access.
Dimensions: 1.75-in. x 19-in. x 8.125-in.
Weight: 7 lbs.
Price: $499.00

Correction on the Wireless Microphone Directory in July/August issue. Please note that incorrect information was listed for HME.

HME Electronics, Inc.

Systems 50, 55 systems have frequency response of 50 Hz-15 kHz ±2 dB, with RF Carrier Frequency of 160-216MHz. Frequency Stability is ±0.005 percent. Less than 1 percent distortion. Dynamic Range is greater than 115 dB ("A" weighted). System requires FCC station license if operated in U.S. or its possessions depending on application.

Systems 515/525. The 515, a body-pac wireless mic system, includes body-pac transmitter, antenna, belt-clip, receiver, whip antenna and 115 VAC power adapter. The 525, a handheld wireless mic system, includes transmitter with dynamic mic element, mic clamp, receiver, whip antenna and AC power adapter. Specifications include frequency response of 50 Hz-15 kHz ±3 dB, RF Carrier Frequency of ±0.005 percent, less than 1 percent THD distortion and a dynamic range of greater than 100 dB ("A" weighted). FCC station license required if systems operated in U.S. or possessions.

HM Electronics, Inc., 6675 Mesa Ridge Road, San Diego, CA 92121

ADDRESSES

Alesis Corporation
3630 Holdrege Avenue
Los Angeles, CA 90016

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

Applied Research & Technology
215 Tremont Avenue
Rochester, NY 14608

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

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

BGW Systems
13130 Yukon Avenue
P.O. Box 5042
Hawthorne, CA 90261-5042

BrystonVermont, Ltd.
979 Franklin Lane
Maple Glen, PA 19002

dbx, a division of AKG Acoustics, Inc.
71 Chapel Street
Newton, MA 02195

DOD Electronics Corp.
5639 South Riley Lane
Salt Lake City, UT 84107

Electro-Voice
600 Cecil Street
Buchanan, MI 49107

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

Oxmoor Corporation
237 Oxmoor Circle
Birmingham, AL 35209

Peavey Electronics Corporation
711 A Street
P.O. Box 2898
Meridian, MS 39302-2898

RolandCorp US
200 Dominion Circle
Los Angeles, CA 90040-3647

Sound Concepts
P.O. Box 135
Brookline, MA 02146

Soundcraftsmen
2200 South Ritchey
Santa Ana, CA 92705

Symetrix
4211 24th Avenue West
Seattle, WA 98199

Tascam-Teac Corporation of America
7733 Telegraph Road
Montebello, CA 90640

White Instruments
P.O. Box 698
Austin, TX 78767

Yamaha
P.O. Box 6600
Buena Park, CA 90622

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"Tonmeister Technology"—the first book that fully integrates music and engineering in the German Tonmeister concept. Equally useful to recording musicians and mixers. Send $21.95 for soft, $36.95 for hard cover (plus NY tax). **Temmer Enterprises Inc. 767 Greewich St. New York, NY 10014.**

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WANTED

RADIO TRANSCRIPTION DISCS: Any size, speed. Drama, comedy, music, variety, adventure, soaps, children's, AFRS, big band remotes, library services. **KINER-db, Box 724, Redmond, WA. 98073-0724.**

WE BUY USED DOLBY CAT. #22 CARDS! If you have replaced your cards with Cat. #280 SR cards, we will buy them. Please let us know how many you have and the asking price. **Smart Theatre Systems, P.O. Box 80361, Atlanta, GA 30341. (800) 45-SMART.**

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

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

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

ALL CLASSIFIEDS MUST BE PREPAID

Send copy to:
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People, Places... & Happenings

• **Ed Learned, a db Magazine** contributing editor, will not be featured in this or the November/December issue, as he is on tour with the *Charlie Byrd Trio* from Aug. 12 to Sept. 14, and *Wayne Toups* and *Zydecajun* from Sept. 22 to Oct. 30.

His complete tour schedule follows.

• *Aug. 12-16* Manila, Philippines , *Aug. 17* Cebu, Philippines, *Aug. 18-19* Davao, Philippines, *Aug. 20-23* Singapore, *Aug. 24-25* Chiang Mai, Thailand , *Aug. 26-27* Bangkok, Thailand, *Aug. 28-30* Jakarta, Indonesia, *Aug. 31* Surabaya, Indonesia, *Sept. 1-2* Medan, Indonesia, *Sept. 3-4* Penang, Malaysia, *Sept. 5* Kuala Lumpur, Malaysia, *Sept. 6-9* Seoul, South Korea, *Sept. 10* Kwangju, South Korea, *Sept. 11* Chonju, South Korea, *Sept. 12-13* Taegu, South Korea, *Sept. 14* Pusan, South Korea, *Sept. 15-21* Beijing, China

While in Beijing, Learned will conduct lectures and demonstrations on modern sound reinforcement techniques and rock 'n' roll mixing techniques.

Sept. 22-26 Bangkok, Thailand, *Sept. 27* Khon Kaen, Thailand, *Sept. 28* Chiang Mai, Thailand, *Sept. 29-30* Bangkok, Thailand, *Oct. 1-3* Manila, Philippines, *Oct. 4* Cebu, Philippines, *Oct. 5* Davao, Philippines, *Oct. 6* Manila, Philippines, *Oct. 7-13* Sri Lanka (schedule to be announced—includes Colombo), *Oct. 14-17* Singapore, *Oct. 18-21* Auckland, New Zealand, *Oct. 22* Christchurch, New Zealand, *Oct. 23-24* Wellington, New Zealand, *Oct. 25* Nadi, Fiji, *Oct. 26* Suva, Fiji, *Oct. 27-28* Sigatoka, Fiji, *Oct. 29* Lautoka, Fiji, *Oct. 30* Depart for USA

• A new, state-of-the-art stereo surround sound system, featuring an array of **Tannoy P100 Dual Concentric loudspeakers**, has been installed at the **Space & Rocket Center** theater in Titusville, FL. The system, developed around the Tannoy P100, is designed to render dialogue and high dynamic sound effects from a CD-quality audio source, specifically rocket launch sounds and audio program information associated with the field of space exploration...A **Sony digital two-track PCM-3402 DASH** recorder has been delivered to **Foto-Kem/Foto-Tronics**, a motion picture laboratory and post-production facility in Burbank, CA. The unit has already served in several applications, including a project to save rare film footage made 15-20 years ago of some of Neil Young's earliest performances. Sony has also delivered a PCM Digital mastering system to **Lion and Fox Recording** in Washington, D.C. The system, which Lion and Fox will use for its Mastering Studio B, is the standard mastering format for CD production. The system includes a PCM 1630 Digital Processor, DMR 4000 Digital Master Recorder and a DTA 2000 Tape Analyzer

Shure StereoSurround has made its Michigan debut at **Ron Rose Productions** in Farmington Hills. The equipment was installed at the company's Studio Center "Room With A View" suite by Shure HTS, which was looking for a commercial production house to test market its new StereoSurround audio system...**Gauss loudspeakers** have been installed at Olympic Park in Seoul, South Korea. **Young Nak So Ri Sa**, the firm that designed and installed the sound system, previously supplied Gauss 18-

inch and 15-inch woofers for most of the low frequency systems in the Olympic Stadium...**Blank Productions** of Stamford, CT, has updated its studios with a variety of new technology, including the latest mixing software from **Steinberg, DOD 24-channel noisegates, two Proteus X units and Crown PZM mics** for their drum booth...**Crown International** engineers have begun the initial phase of a long-term project that will bring state-of-the-art electronics to the **Indianapolis Motor Speedway's** audio system. New design elements given to the system in time for the recent Indy 500 are an **IQ System 2000** and microphone assortment, including a custom-built **"tridundant"** mic, **PCC160 Super Cardioid boundary mic, CM200 mic, CM310 differoid mic and new SASS-P mics**...**Remote Recording Services** has replaced its "Black Truck" with the all-new Silver Mobile Studio, which features a **48-input API console** with all-discrete circuitry and **GML automation, Bryston-powered KRK monitors** and **Studer A810 and A820 tape decks**. The Silver Mobile has full video monitoring facilities, and special attention has been paid to infrastructure factors such as A.C. line voltage and filtering, SMPTE time code lock, audio distribution and patching...**Ardent Recordings** in Memphis, TN, has put its new digital **Dyaxis** to work on a number of projects including jingle production, album editing, sound tracks and sound effects editing...**The 19th Hard Rock Cafe**, which recently opened in Singapore, is the first to use Electro-Voice components in 100 percent of its audio system. The equipment makes up what is estimated to be the most powerful audio system ever installed in a Hard Rock Cafe. The total RMS power for the facility is in excess of 30,000 watts.

• **Systems Development Group** announces the opening of Audiomaster, a digital recording facility dedicated to the production of high quality radio and television soundtracks for advertising. The SDG team has designed a facility for adwork and TV post-production that has a 28' x 25' control room with more than enough space to comfortably accommodate ad agency producers and their clients. There are also two separate 16 x 12-foot studios, each with its own visual access to the control room...**International Post**, an advanced international post-production facility, has opened on the East Coast as a division of **Audio Plus Video International**, a Video Services Corporation company. International Post offers multi-standard editing, film-to-tape transfer and audio layback rooms...The groundbreaking and start of construction for a new **SoundTech** manufacturing facility that will total more than 100,000 square feet has been announced. Rudy Schlacher, president of SoundTech, said a new factory is needed in order to keep up with the company's rapid growth of its line of sound reinforcement equipment and efficient delivery to customers. Scheduled completion date of the new facility is November, 1990...**AKG Acoustics, Inc.**, has consolidated its U.S. operation in a new, 77,000 square foot facility in San Leandro, CA. **Richard Ravich**, company president, said the recent acquisitions of **dbx Professional** Products and **Orban, Inc.**, as well as the growth of AKG Acoustics, necessitated the move.

The Institute of Audio Research (IAR) has established an ongoing course in live sound reinforcement. Specifically designed to enable students to learn the necessary techniques of live sound, the course gives students direct, hands-on experience in selecting and assembling systems for all types of venues, from small clubs to large concert halls. In addition to physically setting up and breaking down each system, IAR students learn the proper procedures for sound checks and monitor and house mixing.

• **David S. Mash** has been appointed to Assistant dean of Curriculum for Academic Technology at the **Berklee College of Music** in Boston, MA. Other appointments and promotions at companies nationwide include the appointment of **William C. Mohrhoff** as vice president of Worldwide Sales and Service at **E-mu Systems, Inc.** Before joining E-mu, Mohrhoff was the marketing manager at the **Tascam division of TEAC Corporation of America**...**Nancy A. Calvert** has been appointed manager of Advertising and Public Relations at **Shure Bros**...**Bill Whitlock** has been named president of **Jensen Transformers**, Inc...**Gary Stanfill**, president of wireless technology company **Vega**, has been named a vice president of **Mark IV Audio**. **Hans Tschernig**, president of **Dynacord** in West Germany, has been appointed to the additional position of vice president of **Mark IV Audio, Inc.**, a subsidiary of **Mark IV Industries, Inc**...**John Carey** of **Otari Corporation**, who started his career at Otari in 1981 as Product manager and moved up to Sales manager and Marketing manager, has been promoted to vice president of Sales and Marketing...**David W. Roudebush**, formerly worldwide marketing manager for **Orban Professional Products**, has been promoted to U.S. Marketing and Sales manager for **AKG Acoustics, the Orban Professional Products Division and the dbx Professional Products Division**...**David Hamlin** has been appointed to plant manager of **SoundTech's** Elkhart, IN, manufacturing facility, and he will play a key role in the Elkhart facility's planned relocation...**Intersound Inc**. has promoted **Bryan J. Rusenko** to vice president of Engineering and **Garry Morris** to executive director...**Paul V. Hugo**, National Sales manager of Gauss Loudspeakers, has been promoted to marketing and sales director. He has also been appointed to Gauss' operating and management staff... **George Massenburg Labs (GML)** has appointed **Adriane Benacquista** as the new Sales and Marketing administrator. She joins GML after two years at **Sony Pro Audio** in Burbank, CA, as coordinator of Sales and Administration...**Vic Steffens**, who has been associated with **Allen and Heath** as a product development consultant, has been appointed as Product Specialist and Customer Support. Steffens will continue his activities as independent producer and studio owner with the Horizon group of studios in Connecticut...**Rick Porter** has joined the engineering team at **Apogee Electronics**...**Stephen L. Ingram** has been hired by the **Society of Broadcast Engineers** as the organization's executive director...**Pro Audio Consultants and Engineers** has been appointed as **ARX Systems' (Australia)** representative for the fast-growing Indian Pro Audio market.

Shure L Series brings reliability to affordable wireless.

Why take chances with anything else?

If you're providing wireless microphone systems to churches, schools, or other value-conscious users, you need reliable equipment you can sell at an affordable price—and make a profit doing it.

That's what the new L Series from Shure is all about. The L Series sets a new standard of value in its price range, offering features, performance and reliability other "economy" systems can't match.

We didn't forget the details.

Designed and built by Shure in the U.S.A., L Series systems include many of the features that set professional-quality wireless systems apart from the "toys." L Series receivers are sturdy, metal-cased, and rack-mountable. Antennas are detachable and may be placed in remote locations, providing excellent performance in situations where many other wireless systems have trouble.

Our L1 Body-Pack Transmitter has features like a separate audio mute switch and a universal 4-pin "Tiny QG" connector that accepts a variety of microphone and musical instrument sources. And L Series lavalier systems come with the 839W, a reliable Shure condenser microphone designed for clear, natural vocal pickup.

Performance meets economy.

Even though L Series components are economically priced, they incorporate sophisticated RF technology. The L4 Diversity Receiver utilizes "intelligent" MARCAD™ circuitry to monitor signals from its two independent RF sections, blending them in the optimum proportion—not merely switching them. The result is reliable, uninterrupted audio with no clicks, no pops. And all L Series systems feature Shure "Mirror Image" companding, plus high-gain, low-noise MOSFETs, a high-fidelity quadrature detector, and a 3-pole Chebyshev audio low-pass filter. It all adds up to outstanding audio quality with exceptional freedom from noise and distortion.

Why risk callbacks with anything else?

Other systems may not meet expectations. But you can recommend a Shure L Series system with confidence. So why risk callbacks—and your reputation—with anything else?

For more information about the Shure L Series, call Shure Customer Services at (312) 866-2553.

THE SHURE® WIRELESS

L SERIES

THE SOUND OF THE PROFESSIONALS®...WORLDWIDE
222 Hartrey Avenue, Evanston, IL 60202-3696

Circle 12 on Reader Service Card

See us at the AES in booth 107, 109

NOW YOU CAN LEASE DIGITAL BY THE MONTH FOR THE PRICE OF RENTING ANALOG BY THE DAY.*

The Fostex D-20 operates just like the most expensive analog open reel 2-channel recorders with SMPTE/EBU capability, and now we're offering a lease program which makes it easy for you to have all the benefits of even better performance at lower rates (see details below).

With the D-20 professional digital audio master recorder, you can post-stripe time code on an existing DAT tape (recorded on any DAT machine), or you can record time code and stereo audio on the D-20 and play that tape back on any other DAT machine with complete compatibility.

The 20-pin **synchronizer port** allows interface with all the popular synchronizer systems (ours included) and there's an RS-422 port for control which requires serial communication. There's an external sync input for composite video, plus Word Sync Input and Output capability - all standard on the D-20.

Because of our 4-head recording system the D-20 features **off-the-tape monitoring** so that you'll always know exactly what you have on tape - a very important feature considering the DAT's ability to record for two straight hours (no more multiple reels and alignment hassles).

You'll be able to control all transport functions by remote control—including **punch-in/out**. Built-in cross-fade timing gives you seamless punches. There's even a **pitch control** complete with digital read-out.

Most important of all, the D-20 sounds great. It records and reproduces all the music completely, faithfully, and better than analog alternatives. So plug into the digital master recorder that has the professional features you need now at a price you can afford now.

* The analog recorder referenced is the Studer A-80 1/2" with 3-track head nest; the price comparison is based on option (A) below and an average of published rates of major audio rental companies for the Studer.

Details of the D-20 Lease Program:
* A simple one page application is all that's required. · Maximum 48 hour turnaround approval. · Two attractive payment schedules: (A) **$199.70 per month, 60 months, first and last payments in advance: 10% purchase option.** (B) $287.20 per month, 36 months, first and last payments in advance; 10% purchase option. · Please note that this lease with option to purchase is not offered through Fostex Corporation. All documents and associated paperwork will be completed by **Signet Lease** Group. Call them directly at **(215) 783-6666.** · High approval rating in the audio industry.

D-20

Digital Master Recorder

FOSTEX

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