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Designing a World Class Radio Station

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db magazine

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THE ELECTRONIC COTTAGE

See page 4.

the broadcast engineer

See page 39.

the sound contracting engineer

About the Cover

• At top, Matthews/Grifith Music—the personal studios of two of Los Angeles' most successful music writers featuring multiple Yamaha DMP Digital Mixers. The lower photo is of NW Mobile Video, a state-of-the-art mobile-production house. They are caught in preparation for a live Sacramento Kings basketball broadcast.

Photos courtesy Ron Bennett, C.M.G., Studio City, CA.

Calendar

• **Synergetic Audio Concepts** (Syn-Aud-Con) is beginning its 18th year of training audio professionals. This is their 1990 schedule for their 2-day and 3-day audio engineering seminars:

2-day Seminars:
Anaheim, CA—January 22-23

3-day Seminars at the Farm in S. Indiana:
April 26-28
May 17-19
June 21-23
July 26-28
August 23-25
September 20-22
October 11-13

For more information, contact Synergetic Audio Concepts, RT. 1, Box 267, Norman, IN 47264. Phone (812) 995-8212 or fax (812) 995-2110.

• The **1990 Electronic Distribution Show and Conference** (EDS '90) will take place at the Las Vegas Hilton Hotel, in Las Vegas, Nevada, with conferences and seminars starting on Monday, April 23, and exhibits opening on Tuesday, April 24, continuing Wednesday and Thursday, April 25 and 26. EDS '90, the marketplace of and for electronic distribution, has over 7000 management, sales and marketing people attending each year and over 350 companies exhibiting.

For more information, contact the **Electronic Industry Show Corporation, 222 South Riverside Plaza, Suite 2710, Chicago, Illinois 60606. Phone (312) 648-1140.**

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Editorial

Whatever happened to the 80s? Suddenly it is the 90s!

Welcome, if you are a new reader seeing **db Magazine** for the first time at the Winter NAMM show or the later winter NAB Convention in Atlanta. Of course, our subscribers have received their copies in December, so to them and to all a joyous holiday season, and a happy new year!!

What your are holding is the oldest professional audio magazine, save only the AES Journal, that has been continuously publishing since 1967. A lot has happened since then, of course, as any perusal of even the last ten-year collection of magazines will attest. Perhaps the most significant is the advance of the computer into professional audio as never before.

Even today, post-production and even master recording could hardly exist without the computer. Never mind its growing pervasiveness in music production for film and TV, but so much work today would not exist but for the digital hard disk—a product of the computer world. Digital editing is all computer based.

It is the computer that has made possible the *Electronic Cottage*, featured in several articles in this issue. Each could not exist without the computer and/or its spin-offs.

Look again at our Contents page. The first three stories (actually four stories) each revolve about some aspect of computer control or interface. The listed Lab Report on the Yahama M3TX four-channel cassette recorder/mixer exemplifies a product that "smart chips" have made possible.

To the NAB audience, I would particularly recommend Senior Editor John Barilla's *Designing a World-Class Radio Station*. When New York City's classical radio station WQXR was forced to move to new quarters it was an opportunity to truly start from scratch with an all-new operation that will lead them well into the next century. With the aid of NCC's Alfred W. D'Alessio (who designed and built the new complex) we really do lead you through the building of a world-class operation —and possibly adapting some ideas into a new recording or broadcast studio of your own. LZ

JOHN BARILLA

Designing a World-Class Radio Station

WQXR in New York City is arguably the finest classical music station in the world. A pioneer in classical music programming for over 50 years; WQXR broadcasts an innovative blend of recorded music and live performance, interviews and special network feeds 24 hours a day.

In recent months, WQXR, AM & FM, (The Radio Stations of the New York Times) moved from its old facilities in the New York Times building to a new, more spacious location at 122 Fifth Avenue.

The new complex was designed and engineered under the hand of Northeastern Communications Concepts, Inc.(NCC)—a leader in the development of quality broadcast facilities.

NCC's president, Alfred W. D'Alessio recently spent the better part of a day with **db Magazine's** editors offering our readership a tremendous opportunity to glean some insights into the process of designing a world-class radio station.

ON THE MOVE

One of the first issues we wanted to explore was the reason behind WQXR's relocation. To the listener, WQXR had always pumped out what seemed like a quality signal, but recent advances in digital audio have essentially changed the level of expectation from broadcast audio. Since the advent of the compact disc with its superb noise specs, the lim-

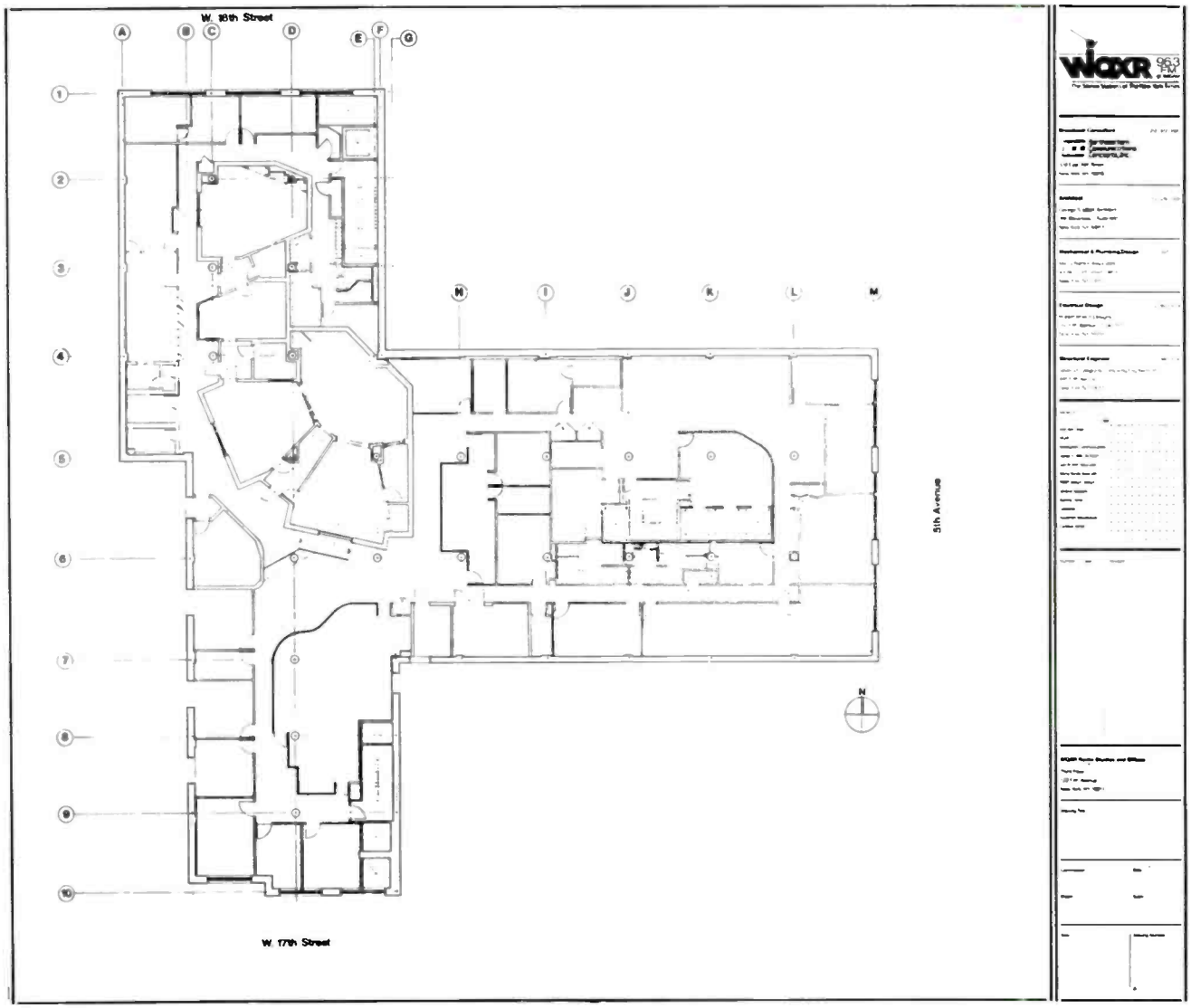

Figure 1. The "T" shaped layout of the WQXR offices. The studios are in the uppper left, entry from the elevators is at reception in the lower left.

iting factor has become the interaction between the other equipment in the broadcast chain. In order to bring noise factors to near theoretical limits requires an integrated acoustical/systems approach: piecemeal retrofittings are not sufficient.

A reshuffling of priorities at The New York Times Company (the parent company of WQXR) led to an unusual opportunity to redesign the facility completely from the ground up. The 43rd street location of the New York Times was simply getting too cramped, so a decision was made that the building should serve strictly the needs of the newspaper. Since the publication had offices scattered in various locations, they planned to consolidate the newspaper in one central location. The radio station however, could exist almost anywhere in the city. It did not need to be located near the editorial offices. So the decree went forth that WQXR was to seek a new location. While this decree could have been received with groanings in some quarters, it actually turned into a golden opportunity for WQXR to move up to a strategically-planned facility.

NCC was in on this move from the earliest stages, indeed, even from before the project was conceived. NCC was originally retained by Doc Masoomian (who was previously chief engineer at WQXR) to provide out-of-house engineering services to the station. At that time NCC was also re-designing WNYC—another of New York's classical and news stations. Herb Squire (now chief engineer at WQXR) was then the telephone-communications consultant to WNYC. So even before the design stages, working relationships between NCC and the in-house engineering staff were being forged. The excellence of communication between staff and design consultant has been cited as an important factor in the success of this ambitious venture.

NCC's Al D'Alessio notes that WQXR's management wanted to make two statements with the move: First, they wanted to make a

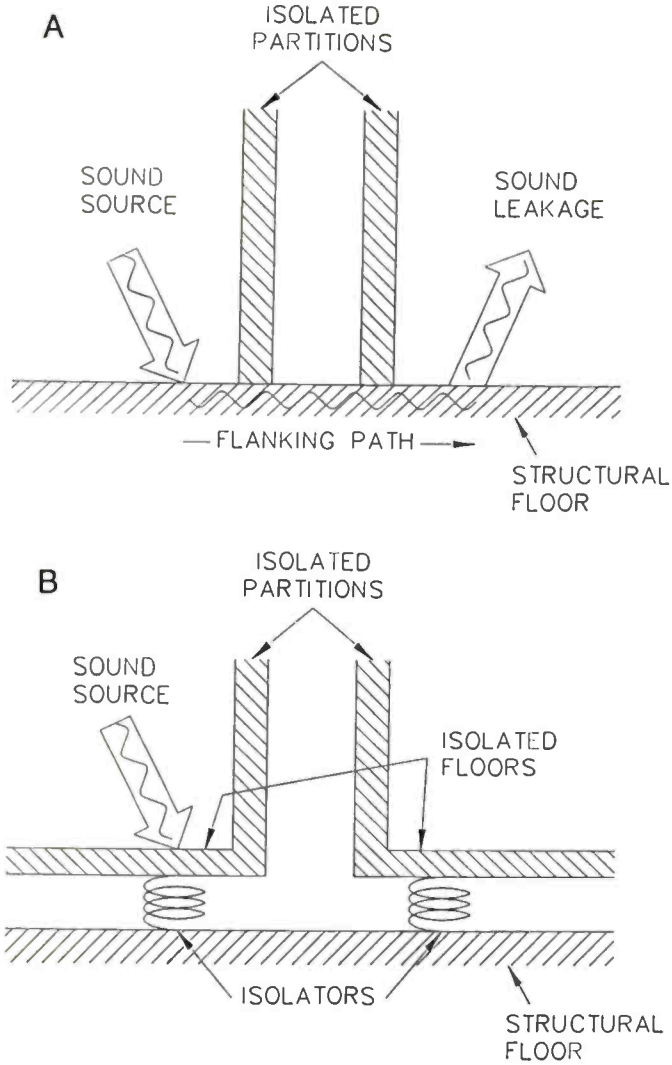

Figure 2. A (A) leakage can occur if partitions are fixed to the floor, but at (B) isolators between the partitions and the structural floor and the isolated floor do the full job. See Al D'Alessio's side bar at the end of this article for further information.

bold pronouncement that WQXR is the pre-eminent classical radio station in New York, and second, that the station was not at all content to sit on their laurels, but instead had their eyes fixed on increasing excellence. In order to achieve this, they required a facility that was technically without parallel, and aesthetically pleasing and user friendly as well.

As it turned out, WQXR's needs were almost a recitation of NCC's design philosophy. According to D'Alessio, NCC specializes in developing one-of-a-kind solutions to demanding design problems: "We put big systems together holistically, taking an 'operations-oriented' approach to designing a media facility. And we always try to have 'form follow function'." The advantage to the client is apparent. Rather than having to hire various individual specialists (designer, architect, acoustical and structural engineers) where the integrity of the functional theme can possibly get lost in a morass of communications: the client here deals with one central coordinator, in this case, D'Alessio—who is responsible for interpreting and clarifying the client's vision, as well as piloting the various specialists required for the project.

THE DESIGN PROCESS

The development of WQXR's new broadcast complex fell into some very definite phases. First, of course, was the choice of location, which is no mean task in New York City. As a whole, the city has a tremendous amount of microwave penetration, but some locations are better than others. The trick is to find a location with line of site to geo-stationary satellite orbits and WQXR's two transmitting sites. A structure capable of supporting massive acoustical construction and the world's largest (and therefore, the heaviest) private classical music library was also needed. While that immediately limits the number of possible locations, side-stepping major sources of RF is a helpful but less important consideration. With the world's most extensive underground transit system criss-crossing the city, putting distance between the facility and the nearest subway became another important factor. A building meeting these requirements was found in the downtown section on Fifth Avenue. It was considered a very fortuitous find since many media facilities and advertising agencies have recently relocated to that area.

The layout of the floor space that was available to WQXR is a "T" formation. This once again was seen as a stroke of good fortune. Since a radio station has three main operations, each operation was (more or less) allocated its own wing. The business office was to utilize the stem of the "T", whereas the horizontal wings were designated as a production area and a broadcast area (including both engineering and on-air personnel). The new location offered a substantial improvement over the old in terms of floor space as well: 18,000 square feet as opposed to 9,000.

But lack of space was not the only problem in the old facility. According to D'Alessio, the old WQXR was a poorly laid-out facility, "not intentionally—it just grew like topsy. When you entered the old 'QXR you went past a reception area for the auditorium (which was unattended), then down a very long "L"-shaped corridor. You ended up walking past all of the studios before you got to the receptionist." The result was a rather confused, inefficient mode of operation where personnel were walking large distances to interface with areas which should have been in immediate communication. Additionally, there was always the chance that unauthorized visitors might interfere with broadcast operations. The formula was not a happy one. It was simply a fact of existence that people learned to live with. Much money (salary) was wasted in what D'Alessio terms "walking wages." Thus, organization of space was one of the first design considerations for the new facility.

THE ARCHITECTURAL PROGRAM

The initial stage of NCC's involvement with the development of the new facility involved architectural planning. Before any electrical or acoustical considerations become possible, the available space must be divided. NCC sat down with

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WQXR's management and engineering people and said basically, What do you need in the way of offices? What do you need in the way of studios? And if you're not sure about that, then just tell us what you want to do and we'll tell you what you need. And so began a dialogue that lasted nearly seven months. Every aspect of the design was planned and sometimes even tested before any construction took place.

Architectural considerations were of major importance. If you put an extra window in a studio or an extra desk or cabinets in a control room, the acoustics would need to be adjusted. So to the extent that these things could be foreseen, they were figured into the design as early as possible. In practice, things changed somewhat as the plans were developed and new needs were ascertained, but once the basic layout was decided, acoustical calculations could then be started in tandem with the architectural development.

During this very critical planning stage, D'Alessio worked in close contact with architect George Ultan. Ultan generated a series of fifty-four drawings that covered every aspect of facility construction: from floor plans and wall sections to detailing, from structural changes to HVAC and electrical specifications. Ultan made many creative contributions to the project, but one is particularly worthy of note. Amongst the architectural considerations was the need to "humanize" this high-tech environment in such a way that it would become a low-key, non-threatening work space. This design objective was made more difficult by several factors: The nature of the "T"-shaped space was that very few offices had windows with available sunlight. Many were in the shadow of another building, and even more had no windows at all. Ultan solved this problem by utilizing a concept known as "borrowed light." Conceptually, what this means is that where no natural source of light is available, you create an aura reminiscent of sunlight by causing a mix of natural and artificial light from a neighboring space to leak into the room. By calling for clear glass transom panels between adjacent rooms (extending down from the ceiling for about two feet), Ultan gave the work spaces a more open feeling. Another technique he used to "warm-up" the entire facility was to call for a bold theme of American cherry wood trim and wainscoting throughout the entire facility. The natural wood in juxtaposition with studio equipment or even office accoutrements gives WQXR a coziness not usually found in broadcast facilities.

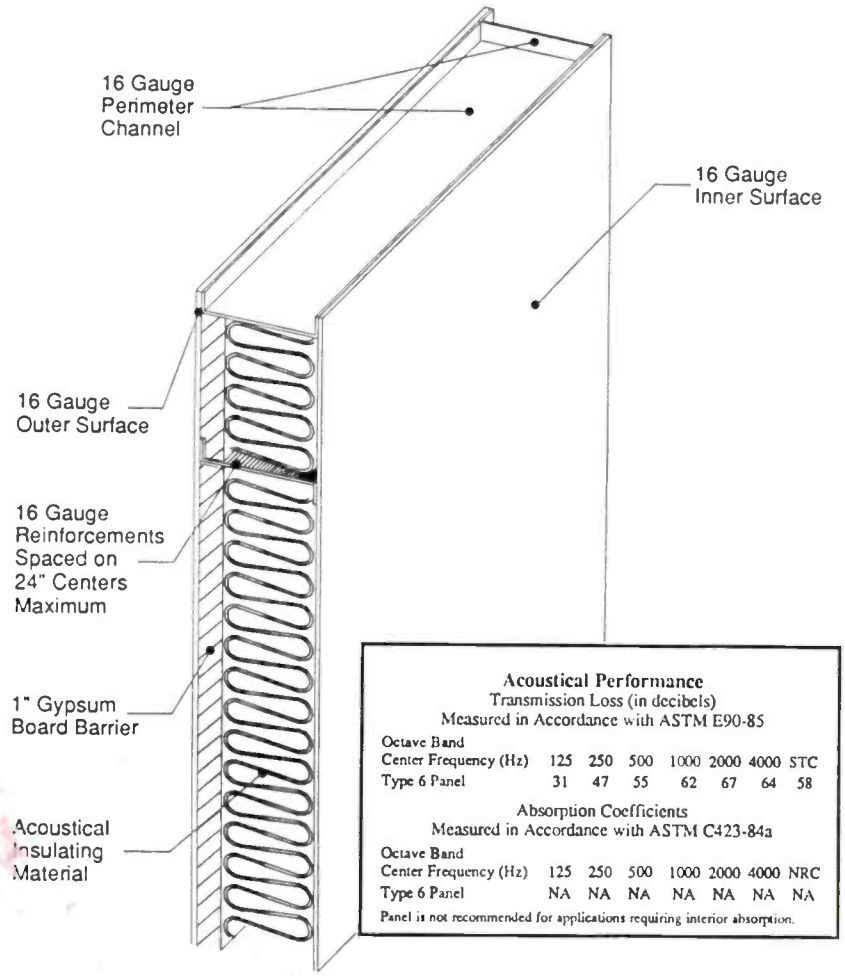

Figure 3. A cutaway view of one of the modular panels.

EQUIPMENT AND ACOUSTICS

NCC was given a large degree of latitude when it came to outfitting the studios. WQXR's chief engineer, Herb Squire, had only one requirement which was, as it were, laid in stone:

All broadcast consoles had to be from Pacific Recorders and Engineering. These units are reputed amongst broadcasters as possessing excellent specs and great simplicity of operation. But beyond this, Squire left it to NCC to suggest the majority of remaining peripherals. When several viable choices were available, NCC performed trial and evaluation on the pieces and made decisions with significant feedback from Squire. While T&E of the equipment was going on, NCC was also working out the acoustics of the rooms. Since NCC had control of the equipment and furniture selection as well as knowledge of virtually all other variables (like how many people are likely to be sitting in the room), they were able to consider these factors before making final acoustical calculations.

One of the most important design objectives was that adjacent studios and control rooms should have acoustical independence, in other words, work going on in one room should not have any interaction with the next room.

D'Alessio says he knew this goal was achievable, but as always, it took a bit of wrestling to make it a reality. He puts it like this:

"I would also have you appreciate the fact that when you're designing in New York City, most of the buildings limit your freedom in terms of

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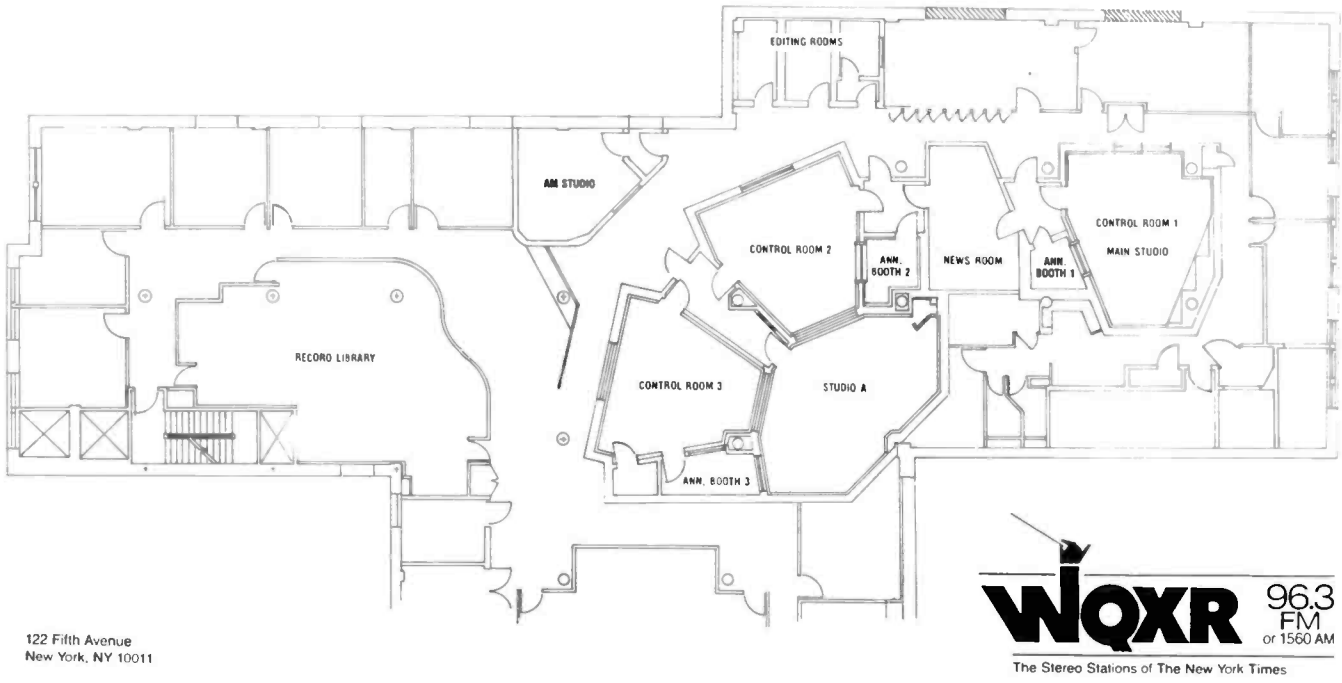

Figure 4. A detailing of the studio portion of the "T" as was seen in Figure 1.

where columns are located, what the floor bearing capacity for heavy acoustical construction is. All of that gets thrown into the same 'design soup' out of which we have to create an environment that's going to be productive for our client. Productivity is a good key here, because a lot of production-oriented enterprises (not just radio stations, but recording studios and production houses), a lot of them can't use two adjacent rooms because they interfere acoustically with each other. That doesn't exist here." Even at the outset, NCC was so convinced that they could deliver an attenuation of 60 dB between adjacent rooms that they issued an "acoustical guarantee" which specifies a substantial price penalty if they didn't meet the criteria. In other words they were promising WQXR that no more than 1/1,000,000 of the acoustical energy (at the human ear's most sensitive mid-range and high frequencies) would be allowed to enter any adjacent room. Music can therefore be played at some rather ear-splitting levels in a control room, yet be barely audible in the adjacent studio.

FLOATING INTO SILENCE

How was this 60 dB figure achieved with all the factors that were aimed against it? Floating construction, of course. (*See sidebar.*) While there is nothing revolutionary about floating construction, NCC did develop an interesting new wrinkle on an old scheme. All of the control rooms and studios at WQXR (with the exception of the AM control room) are floating. But where is the typical isolated concrete slab? There is absolutely no concrete slab to be found. The rooms are not floating on concrete or even on wood. Al D'Alessio explains: "All of the rooms are their own separate modules, which were actually built in a factory, made out of four basic ingredients: steel, gypsum, fiberglass and (where required) laminated acoustical glass." The result is the acoustical equivalent of 12 inches of solid concrete in only 4 inches of space, and a fraction of the weight!

The factory-built modules are in several ways superior to any similar kind of site-built construction. One very obvious advantage can be immediately seen by looking at a sectional drawing (*See Figure 3).* Notice that the modules are a sandwich of gypsum and fiber-glass, in varying layers and thicknesses, bounded by an exterior of rigid steel. The units are bounded on all sides by steel, and are precision built units, easily fitted together. This, of course, makes acoustical sealing a rather easy job. One interesting side effect of the metal-bounded modules is that as large ferric surfaces, they

Figure 5. Studio A. This is also seen in color at the beginning of this article. Both the ceiling and the walls have adjustable acoustical panels.

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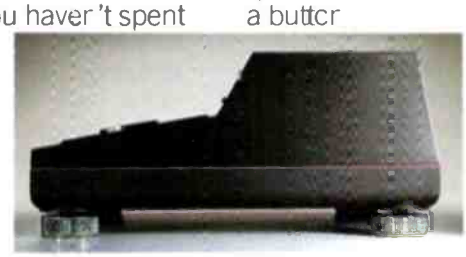

You see, the SL-P1300 is ergonomically designed to give you greater control over playback than you've ever had before.

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Figure 6. NCC's Al D'Alessio at the piano in studio A. This served as an excellent sonic example of the fine acoustics of the studio.

offer a certain amount of Faraday shielding against stray RF, and as Al D'Alessio pointed out, "if the windows and doors had been outfitted with RF screws, the whole studio structure could then be earth-grounded offering even further protection. However, in the case of WQXR, which was already in a relatively quiet RF location, the process would have been engineering overkill."

The typical size of the modular panels was about four feet by eleven and-a-half feet. Panels were used not just for walls, but because of their structural integrity, they were used also for floors (sitting on neoprene isolators) and ceilings as well. In this case, the entire room floats—not just floors and walls as in most conventional applications. D'Alessio explains:

"One of the problems with floating rooms conventionally (where you pour a concrete floor on top of an isolated plywood form) is what do you do with the ceiling? Traditionally, ceilings have been hung by neoprene and spring isolators, which gives you a firm attachment to the floor above. If the floor above is noisy, or has mechanical equipment on it, noise and vibration will be transmitted through the isolator because there is never enough mass in the ceiling that the isolators support, for the isolators to really do their work properly.

"The modular ceilings that form the roofs of the studios and control rooms at WQXR have no mechanical connections whatsoever to the floor above. They are supported entirely by the walls and floors of that unit, which sit on a floor that WQXR has total control of."

While the modular construction forms the "inner-box" (rigid, acoustically sealed, and supported entirely by a floating floor), the addition of an independent "outer-box" with a dampened air space in-between is what enabled NCC to deliver the 60 dB attenuation between acoustical spaces and the outside world. While the critical acoustical interface between studio and control room involved two spaced modular walls, the less critical boundary between control room or studio and hallway utilized a site-built secondary wall. To increase attenuation, the adjacent walls were constructed with different interior thicknesses, and even the air space between them was damped with fiber-glass. The same concept of damping through use of different materials and thicknesses was also carried through on the control room windows. The glass chosen was a laminate of varying thicknesses of glass and plexiglass. Two such glass panels were installed for each window with a damped air space in-between.

With studio isolation well in hand, the room interiors were tuned to control the absorption and dissipation of sound in accordance with the population and use of each space. Here, the American Cherry hardwoods were fashioned into Helmholtz resonators to control low-frequency absorption and to complement the absorbent acoustical properties of the fabric-wrapped wall panels and splayed ceilings.

D'Alessio is resolute about not applying live/dead end acoustics to radio station control rooms. In radio, the control room is also a studio and must be treated with mic'ing as well as monitoring in mind. Since the largest radio control rooms feature less than 3000 cubic feet of volume, control of early reflections by attempting to create a reverberant field confined to a contiguous volume of less than 1500 cubic feet is suicidal. Ironically, the WQXR live performance studio is marked by one of the most popular acoustical materials among LEDE designers. The ceiling features a 200 square foot field of acoustic diffusers, custom built by RPG to NCC specifications. The RPG's, complemented by adjustable room acoustics, permit the studio to be used simultaneously for voice and live music broadcasts and recordings.

QUIETING THE AIR

Air conditioning isolation was accomplished quite easily, says D'Alessio, by doing the following things:

First, "placing the (ducts) entry points where the acoustical flanking (leakage) path would be the least from any adjacent room—as far away as you can get it from any potential noise source."

And second, "use acoustically-lined duct which will attenuate any noise that can seep into the duct itself—including it's own noise."

Third, keep air velocities low with ducts having a large cross-sectional area. Beyond that, he specified using a duct two gauges thicker than that used in normal commercial installa-

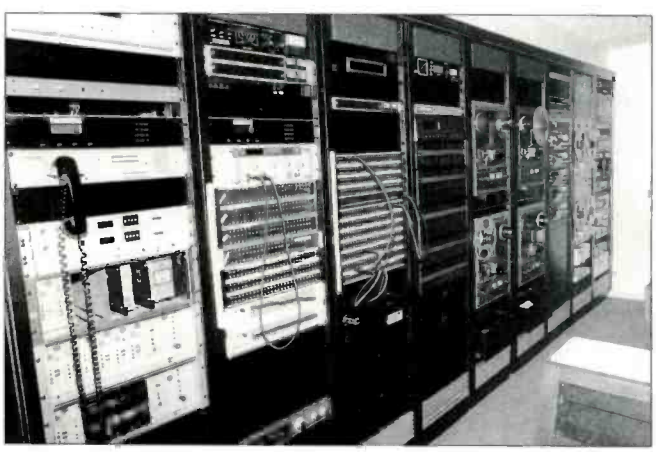

Figure 7. The equipment bays in the Terminal Room. All the gear is fully accessible from the rear via semi-concealed doors (not seen here).

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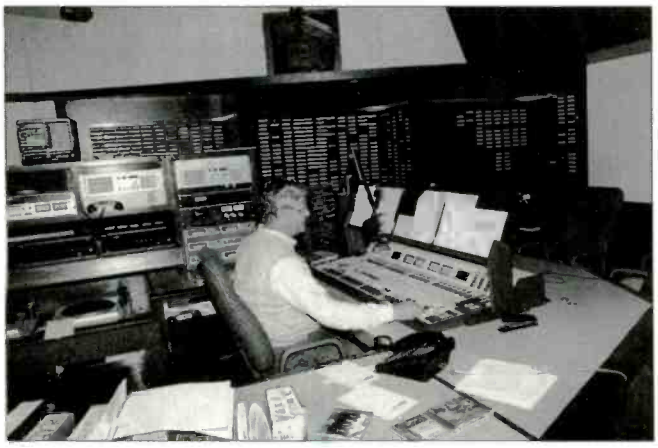

Figure 8. Control Room 1. Announcer Matt Edwards can reach everything that he needs on-air from his seat. Note the cart storage racks on the wall.

tions. Duct attenuators (which are somewhat analogous to a car muffler) were also implemented. When done just right, the result is air conditioning you simply cannot hear. As he graphically puts it: "It's just like somebody is taking a bucket of cold air, gently tipping it over and dumping it into the room."

MIC LINES AND TIE LINES

Any attempt for an acoustical isolation is limited by the weakest link in the chain. If you have achieved an STC of 60 dB between rooms, and then poke a hole between the rooms and create a mechanical attachment by running wires between them, the isolation could suffer by as much as 30 dB. NCC took no chances in this matter. In fact, "there is no firm connection between any of the walls of the inner control rooms and studios to the outside," says D'Alessio. Wires were generally suspended over ceilings or under floors and dropped straight into the room without piercing the wall. "If it was necessary to traverse the gap between inner and outer walls, there would always be a "compliant component," that is, a flexible conduit. Use of such a flexible conduit where cabling exits or enters a room serves to de-couple it and hence preserve the isolation.

Since stray capacitance can be a problem in running long lengths of cabling, NCC specified 26 gauge wire instead of the normal audio 22 gauge. Since long parallel runs of wire can be begin to react much like a capacitor (oppositely charged plates separated by an insulator), a decrease in wire gauge effectively reduced the surface area of the (inadvertently formed) capacitor "plates."

GETTING GROUNDED

Probably the most troublesome threats to a studio's signal-to-noise ratio come from grounding problems. Many studios still endure the illegal and potentially dangerous practice of lifting all electrical grounds. This is mainly a stopgap measure, because the grounding system was never properly designed. In

order to achieve hum-free operation properly, NCC worked out a carefully executed integrated ground scheme where both audio and electrical grounds were referenced to the same ground point.

As D'Alessio points out: "Ground problems exist if either electrical or audio ground is seen by the equipment as being at a different potential due to different resistances to ground. If however, the ground path resistance is made to be less than 1 ohm, there will be no problem."

This was achieved by implementing a classic "star-grounding" configuration for all audio grounds and tying it in at the origination point for the electrical ground. A pleasant side effect of a properly grounded studio is that it does offer some additional RF protection.

FINAL APPOINTMENTS

It's the final touches that really show off a good design. The WQXR installation has its share of them. For example, there is automatic logic switching built into all recording consoles in all control rooms. Anytime a microphone is opened up, "QUIET PLEASE" signs are lit all along the path to the designated area.

There are no "ON THE AIR" signs at WQXR. NCC advised against it, because there are more far more times a studio is used for production then for being literally on-the-air. It was reasoned that it's better to inculcate considerate behavior amongst personnel and visitors than blatantly lie and say you are on the air when you are not. It's perhaps a small point, but it fits well the low-key ambience of the facility. Another small but noteworthy appointment is the use of acoustical analog clocks. Their oil-damped mechanisms are smooth and totally noiseless and of course, all tied together by house sync.

Another consideration is that all floors (being metal clad) are grounded and carpets are computer-grade, which are designed to dissipate static charge. Along with this, humidity is strictly controlled so that nothing will interfere with microprocessor controlled equipment.

Monitoring the various systems that are incorporated in the broadcast chain is humanly speaking, an impossible job. With signal routing chores inside the studio, network feeds from the outside world, and links to transmitter sites for simultaneous FM and AM broadcasts, there are over 120 potential trouble spots. Many of these systems could be down for hours before any human operator was aware of a malfunction.

To avert this kind of broadcast horror story, NCC designed and built a "custom annunciator system" comprised of electronic bulletin boards which keep the engineering staff constantly apprised of the status of each potential trouble spot. If a malfunction should occur, it will specify the location, the nature of the problem, and offer an appropriate course of action to remedy the situation.

NCC's comprehensive set of electronic design documentation paid off by shaving five-and-a-half weeks off the completion date. Without a hitch, Spectrum Broadcast Inc. was able to install and test the equipment

required to get the station cut-over from their old studios in three-and-a-half weeks instead of the originally scheduled eight weeks.

Together with the architectural plans, the documentation will facilitate the maintenance and future growth of the facility well into the future.

All things considered, NCC has created a virtual dream-machine for WQXR. The station is user friendly, easy on the eyes and very, very quiet.

With this sort of edge, WQXR will undoubtedly remain the pre-eminent classical music station for quite some time.

Why Float Acoustical Studios?

Alfred W. D'Alessio, NCC Inc.

Most radio and recording engineers have long realized the value of floating construction. By suspending the walls, floor, and ceiling of a studio or control room resiliently, vibrations in the host building structure set up by mechanical equipment, air conditioners, foot falls, door slams, etc., can be isolated from that room.

However, an often overlooked requirement for floating construction concerns the limitation of sound transmission properties of the partitions separating adjacent studios caused by the building itself. As can be seen in *Figure 2(A)*, even if double wall partitions are used between rooms, some of the sound energy originating in one room can be transmitted through the host building itself, and reappear as sound leakage in another area. The floors and ceilings common to each room are excited by sound waves setting up vibrations in the structure. Because sound or vibration travels over ten times faster and more efficiently in wood, steel, and concrete than in free air; any structural element common to two adjacent rooms becomes an efficient pipeline (flanking path) for transmitting sound around intervening partitions of any design.

When the same two rooms are floated, they have no firm mechanical contact with each other, or with the building structure. As a result, they will be isolated not only from building vibrations, but also from themselves as depicted in *Figure 2(B)*, with the isolators serving as attenuators in the flanking path. Regardless of the materials and cost of the intervening partitions, it is difficult to achieve more than 50 dB of isolation between rooms at 500 Hz without employing a floating constuction. db

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ED LEARNED

Sound Reinforcement in North Africa, Part III

TUNISIA

• Tunisia was another bookend stop for two groups: Terrance Simien began his tour here, while the Sun Rhythm Section ended theirs. The former arrived in Tunis from Paris via Air France; the latter from Jordan via Royal Jordanian Airlines.

Customs clearance proved easy in both cases; entry visas were not required here. Tour scheduling in-country was similar for both groups: we played the southern cities first, finishing up in Tunis at the end. Cities visited were the same except for Kairouan, which was played only by the Sun Rhythm Section.

Kairouan

This city is a hundred miles southwest of Tunis, about a two hour drive due to traffic and road conditions. We played the Maison de la Culture, a facility of small art shops and outdoor display areas containing a small, rectangular-shaped indoor auditorium seating seven hundred in folding chairs. The stage was about three-feet high, and played to the room's long dimension.

It was also raked slightly down from upstage to downstage a la Shakespearean theaters. Remember to block those casters! The room was completely hard, with wood and plaster surfaces, so the reverb time was over 2 seconds. Power was available offstage right from several European-style grounded receptacles; voltage was between 218 and 230 with frequent fluctuations. I went with minimal drum mic'ing here due to the small size and liveness of the hall. Our crowd at show time consisted of twenty people, so we decided to hold an extra half an hour to see what developed. At 7:00 p.m., five minutes before we'd decided to start no matter what, a tremendous influx of kids, around junior high age I'd guess, arrived out of thin air and filled the place to capacity. I even had a circle of kids around me, watching my every move and trying to figure out what exactly I was doing with all those knobs and faders.

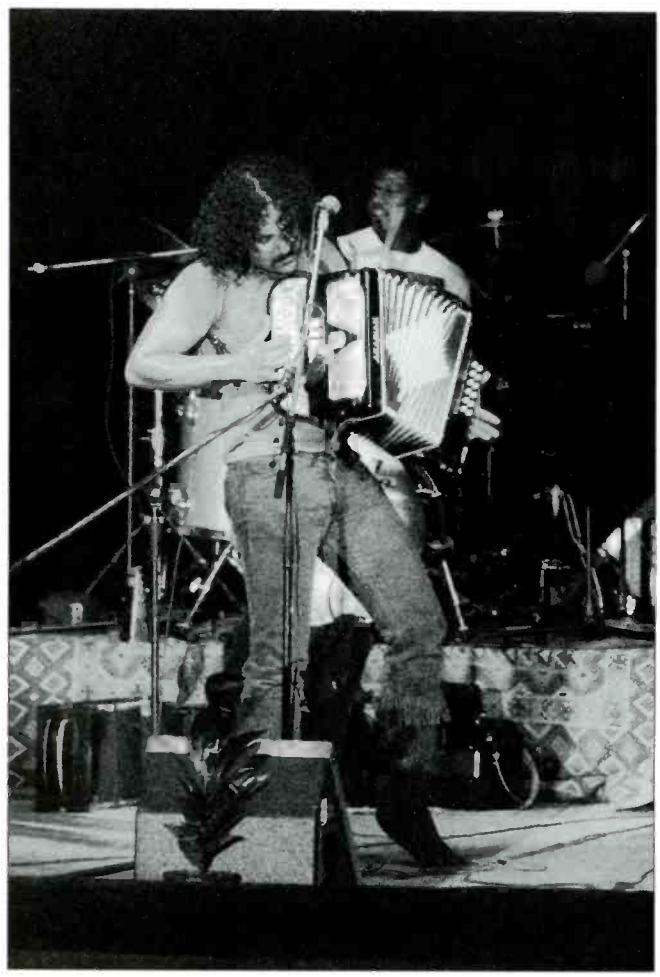
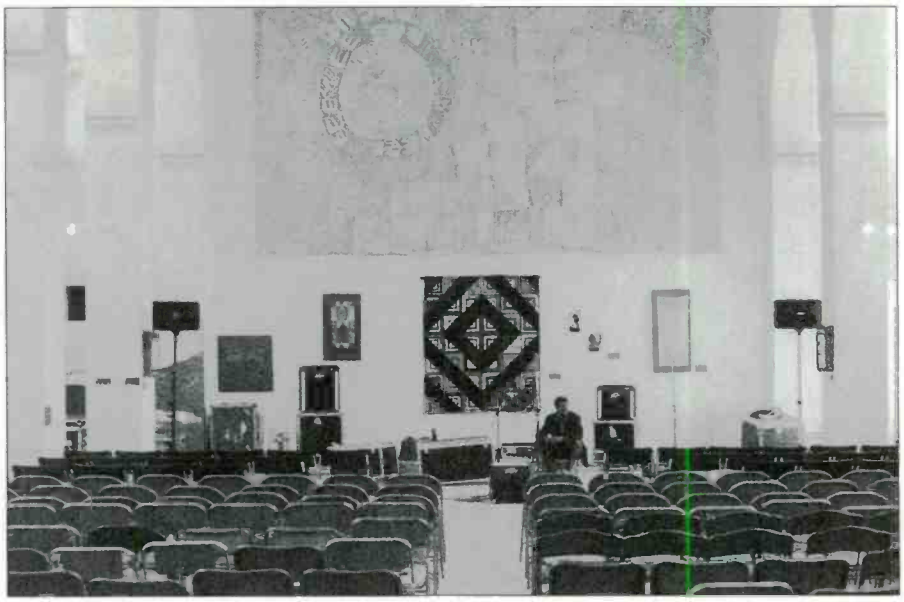


Figure 1. Terrence Simien in performance at the Municipal Theater, Tunis.

Sousse

A resort city on the Mediterranean, Sousse is a three hour drive slightly southeast of Tunis. Spend some time there sampling the lively night scene, and you'll understand the inspiration behind Dizzy Gillespie's tune "A Night in Tunisia." Terrance played the Sousse Palace Theater, a second floor movie house that seated five hundred in a slightly raked main floor area. There was no elevator, so the gear had to go up two flights of stairs. These were fairly wide, but it was an unpleasant experience to say the least. Plentiful wood surfaces and an uncarpeted floor made for "hard" acoustics, with a reverb time of 2 seconds. Power was available offstage right on a European receptacle that supplied 220 volts, but there was no ground. I found a ground wire coming up through the floor beneath the power board and tied to it. The plan for the day involved setting up around a 3:00 p.m. movie, which meant we would have half an hour to set up and sound check before we were scheduled to begin. Ron Minninger,

Figure 2. View of the Centre d'Art Vivant, Tunis, Tunisia.



CAS (Cultural Affairs Specialist), Jamil Halfaoui and I huddled with the theater manager about this conflict in scheduling, and it was agreed that the afternoon movie would be cancelled in deference to our needs.

With a small space to consider, I elected to let the drums and bass go unmic'ed, mixing the band to their acoustic level. This let me get a good sound without crushing volume. A crowd of three hundred attended the show, which quickly became a dance party.

A local policeman attempted to sit the kids down, and actually tried to eject two guys who wouldn't listen! A quick conference with USIS officials and escort Minninger reassured him that it was okay for the audience to dance; the band wouldn't be offended. The concert continued, and a good time was had by all.

The Sun Rhythm Section played the Sousse Palace Hotel, about a block south of the Sousse Palace Theater. We performed in their ballroom, a semi-circular space oriented around a like-shaped stage. Seating capacity could be as high as 1000, although we didn't "play" to that much of the room. The rear wall of the ballroom was a series of windows, which made for some nasty slap-back echoes onto the stage. The room was partially carpeted, but its large size created a reverb time of almost 2 seconds. Power was available on a grounded European outlet stage left, but the house electrician informed me it was only 5 amps, which was not going to cut it for the amount of gear we had. There were power lines for lights in the ceiling over the stage; I had him tie an outlet into an unused circuit where I could get a real 20 amps of 220 volt power. The hall was set for seven hundred fifty seats, yet only about one hundred fifty people showed up; our local sponsors attributed the poor turnout to competition from the many other events going on that night in Sousse.

Sfax

The ninety mile drive south from Sousse can take as long as two hours if you stop a lot for crossing sheep. Both Terrance and Sun Rhythm played the Municipal Theater, a movie house with a stage. Seating capacity was a thousand, seven hundred in the main floor area which included some mid-floor and rear floor "private boxes," and three hundred in a balcony. The theater was on a second floor; equipment was brought up via a chain motor hoist located at the rear of the stage. This is a rope-and-hook arrange-

Figure 3. Terrence Simien and the Mallet Playboys at Ewart Hall, Cairo.

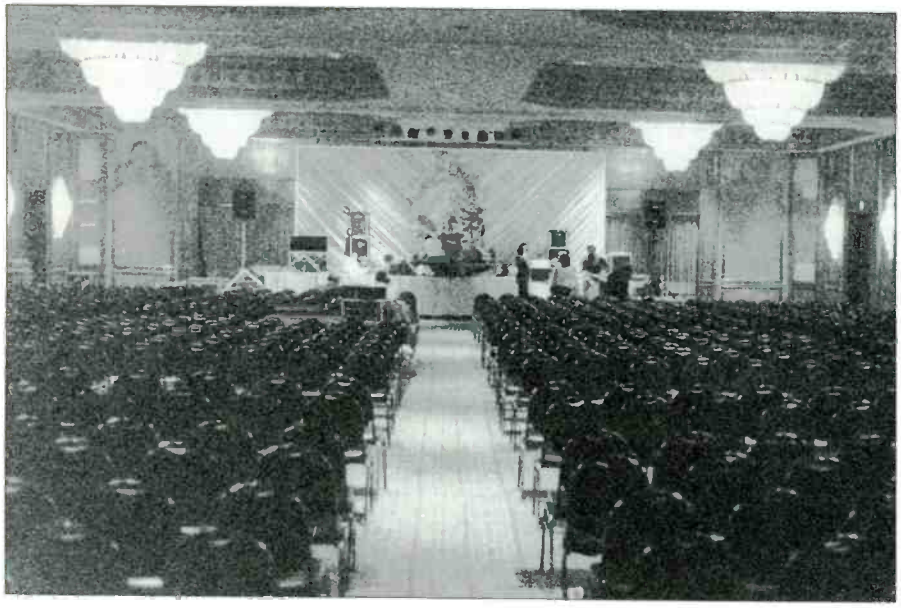


Figure 4. The Aida Ballroom in the Cairo Marriott.

ment that handles one piece at a time, so load-in can take some time. The plush seats and thick carpeting really dampen reverb—reverb time was around 1 second, and overall the acoustics were very smooth and pleasing. I felt it had the best acoustics for an electric band in Tunisia. Power was available on the offstage right wall: 220 volts on a European-type receptacle. These outlets had no ground, but there was a ground lug, located directly beneath the power drop. I set up my mix point in one of the mid-floor boxes house left, giving me both an ideal location for audio and excellent security. The only real drawback to this venue was the movie schedule, which went until almost 6:00 p.m.—this required a late 9:00 p.m. show time to allow for setup time. Audience response to these electric groups was excellent. When Terrance and Sherm asked the crowd to stand up and party after the second number, they never did sit back down! Sun Rhythm had the audience singing along enthusiastically during *Blue Suede Shoes*.

Tunis

Both groups played a major concert at the Municipal Theater that was recorded by Tunisian TV This was an older hall in the European opera house tradition. It seated twelve hundred, including three wrap-around-the-floor-area balconies. There were private boxes at the rear and sides of the floor area as well. Plush seats and carpeting again dampened the room—the reverb time of 1.3 seconds was longer than Sfax due to the higher ceiling. Power was available from stage pockets offstage left. Each pocket contained three European grounded receptacles, but only the middle one of each set actually worked—supplying 220 volts of very stable power. The theater-proper was above street level, so gear came up on a rope block-and-fall hoist through a portal in the floor offstage left. Mix point location proved a bit controversial on my first visit here with Terrance: I'd asked for seats in the floor area, about ¾ of the way back on a center outside aisle. These seats were reserved for my use, yet the hall manager demanded that I set up in a private box at the rear of the hall. The fact that my snake wouldn't reach mattered not at all—this was the way it was always done for safety reasons. After two hours of stonewalling, the fire marshall was finally summoned to solve the dispute. After surveying the situation, he assessed my plan as perfectly safe, and gave it his approval. Fortunately, we didn't have the same

Figure 5. The Jay Hoggard Quintet at an aternooon rehearsal soundcheck at the University of Assuit, Egypt.

Figure 6. Terrence Simien at the Outdoor Auditorium of the Kartoum Army Officer Club.

problem a year later with Sun Rhythm, as there was a new, more flexible hall manager! I gave the TV people a straight PA mix feed; the acoustics were so good and the hall so large that I had everything up in the mix. Tunis audiences were generally "hip" to Western music and ready to have a good time. Both concerts were sellouts, with feverish audience response (*see Figure 1*).

So many people were up dancing to Terrance's music that the TV cameras were blocked. The Sun Rhythm concert had much higher security, with ushers and police preventing any dancing on the main floor in deference to the TV people.

This dampened the fun somewhat, but the people in the balconies were free to boogie the night away—and they did with a vengeance!

Terrance played a smaller concert in Tunis, at the Centre d'Art Vivant. An exhibition of American art was there at the time, and USIS-Tunis thought that a concert of American music would nicely complement this show. The exhibition hall was great for art but lousy for music; the marble floor, plaster walls, and high ceiling combined to make the room deathly bright, with a reverb time of 2 seconds despite its small capacity of 250 (*see Figure 2*).

AC came from a basement European-style receptacle. Voltage fluctuated between 190 and 205 volts, very low for our 220 volt sound gear. There was no ground anywhere; I pounded a copper pipe into some pre-moistened ground behind the arts complex to form a ground rod, and tied my ground to that. I radically cut back on the PA and monitors to make the most of what little power we had. The room was so small that I only amplified vocals and accordion. We even turned Sherm's guitar amp around so that the amp fired into the wall, not the audience. A capacity crowd of invited dignitaries helped our acoustics marginally, but it was the cooperation of the group in consciously holding back on their sound that really allowed us to pull it off.

EGYPT

My very first trip to Cairo in 1984 (with the late jazz trumpeter Woody Shaw) illustrated how the local political climate can affect your travel. Arriving from Syria, we spent three-and-a-half hours in the airport while customs thoroughly took us apart. Relations between Syria and Egypt, strained at the time, translated into rough treatment if you had the misfortune to transit customs. The 1985 Jay Hoggard tour arrived in Egypt from Morocco via Rome.

Figure 7. Again Terrence Simien and the Mallet Playboys, this time at the Ambassador's Residence in Kartoum, Sudan.

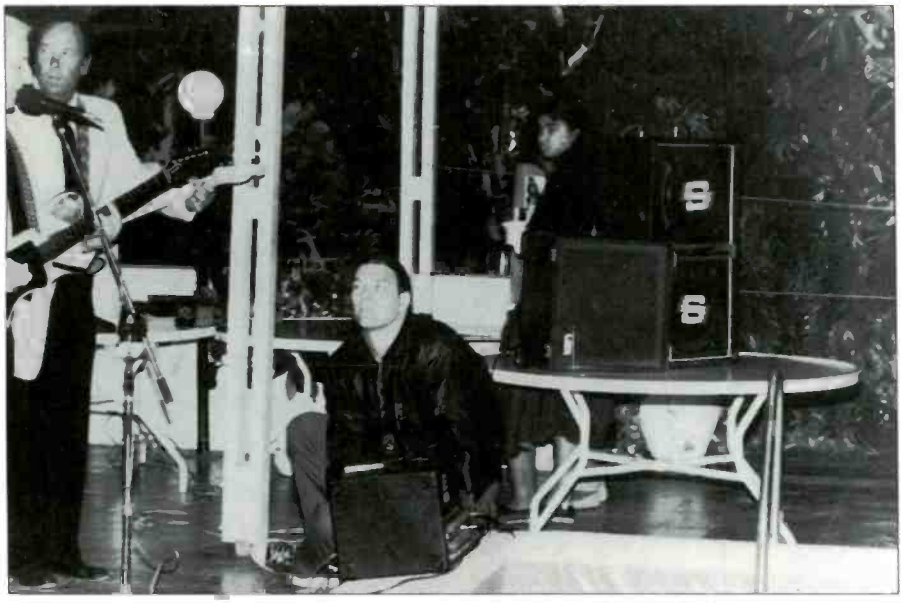


Figure 8.The author mixes on borrowed gear at the Ambassador's Residence in Kartoum. Main and side-fill speakers are visible.

The 1988 Terrance Simien tour came from the Sudan. In each case, we encountered none of the hassles I had on my first arrival.

Knowledge of Egypt's relations with your country of embarkation will give you a clue as to the degree of "hell" you can expect in the Cairo airport.

Cairo

Terrance Simien played two different venues here. Our first was at Ewart Hall, on the campus of the American University of Cairo. This was a theater that seated 1000, including a small balcony. The room was very live, with a reverb time of just over 2 seconds. AC power was available offstage right, from UK-type receptacles that supplied 230 volts.

The ground was not functional, however, so I tied to a water pipe in a bathroom offstage right. The large stage gave the band ample room to spread out, handy because the concert was taped for radio and Egyp-

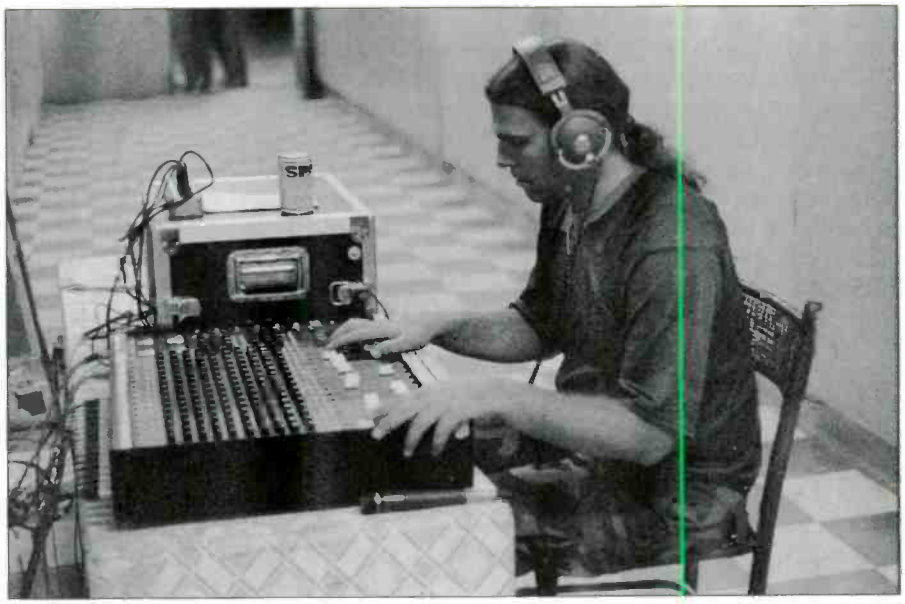


Figure 9. The author is seen mixing Jay Hoggard at the Omdurman TV studios.

tian TV Channel 3. The radio people decided to put up their own mics and take a feed from me, so the stage soon looked like a forest (*see Figure 3*)! The TV people elected to just take a feed from me. From a band standpoint, the show was a great success: people went wild, many of them jumping up on stage to dance. The TV people had a rough night because of this; so many people were dancing on the apron that you couldn't see the band.

The radio mix turned out great, and I was pleased to hear that the radio engineer eventually muted most of his extra mics, electing to use my PA feed as his primary source.

The second Cairo concert was held at our hotel, the Cairo Marriott, in the Aida Ballroom. This was a rectangular room with a low ceiling, seating about eight hundred. We played to the long dimension of the room from a fairly large portable stage (*see Figure 4*). Reverb time was a very manageable 1.3 seconds.

Power was available from UK-type receptacles along the walls, but the house electrician gave me a separate receptacle on a power board with its own 30 amp breaker, this supplied a very stable 230 volts. I had my Bose speakers, on their tripods, placed on sound wings to safely get the extra height I would need to insure projection of sound to the back of this long room.

Our one bit of excitement came from a bomb threat; the hall was cleared for two hours while police swept the place. I was permitted to join them after an hour to reset all the audio gear that was moved. Another capacity crowd turned the concert into a happening. After several songs, Terrance pointed out that the area in front of the stage had been designated for dancing, and he wanted to see it jam-packed on the next number. He got his wish: the crowd never sat again for the rest of the evening.

Alexandria

Both Jay and Terrance played concerts here, arriving by different means. Jay's group travelled by train from Cairo; Terrance by bus. In both cases, the sound equipment and band gear was shipped via truck. Jay Hoggard's concert was held in the auditorium at the El Nasr Girl's College. This small hall seated around six hundred, but what made it memorable was its horrendous reverb time of 3 seconds—amazing for so small a room. Everything in it was wood and plaster, there wasn't an absorptive surface anywhere. There were various UK-style outlets around the stage; the first one I tried, offstage right, measured 75 volts! I finally found one in an alcove entrance offstage left that measured 230, but was really two 115-volt hots and nothing else. I elected to use a single 115-volt hot, tying my neutral and ground wires to a water pipe outside the hall on a wall. We now had a proper 115-volt system, so I needed to bypass all the internal step-down transformers installed in our band gear. I also had to obtain a step-up transformer to run the Crest power amps, which were permanently wired for 220.

Sound check was a real challenge: the live room made it difficult to get any clarity; I could only pray for a full house. I did have the curtain at the rear of the stage pulled down, which helped dampen stage reflections enough so that the acoustic piano could barely be heard. I was forced to completely close the piano's lid to get any usable gain, something I hate to do. Fortunately for me, our concert sounded better; we did have a full house, which allowed me to sneak some of the instruments into the mix. One interesting note: at the end of the first set, Jay announced an intermission in English. Most of the crowd got up and left, thinking the concert was over! The ushers managed to intercept some of the audience with the news that we were only taking a break, but we lost half our audience.

Terrance's concert was held at the Sayed Darwish Theater, an old opera house that seated seven hundred fifty, including two wrap-around balconies. Lack of absorptive surfaces contributed to a reverb time of 1 and

Figure 10. The Jay Hoggard Quintet, again at the Omdurman TV studios. Note the PZM taped to "tune" flat.

¾ seconds that sounded especially muddy in the 200-400 Hz area.

Power was obtained offstage left from a UK-type receptacle: voltage was a single-hot 220, but there was again no ground.

I tied my ground wire to a bathroom faucet about twenty feet from the offstage left wing.

There was a small house PA system available, based on Altec components, but I elected not to use it.

The concert was completely sold out, so I was able to mix more aggressively than I had during soundcheck.

The crowd went wild when Terrance and the group launched into *La Bamba*; we discovered later that the song was number one in Egypt at the time.

Assiut

This city is located about halfway down the Nile River from Cairo to Aswan. Assuit University, the largest university in that part of Egypt, sponsored Jay Hoggard, who was the first American artist ever to play Assuit. We travelled by train from Cairo, a journey that took a little over five hours; our equipment travelled separately by truck. Our venue at the university was a large gym, seating around two thousand. We'd encountered long reverb times on this trip, but this gym was by far the worst: with a time of 4 seconds, any sound was soon rendered unintelligible. A power board offstage right supplied 220-volt power through a strange 4-prong connector. I had the school electrician wire me directly into this service. The band spent most of our afternoon soundcheck working on two new tunes: *Secret of the Desert*, by Onaje Gumbs, and *Jamal's Dream*, by Vernon Reid (*see Figure 5*). The rehearsal confirmed my opinion—this hall would make a great echo chamber!

The group sounded like they were at the bottom of a well, and it was almost impossible to hear the acoustic piano over the din of the other instruments. The guys had turned down as low as they could, but it was still bad. I explained the situation to Onaje; he agreed to only use acoustic piano on quiet numbers. Otherwise, he'd play the DX-7 exclusively. The addition of an audience helped enough so that we were at least partially intelligible.

To add to our woes, Jay's vibes started cutting out in the middle of the second set; he was soloing with one hand while reaching under the vibes to check his pickup with his other hand. He told me later that some screws holding the wires in place had vibrated loose during the long truck ride from Cairo—causing the connection to come apart. He managed to get it working long enough to limp through the rest of the set. A rough night, but a most receptive audience made up for it. The regional governor was in attendance, and he might have been the most enthusiastic of all!

SUDAN

All three groups played the Sudan, primarily Khartoum and the surrounding areas. Conditions in-country were very different for each tour. Jay Hoggard's 1985 tour left Sudan only five days before a coup toppled the Nimery government. Tension was so thick you could cut it with a knife. The Sun Rhythm section tour took place in 1989, and fell afoul of Egyptian-Sudanese relations. Sudan was the first stop on this tour; we arrived from Amsterdam, while the sound system and band equipment was to be shipped from Paris via Cairo.

Due to the unrest in southern Sudan, Egypt has placed an embargo on Sudan—only relief supplies were

allowed to go from Cairo to Khartoum. Despite delivery assurances from the air-freight company, our gear was off-loaded in Cairo, where it sat for five days.

We were forced to do the first concert without our gear, causing some last minute improvisation by your's truly. A word to the wise: if you need to get equipment into the Sudan, make absolutely certain it does not require connections in Cairo. Our equipment was released only after intervention by highly-placed friends of the U.S. government. Carriers like KLM or Lufthansa, that offer direct flights to Khartoum from Europe, are your best bet.

Khartoum

Jay Hoggard's major concert in Khartoum took place at Friendship Hall, a major theater/arts complex built for the Sudanese by the People's Republic of China. The main theater seated twelve in a two-tiered main floor area. It was also a working movie theater, so my setup had to be complete before the 3:00 p.m. movie was shown. I set my stacks and house-mix equipment in place, uncrating monitors and band gear, which I stashed behind the screen to facilitate a quick set-up after the movie.

The room contained a lot of tile, which resulted in a high-frequency reverb time of 2 seconds. Power was available stage left from a stage pocket; it contained two UK-type receptacles, each with a 230 volt, 15 amp grounded circuit. I left around 2:30 p.m., returning with the group at 5:15 p.m. to finish setup and soundcheck. I was pretty happy with the sound here, except for the bass—the room seemed to eat up all the level that I could put out. To compensate, I had Jerome play louder than usual, with added low frequency on his stage amp. Out front, this made the difference in maintaining the bass impact of the overall mix. The evening show was recorded by Radio Omdurman, taking a feed from me via a post-fade aux send. I let my radio mix follow my live mix in this instance, using the aux pots to compensate for the drastically different needs of radio. Both Terrance Simien and the Sun Rhythm Section played the Khartoum Army Officer's Club outdoor auditorium. This facility featured a semi-circular stage, partly covered by a high roof. The audience area was constructed of steeply-raked concrete tiers, rising high above the stage, with seating for five thousand in portable chairs or tiered standing areas (*see Figure 6*). Since the audience area was open air, reverb time was not a factor. The concrete environment did, however, make things slightly bright.

Power came from a panel on the stage right wall: 240 volts on a UK receptacle, but no ground. My ground came from a water pipe

about thirty feet from the stage right side. At this facility, I was hurting for PA—no way do four Bose 802s produce the sound you need for two electric bands outdoors for five thousand people! Handling this space required careful balancing between the band and the PA. Both bassists played "up" as far as they comfortably could, and played with extra low end. Guitar amps were raised on cases and run hotter for Sun Rhythm. We had two guitar amps for one player on Terrance's tour. At the officer's club, Sherm used both amps, angling them to try and cover the widest house area with stage sound.

I tried to get as much out of stage sound as I could, using the PA for vocals and drums; I would then add all I dared of the rest to round out the mix. We did, however, get the idea across. Terrance enjoyed a crowd of twenty-five hundred wildly enthusiastic music fans. Sun Rhythm completely filled the place—the first time it ever happened for an American group. The difference in sound level between the two would make anyone a believer in the ability of people to absorb sound.

Another venue shared by the two bands was the U.S. Ambassador's Residence, where both Terrance and Sun Rhythm were entertainment/guests at an invitation-only concert/party. Invariably, two things happen: you meet many important dignitaries from the diplomatic, governmental, and artistic circles, and then discover that they like a good time as much as anyone!

We played next to the pool, using a corner near a flowered trellis as a stage. A seating area along the pool was covered by one PA stack; the other was pointed at a different seating area with a dance floor (*see Figure 7*).

My mix point was located at the rear of this second seating area. Power was supposed to come from the pool compressor shed, but the voltage there was 268, way too high for our gear. I ended up using a line from the home's auxiliary generator, which provided a steady 242 volts on grounded UK receptacles. I assigned monitor cabinets to our vocalists only. The bands cooperated by playing with restraint.

The Sun Rhythm had a special handicap: our equipment was held up in Cairo, so we had no gear for the concert. The night before our Ambassador's concert, we'd been the guests of USIS at a "Welcome to Sudan" party featuring Lomorika, a band from southern Sudan. I knew our gear was late, so I made arrangements to borrow what gear they had—just in case. This gear was delivered to the Ambassador's Residence at 5:30 p.m., with guests arriving at 6:00 p.m. My PA was a pair of Saund cabinets, each containing two 12-inch woofers. The mixer/amp was a 5-channel PA head with hi-Z inputs. I carried mics and DIs with me, and by combining both with some XLR turnarounds, I was able to come up with two vocal mics. I used the best of Lomorika's high-Z mics for my third vocal. All instruments and drums were on their own, except for Smoochy's borrowed electronic piano, which I ran directly into the mixer. There was only one monitor, an LEM with a small 10-inch woofer and built-in amplifier. I put it up on a table and used it as a side fill. There was no snake, and the cords weren't

long, so I had to mix the show on stage with the group—literally sitting at the feet of guitarists Sonny Burgess and Paul Burlison (*see Figure 8*). Definitely not state-of-the-art, but the bottom line was that we made it work.

Sun Rhythm also played a concert in the ballroom of the Hilton International Khartoum, the hotel we stayed in on every tour. The ballroom was a rectangular-shaped room with a low ceiling; the stage was placed so that we'd play to the short dimension. Capacity for this show was about three hundred, seated around long tables. The small room size and thick carpeting kept the room reverb to a very manageable 1 second. Power was available from the rear stage wall, on UK-type receptacles with functional grounds. Voltage was 240, and very stable for Khartoum. The hotel even had its own generator, a necessity as power outages were a fact of life here.

Terrance Simien's third Khartoum concert was at the Khartoum fairgrounds, where an international trade fair was ongoing. The group was to play a special concert at a site near the exhibition hall housing the U.S. entry. A 16 X 16 stage was set to "play" across the fair's main entrance, adjacent to the fair's main gate in an area framed by trees (rear of stage) and flagpoles. The rear 16 X 8 of this stage was six inches taller than the front 16 X 8—creating a riser effect. Power was run from the U.S. exhibition: it was a 30-amp, 240-volt service terminating in a UK-type grounded receptacle. I had to do a field repair on both our guitar amps here, as neither worked right. One had the power transformer break loose from the chassis, pulling its wires off the amp's PCB; the other had several cracked solder joints in the AC section. This latter problem caused a capacitor to come loose, so the amp had a huge buzz with no input. Once we'd solved our guitar amp problem, the rest of the day was easy. We didn't soundcheck, as our allotted time had been used in amp repair, so I did the show cold. It didn't take me long to get things in the pocket, as I enjoy mixing outdoors. A crowd of around two thousand warmed slowly to the band, but were dancing up a storm by the end of the 75-minute concert.

Omdurman

Located across the river from Khartoum, Jay Hoggard performed a special concert at the TV Omdurman studios. The audio facilities were not extensive; only seven channels were available, so we agreed that I would set up our system, minus the house PA, and mix audio for the TV taping.

The studio itself was quite large, with fairly good acoustics. I arranged to set up my console and rack in a hallway outside the studio, mixing on headphones (*see Figure 9*). Everyone helped set up, but our major

problem proved to be power. There were plenty of UK receptacles around, but none of them had grounds. I improvised one off a water pipe, then discovered that the hot line I was supposed to use was really only 180 volts, with a whopping 40 volts on the neutral.

The TV engineers said they used it all the time, but there was no way that I was going to, as something was very wrong with the neutral. The studio electrician was finally summoned; we searched around until we found a real 225 volt line. All neutrals had some kind of voltage on them, ranging from 3 to 40 volts, so I ended up running my own neutral to my grounded water pipe. This gave me clean power, and allowed me to give a clean signal to the TV people. Once we cleaned our power up, the rest fell rapidly into place. I added some extra mics for the recording, including an ambience mic which was taped to a "tree" flat which was part of the set placed behind the group as a backdrop. We did a quick four-minute soundcheck, so I could set levels, then did a practice tune. Jay and I listened back to this and, after making a few minor adjustments in tone, proceeded to cut six songs (*see Figure 10*).

After the music was done, we cut a short interview with Jay. I mic'ed the interview with my PZM, placed on the floor in front of the seated subjects. It worked fine, with the added advantage of being invisible to the camera. PZM's were new to Sudan; the engineers were quite amazed by their quality and versatility.

Was Medani

Travelling from Khartoum by van and truck took about three-and-a-half hours. Sun Rhythm was the only band to come here; we did two performances, one at Gezira TV, the other at the Gezira Theater. The Gezira TV studios were much more modern than the studio in Omdurman. Again, the TV engineers preferred that I mix audio on my equipment, kicking my output to TV instead of a PA. Power came from outlets located behind the set; I had no problems, as the UK receptacles were all grounded and supplied 240 volts with clean neutrals. We cut the interview with group spokesman Stan Kessler first. The young lady doing the interviewing wore a lavalier mic tied directly into TV audio; I took care of Stan through his vocal mic. The band cut four songs before we adjourned for lunch. While the guys rested, escort Toney Seabolt and I went over to the Gezira Theater to set up for our evening's concert. This theater was open air, with a stage arrangement similar to the officer's club, except that the capacity was only twenty-five hundred in a long, very gradually-raked seating area. Power was available on either side of the stage: grounded UK-type outlets provided 220 volt AC. There were no side walls or high audience area to contain the sound here, so I didn't have to worry about reverb or reflections.

We had only about a thousand people, but as all the Sudanese concerts went—it soon became a happy, dancing happening. db

PAUL HUGO

Speaker Repairs in the Field

The term re-cone is to some extent a misnomer as it is applied to the speaker repair or rebuild process used today. In most cases, a "re-cone job" involves replacing all of the moving parts and gaskets. Relatively speaking, rarely is the cone the only part needed to be replaced.

While failures in the frame and magnet structure of a speaker can be the source of problems, these are most rare and not field repairable. In this article, we will focus primarily on field repairable items. Speaker failure can be divided into two categories: thermal and mechanical. While thermal failure relates exclusively to the voice coil; mechanical failures can occur in any part of the speaker.

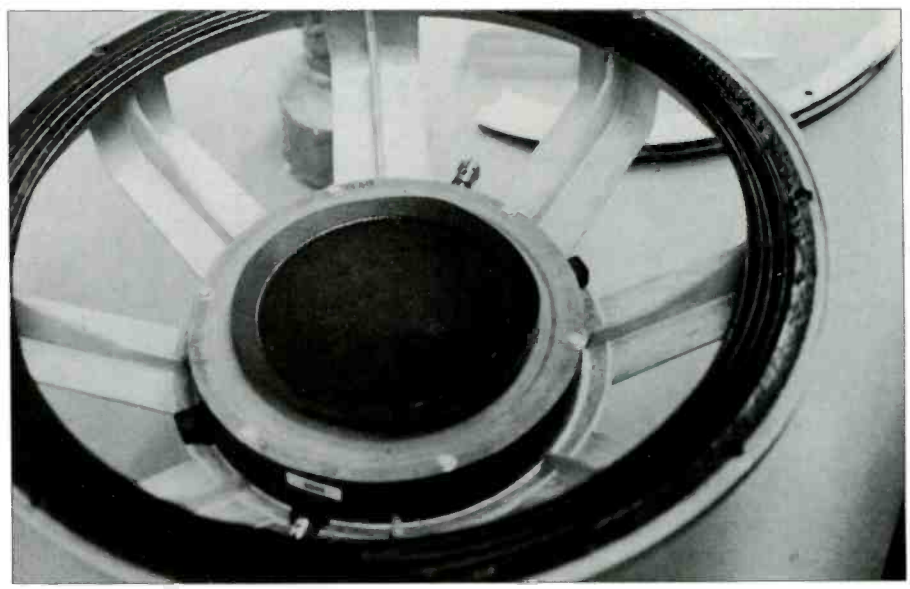
Thermal failures are generally due to misuse either in the form of a distorted signal being applied to the loudspeaker or the misapplication of the loudspeaker. An example of misapplication causing thermal failure would be the use of an extended bass speaker for a mid-bass (50 Hz-250 Hz) application. In the case of a long coil speaker used in a limited range, high-power application, a restricted amount of the coil is used. Since coil cooling is somewhat dependent on movement within the gap, the top and bottom of the long coil are not being cooled. The result is often thermal failure. A shorter coil speaker would have performed better in this application.

MECHANICAL PROBLEMS

Mechanical failures can be either misuse or factory defect. Misuse usually shows up in the form of ripped cones, domes, spiders or surround. Whether from a beer bottle or a foot, rarely is this type of failure covered by warranty. When the bottom of the coil/former is deformed or crushed, it is from overpowering or a large DC component in the signal being fed to the speaker—neither of which is a manufacturing problem. Separations at any joint, frame/surround, surround/cone, cone/former, former/spider, spider/frame, or coil/former are suspect. These are most likely manufacturing defects, but the specific circumstances of the speaker's use should also be considered. For instance, a short coil speaker, intended for use in a horn-loaded enclosure, would be "unloaded" in some ported enclosures and with sufficient power at low frequencies, would take the suspension elements to an extreme without bottoming out the coil. In this case, a spider/former joint failure might not be covered under warranty. On the other hand, all manufacturers of loudspeakers are at the mercy of outside vendors for adhesives and when variations occur, joint failures follow and are usually covered by most warranties.

The re-cone process can be divided into two stages: cleaning and installation. First the old, damaged parts must be removed. The cone gets cut at the surround and down near the

Figure 1. The cone is now removed.



Paul Hugo is the National Sales Manager for Gauss in Sun Valley, California.

dome. Removing this section allows access to the spider (*Figure 1*).

In common speaker designs*, cut the spider all the way around the former, then remove the former/coil assembly and cover the exposed gap with masking tape to prevent foreign material from entering the gap. Remove the gasket material, remaining surround and spider. Using a razor knife, scrape off the material adhered to the frame. A solvent, such as M.E.K. or isopropyl alcohol will remove any remaining surround, spider or adhesive from the frame.

Next, uncover the gap and clean with masking tape. A piece of tape roughly 2-inch by 2-inch attached to a paper or plastic shim should be inserted in the gap and rotated around the pole piece (*Figure 2*). This process should be repeated for inner and outer surfaces of the gap until the masking tape comes up clean. Black material found in the gap is the burnt remains of voice coil insulation. If the coil has separated from the former, special attention must be

*Gauss Loudspeakers are designed around a double spider suspension. The two spiders are affixed to a fiber-glass/plastic mounting ring at the outer edge and to the former at the inner diameter. This design allows for "active centering." The voice coil/spider assembly is not glued down, it is held in place by six screws (*Figure 6*). There is no guess work in coil alignment and no waiting for glue to dry between the spider and the frame. The coil/spider assembly drops into place in only one way. The terminations are an integral part of the mounting ring and come complete from the factory. After the screws are in place and snugged down, some play remains to allow for visual alignment. A low frequency signal (20 Hz) at a low level (7 to 8 volts) is applied to check for rubbing or scraping. After alignment is correct, tighten all six screws in a "star pattern" (like changing a tire on a car). Then, recheck with signal before proceeding as normal. Some people prefer to use shims over visual alignment—this is okay. You can still take advantage of the design by removing the shims after tightening the screws and running a check on coil alignment before going on.

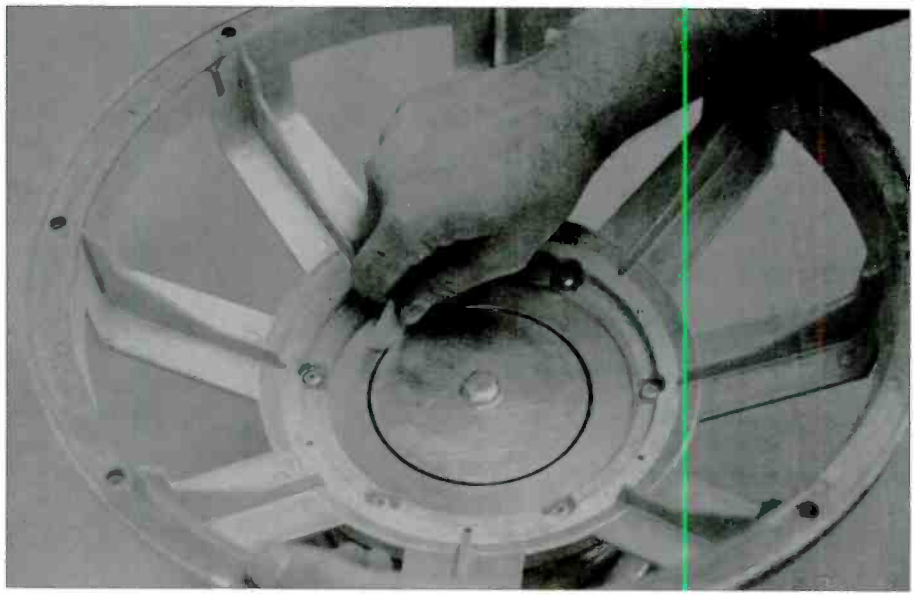


Figure 2. Clean the gap with masking tape.

Figure 3. Apply adhesive carefully and in a bead.

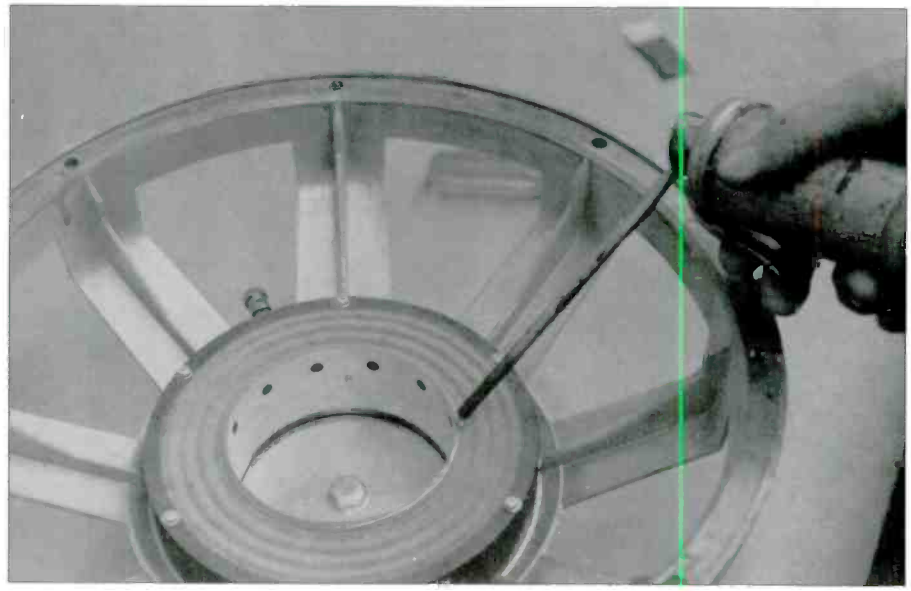
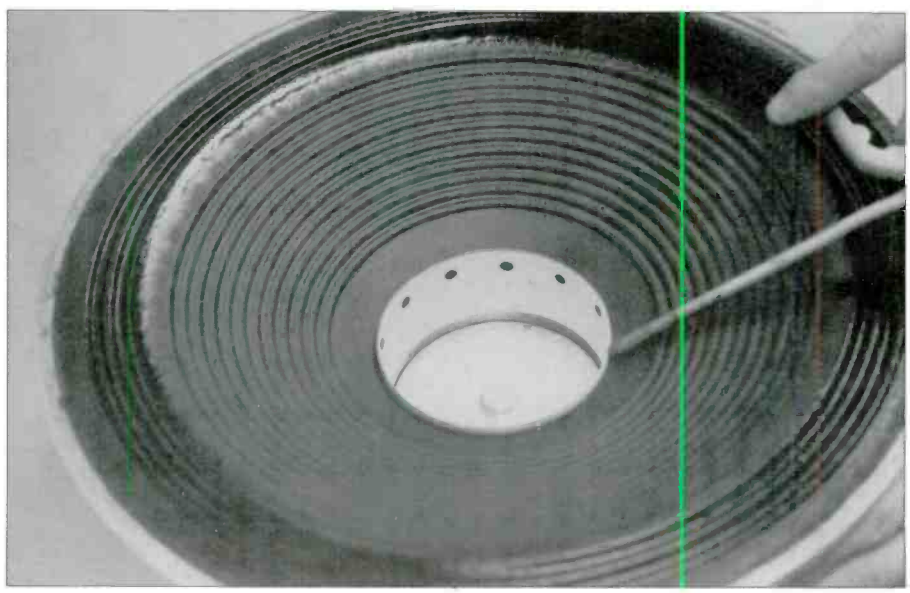


Figure 4. Adhesive now goes on the top of the cone/former joint.



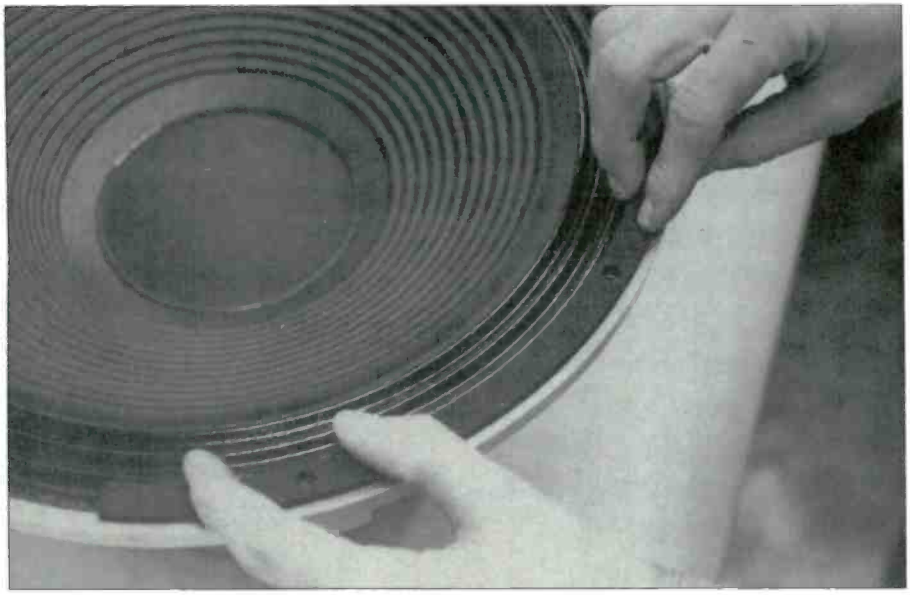


Figure 5. Applying the gasket. Be sure to line up the holes.

Figure 6. At Gauss, the voice-coil spider assembly is not glued down.

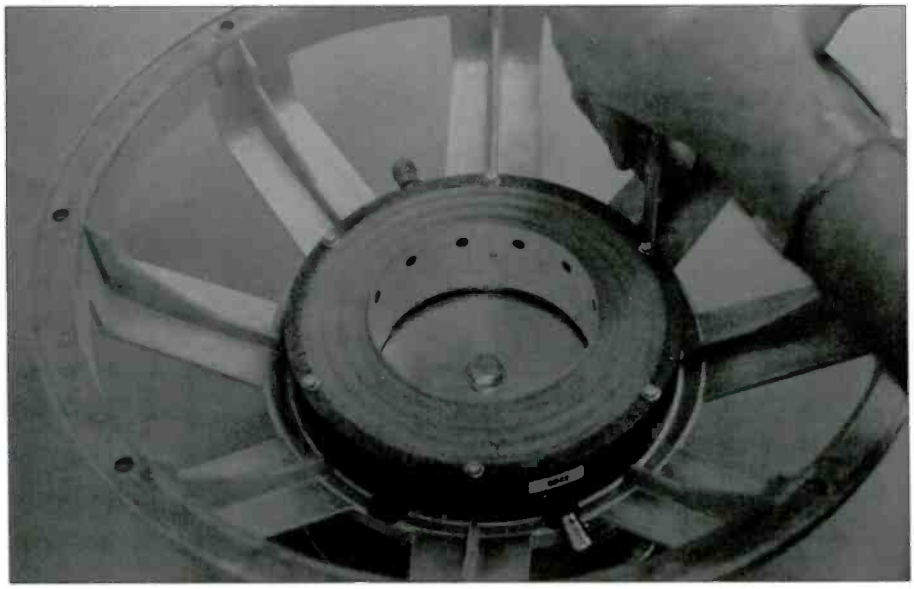
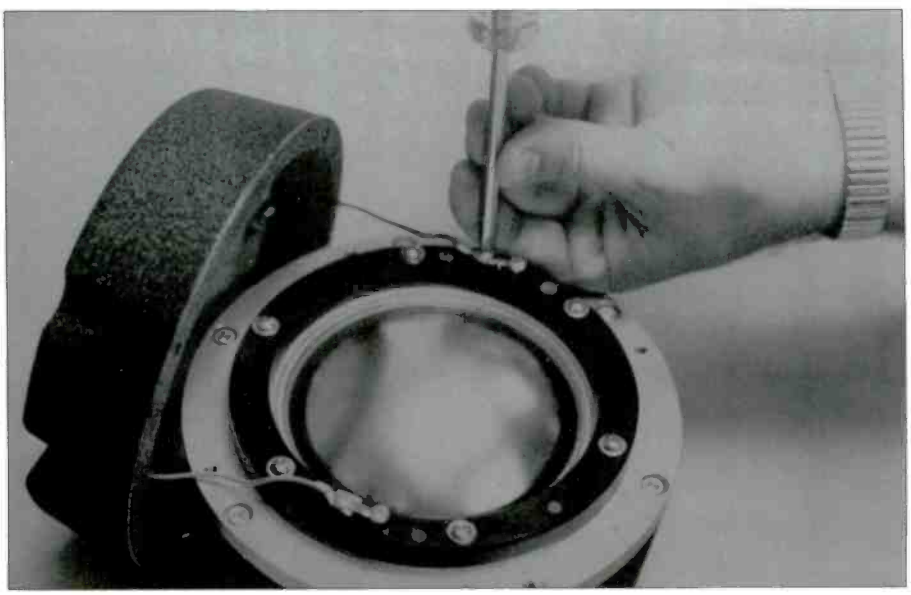


Figure 7. Removing the lead wires from a compression driver.



paid to assure none of these parts are left in the gap. Once all debris has been removed and the gap has been inspected with a high intensity light and magnifying glass, the gap should be checked for alignment. A gap gage inserted into the gap and rotated around the pole piece will point out any centering problem. Proper gap gauges should be available from the manufacturer. Magnet or pole piece shift cannot be field repaired. The magnet assembly must be degaussed, centered and re-magnetized.

INSTALLING THE RE-CONE KIT

Assuming the gap check went okay, the next step is the installation of the re-cone kit. All major manufacturers of raw frame speakers offer a three or four-piece re-cone kit. The voice coil/former and spider are shipped as one part.

The cone/surround, dome and gaskets make up the balance of the kit. In some cases, the cone is already attached to the former.

The coil/spider assembly is installed first. The coil is first inserted into the gap. In common designs, the voice coil leads should be oriented toward the appropriate terminals. Centering is accomplished by placing paper or plastic shims between the coil and pole piece evenly around the pole piece. For most common speaker designs, the spider is now glued to the frame[1]. Making certain that the spider is level and not biased toward or away from the magnet structure, apply a bead of adhesive to the raised portion of the speaker

[1]Adhesives vary from manufacturer to manufacturer. I recommend that you use those specific adhesives called for by the manufacturer of the speaker that you are working on. A particular type of epoxy, for example, may be called for, due to not only the types of materials being joined, but also for thermal, flexure and application properties. In general, an epoxy is used at the juncture of the former, cone and dome. The spider to frame junction (conventional designs) is sometimes made with epoxy and sometimes with a rubber-based adhesive like that of the surround/frame junction.

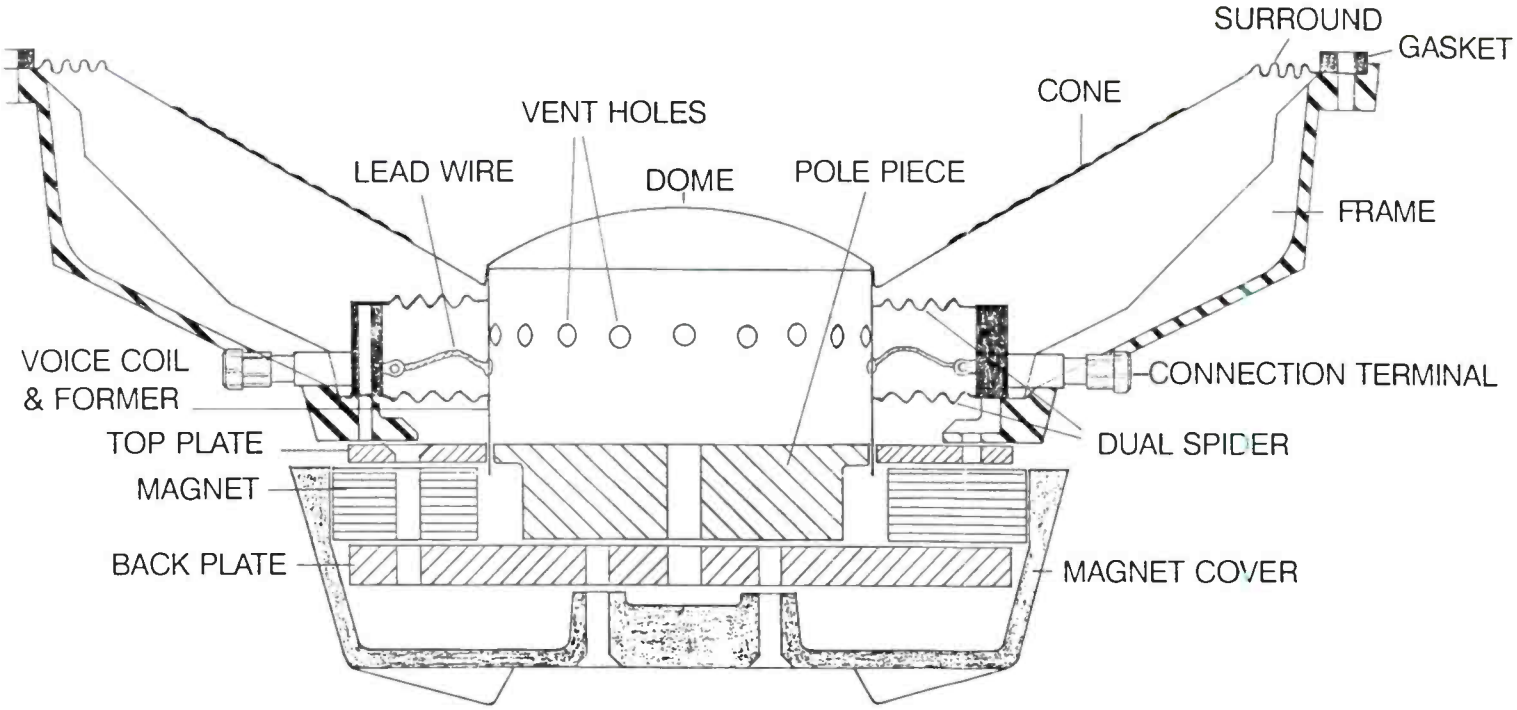


Figure 7(A). A cross-section diagrahm of a Gauss cone speaker.

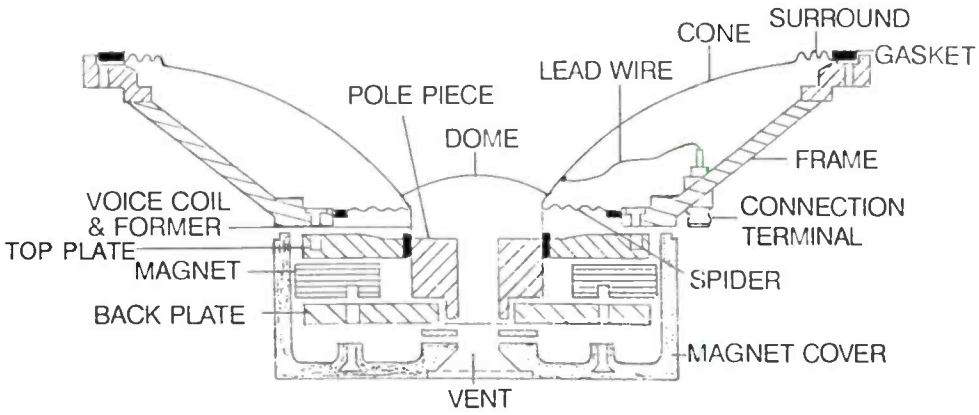


Figure 7 (B). A "typical" cone driver.

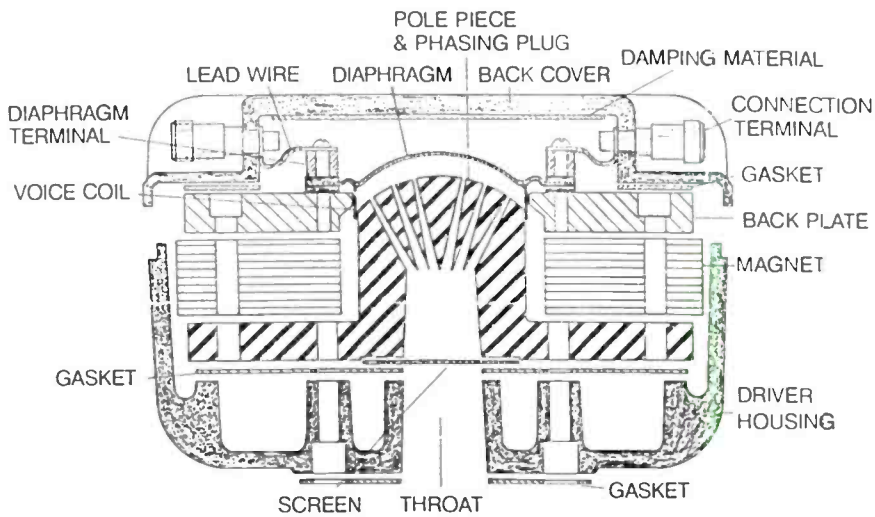
frame. After pressing the spider down to the frame, the speaker should be left to allow the adhesive to dry—usually 24 to 36 hours. It's not a bad idea to place the dome over the former at this point to prevent debris from entering the gap. Do not glue the dome down; I just let it rest over the former.

After the adhesive has dried, remove the shims and apply a low-level signal (see manufacturer's instructions) to the coil. A low-frequency signal is helpful (20 Hz) as this allows detection of rubs or scrapes most easily.

These can occur from a coil/former being "out of round" or undetected debris in the gap. Unfortunately, it's difficult to know this until it's too late. If you have a problem at this stage in conventional designs; you have little choice but to cut the coil loose and try again.

This approach becomes expensive and underscores the importance of cleaning and inspecting the gap. Of course, if the coil is "out of round," it should be returned to the manufacturer for credit.

Figure 7 (C). A two-inch compression driver.



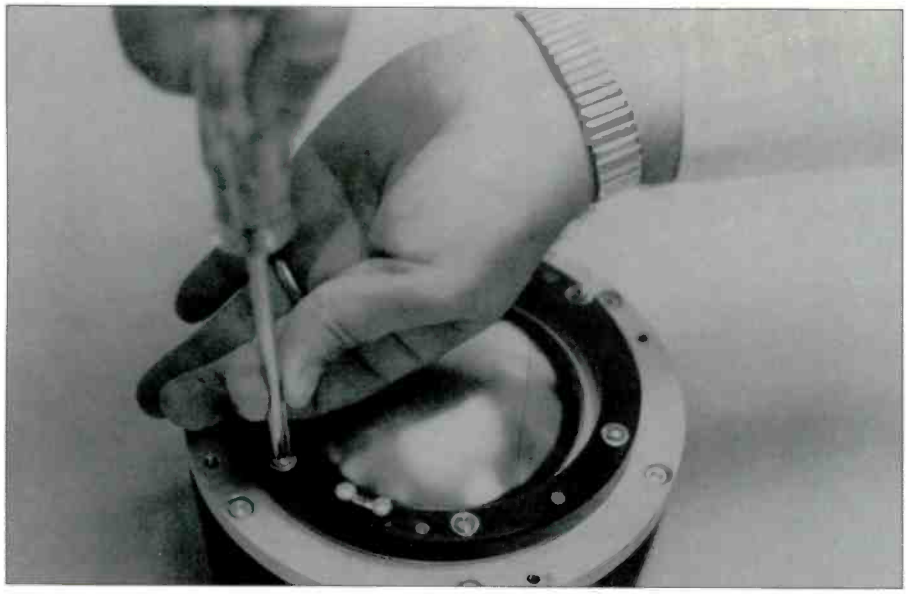


Figure 8. Separating the diaphragm assembly from the top plate.

Figure 9. Now lift the diaphragm assemby by the outside ring.

Figure 10. Clean the inner and outer surfaces of the gap.

THE CONE IS NOW INSTALLED

Now that the coil and spider assembly are in place, the cone gets installed on the former. Place a bead of adhesive around the outside of the former near the top and another where the spider and former meet (*Figure 3*). Do not allow the adhesive to run inside of the former as it might get into the gap. Spread an even layer of adhesive on the rim of the frame. It is important that the adhesive joint between the surround and the frame be complete not only all the way around, but to the inner most edge where surround and frame meet. If the adhesive is not complete to the inner edge, a "ticking" will be audible when the adhesives have dried.

While it is important that the adhesive joint around the dome be complete to assure no air leaks, buzzes or rattles, exact centering is important for cosmetic, not functional reasons.

Place the center hole of the cone down over the former, rotating it slightly back and forth as the cone is pushed down to meet the spider. Press the surround down against the rim. A bead of adhesive should now be applied on the top of the cone/former joint (*Figure 4*). This not only adds to the strength of the cone joint, but in most cases, secures the dome as well. The dome is somewhat fragile, especially aluminum domes, and should be handled carefully while being put in place. In cases where the dome does not seat at the cone/former junction, a small loop of masking tape can be used as a handle allowing centering of the dome on the cone. While it is important that the adhesive joint around the dome be complete to assure no air leaks, buzzes or rattles, exact centering is important for cosmetic, not functional reasons. In most cases, a bead of black adhesive is applied on top of the dome/cone joint. Again, this is primarily cosmetic. To apply the gasket, pull the backing tape off the gasket material, line up the holes with those in the frame, and press into place over the surround/frame joint (*Figure 5*). Lastly,

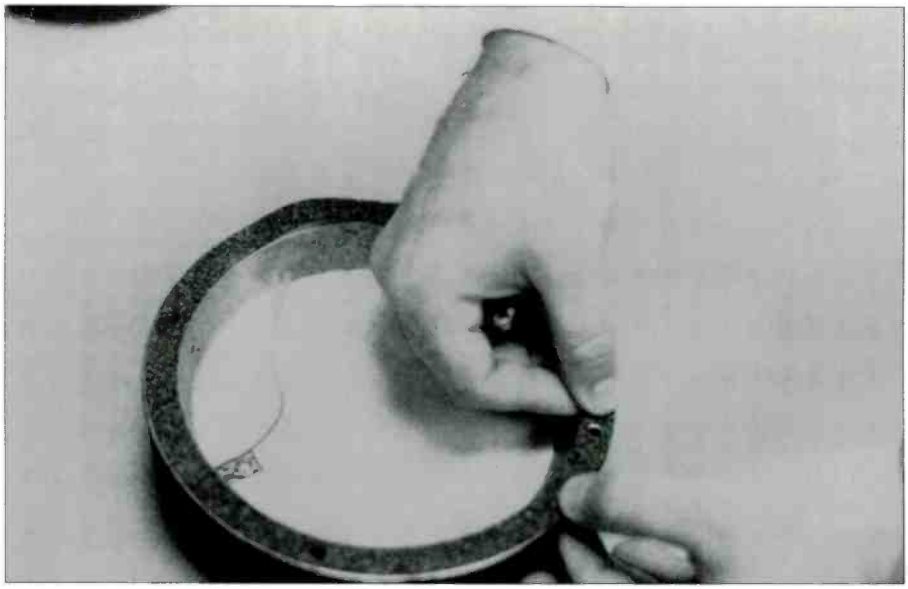


Figure 11. Install a new gasket, again aligning it with the holes.

solder the voice coil leads to the terminals. Leave enough slack to allow speaker movement but not enough to allow leads to short out.

It is common practice to supply light pressure to the cone and a bit more pressure to the gasket while the adhesives set. A roll of masking tape set on top of the cone works nicely to assure that the cone stays in place. Usually, another speaker of the same size is turned upside down and set on top of the speaker being repaired, gasket to gasket. This process provides enough pressure for proper gasket and surround seating. With or without weights, the speaker should be allowed to dry and set up for 72 hours. For a final check, a low frequency (20 Hz) signal applied at a low level (7 to 8 volts) will again show up the most likely problems.

CONSIDERATIONS OF HIGH-FREQUENCY DRIVERS

High-frequency compression drivers suffer more from mechanical than thermal stress. Because of the application they are designed for, lightweight materials must be used. Whether phenolic, aluminum or titanium, the first failure mode is generally mechanical. Knowing how these devices operate should help you understand how they fail. The diaphragm operates in a piston mode at the lower end of its range. In the middle of its range, the diaphragm is in a "break-up" mode. The surface divides into small regions which operate together. At the top of its range, the surround or suspension is the active portion of the device.

The most dramatic mechanical failure in high frequency drivers is a shattered diaphragm. This is generally caused by a stress or crease in the dome. The diaphragm may operate normally at first, but after some use will shatter into small fragments. This stressed condition may be the result of mishandling during installation of the diaphragm or a strong impulse applied to the driver which drives the diaphragm into contact with the phasing plug. An impulse, such as a turn on transient, can also pull the coil away from the diaphragm. One has to remember that the spacing between the diaphragm and pole piece is only about 14/1000 of an inch. It does not take much to cause damage to a device with such small tolerances.

If rubbing occurs, use a flat end punch and small mallet to *gently tap* the diaphragm assembly to correct positioning and stop the rubbing.

Surround failure can take the form of a split in the surround or "oil canned" surround. A split or tear in the surround is fairly self-explanatory.

Oil canning is a permanent deformation of a flexible part. After oil canning has occurred in a suspension element; the diaphragm is positioned further away from the phasing plug than normal. This will result in reduced high-frequency response. Both types of surround failures are caused by excessive movement—usually too much power at too low a frequency.

COMPRESSION DRIVER REPAIRS

To repair a compression driver, the failed diaphragm must be replaced. Remove the screws from the back cover of the driver and lift the cover from the unit. If the cover does not lift easily, use a rawhide or rubber mallet to tap the side of the cover, loosening it for removal. Use caution when usilng a mallet. An improperly-placed blow or too powerful a blow could damage internal structures.

Using a non-magnetic screwdriver, loosen the two screws holding the lead wires to the diaphragm assembly and remove the lead wires (*Figure 7*). Again using a non-magnetic screwdriver, remove the screws and washers holding the diaphragm assembly to the top plate (*Figure 8*). Do not discard the screws and washers as they are necessary for re-assembly.

Now, carefully remove the diaphragm assembly by lifting it by the outside ring (*Figure 9*). Using masking tape, clean the gap thoroughly by rotating a piece of masking tape around the pole piece. Be sure to clean both inner and outer surfaces of the gap (*Figure 10*). Check the gap with a high-intensity light and a magnifying glass to insure all foreign materials have been removed. If the diaphragm has been shattered, extra care must be taken to ensure that all the pieces are removed from the phasing plug assembly. Measure the gap using a pin-type gauge. Proper gauges should be available from the manufacturer. If a gap shift has occurred, the unit must be returned to the factory for proper alignment. Do not attempt to install a new diaphragm.

At this point, if the replacement diaphragm is not to be installed immediately, seal gap and slots in the phasing plug with masking tape to avoid contamination.

DIAPHRAGM INSTALLATION

To install a new diaphragm assembly, orient the two terminal groups on the diaphragm assembly with the small notches on the inside of the top plate. Holding assembly by outside ring, carefully insert the diaphragm assembly into the plate, seating the coil into the gap and maintaining the orientation already described above.

Insert the screws and washers and snug them down using a "star" pattern. Apply approximately 5 volts at 500 Hz to 800 Hz and listen for any rubbing. If rubbing occurs, use a flat end punch and small mallet to *gently tap* the diaphragm assembly to correct positioning and stop the rubbing. If rubbing persists, the unit must be returned to the factory for proper alignment.

Remove old gasket around the throat of the driver and replace, making sure all horn flange mounting holes align with holes in the gasket.

Tighten down the screws, again using a "star" pattern. Be careful not to exceed the force suggested by the manufacturer. Repeat the tone generator test to insure rubbing does not take place.

Remove the old gasket and install a new one around the edge of the back cover, making sure that all screw holes align with the gasket holes (*Figure 11*).

Connect lead wires to the terminals on the diaphragm assembly. Follow phase convention and markings such as the red wire to the terminal with red dot. Be sure lead wires are properly dressed so that they do not touch the diaphragm itself.

Replace cover, insert the screws and tighten down using a "star" pattern. Remove old gasket around the throat of the driver and replace, making sure all horn flange mounting holes align with holes in the gasket.

This article has covered the various failures that can occur in loudspeakers, whether thermal or mechanical, and through detailed instructions, explained the many techniques of repair used to correct these problems, thereby bringing your speakers back up to peak performance. db

JAMES BECHER

Music from an Electronic Cottage

This is what I do. I make music. Sounds simple, right? Maybe. Technology has created the need (and the ability) for incredible precision and slickness in today's pop music, and making music, using MIDI for all aspects of the production and recording process, seems to be the key.

In the 1960's, recording instruments that were out-of-tune, especially guitars and flutes (Can you name those tunes?) were perfectly acceptable. Even "loose" rhythm tracks were often allowed to make the cut. This was due, I suppose, to the organic nature of the bands and their music. There was a certain edge and natural "feel" created by the rawness of those elements. In today's music however, we tend to measure the quality of musicianship by the precision of the instruments.

The level of quality expected for recording (even demos) must be top-notch using state-of-the-art (sorry about the use of an already played-out phrase) equipment.

A NEW BREED

Fortunately, there is now a new breed of musician/producer-arranger/engineer who possess the ability to turn out great music from the home studio, ergo the *Electronic Cottage*. With the use of MIDI keyboards, drum machines, and sequencers, any sound texture can be realized, and complete orchestration is possible—the performance information of which can be digitally recorded and saved on disk to be recalled at any time. By having an extensive MIDI set-up, which virtually eliminates the need of having to record any of the instrument sounds onto analog tape, the only element necessary to record onto the tape is the vocals (except for any live instruments not used in the MIDI configuration).

James Becher operates his Electronic Cottage in Northport, New York.

Therefore, it becomes unnecessary to use large format tape machines (eight tracks are usually just fine). If noise reduction is used on the analog tape machine, and the tracks (both tape and MIDI) are mixed onto a DAT machine, CD quality can be achieved. I say this because there is no signal degradation of the instruments because they are essentially being mixed live direct-to-digital. Essentially the only noise occurring would be the inherent noise of the MIDI instruments themselves, the noise of the mixer and whatever outboard gear is used, and the residual noise (after noise reduction) of the tape.

That's why utilizing eight tape tracks, let's say, instead of twenty-four or even forty-eight becomes a blessing (in terms of keeping the noise floor low), rather than a shortcoming. This is essentially the approach I use for pop music production. It also seems to work for just about any other type of music production, including soundtrack music production.

SONS OF THUNDER

My production company is called *Sons of Thunder* and we have two studios located on beautiful Long Island, New York. Our specialty (of the day) is putting together high-quality songwriter's demos, composing soundtrack music, and designing promotional packages. We recently had a pop song which charted in the top 20.

In addition, we provided the music for the PBS television series entitled *South Africa Now*, a series dedicated to the subject of apartheid in South Africa. I, myself, have had the great pleasure of working (or playing, really) with such artists as Jon Anderson (ex-Yes), Verdine White (Earth, Wind, and Fire), Ed Gagliardi (Foreigner), Bobby Rondinelli (Rainbow), Kenny Aronson and Thommy Price (Billy Idol), and Meatloaf. I served as product specialist and keyboard clinician with M.T.I. (Music Technology, Inc.) working on the original Synergy digital keyboard project with Wendy Carlos, Tom Piggott, and "Stoney" Stockell. I have also been a consultant to European keyboard manufacturers Crumar and Siel, and worked in Japan with the good people of Korg voicing what is currently America's hottest selling keyboard, the M1 MIDI workstation. So you can see not only do I enjoy working with a variety of musicians from all walks, but I thoroughly enjoy the challenge of designing the actual synthesized sounds themselves.

As I have mentioned, Sons of Thunder has two studios, and I own the studio located in East Northport, New York, and it is this studio of which I would like to give you a "Cook's" tour.

My MIDI studio is the quintessential *Electronic Cottage*, and not just a general-purpose recording studio. I make this distinction because I prefer working on a more personal level with the songwriter(s), and because I provide all (or at least, most) of the music.

Many times the songwriter(s) will have written lyrics only, so in that situation I'll compose the music, making sure that the music reflects the images suggested by the words. If a band or large group is interested in doing some recording, I will usu-

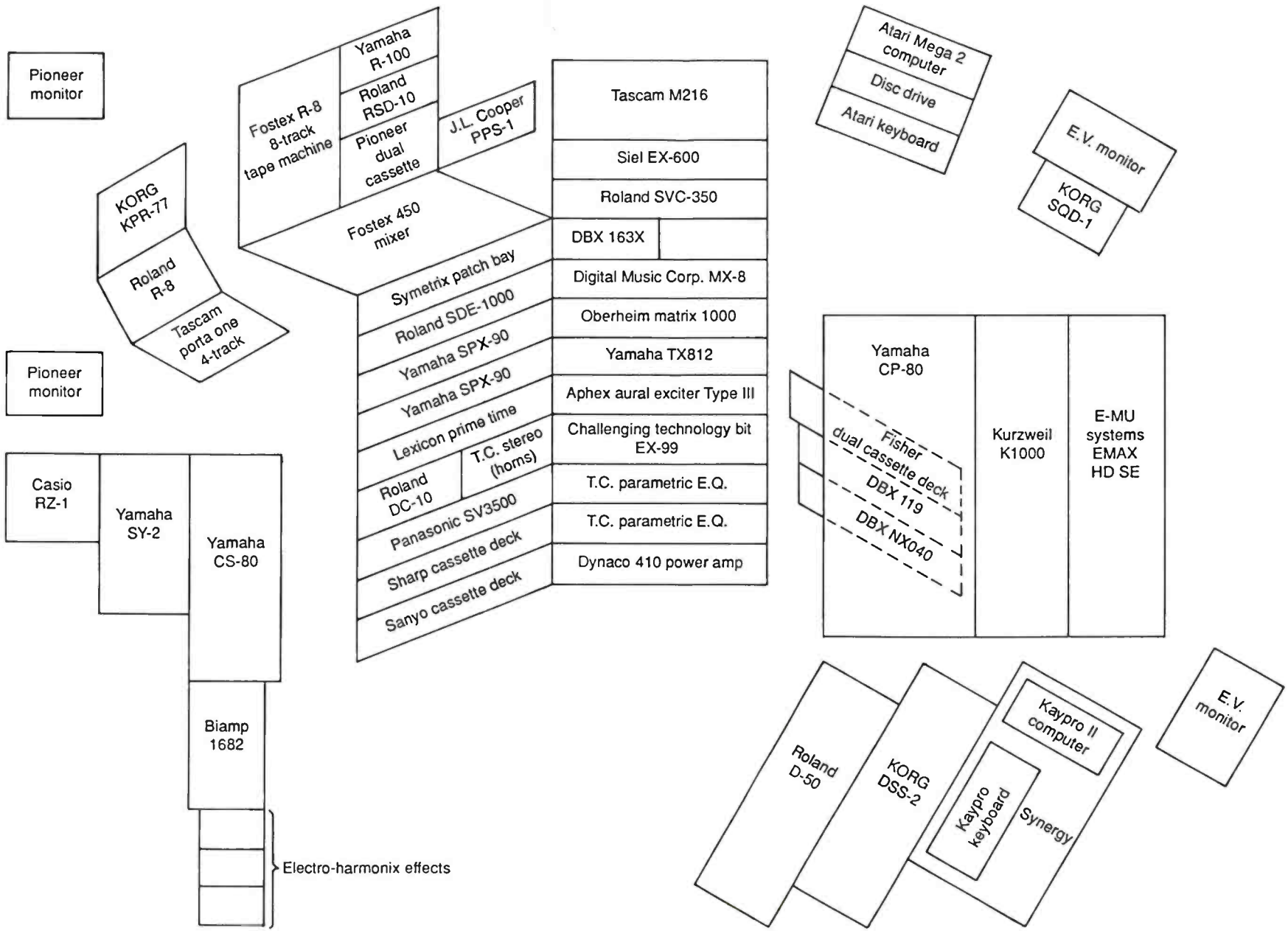


Figure 1. The equipment setup Becher uses.

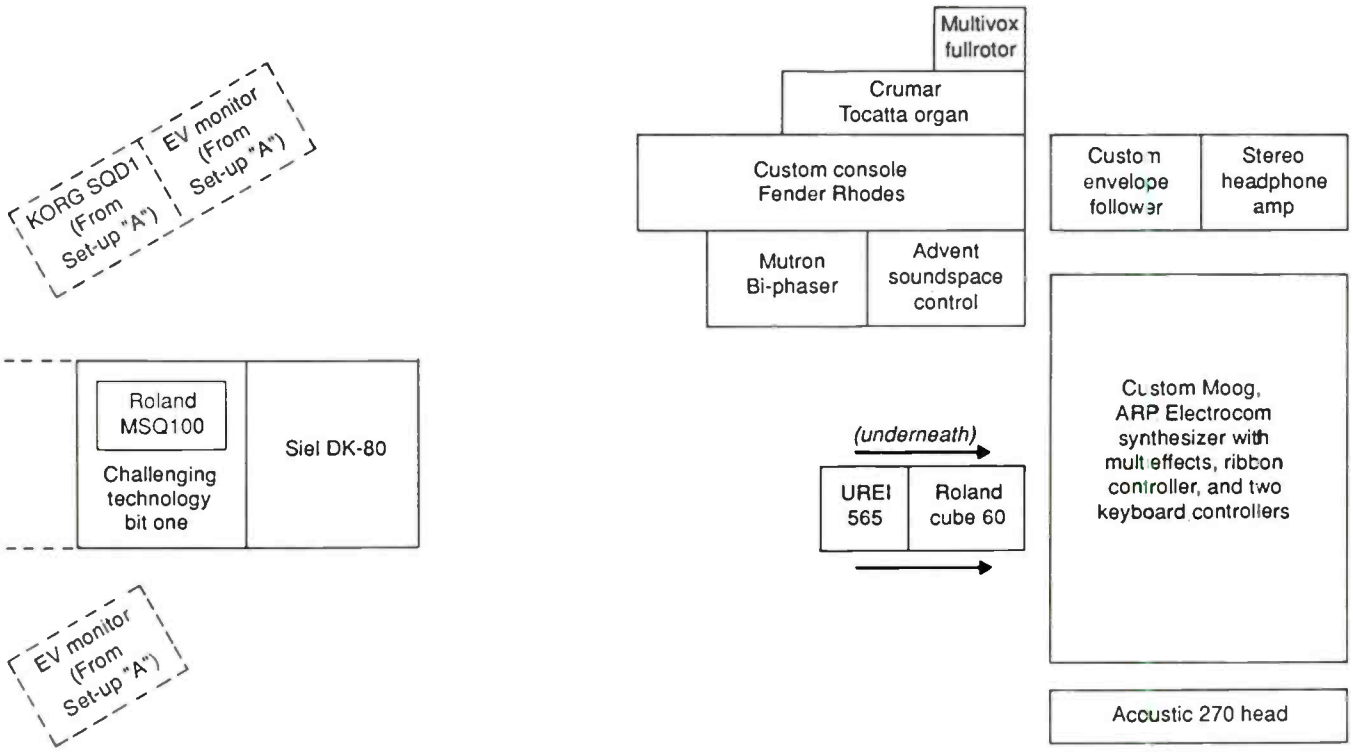


Figure 2. More detailing of the setup used.

ally refer them to another studio that does general-purpose multi-track recording. I try to create a friendly, hassle-free working environment, and I find that it's generally easier to do that with fewer people anyway.

When I am working on soundtrack material, I will usually do that solo. I designed my studio with ergonomics in mind. My keyboard controllers, main computer, multi-track tape machines, drum machines, primary mixers, signal processors, and patch bays (both MIDI and audio), are all within arms reach because of the "U"-shaped layout of the gear. This makes my life easier because not only does this configuration reduce stress when I'm involved in a lengthy production or programming session, but it's just simply comfortable.

POP MUSIC METHODOLOGY

My method for pop music production is similar to soundtrack production, though my approach to soundtrack music will often vary, depending on the nature of the piece. In general though, the first step for putting together a pop song involves mapping out the song(s) with the songwriter(s). We will discuss everything from arrangement, to vocal phrasing, to the types of sound textures that will perhaps be used. Then I do the programming.

I do all of my MIDI programming using the Atari Mega-2 computer with the incredible Notator sequencing program by C-Lab (distributed in the U.S. by Digidesign). I absolutely love that program because of its incredible flexibility, and because you can perform editing functions while it is running. It even prints out music notation in real time. You can't imagine how much time that saves! Notator also has full MIDI automation which makes mixing a dream. The Atari Mega-2 with Notator is literally the heart of my Electronic Cottage. Korg's SQD-1 (affectionately known as "Squid") and Roland's MSQ-100 serve as back-up hardware sequencers, when that becomes necessary. MIDI information is then sent directly to my Digital Music Corp. MX-8 MIDI patchbay/processor. All MIDI info is "looked-at," processed (merged, filtered, transposed, MIDI delayed, etc.) and then assigned to any of the eight outputs, which feed my synths, drum machines, and tape synchronizer.

AFTER THE PROGRAMMING

When my programming is finished, I'll quickly assign temporary, usable sounds to the keyboards and expanders, and then record the vocals. Vocal mikes include a pair of Audio-Technica AT-813 condensers, and a Beyer M-500 ribbon microphone. The Audio-Technica's are surprisingly smooth, and the Beyer has nice top-end. Using compression or some form of limiting on vocals is a must. I personally like the sound of compression on vocals, and the fact that compressing the vocals provides a much more uniform level when recording. This makes level placement when mixing much easier as well. The Yamaha SPX-90, dbx model 119 and 163-X Over-Easy compressors take care of all my compression and gating needs.

I use the Fostex model R-8 eight-track machine for my tape recording. It has extensive programmable memory, auto-locate, and auto-shuttle functions—I love that. It also uses standard ¼-inch Ampex 456 tape, which is cost-effective for the client, and because it does not present any channel cross-talk problems. I say this because as I mentioned earlier, in most situations, the only thing I record onto the tape is the vocals, and the biggest offender of channel cross-talk is the low-frequency energy from the bass track, the kick drum track, and any

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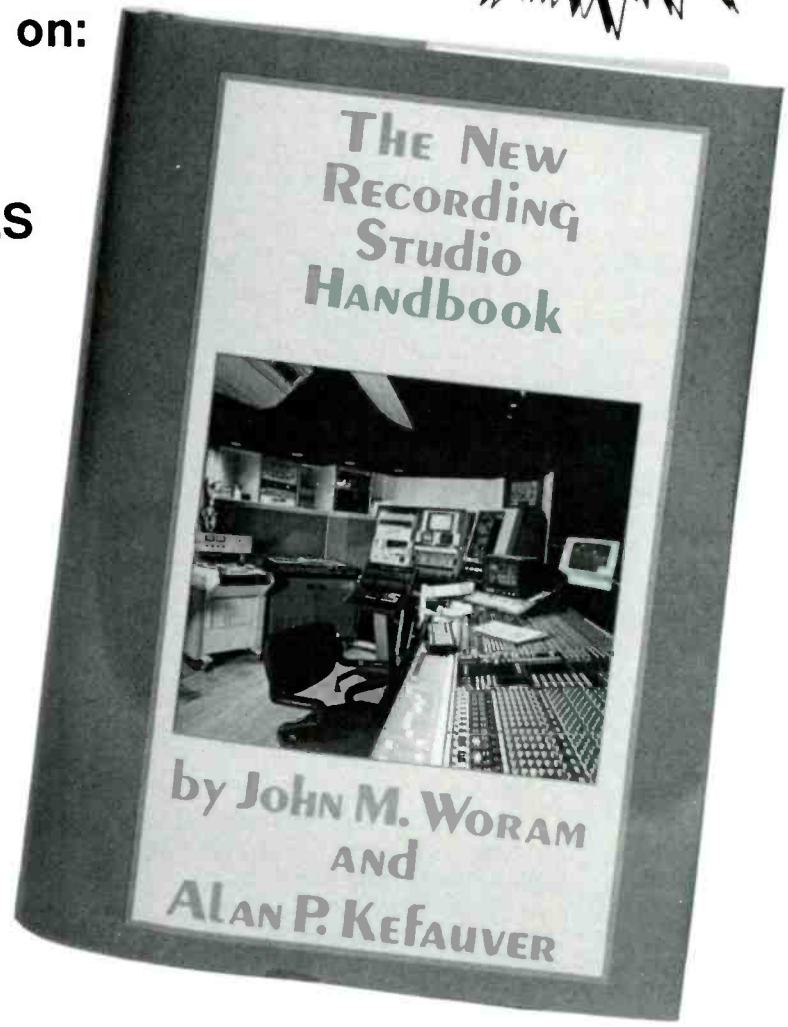

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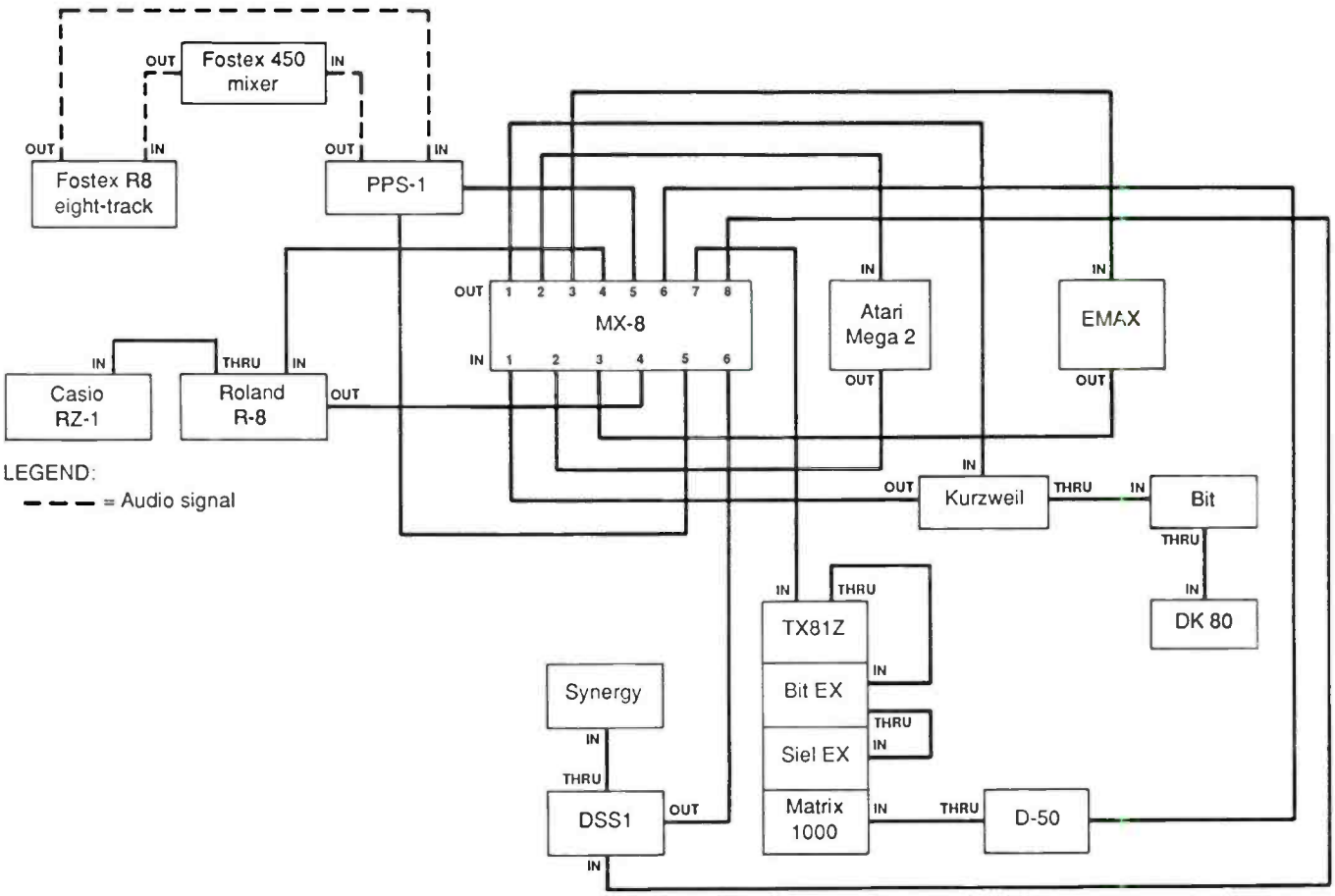


Figure 3. This is the main MIDI flow chart.

low synth stuff. The R-8 will also allow its transport functions to be controlled from the computer via MIDI, and that's pretty "hip." I also have seven tracks available for the vocals (the eighth track being used for synchronization purposes), and that's usually more than most twenty-four track studios have available after all of the instruments are recorded. Seven of the R-8 audio tape outputs are channeled into a Fostex model 450 eight-channel mixing board, while the eighth output is fed directly into a tape-to-MIDI synchronizer which runs my Atari computer. For synchronization, I use J.L. Cooper's PPS-1 (version three) tape-to-MIDI synchronizer, which supports both SMPTE and "smart" FSK sync formats. I personally prefer using "smart" FSK for pop music, because programming a tempo map isn't necessary. If a quick work tape is all that is needed, I'll use my Tascam Porta-One four-track cassette system to log the musical ideas. The Porta-One has built-in dbx type II noise reduction, which makes for very quiet recording.

Figure 4. The main keyboards, computers, mixer, and signal processors at "Sons of Thunder".

THE KEYBOARD COLLECTION

With the vocals recorded, I then begin to pick the "real" sounds to be used, and assign all of the parts to the appropriate keyboards and drum machines.

My keyboard arsenal includes an E-mu Systems Emax HD SE, Roland's D-50, Korg's DSS-1, Kurzweil's K1000, Digital Keyboard's Inc. Synergy (with the Kaypro II computer used for "voicing" and

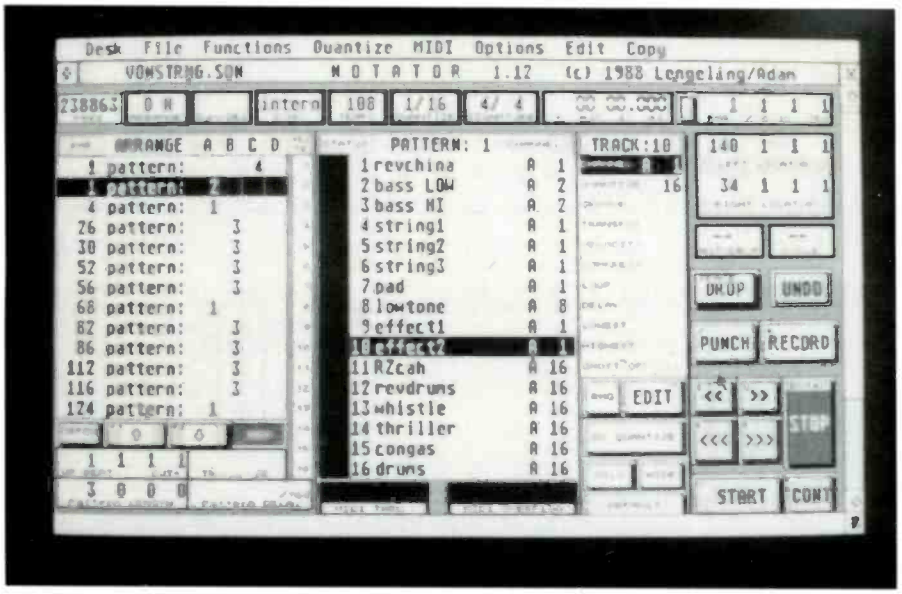


Figure 5. The Atari Mega 2 with the "Notator" sequency and music notation program on screen.

data management), Oberheim's Matrix 1000, Yamaha's TX81Z, Yamaha's CS-80, Yamaha's CP-80 electric grand piano (with custom light-touch, heavy-action), Yamaha's SY-2 lead-line synth, a custom console Fender Rhodes electric piano, a Toccata organ and a few other synths and expanders by European manufacturers Challenging Technology (formerly Crumar) and Siel. I even cascaded a bunch of Moog modules, an E.M.L.-200 Electro-Comp synth, an Arp 16 channel sequencer, a ribbon controller, (Who remembers that?

Hint: Try to get your hands on some vintage Emerson, Lake, and Palmer concert footage—watch Keith Emerson.) and a host of analog signal processors including a phase shifter, an envelope follower, a flanger, and an overdrive module, all into a single (and gigantic, I might also add) road case that takes up the space of an entire wall. All of the instrument audio outputs are then fed into a Tascam M-216 sixteen-channel mixing board, and a modified Biamp model 1682 sixteen-channel mixing board.

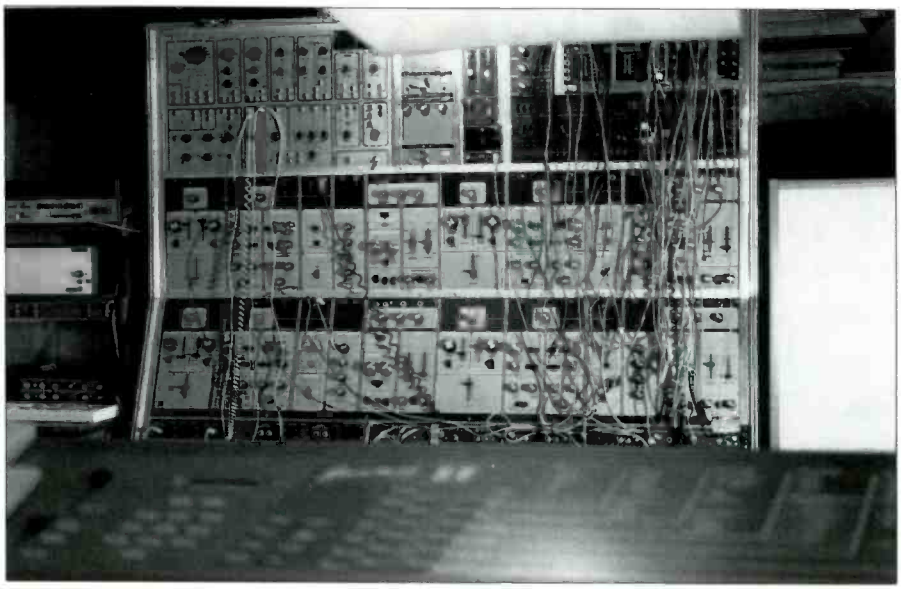
As you may have noticed, I am a firm believer in using both digital and analog synthesizers. One cannot replace the other, mainly because each synthesizer has its own sonic personality. Even the sounds of digital synthesizers vary greatly, which is why I use so many different types.

For example, I'll usually use the D-50 for rich, spectral colors, while I'll use the Matrix 1000 and the CS-80 for fat, punchier sounds. The E-max, DSS-1, and Kurzweil serve as my samplers (the Kurzweil is read-only) providing me with the more traditional and authentic sounds, though I must admit I do have some rather wild samples that are anything *but* traditional. If stereo sampling is needed, I'll use an E-mu Systems Emulator Three sampling keyboard, but I seldom have a need for stereo sampling, and my E-max (which has an internal twenty-megabyte hard drive) and DSS-1 samplers do just fine. The TX81Z gives me that clean FM sound, and makes a great MIDI-mate. The CP-80 is customized with incredible action, and sounds great as a stand-alone piano for pop and rock stuff. The Synergy is probably the most unique keyboard that I use, and its sound is hard to describe. It uses additive and phase modulation/cancellation synthesis, having an edge to its digital sound, yet somehow remaining very smooth-sounding.

Got that? I know, it sounds sought of like that perfume commercial which claimed that its product was "Provocative, yet not too far from innocence" (or something like that). Well anyway, it is difficult to translate sounds into words, though I must take my hat off to the guys and gals who spend hours and hours programming sounds for synthesizers and then tackling the chore of having to name them. I say this because more often than not, the name of the sound attempts to describe some sonic quality or character of the sound.

Sounds like "Nasalfuzz," "Crickettell," and "Stringme" are examples of what a little imagination can produce. In general, I try to use and blend all types of digital synthesis with analog, thus running the gamut of sonic palettes, which seems to keep the music sounding exciting and fresh.

Figure 6. The Moog, ARP, and Electro-Comp looking over the E-mu EMAX HD SE Sampling Keyboard.



DRUM PROGRAMMING

I do all of my drum programming in the sequencer rather than in the drum machines themselves. I do this for a couple of reasons. The first is simply because it happens to be eas-

ier for me to play the drum voices from the keyboard instead of from the drum machine buttons. By playing the drums from the keyboard, I can first record the kick, snare, and high-hat parts, and then when I am satisfied with my basic drum tracks, I'll overdub the toms, cymbals, and percussion.

In this way, I literally build the rhythm section from the ground up. I can also control the dynamic response of the drum sounds better using the keys, because they provide a fairly large target to strike, and their pivoting quality in general tends to provide a nice feel.

There is another advantage to playing the drums from the keyboard into the sequencer. Standard drum programming requires that you create a series of individual patterns (corresponding to verses, choruses, the bridge, etc.), and then to chain these patterns together to create a finished song. Makes sense, right? Well, this kind of repeated pattern playing tends to become predictable and even mechanical, and gets boring real fast.

If you desire to make a subtle (or even not so subtle) variation of a pattern, it must be programmed as a separate pattern, and then plugged into the chain. This is not only time consuming, but cannot be done in real-time, which prevents any spontaneity. The continuity of the song gets lost. Once again this is where using Notator really comes in handy. You see, Notator allows you to simultaneously perform two types of sequencing: the standard pattern style, which is like that of a drum machine, and continuous program sequencing (playing from the beginning of the song to the end of the song). This means that I could program the kick and snare drum parts using the standard pattern style programming.

Then, I could overdub the toms, cymbals, percussion, and even occasional extra snare hits, playing continuously from the beginning to the end of the song. This develops a better flow and sense of feeling for the song. There is still another advantage to recording the drum parts into the sequencer—more efficient data management. That's right. By using the MIDI sequencer to store ALL performance information, all of the information is kept in a common, central location. This eliminates the need of having to save the drum machine data to disk or cassette, in addition to saving the music sequences. Now you could save the drum machine data via a MIDI data dump (using system exclusive messages) into the sequencer and then onto the sequencer's disk, but that's not really time efficient, and why do double the work?

My main source of sound for the drums is the Roland R-8 drum machine, which I think is wonderful. It's multi-timbral, and in addition to functioning as a stand-alone drum machine, it also doubles as another sound module, having the ability to play up to four independent drum voices, all assignable, melodically and polyphonically from the keyboard (picture playing chords with the bell of a ride cymbal). I also have tons of custom drum and percussion samples ranging from African, (not to mention Japanese and Indian), to the "Twilight Zone" in my Emax and DSS-1 libraries. In addition, I keep a Casio RZ-1 sampling drum machine hooked-in to the MIDI system for that occasional, temporary, down-and-dirty sample that doesn't have to have 16 bit quality. There's a very handy little sample in the RZ-1 that I use for doubling the snare drum part on pop music tracks. It's a short sample, and by varying the mid-range center frequency, and boosting a substantial amount of the mid-range gain, different kinds of "cah" type sounds can be gotten. This adds a real "crack" to the snare drum part. I will even find an occasional use for a neat little analog drum box that Korg used to make called the KPR-77. Remember that thing? It doesn't have MIDI, but it does have DIN sync. Still, it's easier just to sample it.

Figure 7. James Becher in his "Electronic Cottage".

FINISHING TOUCHES

Now its time to make the musical monster I've created sound like something. Outboard gear plays such a crucial (and final) role in music production. It's not necessary to have five-thousand reverbs and ninety-three zillion digital delay units for the mix. It *is* necessary however, to effectively and creatively use each of the units that are being employed for the mix. Having one general purpose reverb, a few dedicated reverbs and D.D.L.'s (for both long and short reverb and delay times), and a stereo chorus or harmonizer usually suffice. Creating a three-dimensional sonic depth of field is what's really needed, and that's what I shoot for.

I use a couple of Yamaha SPX-90 digital effect processors, a Yamaha R-100 digital effects processor, a Lexicon Prime Time, a Roland SDE-1000 and Roland RSD/10 digital sampler/delays, a Roland DC-10 analog delay, a T.C. Electronics (out of Denmark) digital stereo chorus, a pair of T.C. Electronics fully parametric four-band equalizers, UREI's 565 notch/peak filter set, a Mutron Bi-Phaser, the Roland SVC-350 vocoder (which I sometimes use to double the background vocal harmonies by "playing" the harmonies

from my Yamaha CS-80 synthesizer, which creates a very ethereal texture), and an Aphex Systems Aural Exciter Type Three, model 250, to create and shape the ambient environment that the music will come to life in.

Most of the keyboards and expanders I use have built-in multi-effect processors, which is great for some of those dedicated effects I spoke of.

Lately, I've gotten more into using shorter reverb times than I used to, and as a result the music does seem to retain a certain "snap" and intelligibility (longer reverb times are still nice for slower tunes and ballads, though). I've come to realize that it's really not necessary for music to have that Phil Spector "wall of sound" in order to be powerful, and that the trick is to orchestrate and arrange the parts effectively, and to carefully choose sounds that paint the appropriate musical panorama which supports the attitude and flavor of the lyrics. A little discipline and restraint (in terms of production) go a long way.

TO THE MIX

And now I mix. Direct to DAT. The Panasonic SV-3500 is the DAT. The SV-3500 has all kinds of programming and looping functions, a digital input/output for digital transfers, and sounds great! I make safety copies of the mix on metal cassettes with dbx type II encoded, recorded as hot as possible, and they sound just fine.

Since I will have pre-programmed the majority of the mixing moves (including the changing of reverb and effect programs) using computer automation via MIDI, the only mix duties I need to perform are to push the DAT record button, engage the eight-track which is now controlling the computer through SMPTE or "smart" FSK sync, and to watch the vocal levels. I like that. There is, in today's music, a definite tendency (and trend) towards over-production or over-kill, and many times the song gets lost or buried in the very production that's designed to bring it to life. One thing is true; fancy production is no substitute for poor songwriting—it only appears to smooth-out the rough edges.

It's so important to have strong melodic "hooks," sensible arrangements, and good lyrical content as the raw elements for pop music. Then, with the proper use of "hip" sounds and tasteful production, the song will sound simply marvelous!

Sitting on the cutting-edge of technology keeps my music diverse and strongly flavored. I enjoy the challenge of pushing music beyond the ordinary into the realm of teeming rhythms and exotic soundscapes. I believe that music should jump out and confront the listener. Such qualities are better described with music than with words, so I'll just cue up my latest piece of music to show that: "This is what I do!"

Equipment

Keyboards
E-mu Systems Emax HD SE sampler
Roland D-50 synth
Korg DSS-1 sampler
Digital Keyboards Inc. Synergy synth with Kaypro II computer
Kurzweil K1000 read-only sampler/synth
Yamaha CS-80 synth
Yamaha CP-80 electric grand piano
Yamaha TX-81Z synth
Yamaha SY-2 synth
Oberheim Matrix 1000 synth
Challenging Technology Bit One synth
Challenging Technology Bit EX-99 synth
Siel DK-80 synth
Siel EX-600 synth
Custom console Fender Rhodes electric piano
Crumar Toccata organ
Custom Moog, Arp, Electro-Comp synths

MIDI Sequencers
Atari Mega 2 computer with C-Lab Notator software sequencing and music notation program
Korg SQD-1 hardware sequencer
Roland MSQ-100 hardware sequencer

Digital Signal Processors
(2) Yamaha SPX-90 multi-effect processors
Yamaha R-100 multi-effect processor
Lexicon Prime Time delay
Roland SDE-1000 delay
Roland RSD-10 delay
T.C. Electronics stereo chorus
Aphex Aural Exciter Type III model 250

Analog Signal Processors
Roland DC-10 delay
Roland SVC-350 vocoder
Mutron Bi-Phaser

Drum Machines
Roland R-8
Casio RZ-1
Korg KPR-77

Tape Recording Gear
Fostex R-8 eight-track
Tascam Porta One four-track
Panasonic SV 3500 DAT deck
Pioneer, Sanyo, Sharp, and Fisher cassette decks
J.L. Cooper PPS-1 tape-to-MIDI synchronizer

Recording and Sound Reinforcement Gear
Tascam M216 mixing board
Fostex model 450 mixing board
Biamp model 1682 mixing board
(2) T.C. Electronics 4-band parametric E.Q.'s
dbx 119, NX-40, and 163X compressors
UREI 565 notch/peak filter set
Dynaco 410 power amp
Roland Cube 60 keyboard amp
(2) Electro-Voice 3-way room monitors
(2) Pioneer near-field room monitors
(2) Audio Technica AT-813 condenser mics
Beyer M500 ribbon mic

Patch Bays
Digital Music Corp. MX-8 MIDI patch-bay
Symetrix 32 point audio patch-bay

BRIAN BATTLES

Ad Ventures

• When it comes to production music, what do you do if you aren't a musician? What do you do if you don't have the ability to play guitar, drums, oboe, panflute, or fancy MIDI keyboards? The simple solution most radio commercial producers turn to is to purchase "canned" music. Production music libraries are available for a variety of applications, in many different price ranges.

In the olden days, I remember going through some dusty production music/sound effects libraries. They featured lush strings and orchestrations comparable in style and brilliance to the finest Muzak offerings. These rugged LPs had somehow survived the decades despite trampling by people wearing golf shoes, being run over by office chairs, dropped behind equipment racks, and being used as coffee cup coasters. It is no exaggeration to state that they still retained their astounding 1 kHz to 3 kHz frequency range and 35 percent distortion. And musicianship! The virtuoso performances found on these masterpieces would someday rival "music-on-hold" performances for its delicate grace, feeling, and sheer intensity.

Nonetheless, times change, and with the advent of the digital compact disc (CD), the ancient analog LP recording went the way of the horse and buggy, rotary-dial telephone, and Billy Beer. Now there are, in addition to old-fashioned LP records, many contemporary production music libraries on the market in modern formats including CD, DAT, cassette, and open reel. Some of the ways radio commercial producers use these special collections to enhance their masterpieces are as jingles, background beds, and to combine canned music with live performances.

If you are, in fact, a confident successful music composer and performer, you may wish to consider producing/selling your own music for fun and profit. If you have prepared a number of music pieces for radio commercials, if you have had some success with your demo tapes, or even if you just would rather record music instead of writing and producing complete advertisements, this could be a good venture for you.

How do you get started? There are two ways to market your service. You can sell some of your existing or custom-recorded material on a contract basis to production companies or advertising agencies in your area, or you can put together a collection of your own original tunes and sell them on the open market. The former approach requires that you make contacts with businesses that need your service. The latter entails that you incur the costs of duplication, advertising, and packaging. If you choose to sell your material to production companies through your own direct solicitation, you can ask more money per selection.

...many well-known production houses got started as simple music production companies and subsequently expanded to encompass copywriting...

On the other hand, if you prefer to go after the wider marketplace, you will make less profit per song, but you may be able to generate a larger volume of sales, i.e., sell the same music to more than one client. It's a decision you can make based on the extent of a commitment you wish to make. You could also start out selling locally, and if you're successful, branch out to offering your service on a broader scale.

The first thing, then, is to take stock of your inventory of music. Do you have a healthy demo tape assembled yet? You can combine jingles, custom-produced music beds, and even material that was prepared "on spec" and never sold; you might even include a few mock-ups done expressly for the demo tape. Add a voice-over track to introduce your service, and briefly describe the contents of the tape. Keep the script for this short and tight, and let the music tell most of the story. Unless you're a professional announcer, spring for a few bucks and hire someone as voice talent.

It's also time to print up some business cards and make sure you have a telephone number that's available (and answered) during the day. (A couple of business tips: Leave your "title" off your business cards, if possible. That way you can present yourself to prospects/vendors/talent as President, Producer, Account Executive, or whatever title is most useful in a given situation. Also, when having a business telephone line installed, most telephone companies will agree to assign you a number you've selected, as long as it begins with the proper exchange prefix for your location, and it's not currently in use. Consider something mnemonically effective like AD-MUSIC [236-8742] or JINGLES [546-4537] or other such suitable number, if you can acquire it in your area.)

As a matter of fact, many well-known production houses got started as simple music production companies and subsequently expanded to encompass copywriting, creating custom jingles, and furnishing complete advertising campaigns. Perhaps you'll be next. db

JOHN BARILLA

THE ELECTRONIC COTTAGE

Using Delay

• This edition of Hot Tips will focus on some creative applications of time delay. It's amazing what you can do with a delay line once you start diddling around with it in earnest. Unfortunately, we're really not encouraged to do much experimentation these days because most out-of-the-box settings appear so awesome. Mostly, we just step through the pre-programmed sounds, find one that pretty well suits our needs and continue recording. Don't get me wrong now, it's nice to have a selection of factory settings at our disposal, but we really miss something if we don't (at least occasionally) customize them to the song we're recording.

Admittedly, it was easier to get involved with "tweaking" a setting when delay lines were simple units with genuine knobs on the front panel instead of just a multi-purpose function button like we find on today's processors. For me, twisting knobs offers more immediate results than the somewhat nauseating task of stepping through a series of parameters. Nevertheless, program modification can and probably should be done for each song that you record. I think it would be wonderful if manufacturers found a way to put more dedicated hardware on the front panels of their digital signal processors. But until that day, don't decommission your old delay lines. They will enable you to run through a lot of creative possibilities very quickly. Who knows, you might even stumble on something really magnificent!

Delay Line Basics. I'm sure many readers have a grasp on this subject already, but it's always worthwhile reconsidering basic premises. So let's take a quick look at the parameters of time delay. On the front panel of almost any delay unit -(or possibly on the LCD readout of a digital signal processor) you will find the following controls:

TIME

This is a fairly obvious parameter. On the older delay lines it's usually controlled by two knobs, a *coarse* tuning control which sets the center point of a range and a *fine-tuning* control which allows the user to set the delay continuously within the specified range. On the newer generation digital signal processors though, you usually have to scroll through the range in consecutive increments. In any case, the result is the same: You can set your basic time delay anywhere from a fraction of a millisecond to several seconds in duration.

FEEDBACK

Feedback (or "regeneration" as some call it) does pretty much what its name says it does:

It literally *feeds back* a proportion of the already-delayed signal back into the delay line enabling one to effectively *echo the echo*. In other words, since the delayed signal is being mixed with the unaffected signal prior to hitting the delay, it too gets echoed again. Depending on how much feedback is integrated with the original signal you may get a long or short series of echoes. And because we usually feedback a much weaker signal than the original, the echoes predictably diminish in amplitude until they are no longer audible.

Of course, if you turn the feedback gain up real high, you will soon find out why they call it feedback, because it will resonate with certain frequencies of the original signal and drive the whole circuit into oscillation. At short delay times, this can be a useful and controllable effect that will allow you to tune the circuit so that it resonates at a particular frequency (I'll give you an application of this later on).

On some units you will have the option to invert the phase of the feedback signal. While by no means an essential function, it is a nice feature which can be utilized in a few different ways. If you are running delays in stereo, kicking the feedback of one side 180 degrees out of phase can impart an interesting *twisting* effect that seems to move from the right channel to the left channel (or *vice-versa*). If you decide to use this effect in stereo, make sure you check for monaural compatibility. The delay settings you choose should be close but not exactly the same setting—or else they may effectively cancel each other out when auditioned in mono. If you decide to use a single delay line placed directly "behind" the original signal in your mix (in other words, emanating from the same direction), the invert feedback switch can enable you to get a whole lot more feedback happening without overpowering your main signal.

MODULATION

This is controlled by two knobs (or parameters) usually labeled *width* and *speed*. The speed control is simply a frequency selector for an LFO (low-frequency oscillator) which generates a waveform (usually ramp or sine wave) in the range of about 0.5 Hz to 10 Hz. This is obviously a sub-audio frequency which is not meant to be heard. Instead its purpose is to modulate the time-base of the delay line. Since the LFO puts out a periodic waveform (it repeats itself going above and below a certain zero point) it is made to vary the delay time above and below the designated setting. While the speed control dictates how rapidly the time delay will oscillate, the width control

dictates how wide of a range above and below the chosen delay time the modulator will be allowed to swing. In other words, will the delay be varied by 5 ms above and below the designated time or 10ms above and below or 100 ms above and below? (If you've ever used a Lexicon Super Prime-Time, the LCD readout will actually show you this swing as a series of constantly changing delay settings incrementally positive or negative about your chosen delay setting.)

If you listen to the delayed signal only, what do you hear? It might be anything from a very subtle vibrato to a tremendous warble. Since we're messing with the time-base (and we know that time and frequency are interrelated), it follows that the pitch of the delayed sound will vary above and below the original sound depending on our settings. Used judiciously, modulation can impart some very interesting and lifelike motion to an otherwise static track. But the modulation settings have to be just right, and the interaction with delay time and feedback needs to be considered, as does the level of the effect relative to the dry signal. It really takes quite a bit of experimentation to arrive at appropriate settings. Lamentably, for most people it seems much less hassle to punch-up a factory preset chorus, double, flange, phase or whatever (all modulation type effects) and forget about fine tuning the parameters. But if you get into developing your own modulation effects, I assure you, it will be rewarding.

OTHER FACTORS

That's about it for the front panel controls. But there are a few other significant factors you should consider when using time delay. One of them is the notion of spatial configuration: where will you locate the delayed signal in the mix relative to the dry signal? It's really a very important consideration, because the same delay setting placed differently in the stereo panorama will render an entirely different effect.

For example, say we have a guitar track that is being sent to a delay with a doubling type setting—say 35 ms of delay with very subtle modulation and no regeneration. The guitar is panned hard left and the delay is also panned hard left. The overall effect in this case would be one very fat guitar—the doubling in this case would meld with the original guitar. But if the delayed signal were panned hard right in opposition to the original guitar, the listener would perceive the distinct differences in timing which were masked when the double emanated from the same position as the original. Hence, the same delay setting will be perceived differently when placed differently in the mix.

Another factor always worthy of consideration is the critical ratio of dry signal to delayed signal. While in most cases aesthetics would dictate considerably less delay than dry signal, in certain cases—such as flanging and doubling type effects, it is mandatory that the dry and effected signal be at very nearly equal volumes before you will be able to really hear them working. Once again, experimentation is the rule.

Finally, equalization is also a factor. Back in the days when delay was accomplished primarily with tape recorders or (later) analog delay lines, the sound came back (because of the nature of the available technology) a little *darker* than it went in. That was not seen as a real deficit back then, because real sound generated in real acoustical environments (concert halls, canyons, etc.) always returned a little dark anyway. High frequencies always seem to lose their "oomph" faster than mids or lows when they are actually bouncing off walls. But today's digital technology allows for full bandwidth even at long delay times—which is actually pretty unnatural. So no matter how many times it bounces off that imaginary wall, the sound never gets any darker. The point being, if you want to get a good density of echoes behind a sound—without actually drawing too much attention to the effect, it may be prudent to roll off a bit of the highs. This is probably more important with long delay times than with short ones, but EQ is always an important factor in crafting a natural sounding delay setting.

A DELAY LINE COOKBOOK

Did you ever sit down and think of how much mileage one can get out of a delay line? Think of all those gaudy-colored stomp-boxes for electric guitarists: the *Super-Flangers*, *Maxi-Choruses*, *Hyper-Echoes* and so on. Of course those names are fictitious, but they do capture the flavor of modern man's need for instant gratification. I don't mean to criticize these boxes—I've bought a few of them myself, and they do have useful applications. But the fact remains, you can perform all of these effects—much more cost-effectively and with greater versatility—with a simple rack mounted delay line.

HERE'S HOW

Flanging with a delay line is an electronic simulation of the classic effect done with two tape recorders running in tandem, and of course, a finger on the flange to vary the tape speed. While the electronic version is not quite as dramatic, it's a lot easier to set up. To do this, set up a short delay—anywhere from a fraction of a millisecond (say 0.05 ms) up to about 10 ms. When mixed with the dry signal in relatively equal proportions this will set up deep regions of frequency cancellation—a phenomenon known as "comb filtering."

If you turn up the feedback, the resonance will increase, thereby deepening the areas of cancellation.

One more factor is necessary to have an authentic flange. You must cause the comb-filtering effect to move up and down—giving that characteristic sweeping sound. To do this, turn to the modulation section of your delay line and dial-in a moderate to slow speed and as much width as you need.

As mentioned earlier, feedback can be used creatively in flange settings. By turning the feedback up just to the point where it begins to ring and simultaneously twiddle the delay time very carefully, you can actually tune the flange to a particular frequency. A couple of applications are in tuning percussion instruments—such as creating a snare drum the "speaks" a tone which is harmonious with the key of the song. Of course with vocals, tuned flanging can impart a *Darth Vader*-like quality which can sound pretty bizarre.

CHORUSING

This is an electronic way to simulate an ensemble sound. In common usage though, the effect comes off as simply a more mellow version of the flange—something that makes guitars and synthesizers sound fatter, and slightly detuned with a nice glistening motion. It's an eminently listenable effect. Simply take the flange setting above, turn off the feedback and run the delay time in the range of 10 to 20 ms and you will have something of the commercial notion of chorusing. If, however, you want to pursue the kind of chorus found on more upscale studio processors, you will have to feed your signal to some additional delay lines set at slightly different delay times and then either mix them all together or spread them around in stereo. The effect will be much bigger and more akin to a large ensemble.

DOUBLING

Doubling capitalizes on the fact that a real, authentic double-track performance—no matter how tight—will still have a certain discrepancies in it; slight timing errors, pitch shifts and timbral shifts. By setting a delay in the range of 20 to 70 ms a certain amount of this native sloppiness can be achieved. Of course, with settings under 30 ms the double will seem rather tight, but from 30 ms on up the desired degree of looseness can be dialed in. A little feedback can help elongate the sound, and it will also cause some increased phase cancellation—all subtle cues of a realistic vocal double. A certain amount of modulation will also help make the performance less predictable. (Without modulation, the delay factor would be constant, but with modulation it would be more variable, hence more natural sounding.) Of course, like all the above-mentioned effects, it is most powerful when mixed with the original sound in about a 50:50 ratio, but even lesser amounts can be used to subtly support and warm up a vocal without drawing attention to itself.

SLAPBACK ECHO

This is an electronic simulation of the original 7.5 and 15 in./sec. tape echo—and of course, everything in between. It's hard to imagine that echo was once limited to two flavors—slow and fast—but it's true (at least until some entrepreneurial person invented the movable tape head). And many classic hit records featured this effect and little else. I once sat down to try and figure out what delay settings would best correspond to tape echo. Using the standard physical relationship $v=d/t$, I plugged in either 7.5 or 15 in./sec. for velocity and an assumed average distance between record and playback heads of about 1.5 inches. Solving for time, I concluded that 15 in./sec. tape echo is about 100 ms long and 7.5 in./sec. is exactly double that at 200 ms. Of course those delay times could increase a bit if the head distance was closer to 2 inches, but in any case a delay setting between say 100 ms and 250 ms should cover the field quite authentically.

I don't recall hearing much feedback in records from the 1950's, but that doesn't mean you can't use it. A little feedback will prolong the decay of the echoes by adding extra echoes of diminishing amplitude right behind it. Sometimes it can provide an interesting trail behind lead instruments. But definitely stay away from the modulation section on this setting. This is meant to be a static and predictable effect. Any wavering will spoil the whole trick. It's the very precision of the echoes that gives tracks a very flattering constant cascade of sound.

LONG DELAY

Long Delay involves settings of any sort beyond the slapback range—250 ms to as much as 1 whole second and more. I find the 400 to 600 ms range most useful. Once again, this is generally used as a static effect—no modulation. But feedback—well that's another story. Long delays coupled with feedback can create a "mystical cloud of action" that can make almost any performance seem intriguing. Notice how rock guitarists use long delay as an integral part of their sound. It makes them sound like they are playing twice as fast as they really are playing. And mistakes? Well, who can point them out when they are drenched in the mystical cloud.

All kidding aside, long delay is a beautiful, intriguing effect that can render insignificant many normal performance glitches. A famous saint once said, "Love covers a multitude of sins." So does long delay.

SYNCHRONOUS DELAY

This is a quite useful application which automatically creates interesting and highly rhythmic embellishments of an existing track. It does this, of course, only after you figure out how to set the delay. This involves a good pair of ears and/or a few mathematical tricks. While I can't do much for your ears, I can show you the mathematical tricks. But before I begin, let me mention one caveat: the tempo of the tune needs to be a steady one. In other words, whether played by MIDI sequencer or live performance, to really get the most mileage from this effect, the rhythm should be constant.

Synchronous delay can be used on almost any track, but deriving your optimum setting is most easily accomplished using a steady rhythm part—such as drums. You can simply start by soloing drums with the delay and listening for the appropriate rhythms to develop, or you can use some simple mathematical calculations as a point of departure. The latter way is usually most efficient.

To get the setting in the ball park mathematically, you must first determine tempo. That may be as simple as looking at an LCD readout on your sequencer or drum machine. If, however, the tracks have been recorded on tape (or if it's a live performance) just get out your stopwatch and count how many downbeats go by in the course of a minute. It's sort of like taking a pulse, so you can use the old nurses' trick: count how many beats occur in 15 seconds and multiply by 4.

You will thereby arrive at a number which will represent the tempo of your song in beats-per-minute (BPM). While various kinds of complex rhythms can be created using synchronous delay, the most commonly used setting—and a good starting point for all other rhythmic excursions—is the simple ¼-note delay.

If you are a trained musician you will immediately grasp this concept. If, on the other hand, you are more of a "techy" and less of a musical type person, a little description might be helpful here. Let's say I already know the proper delay setting. If I play a single hit on a snare drum exactly on the first beat of a song, I should hear the second beat coming from the delay. Then if I turn the

EFFECT	DELAY	FEEDBACK–SCALE OF 1-10	MODULATION SCALE OF 1-10	MIX	COMMENTS
Flanging	From 0.05 ms to 10 ms	0–7	Width:5–10 Speed: 1–4	50/50	Increased feedback will cause filter ringing—for *tuned* flanging. No modulation yields a *static* flange—more akin to phasing.
Chorusing	From 10 ms to 20 ms	0	Width:4–8 Speed:1–4	50/50	Single delays are okay for guitars and keyboards, but for a better effect on vocals set up multiple delays.
Doubling	From 20 ms to 700 ms	0–4	Width:1–4 Speed:1–4	50/50	Use less than 50% delay for a more subtle support.
Slapback Echo	From 100 ms to 250 ms	0–2	Not used	According to taste	Subtle use adds warmth to a track. More overt use adds motion, but overuse may sound gimmicky.
Long Delay	From 250 ms to 1s	2–4	Not used	According to taste	Adds drama and mystery to a track. Excellent on lead guitar and vocals.
Synchronous Delay	Derived by ear or by mathematics	Derived by experiment	Not used	According to taste	First, derive ¼-note delay: (60,000 divided by BPM). For *straight* rhythm use the above setting or multiply or divide by 2. For *shuffle* add 50% to ¼-note figure. Multiply or divide as before. For *triplets* divide by 3 then proceed as above.

A compedium of delay effects.

feedback control way up on the delay, I should hear a long series of beats that keep the actual tempo of the song going perfectly. The delay plays the beat, one beat (i.e., one ¼ note) behind the actual track. ⅛-note delay would be double the timing of the ¼-note delay: for every single note played into the synchronous delay, I would get not one note back, but two. If I turned up the feedback, I would hear a steady in-tempo stream of notes in the ratio of 2:1. A 1/16 note delay would multiply the echoes by another factor of 2 giving 4 hits out for every 1 hit fed into the delay.

How do I figure out a quarter note delay for a given tempo? The formula is quite simple: simply divide the number 60,000 by the BPM of the song. So, for example, if the tempo of my song is 120 BPM, the synchronous ¼-note delay would be 500 ms. To get an ⅛-note delay, I would simply divide the derived ¼-note setting by 2 and arrive at 250 ms. And so, a 1/16-note delay would be 500 ms divided by 4 or 125 ms.

If desirable, more complex rhythms can also be derived once I've figured out the basic ¼ note setting. For example, the ever popular *shuffle* feel frequently heard in blues, jazz and even some rock music is characterized by a perambulating, forwardly propelled rhythmic feeling—much akin to walking or skipping: ergo the term, *shuffle*. Musically, this refers to dividing a beat into two unequal parts, with the first of the two beats being longer in value than the second. While there are definitely several *feel* variations, the weighting of the beats usually comes down to something like this: the first is increased in duration by 50 percent and the second beat is decreased by 50 percent.

To achieve this rhythmic feel with delay, just take the setting you derive for straight ¼-note delay—say for example, the previously mentioned figure of 500 ms—and add 50 percent to it (500 ms + 250 ms = 750 ms). You are now in the realm of the shuffle feel. If you'd like the feel to move twice as fast, divide the setting (750 ms) by two (which gives you 375 ms).

Still not fast enough? Divide that figure also by two (187.5 ms). Get the picture? I have personally used this setting over a rather *straight* drum beat and gotten an hip jazzy feel from the part. What's more, if I wanted an interesting fill, I would turn up the gain of the send to the delay line and sometimes goose the feedback also. The fills that emerged were innovative and much like a real drummer (one with oodles of finesse) might execute.

Finally, you may sometimes want to use a synchronous setting that allows you to put a *poly-rhythmic* feel to it; an odd number of beats would transpire where normally an even number would occur. Something like the musical *triplet*, a ratio 3:2 would be the most commonly used example. By now, you may have developed the mathematical intuition to derive this setting. (Test yourself: think about it for a moment...)

Here's the answer: Simply take the basic ¼ note delay figure—(once again, let's use the 500 ms from our previous examples)—and divide it by 3 (yielding 166.6 ms). Now this is a triplet delay figure, but perhaps—since this is a short delay, it's too fast for your tune. In order to get a longer triplet feel simply multiply this figure by two (yielding 333.3 ms). And if that still doesn't do it, well, multiply it by two once again!

SUMMARY

Perhaps this seems a little complex, but it's really not. Once you derive your basic rhythmic feel—whether it be *straight* (¼-note, ⅛-note, etc.), *shuffle* (add 50 percent), or *triplet* (divide by 3), you then find the appropriate quantum level by multiplying or dividing by a power of 2. It's really pretty simple once you get the hang of it. Don't think about it too much. Just do it. db

BRENT HARSHBARGER

Audio for the Church

• In the last issue, I talked about the mixing console, its functions and how it operates. I also promised a test for this issue. I will then cover some hands on operations that can be encountered on any given Sunday morning.

First the test:

1) Can you add an effects device just on one channel, and if so, would the input signal be affected by the equalization circuit?

2) How many channels can you assign to a group output, and does the EQ circuit affect this function?

The first answer is *yes*: you can add an effects device just on one channel through the insert jack, and *no*: the EQ does not affect the device in this instance. For the answer to the second question; you can assign as many channels as the board has to a sub-group, and *yes*: for every channel you assign to the group, that individual channel EQ circuit affects the sub-group.

TYPICAL SETUPS

Now for the main course of this issue, hands-on operation. I am going to take you through a typical Sunday set-up as if you are visiting my church. The first thing that I do is plug in the mics that are going to be used for the service, pulpit, and lectern (including any special mics also to be used for the service). Then I make sure that all the monitors are plugged in, and in their proper place. Next, I go to the mixing area and check to make sure that the masters are down on the console. Now proceed to power up the mixing console and miscellaneous outboard devices, waiting a few seconds before powering up the amplifier rack.

Starting the power up in this order prolongs the life of the amps and the speakers, and also is less annoying to anyone in the sanctuary.

Now I bring the masters up to about a –5 or –0 dB, and if your console has only a 1 through 10 numbering system—bring it up to approximately 6.5. Now have the pastor step to the pulpit (or whomever uses the pulpit the most), then slide the fader to 0 dB and have the person talk into the pulpit mic as you *slowly* bring up the gain pot until you have a strong, but not overpowering level. Going through this procedure in this way will give you the best signal to noise ratio. By this time, the soloist and the quartet will be ready for their soundcheck.

In this example, the soloist will be using an accompaniment track. There are many types of accompaniment tracks, but you will find that the best are the tracks that are taken from the master tape of the original recording artist. The others vary from poor to acceptable. In a soloist situation, the first thing you do is bring up the slider to 0 dB on the channel with the tape, then slowly bring up the gain until you consistently have a –6 to –9 dB on the board's VU meters. Now adjust the EQ to make the tape sound full. On the tracks that are from the original recording artist; you can usually cut the EQ in the lower-mids approximately around 250 to 400 Hz. Do not cut it any lower than 9 dB, this will take out the boominess.

The only way to set-up the mix for the choir is to be at a rehearsal with the choir.

EQUALIZATION

Next, increase the bass to around 80 Hz, but not more than 100Hz, to about +3 to +6 dB. Finally, boost the highs until they are crystal clear. However, if all you get is tape hiss, back it off some. The third-party accompaniment tapes vary, but the majority of them tend to be bassy and lack highs other than tape hiss. These you will have to judge on a tape-by-tape basis. Before moving on to the singer, double check the VU meters to make sure that you are still averaging between –6 and –9 dB, because you could have increased the level while you were EQ'ing. If your level did increase, turn the gains down to compensate for it. We're now ready for the singer. Again, slide the fader to 0 dB and slowly bring the gain up until you are averaging –3 to 0 dB on the VU meters. From this point, adjust your mix from the channel faders. One way of testing your mix while using an accompaniment tape is to match the soloist with the backup singers on the tape. When the soloist and the backup singers are in unison, you have accomplished a good mix. One word of caution, the third-party tracks may or may not have a good mix between music and backup singers, but the master tracks are ninty-nine percent dead on.

THE SOUNDCHECK

The gospel quartet is now up for their soundcheck. First, bring up the piano to the same VU level as we did for the accompaniment track. We are also going to set the mics up the same way as we did for the soloist with only one variation—how we bring up the gains. Start by bringing up the bass singer's gain until he has a good strong level, but not above 0 dB on the VU meter. Next, bring the first baritone to match the level of the bass singer so they sound like one voice, then bring the second baritone to the first baritone. Finally, bring the tenor up to match the second baritone. Check yourself by listening only to the bass and tenor; if they sound like one voice then you have completed the process. Any other adjustments should be done on the channel fader.

The only way to set-up the mix for the choir is to be at a rehearsal with

the choir: making notes on the levels and keeping them for Sunday. A good tip on EQ'ing the choir is to cut the bass –6 dB at 80 Hz (if your console doesn't have a high-pass filter), and to boost the mid-lows approximately +3 to +6 dB at a frequency of 250-450 Hz. This will give you a full crisp sound.

In the next issue, I will be covering equalization in detail, but for the sake of mixing operation; I would like to point out a rule to follow for equalization on the mixing console. First, try to always cut the EQ instead of boosting it. For example, if the sound is bassy, cut the bass instead of boosting the highs. This is the proper school of equalization, and by doing it this way you can increase your headroom on the mixing console. Audio is full of compromises: there are times you will want to boost the EQ, but only experience will let you know the right time to do it. (You should always try to think where you can cut EQ, however.)

ADDING REVERB

When using sound processing gear such as a reverb or delay or any other special effects devices; the first rule is to use it in moderation, too much can be very distracting. If you are selecting a reverb for a soloist, a small-to-medium bright room will work very well—anything larger can be obtrusive. For quartets and ensembles use a medium room or a small hall setting. Whereas, with choirs a bright medium to a large hall setting works very well.

Since we have gone through a set-up for a problem-free Sunday morning, let's now imagine that things didn't go so smoothly. Let's say that you have some distortion somewhere. Distortion is a raspy sound caused by too much signal going through the electronics of your sound system. The first thing that you want to see is if there are any peak LED's on. If there is one on (let's say channel three), first turn your gain down. If the gain is already down, then you need to check to make sure that you do not have a line level plugged into a microphone input. A line level is over 1 volt in a balanced system, whereas a mic level is approximately 2/1000 of a volt. If all peak lights are out, then you need to check to make sure that you are not overloading the input of your processing equipment. You can do this by using your PFL on your effects return section (or wherever you are bringing the effects back into the console), and listening to see if only one instrument or vocal is distorted, or if all are distorted. If only one is distorted, turn down the effects send pot in the corresponding channel.

Let your ear be the judge and tweak, before the service, as I described in this article.

Feedback is the next area for troubleshooting. There are many causes of feedback and there are different ways to help reduce feedback problems. The first one is to make sure that you are not using too many microphones, because this reduces your gain before feedback. Also, make sure that your microphones are three-times the distance as the source to mic distance. From my observation, these are the two most common mistakes when it comes to the actual source of feedback problems—more than anything else. Others include: having a microphone too close to a very hot monitor, and improper equalization.

The EQ settings that I have described above have been proven to work as a good starting place in a well-designed sound system. Again, the above is only a guideline. Because of different room acoustics, house equalization, and limitations in the equipment in your church, these may not be ideal settings (but the principals still apply in any case). Let your ear be the judge and tweak, before the service, as I described in this article. Practice makes close to perfect.

Finally, I would like to mention an important part of the sound system operation. There is a science that can only be experienced, which I call sound psychology. This is the science of dealing with an audience of "perfect-ear" experts. This can be very discouraging to anyone donating a lot of time and trying for perfection. But, in most cases it's not you. Listen and respond to each complaint, but I'm sure you will find a few people who are never satisfied. I have come to the conclusion that these people can't hear the mix because they are too busy complaining.

One Sunday morning, the music minister at the time came up to me and said the choir was complaining that they couldn't hear, and that he wanted me to turn up the choir monitors. I told him I couldn't because I was on the verge of feedback, to which he replied that he wanted something done about it. So, I measured the SPL (sound pressure level) in the choir loft, then measured the SPL from the house (and don't you know it), the choir loft was 6 dB louder than what the house was getting. I went back and told the music minister what I had found.

This wasn't going to satisfy the complaining choir, however, finding out later that it was only a handful of choir members who can never seem to hear. So, the only thing that I could think of to do was to tell the music minister to give me a sign when the choir monitor was loud enough. He agreed and proceeded to the loft. As soon as he left, I turned down the master monitor send a considerable amount. The choir came in and the pastor came up to the pulpit and started talking; the music minister motioned for me to bring up the choir monitor, so I slowly turned up the master until he motioned for me to stop. The strangest thing happened—the master send was stopping at a lower level than what I was previously setting it at.

The next few Sundays we set the monitors up the same way, and every one could hear with no complaints. What was the difference? Sound psychology. The people heard a level change which satisfied their minds that the speakers were louder. The reason for me telling this little story is to let you know that you are not deaf and crazy as you may sometimes think. I could tell you all the technical information in the world, but you couldn't really master sound system operation without knowing a little psychology.

COMING NEXT ISSUE

In future segments of *Audio for the Church*, we will examine equalization, and an applications look at multi-track in the church, including a visit to a church making good use out of their multi-track tape deck. db

JOHN BARILLA

A Tale of Two Cottages

This is a tale of two cottages. One cottage is in a big city, the other in a quiet suburb. One cottage is occupied by a successful songwriter heading for a banner year, the other by a struggling songwriter seeking his first major cut.

Of course,these are not thatched cottages but Electronic Cottages—the 1990's revival of that ancient concept: the cottage industry. As futurist Alvin Toffler aptly pointed out back in 1980, computer technology has brought us around full circle to a pre-industrial lifestyle where work place and living space can be united once again.

So as we turn the corner of this decade, it seems appropriate to take an in-depth look to help document this quiet revolution in the field of audio.

To this end, we bring you the profiles of two songwriter/producers and the Electronic Cottages that have become the nexus of their creative lifestyle.

Jeff Kent: Conquering The World With Music

A true believer in the Electronic Cottage is Jeff Kent. The compact, but powerful studio in his New York City apartment has seen a lot of use and paid-off for Kent in a big way. He is a professional songwriter—a person who gets up in the morning, sits down at a keyboard in his living room and begins to create a musical product. He is respected for his unique sense of aesthetics and people pay him for it. Sometimes, other artists end up re-cutting his compositions, but lately Jeff Kent has found that a lot of his recordings are going directly from his Electronic Cottage to CD, television or movies. While most of the credit should go to his considerable skills as a writer and producer—a hard won trophy which does not come easily nor overnight, none of it would be possible if Jeff had not invested in a quality, personal-use facility.

Kent, who spent six years as a staff songwriter with two publishing companies, has this to say about his decision to turn his apartment into an Electronic Cottage:

"When I signed a deal with Mary Tyler Moore Music (MTM), that's when I made a decision—rather than do all these demos at different studios, I decided, look, give me all the money you would give me to do all these demos in the studio in one lump sum, let me take some of my own money, put it together and build my own studio and truly become independent for the first time—'cause I found myself writing with people just because they had studios. I didn't want to do that anymore."

Building his Electronic Cottage was a real liberating factor for Jeff Kent. It led him to do more co-writing with people he really admired. And because of his songwriting, and his ability to play several instruments and to make broadcast quality recordings, Jeff also gained access into the world of advertising, doing jingles for several well-known clients.

Jeff notes that he experienced the typical learning curve when it came to figuring out how to interface and operate this collection of equipment, which incidentally, he bought all in one fell swoop, with very little previous technical experience. We can imagine the anxiety he experienced trying to assimilate the technology:

"So I made a commitment to really learn how to use all the technology—at least enough to become independent. I bought all this equipment at one time, so for four months I was just sitting with a stack of manuals. It took a while to become operational, but once I did it was incredible."

Kent stops for a moment and reflects on how autonomous the Electronic Cottage has allowed him to become, something even he would not have envisioned a few years ago. He goes to the shelf and pulls a DAT cassette and plays the new spot he recently finished for Coca-Cola International. It is (believe it or not!) a heavy-metal Coke jingle, slated for world-wide use. Not surprisingly, it is a clean and spacious recording—as a digital master should be. But beyond that, it is fresh and spontaneous sounding, as if it was actually played by a live band. As I listened to more of Jeff's material, it became clear that achieving a "human feel" was a hallmark of his production style.

MUSICAL ROOTS

How did Jeff Kent develop the ability to make electronic instruments really jam? Undoubtedly, his musical past has much to do with it, for he is a musician first and a programmer second. His first successful band,

called *Dreams* recorded two albums for Columbia Records in the early seventies. It was one of the prototypical jazz/rock fusion bands from that era, and featured the likes of Billy Cobham on drums and the famous Brecker Brothers on horns. Like *Chicago* and *Blood, Sweat and Tears* (who Jeff Kent also wrote for), *Dreams* also featured intricate arrangements and smokin' solos. After the demise of *Dreams*, Kent formed another group which included Robin Batteau and David Buskin who, like Kent, have gone on to become successful song and jingle writers.

...it seems that MIDI sequencing has become so dominant that today many producers would opt for a sequenced performance over a live one...

During the mid-seventies Jeff switched from live music to serious songwriting, going through a journeyman period as staff songwriter and in-house producer with April-Blackwood Music (and later on with MTM). In the early-eighties Kent began a very fruitful collaboration with the legendary songwriter Ellie Greenwich from which a host of covers by the likes of Cyndi Lauper, Nona Hendryx and Ellen Foley have generated. It was during this period, in his association with Ellie Greenwich, that Jeff became smitten with the notion of a personal-use studio. He had helped Greenwich choose equipment for her home studio and seeing the benefits, realized that this was the way to go. In 1986, when the money became available from his deal with MTM, he decided to go for it. In just a few months of operation, he had generated a song for a major motion picture, *Roxanne* with Steve Martin and Darryl Hannah. Recorded in Kent's studio, the mix he did went directly into the film soundtrack—a testimony to the practical value of the Electronic Cottage.

COTTAGE UTENSILS

Jeff's choice of multi-track recorder is the Akai 12-14, which he advocates highly, and he mixes down to a Panasonic SV-250 DAT recorder. With digital inputs and outputs and a choice of sampling rates, mixes can be bumped over to professional editing systems for mastering without ever leaving the digital domain. For MIDI sequencing, Kent has made the somewhat unusual transition from software sequencing back to hardware sequencing—a move that some "technoids" might consider to be a bit retrograde.

Figure 1. Jeff Kent with his Mac 51 and Akai 12-14. The Linn 9000 is in the foreground.

But Jeff Kent has his reasons, and this may partially explain why his music has the sound of an inspired first take, rather than antiseptic perfectionism. After spending some time on a Macintosh 512 with Opcode sequencing software, he returned to his trusty old Linn 9000 because (as he put it), "it is musician sensible."

While there is no denying that the software sequencer can do more things, Jeff found the "excessive editing options" to be a distraction. Yes, it was more versatile, he goes on to say, "but that was almost a problem. Like you could spend 20 minutes on a note (tweaking the timing, volume, etc.). I don't have those choices (with the Linn 9000). Either it sounds good or it doesn't. If it feels good in real-time, yeah, I could move it a little bit, but is that going to make the difference be-

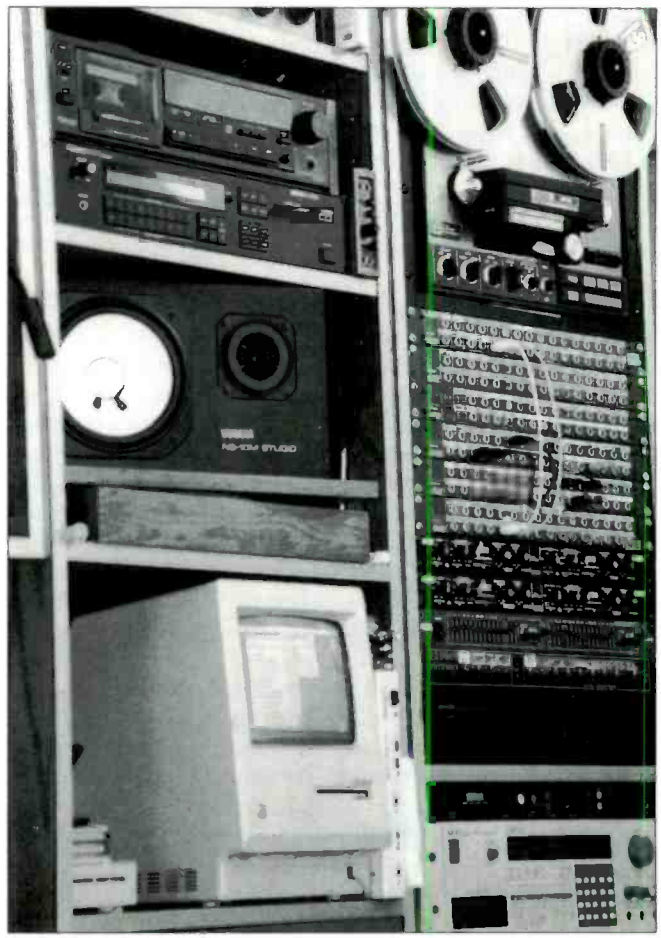


Figure 2. Jeff Kent's equipment rack as seen at the rear of the equipment wall of Figure 1 is here shown in detail.

tween my song sounding like a great song or not? I don't think so!"

If real-time time sequencing is one of Jeff Kent's production secrets, then so is the use of real musical performances on non-MIDI instruments. Was there ever a pre-MIDI age? Alas, it seems that MIDI sequencing has become so dominant that today many producers would opt for a sequenced performance over a live one—because the sample is "fixable." Not so with Kent. While some of his tracks are sequenced, a good deal of his production always involves live performances. Playing keyboards and a variety of stringed instruments, percussion and vocals, Jeff frequently performs many of the live tracks himself—directly to multi-track. (It seems weird that in 1990 we should be patting a guy on the back for having the guts to play live in a recording studio, but here we are...)

Even though he is able to play all of the tracks, Kent is also a savvy producer, and when specific "feels" are required, he calls in a specialist. What really struck me was how specifically Jeff casts his musicians. Over his many years in the business, he has apparently developed a extensive list of players and vocalists, and it is probably for this reason that his recordings (from Spanish music to country rock to heavy metal) always have that authentic ring. In short, because of these live, wild-card elements (the real-time performances), Kent's music does not suffer from the stiff, mechanical feel so prevalent in sequenced music.

Jeff does, of course, utilize MIDI sequencing. Rather than run tape in the beginning stages of his production, he generally uses the Linn 9000 to record basic tracks: drums, keyboard pad and perhaps bass (sometimes bass is done live with Jeff playing his ESP bass guitar). At some early stage though, Kent voices his synthesizers the way he wants them to sound and records the synths and drums over to tape. He does this because he likes the convenience of set-up for his overdubs. When he powers up the next morning, there are no programs to load, no MIDI channels to set up, nor voice patches to choose. He simply fires up the Akai 12-14 and begins to record his overdubs. While some might argue that one should preserve the pristine quality of the digital synths and sampled drums as virtual tracks, Jeff says that he likes the way these tracks sound better after they hit tape. While his production methods are not normative ones, they work.

It is also clear that Kent's lack of regard for convention is really a function of the rebelliousness in his own personality. For this reason, Jeff makes it a conscious policy to employ a lot of talented younger musicians, because he identifies with their energy level. He explains:

"A lot of people that are my age I don't connect with musically, because they're into a whole other kind of music and I for some reason, just connect with this very raw energy.It doesn't matter whether it be a sophisticated song or just a heavy metal piece. Whereas, I know people of my age group (Kent appears to be chronologically at least, part of that "thirty-something" generation) that if I said, 'heavy metal' to them, they'd go running to the hills, but I happen to love it. I'm not into bad lyrics and stuff like that, but I am into the rebelliousness of it, I am into the energy of it. So I do it, I put it in my music and it seems to be...I seem to be coming into vogue right now."

It just came to him one day in a great flash of *satori* that Spanish, Italian, French and Jewish music share the same origins.

CONQUERING THE WORLD

Jeff Kent has been doing aggressive music for some time now, but he was not universally accepted for it. Now with his new heavy metal Coca-Cola ad, things may be catching up. It is probably most ironic, since Jeff was known for a long time, as a ballad writer. Now he's getting calls for sports music, which is the essence of high energy, hard-edged music. But despite stereotypes, Kent does not seem to be limited to any particular style of music. In fact, he's probably about as close to a musical chameleon as you can get, which is indeed a high virtue in a songwriter. It means quite simply, that Jeff is capable of tackling almost any market, and doing it with a good deal of authority. In fact, Kent's writing and personal publishing company (Jent Music) is headed for that broad, international market.

For example, Jeff has recently garnered three cuts on the debut CD of British artist, Robin Beck. One of the songs features tracks that were recorded in Kent's Electronic Cottage. The tracks were simply too good not to use, so they were transferred from the Akai 12-track in Jeff's living room to a standard 24-track recorder and finished. (This kind of inter-activity between the large professional studio and the smaller personal-use Electronic Cottage is really what recording will be like as we enter the '90s.)

Jeff's most current opus is a very moving Spanish (contemporary-pop) tune which was demo'ed with Julio Iglesias in mind. The idea here, of course, is to pitch Iglesias using a production that sounds tailor-made for his sound. Kent's production succeeds in capturing all of the heart-rending emotionalism Iglesias is famous for. Interestingly, Jeff Kent does not speak Spanish, nor has he studied Spanish music. It just came to him one day in a great flash of *satori* that Spanish, Italian, French and Jewish music share the same origins. Kent, who is Jewish, remembers his grandfather (who was then studying to be a rabbi) singing to him as a child. It was a sound he never forgot. When he started to listen to music from other Mediterranean countries, he realized that in some strange way, they were all playing the same song!

Jeff puts it like this: "It's all Semitic music. It's all based on the same scales, and as they (the Semitic peoples) went to different regions, different instruments were picked up and incorporated into the music. This is why I'm discovering that I'm starting to write a lot more Spanish music. I'm realizing that there are a lot of Jewish folk songs that I grew up playing, that if all I do is turn it into a tango and change the words...! I mean, there's public domain songs which could be giant hits all over again."

With twenty years of professional musical experience propelling him forward like wind on his back, Jeff surveys a new, larger market for his songwriting—larger than he ever might have imagined. Is it possible to conquer the world with a song? Undoubtedly, Jeff Kent will give it a try.

Brett Charles: Burning The Candle At Both Ends

It's not easy burning the candle at both ends, but that's exactly what Brett Charles does. By day he is a skilled craftsman manufacturing custom furniture. By night he is also a skilled craftsman, but his trade is of a different sort. By night he crafts songs in his Electronic Cottage. It's really like having two full-time careers. Both are demanding, both require precision, but only one pays the bills. At least so far.

His profile, of course is not unusual. It's the struggling songwriter syndrome. But for now, Brett has become adept at juggling both careers. He realizes that things take time to happen in the music business—that "overnight sensations" are usually ten years in the making. So he goes in to his shop earlier in the morning (7 a.m.) so that he can leave earlier in the afternoon (3:30 p.m.). And if his timing is right, he may be able to make a few calls to the professional managers at some publishing companies before they go home for the evening. Then when evening comes, Brett can be found working in his bedroom/recording studio/office, writing songs until the sandman rocks him to sleep.

One of the reasons I put together this studio is because I tried to put together several bands, but my experience always was that people would call up at the last minute...

While he hasn't been offered any great deals yet, his initial attempts at marketing songs have been encouraging. Music industry people have been recognizing his talent and doors are opening up him. Brett realizes that success as a songwriter means developing relationships with publishers, and that requires a constant flow of new material. His Electronic Cottage has been an incredible boon to his productivity.

Interestingly enough, until a few years ago, Brett was almost blissfully unaware of the technological revolution that was going on around him. Even though he had been writing songs for many years, he hadn't really pursued it in a professional way. With an old Farfisa organ, some drum-drop records and a tape recorder, Brett wrote songs and recorded them, but it wasn't until he went to The Keyboard Expo at the Sheraton-Centre Hotel in New York City three years ago that he became smitten with "the MIDI bug."

GETTING HIP TO MIDI

He relates an interesting story about how his eyes were opened to the new technology. "I'm a songwriter," he says, "not an engineer. Up until three years ago, I didn't even know what MIDI was. Then I caught this guy demo'ing the Synergy (an additive synthesizer popularized by Wendy Carlos) and I said to him, 'Can I ask you a question? What's MIDI?'"

Apparently, the chap who was demonstrating the keyboard couldn't restrain himself from laughing at Brett's naivete. In any event, Brett never got a straight answer to his question that day. So he went out and bought every book he could on MIDI and recording. He bought pieces of equipment, experimented with them and learned them. As new pieces came out on the market, Brett quickly sold off pieces he already had (while they still retained their value) and purchased additional equipment. The process mushroomed until his room really got out of hand. There were wires literally all over the place.

CUSTOM COTTAGE

Now here's where it comes in handy being a furniture maker in a family-run business. Brett and his mother (who is an interior designer) put their heads together and developed an innovative way to make Brett's room do triple duty: as bedroom, office and recording studio. They arrived at an overall look for the room and then Brett modified the design so that equipment could be located at appropriate heights and distances. Many pieces were designed with roll-out racks so that positions could be adjusted according to need. The result is a studio that can actually disappear from view in less than 10 seconds!

While this concept is not exactly a traditional studio design, it can be put to good use in family dwellings where space may be too precious to dedicate to a studio facility, but can possibly be time-shared with other family activities. Some studio owners have been so impressed with the convenience and efficiency of Brett Charles' studio that they have asked Reacine Interiors of Freeport (the family custom-furniture business) to design a similar unit for their homes.

Regarding the Electronic Cottage era, Brett says: "I'm so happy I'm alive during this time—that I'm able to do this in my home. One of the reasons I put together this studio is because I tried to put together several bands, but my experience always was that people would call up at the last minute (before a rehearsal) and say, 'I gotta go shopping with my wife.' I hate to sound like this, but I couldn't depend on people. I had to do it myself. When I found out about MIDI sequencing and being able to play all the instruments myself, I was like yeah, let's do this!"

MAKING THE PITCH

Brett has been pitching his songs to Warner-Chappell, Jobete, MCA,

Figure 3. Brett Charles is seen in front of his disappearing studio. The cabinets were custom built.

Figure 4. Brett Charles has a lean and clean Electronic Cottage.

Chrysalis and Peer-Southern, thus far, and has received all positive feedback, but as yet, no signing. I did get a chance to listen to some of Brett's material. One song, co-written with Don Bader really stuck out in my mind. It was Brett's newest production, and the publishers had yet to hear it.

If it grabs them the way it grabbed me, I foresee less furniture making and a whole lot more songwriting in Brett's future. The song entitled *It's Over*, has a relentless bass line and incredibly hook-laden vocals. It has the sensuality of a George Michael tune, and the smoky, bluesy melody of Sade. My opinion is, this tune *definitely* has all the "right stuff" to make a hit record.

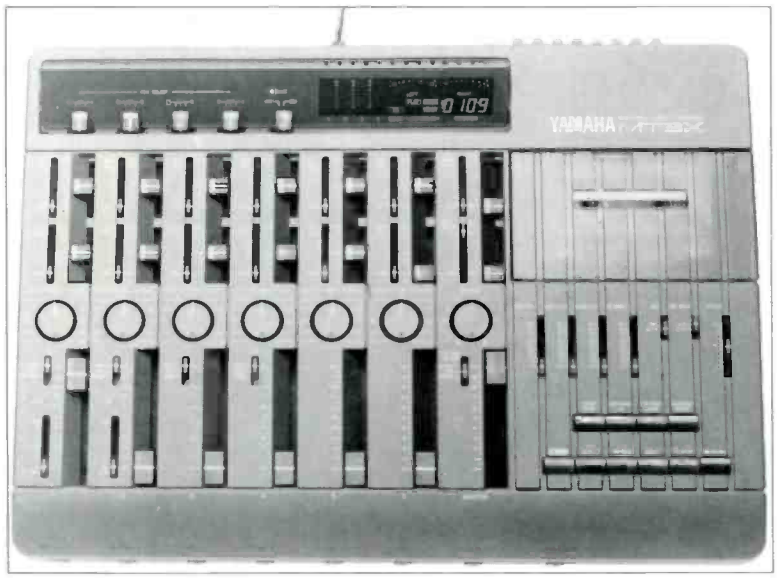
Brett has recently begun to get some visibility as a writer. He did the music on a Franklin Mint TV spot for their new Eagle Watch and is writing songs with producer Richard Bennett on an album project for Michelle Thomas (one of those Cosby kids) who is recording on Apollo Records.

So things are beginning to happen for Brett Charles. The light in his Electronic Cottage will probably be on later and later in the night from now on. db

LEN FELDMAN

Lab Report

Yamaha MT3X Multitrack Cassette Recorder



GENERAL INFORMATION

• The MT3X is a virtual production studio in a tiny package measuring only about 17-½ inches in width by 4-¼ inches in height and 12-¾ inches in depth. Essentially, this versatile little package is a 6-input mixer plus a dual speed 4-track cassette recorder. Line level signals can be applied to all six inputs, two of which also accept microphone inputs. No patching is required to route tracks to faders for mixdown, as selector switches are provided for that purpose. Each channel is equipped with high and low frequency EQ controls. Channel pan controls let you place your sounds anywhere in the stereo sound stage that you create. The MT3X also has two ef-

Figure 1. Frequency response of the cassette recorder section. Narrower bandwith was obtained with tape speed at 4.75 cm./sec.; better response was achieved at 9.5 cm./sec.

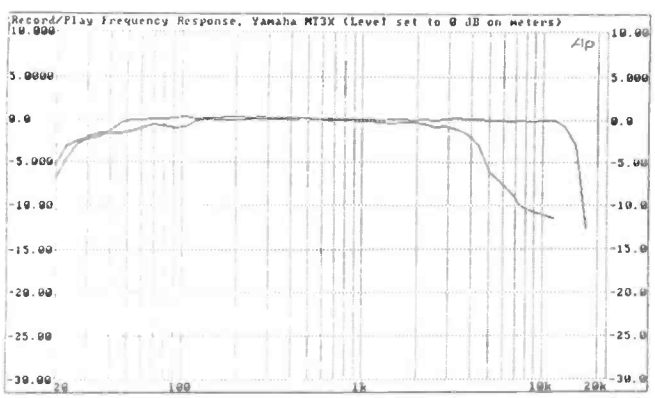
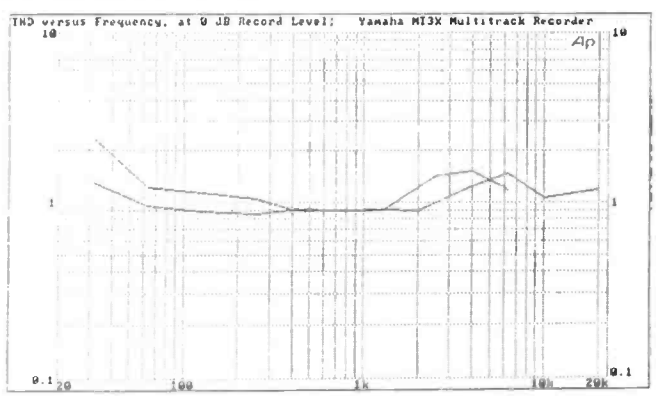


Figure 2. Distortion versus frequency, record level set at 0 dB on the meters of the Yamaha. Higher curve is for 4.75 cm./sec. speed; lower THD curve is for the 9.5 cm./sec. speed.



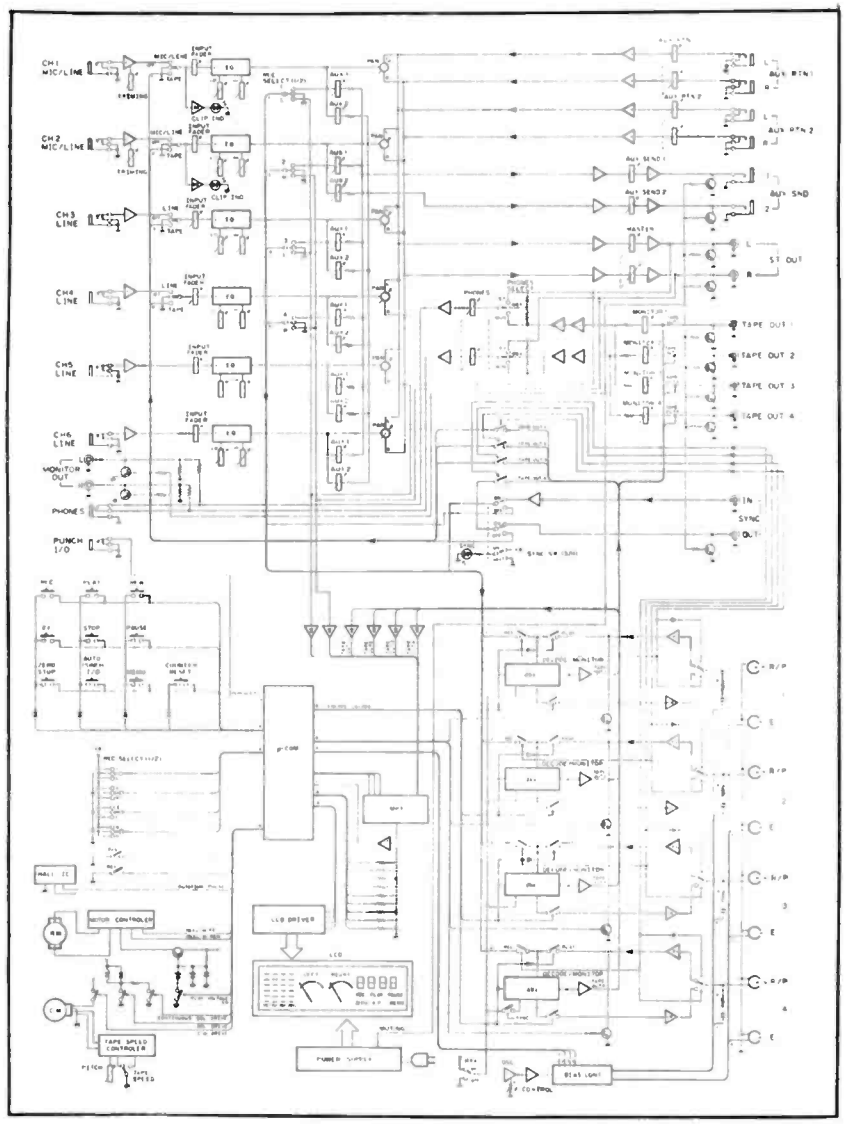


Figure 3. Block diagram of the MT3X.

fects loops allowing you to send to two separate effects devices from any one of the six available input channels.

Each track can be processed with its own effect using the individual tape-out jacks. *Master aux* send and return controls are found right on the front panel. Track assignment is easily accomplished with the input record/select switches located above each input channel. You can record on any or all of the four tracks at once.

The heart of the MT3X is a two-speed cassette transport that employs dual motors, light-touch controls, pitch control, "zero stop" and dbx for noise-free, hiss-free recordings. Yamaha recommends that the higher of

Figure 4. Azimuth alignment was analyzed by recording sweeps on two adjacent tracks and then playing back the results using an azimuth angle test between tracks.

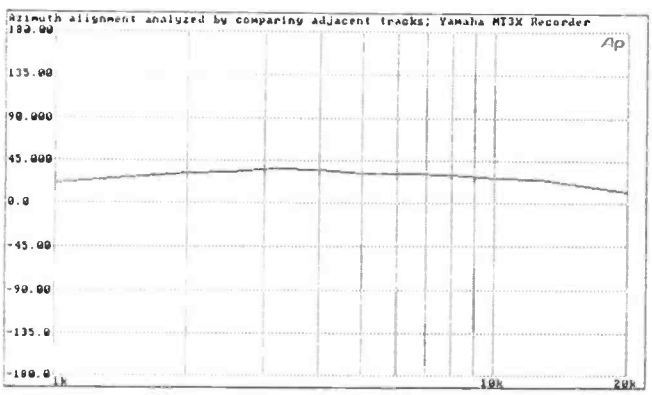
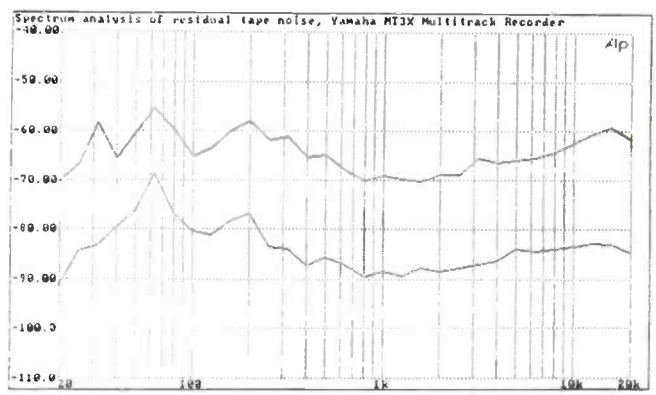


Figure 5. Spectral distribution of tape noise. Upper trace is noise without dbx, lower trace with dbx. Tape speed had a negligible effect on the test results.



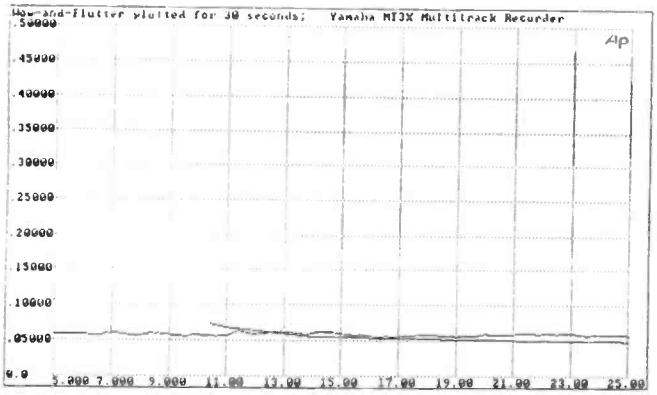


Figure 6. Wow-and-flutter, plotted for a period of thirty seconds, revealed a slight reduction in w-f when the higher speed (9.5 cm./sec.) was used.

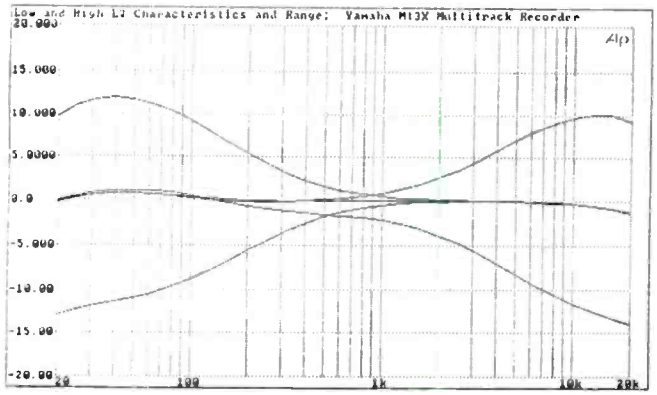


Figure 7. Equalization range and characteristics of the mixer section.

the two speeds (3-¾ in./sec., or 9.5 cm./sec.) be used for making your own master or audition recordings, while the lower, standard 1-⅞ in./sec. (4.75 cm./sec.) speed can be used when it is necessary to play back standard recorded cassettes. The cassette section is also equipped with *Auto Punch-In* capability. You can easily program the recorder to punch in and punch out at specific points in a recording. This allows you to concentrate more on the musicality and artistic consideration of the recording session you are involved in.

The fluorescent display along the upper section of the MT3X is packed with useful information. There is separate stereo and four-track metering, transport status indicators, counter and more. The MT3X provides two sets of left and right output jacks. The stereo output jacks are intended for feeding your mix to a separate mastering recorder, while the left and right monitor output jacks are intended for connection to a separate stereo amplifier for monitoring. An on-board volume control adjusts control room listening level without affecting mix-down levels.

MIDI sequencers and drum machines can be slaved to the MT3X by recording a synchronization code on Track 4. Sync jacks are provided for this purpose. The monitoring capabilities of the MT3X are extremely versatile. You can mix up to six inputs of MIDI gear, two stereo effects and tape tracks simultaneously.

CONTROL LAYOUT

The front lip of the MT3X is equipped with six phone-jack unbalanced input jacks (the first two of which may be used for either microphone or line-level signals), a stereo headphones jack and a *Punch In/Out* footswitch jack. Slider channel faders are on the top surface, in line with the various input jacks. Channels 1 and 2 also have *Mic/Line trim* controls that adjust the sensitivity of the Channel 1 and 2 inputs to accept a wide range of input signal levels—from line to microphone. Channels 1 through 4 are also equipped with *Tape-Off-Mic/Line* (or *Line*) switches that determine whether the input to the corresponding mixer channel is the output from the correspondingly numbered track of the cassette recorder section, or the channel mic/line or line input. The *off* position of these switches turns the particular channel off entirely. Channel 5 and 6 accept line-level input only and are active at all times. Rotary Pan controls associated with each channel input assign the signal from the corresponding mixer channel to any desired position in the stereo sound stage. Slider controls above each pan pot handle low- and high-frequency EQ while *Aux 1* and *Aux 2* controls associated with each channel determine the amount of signal from that channel that is sent to the *Aux Snd 1* or *Aux Snd 2* jack. Those jacks, in turn, can be connected to external signal processors using the aux send/return loops of the MT3X. *Record Select* switches along the upper edge of the sloped front panel assign the signal from each mixer channel either directly to the corresponding track of the recorder or to a different track using the *Pan* controls. A sync switch is nearby, adjacent to the previously-described display area.

The mixer's mastering section incorporates *aux send* (1 and 2) and *aux return* (1 and 2) controls, *monitor* controls, a *headphones select* switch, the *headphones level* control and the master fader control. The area beneath the cassette compartment also houses a dbx on/off switch, the tape speed switch, a pitch control that varies tape speed by about 10 percent in either direction, the *zero stop* button (sometimes referred to as a *memory rewind button*, since when depressed, tape rewinds to the "0000" counter setting), an *auto punch in/out* button, and an *auto punch memo* button that's used to specify the *start* and *end* points for an automatic punch in/out function, and a *counter reset* button. Further down beneath the tape compartment, towards the front of the panel are the usual tape transport buttons.

The rear panel of the MT3X is equipped with four tape out jacks, stereo L and R output jacks, sync in and out jacks, aux send and return jacks, L and R monitor output jacks and a power *on/off* switch. The power on/off switch was wisely placed here so that it would not be inadvertently pushed (thereby shutting down power) in the midst of a recording or mix-down session.

Before commenting on the performance of the MT3X I should point out that Yamaha wisely packed in a short musical tape with each MT3X. The tape consists of four separate tracks, in which a musical group called *The Hot Heads* demonstrate how the unit can be used. The vocalist is recorded on one track while other instrumentalists are on two other tracks. The fourth track (actually track number 1) is reserved for the narration of a very attractive-sounding young lady (dare I say *sexy* sounding in this age of waning male chauvinism?) who "walks you through" the capabilities of the MT3X in a most appeal-

ing manner. Her narration is all the while augmented by the music of the *Hot Heads* in the background, but you can easily tune them out by lowering the monitor controls for tracks 2, 3, and 4. After her very detailed description of the MT3X's capabilities, she invites you to do a mix-down of the next selection recorded by the group, using an external two-track mastering recorder. The whole thing (cassette, music, and narrated explanation) is a nice touch added by the people at Yamaha and one that should make familiarization with the MT3X a lot easier.*

Nevertheless, I would strongly urge users of this interesting piece of equipment to read the well-written 30-page owner's manual carefully. The step-by-step setup and application instructions will save even the most experienced mixer/recordist time and, in some cases, will prevent frustration.

LABORATORY MEASUREMENTS

The *Vital Statistics* chart at the end of this report summarizes all of the measurements made for this unique little "studio in a small package." Most of our measurements concerned the performance of the cassette recorder, since the mixer's functions are essentially those of a pass-through preamplifier whose frequency response, noise and distortion characteristics were bound to be better than those of the tape recorder section. *Figure 1* shows the results of two plots made for the tape recorder section. The curve having the wider bandwidth was made using the 9.5 cm./sec. (3-¾ in./sec.) tape speed while that with the poorer response was made using the standard 1-⅞ in./sec. cassette tape speed. The deck is configured to use only type II (chromium dioxide or other high-bias) tape formulations, and that is what we used to make these tests.

At the higher tape speed, if you apply a ±tolerance of 3 dB, response extended to around 18 kHz, as claimed by Yamaha while low-end response extended down to 20 Hz for the same overall tolerance, as opposed to the 40 Hz limit stated by the manufacturer.

Figure 2 is a plot of harmonic distortion versus frequency, using an input level that caused the level meters on the MT3X to read 0 dB. The upper trace resulted when we used the slower tape speed, while somewhat lower distortion was observed at the higher tape speed, where, over much of the frequency range plotted, THD hovered at or below the 1.0 percent mark. Of course, the distortion readings for any cassette deck are also dependent upon the tape used, so you may expect to get somewhat better or, in some cases, somewhat poorer distortion.

The MT3X recorder is a two-head machine. In other words, a single head serves both for recording and playback. Accordingly, we would have expected excellent azimuth alignment when recording a tape and then playing it back. Our expectations were confirmed, as shown in *Figure 3*.

We recorded a couple of silent tracks on our sample tape in order to measure the signal-to-noise ratio performance of the recorder section of the MT3X. Referred to 0 dB readings on the meters, A-weighted s/n measured 53.9 dB without dbx turned on. Repeating the measurement with dbx active, S/N now measured more than 73 dB, an improvement of nearly 20 dB. *Figure 4* shows two plots in which a spectrum analysis of residual noise versus frequency was made. The upper plot represents noise distribution observed during playback of the tape made without dbx, while the lower trace shows the spectral distribution of noise when dbx was used. Notice that the major noise peaks in both cases occurred at the power line frequency (60 Hz), indicating that the random noise contributed by the tape itself may have been even lower than the values mentioned earlier.

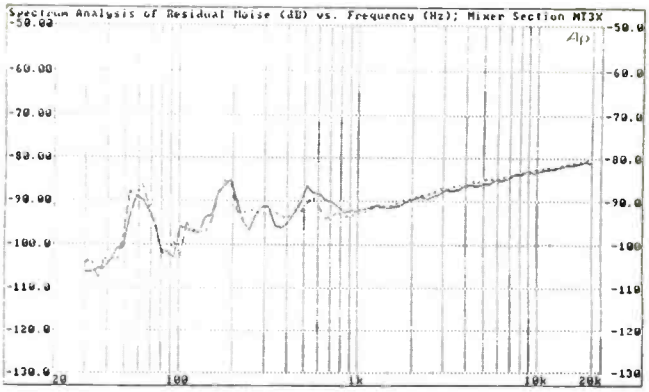


Figure 8. Spectrum analysis of residual noise, referred to 775 mV input and with levels adjusted to produce 0 dB reading on the unit's level meters.

Figure 5 shows two plots of wow-and-flutter, taken over a period of 30 seconds. At the higher tape speed, wow-and-flutter settled into a value of 0.05 percent, exactly as claimed by Yamaha. At the slower tape speed, a very slight increase to around 0.058 percent was observed (represented by the upper trace in *Figure 5*).

We did make a few measurements concerning the channel inputs and mixer circuitry. *Figure 6* shows the maximum boost and cut range of the low- and high-frequency EQ circuits available for each of the six channel inputs. Results are similar to those obtained with ordinary bass and treble tone controls. We also measured the signal-to-noise ratio of the mixer section of the MT3X and obtained a reading of –74.3 dB relative to 0 dB readings on the meter. Yamaha's manual tells us that 0 dB on the meters of this instrument corresponds to 0.775 volts rms or, in other words, 0 dBm. Again, we were curious to see what particular frequencies were contributing to the residual noise and so another spectrum analysis plot was made, this time involving only the input-to-monitor output signal paths. Results are shown in *Figure 7* and, once again, the major contributors seem to be at power-line frequency and at its third harmonic.

CONCLUSIONS

Once we understood the signal paths established in the MT3X (an excellent block diagram is provided in the owner's manual), operating the MT3X was relatively simple and in less than one hour we became thoroughly familiar with its capabilities.

They are, to put it mildly, very impressive. Having such sophisticated features as Automatic Punch In/Out is unusual for a product of this type. The unit is also arranged

* We have learned from Yamaha, after this report was submitted, corrected and typeset, that the tape Len Feldman refers to was sent to him by the company as an aid in evaluation, and is not available with the MT3X.

for easy ping-pong recording, so that if you follow the procedure outlined in the manual it is actually possible to record up to ten individual tracks without re-recording any single track more than once. For less ambitious recording sessions, simple overdubbing is accomplished with ease using this product.

Clearly, the MT3X is either a rehearsal/demo-making system—or can be used as a means for creating innovative and professional-sounding tapes. It is about the most flexible and well thought-out product of its kind that we have seen. Yamaha is, as we all know, no newcomer to the pro-audio field and the engineering and logical layout that they have applied to the MT3X is further proof of that fact. This is one of those products that a reviewer is reluctant to part with—fun to use, and yet capable of providing surprisingly good quality mix-downs for creating two-channel masters on another recorder to which it may be connected. db

VITAL STATISTICS

SPECIFICATION	MFR'S CLAIM	db MEASURED
Tape Transport Section		
Tape Type	Chrome (70 μsec, EQ)	Confirmed
Tape Speed	4.75 or 9.5 cm./sec.	Confirmed
Pitch Control	±10%	+9, –11%
Wow and Flutter	0.05% WRMS	0.05/0.058%
Rewind Time (C60 Tape)	100 second	95 seconds
Mixer Section		
Inputs 1 & 2		
Rated Level	–10 dB to –50 dB	Confirmed
Max. Input Level	+10 dB	Confirmed
Inputs 3 to 6 and AUX Returns 1 & 2		
Rated Level	–10 dB	Confirmed
Min. Input Level	–16 dB	Confirmed
Stereo Out (L&R), Tape Out (1 to 4) and Monitor Out (L&R)		
Rated Output	–10 dB	
Output Impedance	1k ohm	Confirmed
Load Impedance	Greater than 10k ohms	Confirmed
Headphones Out		
Load Impedance	8 to 40 ohms	
Max Output Level	100 mW/40 ohms	Confirmed
Equalizer Range	±12 dB @ 100/10k Hz	Confirmed
Recorder Section		
Freq. Response 4.75 cm/sec.	40 Hz to 12.5 kHz ±3 dB	20 Hz to 6 kHz ±3 dB
9.5 cm./sec.	40 Hz to 18 kHz ±3 dB	20 Hz to 18 kHz, ±3 dB
Distortion	1%, 315 Hz	0.88%, 315 Hz
Channel Separation	55 dB at 1 kHz	58 dB at 1 kHz
Erasure Ratio	70 dB at 1 kHz	Confirmed
S/N Ratio, dbx ON	85 dB	73.5 dB re 0 dB
General Specifications		
Power Requirements	120V AC, 60 Hz, 24 W	
Dimensions (WxHxD, in.)	17-7/16 x 4-3/16 x 12-13/16	
Weight	8 lbs. 10 oz. (3.9 kg.)	
Price:	$995.00	

RANDY HOFFNER

Broadcast Audio

ADVANCED TELEVISION: WHAT'S IN IT FOR AUDIO?

• The reader who has any contact with the television industry is doubtlessly aware of the large amount of current activity devoted to the development and standardization of advanced television production and transmission systems. Advanced television or high-definition is a hot topic in the trade press, and the subject has shown up in the popular press as well recently, largely for non-technical reasons. I am going to keep this article in the technical arena, however. Advanced television (ATV) is a broad term applicable to systems that represent improved-definition television (ITV), extended-definition television (EDTV), and high-definition television (HDTV). That's the alphabet soup.

CLARIFYING THOSE TERMS.

Improved-television refers to improvements to NTSC television which remain within the general parameters of NTSC emission standards. They may be implemented by broadcasters with little or no Federal Communications Commission action, and consist of improvements in such picture qualities as definition, with no change in picture aspect ratio. An example of ITV is the Faroudja system. Extended-definition television refers to improvements such as wide-screen aspect ratio and improved definition that may significantly modify NTSC emissions, but are still compatible with NTSC receivers. An example of EDTV is Advanced Compatible Television 1 (ACTV 1), as proposed by the David Sarnoff Research Center. High-definition television refers to a system that generally accepted guidelines specified as comparable in definition to 35 millimeter film or about double the resolution of present NTSC. All proposed HDTV systems have about twice the number of scanning lines of NTSC and a wider aspect ratio, usually 16:9 rather than 4:3. An example of a true HDTV transmission system is ACTV II.

A discussion of advanced television must be divided into two discrete subtopics: production and transmission. The current NTSC video standard is generally applicable to both production and transmission. In a majority of cases (but not all cases) the same NTSC video waveform is employed both in the production and recording stages and in the system used to broadcast television pictures to the home. There are exceptions to this, of course. Graphics devices, which are really computers, often generate images in RGB (red, green, blue) component form, converting the components to the NTSC composite waveform for use in further production and broadcast. The half-inch professional video recording formats M-II and Betacam, and the Rolls-Royce digital video recording format, D-1, record video in color-difference (Y, R-Y, B-Y) component form. The NTSC composite video signal has a bandwidth of about 4.2MHz, which when combined with an FM audio signal, fits into the U.S. television channel allocation of 6MHz.

GREATER BANDWITH FOR HIGH DEFINITION

In the case of advanced television, on the other hand, a production system that qualifies as "high-definition" as previously defined will have a much greater bandwidth than NTSC. A high-definition video signal will reach or exceed 30 MHz in bandwidth, and this signal obviously cannot be transmitted via a 6MHz terrestrial television channel, or even reasonably by cable or direct broadcast satellite. This produces the requirement for an emission system that transmits a video signal distinctly different from that used in the production system. The transmission system must be capable of emission within the constraints of a limited bandwidth. Thus, the full high-definition production signal must be reduced in bandwidth by some means in order that it may fit into the allotted transmission channel, whether that is a terrestrial broadcast channel, a cable channel, or a dbs channel. There are two categories of systems proposed for the transmission of HDTV signals over terrestrial broadcast systems: augmentation and simulcast. Augmentation implies the transmission of an NTSC receiver-compatible EDTV signal on a 6MHz television channel, accompanied by an augmentation signal transmitted via an additional block of spectrum such as a second television channel. The augmentation signal itself is essentially the difference between the EDTV picture and the HDTV picture. When the two are added in the receiver, the full HDTV picture is recovered. The simulcast approach is the transmission of a separate, non-compatible, full HDTV signal on a separate channel, simultaneously with the NTSC or EDTV signal. There is a general agreement that advanced television production and transmission systems demand improvements in their audio characteristics as well as video. There are a number of considerations involved in audio for ATV. Many issues are common to production and transmission systems, and each area has some unique requirements.

AUDIO ISSUE

In the area of high definition production, the audio issues include: the number of discrete channels necessary, the audio bandwidth limits re-

quired, and the quality level of such distinctions as dynamic range and distortion. The number of channels required depends upon the application of the product. Current production for stereo television often involves 6 channels of source material—two each for dialogue, music, and sound effects. These 6 channels are then mixed down to two-channel stereo.

In addition to television, another proposed application for high-definition video production is the electronic production of cinematic features, which would subsequently be transferred to film for theater projection. If this application materializes, we are told that a minimum of "5.1" perceptually separate channels is appropriate. These channels include three signals behind the screen: left, center, and right, and two rear or surround channels.

The remaining "0.1" channel is the subwoofer signal, and generally includes audio material from about 100Hz down. Fortunately, as audio frequencies get lower, directionality is progressively less perceptible, so a common bass channel derived from a mix of all channels may be employed without perceptual detriment. In addition to theatrical presentation, surround sound will become more important in advanced television because of the larger and wider screens and the subsequent demand for a more realistic sound stage that will result from these systems.

The 5 channels of surround sound are often reduced to 4 channels in the theater when 35 millimeter film is the source material, with the left and right surround channels being reduced to a single, diffused surround signal. In this case, the 4 channels are mixed by a matrix process to yield a two-channel product. This process is called 4-2-4 matrixing, because 4 signals (left, center, right, surround) are matrixed into two discrete stereo tracks which are decoded into the four original signals upon playback. Matrixed encoding is the process that yields 35 mm *Dolby Stereo* in theaters and *Dolby Surround* and other surround sound processes in consumer surround sound decoders. Surround sound encoded with the 4-2-4 matrixing system may be transmitted on normal stereo transmission systems, and in fact, this is done now to a limited degree on television. A production system must have the requisite number of discrete audio channels, however, because editing will require the signals to be in their discrete form. Other considerations regarding number of channels are the possible inclusion of multiple languages and requirements for data or ancillary signals. Proposals for audio for advanced television production have mentioned from three to eight discrete audio channels.

IT WILL BE DIGITAL AUDIO

It is a matter of general agreement that the audio accompanying any advanced television production or transmission format will be digital. It is also agreed by many that audio for an advanced television production system should be of a quality level at least comparable to compact discs, which means frequency response flat 20Hz-20kHz, very low distortion, and a dynamic range approaching or exceeding 90dB. In the professional production world this implies a digital audio word length of at least 16 bits and the professional sample rate of 48kHz. Quantization number and sample rate define the quality level of a digital audio signal. Sixteen bit quantization yields a theoretical maximum dynamic range of 98dB. Allowing 20dB of headroom from operating level up to the "digital full-scale" point, this places the noise floor 78dB below operating level. Those who are involved in the digital audio field will attest that if 16 bits is desired as an end product, a greater digital word length should be used in the production steps to provide adequate headroom for processing. A most desirable method would be to use 18 or 20 bit quantization in the production system, later reduced to 16 bits in the end product.

Specification of audio for an advanced television transmission system involves many of the same considerations as specification of audio to accompany advanced television production systems—with some additional factors to consider. A working party of the FCC Advisory Committee on Advanced Television Systems (ACATS) has demonstrated strong resistance to establishing any minimum number of audio channels or minimum data capacity. That committee agrees that current NTSC audio capabilities must be continued, and that any ATV transmission service should provide new digital audio channels of near compact disc quality, in number at least as great as the number of analog channels provided by the existing MTS system. In addition, captioning and other present vertical blanking interval signals must be preserved. If these criteria are met, an ATV transmission system will have a total of eight audio channels, four analog and four digital. Of course, mono compatibility must be maintained to preserve compatibility with present MTS.

The inclusion of even two channels of digital audio in an advanced television transmission system for use within the present 6MHz television channel will necessarily involve a data compression system to reduce the bandwidth requirement of the digital audio data stream to a manageable level. A 44.1kHz sample rate and 16 bit linear quantization, as in compact discs, will produce a data rate around 1 megabit per second, which is far more data than may be readily sandwiched into an already fully-packed television channel allocation.

There are a number of perceptually-based digital audio data reduction schemes in various stages of development.

One system currently in use is NICAM-3, the source coding scheme used by the British Broadcasting Corporation's digital stereo television transmission system. This system uses a base quantization number of 14 bits, compressed to 11 bits, requiring about 730 kHz of bandwidth for transmission. There are systems currently under development which promise even further data rate reduction, in some cases down to an average quantization number of two bits, which compares favorably in perceptual terms with compact disc performance.

Some of these systems are considerably closer to the affordable and size-efficient hardware stage than others. Many of the advanced television system proponents to be tested by the Advanced Television Test Center (ATTC) are rather vague on the point of "digital audio" but those that do mention a data rate generally specify a 500 kilobit-per-second transmission channel. Such a data capacity demands a rather serious data reduction scheme to yield audio

quality comparable to compact discs.

ADDITIONAL SPECTRUM SPACE

In addition to a source coding system, spectrum space must be found in which to locate a carrier for the digital audio data. A high definition transmission system using the augmentation approach may possibly have the required spectrum space for full compact disc quality audio that is not digitally compressed, but any ATV transmission system that is compatible with current 6 MHz channel is going to present a real challenge in finding a place to put even a data-reduced digital audio signal.

It must be remembered, however, that space was found in the monochrome television signal to sandwich in color, and now space has been found to sandwich in a wider aspect ratio and higher definition. The chances of finding a way to sandwich in digital audio are pretty high.

It's apparent that there are a number of obstacles to overcome before advanced audio for advanced television is a reality. Eventually, of course, they will be overcome, and the consumer may look forward to the many advantages of digital audio to accompany advanced television pictures. db

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

- On the pages that follow, you will find a Guide to speakers, both studio monitors and performance/stage types, each treated separately. The Guide is in chart form and is immediately followed by manufacturers' adressses.
- As usual, be aware that we attemp to contact every manufacturer but not all are prompt or cooperative enough for our necessary deadlines.

MONITOR SPEAKERS

MODEL	DIMENSIONS, HWD	OUTSIDE FINISH	GRILL SCREEN, FINISH	IMPEDANCE, OHMS	FREQUENCY RESPONSE, dB	BASS DRIVER, DIMENSIONS	BASS DRIVER, TYPE	MID DRIVER, DIMENSIONS	MID DRIVER, TYPE	HIGH FREQUENCY DRIVER, DIMENSIONS	HIGH FREQENCY, TYPE	CROSSOVER(S)	WEIGHT, LBS	PRICE	FEATURES
ALTEC LANSING CORPORATION															
9842-8A	30 34 18	gray lacq	black nylon	8	35-20k	12	cone				horn	1.5	60		Vented 2-way studio or broadcast monitor system.
9842-8D	30 34 18	oak venr	black nylon	8	35-20k	12	cone				horn	1.5	60		2-way vented studio or broadcast monitor
9844-8B	30 40 20	gray lacq	black nylon	8	35-20k	12	cone				sect horn	800	80		2-way vented monitor system.
APOGEE SOUND															
AE-1	16 10.25 8	black text paint	gray foam	8	63-19k 4	8	cone			1	horn	4k	18	$420.00	Exceptionally clear reproduction of voice and music in foreground music applications.
AUDIO MEDIA RESEARCH (PEAVEY)															
PRM 312A	25 15.88 13	black	black fabric	8	42-18k 3	12	cone	6.5	cone	1	soft dome	300 3k	58	$439.50	3-way system with rear level control for both the high and midrange drivers. Power handling is 200 watts program.
PRM 310S	21 14 11.25	black	black fabric	8	44-18k 3	10	cone	5	cone	1	soft dome	300 3k	40	$399.50	3-way system with a power handling capacity of 150 watts program.
PRM 308S	17.5 12.25 11.5	black	black fabric	8	45-18k 3	8	cone	5	cone	1	soft dome	300 3k	26	$299.50	3-way system with response mode switch, power handling capacity of 180 watts program.
PRM 205A	8.25 11.5 6.75	black	black fabric	8	79-18k 3	5	cone			1	soft dome	2.5	7	$99.50	2-way system with drivers that are mounted on the same vertical axis. Power handling capacity is 50 watts program.
PRM 208S	9.75 15 8.25	black	black fabric	8	68-18k 3	8	cone			1	soft dome		6	$199.99	2-way system with drivers mounted on the same vertical axis. Power handling capacity is 60 watts program.
ELECTRO-VOICE															
Sentry 100A	17.25 12 11.13	black vinyl	gray cloth	6	45-18k 3	8	cone			1.5	dome	2k	27	$360.00	Incorporates front mounted high-frequency shelving control. Rack mountable.
Sentry 100EL	17.25 12 11.13	black vinyl	gray cloth	6	45-18k 3	8	cone			1.5	dome	2k	33	$750.00	Internal 50 watt MOSFET power amplifier. Rack-mountable.
Sentry 500	23.75 27 13	black vinyl	gray cloth	8	40-18k 3	12	cone			1.5	dome	1.5	70	$714.00	Constant directivity and time-coherent studio monitor.
Sentry 505	19.5 25.6 18.7	black vinyl	gray cloth	8	40-18k 3	12	cone			1.5	dome	1.5k	60	$714.00	Constant directivity and time-coherent studio monitor suited for 1/4 space placement.
MS-802	17.2 12 11.1	oak vinyl		6	45-18k 3	8	cone			1.5	dome	2k	27	$599.90 pair	Rack mountable studio monitor. Incorporates front-mounted high-frequency shelving control.
FOSTEX CORP. OF AMERICA —See our ad on Cover III															
RM800	15 8.5 8.6	black vinyl	black cloth	8	55-25k	6.5	cone			1.25 sq.	RP	7k	14.75	$500.00 pair	Coaxial two-way system. Employs constant directivity RP square horn.
RM900	17 10.25 10.43	black vinyl	black cloth	8	50-25k	8	cone			1.25 sq.	RP	7k	18.75	$788.00 pair	Has high frequency attenuation and field switch.
RM1000	24.5 14.5 11.75	black vinyl	black cloth	8	40-25k	8	cone			2.0 sq.	RP	7k	49	$1000.00 pair	Double spyder woofer. Binding posts for inputs.
JBL PROFESSIONAL															
Control 1	9.25 6.25 5.63	charc gray paint	black metal	4	120-20 3	5.3	cone			.75	dome	6k	4	$200.00	Shielded magnet structure, optional mounting hardware. Miniature system, Control Series family.
Control 5	15.25 9.88 9	charc gray paint	black metal	4	75-20k 3	6.5	cone			1	dome	3k	10	$335.00	
4425	16 25 12.25	oiled walnut	cloth	8	40-16k 3	12	cone			.88	bi-rad horn	1.2	57	$950.00	Smooth, accurate response, flat power response, 200 watts cont. program capacity.
4435	35.75 38 17.13	oiled walnut	cloth	8	30-16k 3	15 (2)	cone			1	bi-rad horn	1k	250	$2,095.00	Stable stereo imaging that remains fixed over a wide range of horizontal positions.
4406	15.38 9.38 8.5	oiled walnut	cloth	8	55-20k 2	6	cone			1	Ti dome	3k	17	$225.00	Ideal for console or close-in listening. Deep, powerful bass.
4408	17.25 12 11.63	oiled walnut	cloth	8	50-20k 2	8	cone			1	Ti dome	2.5	26	$275.00	Ideal for broadcast and general monitoring applications.
4410	23.5 14.25 11.25	oiled walnut	cloth	8	45-20k 2	10	cone	5	cone			800 4.5	43	$425.00	Vertical line array that gives maximum special detail at greater listening distances.
4412	14.25 23.5 11.25	oiled walnut	cloth	8	45-20k 2	12	cone	5	cone			800 4.5	47	$650.00	Tight cluster arrangement and superior power efficiency.

MODEL	DIMENSIONS, HWD	OUTSIDE FINISH	GRILL SCREEN, FINISH	IMPEDANCE, OHMS	FREQUENCY RESPONSE, dB	BASS DRIVER, DIMENSIONS	BASS DRIVER, TYPE	MID DRIVER, DIMENSIONS	MID DRIVER, TYPE	HIGH FREQUENCY DRIVER, DIMENSIONS	HIGH FREQENCY, TYPE	CROSSOVER(S)	WEIGHT, LBS	PRICE	FEATURES
KLARK-TEKNIK															
Jade One MKII	21.5 15.5 10	black text epoxy	no grille	20k 10k	55-17k 3	8	cone neoflex			1.25	soft dome	2.5	48.5	$2,295.00	Optimum phase linearity, creating an accuarate stable soundfield for mixing, mastering, etc.
KLEIN & HUMMEL (GOTHAM AUDIO CORP.)															
098	15 10 7.25	gray	black cloth	4.7	50-16k 2.5	8	cone	1.5	dome	.75	dome	800 6k	26.4	$1295.00	Active, tri-amplified, 2 electronic X-overs with location dependent equalizer. (amps:1-100W, 2-50W).
096	20.7 12.4 11.4	gray	black cloth	4.7	50-20k 2.5	10	cone	2	dome	.75	dome	600 4k	48.4	$2,270.00	Active, tri-amplified (3-60W), 2 electronic X-overs with location dependent equalizer.
092	31.5 17.3 11.8	gray	black cloth	4.7	50-17k 2.5	10 (2)	cone metal	3.5	cone	1	dome	500 3k	66	$4,195.00	Active, tri-amplified (1-120W, 2-60 W), 2 electronic X-overs with location dependent equalizer.
KLIPSCH AND ASSOCIATES															
KSM	16.0 19.5 22.0	text black	black grill cloth	8	35-15k 5.0	12	reflex			2.75 8.75	horn	1.5k	48	$435.00	
KSM-1	18.0 text 20.0 22.0	black	black grill cloth	8	50-15k 5.0	15	reflex			3.25 10.25	horn	1.5k	59	$529.00	
KSM-2	24.0 25.5 20.0	text black	black grill cloth	8	50-15k 5.0	15	reflex			3.25 10.25	horn	1.5k	75	$549.00	
MEYER SOUND LABORATORIES INC.															
833	32 20 19.75	satin black lamin	foam	8	35-18k 4	15	cone				rad horn		115	$6,300.00	Studio reference monitor system includes electronic control unit.
834	38.5 24.13 20.13	satin black lamin	foam	8	30-100	18	cone						127	$1,600.00	130 dB SPL. Used as a subwoofer with the 833 monitors.
PANASONIC RAMSA															
WS-A10	8 12.5 8	black resin	black metal	8	80-16k	6	full-range cone						6	$120.00	Compact full-range, near-field monitor. Magnetically shielded. Accessories opt.
WS-A10	6.1 6.1 9.8	white black mold	white black metal	6	80-16k 5	4.7	cone						5.7	$120.00	
PROFESSIONAL AUDIO SYSTEMS															
SM-1	25 19.5 18.5	oak black	black cloth	8	37-20k 3	15	cone			1	horn	1.2	90	$4,600.00	Studio monitor with time offset correction.
SM-2	36 36 24.5	black		4.8	25-20k 3	15	cone	15	cone	2	horn		190	$10,500.	3-way studio monitor with time offset correction.
STUDER REVOX AMERICA INC. —See our ad on page 30															
2706	12 15.1 13.5	dark gray nextel	black anod alum	4	42-20k 3	12	cone	2	dome	1	dome	720 5k	48	$825.00	Compact. Can be installed on floor stands or suspension brackets. High power dome transducers.
A723	22.9 12.7 16	wal-nut venr	black cloth		40-20k 3	12	cone	5	cone	1	dome	300 27k	68	$1850.00	Each crossover has dedicated 100w amp. Useful in small to medium sized listening environments.
TANNOY NORTH AMERICA INC.															
PBM 8	12 8 8.5	vinyl	black material	8	57-20k	65	poly			.75	dome	2.4	18.5 pair	$338.00	Small size. Ideal for portable desktop reference monitor.
PBM 8	15 10.13 10.63	pewter gray vinyl	black material	8	47-20k	8	poly			1	dome	2.4	18.5	$478.00 pair	Deep extended low frequency. Ideal for acoustic and classical music.
AUM-TP1 10	15 ash 8.5	black material vinyl	black	8	55-25k	6.5	poly			1	dome	3k	12	$548.00 pair	DMT -("Differential Material Technology"). Fully shelided. Ideal for broadcast and audio for video.
NFM8-DMT	18.5 10 8.5	Nextel grey	black material	8	48-25	8	poly		dual conc	1	comp horn	2.3k	25	$998.00	Near-field reference monitor for pairphase coherent mixing. Ideal for portable reference.
LGM 12B	23 16 11	walnut venr	black material	8	55-20k	12 conc	paper		dual horn	2	comp	1.2k	52	$1,998.00 pair	Point source primary mix monitor for portable reference.
SCM 10B	20.5 14.5 8.5	walnut veneer	black material	8	55-20	10	paper	dual conc	2	comp horn	1.2k	40		$1798.00 pair	same as above.
TASCAM (TEAC CORP. OF AMERICA) —See our ads on page 22-25															
S1010 M		black metal	black metal	25k	100-18	4	cone						5	$300.00	Reference monitor system with built-in amplifier.
CM-4	9.2 5.8 5.6	black metal	black metal	8	80-20k	4	cone			1	dome	2.5	5	$125.00	2-way reference monitor speaker system.

MODEL	DIMENSIONS, HWD	OUTSIDE FINISH	GRILL SCREEN, FINISH	IMPEDANCE, OHMS	FREQUENCY RESPONSE, dB	BASS DRIVER, DIMENSIONS	BASS DRIVER, TYPE	MID DRIVER, DIMENSIONS	MID DRIVER, TYPE	HIGH FREQUENCY DRIVER, DIMENSIONS	HIGH FREQENCY, TYPE	CROSSOVER(S)	WEIGHT, LBS	PRICE	FEATURES
TOA ELECTRONICS															
22-ME	6.7	gray	black	8	100-17k5	cone							5.1	$74.50	Near-field cube. Audio/video sheilded
AV	6.7	poly	jersey		3										and mountable.
	5.8														
265-	13.8	gray	black	8	60-20k	6.3	cone			1.2	dome	3k	11.5	$194.50	120-degree even dispersion,near-field,
ME-AV	8.1	poly	jersey		3										A/V shielded, mountable, symetrical
	9.6														component placement.
280-	15.7	gray	black	8		8	cone	1.2	dome	0.79	dome	1.5k	15.4	$249.50	Tweeter level control. 120-degrees
ME	9.3	poly	jersey									14k			even dispersion. A/V shielded,mount-
	9.3														able.Symetrical component placement.
312-	22.88	gray	black	8		11	cone	4.7	cone	1.2	dome	500	35.7	$419.50	Tweeter and mid-range level controls.
ME-AV	13.2	poly	jersey									5k			120-degrees even dispersion. A/V
	11.6														shielded. Symetrical component place.
TRANSDUCER DEVELOPMENTS LIMITED —See our ad on page 2															
Ref	20	black	black	8	16-35k	9.5	cone	4.5	cone	1	dome	200	154	$6,995.00	Transmission line design. Low freq.
Stnd	22	wood	brown		6.5			(2)				3.5			separately terminated.
TLS	48	walnut	cloth									13k			
Mon-	18	black	black	8	18-35k	9.5	cone	4.5	cone	1	dome	300	103	$4,595.00	Transmission line design. Low freq.
itor	12	wood	brown		6.5							13k			separately terminated.
TLS	47	walnut	cloth												
Stu-	16	black	black	8	20-20k	8	cone			1	dome	3k	59	$1,995.00	Transmission line design, featuring
dio	12	wood	brown			(2)					fluid				magnesium alloy suspended dome with sep-
TLS	39	walnut	cloth												arate chamber.
Stu-	15	black	black	8	24-20k	8	cone			1	dome	3k	48	$1,595.00	Same design and high frequency compo-
dio	14	wood	brown												nent as the Studio-3.
TLS	35	walnut	cloth												
Super	14	black	black	8	28-20k	8	cone			1	dome	3k	26.5	$1,095.00	Reflex design.
Com-	11	wood	brown												
pact	20	walnut	cloth												
Com-	12	black	black	8	30-20k	6	cone			1	dome	3k	22	$935.00	Reflex design.
pact	9	wood	brown												
	18	walnut	cloth												
UREI-JBL PROFESSIONAL															
809	23	flat	opt.	8	50-17k	12	cone			.88	comp		60	$750.00	Time aligned, flat power response main-
	16.5	black	grille		3						driv				talning freq. bal. at low distortion
	13.5	paint													over wide range output levels.
811C	20.75	flat	opt.	8	70-17k	15	cone			1	comp		110	$1,590.00	Minimizes listening fatigue, accurate
	26.5	black	grille		3						driv				stereo imaging.
	19	paint													
813C	36	flat	opt.	8	50-17k	15	cone	15	cone	1	comp		198	$2,290.00	Minimizes listening fatigue, accurate
	31	black	grille		3						driv				stereo imaging.
	23	paint													
815C	32	flat	opt.	8	40-17k	15	cone	15	cone	1	comp		260	$2,990.00	Minimizes listening fatigue, accurate
	43.5	black	grille		3	(2)					driv				stereo imaging.
	21	paint													
YAMAHA CORPORATION OF AMERICA —See our ads on page 14-15															
NS10	8.5	black	black	8	60-20k	7	cone			2.4	soft	2k	13.2	$237.00	Industry-standard studio close-field
MS	15	wood	cloth								dome				monitors.NS10MC is commercial version.
	7.75														
NS40	23.5	black	black	8	30-20k	7	cone	2.4	soft	1.2	soft	1.2	37.4	$465.00	Bigger version of the NS10MS with great-
M	11.5	wood	cloth			(2)			dome		dome	5k			er low-end response.
	12														
S10X	6.1	black	black	8	65-20k	4	carb						6.2	$145.00	Very compact wide range system. Handles
	9.5		metal				fiber								up to 150 watts of program material.
	6.5														
S20X	7.5	black	black	8	65-20k	4	carb						4.6	$210.00	Compact dual-driver wide range system.
	11.63		metal			(2)	fiber								Handles up to 300 watts of program
	7.75														material.

PERFORMANCE SPEAKERS

ALTEC LANSING															
937	30	black	black	8	70-15k	12	cone			16x9	horn	3k	45		Multi-purpose, vented, two-way moni-
	23	text	nylon								const				tor loudspeaker system.
	20.7										direc				
9872	30	unfin	neut	8	80-20k	12	cone			18x9	horn	2k	42		Two-way, vented, full-range.
8A	23	birch	brown								const				
	14										direc				
9872	30	tan	neut	8	20-20k	12	cone			16x9	horn	2k	42		Two-way, vented, full-range.
8F	23	text	brown								const				
	14										direc				
9812	34	gray	black	5?	60-13k	15	cone			30x	horn	500	80.1		Two-way, vented, full-range.
8A	23	text	nylon							17	const				
	22										direc				
APOGEE SOUND															
3X3	45	black	gray	8L	53-19k	15	cone	2	horn	28	horn	1k	250	$4,400.00	Extremely high power, wide range,
	29	text	foam	8M	4							7.6k			fully horn-loaded, trapezoid design
	30	paint		16H											will array for concert reiforcement

MODEL	DIMENSIONS, HWD	OUTSIDE FINISH	GRILL SCREEN, FINISH	IMPEDANCE, OHMS	FREQUENCY RESPONSE, dB	BASS DRIVER, DIMENSIONS	BASS DRIVER, TYPE	MID DRIVER, DIMENSIONS	MID DRIVER, TYPE	HIGH FREQUENCY DRIVER, DIMENSIONS	HIGH FREQENCY, TYPE	CROSSOVER(S)	WEIGHT, LBS	PRICE	FEATURES
AE-2	10.5 32 10	black text paint	gray foam	16	63-19k	8	cone			3x5	horn	4k	30	$1,095.00	165 degree horizontal dispersion enables underbalcony applications or in facilities with low ceilings.
AE-3	16 12.5	black text paint	gray foam	8	70-18k 3	10	cone			3x5	horn	4k	40	$750.00	Tremendous power/size ratio in vocal range. Works well with subs for club and disco installations.
AE-4	23 14 14	black text paint	gray foam	8	55-16k 4	12	cone		horn	5x 13.5	horn	1k	65	$1,495.00	High power, full-range, single-amped speaker system. Ideal for churches, auditoriums, PA systems.
AE-5	23 14 16	black text paint	gray foam	8	53-17k 4	12	cone			5x 13.5	horn	1k	78	$2,150.00	Exceptionally clean, natural high frequency response. Trapezoid design enables multiple coupling.
AE-6	14 23 15.5	black text paint	gray foam	8	53-17k	12	cone			5x 13.5	horn	1k	78	$2,170.00	High power stage monitor. Multiple angle design, polar response for optimum stage coverage.
AE-10	22.5 32 24	black text paint	gray foam	8	36-120	15 (2)	cone					100	138	$1,785.00	Powerful bass reproduction from compact enclosure for small club applications.
AE-12	30 44.8 22.5	black text paint	gray foam	8	35-120 4	18 (2)	cone					100	60	$2,110.00	Powerful bass response for large clubs and concert venues. Built-in rigging points.
ARX SYSTEMS —See our ad on page 2															
1812	40 40 22.7	poly paint	acous foam	8 16	40-20k	18 (2)	reflex (2)	12	dome	2	horn	250 2k	260		One-box concert speaker in trapezoid cabinet. Comes with flying points. Optional dolly set.
212	34 20 23.5	poly paint	acous foam	8 16	200- 20k			12	dome	2	horn	2k	100	$2,699.00	
912	46 22	poly	acous 20	8 paint	40- foam	15	bass 18k	12	dome horn	1	horn	250 3k	132	$3100.00	Offers the amplifier efficiency of a 2-way system with improved frequency response of 3-way system.
KA118	22.5 40	poly paint	acous foam	8	40- 200	18	bass horn					80 250	195	$2299.00	Extends bass response of 912 system; or use as full range system. 4.5
KA115	37 22	poly paint	acous foam	8	35- 100	12	heavy duty				100	94			To be used with 303X where extended frequency response is needed.
306	22.4 15 15	poly paint	acous foam	8	50- 20k	12	cone			1	CD flare	2k	50	$1299.00	Operates at continuous high SPL. Has three angles of profile.
303X	24 15 15	poly paint	acous foam	8	50- 20k	12	cone			1	CD flare	2k	50	$1299.00	Compact 2-way speaker suited for convention centres, nightclubs, theatres,etc.
CARVIN CORPORATION															
973	30.5 22 15.	grey Ozite	black metal	8	50-19k 3	15	cone	6.5	cone	3.5 4.5	horn	400 4k	85	$369.00	Full-range 3-way speaker system. Handles 400w.
993	47 25 19.25	grey Ozite	black metal	4	45-19k 3	15 (2)	cone	6.5 (2)	cone	3.5 4.5 (2)	horn	400 4k	146	$629.00	3-way system will handle 800w of power. Projects sound accurately at full power.
852	26 20 14	grey Ozite	black metal	8	65-17k	15	cone			6 16.5	horn	2k	58	$299.00	Has large 45/90 degree exponentiai horn. Deep bass response. For medium size club use.
962	31 24 15	grey Ozite	black metal	8	65-17k	15	cone			6 16.5	horn	2k	90	$349.00	Designed for maximum projection. Tuned horn-loaded and port-loaded. Handles 400w continuous power.
1331 s	32 28 24	grey Ozite	black metal	8	60-3k	15	cone					2k	130	$349.00	Horn and port loaded woofer speaker will handle 400 w continuous power.
1210	23 28 24	grey Ozite	black metai	4	100- 3k			10 (2)	cone			150 2k	85	$299.00	Midrange speaker system for use with a tri-amped speaker array
3018	24 42 24	grey Ozite	none	8	60- 500	18	cone					150	145	$449.00	Folded horn 18in. bass speaker. Used for high level sound reinforcement. Strong subsonics.
752	22 18 13	grey Ozite	black metal	8	80- 17k	12	cone			3 10	horn	2k	55	$239.00	2-way monitor will handle 200w. Excellent vocal monitor.
CELESTION INDUSTRIES,INC.															
SR- Com- pact	7 9 7	grey	black mesh	8	80- 20k 3	1x 5	dual conc						7	$199.00	
SR- 3	10 13 9	grey	black mesh	8	60- 20k 3	1x 8	dual conc						20	$330.00	Requires SRC3 controller for proper operation.
SR-1	12 22	grey	black mesh	8	50- 20k	2x 8	dual conc						20	$545.00	Requires SRC1 controller for proper operation.14

MODEL	DIMENSIONS, HWD	OUTSIDE FINISH	GRILL SCREEN, FINISH	IMPEDANCE, OHMS	FREQUENCY RESPONSE, dB	BASS DRIVER, DIMENSIONS	BASS DRIVER, TYPE	MID DRIVER, DIMENSIONS	MID DRIVER, TYPE	HIGH FREQUENCY DRIVER, DIMENSIONS	HIGH FREQENCY, TYPE	CROSSOVER(S)	WEIGHT, LBS	PRICE	FEATURES
COMMUNITY LIGHT AND SOUND —See our ad on page 27															
CS70	26.7 33.5 18	black carpt	black metal	4	45- 18k	12	ferr cone	2	comp drvr	1	horn	100	135	$979.00	3-way passive, full range speaker.
CS52 black	38.5 black 18	4 carpt	40- metal	15	ferr 20k	6.5	ferr cone		horn cone	500		82 5k		$598.00	3-way full range passive speaker.
CS50 black	33.5 black 26.7 18	8 carpt	35- metal	18	ferr 500		cone					100		$599.00	Compact subwoofer. Internal 150Hz crossover with dual high pass outputs.
CS45 3k	33.5 80 18	black carpt	black metal	4	50- 18k	12	ferr cone				horn			$545.00	2-way passive full range system with HF section. 18
CS38M	17.5 17.5 25	black carpt	black metal	8	60- 18k	15	ferr cone			15.5 x8	horn duct	25k	11	$399.00	Wedge stage monitor.
CS35	23.7 17.5 13.5	black carpt	black metal	8	60- 18k	15	ferr cone			15.5 x8	horn duct	25k	13	$349.00	2-way full range speaker.
CS28M	14 15 22.5	black carpt	black metal	8	70- 18k	12	cone			2x 16	horn	3k	29	$327.00	2-way wedge monitor.
CS25	18.5 15 13.5	black carpeti	black metal	8	70- 18k	12				2x 6	exp. horn	3k	32	$295.00	2-way full-range system.
ELECTRO-VOICE															
DML- 1122 A	23 14.6 14	black text paint	black steel w/foam	8	67-20k 3	12	cone			2	horn	1.3k	68	$1,860.00	High output electronically controlled system featuring high excursion components. 2-way system.
DML- 1152 A	29.88 18 16.28	black text paint	black steel w/foam	8	50-20k 3	15	cone			2	horn	1.1k	96	$2,184.00	High power wide range 2-way electronically controlled system prevents over-heating/over excursion.
DML- 2181	36 22.5 29.88	black text paint	black steel w/foam	8, 8	36-100 3	18	cone					100	164	$2,328.00	Electronically controlled subwoofer utilizing Manifold technology, high excursion drivers.
SH18- 10ER	49 25.8 25.8	black carpet	black steel	8	46- 20k 3	18	cone	10	horn		horn	250 2.5k	163	$2,249.60	Subscoop enclosure with reverse mounted ProLine 18B.DI10X mid and DH3 tweeters.
SH150 2ER	31.9 24.7 16	black carpet	black steel	8	62- 20k 3	15	cone				horn	1.6k	81	$629.90	Horn loaded. New EVG-15 woofer and 2010A tweeter.
S-120 black 2ER	25.5 black 19.5 12	carpet	steel	8	75- 20k	12	cone				horn	1.6k	66	$709.90	Proline 12S and 2010A on HT94
FM12 black 02ER	19.5 black 19.5 24.8	8 carpet	75- steel	12	come 20k 3						horn	1.6k	65	$669.90	Floor monitor incoporates Proline 12S and 2010A on HT94 horn.
S-200	24 15 8.5	black plastic	grey cloth steel	8	50- 18k 3	12	cone				supr dome	2k	36	$1250.00 systm	Pair of speakers and stereo rack-mountable equalizers.
FORMULA AUDIO INC.															
DR- 4401	24 14.5 13	gray Ozite	black metal foam	16H 8M	63-16k 3	12	dir rad			1	horn comp drivr	1.2	55	$1,300.00	Flying hardware standard, trapezoidal, 310 watts RMS.
DR- 4402	36 19.5 22	gray Ozite	black metal foam	8H 8L	43-20k 3	15	dir rad	2	horn	1	slot twt	800 7k	100	$2,100.00	Flying hardware standard, trapezoidal, 370 watts RMS.
DR- 4403	36 36 22	gray Ozite	black metal foam	4H 4L	43-20k 3	15 (2)	dir rad	2 (2)	horn		slot (2)	800 7k	215	$3,500.00	Flying hardware standard, trapezoidal, 740 watts RMS.
DR- 4428	45 22.5 30	gray Ozite	black metal foam	8	30-250 3	18 (2)	dir rad					80	156	$1,800.00	130 dB continuous, 133 peak, 1000 watts RMS.
SM22	15.5 22.7 17.7	black text poly	black metal foam	16H 8B	65-16k 3	12	dir rad			2	horn comp drvr	1.2k	55	$1,400.00	
SM25	15.5 25.7 18.5	black text poly	black metal foam	16H 8B	63-16k 3	15	dir rad			2	horn comp drvr	1.2k	65	$1,600.00	
PLUS 2HM	43.2 22.5 26	black text poly	black metal foam	8M 16H	160- 20k 3			12	horn comp drvr	2	horn comp drvr	1.6k	156	$3,600.00	Fully horn loaded system. Will exceed levels of 138 dB.Can be flown in tight cluster.
PLUS 30B	43.2 22.5 26	black text poly	black metal foam	8B	30- 200	15 (2)	horn					160	138	$2,800.00	Fully horn loaded system. exceed levels of 138dB. Can be flown in tight cluster.
RAM- 300	54 22.5 22.5	black text poly	black metal foam	8B 8M 16H	30- 20k 3	15 (2)	dir rad	8 (2)	dir rad	2	horn	250 1.5k	180	$2,800.00	RAM (Rotating Acoustic Module) can rotate 90 degrees. Handles 130dB. SPL.

MODEL	DIMENSIONS, HWD	OUTSIDE FINISH	GRILL SCREEN, FINISH	IMPEDANCE, OHMS	FREQUENCY RESPONSE, dB	BASS DRIVER, DIMENSIONS	BASS DRIVER, TYPE	MID DRIVER, DIMENSIONS	MID DRIVER, TYPE	HIGH FREQUENCY DRIVER, DIMENSIONS	HIGH FREQENCY, TYPE	CROSSOVER(S)	WEIGHT, LBS	PRICE	FEATURES
FOSTEX CORP. OF AMERICA —See our ad on Cover III															
SP11	7 14 8.4	black	black metal	8	60-18k	4 3 (2)	cone						12	$270.00	Built-in power amplifier.
SPA11	7 14 8.4	black plas	black metal	8	60-18k	4 3	cone						17	$399.99	
GAUSS															
3588				8	40-18k	15	cone			2	comp	1.2k	25	$990.00	Coaxial raw loudspeakerhandles 200watts RMS. Sensitivity of 96dB. For studio and live use.
3288				8	40-18k	12	cone			2	comp	2k	24	$925.00	Coaxial raw loudspeaker handles 200watts RMS.Sensitivity of 91dB For studio and live use.
3285				8	70-15k	12	cone			2	comp	2k	24	$925.00	Coaxial raw loudspeaker handles200watts RMS.Sensitivity of 99dB. Flush-mountable in ceilings.
HARTKE SYSTEMS —See our ad on page 26															
HS401015 B	black 24 24	black carpt	8 metal	30-4k	10	alum cone							87	$999.00	Drivers exhibit quick transient response.
HS11SB15	black 24 24	black carpt	8 metal	30- 2.5k	15	alum cone							66	$799.00	Drivers exhibit quick transient response.
HS412612	black 28 28	black carpet	8 metal	40- 5k	12	alum cone							90	$1150.00	same as above
HS210	12 24 14	black carpet	black metal	8	50- 4k	10	alum cone						45	$699.00	same as above
JBL PROFESSIONAL															
4612B	18.5 21.5 10.25	multi lamin hdwood	cloth	8	60-21k	8 (2)	cone			3	ring bi- radial	3k	45	$665.00	Wide, tightly controlled dispersion, extended freq. res.,high power, high efficiency.
G-734	26.25 20.25 12.75	gray tolex	black metal	8	50-17k 10	15	cone			.88	horn comp driv	1.5	53	$555.00	2-way portable system designed to deliver wide bandwidth and high SPL
G-791	5 15.25 9.75			8	1.5-17 10							1.5	10	$260.00	Contains horn, driver and X-over for musicians who want to custom load components in own cabinets.
G-730	21.75 18.25 11.75	tolex	black metal	8	70-17k 10	12	cone			0.88	horn comp driv	2k	45	$495.00	Compact, 2-way system ,high yield acoustic output from small/portable package.
G-731	16 18.25 21.75	tolex	black metal	8	70-17k 10	12	cone			0.88	horn comp driv	2k	47	$525.00	Lightweight, 2-way floor monitor designed for high acoustic output from small/portable package.
G-732	33.25 24.75 17.75	tolex	black metal	8	45-17k 10	15	cone			0.88	horn comp driv	1.5	77	$660.00	High efficiency system utilizing a G-135A-8 low freq. driver mounted in a horn enclosure.
G-733	20.25 26.25 12.25	tolex	black metal	8	50-17k 10	10	cone	8	cone			3k	52	$595.00	3-way keyboard system.
4628B	30.19 20.13 18.8	multi lamin hdwood	cloth	8	35-21k	15	cone	8	cone	3	ring rad	800 3k	108	$950.00	Specially designed for synth, piano and organ with superb bottom end for clean pedal tones.
JOE'S SOUND & SALAMI CO.															
18UL	24 21 16	black pebble	metal black		35-1k	18	cone						35	$475.00	Ultra-light empty cabinet.
D18 PRO	48 21 16	black pebble	metal black		35-1k	18	cone						100	$899.00	Perfect for sub-woofer applications.
PRO 15	24 21 18	black pebble	metal black		45-2k	15	cone						50	$399.00	High output empty cabinet. Made of wood/chemical laminate.
RL 15M	24 21 16	black pebble	metal black	6	45-15k 3	15	cone			4x 10	horn	2k	50	$449.00	High volume monitor where low frequency punch is desired.
RL	21 12 16	black pebble	metal black	6	75-15k 3	12	cone			4x 10	horn	2k	38	$399.00	High output speaker for space restricted applications. Ideal for portable PA and discos.
RL15	24 21 16	black pebble	metal black	6	45-15k 3	12	cone			4x 10	horn	2k	50	$449.00	Stand-mounted speaker with good low-frequency punch. Designed for DJ installations.
RL 12M	21 16 16	black pebble	metal black	6	75-15k	12	cone			4x 10	horn	2k	38		High output from a small package.
410- CBB	26 24 14	black pebble	metal black		75-2k	10	cone						45	$399.00	Designed for bass players looking punch. Smooth transients.
15CBB	24 21 16	black pebble	metal black		45-2k	15	cone						35	$330.00	High output, cost effective unit.

MODEL	DIMENSIONS, HWD	OUTSIDE FINISH	GRILL SCREEN, FINISH	IMPEDANCE, OHMS	FREQUENCY RESPONSE, dB	BASS DRIVER, DIMENSIONS	BASS DRIVER, TYPE	MID DRIVER, DIMENSIONS	MID DRIVER, TYPE	HIGH FREQUENCY DRIVER, DIMENSIONS	HIGH FREQENCY, TYPE	CROSSOVER(S)	WEIGHT, LBS	PRICE	FEATURES
KLIPSCH & ASSOCIATES															
KP-201	28.6 16.3 13.5	text black paint	black grill cloth	8 4	50-20k	12	seal encl	2.7x 8.7	horn	4.5x 2	horn	700 7k	38	$425.00	Sensitivity 96dB (2.83v)
KP-250	22.6 16.6 13.1	text black paint	black grill cloth	8 4	65-20k	12	bass reflx	2.7x 8.7	horn	4.5x 2	horn	700 7k	49	$592.00	Sensitivity 101dB (2.83v)
KP-301	32.6 20.8 16.8	text black paint	black grill cloth	8 4	45-20k	15	bass reflx	3.3x 10.3	horn	4.5x 2	horn	650 7k	82	$759.00	Sensitivity 101dB (2.83v)
KP-450	55.7 27.7 18	text black paint	black grill cloth	8 4	45-20k	15 (2)	bass reflx			8.5x 21	horn	650	190	$1748.00	Sensitivity 104dB (2.83v)
LS1	35.1 24 24.7	text black paint		8 5	55-17k	15 (2)	horn	5.7x 16.7	horn	1.7x 4.3	horn	400 6k	151	$1071.00	Sensitivity 104dB (2.83)
MARTIN AMERICA															
YRS 1000	22.5 51 26	black	black metal	8B 8M 16H		18						220 1.5	238		3-way all horn loaded. Vertical format. One box system.
F2	22.5 42 25	black	black metal												2222lbs is unloaded weight.Modular system can accept variety of horn and driver combinations.
F2 Bass	22.5 42 36.5	black	black metal	8	15								235		
MEYER SOUND LABORATORIES INC.															
UPM-1	6.75 18.5 7.5	black text	exp metal	16	60-20k 4	5	cone	5	cone	2x5	horn		17	$893.00	Ultra-compact and light weight, efficient high power.
UM-1A	14 22.5	black	metal foam	8,	60-16k	12	cone				CD horn		66	$2,390.00	Efficient high power, ultra-low distortion, flat response, requires M-1A control electronics.
UPA-1A	14.5 22.63 14.5	black text	metal w/gray foam	8, 8	60-16k 4	12	cone				rad horn		66	$2,490.00	Efficient high power, ultra-low distortion, compact and versatile.
MSL-3	31.25 56.75 30	black text	metal w/gray foam	8, 4	75-20k 4	12 (2)	cone				rad horn		265	$4,820.00	High power, high clarity and coherence, arrayable, requires electronic control unit (M-3T).
USW-1		black text	metal w/gray foam	4	40-100 4	15 (2)	cone						115	$1,785.00	Sub-woofer, high power, low distortion.
650-R2	45 30 22.5	black text	metal w/gray foam	4	30-100 4	18 (2)	cone						180	$2,215.00	Sub-woofer, high power, low distortion.
UPA 1A black	22.3 metal 14.5	text	60-16k with foam	12	cone 4						horn		1.6k6	$2,390.00	Compact, high power, arrayable.
14.5 833	32 20 14.7	waln or black	acous trans mater	8	35-18k 3	15	cone				horn		115	$6,300.00	High continuous output. Low distortion.
834	38.1 24.1 20.1	black satin	acous trans mater	8	30-100 4	18	cone						127	$1600.00	Highoutput subwoofer for 833.
PANASONIC RAMSA															
WS-A70	13 21 12	black or wht paint	black or wht cloth	8	50-18k	8	cone			120x 120	horn	2k	14	$240.00	Power capacity is 80 watts. Magnetically shielded, variety of mounting options.
WS-A80	14 21.5 12	black or wht resin	black or wht metal	8	65-18k	8	cone			60x 40	horn	2.5	16	$280.00	High SPL, compact, stackable, variety of mounting options.
WS-A200	28 20 16	black or wht resin	black or wht metal	8	70-20k	12	cone			60x 40	horn	2.5	35	$590.00	125 watt power handling, compact, high SPL, stackable, variety of mounting options.
WS-A240	28 20 16	black resin	black grille	8	35-X-over	12	cone						35	$510.00	Subwoofer system requires model WS-SP2A electronic crossover.
WS-A500	22 10.7 16	gray or white	metal grille	8	100-20k			12	cone		horn	1.5k40		$1200.00	A mid/high system for use with two WS-A550 low-frequency systems and WS-SP2A crossover.
WS-A550	22 10.7 16	gray or white	metal grille	8	35-X-over	12	cone						35	$680.00	

MODEL	DIMENSIONS, HWD	OUTSIDE FINISH	GRILL SCREEN, FINISH	IMPEDANCE, OHMS	FREQUENCY RESPONSE, dB	BASS DRIVER, DIMENSIONS	BASS DRIVER, TYPE	MID DRIVER, DIMENSIONS	MID DRIVER, TYPE	HIGH FREQUENCY DRIVER, DIMENSIONS	HIGH FREQENCY, TYPE	CROSSOVER(S)	WEIGHT, LBS	PRICE	FEATURES
PEAVEY AUDIO MEDIA RESEARCH															
HDH2	30 27 27	carpet	black mesh	8	40-16k 3	15	cone			2 (4)	horn	1.2	108	$749.99	High-level, 2-way, direct radiating enclosure. Processors and bi-amp capable.
3020 HT	38 36.5 18	black carpet	black perf steel	4	45-20k 3	15 (2)	cone	10 (2)	cone	2 1	horn twt	400 2.2k 8k	161	$949.00	High-level, full-range or triamped enclosure. Internal 4-way passive X-over. SPL is 99.5 dB.
HDH1	59 27 27	black carpet	steel		40-16k 3	15 (2)	cone	12	horn	2 (4)	horn	300 1.2	250	$1,899.99	High-level, 3-way enclosure,tri-amp only. Processor capable.
1245M	17 20 17	black carpet	black perf steel	8	100-14 3	12	cone			2	horn	1k	53	$449.99	2-way, biampable stage monitor. Built-in passive X-over. SPL is 102 dB.
1545M	20 20 17	black carpet	black perf steel	8	60-16k 3	15	cone			2	horn	800	59	$479.99	2-way, biampable stage monitor. Built-in passive X-over for full-range operation. SPL is 103 dB.
HDHM	24 22 33	black carpet	steel	8L 8H	60-16k	15	cone			2	horn (4)	800	95 1.2k	$749.99	Biampable, high-level, 2-way stage monitor with internal passive X-over for full range operation.
SP- 2A	31 23.7 17	black text	black mesh	8	60-16k 2	15	cone			2		800	86	$499.99	Front loaded, direct radiating,hi-level passive crossover. Biampable. Handles 300 watts.
SP-4	52.5 25.7 19.3	black carpet	black mesh	4	40-16k	15 (2)	cone			2		1.2k	135	$699.00	Front loaded, direct radiating,hi-level passive crossover.Biampable. Handles 300 watts.
PROFESSIONAL AUDIO SYSTEMS															
MRS-2	51 32 23	gray carpet	black metal	8	40-15k 3	15	cone			2	horn	1k	185	$1.490.00	Modular, full-range with time offset correction.
CX 15-3	25 19 10	gray carpet	black metal	8	60-15k 4	15	cone			1	coax horn	1.2k	70	$684.00	Compact, full-range, high output stage monitor with time offset correction.
RENKUS-HEINZ															
FRS 1S1C	20.1 30.2 16.2	black carpet	black metal	8	50-17k	15	cone			2	comp	1.6k	75	$1198.00	Compact 2-way,passively crossed-over,with 90 degree horn. Stand mountable.
FRS 1S2 CD6	48 20 17.7	black carpet	black metal	4	45-17k	15 (2)	cone			3.3	comp	1.6k	123	$1860.00	Double 15 inch system with large format compression driver. Box passively crossed-over.
SUB 152	48 20 17.7	black carpet	black metal	4	45-200	15 cone (2)							138	$1700.00	Low-profile, band-passed,tuned subwoofer.
SR 121D	24.3 15.7 14.2	black carpet	black metal	8	50-17k	12	cone			3.3	comp		52	$1820.00	Compact trapezoidal enclosure. Max continuous output of 126dB @ 1M.
SR2A	51 23.7 17	black carpet	black metal	4	40-17k	15 (2)	cone			3.3	comp		52	$2890.00	Trapezoidal 2-way system under processor control. Max continuous output of 134dB @ 1M.
C-1 M/H	36.5 31.5 28	black carpet	black metal	8	300-17k	15 (4)	cone			5.6	comp		138		$4900.00Large-format coaxial point source with constant directivity control down to 300Hz.
C-1 LOW	36.5 31.5 28	black carpet	black metal	4	50-300	15 (4)	cone						185	$3960.00	Four 15inch low frequency section of C-1 system. -3dB down point is 5 0Hz.
C-1 SUB	24 31.5 31.5	black carpet	black metal	4	36-80	18 (2)	cone						175	$3505.00	Band-pass tuned, double 18inch subwoofer will produce 136dB @ 1M.
ROSS SYSTEMS															
H218 EV	38 38 18	grey carp	black metal	4	35- 18.5k 3dB	18 (2)	cone	14x 23	comp drvr	8x 10	CD horn	300 3k	176	$1249.95	3-way biampable in Thiele aligned vented enclosure. 500Watts RMS, 1000watts peak.
H118 EV	43 26 18	grey carp	black metal	4	45- 18.5k 3	18	cone	14x 23	comp drvr	8x 10	CD horn	300 3k	125	$895.00	3-way biampable in Thiele aligned vented enclosure. 300watts RMS, 600watts peak.
T183 EV	30 24 17	grey carp	black metal	4	45- 18.5 3	18	cone	8	cone	8x 10	CD horn	300 3k	100	$699.95	3-way biampable. 300watts RMS, 600watts peak.
T153 HS	28 23 16	grey carp	black metal	8	55- 18.5k 3	15	cone	8	cone	4x 10	CD horn	300 3k	88	$549.95	3-way blampable, with constant directivity horn. 350watts.
T152 HS	24 18 17	grey carp	black metal	8	55- 18.5k 3	15	cone			4x 10	CD horn	3k	65	$399.95	Hurricane LF-15 and HF01/CDT in a compact cabinet. 300W.
T122 HS	21 16 16	grey carp	black metal	8	60- 18.5k 3	12	cone			4x 10	CD horn	3k	45	$249.95	Premium 12inch speaker plus HF01/CDT horn. 200watts.
T122	21 16 16	grey carp	black metal	8	65- 18.5k 3	12	cone			4x 10	HF 44	3k	45	$199.95	2-way, 100 watt. Features HF44 compression driver.
M152 EV	28 17 14	epoxy	black metal	8	55- 18.5k 3	15	cone			8x 10	CD horn	3k	56	$499.95	HT-94 horn flare for high power and extended bandwidth stage coverage. 200watts RMS.

MODEL	DIMENSIONS, HWD	OUTSIDE FINISH	GRILL SCREEN, FINISH	IMPEDANCE, OHMS	FREQUENCY RESPONSE, dB	BASS DRIVER, DIMENSIONS	BASS DRIVER, TYPE	MID DRIVER, DIMENSIONS	MID DRIVER, TYPE	HIGH FREQUENCY DRIVER, DIMENSIONS	HIGH FREQENCY, TYPE	CROSSOVER(S)	WEIGHT, LBS	PRICE	FEATURES
SHURE BROTHERS INC. —See our ad on Cover IV															
3200	25 17 13	black vinyl	black metal	8	60-13k 5	12	cone			12x 8	CD horn	2k	38	$416.00	120 watts, protective corners, handle, stand mounting bracket, time-sync crossover.
3100	24 16 12	beige paint	brown fabric		60-13k 5	12	cone			12x 8	CD horn	2k	52	$470.00	120 watts, paintable finish, T-nut hanging hardware, time-sync cross-over.
SUNN															
SPL 1225	28 25 19	gray carpet	black metal	8	50-20k 3	15	cone			comp driver	horn	1.3k	74	$499.99	Trapezoid plywood cabinets, 200W power handling, 100 dB SPL.
SPL 1226	48 25 19	gray carpet	black metal	4	38-20k 3	15 (2)	cone			comp driver	horn	1.2k	116	$699.00	Trapezoid plywood cabinet, 400W power handling, 100 dB SPL.
1282	17 25 19	gray carpet	black metal	8	60-15k 3	12	cone			comp driver	horn	1.2k	52	$449.99	150W power-handling, 100 dB SPL.
1285	17 25 19	gray carpet	black metal	8	50-15k 3	15	cone			comp driver	horn	1.2k	56	$479.99	200W power-handling, 100 dB SPL.
1201		gray carpet	black metal	8	60-17k	12	cone				dual piezo	1.5k	41	$239.99	Cost effective 2-way system with 100 watts power handling.
1205		gray carpet	black metal	8	60-17k	15	cone				dual piezo	1.5k	48	$269.99	Features similar to 1201.
1211		gray carpet	black metal	8	50-20k	15	cone				comp drivr	1.2k	57	$399.99	Full range 2 way system. 150watts-95dB SPL.
1228		gray carpet	black metal	8	45-20k	18	cone				comp drivr	1.2k	82	$599.99	Ideal DJ speaker. 150watts, 95dB SPL.
TURBOSOUND															
TXD- 520	16. 11. 10.	blue paint	perf steel	8	100- 18k 4	10	cone				soft dome		26	$665.00	Low frequency enclosure for TXD series.
TXD- 530	12 25 13	blue paint	perf steel	8	90- 20k 4	10	cone				slot twt		45	$1,064.00	Wide dispersion.
TXD- 580	45 21 17	blue paint	perf steel	8H 8L	45-18k 4	18	cone	10 (2)	cone	1	horn		150	$2,244.00	Full-range high power wide dispersion enclosure for bands, club, cinema, DJs, mobile systems.
TFM-2	22 14 33	black foam	black	3	60-18k	15	cone	15 conc	turb conc	2	turb		125	$2,693.00	Very high power. Concentric dispersion.
TMW- 215	14 34 20.7	black paint	black foam	4L 16H	90-17k 3	15 (2)	cone			2	horn		92.6	$1647.00	High power, low profile.
TMW- 212	12 16 27	black paint	black foam		110-18 3	12 (2)	cone			1	horn		66	$999.00	Low profile, 2-way passive monitor for vocal, keyboards, percussion.
TMW- 210	10 23 13	blue paint	black foam		150-18 3	10 (2)	cone				twt		33	$895.00	Extra low profile vocal wedge for acoustic and jazz applications.
TXD- 518	29 21 17	blue	perf steel	8	45- 250 (act)	18	cone						82	$999.00	Low frequency enclosure for TXD series.
TOA ELECTRONICS															
380- SE	29.8 19.6 16.1	gray poly paint	black jersy	8	50- 20k 5	15	cone		horn		exp supr twtr	800 8k	79.2	$899.00	Bi/triampable, MF and HF attenuators. High efficiency. 360watts RMS continuous.
480- SE	32.3 22 17.7	gray poly paint	black jersy	8	40- 20k 5	18	cone		horn		exp supr twtr	600 2k	99.2	$1179.00	Bi/triampable, MF and HF attenuators. High efficiency. 360 watts RMS continuous.
380- SD	26.8 19.4 15.4	gray tolex	black metal	8	50- 20k 5	15	cone		horn	3.5	horn	1k 10k	59.5	$678.00	Bi-ampable, MF attentuator, slot-loaded LF driver. Stand mountable. 360watts RMS continuous.
300- SD	23.1 18.1 12.3	gray tolex	black metal	8	60- 20k 5	12	cone		horn	3.5	horn	1k 10k	46.3	$619.00	Bi-ampable, MF attenuators, slot loaded LF driver. Stand mountable. 360watts RMS continuous.
SL- 150	23.6 18.1 12.2	gray tolex	black metal	8	70- 20k 5	15	cone				horn	2k	24.3	$359.00	Interlocking corners for stacking. Stand mountable. 240watts RMS continuous.
SL- 120	20.5 15.0 12.2	gray tolex	black metal	8	70- 20k 5	12	cone				horn	2k	28.7	$299.00	Same features as SL-150.
RS- 20	17.5 13.7 13.2	black vinyl	black jersy	8	90- 20k 5	5 (4)	cones				horn	1.5k	29.8	$299.00	Front latching covers, stand mountable. 400watts RMS continuous.
SEB	19.7 26 16.1	gray poly paint	black jersy	8	40- 150 5	18	cone					120	103		Subwoofer, bass reflex, passive crossover and direct input.

MODEL	DIMENSIONS, HWD	OUTSIDE FINISH	GRILL SCREEN, FINISH	IMPEDANCE, OHMS	FREQUENCY RESPONSE, dB	BASS DRIVER, DIMENSIONS	BASS DRIVER, TYPE	MID DRIVER, DIMENSIONS	MID DRIVER, TYPE	HIGH FREQUENCY DRIVER, DIMENSIONS	HIGH FREQENCY, TYPE	CROSSOVER(S)	WEIGHT, LBS	PRICE	FEATURES
YAMAHA CORPORATION OF AMERICA—See our ads on page 14-15															
S4115 HIII	30.7 25.2 18.2	black	black metal	8	45-16k	15	carb fiber			1.7	comp drvr horn	1.6k	97	$695.00	New version of the 2-way stage monitor. Rugged cabinet with recessed handles.
S2115 HIII	21 w/blk 23.3 26.2	ply	black metal	8	50-16k	15	carb fiber			1.7	comp drvr	1.6k	77	$695.00	High power-handling and excellent low frequency response. Rugged two-way system for stage monitoring.
S115 MT	22 28.6 13.9	ply w/blk	black metal	8	40-20k	15	cone	6.5	cone	3.2	bullet	1.2 5k		$445.00	All-cone 3-way system for close-field reinforcement applications. Available in oak finish.
S112H	18 24.3 11.8	ply w/blk	black metal	8	60-16k	12	cone				comp drvr horn	1.6		$345.00	Compact, 2-way system for medium-throw applications. Ideal for tight stage situations. Opt. oak finish.
SM15H	20.5 25.75 15.5	ply w/blk	black metal	8	30-16k	15	cone				comp drvr horn	1.6		$395.00	2-way slant-front stage monitor with wide high frequency dispersion
SM10H	14.75 18.75 14	ply w/blk	black metal	8	60-16k	10	cone				comp drvr horn	2k		$295.00	Ultra-compact stage monitor with high power. Ideal for applications requiring "invisible" speaker.
MS-101	8.5 5.8 7.7	black	black metal		30-20k	4	full range						4.9	$125.00	Compact self-powered (10watt) monitor with mic and two line inputs. Volume and tone controls.
MS-202	8.5 11.5 7.7	black	black metal			4 (2)	full range						8.8	$195.00	Wide-range self-powered monitor (20watt) with mic and 2 line inputs. Volume and tone control
YORKVILLE SOUND															
MX-1000	40.2 27.2 19.1	black ozlte	black perf steel	4	55-16k 3	15 (2)	cone			16x5	horn	2.3	143	$1,250.00	Sensitivity is 105. SPL is 134 dB (1W/1M). Protection circuitry. XLR or 1/4 inch connectors.
SW-600	27.8 23.2 22	black ozite	black perf steel	8	50-300 3	18	cone						100 (rec)	$675.00	Program power is 600watts, efficiency 100dB. Overload breaker. Stand socket on top.
SW-1000	43.3 28.5 24.5	black ozite	black perf steel	4	45-300 3	18 (2)	cone						100 (rec)	$1,299.00	Program power is 1200 watts, efficiency of 103dB. Overload breaker, tilt-back dolley wheels.
M-160	17.9 14.6 13.6	black ozite	black perf steel	8	65-19k 3	10	cone			9x5	horn	4k	33	$440.00	Program power is 160watts, efficiency is 99dB. Stereo processor, subwoofer crossover.
M-400	23.5 20.5 13.8	black ozite	black perf steel	8	60-20k 3	10 (2)	cone				horn	4k	62	$699.00	Program power is 400watts. Efficiency is 97dB. Stereo processor, subwoofer. Overload breaker.
M-600	19.5 23.2 13.5	black ozite perf	black steel	4 3	50-16k (2)	10	cone				horn	1.8	77	$899.00	Program power is 600 watts, efficiency is 103dB. Stereo processing, subwoofer crossover.
MX-401	27.2 21.7 16.1	black ozite	black perf steel	4	45-19k 3	15					horn	2.k	66	$825.00	Program power is 400 watts. Efficiency is 102dB. Stereo processor and subwoofer crossover.
MX-160M															

ADDRESSES

Altec Lansing Corporation
10500 West Reno
Oklahoma City, OK 73128

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

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

Audio Media Research (Peavey)
711 A St.
Meridian, MS 39301

Carvin Corporation
1155 Industrial Ave.
Escondido, CA 92025

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

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

Electro-Voice
600 Cecil St.
Buchanan, MI 49107

Formula Audio Inc.
RT. 5, Box 4403, Hwy. 39
Zebulon, NC 27597

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

Gauss
9130 Glen Oaks Blvd.
Sun Valley CA 91352

Hartke Systems
485-19 S. Broadway
Hicksville, NY 11801

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

Joe's Sound and Salami Co.
303 Clymer Ave.
Morrisville, PA 19067

Klark-Teknik (Turbosound)
30 B Banfi Plaza
Farmingdale, NY 11735

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

Klipsch & Associates
Box 688
Hope, AR 71801

Martin America
21000 Devonshire St., #205
Chatsworth, CA 91311

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

Panasonic/Ramsa
6550 Katella Ave.
Cypress, CA 90049

Professional Audio Systems
1224 West 252nd St.
Harbor City, CA 90710

Renkus-Heinz
17191 Armstrong Ave.
Irvine, CA 92714

Ross Systems
1316 E. Lancaster
Fort Worth, TX 76102

Shure Brothers Inc.
222 Hartrey Ave.
Evanston, IL 60202-3696

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

Sunn
1130 Columbia St.
Brea, CA 92621

Tascam (Teac Corp. of America)
7733 Telegraph Rd.
Montebello, CA 90640

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

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

Transducer Developments Ltd.
652 Glenbrook Road
Stamford, CT 06906

Yamaha Corporation of America
P.O Box 6600
Buena Park, CA 90622

Yorkville Sound
Witmer Industrial Estate
Niagara Falls, NY 14305

HOT LINE. . . . . to the experts.

Dear Mr. Learned:

I have read with interest your articles on Sound Reinforcement around the world. What I would really like to know is how do you measure the reverberation time in a certain room? What instruments do you need? Is there a simple method to do this measurement?

J.S. Jusak
Indonesia

To which Mr. Learned replies:

Your letter made me realize that my choice of reverberation terminology may have confused some of our readers. Strictly speaking, there is no such thing as "room reverb time." There is, however, a very specific reverberation decay time for every individual room. Each individual frequency of sound emitted into a room decays at its own rate—defined as the time it takes for the amplitude of a given frequency to fall 60 dB below its original emitted amplitude. Measurement of a room's reverb decay time must take into consideration each frequency within the range of human hearing individually, as it is quite common to encounter rooms where certain frequencies decay faster than others. This phenomenon can wreak havoc with amplified sound: if a speaker system in a particular room happened to be over-enhancing frequencies that take longer to decay, the room would, in effect, amplify these frequencies further by increasing their duration time. This would reduce intelligibility and "color" the sound.

There are many sophisticated audio analyzers that can perform these measurements. TEF or FFT analyzers provide the most accurate measurements, but at the greatest cost. Certain types of real-time analyzers can perform these measurements if they are equipped with an RT60 package. RT60 is the term used in reference to this reverb decay time measurement. While there are many schools of thought on the ideal sound source for these measurements, my personal preference is a pink-noise burst. My use of the term "reverb time" in reference to the venues in my articles refers to my own simple method of assessing an average RT60. I stand in the center of the audience area, and loudly clap my hands. I then use a stopwatch to time the duration of the reverb, stopping the clock when I can no longer hear any. Something to remember: individual reflections or flutter echoes are not reverb. I also shout, using words like "you" or "hoe" that contain lower frequencies; this gives me some idea of the relative differences in decay times between high and low frequencies. When the sound system is up and operating, I usually play a tape or CD of music that has excellent fidelity and a wide range of sounds. I quickly mute this signal, again observing the time it takes for the reverb to die away.

With experience, this method can tell you a lot about where your problem areas will be. Remember, however, that colored system response could give you a false read on room reverb. As a very general rule, if a sound system exhibits the same sonic coloration in rooms of different size and composition, the problem lies with the speaker system.

If a specific coloration occurs only in a certain room, the problem may lie with the room's decay time at those frequencies. The bottom line in all of this is to remember that what you hear from a sound system is a combination of what the system puts out and how the listening area colors it. Remember that more level injected into a room will result in more reverb. Often, turning down the overall volume is an excellent solution to intelligibility problems caused by long reverb decay times. db

Write to HOTLINE!, db Magazine, 203 Commack Road, Suite 1010, Commack, NY 11725. All letters become the property of db Magazine.

New Products

INTERFACE

• The RW760 is a rack-mountable unit which will allow the HME 700 Series Intercom products to connect to two-wire, non-compatible three-wire, four-wire or telephone intercom systems. The RW760 includes a modular phone plug and the feature of capturing and holding the telephone line. An AC adapter is available to power the unit if not being powered by the three-wire intercom line.

Mfr.—HM Electronics, Inc.

Price: available upon request

Circle 31 on Reader Service Card

VOCAL MIC

• Electro-Voice has introduced the MS-1000 wireless vocal microphone with the EV N/D757 capsule. The MS-1000 is a dual-receiver true-diversity system with two separate receivers on the front end. To virtually eliminate dropouts, the receiver with the strongest signal is automatically chosen. To assure interference-free operation in harsh rf environments, the receivers provide high selectivity for rejection of adjacent and interfering signals. The transmitter in the MS-1000 delivers 50 mW of output power. To eliminate popping noises, the transmitter has separate rf-on and mic-on switches and operates for up to 10 hours on a single standard nine-volt alkaline battery. The 19-inch, rack-mountable MS-1000 operates in the VHF range, 165-216 MHz. The system has a switchable mic/line XLR output, line level ¼-inch output, detachable antenna and an internal power supply.

Mfr.—Electro-Voice

Price: $1,368.00 (includes receiver and microphone)

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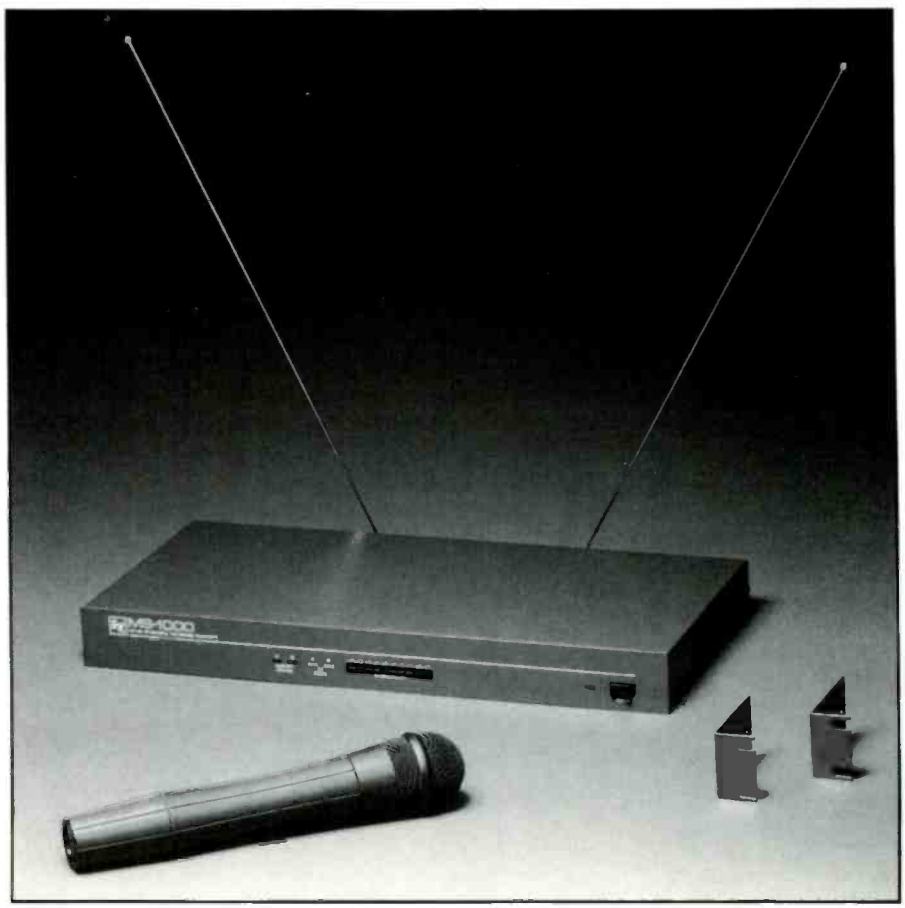
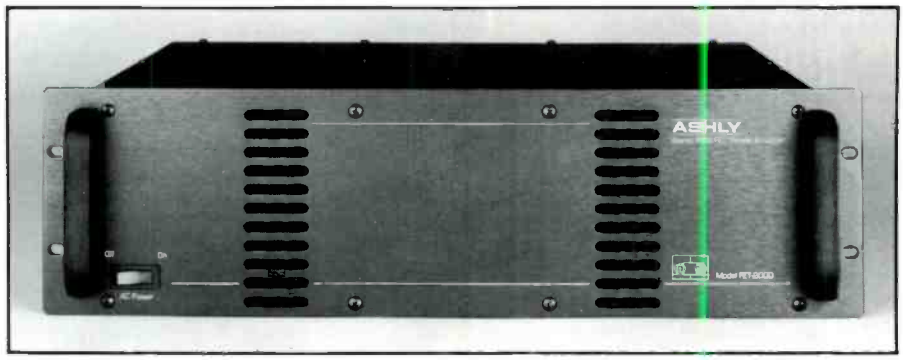


POWER AMPLIFIER

• The FET-2000C power amplifier features up to 500 watts RMS per channel into four ohm loads, with the FTC rating of 250 watts RMS per channel into 8 ohms. The FET-2000C is meterless and features barrier strip input connections for permanent installations. Paralleled phone jack input connectors are standard for bench testing or for use as alternative inputs. It meets all Underwriters Laboratories' testing requirements and is now U.L. Listed.
Mfr. Ashly Audio, Inc.
Price- $999.99



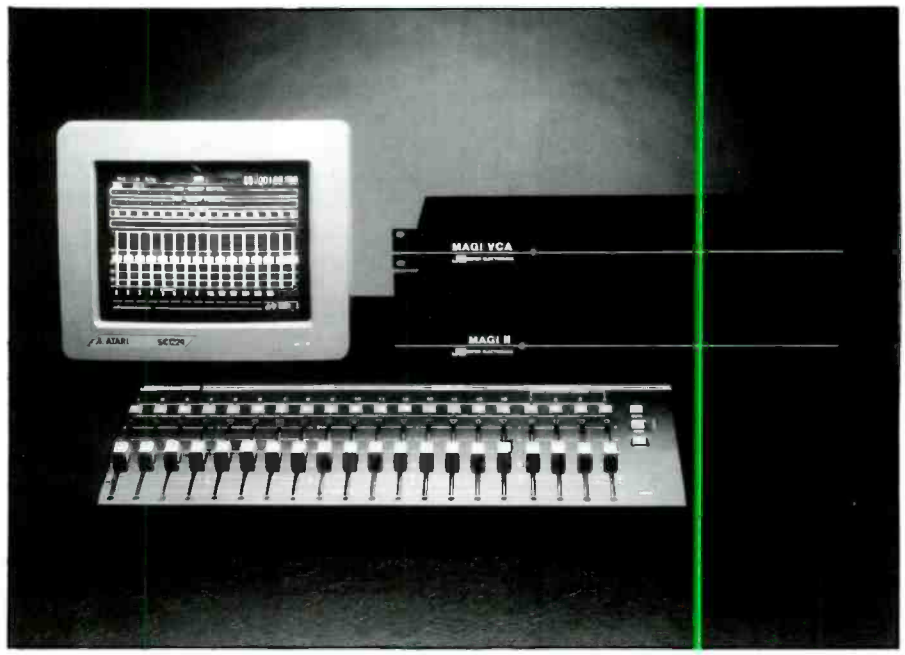
Circle 33 on Reader Service Card

AUTOMATION SYSTEM

• The MAGI II Console Automation System is an integrated state of the art automation system that easily interfaces with any mixing console to provide SMPTE-locked fader and mute automation. MAGI II consists of rack mounted, high quality dbx VCAs, the MAGI II Controller unit, a MAGI remote fader unit, and software. The MAGI II software runs on either an Apple Macintosh (SE, Plus or 512) or an Atari ST (520, 1040 or Mega) computer. The system is available in 16, 32, 48 & 64 channel configurations. MAGI II also features a built in SMPTE generator that can read and write all formats of SMPTE time code.

Mfr:—J.L. Cooper Electronics
Price: available upon request

Circle 34 on Reader Service Card



AUDIO SYSTEM

• The KF300 is the latest addition to the Forsythe Series of high definition audio systems. The KF300 incorporates the company's unique Virtual Array technology in an ultra-compact package designed for near-field applications. Dimensions are: 27 ½-inches high, 14 ¾-inches wide and 14 ¾-inches deep. A separate subsection reproduces the entire mid-range. The KF300 is composed of a 12-inch woofer, a VA design mid-bass horn loading a carbon-fiber 6 ½-inch cone driver, and a constant coverage HF horn loading a titanium diaphragm compression driver. The 3-way system is capable of SPLs in excess of 125 dB over a frequency range of 80 to 20 kHz.

Mfr:—Eastern Acoustic Works
Price: $2,125.00

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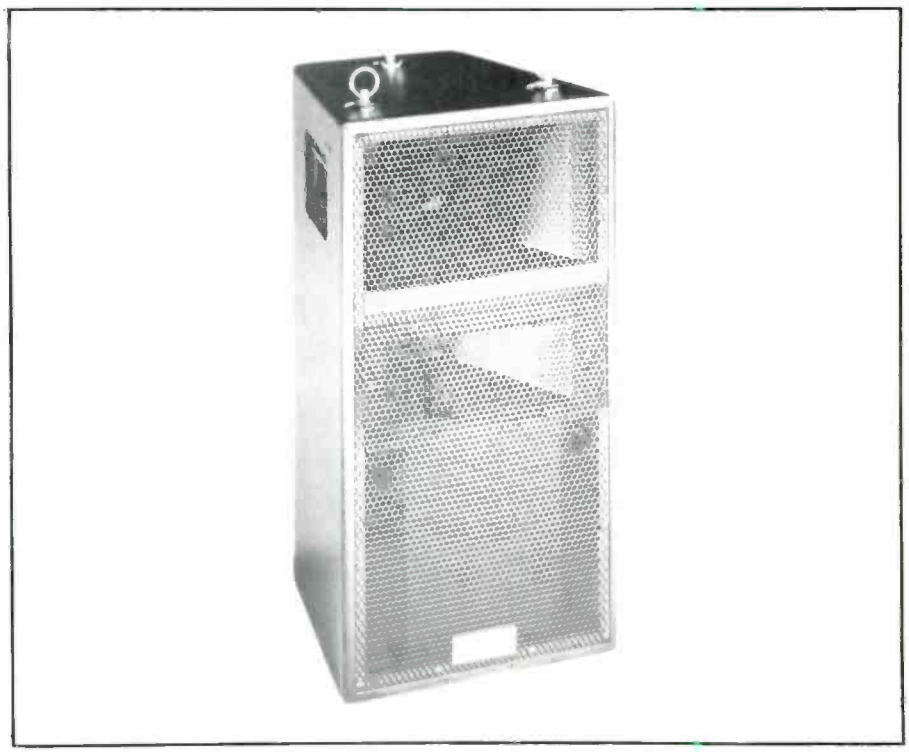
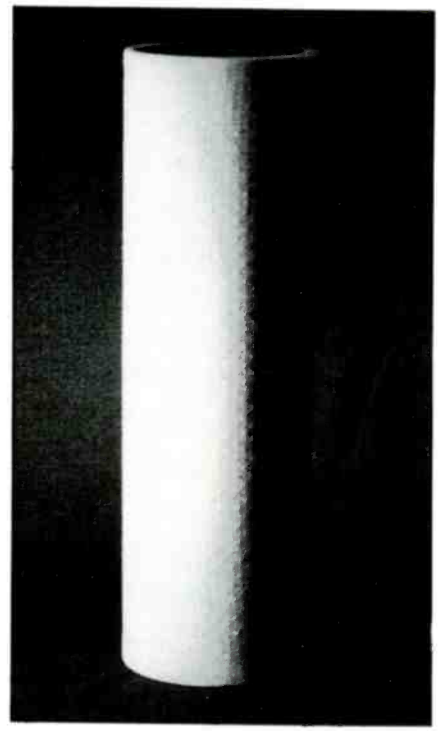


SOUND ABSORBER

• The 11-inch Super Trap, a high-efficiency broadband sound absorber, is a second generation of ASC's popular 11-inch Tube Trap. It features extended bass response and enhanced damping of standing waves into the 70 Hz range. 11-inch Super Traps are specified for rooms with an 8-foot ceiling, to control fundamental floor-to-ceiling resonances that are often accentuated by dynamic speakers and subwoofers. They are available in three and four foot standard or custom lengths, and are covered with Guilford 701 commercial standard fire-resistant fabric, available in a range of designer colors.
Mfr. Acoustic Sciences Corp.
Price- Dependant upon room specs.

Circle 36 on Reader Service Card



PORTABLE SOUND SYSTEM

• The FMR-25P is a portable sound system with wireless microphone capabilities. It is a complete, self-contained system with amplifier and speakers housed in a compact enclosure. The unit weighs only 18 lbs. and is 9 inches high by 13 inches wide by 9 inches deep. It comes with a protective cover with handle for easy carrying. The FMR-25P features a powerful 75 watt MOSFET amplifier and a tri-radiant speaker array in an acoustically-tuned durable roadie case. It is easily capable of handling crowds in excess of 500. Bass and treble tone controls and separate level controls for each input allow complete control over the program quality and mix. It also has an external speaker output that can drive an external, unpowered speaker.
Mfr. Telex Communications, Inc.
Price- $940.00

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RACK-MOUNTABLE DRAWER

• Rackdrawer is a "two space" rack-mountable drawer that will fit into any 19-inch equipment rack. Shipped fully assembled, it mounts instantly and securely into a studio or mobile rack. Rackdrawer has been constructed for on the road durability, and provides a storage space for cables, adapters, tuners, effects pedals, tapes and more.
Mfr. Four Designs Co.
Price- $44.95

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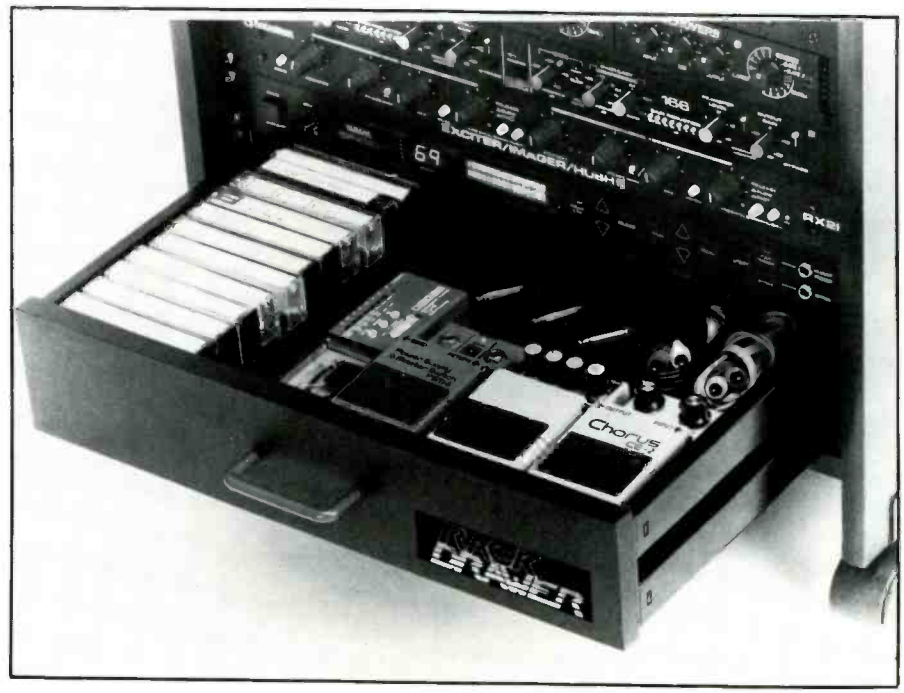
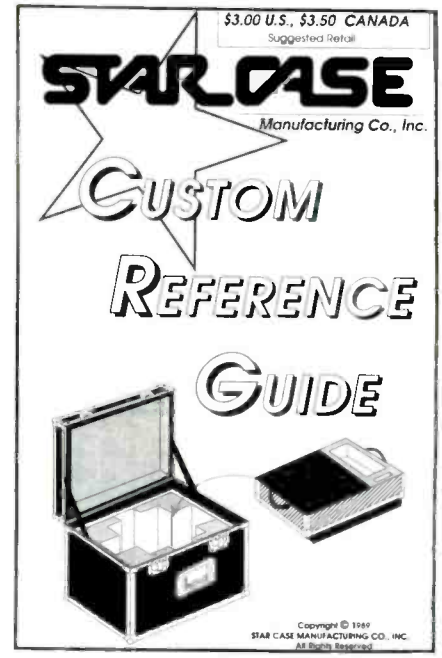


REFERENCE GUIDE



• The Custom Reference Guide devotes nearly 100 pages to a graphically-rich layout of case and container types, styles, optional attachments, interior treatments, shipping costs and time estimators. The guide also features a comprehensive "how-to" chapter on "Measuring, Templating & Design Conception."
Mfr. Star Case Manufacturing Co., Inc.
Price- $3.00 (U.S.), $3.50 (Canada)

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CARDIOID MIC

• The Type 4012 Professional Microphone is a pre-polarized condenser microphone with a first-order cardioid directional pattern, powered from Bruel & Kjaer Dual-Channel Power Supply Type 2812. Type 2812 supplies 130 V to the preamplifier of the Type 4012, which enables the microphone to handle up to 168 dB SPL before clipping occurs. The on-axis frequency response measured at 30 cm distance is flat from 40 Hz to 20 kHz (+1,–2dB). It also has a high output level and the ability to drive very long cables without noticeable deterioration of the signal.

Mfr.—Bruel & Kjaer
Price: $1,633.00
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SPEAKER ENCLOSURE

• The redesigned 112T two-way speaker cabinet now contains a new enclosure featuring a 12-inch STS speaker vented through a pair of tuned ports. Highs are handled by a Motorola tweeter focused through an acoustic lens. The lens controls the vertical dispersion pattern to 45 degrees while dispersing the sound over a horizontal plane of 80 degrees, wider than could be obtained with a horn alone. At only thirty pounds, the 112T is easily transported.
Mfr. SoundTech Int'l.
Price- $179.50
Circle 41 on Reader Service Card

CORD FASTENERS

• Cord-Lox will color code your cables, sort by size, length or type, and keep them neatly coiled for protection and safety. Cord-Lox are designed in such a way as to easily attach to your cords, and then wrap around them to keep securely bundled and tangle-free. There are currently 19 standard models available, to fit the smallest or largest cords. The company also custom builds virtually any size or type you would require for special needs.
Mfr. Toleeto Fasteners Int'l.
Price- Various

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db magazine

serving: recording, broadcast and sound contracting fields

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Lab Report: Fostex D-20 Digital (DAT) Recorder. Len Feldman. November/December 1989, p. 57.

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BUYER'S GUIDE INDEX

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People, Places... & Happenings

• **Ben Rizzi of Master Sound Astoria**, New York has signed an agreement for a joint venture with the Republic of Latvia, which will complete the design, construction and installation of the new recording studios to be called **Master Sound Riga**. Rizzi, along with **Maxine Chrein**, will manage the new operation.

The facility will open with two fully-equipped, state-of-the-art recording studios. "This will mark the first opportunity for artists from Western Europe, the Baltic Republics and across the Soviet Union to have their product compete more effectively in world markets from a technological perspective," said Rizzi.

In 1987, **Master Sound Astoria** pioneered the world's first interactive all digital, bi-coastal recording session via fiber-optic and satellite transmission. Plans are being made to create a permanent link between the Riga and New York facilities utilizing similar technology. This will enable musicians to record simultaneously in Riga and in New York. The digital signal will be sent between the two locations without any loss in sound quality.

Additional features of **Master Sound Riga** will include multitrack analog and digital recording, computerized mixing, internal fiber-optic interfacing, and superior acoustics. The contract signed recently in Riga by **Raimonds Pauls**, Minister of Culture of the Republic of Latvia, signifies the first joint venture agreement of this magnitude entered into with the United States by a member of the Baltic Republics since they were promised economic autonomy from the Soviet Union this summer. The deal further positions the Baltic Republics on the front-line of *glasnost* and *perestroika*.

"This venture between **Master Sound Astoria** and the Republic of Latvia will allow all musicians in the Soviet Union access to one of the highest quality recording facilities in the world," said Pauls, after signing the agreement. Pauls, besides being Minister of Culture of Latvia, is one of the most well-known musicians in the Soviet Union. He is renowned for composing jazz and popular music as well as being a virtuoso pianist and bandleader. The Baltic Republics share the love and appreciation of the rich cultural diversity of the USSR's music and arts, and yet the Baltic Republics' cultural heritage is considered more westernized than the rest of the Soviet Union.

• **Bob Herrold** has accepted a new position with **Crown International** as Microphone Product Line Manager. In this capacity, Herrold will determine product needs for the broadcast, recording, film, pro sound, and MI markets, and work with Crown's engineering and marketing departments to design and manufacture products to meet those needs. He will also determine pricing and corporate programs, and conduct educational seminars on both the dealer and end-user levels. Prior to taking on these new duties, Herrold spent 12 years with **Audio-Technica U.S., Inc.** as a design engineer and product manager. Previous to that position, he spent 12 years being employed by **Electro-Voice, Inc.**

• **Tannoy North America** is now **TGI North America**. In January 1988, Tannoy, **Goodman's** and **Mordaunt-Short** joined forces to become a new publicly-listed company in the U.K. under the new name of **TGI plc**. **TGI North America** will continue to distribute Tannoy products in North America as well as expand its distribution network to include other TGI companies.

• **Electro Sound Inc.** has sold music duplicating systems to three leading companies in the People's Republic of China. **Simex,** in Shanghai, and **Kong Sen Tape Factory,** located in Panyu, Guanzhou and Guandong, have acquired new duplicating systems, while **Shaanxi Audio,** based in Xi'an, has acquired equipment to expand its duplication capability. "The penetration of high-speed tape duplicating systems and equipment in China is significant because it indicates that China is advancing its technology in the music and tape industry," said **Jim Williams**, President of Electro Sound.

• **Keith Clark** has been named Public Relations Director for **Electro-Voice, Inc.**, as announced by **Jesse Walsh**, Director of Advertising. Clark's responsibilities with Electro-Voice include corporate communications and trade publication relations.

In related news, a new full-line service center for repairing Electro-Voice electronic components, microphones and speakers has opened in Richmond Hill, N.Y. The facility, operated by ECS, Inc., provides EV with a service center for the eastern region of the U.S., to go along with existing facilities in Buchanan, MI, and Visalia, CA.

People, Places... & Happenings

• Deane Ellsworth Jensen died suddenly in his private audio engineering laboratory at **Jensen Transformers, Inc.**, North Hollywood, CA, the weekend of October 21, 1989 at the age of 47. Deane was well-known for his efforts to improve the fidelity of sound in the recording, reinforcement, and broadcast industries. Jensen Transformers was dedicated to high quality, both in its hardware and software. The company will continue Deane's work: building quality products and serving the audio community with engineering leadership. Jensen was born in Annapolis, MD, 15 October 1942. He lived in Princeton, NJ and Baltimore, MD. He attended the University of Pennsylvania where he started in audio technical engineering and operations at WXPN, the student-operated radio station. He founded **Jensen Transformers, Inc.** in 1972, a few years after he moved to Hollywood, CA, and led it as owner and CEO throughout its successful growth to its present prominence in the industry. In lieu of flowers, etc., the family requests that any contributions be sent to the **Richard C. Heyser Memorial Scholarship Fund**, c/o the **Audio Engineering Society**.

• My personal thoughts on the passing of a friend, By Gary D. Davis.

Deane Jensen and I met in 1972 when I was working as a technician for Automated Processes, Inc. in Farmingdale, NY. Deane had been sent there as a consultant for **Wally Heider Recording Studios** to do the final acceptance/checkout on a large recording console API was building for Heider. Deane ended up staying in New York for several months to oversee completion of the console through many bug-fixes and improvements. Due to the length of the unplanned stay, he was invited to move out of the motel and share a house with my roommate, engineer **Fred Addison**, and I for a couple of months. I learned a lot from Deane about the application of scientific principles and ethics—at a point in my early career where a lasting impression was made. Our friendship grew, and when I moved to California a year later, Deane helped me get established. Jensen Transformers became one of our first clients, and our relationship was always one of mutual respect. I especially learned from Deane that there is a best way to do something—that "good enough" is not really an option when perfection is the goal.

Those who did not know him well sometimes thought of Deane as being somewhat aloof. In fact, he was very warm and concerned about his friends. If he thought he said something that might have been in some way offensive, he would often phone back and make sure there was no insult or misinterpretation. He went out of his way to make sure he was clearly and fully understood. Magazine editors winced at his insistence of a "no edit" policy on anything he wrote, but he refused to have his concepts truncated or mistakenly "improved." This was, perhaps, an extension of his attitude of not accepting anything at face value. If some equation or theory had always been assumed to be a certain way, Deane would do his own basic research to affirm to disprove it. He seldom took anybody's word about an engineering principle without a grain of salt, and this quality served him well. It enabled him to make breakthroughs in many areas. It is why, for example, he ended up making transformers in the first place. Deane simply thought that better ones could be made, and he wanted to have better transformers to use in the audio circuits he was designing. His true goal was better audio, not just better transformers. Ultimately, he devised new ways of winding coils, and was involved in formulating special new alloys to achieve orders of magnitude improvement in transformer performance.

Deane was dedicated to his work. He was not a *workaholic* in the traditional sense, but he was driven to complete a number of ambitious projects. Many of them involved basic scientific work not specifically aimed at development of a new product. It was this unselfish, persistent, inquisitive attitude that will forever remain my most treasured memory of a good friend.

• Gary Davis is President of **Gary Davis & Associates**, Santa Monica, CA. The firm has provided technical writing, advertising, and consulting (primarily for professional audio equipment manufacturers) since 1974.

• The **Audio Engineering Society** regrets to inform our friends and associates of the death of **Ann Marie Smyth**, Wednesday, November 8, 1989 in New York City. Ms. Smyth joined the AES in the late seventies, and was an effective member of the AES convention and administrative staff at AES International Headquarters in New York. Ann was involved in all domestic and international AES conventions and conferences. Her sudden death in the lobby of the Lincoln Building—site of AES headquarters—in New York was symbolic of her dedication to the organization. Her previous career included sales positions in the hotel field. Prior to joining the AES, she was associated with the Cahners and Snitow groups in the organization of high-fidelity shows and the consumer electronics show (CES). Sudden death at the age of 46 has cut short a productive life, a life that touched so many in our industry with concern and compassion toward all who had the good fortune to work with her.

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.

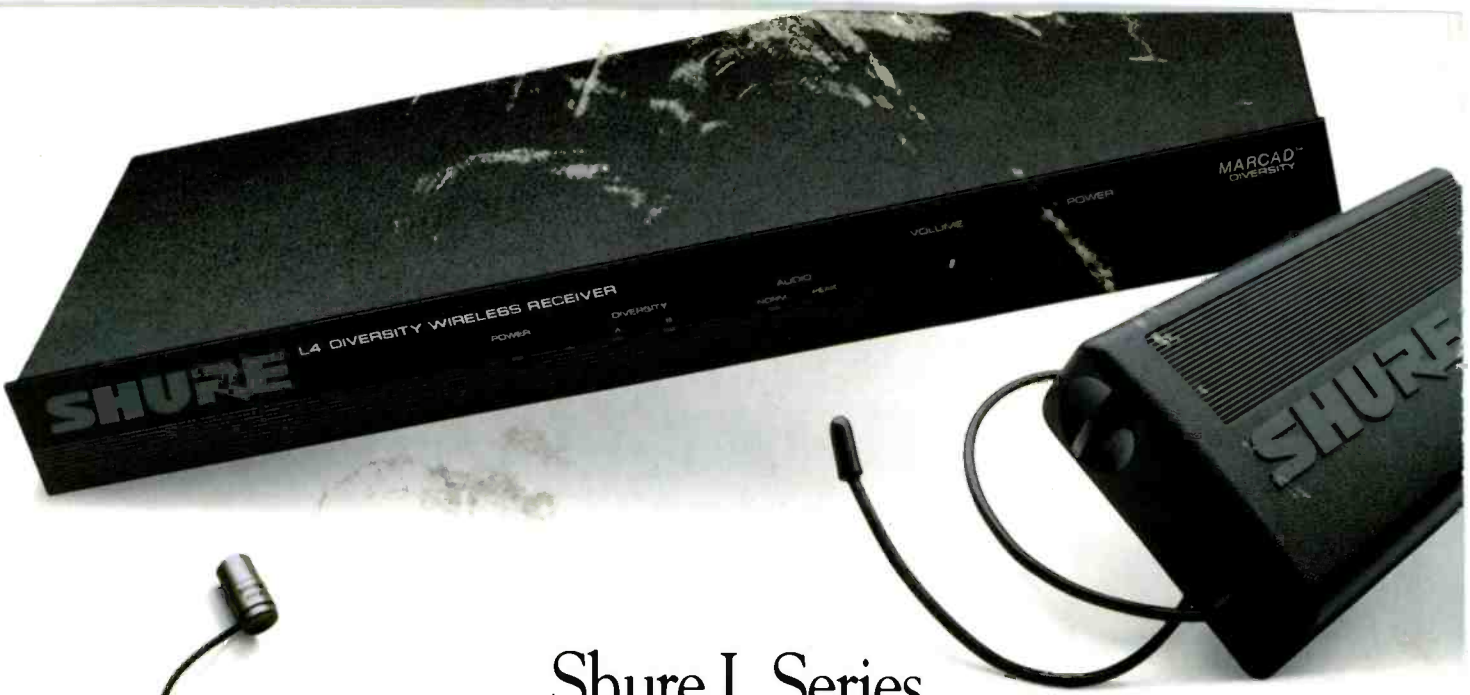
* The analog recorder referenced is the Studer A-80 1/2" with 3-track head nest; the price comparison is based on option (A) below and an average of published rates of major audio rental companies for the Studer.

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®



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 (708) 866-2553.

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