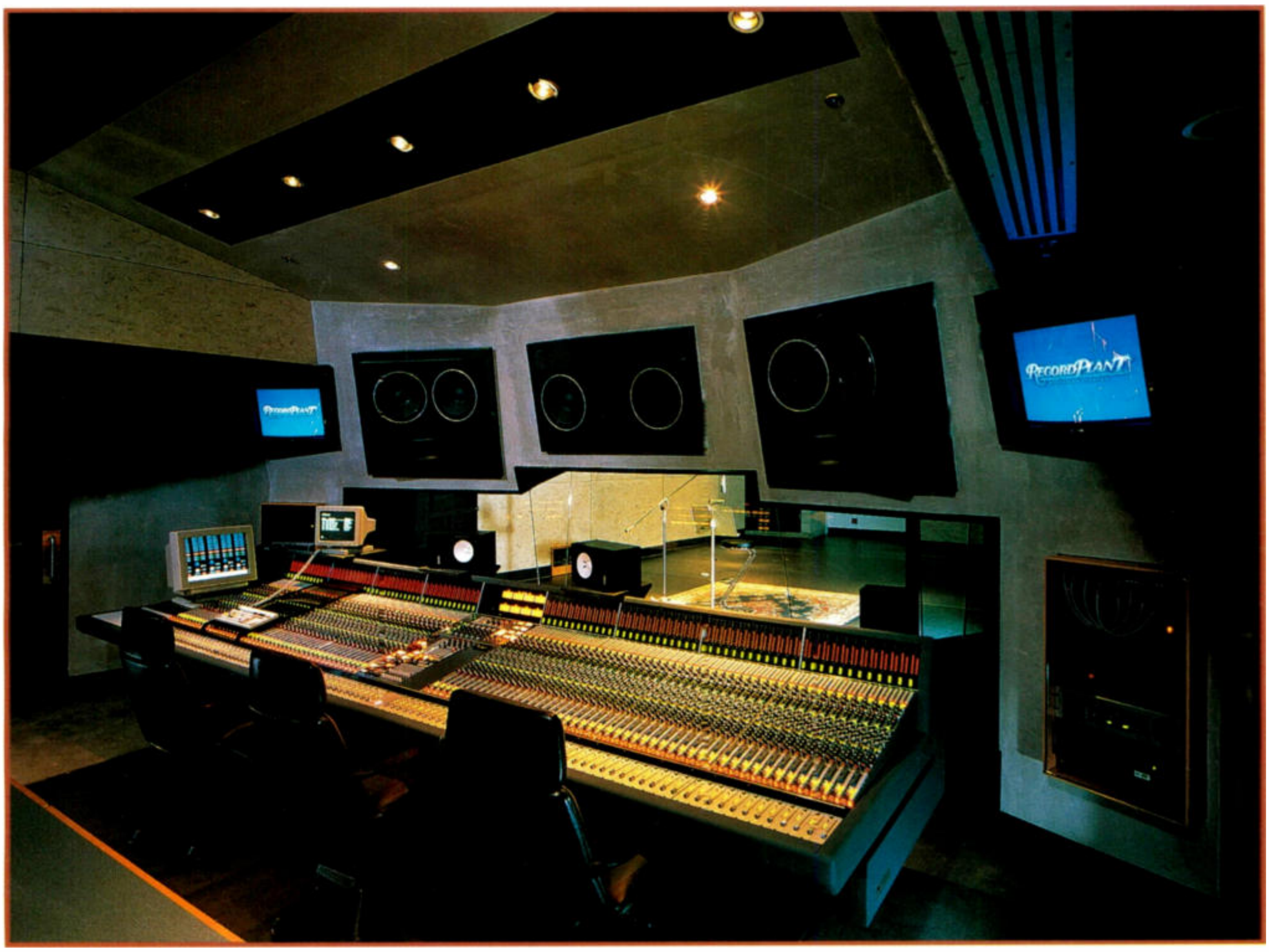
September/October 1993 $3.50

db magazine

serving recording, broadcast and sound contracting fields

Mobile Church Audio
Quality Cassette Duplication



The LA Record Plant at 25 Years

September/October Vol. 27, No. 5

db magazine

serving recording, broadcast and sound contracting fields

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The Recording Engineer

See page 26

AUDIO for the Church

The Sound Contracting Engineer

Departments

About the Cover

● One of the Record Plant's new Studio Suites is called *Neve 1* because it houses a 96-input Neve VRSP Legend console equipped with GML Moving Fader automation and recall automation. Designed to function as a self-contained environment, each Studio Suite includes a private lounge, kitchen and bathroom. See Adrian Zarin's story on the Hollywood Record Plant's 25th year, beginning on page 18.

Calendar

1 **Sound in Entertainment: Specifying and Designing Audio Systems to Create Sound Environments** is the long title of a series of seminars, sessions and exhibits that will take place November 13-15 at the **Orange Country Convention Center** in Orlando, Florida.

The conference is supplemented by the **LDI93** show. Exhibits will be open 10:00 am to 6:00 pm November 13th, 14th, and 15th 1993. LDI93 is sponsored by *Lighting Dimensions* magazine. For information call **Jacqueline Tien** at 212 677-5997 or

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fax 212 677-3857. You can write them at 135 Fifth Avenue, New York, NY 10010

l It's almost too late to start thinking about the **95th AES Convention,** which will be held this year on October 7-10th in New York City. This time however, it will be held at the **Jacob Javits Convention Center**.

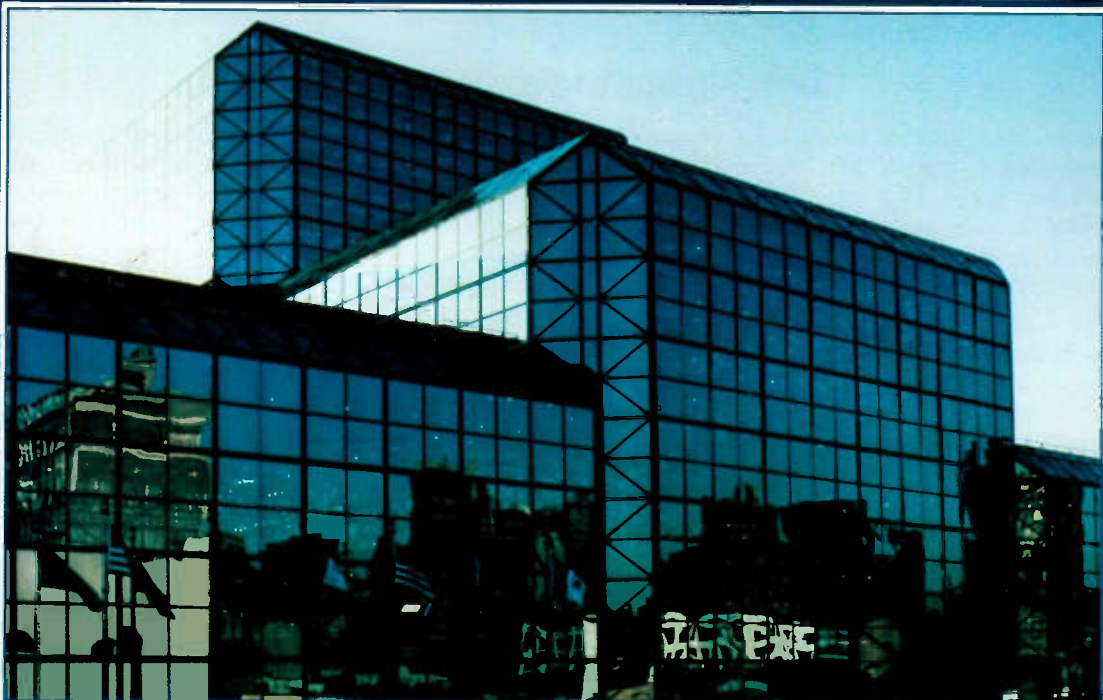
"The AES Convention Committee members agreed that the AES exhibit requirements have outgrown the **New York City Hilton Hotel** where AES Conventions have been held for several years,"according to Convention Chairman **Leornard Feldman**. "Our experience at the 93rd AES Convention in San Francisco confirmed the advantage of having all exhibits on a single floor—a benefit that could not be achieved at the Hilton."

The dates include a Saturday stay, offering advantages to attendees for lowest discounted air fares, while exhibitors will benefit from weekdays being used for setup and breakdown, thus avoiding overtime charges. According to Len Feldman, "the dates of October 7-10 were not available at the Hilton."

For information, contact the AES at 60 East 42 Street, New York, NY 10166, or call at 212 661-8528.

l **SMPTE's Technical Conference and Equipment Exhibit** takes place from October 30th thru November 2 at the Los Angeles Convention Center. **Contact SMPTE at 595 W. Hartsdale Avenue, White Plains, NY 10607 or call SMPTE at (914) 761-1100.**

db



AES 95th Convention

AUDIO IN THE AGE OF MULTIMEDIA

October 7–10, 1993
Jacob K. Javits Convention Center • New York

Science emphasizes detached observation, objectivity, and logical deduction, but most who come away from the AES 95th and New York City this fall will find themselves feeling anything but detached—the combined dynamics of this city and this event simply won't allow it.

JOE CICCARELLO

Mobile Church Audio

Good Things in Small Packages

Mobile Audio has become increasingly popular in the church market. With new technology creating better and smaller packages, quality, state of the art studios can pull right up to the church door via mobile audio.

A company in the forefront of mobile audio is A.S.L. (Aura Sonic LTD.) of Flushing, N.Y. Steve Remote (yes, that's his real name!) is the owner, operator and creator of this amazing 30 foot truck, that has more equipment in it than most trucks twice its size.

HUMBLE BEGINNINGS

Steve began his business 16 years ago when a friend asked him to engineer and produce a local band. Steve however, thought it would be wiser to buy some recording equipment and build a studio so they could produce as many bands as they wanted. Not content with operating a conventional recording studio, Steve decided to create a studio on wheels so they could go to the client instead of the client going to them. Steve's friend bought the idea and not too long after they purchased a used Con Edison truck for $ 770.00 and began to build. The first year was one of growth both in business and experience. Much of what Steve learned about audio did not come through formal training but through hands on experience and reading a particular audio magazine - any clue as to which one that might have been? Steve said that db Magazine helped him learn from the big guys. Reading articles about what they were doing, what equipment they used and what resources were available, helped Steve to directly apply their knowledge to his work. Well, it must have paid off because by the end of the first year of business, Steve had the money to redesign and update his truck. A prominent producer also made some suggestions as how to better utilize the truck's space and Steve implemented them. His motto was (and still is) "continue to update and improve so as to best serve the client's needs".

STAYING CURRENT

Through the years, with Steve's foresight, A.S.L. has been at the forefront of audio's latest technology. A.S.L.'s most recent purchases are six Tascam DA-88 8 track digital hi-8 machines, providing 48 tracks of digital recording (Figure 1). Steve points out that the DA-88s have many advantages, the first being expandability. No longer do you have to sell one machine and buy another to update to a machine with more tracks. If the project grows, just add another machine. Up to 16 machines can be linked together without syncing problems, providing 128 tracks of recording. Another advantage is size. When you're dealing with limited space like a truck, you want to make every cubic inch count. The DA-88s are about one forth the size of an analog 8 track machine enabling you to hop on a plane with two DA-88s, a mixer and a pair of his Genelec (powered) speakers

Figure 1. One of the six Tascam DA-88s that are located in the truck. Linked together these provide 48 tracks of digital audio.

Joe Ciccarello is the Music Director for a 500-member church, the Christen Victory Center in Hempstead NY. He has a Masters of Art in Communication from Regent University.

Figure 2. In the rear of the truck as the customized bi folding doors are opened, we find to our left, two 250 foot reels of cable nestled against the power supplies.

and do a session just about anywhere.

THERE'S NO PLACE LIKE CHURCH

Churches, independent gospel producers and record companies supply A.S.L. with a healthy amount of business throughout the year. Unlike secular clients, churches have unique requirements when undertaking a recording project. It has been said, "You can take the studio to the church but you can't take the church to the studio". Steve's experience has shown him that there is a special vibe or feel in churches that would get lost in the recording studio environment. Many church musicians and singers are often uncomfortable and intimidated in the unfamiliar setting of a studio. Also certain instruments, especially organs, which are so important to gospel music are difficult and in some cases impossible to transport into a studio. Then there's the natural acoustics of some churches and cathedrals that cannot be duplicated in any studio. All of these disadvantages add up to the popularity of mobile audio in church recordings. Steve described a recent scenario that illustrates how the best of both worlds can be achieved through mobile recording in the church. A church was preparing to record a CD. Members of the congregation who were involved with audio asked Steve what they could do to make their church acoustically sound in order to attain studio results. Steve suggested (among other things) the use of baffles to produce isolation between the different sections of the band and choir. Upon his next visit he not only found sturdy customized baffles but ones that were aesthetically pleasing made from clear Lexan plastic in order to facilitate visual contact between the music director and the entire choir and band. The end result was a recording that was studio quality, yet recorded totally within the confines and comforts of the church.

OUTSIDE IN

To really appreciate what's inside of the A.S.L. 'Super truck' we'll take a look at it from the outside in. In the rear of the truck as the customized bi folding doors are opened, we find to our left, (Figure 2) two 250 foot reels of cable. Each cable equipped to carry 52 channels with a multi pin connector at each end. These cables are connected by two snakes within the venue making available 104 separate inputs into the truck. In the center of the rear of the truck are eight individual video lines, five phone lines, two of which are cellular, numerous studio stereo feeds

Figure 3. Steve Remote sits along side of air ducts that are within reach for future hook up for the truck's rear extension

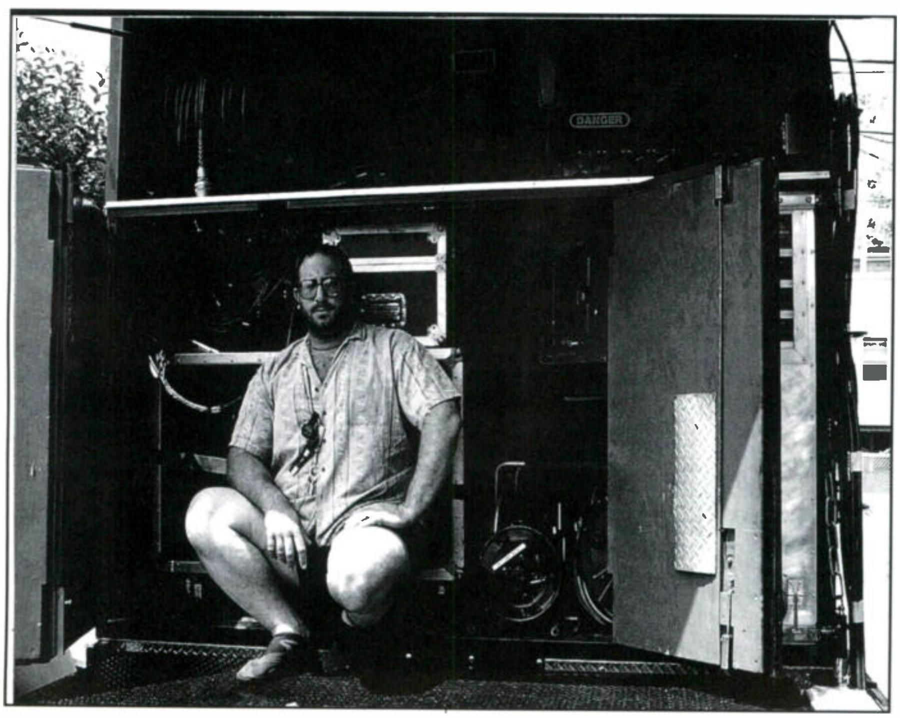


Figure 4. The dominant feature at the front of the room is the 36 channel Harrison MR-4 console.

and four different communication systems; ClearCom, which is used by most sound companies, RTS, which is used by most video companies, KS Audio which is A.S.L.'s own system and an auxiliary system. Finally to the right are the power hook ups; Shore power which is Navy terminology for land power or power that's taken from the venue and Generator power.

Steve has plans to enclose this entire area with a canopy so work can be done out there in all kinds of weather. Even air ducts are within reach for future hook up for the truck's rear extension (Figure 3). A common problem for trucks of this type occurs when repairs have to be made on equipment and accessibility is difficult. Steve however, had enough foresight to design his vehicle with rear entry doors that make all the equipment accessible so that repairs can be made quickly and without pulling apart the inside of the truck.

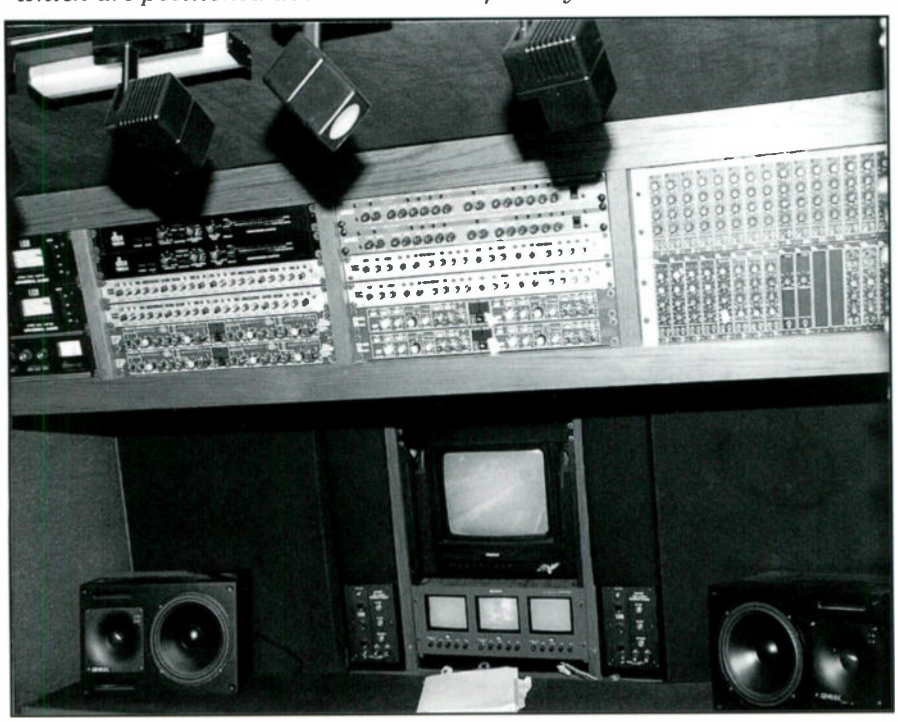
INSIDE

Upon entering the air conditioned control room of the A.S.L. truck, I was surprised how uncluttered the 17 x 7 ½ room appeared. The dominant feature at the front of the room is the 36 channel Harrison MR-4 console (Figure 4). Above and slightly angled back are the reverb units which are fed into a Roland M-160 line mixer. Sitting atop the console are a pair of Genelec 1031A powered monitors and directly behind them are a pair of UREI 813s. In the center of the wall are seven video monitors. During an event these monitors can be connected to various video feeds and cameras including a camera that can turn 360 degrees, operated by a joy stick. Above the monitors are a host of compressors and noise gates which are positioned above the console for easy access (Figure 5) while other outboard gear such as EQs, space echos and other gadgets which don't require constant attention are positioned on the rear wall (Figure 6). Below are two Otari MTR 90 24 tracks. These machines are connected to the inputs of the board with Elco multi pins so all 24 track can be plugged and unplugged with one connection (see Figure 6 above Steve's left arm). Steve takes great care in making backups for much of what he works on. When making a digital audio tape master, he doesn't just make one master but two and therefore owns two Sony DTC 1000ES digital audio tape decks as well as two additional ones by Panasonic. In his own words, "backups to me are paramount even for the client who doesn't know better".

LOOKING TO THE FUTURE

If there is one thing that impressed me about Steve is that he is a man with vision. He loves his

Figure 5. Above the monitors are a host of compressors and noise gates which are positioned above the console for easy access.



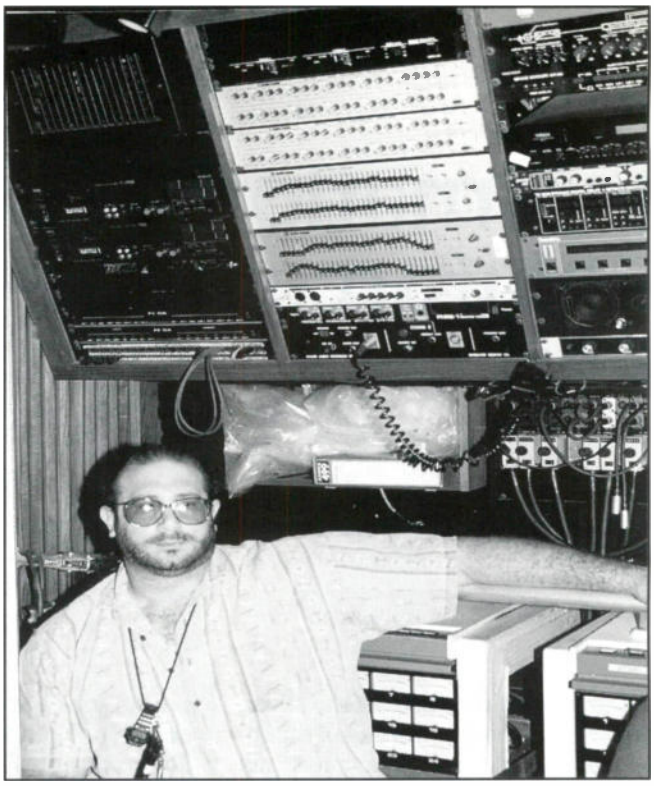


Figure 6. Steve Remote sits among the outboard gear such as EQs, space echoes and other gadgets which don't require constant attention and so are positioned on the rear wall.

work and appears to be challenged by his limitation of space. His future plans include the possible purchase of a digitally controlled analog console which would be able to take snapshots of numerous different mixes.

The big payoff however would be the increase in the amount of tracks. Where 36 tracks now exist in a 7 ½ foot space, the new console would provide 96! Sixty more tracks in the same amount of space.

Where will Steve be ten years from now?

Who knows, he may pull up to your church to record your latest CD in a fully equipped, state of the art Geo Metro. db

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International Sound Reinforcement—West Africa, Part V

Author's Forward: Some of you may have wondered where I disappeared to during the past six months. I recently completed an extensive tour of East Africa (my first visit to this region); then I got busy with festivals and other sound jobs here in Michigan upon my return. My apologies for the delay in completing my tales of West Africa. Also, my thanks to those **db Magazine** readers who have contacted me; in deference to your wishes, I'll gladly talk more about the countries I visit. Enjoy this new format; and look for my series on East Africa in upcoming issues.

● The *Pharoah Sanders Quartet* tour of 1992 included visits to the countries of Congo, Gabon, and Sao Tome. It was this region, culturally beholden to both West and Central African traditions, that proved the most exciting. The old Chinese curse of "may you live in interesting times" certainly came to mind many times—by far, this was the most "interesting" section of our tour. My story begins in Lagos: after our Nigerian concerts were completed, we were scheduled to fly to Libreville, capital of Gabon. We would stay there that night, continuing on to Sao Tome the following day. When I talk about this, I usually get a reaction like "where the *%#! is Sao Tome?" The Republic of Sao Tome and Principe is comprised of two large islands (and a few smaller ones), located about 180 kilometers due west of Gabon out in the Atlantic Ocean. It is one of the few African countries using Portuguese as an official language; it is also one of the hardest countries to reach. There are direct international air connections with Portugal, Gabon, Cameroon, Angola, and Russia only; none of these operate daily. This dearth of regular flights puts a premium on making your connections; transportation machinations are the most important part of any touring project to the RSTP. We should have guessed things were going to be "interesting" when Air Gabon inexplicably changed our departure time from Lagos to Libreville the night before we left, without bothering to notify passengers. Fortunately, the US Embassy expediter in Lagos, at the airport to meet another incoming group, found out about the change and warned us to change our departure plans accordingly. Little did we know there were many more surprises ahead.

Lagos-Libreville entailed a 1 hour 40 minute flight; when we arrived on Sunday, February 2, we were met at the plane by USIS PAO (Public Affairs Officer) Jan Hartman, CAS (Cultural Affairs Specialist) Dieudonne Nzue, and a host of other staffers. Instead of moving into the terminal, we were ushered across the airport directly to another aircraft—plans had changed, and we were now scheduled to leave for Sao Tome *that same day*! Equatorial Airlines operates only 5 flights a week from Libreville to Sao Tome—our flight was a special charter arranged by Equatorial, USIS, and the RSTP government. Dieudonne rushed our passports into the terminal, where Sao Tome *Charge d'Affaires*

was waiting to issue our Sao Tomean visas; the band was briefed on our revised schedule in the air-conditioned comfort of a USIS van. While all this was going on, I was chasing down our band and PA equipment: the Air Gabon flight we'd recently departed continued onward to Brazzaville, we could not afford to have any equipment doing that. I was escorted by the Air Gabon ground crew directly to the plane, where I helped them identify and off-load our equipment and baggage. I then located the Equatorial pilot; together, we dealt with our next problem: lack of cargo capability. Our aircraft was a twin-engine 22 seat propeller-driven Nord (see Figure 1), with a tiny forward cargo hold and a 500 kilo cargo limit; substantially less than the 800 kilos we'd been promised. To make matters worse, it was obvious that our road cases were too big for the cargo area. Since this flight was a charter, our group and USIS-Libreville staff were the only passengers; there were going to be lots of open seats. We agreed that I could use these for extra cargo if I kept the weight down. I did a quick repack of the system right there on the tarmac: the front-of-house rack, amp rack, and console were loaded as is, the cable/stand was shorted down to the bare minimum. These items, plus the piano and bass amp, filled the cargo area to overflowing. I removed the floor monitors and Bose house cabinets from their road cases, placing these in seats on the right-hand side of the cabin; this conglomeration of loose audio gear, drums, instruments, and baggage piled into the seats was then covered with a cargo net and lashed down (see Figure 2). The leftover cases and shorted gear stayed behind with Dieudonne; we would link up with them when we returned to Gabon in 2 days. With a watchful eye on the gear, we taxied, took off, and enjoyed perfect flying weather for the short flight to the island. I spent an enjoyable 20 minutes in the cockpit, ogling the African coastline and the ocean—and it was interesting to see the flight engineer sitting on my cable case, which jutted into the cockpit slightly. Yes—that's how packed we were! I can't thank Equatorial enough for the fine job they did.

Figure 1. Our Equatorial charter and flight crew, on the ground in Libreville.

Figure 2. Making it all fit!

Landing in Sao Tome, we quickly cleared customs and had the gear loaded onto a waiting flatbed; all this courtesy of Guilherme Neto, our representative from Sao Tomean National TV. Sao Tome doesn't get many foreign visitors, and customs facilities at the airport are meager as a result—our rapid processing was a tribute to Guilherme's advance work. Our hotel was the Miramar, the only major hotel in the city. Monday's concert was held at the Palacio dos Congressos (see Figure 3), just a short drive from the hotel.

This hotel was built by the Chinese, and its acoustic signature was very similar to other Chinese-constructed auditoriums I'd encountered in Africa: tile and plaster surfaces contributed to a very bright overall room sound; room decay was in excess of 2 seconds. It seated 1,000 in a very steeply raked single audience area. Power was 223 volts; I tapped off an un-grounded European style round pin receptacle off stage left.

Ground came from a wire run to the backstage bathroom's water pipe. After we completed set up, Guilherme and I climbed into his jeep and took off in search of the band; they were reportedly at the beach. We

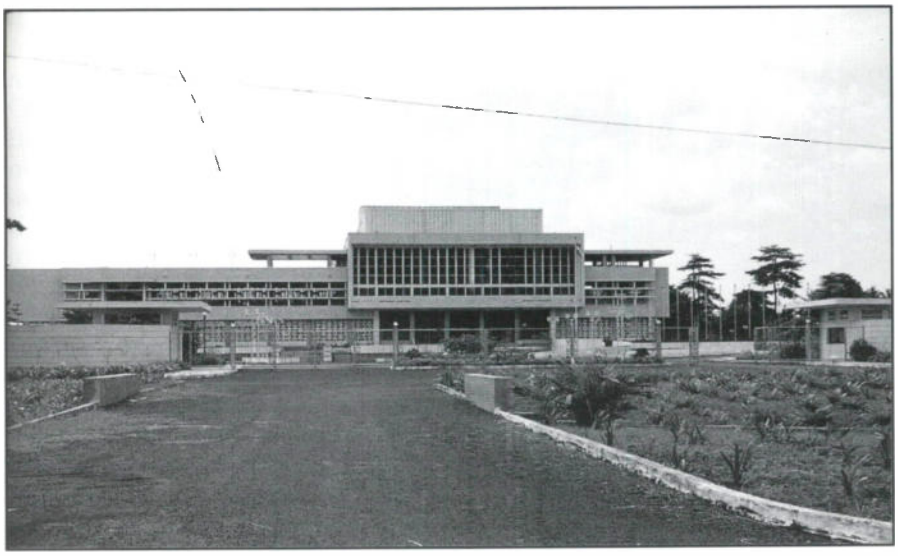

Figure 3. The Palacio dos Congressos in Sao Tome.

twisted through tropical forest and skirted sugar-cane fields on dirt roads that were badly rutted and muddy; several times we had to leave the road to by-pass flooded sections. Without 4WD, you don't get around on Sao Tome. When we arrived at the beach, there was no sign of the band—*but everybody else on the island was there*! Or, at least it seemed that way. People from every nearby village walked—others came crowded in the back of trucks. February 3 was a holiday in Sao Tome, commemorating the Batepa Massacre—the last slave uprising that led to Sao Tome's independence from Portugal. This particular beach was where the official ceremonies and subsequent celebration took place—and what a celebration it was! Everybody was partying, barbecuing, drinking, dancing, etc. I was immediately kidnapped by Equatorial Airlines and forced to join their party; poor me! It turned into a real fun day—I didn't rendezvous with Pharoah and the guys until our 4 PM sound check. Sao Tome doesn't get much outside entertainment; a visit by a musician of Pharoah's stature is special, but having it concurrent with the most important holiday in Sao Tome made in an event for the ages. The place was beyond full: people were sitting in aisles, standing in the back. I'd experienced acoustic hell, but this many people did wonders for the room sound, warming it up considerably and lowering room decay to the point where I could actually mix *all* the instruments into the PA. This audience was fantastic: very quiet and attentive during ballads, they got rocking as Pharoah picked up the tempo towards the end of the set. All songs were concluded to deafening applause. At a reception in our honor après concert at the Miramar, the band continued to bask in the glow of these most hospitable and beautiful people.

Figure 4. Ngonfi.

At breakfast Tuesday morning, I was approached by tour escort Ron Mininger and PAO Hartman with some serious news regarding our Gabonaise program. Apparently, there had been rioting in Libreville over the weekend: students at *Omar Bongo University*, angered by new government policies regarding financial aide and curriculum, had gone on a rampage which included torching the university auditorium. An interesting state of affairs, because *the Omar Bongo auditorium was where our Wednesday concert was supposed to be held*! We were now without a venue, and the concert less than 48 hours off. Jan mentioned that she would try and find us another site, but wasn't optimistic that this could be accomplished on such short notice; we agreed. Of course, our first hurdle was to get back to Libreville: we again used a special Equatorial charter flight.

Leaving at 10 in the morning this time, we arrived at the Libreville airport just after noon: our lost hour was the result of the hour time change between countries. Ron and Jan took the band and their luggage over to the hotel, while Dieudonne and I put the gear back in its cases and re-united the sound system. When inventory was complete, we loaded everything onto an embassy truck for the drive over to the U.S. Ambassador's residence, where we were scheduled to play that evening. Our concert site, located just behind the residence, was situated on a hill overlooking a good portion of Libreville.

Our "stage" was the back patio: this was covered with a small tent as protection against the cruel sun and possible rain. Set-up was rapid and without problems—I actually had a chance to return to the hotel for some relaxation before our 5:30 sound check. The concert was a smash: a large cross-section of the diplomatic, business, and artistic community of Libreville was in attendance, united in their love for jazz—some even had Pharoah Sanders records for Pharoah to autograph. Of course, what most people wanted to know about was the fate of our scheduled public concert tomorrow. Several people offered suggestions on possible sites, and we listened to all of them. Jan agreed to follow up on some of these ideas; we decided that she, Ron, Pharoah, and I would make a group decision on what to do Wednesday morning.

The group began their Wednesday with a press conference at the Intercontinental Hotel, where we were staying. While this was going on, Ron, Jan, and I were discussing our aborted evening program. The only appropriate venue available, a secondary school auditorium, was vetoed as too risky; any educational facility was a likely target for violence under the current tensions, so that was that—we now had an unexpected day off. Never in my 20 years of sound reinforcement had I ever had a gig canceled because the venue was torched—there were the usual "disco inferno" jokes and choruses of Bob Marley's "Burning & Looting", but we were really relieved. We'd almost elected to store some of our left-behind gear at *Omar Bongo Auditorium* during our Sao Tome trip; our schedule change made this unwieldy, thank goodness. We spent the rest of the day relaxing poolside and/or hitting the market—the night found us enjoying African cuisine as guests of Le Perroquet restaurant. Afterwards, some of us hit the clubs for a taste of Libreville night life—which, in typical African tradition, didn't really get rolling until after midnight! Fortunately, we could stay out late, secure in the knowledge that our Thursday flight for Congo didn't leave until 7 PM.

Figure 5. Ngoumi.

Thursday's work schedule involved Ron and me heading for the airport around 3 PM to begin the involved check-in procedures governing all our equipment. The customs checks were much more rigorous now, as was the airport's security. Shows you what a good riot will do. One might think we were exchanging tension for tranquillity traveling to Congo, but it had potential problems too. Brazzaville had experienced riots between supporters of ousted Premier Andre' Milongo and army units loyal to someone else only three weeks before our scheduled arrival. This "show of force" turned bloody, and the resultant turmoil threatened our entire program. Fortunately, the army finally returned to the barracks and, although things were still tense, the overall feeling was that the worst had passed, cooler heads had prevailed, and we were free to come and perform without fear. Libreville-Brazzaville took 1 hour 25 minutes on Cameroon Airlines; PAO Mary Johnson and her staff were there to greet us. Ron took the group over to the hotel, the PLM Mbamou Palace, while I stayed behind to deal with our equipment. I'd expected a hassle, but customs clearance here was perfunctory. Mary had Ministry of Culture clearances for us, which helped, but considering the tense situation of a few weeks ago, I was surprised. I definitely expected a shakedown; just goes to show how quickly political situations can change in Africa. The drive to the hotel was quick: I separated luggage and personal instruments from PA equipment, sending the latter off for safekeeping at the US Embassy. We had a quick briefing with Mary, who informed us that things had indeed "lightened up", and that Brazzaville was eager for our performance, scheduled for Saturday, February 8.

Friday's only commitment was a workshop with local musicians, held that morning at the American Cultural Center and hosted by USIA CAA *Jean-Jacques N'ganga*. I usually don't get to attend these, as typically they are scheduled on a concert day; I go to set up while the group gives the workshop. I'm certainly glad that I attended this one—Congolese jazz musicians were amazed by the facility and imagination of Pharoah, William, Russel, and Greg. We were all impressed at the jazz talent here—and were especially pleased that several traditional instruments were demonstrated for us. I even had several people interested in sound; our exchange was very lively as I answered their technical questions while they, and the local musicians, showed me how they mic'd the *Ngonfi, Ngouomi, and La Sanza* (see Figures 4, 5, and 6). A jam with both modern and traditional instruments was inevitable; it proved so magical that Pharoah arranged then and there to include a similar jam as part of our public concert.

We spent the last portion of our workshop planning how we'd handle that. Once the post-workshop press conference was completed, we had the rest of the day off.

Mary was a paragon of efficiency: a car was waiting for me Saturday at 10 AM, and I was soon at the *Palais des Conges*, a large hall located between the Moungali and Poto-Poto districts of Brazzaville. I wandered into the auditorium, and was quite pleased to find all our equipment waiting in the off stage right wing; there were even two embassy guys there to help me. What I found puzzling was the lectern on stage—there were also several tables flanking this lectern, and people were very busy on stage setting up mics and hanging scrims. What was this all about? "Oh", said the hall manager, "we have a political rally scheduled for today". When I asked how long it would last, I was told 3 PM, followed by the words I dreaded: "no problem". Ron, Jean-Jacques, and I discussed options: under the best of circumstances, set up would take 2 hours, tops. If we could have access to the stage by 5 PM, it would still be possible to do the show. With a firm deadline in mind, we went off to Mary's house, where the band was enjoying an informal lunch with several local musicians, to break the good news. Mary was not a happy camper—she immediately called the Ministry of Culture. I heard "contract", "commitment", and "guaranteed" mentioned several times, with no apparent effect. Whatever arrangements had been made were now overrun by the engine of politics. We were at the mercy of the rally schedule.

Figure 6. La Sansa.

I returned to the hall with Ron and Jean-Jacques around 4:30, to find the party was in full swing. The hall was packed, loud speeches and cheers filled the air, and the huge parking area in front of the Palais was a mass of humanity. We were informed that "our leader's flight has not yet arrived from Paris"; at that point, we knew it was hopeless. Mary arrived at 5 to get the bad news—I waited an extra half hour, but with no change in the rally at that point we officially had to scrub the show. We made arrangements with the hall manager to reschedule the show for Monday, which had been scheduled as our rest day. It was probably just as well—nothing could possible compare with the energy and craziness of a Congolese political rally. Ron and I took a half-hour walk through the crowd outside the Palais and take it all in. Entire villages gathered in giant circles, dancing wildly to drum beats and waiving signs while they did it. Long lines of people paraded through the crowd, banging tambourines and drums; they had the candidates photo stuck to their foreheads. People on tall 10-foot stilts danced to wild music, spinning crazily high in the air while others danced below. It was a *serious* party; cross a US national political convention with Mardi Gras and you have a general picture of what went on in Brazzaville that day. It was yet another first for me: I'd never had a gig canceled because a political rally usurped the venue! The positive side to all this was another night off; no better way to spend it then hangin' on the terrace of the *Kebe-Kebe* nightclub, high atop the *Mbamaou Palace Hotel,* enjoying the creature comforts and the view: a clear tropical night, gazing across Stanley Pool at the twinkling lights of Kinshasa, Zaire, just across the river from Brazzaville.

Figure 7. Setup at the Palais des Congres in Brazzaville, Congo.

Sunday's private concert was held at the US Ambassador's residence. At least here, we'd have no problems—or so I thought. Our stage was the patio behind the house—this was covered with a large canvas tarp. The audience area was covered with a different series of tarps—but there was an open, un-tarped area of about 25 feet between "stage" and the audience. This made for problematical coverage: if I left the speakers in their traditional spot close to the band, very little sound reached the back half of the audience area, and what *did* get back there was lacking any highs. I elected to set one pair of Bose speakers just outside the stage tarp; I set the other pair halfway back in the audience tent, delayed with respect to first pair. I normally prefer to maintain point-source between PA and band; faced with a rear drop off *this* severe, a distributed system seemed to be the only answer. Necessity was definitely the mother here. During our setup, the sky turned threatening; since my Bose house speakers were either outside or near the edge of the tarps, I covered them with plastic garbage bags while still on their tripods. The rain held off though; with Ron's help, we were done early. At this point, we left with Jean-Jacques to go down river for lunch. The views of the rapids were spectacular—we found a place close where we could eat to the roar of the water. We were about half way through when the skies opened up and it just poured. Eventually, the rain slackened, and we prepared to return to our hotel when Jean-Jacques radio began to crackle with the news of "a catastrophe at the residence". We detoured there, to find that the audience tarp had caved in, soaking everything inside and knocking down my delay stacks. The bags protected the speakers from the rain, and except for a bent ¼-in. speaker jack, everything worked just fine. We returned with the group at 5:30 PM for sound check—we hit at 7 PM to a fully-packed tent. My attention to detail on coverage was appreciated—several musicians later told me it was the first time anyone could remember hearing clearly in the rear of the tent.

Monday's rescheduled concert proved to be of the tour's highlight performances. The hall manager at the Palais des Congres was appropriately contrite after Saturday's fiasco—there was plenty of help for me, so setup was the fastest of the tour. The hall seated 1200 in a single seating area with a mild rake (see Figure 7).

Reverb time was less than 2 seconds; our problem here was echoes off the back wall. It was also the only place I'd been in Africa that used Australian-type AC receptacles—the hot and neutral blades are at 45 degrees to a vertical ground blade. Voltage here was 222; I observed spastic fluctuations as high as 8 volts with no load. Most of the stage receptacles were un-grounded; I found one with a good ground on the upstage left wall. We were scheduled to start at 6 PM, but in deference to typical African style, we delayed until almost 7.

The wait fazed no one; the crowd was totally into it, applauding immediately after Pharoah's first solo, and it only got better! Pharoah's cadenza on "Soul Eyes" evoked gasps from the crowd, but the biggest applause went to the mass jam, which featured local trumpet and guitar players along with the *ngonfi* and la *sanza*. The good feeling continued the next morning as we left Congo for Cameroon—it turned out that the customs crew had attended our concert—and they seemed more interested in autographs than in searching our gear.

After one of the fastest check-ins I've seen in Africa, we were on our way—with very good memories of Brazzaville and lots of stories to tell.

AUTHOR'S POSTSCRIPTS

As this issue gets out, I'll be preparing to get out too! Fall '93 will find me touring North and West Africa with the Rebirth Brass Band. At press time our itinerary looked like this:

10/14-21/93 *Cairo & Alexandria, Egypt*
10/22-26/93 *Tunis, Souse, and Fax, Tunisia*
10/27-31/93 *BD, probably Morocco or Niger*
11/01-03/93 *Bamako, Mali*
11/04-07/93 *Coition, Benin*
11/08-15/93 *Accra, Chums, and BD, Ghana*
11/16-21/93 *Dakar, Ziguinchor, and BD, Senegal*

db

Tools Of The Trade

SHELLEY HERMAN

● James Boyk is both pianist in residence, and Lecturer in music in the Electrical Engineering Department at California Institute of Technology (known locally as Caltech), perhaps one of the two most prestigious technical universities in the USA.

As his title states, Jim is both a fine musician and a technical wizard. His technical forte is making good sound better. (He won't even tolerate bad sound, you can do that if you're on staff at Caltech.) Over the years his projects have added to our knowledge of *how* and *why* we hear as we do and he has constructed some awesome machinery to demonstrate his theories.

Several years ago, it occurred to him that microphone placement was pretty important as to how we perceive stereo. He maintains, correctly, that 99 percent of the recordings we hear today are not stereo and to quote him speaking about modern multi-channel recordings:

"It's something like the difference between taking a group picture in the normal way and taking close-ups of each member and then pasting these into a simulated group shot."

In the few real stereo recordings that are made, there are about as many different microphone placement techniques as there are recordists. Jim Boyk's label, *Performance Recordings* has released a CD that is a "must own" for any serious recordist who does stereo recordings.

The CD is titled *A Demonstration of Stereo Microphone Technique*, a recording Jim, and three of his *Projects in Music and Science* course students, conceived, designed and executed. On the recording, there are a total of forty-seven tracks, the first eleven are introductions and explanations. The balance is arranged into eighteen pairs of tracks, on the first track of the pair there is a description of the microphone type and layout with announcer Peter Sutheim speaking into the microphones under demonstration. The second track of the pair has a series of sounds that will allow the listener to evaluate that particular microphone arrangement. The various arrangements cover condenser, dynamic and ribbon microphones in cardioid, bi-directional, and omni-directional patterns arranged in every imaginable geometry including spaced, NOS, ORTF, and Blumlein.

If carried to extremes, there could be Carl Sagan type numbers of recordings and Jim and his students would grow old and still be recording. These eighteen tracks give the recordist the information needed to help decide how the microphones should be arranged at their next stereo session. The CD comes with a very comprehensive 20 page booklet that is aimed at the novice, but has information useful to professional. It carefully explains what is on the disc and how to use it.

It would not be constructive to actually review this disc, as it is a teaching device and not a performance, but for recordists, both new and experienced, and it is certainly a valuable **Tool Of The Trade**.

Another Jim Boyk project was a device he called *Magnesaurus* (tm) long before Jurassic Park raised its toothy head. The model Ampex 351-2 tape recorder, manufactured by Ampex from about 1952 until about 1965, was the mainstay of most recording studios until the middle 1970s. As solid-state devices became quiet and reliable enough for professional audio, these venerable tube devices were first delegated to making copies, then editing, and finally, ignominiously, some of them became tape winders. Jim was able to obtain a very clean 351-2 and proceeded to do what the hot rodders call blueprinting.

The deck was completely disassembled, every moving part was re-bearinged, the motors were rebuilt to higher tolerances than ever specified by Ampex, then balanced. The head assembly was replaced by a half-inch two-track assembly on a special mount conceived by John French of JRF Magnetics, and finally the tape path was carefully tuned with the use of some very sophisticated Hewlett-Packard test equipment.

Meanwhile, back at the electronics, Steve Haselton of The Mastering Lab was busy designing an external power supply to reduce hum and noise from that source. Steve also tweaked the electronics for maximum performance.

The first *Magnesaurus* (tm) project was a recording of Jim playing Mussorgsky's *Pictures At An Exhibition* in Caltech's famous Dabney Hall. Using Jim's favorite Coles ribbon microphones and some custom modified BBC preamplifiers, the signal was then sent about a quarter of a mile to Jim's lab where the tape recorder was located. The author was at the controls and fortuitously had brought a Panasonic 255 portable dat recorder, and so he made a simultaneous digital recording.

Although the *Magnesaurus* performed perfectly having the digital backup gave Jim an idea, and in the released recording the CD has both the analog recording from the *Magnesaurus* and the digital recording from the Panasonic 255, allowing the listener to compare the two recording methods. The vinyl disc has only the analog recording.

Although not conceived as a **Tool Of The Trade**, these recordings are a valuable lesson in the differences between analog and digital. Additionally, the music and performance are excellent, and you are liable to enjoy yourself while listening to the recording.

Both recordings are available from:

Performance Recordings,
135 Holmby Ave.
Los Angeles, CA 90025-5915,
(510)-475-8261.

db

Cover Story

ADRIAN ZARIN

A Revitalized Record Plant Celebrates Its 25th Year

In September of this year, a rock and roll legend celebrated twenty-five years in the entertainment business. Unlike some of the graying, gracefully aging performers who also got their start in 1968, the Los Angeles Record Plant truly looks younger and fitter than ever.

IT STANDS ON A DIFFERENT SITE THAN it originally did and is under the direction of a new owner. But the spirit and overall concept of this recording industry institution remains the same: It is a state-of-the-art living room where the top musical talents of our time come to play. (Some call it work.) Over the years, the Record Plant has become a home not only to recording artists but also to the audio cadre of the film and television industries. In an economic climate where many legendary studios have closed their doors, the Los Angeles Record Plant is a diversified, flourishing business.

Walking through the stylish, bustling facility that is today's Record Plant, it's hard to believe that the studio was in grave peril just three years ago, with industry insiders predicting an imminent closure. As it turned out, however, that sad fate was not to be.

The person responsible for the Record Plant's rescue and triumphant reversal of fortune is an energetic, affable man named Rick Stevens. Formerly president of the *Summa Music Group* publishing firm and a veteran of the artist management and A&R fields, Stevens and his long-time Wall Street partners, Tom Kirch and Michael Beder, purchased the facility from its former owners in June of 1991. The new owners immediately embarked on an exhaustive, four-million dollar studio upgrade and expansion. "Record Plant Phase Two" added two new state-of-the-art studios to the premises: a mix suite equipped with a 96-input SSL SL-8000 G Series console and a large recording/scoring room boasting a 96-input Neve VPSR Legend console.

These new studios joined Record Plant's two previously existing rooms to form a deluxe working environment complemented by a light, bright new atrium area complete with Jacuzzi, billiards table and all the other requisites of a comfortable, lively social area.

Completed under the supervision of architect Peter Grueneisen of the Los Angeles design firm *studio bau:ton*, the upgrade quickly re-established the Record Plant's preeminence as one of Los Angeles's hippest recording spots. Not content to rest on his laurels, Stevens recently added a fifth room: an overdub/pre-production/MIDI suite that mates a vintage API Di Medio console with both analog multi-tracking and a Pro Tools digital audio workstation.

"The Record Plant has historically catered to top-echelon music stars," says Stevens. "I very much wanted to continue in that tradition because that's the only segment of the studio business where you're competing on the basis of client service rather than rates. So many studios in the Los Angeles market are competing to offer slashed rates right now, and that's why many of them are struggling. I just realized that most stars come down a notch in their life when they walk into the average recording studio. If they're in from New York or London, they're staying at the Beverly Hills Hotel or the Mondrian or Bel Age, where they're used to a high caliber of service. I wanted to bring that same level of service into the studio. That's what makes us different."

A BRIEF HISTORY OF THE RECORD PLANT

Stevens' operating philosophy carries on in the tradition established by Chris Stone, the Record Plant's founder and former owner. Together with the late Gary Kellgren, Stone opened the first Record Plant in New York, in 1968. The Los Angeles Record Plant was built the following year. Right from the start, the concept was to provide an environment at once more hip and more homey than what was to be found at the label-owned studios and old-style independents of the Sixties. Jimi Hendrix' groundbreaking *Electric Ladyland* album was the Record Plant's in-

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Adrian Zarin is a Los Angeles-based free-lance writer and musician.

augural project. With its technological adventurousness and pioneering FM rock vibe, this landmark recording site set the pace for the Record Plants' future. From Frank Zappa and the Velvet Underground in the 60s, to the Eagles and Stevie Wonder in the 70s, to Tears for Fears, Whitesnake and Bruce Springsteen in the 80s, the Record Plant grew by leaps and bounds. Chris Stone had assumed sole leadership of the business after Gary Kellgren's death in 1977.

In the 80s, the Los Angeles studio became the focal point of Stone's Record Plant enterprises. In 1986, he closed the original studio site on Third Street and opened the present-day Record Plant on Sycamore Avenue, in a district that houses many of Hollywood's leading film facilities. The building chosen for the new Record Plant was originally an annex to the historic Radio Recorders. Stone converted the structure into a top-flight, two-studio facility equipped to do a diversified mixture of record and film work. In 1989 Chrysalis Records purchased the Record Plant from Stone and operated the studio up until its 1991 purchase by Rick Stevens.

Figure 1. Pride of the place in the new SSL 1 Studio Suite is this 96-input SL-8000G.

"It quickly became evident that a major renovation was in order," says Stevens. "You couldn't just buy this place, upgrade the service and expect the Record Plant name to mean what it meant ten years prior. So right from the start, our business plan was not just to acquire the Record Plant, but also to implement the now completed improvements."

The first priority was to restore the existing rooms, which had undergone slight modification during the Chrysalis years. The Record Plant was closed for a three-week period for what Stevens calls "cosmetic and substantive improvements: One of the first things we did was bring in [noted acousticians] George Augspurger and Steve Brandon to take a look at the acoustic signature of the rooms and do some fine-tuning. Another thing we did was upgrade the look of the complex and create new, more luxurious private lounges for each of the studios. For a facility that invented the concept of the private lounge, this place was sadly devoid of creature comforts when we took it over."

With these immediate concerns satisfied, attention turned to Stevens' master plan: converting 11,000 square feet of office space at the rear of the building into two new studios and a large common lounge area. An important design objective was to make each of the new studios a completely self-contained work-space, each incorporating a private lounge equipped with a kitchenette and bathroom. This way, projects could proceed in complete isolation from other activities around the studio complex. The design contract for the upgrade was awarded to Peter Grueneisen of *studio bau:ton*. Stevens credits *studio bau:ton's* contractors, Pridemark Inc., for the fact that the Record Plant's two original studios were able to remain open during most of the construction, hosting sessions with artists such as Mick Jagger, Michael Bolton, the Black Crowes, and the Red Hot Chili Peppers. When the time came to build the new Mini Plant

Figure 2. The other new studio, Neve 1, houses a 96-input Neve VRSP Legend, equipped with GML Moving Fader automation.

pre-production/overdub/MIDI facility, the talents of Grueneisen and *studio bau:ton* were again enlisted.

"We wanted the Mini Plant to maintain the acoustic and visual style of the other new rooms at the Record Plant," Stevens comments. "In its look and sound, it's very much a smaller version of our new SSL room."

A WALK THROUGH THE RECORD PLANT

The social and aesthetic heart of the Record Plant is its spacious atrium lounge, located at the rear of the building. Here, clients can get away from their projects and mingle with other clients beneath a large skylight which admits the plentiful California sunshine. Meals, rides and other outside amenities can be ordered at the client service desk. There's also a fully equipped kitchen, billiards table and of course the Record Plant Jacuzzi. This enduring monument to recording studio hedonism is enclosed by dramatically sloping glass walls fitted with mini blinds which can be drawn for privacy. "The Jacuzzi at the old Record Plant was somewhat a symbol of the place," says Peter Grueneisen; "Rick wanted to bring that back." Grueneisen adds that the new Record Plant operates on a "two-tiered lounge concept." Clients can stay in their private studio lounges—"their own little kingdom," as the architect puts it—or come into the common lounge to mingle with people working on other projects.

The Record Plant's four studios are located along the long corridor that runs from the front entrance back to the atrium lounge. As you enter the building, facing the atrium, you'll find one of the original studios on your left. Now called Neve II and equipped with a 60-input Neve V Series console with Massenburg automation and a custom film monitor module. Neve II contains the facility's largest recording area, a 40-ft. by 35-ft. space with a 25-foot high ceiling which was originally designed to handle scoring dates with as many as forty players.

There's also a 21 by 21-foot "Super Live Room" with a 14-foot ceiling and a 6 by 8-foot ISO booth with a 10-foot ceiling.

Across the corridor lies the other room that acoustician Tom Hidley built for the Record Plant in 1986. Now called SSL II, its 14 by 22-foot tracking room is smaller than Neve II, although its control room is identical in size: both are 30 by 20 feet.

The studio is equipped with a 72-input SSL G Series console with a G Series automation computer and a combination of E and G Series EQ modules. Both SSL II and Neve II have 25-inch video monitors in their control rooms and projection TV systems in their tracking rooms.

The two new studios—Neve I and SSL I—are also on the right-hand side of the corridor, more toward the rear of the building. Let's go in for a closer look, shall we?

Figure 3. A view from the front of SSL 1. It shows the well-equipped outboard rack situated behind the engineer's mix position.

SSL I: DESIGN AND TECHNICAL FACILITIES

SSL I, an mix/overdub suite, was the first of the two new studios to be completed. Prince was the room's inaugural client. The studio's centerpiece is a 96-input SSL SL-8000 G-Series console with Ultimation, one of the two largest SSL consoles ever to be installed in an audio facility. SSL I's control room measures 23 by 20 feet and is supplemented by a 14 by 12-foot vocal booth.

The design of the room is based in part on the inhouse mix room at Summa Music, Rick Stevens' former company.

"We knew we wanted the console to be very close to the front wall by conventional standards," explains Stevens. "That is a formula we just lucked onto in the Summa mix room because space was limited. But the room became a very successful facility when we opened it to outside clients, so we decided to stay with that basic Summa formula in SSL I. With the board that close to the front of the room, we knew the large room monitors would be functioning almost as giant near fields. We wanted the front of the control room — say up to the outboard gear rack—to be very neutral acoustically."

Implementing this idea took a certain amount of design finesse, since SSL I is significantly larger than the Summa mix room. Also, a 96-input mixing console is a rather lengthy piece of gear, so it was necessary to maintain a fairly wide "sweet spot" in the main mix position. Peter Grueneisen worked with George Augspurger and Steve Brandon in designing and fine-tuning the room acoustics. What worked out best, says Grueneisen, "was to have a soft front wall with fabric on it and RPG diffusers in the rear wall. Monitor placement was critical too."

The room's Augspurger monitors are mounted in wooden soffits. "We made soffits that were bigger than the actual speaker enclosures," Grueneisen explains. "This way, we had room to adjust the position of the monitors. Once we found precisely the right position, we filled in the gap around each monitor with sand and closed up the soffit. The sand really helps anchor the monitor and helps provide isolation. There's also lead around the loudspeakers, a heavy layering of plywood, a material called MDF (medium density fiberboard) and lots of drywall. By making the speaker soffit larger than the speaker we not only provide leeway for positioning the monitor with maximum accuracy, we also give the clients more options for the future."

The video facilities in SSL I are no less impressive than the audio gear. The front wall of the control room houses three video monitors. The central monitor, a 50-inch rear-projection Mitsubishi 5017-S is mainly intended for mix-to-picture applications. It is flanked by two smaller Mitsubishi AM-2752 video monitors which are set up to provide a variety of displays—everything from local cable TV programming to Ultimation and Macintosh display screens. Many of these functions can also be displayed as a picture-in-picture window in the lower right-hand corner of the big 50-inch monitor. A video switcher located at the console allows any one of four different inputs to be routed to the monitor, including a cable box, VHS VCR, laserdisc player, a ¾-inch U-Matic VCR and other video devices. Alternately, the monitor can be switched to RGB mode, and used as a computer screen for the SSL Ultimation or other displays.

NEVE I: DESIGN AND TECHNICAL FACILITIES

Neve I, the second new room at Record Plant is designed to accommodate a wide variety of tracking, scoring and mixing projects, including mixes to video. "Knowing we were going to wind up with the largest Neve on the planet, we estimated that probably 70 percent of our business in that room would be mixing," says Stevens. "But because it is a Neve board, we knew people would want to track in there as well; we wanted to accommodate that kind of clientele too."

Responding to the need for a versatile, good-sounding tracking space in Neve I, Peter Grueneisen designed a 24 by 35-foot room with a sloping ceiling that reaches a height of 22 feet. Room acoustics are fairly live. "That's the way we like to do all our rooms," says Grueneisen. "We feel that's the trend. Also you can always deaden a live room very easily by using drapes. But when a room has been built dead, it's very hard to make it live."

A large portion of the walls are surfaced with wood panels: "A substance called strand board," Grueneisen elaborates. "It's a very economical material that we treat in a special way by sanding and then staining it. And then we used concrete acoustical blocks in the back corners of the room. Those are raw concrete blocks with slots in them that make them act as acoustical resonators."

Concrete acoustical blocks were also used in the studio's 14 by 11-foot vocal booth (located off the control room) and to construct two small ISO chambers in the two rear corners of the room for micing guitar and bass amps. Details like this, along with the overall room acoustics, make the Neve I tracking room an ideal space for cutting rock bands as well as small to mid-sized orchestral ensembles. To achieve the tracking room's lofty 22-foot height, it was necessary to take out the existing roof trusses and raise the roof by about five feet. According to Peter Grueneisen: "since we were raising the roof, we decided to construct a second-story lounge for Neve I and install a window so that the lounge overlooks the Jacuzzi and the whole atrium area."

The focal point of Neve I's control room is its 96-input Neve VRSP Legend console with GML Moving Fader automation and recall. The Legend features a custom-designed combination of Neve's Scoring and Post-production User panels—hence the "S" and "P" designations. Minor modifications were also made to the board to provide an optimal interface with other equipment in the room, including five full rows of patching to accommodate various configurations of tape machines and outboard gear. Neve's VSP section, which handles output bus-to-speaker assignments for film mixing, also added a few more patchbays to the board.

"The great thing about the Neve VRSP Legend and the new SSL SL-8000," confides Rick Stevens, "is that either can easily handle discrete film-style or Dolby surround-sound mixing. The engineer

Figure 4. A close-up detail of part of SSL 1's well stocked outboard rack as seen in Figure 3.

Figure 5. The back of each outboard rack is fitted with audio, video and MIDI tie-lines. This allows processors and musical equipment to be used free-standing on the table top surface.

just pushes a button, and every bus is connected the appropriate speaker channel. They don't have to fuss with re-patching an LCRS (left-center-right-surround) system. The console automatically covers all of that re-assignment for them."

Video monitoring in the control room is provided by two Sony PVM-2530s. Unlike SSL I, however, it wasn't necessary to install an extensive video monitoring system. Because Neve I was designed to function as a conventional recording studio, most of the front wall is taken up by a conventional control-room window. Two monitors mounted in the wall are dedicated for video playback. Since the GML automation and recall systems feature their own monitors and high-resolution graphics, these video monitors are not required to display computer information, as in the case in SSL I.

Rick Stevens explains that "the acoustic signature of the Neve I control room is much more like the existing Hidley rooms in the Record Plant. "But it's a smaller control room at the front," he adds. "Consequently, it's somewhere between the style of SSL I and the style of the existing Hidley rooms."

"One difference is that the front wall is plaster," adds Grueneisen. "So it's a hard, reflective surface rather than a soft one like in the SSL I control room. Also it has more of a soft ceiling in the back, as a result of the harder front wall. So some of the details are flipped, but the goal is the same: to maintain a large reflection-free zone."

THE FACILITY-WIDE PERSPECTIVE: HARDWARE

Both SSL I and NEVE I feature identical custom George Augspurger monitor systems consisting of left, center and right enclosures. The enclosures are loaded with TAD components: two 15-inch drivers and a 2-inch horn. The high-end components are driven by Boulder 500 power amps. Australian 1K2 1.5 kilowatt power amps are used on the low end. A modified White Instruments 4000 Series ⅓ octave EQ/crossover completes the system. JBL 8330s are used as surround sound monitors for the rooms' Dolby stereo matrix encoding units.

The two older studios (Neve II and SSL II) have the exact same power amp/crossover/room EQ configuration, installed during the initial three-week upgrade of the Record Plant's facilities when Stevens acquired the studio. In the older rooms, however, the amps drive custom Hidley/Kinoshita monitors and JBL 4312 surrounds. In all four studios, Yamaha NS-10Ms and Auratones are the standard near field monitors; KRK, Tannoy and AR near fields are available on request. The smaller monitors in each studio are driven by Krell stereo power amps.

The Record Plant boasts an exceptional selection of microphones and outboard equipment, including a fine cache of vintage gear.

The standard tape machine complement for each studio consists of two Studer A800 MkIII analog 24-track machines, vintage Ampex ATR 102s and 102s (half-inch and quarter-inch), two Panasonic SV-3700 Pro-DAT machines, two Panasonic VHS Hi-Fi VCRs, two Nakamichi MR-1 cassette decks and one Rotel 955AX CD player. The Record Plant has Sony PCM-3348 digital 48-track machines, a Mitsubishi X880 digital 32-track and several Studer A820 analog two-tracks that are allocated to the rooms on a floating basis. Additional digital multi-tracks and other ancillary machines are hired in as needed. The machine bays in each studio are designed to accommodate up to two additional 48-track machines without having to disconnect any of the standard tape machine complement.

Synchronization is provided by upgraded Timeline Lynx II modules (one per machine). The console's on-board machine controls are used to drive the tape machines, which then simply chase-lock via the Lynx II units. But each room has also been pre-designed to run with other master control systems, including the Lynx Keyboard Control Unit., as well as any system that operates via an RS422 interface bus, which is provided throughout the complex.

The Record Plant boasts an exceptional selection of microphones and outboard equipment, including a fine cache of vintage gear. Much of it comes from the stock of Livingstone Audio, the old Record Plant's satellite rental company. On acquiring the Record Plant assets, Stevens decided to distribute the Livingstone wealth among his four rooms. Thus the equipment rack in each studio comes with a generous selection of Pultec EQs, Fairchild limiters, UREI Los Angeles-2As and 1176s, as well a good supply of modern gear including Drawmer gates, GML 8200 EQs, Lexicon 480 SLs and PCM 70s, Eventide H3000s, AMS RMX16s, dbx 902s and Yamaha SPX 90s, 900s and 1000s. Other pieces, such as the Record Plant's EMT 250 and 251 and its two Fairchild 770s float from room to room.

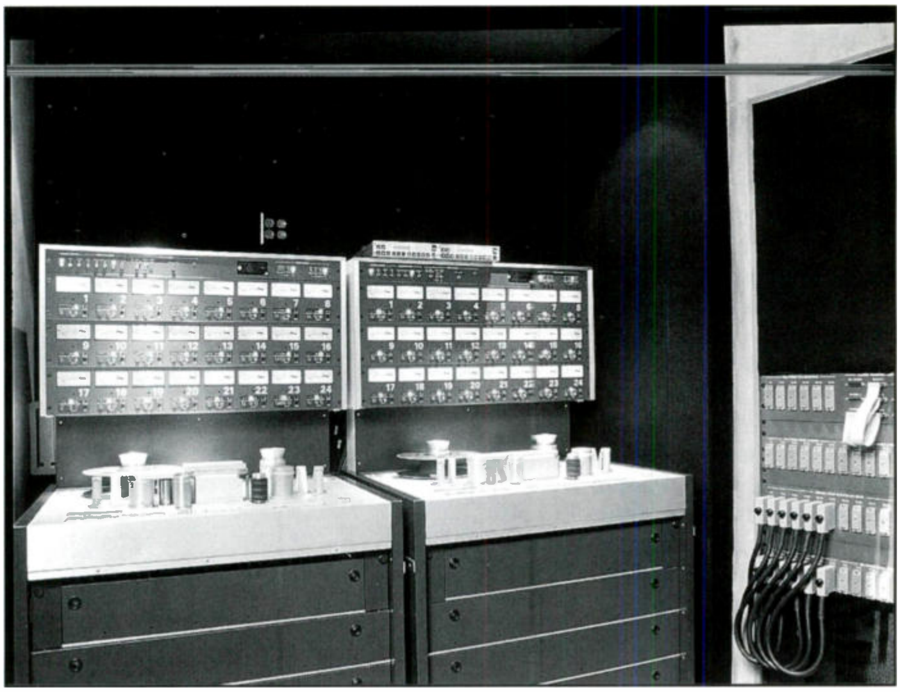


Figure 6. Each tape machine soffit features pre-wired multi-way connectors that enable tape machines to be installed easily and quickly.

THE FACILITY-WIDE PERSPECTIVE: STRUCTURAL AND INTERFACE DESIGN

Optimal physical isolation between studios with maximum electronic inter-connectivity: these were two major design goals when the Record Plant was refurbished. The task of isolating the studios acoustically was part of Peter Grueneisen's brief.

"One of the big logistical problems was that the studios are all so close to one other," the architect notes. "Three of the rooms are only separated by walls. There are no corridors in between. So we had to build very heavy walls with many layers between the studios. One of them is made out of five independent wall structures, each constructed of wood framing, metal studs, plywood, lots of leaded drywall and plenty of air spaces. Each layer floats on rubber mounts. There are no hard connections anywhere: no screws or bolts."

Grueneisen also deemed it necessary to take out the concrete floor slab in the rear portion of the building to avoid the need for cumbersome access ramps. This offered an acoustical advantage, since it enabled the architect to design floating floor slabs for both the new studios, decoupling them from the other rooms for further acoustic isolation. In addition, adds Grueneisen, "the air conditioning was redesigned from scratch for the back part of the building, so that there are individual units for each of the rooms. It's all designed to minimize transmission of sound from room to room. There were some cases where we had to put two rooms on one air conditioning unit; but in those instances, we put in muffler boxes and an elaborate system of ducting to make sure no sound travels from one room to the next."

Figure 7. Neve 1's recording and tracking area is large and airy. A voice-over booth, and main Studio Suite entrance is to the left.

A primary design goal was to achieve maximum "plug-and-play" compatibility throughout the five rooms, allowing audio, video and synchronization hardware to be patched into any of the studios, and be ready for action with a minimum of hookup time. Each studio is equipped with two machine bays, each with ample ALCO connectors for patching in multiple audio and video tape machines over and above the room's stock machine complement.

Interfaces for the new rooms were designed to conform with the existing two studios. Although each room is completely self-contained, audio and video tie lines are provided between them, as well as machine-control tie lines. In this way, each room can be interfaced without additional noise and ground-loop problems. To satisfy the latter requirement, a star grounding system ensures that all equipment is isolated. A separate clean ground can then be connected, if necessary, to the console in each room. So far that hasn't been necessary, however, since the AC grounding scheme is well-distributed.

And although the Record Plant is—as its name implies—primarily a record house, it is also equipped with mix-to-video facilities for those record clients who might have to do special mixes of their music for film soundtracks, videos, etc. For this reason, distributed house video sync is provided in each room to ensure a clean, consistent sync source for the various time-code synchronizers, digital tape machines or video decks used during a session.

Figure 8. A close-up view of the Neve Legend's center area. A VSP Section handles complex speaker configurations for film mixing

THE MINI PLANT: A TIMELY AFTERTHOUGHT

With the massive Phase II upgrade brought to a successful conclusion, it would seem that the Record Plant was complete. But Rick Stevens had another idea. When some second-story space became available in '93—it had formerly been let to a tenant whose lease had expired—Stevens knew just what to do with it. He wanted to provide a place where clients could do pre-production, overdubs, MIDI sequencing and other "off-line" work at a slimmer rate than that commanded by any of the Record Plant's other rooms. Construction of the Mini Plant began in late July and was completed within a month.

The studio features a control room and overdub area. Mixing is provided by a 36-input custom API Di Medio console that formerly belonged to Steven's Summa Music Group: "The console was originally built for Paramount Studios in the early 1970s by Frank Di Medio," explains Stevens. "When Summa bought the board in 1989, we brought Frank out of semi-retirement to restore it: both in terms of electronics and cosmetics. It's quite a handsome piece of vintage gear."

The control room offers near-field monitoring only, via a selection of Yamaha NS-10Ms, KRK 9000s, Westlake and Tannoy monitors powered by either Studer, Krell or Yamaha amplification (the client's choice). Tracks can be laid onto a Studer A800 analog 24-track, or to the room's Digidesign Pro Tools eight-track digital audio workstation with Sound Designer II editing software. The system is based around a Macintosh Quadra 800 with 32 meg of RAM and a 230-meg internal drive. Its twin 1.7 gigabyte external hard drives and 8 mm tape backup system are by Pacific Coast Technologies (PCT). Visual display is provided by a 20-inch monitor, and a Roland CS-10 virtual controller can be used by clients who prefer the hands-on feel of faders and buttons to the feel of a Mac track ball. Apogee A/D and D/A converters are used at the input and output stages of the system.

"We made a very conscious decision to combine the warmth of an old API console with digital audio technologies," says Stevens. "We think it makes for a powerful combination."

A second Macintosh—a Centris 650—handles MIDI sequencing and editor/librarian functions. It too has 32 Mbytes of RAM and a 230 Mbyte hard drive. Resident on the drive are Mark of the Unicorn's Digital Performer sequencer/digital recording program and Unison editor/librarian, Opcode's Studio Vision sequencer/digital recorder and Galaxy librarian. (All these programs are installed on the Quadra 800 as well.) The Mini Plant's stockpile of MIDI gear includes a Roland A-80 keyboard controller, JD-990 and D-550 tone modules, a Korg Wavestation AV, E-mu's Proteus 1, Proteus 2 and Vintage keys, Alesis IV drum module, and Akai S1000 sampler with 18 Mbytes of RAM and a Dynafex CD ROM player with library.

The Mini Plant is tied into the audio and video network that links the studio's other four rooms, as Stevens explains: "We've created a system where you can basically do a D-to-D transfer into Pro Tools from a digital machine in one of our downstairs control rooms. In Pro Tools, the tracks can be edited and otherwise manipulated and then sent back down to the multi-track, without ever leaving the digital domain."

Rick Stevens speaks with discernible pride as he explains the ins and out of his newly invigorated facility. The venerable Record Plant is clearly ready and eager for the next 25 years. db

CONRAD COOKE

Packaging, Packaging, Packaging

"That's what seems to be the deciding factor as to whether a product is successful in the market place, or whether it bombs—simply because it, (whatever "it" happens to be) doesn't have the right look. This is a common problem for both individuals and small companies who don't have the capital to sink into fancy or "slick" promotional packaging."

We are living in the MTV generation, where hi-energy, fast-paced advertising and imagery are bombarding us with unrelenting hammering. Our brains have been conditioned to absorb unbelievably complex visual and audio messages, and so we find ourselves slipping into the head-space that "more" of something is better than "less" (as opposed to the old adage that "less" is "more"), and that "bigger" is "better."

When packaging a product (any product), one has to find the tricky balance between "done" and "overdone." Mis-representation or poor visual representation of a product can completely kill that product's chances for success. Remember the oriental train of thought for preparing a successful meal: Fifty percent is feast for the mouth, and fifty percent is feast for the eye. When it comes to putting together an audio cassette package, this same thinking holds true. I was visiting the Long Island NY home of Ariel Music Design and talking with Jim Becher, its owner.

"Even before an audio cassette tape is listened to, it is looked at. It is this first glance which plays an important part of how the tape is perceived; that is, whether or not the cassette tape was professionally recorded and packaged. If the tape does not look good, it can count as a strike against an individual or individuals who are attempting to shop the tape to record companies and music publishers, or selling and distributing their product to the masses. It is very important that the audio cassette tape package looks finished, and not like a work tape or rough draft.

"The quality of the audio recorded on the tape may indeed be outstanding, but if the same level of "quality control", or level of integrity from sound to sight, so to speak, is not maintained throughout the entire presentation, the overall impact of the presentation may suffer,"said Jim.

THE END JUSTIFIES THE MEANS

He went on, "When all is said and done, the only thing that matters when listening to a tape, record, or compact disc is whether or not the end result, that is, the sound coming from the speakers is pleasing to the ear.

More times than not, a lot of time and expense is given to creating a master product in the recording studio, which includes mixing onto dat (digital audio tape), only to have the final mixes get "lost in the translation.", when making multiple audio cassette runs.

What is the point of going to all the trouble to make such a wonderful sounding master tape, and then making horrible, muddy cassette dupes? It doesn't make sense.

The final mass-produced version of the master tape *has* to sound as good as the original dat master. The most difficult format for mass-duplication to make sound as good as the original dat master, is the cassette."

ENTER ARIEL MUSIC DESIGN

Jim, in order to explain Ariel Music Design, tell us first where you come from and how it started. This paragraph following is his biography.

James Becher is a multi keyboardist/producer/engineer who has worked with such notables as Jon Anderson (Yes), Verdine White (Earth, Wind, and Fire), Ed Gagliardi (Foreigner), Kenny Aronson and Thommy Price (Billy Idol), Meatloaf, and the Weather Girls. He also served as product specialist and keyboard clinician for Music Technology Inc., and was involved with product development and sound design for Korg's M1 - MIDI Workstation electronic keyboard. He is currently the owner of Ariel Music Design, a full service music production company.

Jim then began, "Making the ultimate hot tape copy was a quest of mine for some time. I've succeeded, and I'm happy to be able to make the finest sounding audio cassettes—anywhere, and to be able to offer my services to the public. Ariel Music Design is a complete music production and recording service, with two facilities—a recording and music production studio, and a fully equipped, state-of-the-art cassette duplication facility. I would like to talk with you, and just give a "Cook's tour" of the cassette duplication facility." (Editor's note: A similar tour of his recording facility is in the January/February issue of **db Magazine**).

KEEP IT CLEAN

I asked him how do you keep things clean?

Conrad Cooke is a free-lance writer, base on Long Island, His response,"One of the most important rules in doing mass tape duplication is keeping the noise floor low. I say this because many tape duplicators use a method of wiring the tape decks together known as *daisy-chaining*. This simply means that the audio signal from the master dat machine is routed to the input of the first tape deck, and the output of the first tape deck is fed into the input of the second tape deck. The output of the second tape deck is then fed into the input of the third tape deck, and the output of the third tape deck is sent to the input of the fourth deck, and so on. The audio signal may or may not be split at some point in the chain using "Y" cords, but the output cords of the "Y" splits are still fed to the tape decks using the daisy-chaining method. The major problem with this method of signal routing is that cumulative noise build-up is experienced, as each tape deck in the chain receives the audio signal *plus* the summed inherent amplifier noise in each of the previous tape decks. This also means that not all cassette tapes will sound the same; some will be noisier than others. At Ariel Music Design, we use high-quality distribution amplifiers to evenly split the audio signal so that every tape deck is fed the exact same signal.

The reason for the use of distribution amplifiers is because we are running fifty hot-wired modified Tascam cassette decks (which run in real-time, and it is not possible to split any audio signal that many times without the use of some type of line-level dats. The distribution amplifiers we use are Rane model SM26's. They are unbelievably quiet. Another key factor in keeping the noise floor low, is our use of the Mackie 1202 and 1604 mixers. The hi- and low-shelving equalization controls have center frequencies that are much more usable than most mixers. The hi-center frequency on the Mackie mixers is shelved at 12.5 kHz. (as opposed to 10 kHz. on most other mixers). This is a perfect frequency to add a little sparkle or crispness to the cassettes when the master dat for whatever reason was mixed a little "dark" (There is perhaps an even more important other reason for adding a touch of 12.5 kHz to the

Figure 1. Computers, and some of the stacks of slave cassette machines at Ariel Music Design. Note the U shape of the work area.

cassette tapes, which I will talk about shortly). Ten kHz tends to be a little "ratty" sounding when used brighten-up program material. The low shelving frequency on the Mackie is set at 80 Hz. (instead of the usual 100 Hz on most other mixers). Again, this seems to be a much warmer frequency for adding low frequencies than 100 Hz,, because 100 Hz can tend to make the bass sound *boomy* instead of *sweet and low* (sounds like the name of something...doesn't it?...er,...umm...). Oh, did I mention that the Mackie series mixers are absolutely the quietest mixers around, and have been compared to Neve?"

JUST A TRIM PLEASE

How do you handle the master when it comes in, particularly if it is not quite perfect. Said Jim, "There is nothing in tape duplication that is more important than having multi-band equalization. Having the ability to take off a little gain on only one band of frequencies, or to add a little gain to another is one very important facet of making smooth sounding tapes. Precise control over every aspect of the sound allows us to fine-tune the music or voice as it goes onto the audio cassettes. We use a pair of Alesis MEQ 230 dual thirty band graphic equalizers to address the individual frequency bands. The reason two are used is because one stereo equalizer is used to control the overall shape of the output signal, and the other equalizer is used as a control signal which is fed into the side-chain input of a model 3630 dual channel compressor/limiter/gate also made by Alesis. Again, the Alesis equipment was chosen because of the quiet, *transparent* way in which it processes the audio signal. The Alesis model 3630 compressor/limiter/gate has a very gentle way of limiting the loud, transient peaks that some recorded material contains.

The RMS mode of compression/limiting using *soft-knee* compressing/limiting handles overall program material much better than peak *hard-knee* compression/limiting, so therefore the action of the limiting is not noticed by those listening to the cassette tapes. Simply put, this means that because there are no out-of-control bursts of music (or whatever), to distort and saturate the tape, a much hotter recording level can be achieved, thus greatly improving the signal-to-noise ratio. The noise floor is therefore much lower than it would be if no limiting was used."

THE PEAK OF PERFECTION

Do you get perfect masters? Jim replied, "It's hard to say just what the perfect recording environment is or which recording studio has the best sound. One thing is certainly true; music and voice recording is a greatly varied medium, and barring an obvious screw-up or terri-

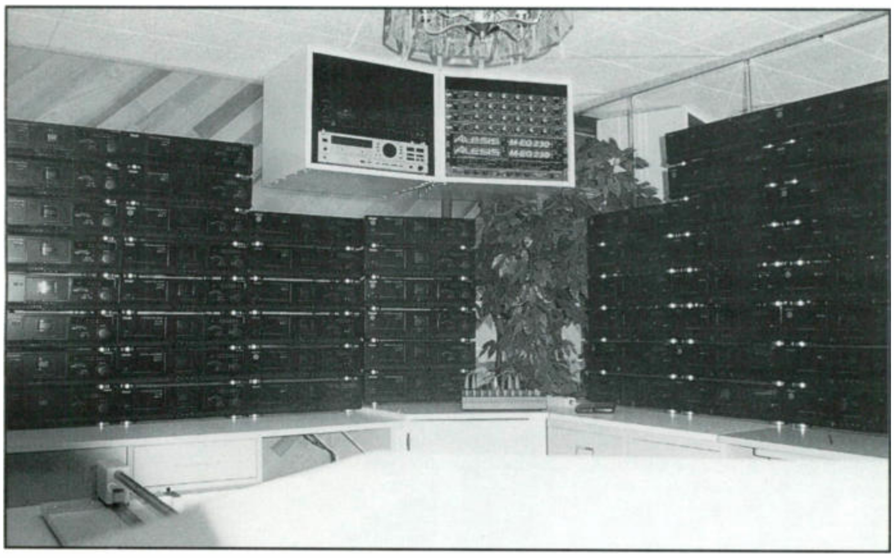


Figure 2. A close-up of the slave setup. Note the overhead equipment rack.

bly recorded master tape, it is a completely subjective issue. Because there are those individuals who prefer very tame mixes, and those who prefer a slamming recording loaded with tons of upper mid-range and high frequency content, at some point, selective frequency-dependent limiting rather than overall frequency broad-band frequency limiting may become necessary.

For example, if the program material on the master dat is hard-rock or heavy-metal rock music, there may be an excessive or at least emphasized amount of 5 to 6 kHz frequencies present which, when copied for mass cassette tape duplication, may distort or saturate the cassette tapes. Audio cassettes tend to be not terribly forgiving when it comes to recording music or voice with lots of highs, especially if there are sudden sonic peaks in the program material. With the use of frequency dependent limiting, 5 kHz and/or 6 kHz for example, may be the only frequency bands that are limited or even squashed. The limiting of the dynamic range of these frequencies allows a much hotter recording level without tape saturation or distortion. This method of specific frequency band limiting is achieved by splitting the signal as it comes out of the mixer into two feeds. The first is allowed to pass through unaffected into the first MEQ 230 equalizer, the 3630 compressor/limiter/gate and then to the distribution amplifiers, which, of course, feed the signal to the tape decks. The second feed is patched into the input of the second MEQ 230 equalizer. The audio output of the second MEQ 230 is then patched into the side-chain input of the 3630 compressor/limiter/gate. By kicking in the second equalizer, any boosting or cutting of any of the frequency bands of that equalizer directly affects the action of the compressor of the 3630 because it is those specific frequency bands *only* which the compressor sees. The second MEQ 230 essentially acts as a control signal to the 3630 compressor/limiter/gate.

"The gate portion of the 3630 compressor/limiter/gate is as smooth working as they come. With a smooth preset attack setting and a variable release time, the gate may be used in a number of recording situations to clean-up noisy dat masters during silent portions of the tape, or the blank space between song selections."

GETTING EXCITED

I asked, what works when you get a less-than-perfect master? Jim's answer,"Sometimes it becomes necessary to "fix-it-after-the-mix", when, for whatever reason, the music or voice doesn't jump out of the speakers and confront the listener. This is when the use of an aural exciter may be just the answer. The perceived effect of aural exciting is different than that of just boosting high frequencies with equalization. Aural exciting gives not only high-frequency presence, but adds more definition to the original program material. At Ariel Music Design we have the option of using two different aural exciters. Each has its strengths in different areas of dealing with the audio spectrum. Both aural exciters we use are made by Aphex. The first which is the model Type C, is the simpler of the two. The Type C is useful when the program material feeding it is a little dark or muddy and just needs a tad more presence. However, the Type C is sensitive to high frequency transients and may distort if any high frequency transients are detected. If that is the case, the second and more elaborate exciter which is the Type III is the exciter of choice. The Type III will not go into distortion when strong upper mid or high frequencies are detected because it uses a different method of exciting altogether. The Type III actually synthesizes harmonics, even when the original program content is muddy and has very little high-frequency content to begin with. This method of putting back clarity into a questionable master tape is often a "fix-it-after-the-mix" life-saver.

"I mentioned earlier that there is a very important reason why remastering by adding a little extra sparkle using either equalization, limiting, the exciter, or any combination of the above, is usually necessary when doing mass duplication of audio cassettes. When a final master tape is put together and recorded onto dat, the listening environment for the dat is usually a controlled high-end situation, such as a recording studio. The average consumer does not typically own or have access to a dat machine. Cassettes are a much different story, however. The listening environment for the audio cassette can and does vary greatly. The same audio cassette may for one listener be played on a car stereo player, and for another be played on a home hi-fi stereo system. The third listening may be on a portable boom box at the beach or in the woods, there is no telling before hand what the listening situation will be. It therefore becomes necessary from my perspective, to attempt to allow for each of these listening environments when doing a large cassette run. By adding a touch of clean, smooth top end it becomes possible to build in a safeguard into the cassette mass duplication. The audio cassette that the end-user plays will sound great when played in almost any tape deck. You see, it is much better (from a sonic point of view) to make a bright tape copy rather than one that appears to be masked by an audio veil."

Figure 3. Jim Bescher sits at his control center. The computers are behind him, the box to the right is the HP-4 laser printer.

ARTFUL FABRICATION

When the copies are made,what can you do about the packaging? Said Jim, "Remember "packaging, packaging, packaging?"Now that we know that the sound recorded on the tapes (or CDs or records) is happening, the same level of integrity must be maintained for the packaging. This is the stage where all "fabrication" is done. It is so important that the final package has visual appeal and continuity so that when the tape gets thrown into a pile with other tapes, yours will catch someone's eye.

At Ariel Music Design we offer complete packaging which includes BASF pure chrome high bias tape which has excellent low- and high-frequency output combined with minimal distortion, J-card inserts, Norelco boxes, custom printed cassette labels or on-cassette printing, and the "cigarette-pack" style of shrink-wrapping. We provide 4-color separation for full-color cassette J-cards. A full range of coatings is available from standard press varnish and aqueous base gloss or matte finishes, to high gloss ultra violet coating. For those working on a somewhat limited budget, we can print Laser-Jet labels, which are black on any color or textured stock, using the Hewlett-Packard Laserjet 4 printer at a slick-looking 600 dots per inch. Though more inexpensive than 4-color separation, the Laserjet labels look classy, and certainly provide a cost-effective way of presenting yourself as a professional in the marketplace."

ALL BASES COVERED

Do you handle much in he way of less critical work? How do you handle that? Jim responded, "In addition to providing the most complete real-time cassette duplicating service available, Ariel Music Design now offers high-speed tape duplicating for less sonically critical situations such as voice-only narration, or any music that does not need to be digitally "pristine". Again for high-speed tape duplicating we use BASF tape. It gives faithful reproduction with a high amplitude at high frequencies.

"We have three digital dat machines in-house (Panasonic SV-3500, Panasonic SV-3200, and the Sony DTC-700) that we use for playback and for making digital back-up dats.

For the record industry we offer compact disc booklets and tray cards, compact disc boxes, record jackets, record labels and backgrounds.

"For the video industry we can provide laser disc jackets, video slip cases, bottom load, side load, or closed end, spine and face labels on pressure sensitive rolls, and labels for vinyl video boxes.

"In addition, we have folding boxes, envelope boxes, blister cards, posters and brochures, floppy disc labels, and pressure sensitive labels and stickers on standard rolls or pin-fed for computer imprinting."

FOOD FOR THOUGHT

Do you have any final thoughts Jim? "Yes, with this fast-paced, Madison Avenue, quick-fix society we are in, we as artists and people struggle to survive, it becomes more and more evident that in order to stand out from the rest and become noticed, we must indeed be individuals. This country was built on rugged individualism; by those who were not swayed by the masses, but remained steadfast in their personal convictions. The first key to success must be an absolute belief in yourself and your goals; how can you earnestly push for something that you don't wholeheartedly believe in? My point is simply this. Whether you are selling large quantities of something—audio cassettes for example, or selling a service, such as mine, there is one thing that is common to both: selling. You are selling yourself *first*, and your product or service ***second.*** Believe in what you are doing, and most importantly, persevere. Never give up. It is very easy to become discouraged and despair when it seems like all the odds are against you. You'll never taste success if you don't ever try. This is why I encourage people to make a concerted effort to sell whatever it is that they believe in.

"Audio cassettes have always been a perfect calling card for gaining visibility and grabbing a healthy share of a particular market (they also don't get lost or misplaced as easily as business cards do). Any form of business or service can be represented on tape. Everything from music to sales promotions, to an instructional series, to preaching can be recorded on cassette and used as an effective means of getting your product or service to the masses. Think about it." db

NEW PRODUCTS

FEEDBACK EXTERMINATOR

● The ADF-1200 combines automatically-controlled feedback with a parametric equalizer, shelving filters, programmable digital delay and a real-time analyzer. Features include 12 independent and concurrent digital filters that can be placed anywhere in the audio spectrum. The user can assign any combination of automatic feedback control filters. The parametric filters can be set at 3 Hz wide and 89 dB deep to eliminate 60 Hz hum, the 31-band real-time analyzer graphically displays the power spectrum on a back-lit LCD display with the RTA operating concurrently with all 14 filters, The LCD display and 8-button keyboard provides the user with control over all filter parameters.
Mfr: Sabine Musical Manufacturing Co. Inc.
Price: $1,695.00
Circle 60 on Reader Service Card

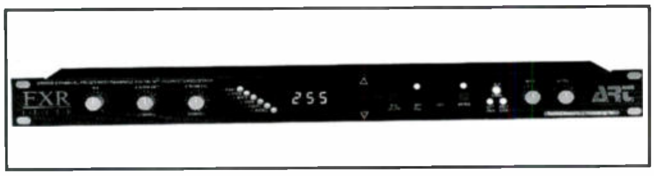
PORTABLE STEREO MIXER

● The new FP32A portable has three mic/line inputs and stereo cabability. Self noise has been reduced from the company's popular FP32 by 30 dB to make the unit compatible with digital recording formats and transmission schemes. 48V phantom and 12V phantom as well as 12V T(A-B) power are standard. Active input gain controls have been added that simultaneously lower an input's volume level and increases its clipping point. A front panel switch activates peak limiters for the left and right outputs, another switch links inputs 2 and 3 together in a stereo pair—controllable by the channel 3 knob. Bi-color LEDs are provided for each input, indicating signal presence and peak levels. A 1-kHz tone oscillator helps set reference levels, and both a one second slate tone and built-in condenser slate mic with automatic gain control assist the user.
Mfr:Shure Brothers Inc.
Price: $1,795.00
Circle 61 on Reader Service Card

SIGNAL PROCESSORS

● The FXR and FXR ELITE are two new signal processors. The FXR offers 250 preset combinations, up to four effects simultaneously, and fully discrete two-channel stereo. Because it now features on-board random-access memory, it can now do complex reverbs and pitch-bend effects. The FXR ELITE version is a programmable version of the FXR. The front panel includes an LED matrix which tells the user which parameter is being controlled, and an auto store button isused for saving the setting. Both units offer reverb, gated reverb, multiple types of delay, chorusing, and flanging. Inputs and outputs are electronically-coupled ¼-in. jacks.

Mfr: ART
Price: FXR—$219.00
FXR ELITE—$299.00
Circle 62 on Reader Service card



BOOM STAND

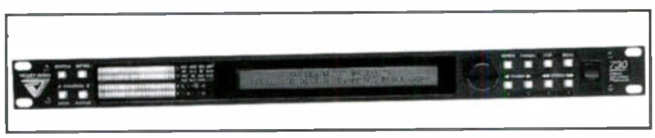
● This new boom stand, Model SB-11WE, is part of the Performer Series and features 60-inch long, two-piece horizontal boom assembly constructed of seamless steel tubing. The boom includes a swivel clamp and an adjustable 2 lb. counterweight for effortless boom operation. The tubing also incorporates a 900 degree angled end for optimum microphone positioning and terminates in the standard ⅝-in-17 thread. Vertical tube assembly is 1⅛-in. and ⅞-in. diameter and adjustable from 43-68 inches.

Mfr: Atlas / Soundolier
Price: $267.46
Circle 63 on Reader Service Card

DIGITAL PROCESSOR

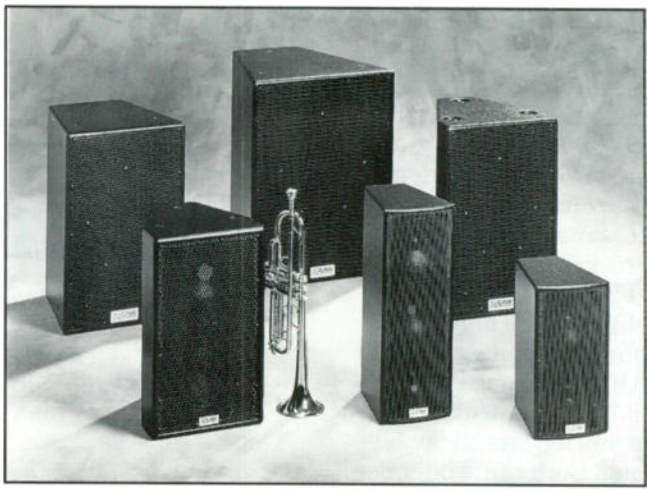
● Model 730 is a digital dynamics processor. It provides stereo compression, keyable expansion and gating, limiting and sibilance control. The unit is entirely digital and will support all digital and/or analog inputs/outputs. at any level or protocol: -10/+4 balanced/unbalanced, AES/EBU, S-DIF,S-DIF-2, OPTICAL, MIDI, RS-232, RS-422. It is single rack space.

Mfr: Valley Audio Products, Inc.
Price: $2,000.00
Circle 64 on Reader Service Card



COMPACT FULL-RANGE SYSTEMS

● The JF Series of high-definition ultra-compact systems are compatible with the company's sub-woofer system. The latest additions to the line are the JF260 and JF650. Both use a 2-inch exit compression driver coupled to a 60-degree constant coverage horn, The JF260's 12-inch cone and the JF560's 15-inch woofer have been developed for these systems. There are other models in the JF Series, all ultra-compacts.
Mfr: Eastern Acoustic Works
Price: JF260—$1,695.00
JF560—$1,895.00
Circle 65 on Reader Service Card



NEW LITERATURE

The P.I.P. Cookbook is a sixteen page guide to the company's Programmable Input Processor modules. These modules attach directly to the back of any P.I.P.-compatible power amplifier and provide a wide range of functions, including computer control of amplification. The booklet includes in depth details of specific applications and capabilities of each module.
Mfr: Crown International, Inc.
Price: free
Circle 66 on Reader Service Card

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If you are a manufacturer and want your new product listed in this section, send the release, include the suggested list price and there must be a photo or diagram included.

Send to New Products Department, db Magazine, 203 Commack Road, Suite 1010, Commack NY 11725.

Historical Perspectives

On this page and page 62, we present two photos from our historical files. You can safely assume that after twenty-five years of continuous publishing, our files have many pictures, some much older than these.

Jack Mullin returned from Germany right after the surrender in 1945 with two Magnetophon audio recorders, and a quantity of mylar-type audio recording tape. He went to a fledgling Ampex to see if they could make a new American machine, and with his partner Bill Palmer to 3M company and others to make a new tape.

Jack is shown here in this photo taken in the early sixties. Ampex had gotten into the tape business by buying the Alabama-based maker of Irish tape. At about the same time, 3M Company decided to get into the audio tape recorder business. Jack became the chief engineer of the new Mincom Division and is shown here with an Ampex 300-2 (two tracks on 1/4-inch tape). Mincom had apparently 5 tracks on this laboratory deck, here shown covered by reels of tape. An Ampex 600 is also mounted in those impressive wooden cabinets.

Why Are People Falling Asleep in the Church?

or

Why Is Church Sound So Confusing?

The Issues and Answers Book for Church Sound Systems and Church Acoustics

by Joseph De Buglio, Second Edition

JOSEPH DE BUGLIO

Why is Church Sound So Confusing?

Nuts and Bolts and More Tips

This article is Chapter 4 in a new book "Why is Church Sound So Confusing?"by Joseph De Buglio. See page xx for information on the complete book. It will shortly prove to be the complete word on church audio installations.

WHEN TO START TRAINING?

Before we can suggest a sound system design, we must look at the larger picture. It is one thing to give a church a sound system that finally does what it is supposed to do, it is another thing to know how to use the system to enhance the worship, rather than degrade it.

The operation of a church sound system is a team effort. That team includes the Minister, the Organist, Pianist, Choir Director, Song leader, Soloist, Musician, Lay readers, Guest Speakers, Guest Soloists, Guest Musicians, Children, Teen-agers, Lay members, Ushers, Deacons, Elders and of course, the sound person. (Some women have proven to be the best sound system operators in North America.) Every person in a church needs to be taught how to use a sound system, what to expect of it and how to get the same high quality results every time.

We are now approaching the third generation of churches with audio and it seems that 98 percent of the people haven't a clue about what they are doing. The purpose of the sound system is to have the people in the pews intimately involved with every part of the worship.

Example: In some churches there are lay people who pray out loud during a planned part of the service. Often, they will simply pray out loud from where ever they happen to be standing. In small churches this works just fine but in larger churches, a person's voice needs to be amplified. It is interesting to see what happens when a microphone is put in front of most people. Give a person a microphone and their automatic volume control turns down. Furthermore, they often hold the mic so far away they give the impression that the mic has teeth. Sometimes they hold the mic as if it has the pickup capsule in its side. Now, if you combined these two responses, you may conclude, as I often do, "what that person has to say must be so personal they don't want to share it with others." The only reason for these responses is lack of education. People have to be given permission to speak out with the equipment they have. Would you build a church to protect you from the elements and then have all of your services out of doors because you're afraid to wear out the building!

The worst offenders of not using a microphone properly are ministers. Most ministers act as if the microphone is a hindrance or an obstacle that well meaning people put in front of them. Considering what Bible Colleges teach and what those institutions own, it is no wonder ministers are afraid of using audio equipment. A poor system in a church literally drives people away. All the mic techniques in the world will not help.

However, in a proper sound system, the microphone becomes an extension of your voice so everyone can participate. If you treated a microphone like an ear of a very close friend, you would be well on your way to taking advantage of your ministry.

HOW TO USE A MICROPHONE

The way this works is very simple. If you were speaking softly, as you would with a close friend with whom you have an important message to share, you move closer. This means getting within 4 to 12 inches from a mic. With a good system this is whispering. When you are talking as to a friend across a living room with general comments, you should be 9 to 20 inches from the microphone. When you have important statements or a special point on which you want to raise your voice, you should be 15 to 36 inches from the microphone.

The reason for using this method is straight forward. When your voice is low, there is a lot of detail that people often miss. This detail is needed because it's not what was said but how you said it that will carry the largest impact. Your words of comfort are meaningless if people only hear 30 percent of them. Therefore, getting closer to the mic allows people to fully understand the importance of those supporting words. Likewise, when you are excited, you need to move away from the microphone.

When you speak louder, you will blow more air. As you blow more air, you will pop the mic with strong bass sound. This is annoying, distracting and easily avoidable. If you are able to move away from the mic (if you are hand holding it, it works the best), you will also keep the volume down while getting your message across more effectively.

Getting louder often does not mean getting clearer. In fact, as you increase the volume of a sound system, the room starts to fight back. If the acoustics are good, the sound can be louder but, as in most churches, there is a narrow window at which the volume can be set and it is up to the person speaking or performing to know the limits. It is insulting to everyone when a sound system is so loud you can not understand what the minister is saying.

There are many churches today that are abusing our hearing. Did you know the average sound pressure level of congregational singing in a conservative Protestant church is about 95 decibels (dB) and in a Pentecostal type church the singing is about 5 to 10 dB louder? According to most of the Health and Safety Acts around the world, long-term exposure to sounds at these levels will cause some hearing loss. In one church, the congregational singing could last up to an hour and a half. In this 2,000 seat church, the sound levels were often over 108 dB. According to OSHA, your exposure time is 30 minutes without hearing protection. Perhaps if people started to ask for hearing protection in some churches, the sound levels would come down.

(Many of these churches have serious acoustical problems and most sound men and audio companies have the attitude that if you turn up the sound loud enough, eventually, the room will not have any affect on the sound and it will eventually get clearer. Folks, the opposite is true. Turning the system down will not only make it clearer but people would get more out of it than just abuse.)

Distance	SPL dB	Increase
32 in	60	0 dB-Performance of a HIS System
16 in	66	6 dB-Normal Speaking Distance
8 in	72	12 dB
4 in	78	18 dB-Typical singer holding mic
2 in	84	24 dB
1 in	90	30 dB
½ in	96	36 dB
¼ in	102	42 dB
⅛ in	108	48 dB-Lips touching the windscreen

Sound Pressure Change When Moving Into A Microphone.

This chart shows the amount of change that occurs as a person moves around a microphone. Just as doubling your distance changes sound levels 6 dB at the speaker end, the same happens with input. When a person moves into a microphone room 16 inches to 1 inch, the sound pressure change is 24 dB. In this close distance, there is little or no room effect on the input signal.

A CONSTRUCTION DETAIL TO CONSIDER—AND MUCH NEEDED

Cable chase ways: Since there has been no standard in church sound systems, the idea of a cable chase way seem pointless. However, if a cable chase way for lighting and sound were planned, then the church could have flexible options in the future.

Cable chase ways are not new. In fact, offices have had them for years. Cable chase ways cost more during construction but later when changes are done, there would be no need for expensive conduit work after the fact.

IS THE "Q" IMPORTANT?

It is now generally accepted that a church is best served with a central cluster speaker system. The speaker system is often placed somewhere over the front of the pulpit or platform area using directional speakers that have a predictable and constant dispersion of sound. Omni directional speakers such as spheres, column speakers, low "Q" speakers and Hi-Fi speakers fail to perform in so many critical ways they are not worth the paper to criticize.

Not all speakers are the same and these other speakers work better in application where non-critical listening is required. Finding so many Soundspheres in church closets, for a "State of the Art Technology" that is considered current, is a clear case of buying the wrong tools for the wrong job. Between spheres, column speaker, flat speaker other speaker designs and audio products, if all of the churches who owned them donated this equipment to other churches who are determined to waste their money on this inappropriate equipment, the equipment could be recycled many times and churches would spend less of my money foolishly. However, as one church board chairman said, "I wouldn't want that equipment wished upon anyone else...." Those strong opinions are often made after a church has had the opportunity to compare a proper church sound system with one designed with good intentions.

In the early days of church sound, speakers generally were all horn types with horn loaded woofers. Amplifiers were expensive and rarely over 100 watts in power. (Today, many amplifiers can produce over 400 watts X 2 channels at a 4-ohm rating.) A term often used and often misunderstood is "Q". "Q" is the directivity rating of a speaker.

High "Q" speakers are designed to project sound over large distances. A low "Q" speaker is a speaker that allows the listener to get comfortably close without serious degrading of the sound. A hi-fi speaker is a low "Q" speaker. A police Bull horn is a high "Q" speaker.

Just as the installation of the Organ was resisted in the church because it was once played while Lions were feeding on Christians in Roman arenas, churches have also been slow to embrace the sound system for similar reasons. At first, churches used their sound systems strictly for speech only. Today, some of the best sounding systems in public places are in churches.

In the beginning of church sound, people soon learned that a sound system was very limited and often the room had to be fixed. Many churches did fix their sanctuaries with very good results. In the mid 50s, bigger and less expensive amplifiers arrived in the market place. At the same time, speaker engineers began to understand what "Q" was and how to measure it. A high "Q" rating could be from 10 to 25. A low "Q" is from 2 to 9.

There is a direct relationship between reverberation, "Q" and how much direct sound arrives to the people in the pews. Poor reverberation limits the performance of a speaker system. By increasing the "Q" of the speaker, you can compensate for the increase in room reverberation. However, there is a limit as to how high you can increase the reverberation of the room and have a speaker with enough "Q" that will still give the best sound coverage possible.

As more churches began to accept and use sound systems, audio contractors suddenly had to deal with churches that had reverberation times of over 1.5 seconds. (Most audio contractors rarely have to deal with as hostile an acoustical environment as that which occurs in churches.)

There are some speakers that have a "Q" of 50 . A police bull horn has a "Q" of over 50. Police bull horns don't sound very good. Many church sound systems sound like bull horns but regular two way speakers are being used which should sound very natural. Once a speaker "Q" rating passes 25, it no longer sounds natural due to too much compression of the sound within the bell of the horn. (Horns compress sound in terms of ratios. A highly projecting horn can have a compression ratio of 12-15 to 1. A natural or musical sounding horn has a compression ratio of 4-8 to 1. Low compression, high "Q" speakers give the best overall performance in a church.) Speakers with a "Q" of over 22 generally are not very musical. They have to be supplemented with woofers. The end result is the woofer drags the "Q" rating, over the full range of the speaker system, down.

Later, someone discovered that if you stacked several horns on top of each other, you could increase the "Q" of the speaker system and maintain a reasonable quality of sound.

This works fine but where do you draw the line? When is it time to stop looking to sound equipment as a magic wand to solve poor building designs or poor use of construction materials?

In one photograph (page 348 in the book *Sound System Engineering* by Don Davis), it shows a speaker cluster of 4 horns and 4 woofers stacked one on top of the other. It was demonstrated that the test worked very well but the speaker system looked awful. This was an example of creating a high "Q" array using a column speaker approach. In this case, it would have been better to fix the room rather than having a speaker system that looked like a monster.

There have been several studies and charts suggesting the limits of reverberation and the "Q" of the speaker system for churches. (These charts can be found in the books mentioned earlier. Klark Teknik has a manual that has excellent charts on this subject.)

From experience, a church with a reverb time of 2.4 seconds or longer at 200 hertz will not be able to count on their speaker system to work properly for every part of the worship service. "Q" is important but there is a limit. The higher the "Q", the more speaker compression.

For churches that have ambitious music programs, a medium to low "Q" speaker that sounds very musical and low reverberation are a must because in a loudness war, the room always wins. (A list of the 5 top speakers for the HIS System discussing their good points and bad points and when to use them will be in the next update.)

Note: Dome tweeters and bullet horns have a very limited use in a church. They have two major problems. If the speaker system was only controlling prerecorded music all day, they would work very well. However, speech requires a lot of speaker output in the mid range. Many domes and bullets fail and fall short of the power demands of speech. Secondly, most domes and bullet horns have such wide dispersions that gain before feedback is seriously limited. Although domes and bullets allow you to make a lower cost speaker, the performance limitations and high rate of failure make them impractical for use in a church.

ARE ALL SPEAKER CLUSTERS THE SAME?

It would be fair to say there are huge differences in speaker components and the jobs they are designed to do. It is also important to know that the position of a speaker in a room is super critical. Every room has a sweet spot. A speaker out by 1 foot can make $2,000 speakers sound like $50 A.M. radio speakers.

Some speaker systems are less critical than others and in some rooms there is no room for error or flying by the seat of your pants in design. You should have a detailed knowledge of your sound system supplier. The following is a list of things you should know about your supplier:

1. Learn everything about the speakers you are being recommended. An audio contractor doesn't have time to know about every speaker that is available or how they sound.
2. Avoid custom built speakers. A custom built speaker is often designed on a given set of assumptions.
3. If you have a custom built speaker for an original design, ask for the printed test results. Usually, there is no laboratory testing on custom speakers so you have no way of knowing 100 percent how the speaker will behave under church conditions.
4. A good sound contractor will have a limited number of speakers they use and will know them inside and out.
5. Learn how to read a spec. sheet and understand what it all means.

After selling over 100 complete sound systems, it never ceases to amaze me how a church can spend huge amounts of money based on a proposal without going to hear the finished product in other churches. Many speaker clusters look the same. Many audio companies use the same equipment. The end results are like day and night. When you buy a church speaker system, it must be assembled and strategically located in your church. What good is it to have a speaker shoot-out at your church when the audio company doesn't know how to install it for the best performance. Speaker testing is often a waste of time for the client. It is very important for the contractor.

In many speaker demonstrations, people are often caught up in how a speaker sounds rather than knowing how a speaker performs. In reality, the differences between speakers from the major speaker manufacturers is very small. It is the ability of choosing the right speakers for a given job, that takes skill.

In a recent speaker demonstration in a church, three different horn speakers from three manufactures were represented. Before equalizing and before setting the volumes at equal levels, all three speakers had very distinct sound differences. However, after equalizing the speakers and setting the volume levels to equal loudness, the differences were very small. In each case, the speaker system was limited by the performance of the room. Once the room was factored in, the sound quality of the speakers became very similar. What was more noticeable was speaker coverage and control. One speaker was more musical, the other could play louder and the final speaker had better control.

WHAT DOES THE CHURCH NEED? DOES THE CLIENT KNOW?

Speaker clusters have a very small window in which to work. When you buy a sound system for your church, it must include the skill of your contractor and his sensitivity to the needs of the church and the acoustics of the room. It is nice to buy the speaker system that sounds the best, but if the best speaker system can't sound better than the others because the room gets in the way, isn't it better to find the best combination supplier, speaker and room repair?

When a church asks for a demo in their sanctuary, the smartest sales person is going to win. When a church takes the time to visit other churches to hear the system in action during a worship service, the best performing system will win. Not all speaker clusters are the same and not all audio contractors know how to find the room's sweet spot.

ONE RULE—ONE FACT

One of the most important rules for all sound operators is simple: Once the minister begins to preach, you are allowed only one volume level adjustment throughout the whole sermon (unless your minister instructs you otherwise). With a good sound system, the minister should be able to control their own volume by using the microphone properly (more effectively). However, there is one adjustment most ministers won't mind.

Most sermons begin 3 to 7 minutes after singing. Since most singing is very loud, your hearing of low sounds is stressed for a while. Speech is intermittent. Speech gives our ears a chance to rest between words. When the volume is set right after a Hymn, chances are it is too loud. Some people get the impression the sound system has been turned up. Rather, it is our hearing that became more sensitive to lower level sound.

Often the sound person can lower the volume of the system by 3 to 5 dB at about 5 to 10 minutes into the sermon. The interesting thing is, if you don't do this and you need more volume from the sound system, it won't be there.

Another rule is to turn off all other microphones not in use during the sermon. The more mics that are on, the less gain you have before feedback. This is important for the times when your minister has a throat problem but is determined to continue in a lower voice. Furthermore, the sound tends to sound clearer because there are no open mics to reinforce the first sound. There are other reasons for turning things off or down, but for now, these reasons will do.

A POWERFUL PERSON

One of the most important facts is, the soundman, in churches with a good sound system, is the most powerful person during a worship service. The soundman can either enhance everything and assist people sitting in the pews to be more involved with the service or, the soundman can undermine everything the minister does without the minister knowing it. It is time we wake up to the fact the 95 percent of all churches have a sound system of some kind. Unfortunately, only 5 to 10 percent of these churches have a sound system really usable as a tool. All other churches have sound systems that get in the way. When something gets in your way, it undermines everything. Even a good sound system can have the same problems of getting in the way.

For example: In a good system where people expect to hear properly, the minister or a lay person may move to a mic that has been turned off to give the other mic better control. The sound person should be alert and see the minister moving to the other mic.

If the mic is set right, the mood of the service continues. If the minister speaks for 3 seconds or longer with the mic off, and the minister notices it, what happens? It breaks the mood and often, his concentration. This single event upsets more ministers than anything else. Moving from one mic to another at the closing of a sermon may be just what the minister needed to do to make his final plea or point. A soundman asleep at the controls is unacceptable.

Of course some will suggest using a wireless lapel mic. In my opinion, a wireless lapel microphone has some noticeable limitations. Most churches do not have the acoustics needed to use a lapel mic effectively. (Usually this means a dead room and most churches don't want dead rooms.) With a lapel mic you cannot raise or lower your voice too much without losing something or distorting the wireless. (Automatic volume controls will cause other problems.) With a wireless, all you get is plain vanilla in the presentation. Just as some ministers wear the well-earned title of word smith, a person can also be an *amplified voice smith*—a person who knows how to present a message using all of the inflections of voice. Casey Kasim has the most recognized voice on TV today. He has made millions of dollars selling his voice to television.

On PBS, the public broadcasting service, there is a weekly show called *Nature*. The success of *Nature* and its continuance is based on the voice of George Page. This is a voice that brings life to the screen. Jason Robards, Lorne Greene, Robin Williams and Dick Van Dyke are all examples of the use of voice. As they say in the business world, it's not what you say but how you say it. A good sound system accurately reproduces everything you say and how you say it — the key to a successful presentation.

A lapel mic can not give you anything more than a vanilla presentation. Then again, there are many gifted ministers who know how to make plain vanilla taste like chocolate. Of course, this is only an opinion based on many experiences. What have your experiences been?

This is not to say wireless are not effective. For drama presentations, plays and other specialty events, a lapel wireless setup is just the ticket. Today, some ministers make the usage of a wireless microphone a condition of their employment. Getting a wireless mic just to have "freedom" must be combined with planning to be more dynamic with body language. What a church should avoid is getting a wireless lapel mic for a minister who does not use body language in their presentation when they could have done better buying 6 regular microphones for most of the other events they wish they could do better. A good wireless lapel is a lot of money. If all you have is $1,000 to spend, 6 mics with stands and cables could be the wiser choice.

ASSESSING THE SANCTUARY

Room Shapes

There are six basic room floor plans with hundreds of variations of each. There are rectangles, squares, diamonds, ovals, triangles, circles and pentagons. In roof designs there are several basic shapes with many variations of each. There are Domes, "A" frames, Flat, Sloped, Vaults and waves. Almost all the known shapes will work but you need to know at which end of the church you will preach and at which end you will listen.

The type of room shape must match the nature of the service or denomination. There is no such thing as an ideal or perfect space but, it is important to recognize the shape, how it works and where the speaker system must go. Don Davis wrote in his book *Sound System Engineering* that a speaker system often goes where a wall or ceiling should be. This is very true. But as you get into larger spaces that seat 200 people or more, you need more than a reflector. The wall must amplify as well.

SOUND SYSTEM DESIGN TYPES

In churches, you will see a variety of system designs used. Some designs are chosen because of appearances, others are chosen because of the perceived cost savings but most are installed to compete with room acoustics. Unless you have a ceiling lower than 12 feet, the best system in any church is a Central Cluster design.

The Central Cluster design forces you to look at the person speaking or singing. It offers the highest levels of intelligibility. It has the lowest levels of Listener's Fatigue.

It is usually the best layout for operating and hearing the speaker system. It usually has a higher level of gain before feedback. It is a system that you notice the least.

Therefore, you can say a central cluster system is the least obtrusive available. The other sound system designs are either creators of dead spots or very expensive if done correctly.

The following is a description of the various system types and why they are not appropriate in a church.

PEW BACK SYSTEMS

The pew back system is most often attempted in churches with long reverb times. It is based on the idea that if you get the sound sources closer to the people, the listener will get more direct sound and less interference from the reverberation of the room. If it is to work, it can only be a speech system. It will not work well for music because music has to be played loud enough for the musicians to hear themselves. This means, in the pew back system, a listener could hear both the direct sound and the amplified sound coming from many directions, depending on how the listener moved their head during the live music. Music, at a medium level in a pew back system, will increase the reverberation time of the sanctuary. Remember the reason for the pew back system?

From the experiences known, there is no church that has a pew back system that works as good as a properly designed cluster system. Generally, a pew back system is abandoned after several years of trying to make it work. Robert Schuller's Crystal Cathedral in California is a prime example. They went from a cluster system to a pew back system and now they are back to a cluster system. The next step is to find a method of reducing the reverberation without interfering with the appearance of the "Glass Slipper".

DESIGN PRINCIPAL

The pew back system will have speakers mounted either on the back of the pew, under the pew or on the floor under the pew. A speaker is placed every few feet to give even coverage in each row. Each row, or in some cases, every other row gets speakers. Some people have tried one speaker for every other person or a 2 to 1 ratio, while others have tried up to 10 to 1 ratios.

For every 2 or 3 rows of speakers, you have to install a delay system. The delay is used to delay the electronic sounds from one row of speakers to the next and to delay the electronic sound from the source of the sound. Without the delay, you will hear echoes. Costs can vary. For a speech only system that works reasonably well, the cost per seating position can be well over $80. This makes it the most expensive way of doing sound. Would it not be better to spend $35 per seat on sound and $45 per seat on acoustics?

Contractors who have tried pew back systems have used all types of speakers at all angles. It doesn't seem to matter whether it is a $10 speaker or a $200 speaker, overall performance is low. Some systems are on a 70 volt or 100 volt distribution layout, while others have tried a mini amplifier for every 2 speakers. Others have also tried to series speakers together to try and keep the amplifier cost down. Running an audio signal from speaker to speaker does create other sound quality problems as well.

The only place for a pew back system is in town halls, city hall counsel chambers and in churches that have no music in their services.

THE LEFT/RIGHT MONO SYSTEM

The left/right mono system has always been called the poor man's system. It is installed out of convenience, lack of planning and a copy of what entertainers do when they only have a few hours to put on a show. Of all the books available on sound reinforcement, not one book shows how to install such a system. Rather, they go to great lengths to explain what is wrong with it. Instead, they all support the cluster system as the best way to install a permanent system. The left/right mono system is strictly a portable or temporary system setup. It was (and still is) a fast and convenient system by which entertainers could put on a show for their public.

There are four main reasons for not using such a system in a church:

1. Dead Spot

When you have two mono sounds separated by 10 feet or more, the left and right speakers will start canceling each other out in the overlap areas and whenever you are at a different distance from each speaker. When you are at an equal distance to each speaker, the sounds are summed together, often increasing the sound level 6 to 9 dB.

When you include wall reflections, the problem is compounded and it creates additional dead spots. When the speakers are 25 feet apart or wider, the areas of overlap increase dramatically. A dead spot can be easily measured with an inexpensive sound level meter such as the one Radio Shack stores sell.

In many testing experiences with computerized measurement systems, some very interesting pictures of sound began to appear. In many churches with a left/right system, it could be seen how the sound from the left speaker was louder than the right speaker but, the test microphone had been placed in front of the right speaker. This position would be about 30 to 55 feet out and about mid point of the right side of the church. By standing at this position and running the test signal, you could indeed hear which speaker was louder. Please, remember that dead spots are often frequency dependent. That is, since every frequency has a different wave length, not all sounds or notes will cancel or boost at a given position. A sound level meter can tell you whether the sound is lower or louder. A computer system is needed to determine if certain sounds are missing in a given position. In translation it means that in this pew the vowel "a" and a "Mc" are not audible and all "b" and "ch" sound are too loud. Some people can fill in the blanks better than others. People with hearing aids have problems in this setting.

2. Gain Before Feed Back

In a left/right speaker installation, the speakers are often placed behind the pulpit at equal height or slightly higher than the pulpit. Most speakers have great vertical dispersion control but poor horizontal control. This is like placing a microphone in front of the speaker, and we all know that will cause the system to feedback. The causes of feedback are a combination of acoustics, angles of incident, proximity effects and oscillation. As you turn up the level of a microphone, it is able to pick up and amplify everything. This includes any low level noise generated by your electronics (mixer, amplifier, equalizer, preamps, effects devices and other pieces of electronics connected to the sound system that is unstable. Poor wiring and electrical induction are other causes of noise.) As a result, any speaker close to a microphone will cause feedback. Any speaker that has a signal path with an angle of incident that reflects sound back to the microphone will cause feedback. When the gain of the mic is turned up so loud that the noise from the system is being recycled (that is when the noise in the system is being produced by the speakers and picked up by the microphone, the level increases in a cycle many times until you hear it. This can appear to happen instantaneously), it causes feedback.

The acoustics part of the picture is more complex. All large rooms are constantly flexing and moving. This is a constant effect altered by room temperature and humidity. Speakers move air. Microphones will amplify everything whether you can hear anything or not. At a level you can not hear, the room's flexing is being picked up by the sound system. As long as the level of sound system is kept low, the sound system is stable. When the levels of the sound system are turned up, decay time of the rooms flexing becomes longer. Now, at the same time, the speaker system is moving air and exciting the room, multiplying the energy output hundreds of times. This creates a chain reaction in which the sound system amplifies the resonant frequency of the room (room flexing) and causing feedback. Acoustics play an equal part in the causes of feedback. When there are multiple speaker locations in a room, the number of wave forms that will excite the room increases. Also, the number of hot spot reflection points increase which causes feedback. This is only a partial explanation.

3. Intelligibility

Intelligibility is the understanding of individual words. As discussed earlier, in a 45 minute sermon a minister can speak about 10,000 words. A sound system with a score of 80 percent intelligibility will mean that 2,000 words in a 10,000 word sermon were missed or misunderstood. Depending on your seating position, one speaker will either boost or cancel certain frequencies. That means every "ch" sound is canceled and every "a" sound is amplified. As a result, many words and word fragments are missed. Fortunately, our brains are usually fast enough to fill in the blanks because of familiarity with the subject matter and the person speaking. A visitor to the church does not have this advantage. For this reason, no church should accept a sound system that scores below 87 percent intelligibility. Below 87 percent people can misunderstand complete phrases. The TEF or MLSSA acoustical measurement systems can test a room's intelligibility in minutes.

4. Localization

Churches always say they don't want to see or hear the sound system, but it must be loud enough and crystal clear. When you have a sound source at ear level, you are automatically programmed to first look at where the sound comes from, then to look for the source of the sound. This is a basic protective instinct all creatures possess — to be able to tell where danger is coming from is natural. Since humans have no natural enemies that attack from overhead, man has his eyes and ears where they are, on the front and sides of our heads. A mouse is most often attacked from overhead. They need their eyes and hears closer together, near the tops of their head. A mouse can also rotate the ears to find the noises.

When the sound is coming from one direction (Speakers) and the visible source is in another direction (the person speaking), the human brain goes into overtime matching up the two events. When you compound low levels due to feedback problems, low intelligibility and multiply sound sources, you have classic listener's fatigue syndrome.

It has been shown that a poorly designed sound system can help people lose their attention span or cause them to nap sooner than when a proper sound system is used.

The difference can be as much as 20 minutes.

When the eyes and ears can focus on the same event, you spend most of your time understanding what was said. When a left/right, or distributed or pew back system is used, you really spend most of your time just trying to hear.

It would be fair to say most church sanctuaries, by themselves, do not have dead spots. Rather, most churches are so large they need amplification. If all you need is extra level, then why would anyone install a sound system that creates dead spots?

WHAT ABOUT A STEREO SYSTEM IN A CHURCH?

Some churches have successfully installed Stereo Reinforcement Systems. With the arrival of electronic instruments, stereo keyboards and tape accompaniment, some churches have felt it necessary for this kind of investment because they had the facility and talents.

Remember, a Stereo system at home is the opposite from a stereo system in a church. At home you position yourself between two speakers and the recordings play tricks with phase to give the impression that sounds are coming from different places within the sound field. In a live situation, you can see where the sound is coming from. Therefore, the sound system must give the impression the sound comes from the same place otherwise the performer will hear an echo and it will disrupt their playing ability. A live stereo speaker system is really many speaker clusters over the performer's head. The nearest cluster amplifies and the other speakers are silent. This gives the effect that sounds are coming from the different parts of the stage where the performer is, giving a better picture of localization. As a result, a true stereo church sound system can cost 4 to 5 times more than a good quality mono single cluster. (Church stereo systems will be discussed in detail in future editions of the book.)

SPEAKER LOCATION

The location of the speaker system in a room is the most important step in clinching the desired results. If the speakers are in the wrong location, the rest of the system will sound mediocre despite the quality of the equipment. In almost all churches, there is only one proper location for a speaker system. Any other location is a noticeable compromise in comparison.

The speaker system in your church is the most important part of the audio link. If this part of the system is not correct, you will not be able to successfully make any improvement through electronics. It is vital therefore, to make your speaker system the first step in correcting your sound problems. Not treating the speakers first will result in needless experimentation and expense. (It is amazing to see all of the gadgets churches try to invent. If only they knew the laws of audio and acoustical physics, they would spend millions of dollars less on audio products that don't work.) In rectangle shaped rooms, a single point is ideal for all church application.

(According to Dr. Dave Eagan and Dr. Don Davis, there is no other shape better than the shoe box or rectangle church. Boston Symphony Hall is a prime example.) Often the speaker location is always several feet in front of the pulpit, overhead. Other times it is directly overhead while in some rooms it is 2 or 3 feet behind the pulpit. The exact positioning must consider ceiling height, pulpit height, width of room and pulpit to back wall distances.

In wide or fan shaped rooms, localization of sound does present a minor problem. Those sitting on the ends will perceive two sound sources. This will not reduce intelligibility. However, it will introduce minor amounts of listening fatigue.

For this reason, it has been our practice to divide the sanctuary into several rooms. This gives all people a point source for listening comfort. Sub clusters are very effective. Phasing problems are controlled by separation, crossovers between horns and sometimes digital delay circuits.

Typical comparison of a Single Cluster Speaker System verses a Multiple Speaker System.

Situation	**Cluster Systems**	**Other Systems**
Dead Spots of 3 dB or more	Almost none	Many
Phase Cancellation	None	10 to 30%
SPL from front to back	6 dB or less	Often 12 dB or more
Realism	98%	15% or less
Articulation	+85%	75% or less
Intelligibility	Great	Poor
Max. working distance before feedback	20" to 40"	4" to 16"
System design life	Permanent-possibly unlimited	Replace when no longer tolerated
Listening fatigue	0.5&	20 to 50 %
Music Quality	Hi-Fi	Limited
Flexibility	Very good	Limited
System headroom	20-30 dB	10-20 dB
System focus	Pulpit area	To each speaker
Echo Amplification	5% or less	10& and up

CLUSTER HEIGHT

The maximum height for a cluster should be no higher than 40' and no lower than 13'. However, height is also determined by the speakers throw distance and other room restrictions. Remember, the closer to the ceiling, the more bass the speaker system will produce. If the room is bass heavy, hang the speaker lower if there is room to do it.

If you require a throw distance greater than 145', a sub cluster system may be required. Ceilings below 13' may require other system designs. This book will not discuss these requirements in detail in this edition. Send for a Supplement for your church.

Height	13'	Max. Length	48'
"	20'	"	72'
"	25'	"	85'
"	30'	"	110'
"	35'	"	130'
"	40'	"	145'

Height Ratios. The target point is usually the third row of pews from the back wall.

SPEAKER SAFETY

Hanging a speaker from a ceiling presents some concerns that need to be addressed:

1. Cabinet construction: Ceilings in many churches have a wide temperature range. During the summer in some churches, the ceiling can exceed 120 degrees for many days on end. Many speakers are only fastened with nails and glue. There are a number of stories circulating the Audio Engineering Society (AES) and the National Sound and Communications Association (NSCA) that describe how speakers are falling apart and falling down. Look for a speaker cabinet that is reinforced for roof suspension.

2.Do not use chain: Speakers vibrate and it can cause metal fatigue to the chain links. Also, a lot of bass sounds are lost with chain suspension.

3.You should not use aircraft cable because of sound quality in the bass. It has the same problem as chain.

4.Speakers should be supported from the side walls, not from the top of the cabinet only. (Some speakers have metal rods or bars that run through the speaker box to support the bottom of the speaker from the inside. Also, look for a speaker with a space frame type of construction.)

SPEAKER BRACKETS.

Custom steel speaker brackets are the best way to support the speaker system and they are very inexpensive to make. A properly welded steel bracket wins in many ways. Depending on the number of speakers in the cluster, there is a wide variety of fast and simple adjust you can make that are often too awkward with chain or cable.

In most single and two speaker clusters, the steel bracket is very cheap to make. A bracket that is safe up to 400 pound can often be made for under $50. That includes the steel, welding and paint.

THE BEST ILLUSION OF THEM ALL!!!

There are three very strong effects only a central cluster can do. The first, which has already been mentioned, is localization. From 80 to 90 percent of all the seating in the church, when someone on the platform of the church speaks, sings or plays, everyone's attention is on that person. Since the cluster is positioned in the vertical axis of our ears, the sound arrives to both ears at the same time. The reflected sounds will give the direction.

Since the reflected sounds are much lower, your first reaction is to look at the pulpit area or center to the altar area. At the same time, your eyes will focus on the first moving object or to the tallest person standing at or around the pulpit. The only people who will notice the cluster are people seated at the extreme right and left of the front three rows of pews.

The second effect clusters give is the illusion that the minister is only four or five feet away. Since you are listening to only one speaker in your seating position, there is no presence cancellation. Generally, this effect is for 70 percent of the seating and when the RT60 is below 2 seconds. When the RT60 is longer, not only does the sound system lose intelligibility but the music program is degraded.

Multi sound source speaker systems can never give this effect. This system gives the illusion the minister is further away than what he is in reality.

The third and most exciting illusion a cluster has to offer is movement of sound. Since 80 to 90 percent of the people seated have sound arriving to both ears at the same time, it is very difficult to say where the sound is being amplified from except from the source, the minister, lay person or singer. When the person moves, the sound appears to move with them. There is a limit to this. If the minister moves from a central pulpit position to the extreme left, talking the whole time with a wireless lapel mic, it leaves the audience thinking the sound followed the minister until he had moved half the distance, then the speaker is high. Example: If the cluster is 20 feet high, the minister can move 10 feet to either side of the pulpit and have the illusion that the sound is following him.

Since it is natural to look at what we hear, the central cluster approach to church sound is the only truly natural method of sound reinforcement.

SOUND OPERATOR TIP

One of the rewards of installing a high quality and affordable sound system is in the listening to the system. When a system is properly adjusted, for the many different parts of a worship service, you will have the impression that the sound system is not on at all. Although many listeners like the effect, a new problem keeps showing up.

For every new system, you try your best teaching the volunteer operators of the system. In recent years, video taping the training session has become a valuable tool in reminding the sound operators of all the tricks to using the new system. However, as good as some sound operators are, training can take many sessions.

As stated earlier, many people with audio experience have generally picked up many bad habits. The one habit that is hard to break is the most obvious. In many retraining sessions, I have found the operator has changed the channel equalizers or the main equalizer in such a way that the sound system sounds like a bull horn.

It seems that hearing the minister as though he were only 2 feet away is not enough. Many sound operators want to hear the sound system sound like a PA system so they will be convinced the system is working. The bottom line is, they don't trust their hearing. Although this is not a serious problem, it has caused some embarrassment to the suppliers of such systems.

If you operate a good sound system, trust your hearing. As long as you can understand the minister, everyone else will too.

MIXER LOCATIONS

The best location for a sound operator in a church is on the main floor, 1 to 3 rows from the back of the church and in the pews. Preferably, just inside of the outside isles.

If there is a balcony with seating under it, the mixer desk is best located 2 or 3 rows out from under the balcony's front edge. Although it is a new concept to most churches of today, historically, churches started the idea of having a sound operator controlling the sound system from within the congregational seating area in the 1940's.

As it turns out, it is impossible for a person to adjust sound levels from one area for people in another area. It doesn't work. The idea of having a mic mixer in a place like a pulpit or a room behind the altar is very awkward. How many times have you been to a church with such a setup and have heard the sound system ringing or sounding like someone is speaking through a tin can through the whole service. People come to church to pray and hear what the Minister/Priest has to say. It is annoying, insulting, and rude to have to put up with something that could have been adjusted in seconds.

And of course, when someone complains, the Minister/Priest says he didn't hear it! Well of course not. You have to be in front of the speakers to know what is happening. Well, enough with people frustrations.

Having a sound operator and a lockable secured wooden or steel mixer desk is the best choice. For some denominations, that have a very structured hour of worship, in most services the levels can be preset with 3 or 4 mics and no one needs to operate anything. The only problem with this or any automated system is that everyone speaks differently from week to week, day to day, minute to minute. Even with the most professional presenters, no one is able to speak at a constant level for everyone to hear all the sounds at the same volume. True, you can get expensive gates, limiters and compressors but their use is very limited. For example, if you set your limiters for a person with a powerful voice who is speaking very close to a microphone and seconds later pass the same microphone over to pick up the choir at 10 feet away, you will hear nothing. Move the mic 5 ft. away and you still hear nothing. But if you bypass the limiter, compressor, and gate you will find that you have more than enough audio level.

If there are several things going on at the same time in which many mics need to be turned on or off, having a sound operator is the most natural and best way to run the sound system. With a good operator, most people will not be aware that any adjustments are being made. Besides, when the operator is in the pews, he cannot day dream or fall asleep. He is forced to stay alert.

It is no secret that the fewer mics that are turned on, the higher the system can be turned up. Most churches that have tried to use an automated mixer system wind up having the bypass switch on all of the time. This translates into a $10,000 expense that is not being used. The only place where an automated mixer will work well is in rooms with RT60s that are below 1.3 seconds and the NC is 15. There is nothing better than a person operating the mixer for live sound reinforcement.

Balconies are an option under certain conditions and if the sound operator is young and in good health. Stairs are not fun. Setting up a service that has special music or concerts takes three times longer when a balcony is the location of the mixer. Operating from the balcony is a two man job unless you are willing to hold up the service from time to time to let the sound man finish the set up.

From a listening stand point, the balcony must not have an arch or beam above the railing. If there is a beam or arch, it automatically reduces the sound level considerably unless the sound operator is in from the beam by 10 feet.

A good quality custom mixer desk that matches to pew design and color can cost from $900 to $1,500 to make. The mixer desk would only contain the mixer, tape machines, wireless receivers and remote controls for lighting and AV (Screens).

SPEAKER, MICROPHONE WIRES AND WIRING

Microphone Wire
Recommended Pin connections should be:
Pin 1 ground, drain wire
[do not solder pin 1 to the shield]
Pin 2 (+) Hot Red wire
Pin 3 (–) Cold Black wire

This is a standard followed by many contractors and audio companies. However, some manufacturers use pin 3 as hot. Check the manufacturer's specification sheets before you interconnect your electronics as it can often cause some hums and noises when pin 2 and 3 are incorrect.

All microphone { LINES } shall be of a LOW IMPEDANCE TYPE.

All microphone { CORDS } shall be of a LOW IMPEDANCE TYPE.

The line will consist of 2 stranded lines with 1 drain wire or ground and foil shield.

Shield *must* be aluminum foil wrap for permanent wiring. This is currently the best available shielding that will give 100 percent protection from RF. This wire is not suitable for mic cords as the foil shield is prone to breakage or unraveling.

For movable mic cords use a stranded shielded wire. Depending on the manufacturer, the best braided shields are between 85 to 93 percent. However, short mic cords will maintain a shield under stress from bending. Braided wire is not suitable for permanent installation work as it will not give a 100 percent shielding from RF interference.

INSTALLATION DO NOTS

1.Do not run parallel to AC (Alternating Current) electrical lines.

2.Do not run parallel to ballast routes or fluorescent lights.

3.Do not run parallel to unshielded speaker lines which operate over 100 watts.

4.Do not twist wire or have any 90 degree turns.

5.Do not splice mic cables. All mic line should run from stage to mixer without breaks. (There is one exception. If you need to split the signal for TV or Radio broadcasting, you need to go through a splitter box first.)

Conduit Tips

Using Beldon 8451 or equivalent and 14/2 stranded speaker cable:

Conduit Size	Mic Line	Speaker Line
½ in	4	1
¾ in	8	2
1 in	12	3
1¼ in	20	4
1½ in	30	6

Never have more than two 90 degree turns in each conduit run. In any system with more than 8 channels as a starter system, you should consider conduits for the following projects that require conduits:

1.Mic cables
2.Speaker cables
3.Video cables
4.Remote lighting
5.Remote platform lighting
6.Electrical
7.Audience mic inputs
8.Distributed system

HOW MANY MIC LINES?

The number of mic lines you need is also the size of mixer you should have. The only thing to determine a larger mic input requirement is the size of your music program.

Generally, you should have 1 mic input for every 70 square feet of altar or platform space. This does not mean you can't group all of your mics to one location. What this does ensure is that for 99 percent of your churches functions through the year, you will have enough mixer channels and mic input so that you should not have to rent equipment for special services.

To calculate this you have to measure the size of your altar area and divide by 70. The following chart will assist you in your decisions.

1 to 9—8 channel mixer
10 to 13—12 channel mixer
14 to 18—16 channel mixer
19 to 26—24 channel mixer
27 to 34—32 channel mixer
34 to 42—40 channel mixer

There is an exception to the rule. For smaller churches, you should always install at least an 8 channel mixer with 8 mic lines.

As it turns out, audio amplifiers are not very efficient but the amount of current generated does raise some safety and fidelity issues.

If you can, avoid patch bays. It is often cheaper to buy a larger mixer than to go through the expense of building a patch bay. Besides, most churches I know of who have a patch bay never use them more than once a year. Rather, these churches wished they had larger mixers.

MIXER TIP

Always number your stage and the mixer the same with mic stage number starting from left to right from the soundman's position. Therefore, if you pulpit is in the middle of the church and you have a12 channel system, the pulpit control on the mixer will be either 6 or 7. This is very helpful for people who only operate the mixer a few times a year.

SPEAKER WIRE

For those people who are looking into the audio industry and wondering what all of the hocus-pocus is about in speaker wire, have we got some bad news for you. Are the claims of the seller of expensive speaker wire telling you the truth about speaker wire?

DOES WIRE MATTER IN A CHURCH INSTALLATION?

For hi-fi people, this book will not settle any disputes. There are dozens of claims that speaker wire manufacturers are making every day. Sometimes one would think that somebody, with nothing better to do, is figuring out what the next scam for wire will be. I wonder how far they will push before they are unable to get away with it any longer?

For the church, there are several solid reasons for doing some of the things needed for a church installation.

Let's start with a high current amplifier. A 200 watt into 8 ohms 2 channel amplifier can generate a considerable amount of current. There is enough current in two of these amplifiers to trip a 15 amp. fuse.

In some of the large current 200 or 300 watt amplifiers that boast they can work at 2 ohms, the outputs either have a 10 or 15 amp. fuse for each channel. That is enough current to run two drills drilling into steel (intermittently). Fortunately, all of that power is momentary. Different frequencies and rapidly changing volume levels often avoids thermal shutdown in a church setting.

However, we must recognize that even for a millisecond, a 200 watt amplifier at 2 ohms can generate a potential of 10 amps per channel of the amplifier. If this amplifier were 90 percent efficient, it would require a 20 amp service to plug into. As it turns out, audio amplifiers are not very efficient but the amount of current generated does raise some safety and fidelity issues.

Will expensive wire help the church sound system to sound better? In most cases no. The reason is simple. Sonic differences in wire is usually subtle.

Many churches have an NC above 25 dB and/or they have a reverb time longer than 1.8 seconds. This is a very hostile environment for playing games with fidelity. If your church has either of these two problems, there will be too much interference to hear the difference. Furthermore, if the speaker is further than 30 feet away, as it is in most cases, room effect will also interfere with the sound quality.

If your church has neither of these problems, then the wire issue becomes a cost consideration. Some high tech wire can cost over $15 per foot. In a 150 foot run, the cost of 1 wire run would be $2,250. The wire costs more than most professional speakers. Therefore, unless you have the speakers to justify expensive wire, common sense should prevail. And one more point, expensive wire on cheap or poorly designed speakers is a waste of money. If you have to chose between speakers or wire, spend the money on the speakers.

It would be fair to say in most cases, the average person in the pew and the musicians will not hear the wire difference in low cost, budget speakers. The magic wire has to offer can easily be defeated by choosing a better sounding speaker.

DOES WIRE SIZE MATTER?

Yes it does. Wire size determines the amount of current you can send over a given distance. In the appendix there is a wire chart that shows the wire gauge to use over distance with 100 watts - 8 ohm, 4 ohm and 2 ohm loads. The length of the cable run, size of amplifier and the speaker's handling capacity have to be taken into account.

The following list and rules should help in choosing the wire you need for the job:

1.Always run 1 speaker wire to each speaker cabinet or speaker component. If your cluster is a 2 way system with 2 woofers and 4 horns, run 6 speaker lines. If you have 2 full range speakers in the cluster, run two speaker lines. This is a real asset in trouble shooting your system.

2.Whenever possible, keep your speaker cable runs under 100 feet. Otherwise, use 14 gauge wire on runs 100 feet or less. Use 12 gauge wire up to 200 feet. For longer runs, double up on the 12 gauge wire.

3.Don't use inch tip sleeve jacks for your speakers. Some amplifiers will not tolerate a momentary short on the output of the amplifier. Either the speaker will be damaged or the amp will fail.

4.Do not use 16 gauge wire or smaller for amplifiers with an output of 75 watts or higher.

HEARING IMPAIRED SYSTEMS

There are four basic types of hearing impaired systems. Each one has an advantage over the other. In most cases, the FM systems do better mainly because of price but an FM system is not for every church.

The four system types are:

1.Hard Wired Systems
2.Loop Systems
3.FM Systems
4.Infrared Systems

Hard Wired Systems are by default the best quality of the 4 systems. However, it is the most restrictive.

The design of the system is simple. From the mixer you run a distributed cable under the pews you want covered. At each seating position you mount a box with a volume control, tone control and headset input. This can be a line level system or 70 volt system. With a good quality full ear cup headset, you have the best signal to noise ratio.

Drawbacks on the system are obvious, you can't move. Therefore, you have to provide many seats with input boxes which drives the cost up. Churches with concrete floors or closed basement ceilings can not use this option. Installation is labor intensive. Cost of the system varies. For 10 people on two rows of pews, you can spend as little as $400 plus installation. A good quality system will cost about $1,000. At 12 seats, a wireless system becomes more attractive.

FM Systems have recently become the most popular system while many public facilities have standardized with infrared systems.

Loop Systems are making a strong comeback. A popular system in the 60s and early 70s, the loop system almost disappeared in churches. In recent years, loop systems have been very effect in simultaneous translation systems, school classrooms and business meetings.

The design of the system begins with an amplifier, a coil of wire around the area people are seated and various types of receivers. A person with a "T" switch on their hearing aid will not need any additional equipment to hear.

Problems with the loop system are frequency range and uneven coverage. Where the loop system wins out is in situations where you need more than one program taking place at the same time. For example, you can have as many simultaneous translation languages you want for as much space you have available. There is no limit. This may mean people have to sit in designated areas but no wires are required. The other advantage of the loop system is privacy. Once you step out of the field there is no further signal pickup. For some churches, this is an important issue.

FM Systems have recently become the most popular system while many public facilities have standardized with infrared systems. FM systems are an alternative to the infrared systems, which are costly in comparison. The FM systems are every bit as good as the infrared system but they have one draw back. When a person leaves the sanctuary, the signal continues. Some FM systems can transmit over 1,000 feet under good conditions. That means someone could leave to go to the bathroom during the service and not miss a word. It also means that you are subject to public airwaves being listened into with radio scanners. This also applies to FM microphones. If privacy is important, FM will not give you that kind of security. For simultaneous translations, you can have up to 32 channels at the same time.

The *infrared system* has been around long enough to become a standard in public places. Some churches choose this kind of system because the theatre or concert hall down the street uses infrared. Most infrared systems seem to be compatible. The infrared system is secure for privacy and it is very good quality sound. Some of the drawbacks are light and line of sight. In some churches with large windows, the sunlight can add noise to the audio signal. Relocating the emitter can help but a second emitter is often the solution. Another problem concerns the elderly.

Some elderly do not stand when everyone else does. Depending on the receiver, a person standing in front of them can block the infrared signal completely.

As a personal preference, the FM system is the best buy where privacy is not a problem. However, newer infrared systems have been coming down in price. The hardwired system is the lowest price as long as volunteers install it. The loop system is very useful if most of your church members already have "T" switches and then they don't need anything attached to them.

Pop = - a father - a soft drink—a lot of air vibrating the surface of the microphone diaphragm generating an undesirable, low frequency rumble or bang. Words with the letter "P" or "B" are often the cause of blowing too much air = pop

OSHA = Occupational Safety and Health

Poor reverberation is usually when reflected energy is focused back onto the microphones in the platform area. This inhibits the choir, organ and sound system.

Note: Some of the best looking and most impressive spec. sheets in the industry have sometimes been found to be the worst sounding speakers.

Shoot out—OK coral—Frisbee contest

—various speakers set on a stage for side by side comparisons. All speakers must be equalized and set at the same volume using pink noise and a SPL Meter.

Sound travels through wire faster than through the air. Sound travels at 1125 feet per second or 660 miles per hour. A frequency is speed of sound divided by the length of the sound wave.

1125 x 10 ft. = 112.5 hertz
1125 x 3 ft. = 375 hertz
1125 x 6 ins = 3,125 hertz

Series wiring is when you take the negative terminal of a speaker and contact it to the positive terminal of the second speaker. This can be with full-range two way speaker boxes or from driver to driver. This is a low cost method of matching an impedance load. 8 ohms series to 8 ohms = 16 ohms. 8 ohms parallel to 8 ohms = 4 ohms. This most often degrades the overall signal because the signal path includes the voice coil and the crossover in two or three way speakers.

Always leave yourself a way out. By not soldering pin 1 to shield, you can easily isolate your audio components to trouble shoot your system for noises, hums and levels.

It was once said that if you bring too much attention to a problem, people will either try to prove you are wrong or they think that you are hiding something. Isn't this like watching someone else burn their hands in the fire and then putting your hands in the fire to see if you will burn too! Is experiencing it for yourself more important than learning from others?

SPL = Sound Pressure Level

Remember, you must always think of the cosmetics. Everything you do must appear as if it was meant to be there and not just something added on.

To be finished.

During the late 50s and early 60s, many churches did live radio broadcasts of the service. This resulted in many churches building an enclosed sound booth combined with the broadcast and live sound. This was not just a compromise, it was a handcuff to both the live and radio sound.

Always leave yourself a way out. By not soldering pin 1 to shield, you can easily isolate your audio components to trouble shoot your system for noises, hums and levels. If you need additional grounding to reduce a specific problem, pin 1 to ground in the right location can make a world of difference. However, if pin 1 is already grounded throughout the system, trouble shooting can be a nightmare.

Tip—If you don't have any 12 gauge wire available for runs up to 200 feet, you can double up on the 14 gauge wire. db

db Buyer's Guide

Crossovers, Delays, Reverbs, Equalizers, Multi-Effects Processors

On the pages that follow, we present this issue's Buyer's Guide on Crossovers, Delays, Reverbs, Equalizers, and Multi-Effects Processors. The information contained is supplied by the respective manufacturers. Further, if a manufacturer that you seek is not listed, the chances are strong that, as many times as we tried, we could not get information from them.

CROSSOVERS

ALTEC LANSING CORPORATION

Model 1631A is a two-way electronic crossover using plug-in modules to select crossover frequency and configure specific equalization to provide flat power response for various horn/driver combinations. The high-pass output has a level control and the low-pass output has a delay adjustment of 0 to 25 ms.
Dimensions: 1.75 in. X 19 in. X 4.875 in.
Weight: 4.74 lbs.
Price: $708.00
The 1632A Electronic Dividing Network is a dual channel two-way or single channel three-way active crossover, 24 dB/octave, selectable from 50 Hz to 10 kHz; elect. balanced in/out with xfmr in/out optional; 30/60 Hz HP inputs, hard limiters on all 4 outputs; sub-modules to customize response.
Dimensions: 1.75 in. X 19 in. X 9.75 in.
Weight: 8 lbs.
Price: $1,244.00
The 15594A Low Pass Crossover/Equalizer Module is a plug-in module for the 9400 series power amplifier; has 18 dB/octave roll-off pre-programmed at 125 Hz, 500 Hz, 800 Hz, 1250 Hz; customer programmable for other frequencies; programmable 12 dB HP roll-off with pre-sets at 16 Hz or 32 Hz.
Dimensions: 1.6 in. X 2 in.
Weight: 1.6 oz.
Price: $102.00
The 15595A High Pass Crossover/Equalizer Module is a plug-in module for the 9400 series power amplifier; 18 dB/octave roll-off pre-programmed at 125, 315, 500, 800, 1250 Hz; customer programmable for other frequencies; sub-modules available to customize frequency response to horn/driver.
Dimensions: 1.6 in. X 2 in.
Weight: 1.6 oz.
Price: $102.00

ARX SYSTEMS See our ad on page 2

The EC-1 is a low noise Linkwitz-Riley 24 dB Phase correct electronic crossover. It is ideally suited for all studio installation and live sound applications.
Dimensions: 1.75 in. X 19 in. X 6 in.
Weight: 5 lbs
Price: $325.00
EC-2 is dual-channel 2-way/mono user configurable 24 dB Linkwiwitz/-Riley electronic crossover, balanced XLR in and out, also features mono low-frequency sum output, ultra low noise and distortion.
Dimensions: 1.75 in X 19 in. X 6 in.
Price:$425.00

BRYSTON/Brystonvermont Ltd.

The 10PBX is a 2-way stereo 3-way mono crossover with 12 switchable turnover points, 3 switchable slopes of 6, 12 +18 dB per octave; balanced input/output; high frequency gain or cut control and mute switches; S/N ratio of -90 dB; distortion of 0.005 percent; 20 k ohm input impedance and output impedance of 100 ohms.
Dimensions: 1.75 in. X 19 in. X 10 in.
Weight: 12 lbs.
Price: $1,295.00
The 10PBX LR is a 2-way stereo, 3-way mono Linkwitz-Riley slopes with fixed cross-over points; high frequency gain or cut

control; S/N ratio of -90 dB; distortion of 0.05 percent; 20 k ohm input impedance and output impedance of 100 ohms.
Dimensions: 1.75 in. X 19 in. X 10 in.
Weight: 12 lbs.
Price: $1,350.00

DOD ELECTRONICS CORPORATION

The Audio Logic X 34 Stereo 3-way, Mono 4-way Crossover features 24 dB per octave Linkwitz-Riley filter topology; continuously variable, extended range, crossover points; independent level; mute and polarity controls on each output; and 15 Hz, 4th order Butterworth high-pass filters on each input.
Dimensions: 1.75 in. X 19 in. X 6.5 in.
Weight: 3 lbs.
Price: $425.00

ELECTRO-VOICE, INC.

Model XEQ-3/Electronic Crossover features 3-way configurations; allows low-frequency signal delay for source alignment; low-frequency boost for extended bass; step-down operation of TL bass system. Has simple, easy to install modules for compression-driver high-frequency equalization.
Dimensions: 1.73 in. X 19 in. X 7.28 in.
Price: $889.00

FURMAN SOUND, INC.

The Model TX-324 stereo 2-way/mono 3-way crossover features 24 dB/octave rolloff slopes. Field Select allows optimizing filters for long-throw (Butterworth) or near field (Cauer); hard limiters on each output with adjustable threshold provide speaker protection; includes on/off transient muting; ground lift switch; in/out level controls; limit threshold indicators. Optional balanced configuration.
Dimensions: 1.75 in. X 19 in. X 8 in.
Weight: 7 lbs.
Price: $419.00
Model TX-424 stereo 3-way/mono 4 or 5-way crossover has features similar to the Model TX-324.
Dimensions: 3.5 in. X 19 in. X 8 in.
Weight: 9 lbs.
Price: $549.00
Model TX-524 stereo 4-way crossover has features similar to the TX-324.
Dimensions: 3.5 in. X 19 in. X 8 in.
Weight: 9 lbs.
Price: $679.00
The Model TX-3A is a 12 dB/octave tunable crossover that may be used for either stereo 2-way or mono 3-way applications. Includes calibrated input/output level controls, power indicator and ground lift switch. Optional balanced configuration.
Dimensions: 1.75 in. X 19 in. X 8 in.
Weight: 7 lbs.
Price: $319.00

LT SOUND

The ECU-2 is a stereo electronic crossover unit capable of stereo bi-amping as well as stereo tri-amping. Crossover points are continuously variable from 70 Hz to 11 kHz. It has 12 dB/octave Butterworth filters; summed mono output for subwoofer operation; individual phase switches on mid and high bands.
Price: $295.00

PANASONIC PRO AUDIO SYSTEMS

The WS-SP2A Subwoofer Processor (crossover) is networked for use with Ramsa loudspeakers. Includes 6th order alignment network for Ramsa subwoofers; has frequencies of 50 Hz, 80 Hz and 120 Hz; A and B (left and right) inputs; XLR; +4 dB balanced; A and B outputs; phone jack; +4 dB unbalanced; VLF is sum of left and right passed through crossover filter network; mono has phone jack, +4 dB unbalanced.
Dimensions: 1.75 in. X 19 in. X 7.875 in.
Weight: 6 lbs.
Price: $275.00

PEAVEY ELECTRONICS CORPORATION

The V4X is a variable 4-way electronic crossover; low, mid, high and very high level controls; switchable high EQ; balanced outputs; high and low pass filters; at 40 Hz and 20 kHz, calibrated System Gain Control; balanced XLR and ¼ in. input jacks; transformer-balanced XLR and ¼ in. output jacks for all four bandpass outputs.

Dimensions: 1.75 in. X 19 in. X 9 in.

Weight: 8 lbs.

Price: $399.99

The PC4-XL is a totally programmable, all digital four-way (mono) crossover; three-way mono with 4th output as additional LF out; MF out or HF out; two-way mono or stereo; 48 kHz sample rate; 24-bit internal processing; 64 times oversampled A-D; 70 to 650 ms of pre-delay time; up to 10 ms of delay on each output for driver alignment; two balanced inputs, four balanced outputs; selectable filter type.

Price: $799.99

RANE CORPORATION

The AC 22 and AC 23 State Variable Time Correcting crossovers feature 24 dB/octave Linkwitz-Riley filter performance via 41-detent frequency selector controls; built-in variable time delay for phase correction; automatic internal configuration switching; mute switches and input/output level controls with 6 dB gain each.

Dimensions: 1.75 in. X 19 in. X 5.25 in.

Prices: $389.00 and $499.00

The FAC 24 Flex Series Crossover features 24 dB/octave Linkwitz-Riley performance; 24-position digital frequency selector switch for plug-in card accuracy; electronic phase alignment; built-in adjustable CD-horn EQ; mono sub-bass input; and fully balanced ins/outs in half rack package.

Dimensions: 8.5 in. X 1.75 in. X 8 in.

Weight: 4 lbs.

Price: $339.00

The FAC 28 Flex Series Crossover is identical to the FAC 24 except that it features 48 dB/octave slopes to minimize the crossover region and associated problems.

Price: $449.00

SYMETRIX

The 524E multi-mode crossover has four configurable bands; precision cards that set frequencies and slopes; limiter attack/release times; HF horn EQ in/out; flat response from 20 Hz to 50 kHz; 0.01 percent distortion; threshold, gain, mute, phase reverse and phase adjust controls.

Dimensions: 1.75 in. X 19 in. X 9 in.

Weight: 10 lbs.

Price: $1,095.00

WHITE INSTRUMENTS

The DSP 5000 has a digital crossover, delay and parametric equalization all in one rack space; 19 bit, user configurable single channel in, 4 out; remote control capability via PA-422; MIDI or contact closures.

Dimensions: 1.75 in. X 19 in. X 12 in.

Weight: 9 lbs.

Price: $3,400.00

DELAYS

EVENTIDE, INC. **See our ad on page 3**

The BD980 broadcast delay has stereo; 20 kHz frequency response; 10 seconds maximum delay; dump; wait & exit; ramp to zero functions.

Dimensions: 3.5 in. X 19 in. X 12.5 in.

Weight: 15 lbs.

Price: $5,495.00

The BD941 broadcast delay features mono; 20 kHz frequency response; fixed delay; delete and bypass functions.

Dimensions: 1.66 in. X 19 in. X 9.4 in.

Weight: 13 lbs.

Prices: $1,795.00 (6 seconds) or $2,195.00 (12 seconds)

The BD942 broadcast delay features stereo; 20 kHz frequency response; fixed delay; delete and bypass functions.

Dimensions: 1.66 in. X 19 in. X 9.4 in.

Weight: 13 lbs.

Prices: $1,995.00 (3 seconds) or $2,395.00 (6 seconds)

KLARK-TEKNIK ELECTRONICS, INC.

The DN716 is a one in, 3 out 16 bit digital delay line with less than 90 dB dynamic range, 20 Hz-20 kHz, unweighted. Delay times from 0-1.3 seconds in minimum increments of 20us; input level indicator and level control; non-volatile memory; electronically balanced input; unbalanced outputs; transformer balancing optional.

Dimensions: 1.75 in. X 19 in. X 11.75 in.

Weight: 5.5 lbs.

Price: $1,657.00

The DN726 has two in, two out stereo 16 bit digital delay; 100 percent stereo tracking; control functions lock out and non-volatile memory; dynamic range of less than 90 dB; 20 Hz-20 kHz unweighted; electronically balanced inputs; unbalanced outputs; transformer balancing optional.

Dimensions: 1.75 in. X 19 in. X 11.75 in.

Weight: 5.5 lbs.

Price: $2,995.00

The DN726V is very similar to the DN726, but will display in either milliseconds or fields, and is switchable between PAL and NTSC standards (internally). Also has a (4) GPI control function to automatically follow delay introduced by other devices. For use in video applications.

Dimensions: 1.75 in. X 19 in. X 11.75 in.

Weight: 5.5 lbs.

Price: $2,995.00

The DN775 is a stereo disc-cutting delay, switchable to select 33 or 45 RPM; 100 percent stereo tracking; less than 90 dB dynamic range; 20 Hz-25 kHz; unweighted, electronically balanced inputs; transformer balanced outputs (standard); frequency response of 20 Hz-25 kHz + 1 dB, any level, any delay.

Dimensions: 12.75 in. X 19 in. X 11.75 in.

Weight: 5.5 lbs.

Price: $3,995.00

LEXICON, INC.

The LXP-15 combines range of reverb, pitch shifting and delay effect with fast editing of presets, MIDI control in a single rack-space package and user interface. Offers 128 preset effects with up to five pages of parameters per effect, and the ability to store 128 of your own effects and five external analog inputs for foot switches or pedals.

Dimensions: 1.75 in. X 19 in. X 13.9 in.

Weight: 12 lbs.

Price: $1,050.00

PANASONIC PRO AUDIO SYSTEMS

The WZ-9375 has 2 inputs with 2 outputs, alternately, 1 input with 4 outputs; up to 654 msec @ 100 kHz sampling rate; 10 microseconds to 1 millisecond of delay time steps; 50 kHz or 100 kHz sampling rate; frequency response of 20 Hz to 20 kHz, +0.5, -2 dB at 100 kHz sampling rate; dynamic range of more than 90 dB; less than 20 micro-seconds of group delay; less than 0.03 percent at 100 kHz sampling rate T.H.D.

Dimensions: 3.5 in. X 19 in. X 13.75 in.

Weight: 19.5 lbs.

Price: $4,500.00

ROLAND PRO AUDIO/VIDEO GROUP

Features digital companding PCM system equivalent to a 16-bit A/D/A converting system; dynamic range more than 100 dB; frequency response ranges from 10 Hz to 17 kHz with delay time from 0 to 1,500 ms; can store up to 8 different programmable memories.

Dimensions: 1.75 in. X 19 in. X 11.75 in.

Weight: 11 lbs.

Price: $1,095.00

SOUND CONCEPTS, INC.

The SSD550 surround and ambience delay unit features two channels of delay; 5 to 50 ms; may be switched to sequential for 10 to 100 ms.. variable mix of original and delayed signals available; passive surround decoder for film; S/N 90 dB; response 10 to 8000 Hz.

Dimensions: 3.5 in. X 19 in. X 9 in.

Price: $975.00

SYMETRIX, INC

Model 402 is a 1 input, 2 output digital delay with a 104 dB dynamic range, 12 Hz to 20 kHz frequency response, 19-bit A/D conversion, 884 ms. maximum delay; display is in msec, feet and meters.

Dimensions: 1.75 in. X 19 in. X 7.5 in.

WHITE INSTRUMENTS

The DSP 5000 has a delay, parametric equalization and digital crossover all in one rack space; 19 bit, user configurable single channel in, 4 out; remote control capability via PA-422, MIDI or contact closures.
Dimensions: 1.75 in. X 19 in. X 12 in.
Weight: 9 lbs.
Price: $3,450.00

EQUALIZERS

ALESIS STUDIO ELECTRONICS

The MEQ-230 Precision Equalizer has dual 30 band, ⅓ octave EQ in single 19 in. rack space; interface provided by means of ¼ -in. and RCA jacks; center frequencies range from 25 Hz to 20 kHz and are set to ANSI/ISO standards; each band provides 12 dB cut/boost; in/out switch.
Dimensions: 1.75 in. X 19 in. X 4 in.
Weight: 2.5 lbs.
Price: $249.00

ALTEC LANSING CORPORATION

The 1412A stereo 15 band ⅔ Octave Equalizer is two-channel with 15 variable-Q filters, 6 or 12 dB boost/cut, low-cut sector switch, balanced in/out, barrier-strip only.
Dimensions: 1.75 in. X 19 in. X 9.3 in.
Weight: 6 lbs
Price: $480.00
The 8558B Programmable Microaudio Equalizer offers eight memories; only one rack space; no front panel controls; 28 ⅓ octave filters with 12 dB of cut/boost; fixed HP/LP filters; elect. balanced in/out; xfmr in/out optional barrier strip only.
Dimensions: 1.75 in. X 19 in. X 7 in.
Weight: 5.9 lbs.
Price: $2,070.00
The 1750A Cut-Only ⅓ Octave Mono Equalizer has 28 constant-Q filters from 31.5 Hz to 16 kHz; 15 dB of attenuation per filter; 20 dB of broadband gain; variable HP/LP filters; elect. balanced in/out with optional xfmr, XLR and barrier strip.
Dimensions: 3.5 in. X 19 in. X 9.75 in.
Weight: 10.7 lbs.
Price: $1,250.00
The 1753A Boost-Cut ⅓ Octave Mono Equalizer has 28 constant-Q filters from 31.5 Hz to 16 kHz; 12 dB cut/boost per filter; 20 dB broadband gain; variable HP/LP filters; elect. balanced in/out with optional xfmr, XLR and barrier strip.
Dimensions: 3.5 in. X 19 in. X 9.75 in.
Weight: 10.7 lbs.
Price: $1,250.00

APPLIED RESEARCH AND TECHNOLOGY

The HD 31, Model 350 is an active balanced ⅓ octave 31 band equalizer featuring constant Q filters; 60mm sliders; switchable 15 and 7.5 dB level scale; switchable subsonic and ultrasonic filters; hard bypass at no power and S/N of 115 dB.
Dimensions: 3 in. X 19 in. X 6.25 in.
Weight: 8 lbs.
Price: $425.00
The HD 15, Model 340 is an active balanced ⅔ octave 15 band equalizer with constant Q filters; 60mm sliders; optional XLR connections; switchable subsonic and ultrasonic filters; hard bypass at no power and S/N of 115 dB.
Dimensions: 3 in. X 19 in. X 6.25 in.
Weight: 8 lbs.
Price: $425.00

ARX SYSTEMS See our ad on page 2

The EQ 30 and 60 are ultra low noise constant Q ⅓ octave graphic equalizers featuring balanced XLR and jack inputs/outputs and switchable 6 dB or 15 dB of cut/boost.
Dimensions: The EQ 30 is 3.5 in. X 19 in. X 10 in.
The EQ 60 is 5.25 in. X 19 in. X 10 in.
Weight: Each model is 8 lbs
Prices: EQ 30—$899.00; EQ 60—$1,349.00

The Multi Q is a six channel/band fully variable parametric EQ. Featuring ARX Auto Patch, the Multi Q allows the user to select any number of channels without the need for patch cables.
Dimensions: 1.75 in. X 19 in. X 6 in.
Weight: 4 lbs
Price: $710.00

dbx PROFESSIONAL PRODUCTS, A DIVISION OF AKG ACOUSTICS, INC.

The 905 three-band parametric equalizer features instant before/after comparisons available by switch bypass; switchable notch mode on each band, and a choice of shelving or symmetrical peak/dip on outerbands.
Dimensions: 5.25 in. X 1.5 in. X 9.5 in.
Weight: 0.75 lbs.
Price: $499.00
The 1531X graphic equalizer has selectable 15 band stereo (2/3 octave) or 31 band mono (⅓ octave) equalizer on ISO centers; constant-Q and symmetrical peak/dip curves with selectable 7.5 or 15 boost or cut; and switchable HP filtering at 20 Hz, 60 Hz or 120 Hz, XLR and ¼ in. inputs and outputs.
Price: $419.00

DOD ELECTRONICS CORPORATION

The DigiTech MEQ 28 Mono 28-band MIDI Programmable Graphic EQ is a two space, rack-mount mono graphic EQ that is fully MIDI controllable and programmable with 99 user-definable programs. It features 28 bands of 12 dB cut/boost equalization.
Dimensions: 3.5 in. X 19 in. X 8.5 in.
Weight: 7 lbs.
Price: $599.95
The DOD 830 Stereo 15 band per channel, ⅔ Octave Graphic EQ is a two rack space EQ featuring 20 Hz to 20 kHz equalization; 12 dB cut/boost; low cut filter; 90 dB S/N; THD 0.006 percent and 5 percent frequency tolerance.
Dimensions: 3.5 in. X 19 in. X 8.5 in.
Weight: 7 lbs.
Price: $249.95

ELECTRO-VOICE, INC.

Model 2710 ⅓ octave graphic EQ features 27-band, ⅓-octave equalizer; constant range variable-Q filters; minimal interference between adjacent filters; user-selectable high- and low-pass filters; built-in pink-noise generator for noise masking; system equalization and other applications.
Dimensions: 3.5 in. X 19 in. X 10.25 in.
Weight: 11.5 lbs.
Price: $1,130.00

FURMAN SOUND, INC.

Q-Series Graphic Equalizers Long Throw Slider Models feature maximum length 60 mm sliders, constant-Q response, "Straight Wire" bypass, input and output level controls, 4-segment LED meters for both input and output, low and high cut knobs, ±12 dB of equalization. Both models available with XLR-balanced in/out option.
Dimensions: 3.5 in X 19 in. X 8 in.
Weight 10 lbs.
Price: Q-152—$489.00; Q-302—$469.00
Q-series Graphic Equalizers Short Throw Models feature Constant-Q response, "Straight Wire"bypass, output level controls, 4-segment LED output meters, low cut buttons, ±6 dB/±12 dB rear-panel range switches with front panel indicators (exept Q-541 which is fixed at ±10 dB). All but the Q541 are available with an XLR-balanced input/output option.
Dimensions: 1.75 in. X 19 in. X 8 in. (single rack space units; 3-5 in. X 19 in. X 8 in. (double rack space units)
Weight: 8 lbs. (single rack space); 10 lbs. (double rack space)
Prices: Q-151—$399.00; Q-301—$389.00; Q-602—$699.00; Q-541—$399.00.
The Model PQ-4 parametric equalizer has constant-Q equalization curves; peak/shelf switches on top and bottom bands; extra wide range of bandwidth and EQ adjustment. Includes input level control; EQ in button, as well as overload; EQ status and power indicators; high and low level inputs/outputs; and footswitch jack, allowing use as a preamp. Balanced configuration is optional.
Dimensions: 1.75 in. X 19 in. X 8 in.
Weight: 6 lbs.
Price: $379.00

KLARK-TEKNIK ELECTRONICS, INC.

The DN410 is a dual (5) band/Single (10) band parametric equalizer with 100 percent frequency overlap on all bands; +15/-25 dB boost/cut; 1/12 to 2 octave bandwidth; separate variable high/low pass filters (each channel); separate EQ in/out switch on all bands plus overall noise less than -94 dBm; 20 Hz-20 kHz, unweighted.
Dimensions: 3.5 in. X 19 in. X 9.25 in.
Weight: 10 lbs.
Price: $1,275.00
The DN405 is the same as above, but with single (5) band only.
Dimensions: 1.75 in. X 19 in. X 9.25 in.
Weight: 7.7 lbs.
Price: $850.00
The DN360 is a dual channel 30 band 1/3 octave graphic equalizer with switchable 12 dB/6 dB scale on faders; switchable high pass filters; electronically balanced inputs, unbalanced outputs; transformer balancing optional; noise less than 90 dBm; 20 Hz-20 kHz unweighted.
Dimensions: 5.25 in. X 19 in. X 8 in.
Weight: 10 lbs.
Price: $1,795.00
The DN300 is a single channel 30 band 1/3 octave equalizer with continuously variable high and low pass filters; switchable 12 dB/6 dB fader scale; noise less than 90 dBm; 20 Hz-20 kHz unweighted; electronically balanced input; unbalanced output; transformer balancing optional.
Dimensions: 3.5 in. X 19 in. X 8 in.
Weight: 7.7 lbs.
Price: $1,150.00
The DN301 is a single channel 30 band 1/3 octave Cut only graphic equalizer with continuously variable high and low pass filters; switchable 12 dB/6 dB fader scale; electronically balanced input; unbalanced output; transformer balancing optional; noise less than 94 dBm; 20 Hz-20 kHz unweighted.
Dimensions: 3.5 in. X 19 in. X 8 in.
Weight: 7.7 lbs.
Price: $1,150.00
The DN332 is a dual 16 band 2/3 octave graphic equalizer with +12 dB boost/cut; switchable high pass filters; electronically balanced inputs; unbalanced outputs; transformer balancing optional; noise less than -90 dB;, 20 Hz-20 kHz unweighted.
Dimensions: 3.5 in. X 19 in. X 8 in.
Weight: 7.7 lbs.
Price: $1,095.00

LT SOUND

The PEQ is a dual-channel, 4-band parametric equalizer with selectable peak/dip or shelving response on upper or lower bands, overall hard-wire bypass and individual bypass on middle 2 bands. Bandwidth variable from 0.15 to 2 octaves.
Price: $595.00
The PEQ-1 is a single-channel version of the PEQ-2. Utilizes a single-rack space.
Price: $349.00

ORBAN, A DIVISION OF AKG ACOUSTICS, INC.

The Model 642B dual channel/stereo is a fully parametric equalizer with 4 bands per channel, switchable to 8 channels mono; each band with separate bypass, Q, frequency and fine tuning control; high pass and low pass filters per channel; minimum 40 dB notch per channel.
Dimensions: 3.5 in. X 19 in. X 11.25 in.
Price: $1,200.00
The models 672A/674A mono/stereo 8-band graphic parametric equalizers have long throw faders controlling boost and cut for each band; high pass and low pass filters with separate outputs for use as 2-way crossover.
Dimensions: 3.5 in. X 19 in. X 5.25 in.
Prices: $725.00 for the 672A
$1,525.00 for the 674A
Model 764A features programmable, digitally-controlled parametric equalizer version of the 642B; controls up to 99 channels of masters and slaves; stores up to 99 presets; has four bands, dual channel, with high and low pass filters; and programmable input attenuator.
Dimensions: 3.5 in. X 19 in. X 9.625 in.
Price: starting at $1,900.00, depending on configuration

OXMOOR CORPORATION

The DEQ-1 High Resolution programmable ⅓ octave equalizer has 29 ⅓ octave filters adjustable in ½ octave spacing; 8 presets with security; balanced inputs/outputs; PA-422.
Dimensions: 1.72 in. X 19 in. X 13.5 in.
Weight: 13 lbs.
Price: $1,060.00

The DEQ-II High Resolution Programmable ⅓ octave Equalizer has 29 ⅓ octave filters adjustable in ½ dB steps; high/low pass filters selectable on ⅙ octave spacing; large LCD display and front panel controls make programming simple; 8 presets with security; balanced inputs/outputs; PA-422.
Dimensions: 3 in. X 19 in. X 13.5 in.
Weight: 15 lbs.
Price: $1,400.00

PEAVEY ELECTRONICS CORPORATION

The Autograph II is a programmable, MIDI controllable 28-band graphic eq with RTA , pink noise generator, multiple discrete room sampling, continuos room sampling, factory and user defined target curves, curve compare and averaging, feedback finder, low-cut filter.
Dimensions: 1.75 in. X 19 in. X 9 in.
Weight: 6.25 lbs.
Price: $499.00

The EQ 215 has two ⅔ octave graphic equalizers; 6 or 12 dB ranges; level control; EQ bypass; low and high cut filters, balanced/unbalanced inputs and outputs, ISO centers.
Price: $399.99

The EQ 31 has 31 bands of graphic EQ; IS0 centers; 6 or 12 dB ranges; level control; low and high cut filters; balanced/unbalanced inputs and outputs.
Price: $399.99

RANE CORPORATION

The ME 30 and ME 15 MicroGraphic Equalizers feature constant-Q ⅓ and stereo ⅔ octave performance in single rack space packaging; with switchable 6/12 dB boost/cut; input level; hard-wire bypass and 20 mm center-detent sliders.
Dimensions: 1.75 in. X 19 in. X 5.25 in.
Weight: 5 lbs.
Prices: $499.00 for either

The GE 27 and GE 14 Graphic Equalizers feature constant-Q ⅓ and ⅔ octave performance in two rack space packaging; with 45 mm center-detent sliders; level control; hard wire bypass; low noise and low distortion circuitry.
Dimensions: 3.5 in. X 19 in. X 8.5 in.
Weight: 9 lbs.
Prices: $499.00 for GE 27 and $529.00 for GE 14

The MPE SERIES Programmable Equalizers feature the MPE 28 ⅓ octave and MPE 14 Dual ⅔ octave equalizers with 128 memory locations plus a software package that enables curve weighting (adding 2 curves together); real time program changes; remote control; copying; data-dumping, full MIDI mapping and other functions.
Dimensions: 1.75 in. X 19 in. X 8.5 in.
Weight: 6 lbs.
Prices: $749.00 for the MPE 28 and $799.00 for the MPE 14.

NEQ 128 NEQ 228 and NEQ 256 Network equalizers. The NEQ 128 is a mono 1/3 octave, the NEQ 228 is stereo, and the NEQ 156 is a mono 1/6 octave equalizer. Features include almost dead-fron panels, controllable by RS-232 (standard, software included),. Cards available to communicate via other network protocols, 16 internal non-volatile memories recallable by contact closure. Balanced terminal strip connectors.
Dimensions: 1.75 in X 19 in X 8.5 in.
Weight: 6 lbs
Prices: $699.00, $9990.00, and $999.00

PE 15 and PE 17 parametic equalizers; the PE 15 is a 5-band with overlapping bands, 1 and 5 can be switched to shelf mode, the PE 17 is a 5-band with additional high and low cut filters with all bands sweepable from 20 Hz to 20 kHz, bandwith range is from 1/30th to 1.5 octaves, balanced terminal strip connector.
Dimensions: 1.75 in. X 19 in. 5.25 in.
Prices: 399.00 and $599.00

SABINE MUSICAL MANUFACTURING COMPANY, INC. **See our ad on page 4**

FXB-900 Feedback Exterminator utilises digital processing technology and automatically detects and eliminates feedback quickly by assigning very narro notch filters to the ringing frequencies, without changes in sound quality.

Dimensions: Single space rack mount
Price: $599.00
ADF-1200 and ADF-2400 Workstations are the first complete set of adaptive digital filters. They include an extensive array of fully programmable functions and contain the FXB Feedback Exterminator.
Dimensions: 2 U rack mount
Prices: ADF-1200—$1,695.95; ADF-2400—$2,627.95

SOUNDCRAFTSMEN

The PRO-EQ 440 PRD is a C-MOS 0.1 dB Differential/Comparator Third Octave featuring C-MOS digital switching; two independent channels of EQ; ⅓ octave 40 Hz/1 kHz; alternate ⅓ octave 1 kHz/16 kHz; exclusive differential/comparator unity-gain circuits; balancing LEDs for instant adjustment to unity gain; pre-post EQ loops and EQ defeat switch.
Dimensions: 3.5 in. X 19 in. X 11 in.
Weight: 15 lbs.
Price: $549.00

SUMMIT AUDIO, INC.

The EQP-200A is a dual program equalizer utilizing tube gain make-up stages with 990, balanced output. All units are hand-crafted and burned in for ten days or more.
Dimensions: 3.5 in. X 19 in. X 10.5 in.
Weight: 19 lbs.
Price: $2,100.00
The EQF-100 Full Range Vacuum Tube Equalizer is a full-range, single channel, four band equalizer with Hi/Lo pass filter section; musically selected center frequencies; with bands one and four peaking of shelving selectable; 990, balanced output.
Dimensions: 3.5 in. X 19 in. X 10.5 in.
Weight: 21 lbs.
Price: $2,200.00

SYMETRIX

The SX201 parametric EQ has three overlapping bands; +15 dB boost; -30 dB cut; 0.05 octave to 3.3 octaves bandwidth; 119 dB S/N ratio; 20 Hz to 20 kHz response (+0, -1 dB).
Dimensions: 1.75 in. X 8 in. X 5.5 in.
Weight: 5 lbs.
Price: $259.00

WHITE INSTRUMENTS

The Model 4700/4700-2 is a digitally-controlled ⅓ octave equalizer; has one or two channel; controllable from the front panel with password protection or software control via RS-232 or PA-422 interface with Pilot 447 software provided. Model 4700IQ, 47002IQ are the same as the model 4700,4700-2 above except they come with Crown IQ 2000 control.
Dimensions: 1.75 in. X 19 in. X 12 in.
Weight: 9 lbs.
Prices: $950.00 mono/$1,425.00 dual; IQ Versions to be announced.
Model 4710 is a digitally-controlled ⅙ octave 55 band equalizer in one rack space. Controllable from the front panel with password protection; has 10 memory locations and 10 separate preset locations in non-volatile storage.
Dimensions: 1.75 in. X 19 in. X 12 in.
Weight: 9 lbs.
Price: $1,550.00
The Model 4650/4660 is a 60 mm slider controlled ⅓ octave filters 31.5 Hz-16 kHz; 12 dB, 10 dB gain; variable high/low pass on 4660; XLR and ¼ jack connectors; input/output transformer available (4622).
Dimensions: 3.5 in. X 19 in. X 5 in.
Weight: 7 lbs.
Prices: 4650 is $699.00/4660 is $750.00
Model 4675 is a 60 mm slider controlled stereo ⅔ octave; filters 40 Hz-16 kHz 12 dB range, 10 dB gain; variable high pass, fixed low pass; XLR connections; servo-balanced differential input/output circuit.
Dimensions: 3.5 in. X 19 in. X 5 in.
Weight: 7 lbs.
Price: $795.00
Model 4400 has L-C active ⅓ octave filters 31.5 Hz-16 kHz; 10 dB range; variable high/low pass; 3 outputs and crossover socket for optional bi-amp/tri-amp operation; input/output transformers available; noise -90 dBu worst case.
Dimensions: 3.5 in. X 19 in. X 8 in.

Weight: 15 lbs.
Price: $1,050.00
The DSP 5000 has 12 bands of parametric equalization, digital crossover and delay all in one rack space; 19 bit, user configurable single channel in, 4 out; remote control capability via PA-422, MIDI or contact closures.
Dimensions: 1.75 in. X 19 in. X 12 in.
Weight: 9 lbs.
Price: $3,400.00

MULTI-EFFECTS PROCESSORS

APPLIED RESEARCH AND TECHNOLOGY
The SGX-2000, Model 500, is for guitar. Tri-channel programmable tube and solid state preamp with stereo digital effects; full 20 kHz bandwidth; 24 bit processing; seven band equalizer.
Dimensions: 3 in. X 19 in. X 9 in.
Weight: 15 lbs.
Price: $829.00
The SGX NightBass, Model 490, is for bass guitar. Tri-channel programmable tube and solid state preamp with stereo digital effects; full 20 kHz bandwidth; 24 bit processing; seven band equalizer and selectable crossover.
Dimensions: 3 in. X 19 in. X 9 in.
Weight: 15 lbs.
Price: $839.00
The Power Plant, Model 410, is a dual channel guitar preamp. Channels are switchable between clean and dirty with their own separate EQ effects loop; separate guitar, line and power amp and headphone outputs.
Dimensions: 1.5 in. X 19 in. X 10 in.
Weight: 11 lbs.
Price: $329.00

DOD ELECTRONICS CORPORATION
The Digitech TRS-24 digital reverb and multi-effect processor, true stereo with multiiple input/output configurations, 128 factory presets, 128 programmable settings, 24-bit signal path and 48bit internal datadigital sampling, 4-octave pitch shifter, choruses, flangers, programmable eq, noise reduction, auto panning...
Dimensions: 19 in X 1.75 in. X 9 in.
Price: $799.00
The DigiTech DSP 256XL Digital Effects Processor features 21 different studio-quality effects, up to 4 simultaneously, and is built tough enough for road use.
Dimensions: 1.75 in. X 19 in. X 8.5 in.
Weight: 5.5 lbs.
Price: $439.95
The DigiTech DSP 16 Effects Processor contains 128 MIDI changeable programs utilizing 16 different reverb and delay effects; a 3-band EQ provides tailoring of the sound.
Dimensions: 1.75 in. X 19 in. X 8.5 in.
Weight: 4.5 lbs.
Price: $299.95

LT SOUND

The ECC is a digital delay system with microplate reverb. Delay and reverb may be used simultaneously or independently; delay range is from 1 ms to 1 s; effects include doubling, chorus, flange, plate reverb with delay, acoustic chamber and tremolo.
Dimensions: 1.75 in. X 19 in. X 7.5 in.
Price: $995.00

PEAVEY ELECTRONICS CORPORATION
The Ultraverb II is a digital multi-effects processor with extended bandwidth; 256 internal programs; 128 user editable effects; real time MIDI control; each preset transfer step reversible up to last keystroke; full MIDI access.
Dimensions: 1.75 in. X 19 in. X 6.5 in.
Weight: 6 lbs.
Price: $349.99
All effects of the AddVerb II, except reverbs and specials, may be modified and stored at any of 199 program presets; full MIDI control capability; 50 reverb presets; 40 programmable delay/echo and modulated presets; 10 combinations; presets may be mapped to any of 128 MIDI program numbers.
Dimensions: 1.75 in. X 19 in. X 8 in.
Weight: 6 lbs.
Price: $319.00
ProFex II is a programmable MIDI controlled multi-effects preamp featuring digital stereo effects; RAM slot; independent

effects blocks can be combined in series or parallel in any order to form multi-effect chains; each effect block has independent mix and level control; programmable noise gate in all programs; 128 presets mapped to 128 user patches, MIDI or footswitch access.
Price: $549.99

SYMETRIX INC.
Model 601 Digital Voice Processor includes fully parametric equalization, dynamic filter/noise reduction, de-essing, compression, automatic gain control, expander gate, and stereo echo.
Dimensions: 1.75 in. X 19 in. X 10 in.
Price : $1,995.00
The 528 Voice Processor has a mic preamp with 48V phantom, compressor, limiter, downward expander,, 3-band parametric equalizer/notch filter and de-esser all in one rack space.
Dimensions: 1.75 in. X 19 in. X 8.5 in.
Price: $679.00

REVERBS

ALESIS STUDIO ELECTRONICS

The Quadraverb Plus features 1.5 seconds of delay memory for sampling; independently adjustable multi-tap delays; programmable panning; new ring moduator and resonator configuration along with the 20 K bandwidth reverb; delay; chorus; flanging; parametric EQ; leslie simulator; and comprehensive onboard digital effects mixing system of the original Quadraverb.
Price: $499.00
The Microverb III, a 16-bit stereo digital reverb and effects processor, has 256 preset programs: 112 reverbs; 32 gated/reverse reverbs; 80 delays; and 32 multi-tap and effects programs. The 19 in. rack mountable unit features 15 kHz bandwidth and two bands of EQ (100 Hz and 4 kHz) for fine tuning of programs.
Price: $249.00
The Midiverb III is a digital stereo multi-effects unit capable of generating four effects at a time: delay; reverb; and chorus or flange. Features 200 memory locations, with 100 reserved for factory presets. Real-time MIDI control.
Price: $399.00

APPLIED RESEARCH AND TECHNOLOGY

The Multiverb ALPHA, Model 470, is a 24 bit full 20 kHz digital signal processor capable of combining seven effects at once. Has programmable seven band EQ; reverb; two octave of Pitch Transposing; 20 delay types including sampling.
Dimensions: 1.5 in. X 19 in. X 9.25 in.
Weight: 11 lbs.
Price: $499.00
The Multiverb LT, Model 420, is a studio digital effects signal processor with instant access to 192 pre-programmed presets of up to three effects at once. Effects include reverb; delay; chorus; flanging; gated and reverse reverb and panning.
Dimensions: 1.5 in. X 19 in. X 9.25 in.
Weight: 10 lbs.
Price: $299.00

EVENTIDE, INC. **See our Ad on page 3**

The DSP4000 Ultra-Harmonizer has AES/EBU digital in/out for all-digital signal manipulation and a modular software architecture.
Dimensions: 3.5 in. X 19 in. X 12.5 in.
Weight: 12 lbs.
Price: $4,495.00
The H3500 Ultra-Harmonizer features new dynamic effects; 16-bit, 44.1 kHz sampling, pitch shifting, dense reverbs, stereo delays and choruses, 24 algorithms, and 400 presets; sampling is available in 22 or 95 seconds.
Dimensions: 3.5 in. X 19 in. X 13.5
Weight: 13 lbs.
Price: $3,495 (H3500-sfx w/22 second sampling); $4,495.00 (H3500-dfx/e w/95 second sampling)
The H3000SE Studio Enhanced Ultra-Harmonizer has 19 algorithms, including vocoder; dense reverb; multishift; band delay; string modeller; phaser, stutter and patch factory; 200 presets; function generator (programmable parameter modulation); soft functions (user-definable Soft Keys).
Dimensions: 3.5 in. X 19 in. X 13.5 in.
Weight: 13 lbs.
Price: $2,995.00
The H3000B Broadcast/Post Ultra-Harmonizer has 14 algorithms, including TimeSqueeze (stereo time compression/expansion with machine control); stutter and patch factory (white noise generator, filters, pitch shifters, delay lines and more); 80 presets; function generator, soft functions.

Dimensions: 3.5 in. X 19 in. X 13.5 in.

Weight: 13 lbs.

Price: $2,995.00

The H3000S Studio Ultra-Harmonizer has 11 algorithms including diatonic shift; dual shift; layered shift; stereo shift; reverse shift; swept combs; reverb factory; ultra-tap; dual digiplex; long digiplex; 48 Steve Vai presets; 58 factory presets.

Dimensions: 3.5 in. X 19 in. X 13.5 in.

Weight: 13 lbs.

Price: $2,495.00

KLARK-TEKNIK ELECTRONICS, INC.

The DN780 offers full control over several parameters including predelay time; level and pattern of reflections; low and high frequency decay times; and room size. Supplied with remote controller; has 50 non-volatile user memories. 32 bit circuitry.

Dimensions: 3.5 in. X 19 in. X 12.25 in.

Weight: 16.5 lbs.

Price: $2,995.00

LEXICON, INC.

The 300 Digital Effects System is designed for the small professional studio. Features include two stereo inputs/outputs (balanced XLR) and digital inputs/outputs in the AES/EBU and SPDIF formats. The 300 features 50 event effects recall via SMPTE time code; full MIDI implementation; and 96 dB signal-to-noise ratio; and reverb; ambiance; stereo pitch shifting and mastering type algorithms.

Dimensions: 3.5 in. X 19 in. X 13.6 in.

Weight: 18.9 lbs.

Price: $4,795.00

LT SOUND

The RCC reverb control center is a complete microplate reverb system for use with or without a mixing board. It has 2 mic inputs; inputs for 2 additional stereo sources; and output for a tape recorder, plus 3-band equalization.

Dimensions: 1.75 in. X 19 in. X 7.5 in.

Weight: 7 lbs.

Price: $695.00

PEAVEY ELECTRONICS CORPORATION

The Univerb II has 128 stereo 16-bit effects; bandwidth of 20 Hz to 12 kHz; VLSI technology; remote bypass capability; stereo and mono to stereo capability; single rack space chassis.

Dimensions: 1.75 in. X 19 in. X 8.125 in.

Weight: 5 lbs.

Price: $249.99

ROLAND PRO AUDIO/VIDEO GROUP

The R-880 digital reverb has four independent DSPs; reverb; non-linear reverb; early reflections; chorus; delay; EQ; compression; flat frequency response; 90 dB dynamic range; analog, AES/EBU digital I/O connections; accommodates 48 kHz, 44.1 kHz signals.

Dimensions: 3.56 in. X 19.18 in. X 16.56 in.

Weight: 22 lbs.

Price: $3,995.00

The GC-8 is a graphic controller remote control unit for the R-880 featuring large, 256 X 64 dot LCD; five rotary knobs and numeric keypad for easy programming; memory card slot for storing and loading programs.

Dimensions: 2 in. X 13.125 in. X 6.94 in.

Weight: 2 lbs., 10 oz.

Price: $850.00

Addresses

Alesis Studio Electronics
3630 Holdrege Avenue
Los Angeles, CA 90016

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

Altec Lansing Corporation
10500 West Reno Avenue
P.O. Box 26105
Oklahoma City, OK 73126

Applied Research and Technology
215 Tremont Street
Rochester, NY 14608

ARX Systems
28271 Bond Way
Silverado, CA 92676

Bryston/Brystonvermont Ltd.
979 Franklin Lane
Maple Glen, PA 19002

dbx Professional Products, a division of AKG Acoustics, Inc.
1525 Alvarado Street
San Leandro, CA 94577

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

Electro-Voice, Inc.
600 Cecil Street
Buchanan, MI 49107

Eventide, Inc.
One Alsan Way
Little Ferry, NJ 07643

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

Klark-Teknik Electronics, Inc.
200 Sea Lane
Farmingdale, NY 11735

Lexicon, Inc.
100 Beaver Street
Waltham, MA 02154

LT Sound
7980 LT Parkway
Lithonia, GA 30058

Orban, a division of AKG Acoustics, Inc.
1525 Alvarado Street
San Leandro, CA 94577

Oxmoor Corporation
2111 Parkway Office Circle
Birmingham, AL 35244

Panasonic Pro Audio Systems
6550 Katella Avenue
Cypress, CA 90630

Peavey Electronics Corporation
711 A Street
Meridian, MS 39301

Rane Corporation
10802 47th Avenue West
Everett, WA 98204

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

Sabine Musical Manufacturing Company, Inc.
4637 Northwest 6th Street
Gainesville, FL 32609

Sound Concepts Inc.
Post Office Box 135
Brookline, MA 02146

Soundcraftsmen
2200 South Ritchey
Santa Ana, CA 92705

Summit Audio, Inc.
P.O. Box 1678
Los Gatos, CA 95031

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

White Instruments
1514 Ed Bluestein Boulevard
Austin, TX 78721

From page 33 Historical Perspectives

Crooner Bing Crosby was the first popular entertainer to foresee the advantages of audio tape over then then widely used 16-inch transcription discs. A check from him started Ampex going in pro audio with the first American machine, the Ampex 200. By the mid-fifties when this promo picture was taken, Ampex had produced many 300s and their first portable, the Ampex 600 which this pose was promoting.

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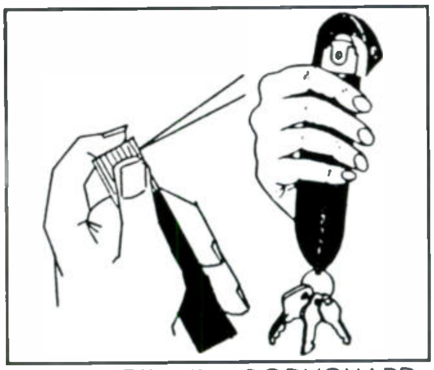
Recording Studios and Broadcast stations work around the clock, so more than most need a safe personal protection device to ward off those who present a threat to safety.



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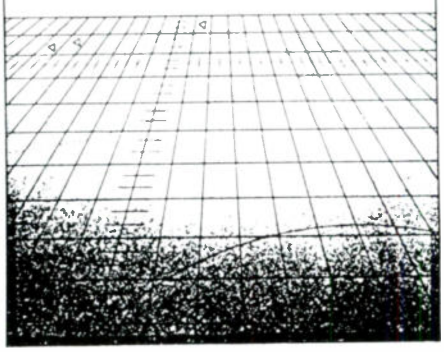

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PEOPLE, PLACES & HAPPENINGS

• **Community Professional Sound Systems** announced the appointment of **Todd Rockwell** as the new engineering and marketing liaison for the company. He serves as a link between the minds and design resources available at the company's Chester, PA operation and contractors and consultants across the country. Rockwell formerly was with **Electro-Voice** and **Mark IV Cinema**, where he held several positions. He has a BS. in Electrical engineering from Michigan Technical University in 1987.

• **Ampex Recording Systems Corporation** of Redwood City, California has formed a new division to be known as **Ampex Digital Media**. The division will concentrate on developing and supplying specialized high-performance magnetic media for use with the company's current digital video and mass data storage recorders. **Dave Davies**, formerly Vice President of Engineering at Ampex Recording Media Corp has been named Vice President of the Digital Media Division. Michael Wilke, formerly Marketing Manager at the Corporation has been appointed General Manager of the division.

• **Rose Mann** is back at the **Record Plant.** A mainstay at the legendary Hollywood Record Plant during the Seventies and Eighties, Rose is now Vice President. Studio Manager, according to the announcement by **Rick Stevens**. Record Plant will also be celebrating its silver anniversary (25 years) in September. The studio has just completed an ambitious $4 million upgrade that more than doubled the client areas of the building, adding two new state-of-the-art Studio Suites, and a new digital editing/MIDI overdub suite to the existing two studios.

• At **University Sound, Ken Koceski**, engineering manager, has been promoted to market development managers of the company's new **USI Audio** branded products; announced **Doug MacCallum**, president of University. Koceski, who has been at the company for five years, will be fully involved with USI Audio products.

• **Audio Pus Video International, Inc.** and **Tahoe Productions** have formed an alliance to provide post-production services for the New York video market. **Scott Irwin**, Director of Post Production for Tahoe has established a base of operation an APVI New York, situated in mid-town Manhattan. APVI specializes in standards conversion, duplication, film-to-tape, audio and international post-production with two other facilities in Northvale, New Jersey.

• **Allan Nichols** has been named director of sales and marketing for the Mark IV Pro Audio Group. The announcement was made by Ivan Schwartz, general manager. The Group, headquartered in Buchanan Michicagan distributes in the U.S. Klark-Teknik, DDA and Midas products, as well as Electro-Voice Concert Sound products.

• As part of its commitment to meeting the audio/video musical-instrument rental needs of facilities around the country, the **Toy Specialists** of New York City has opened a new office specifically to serve Florida and Southeast Markets. The announcement was made by **Bill Tesar**, Vice President of the company. The new office will be headed by **Mark Prater**, and it is located at 1211 N. 56th Street, Tampa, Florida. Their new phone number is **800 445-3330**.

• There will be four sub-committees and twenty-three working groups of the **AES Standard Committee** at the **AES Convention** in New York. The groups will be preparing documents on subjects ranging form computer control sound systems to low-noise grounding and wiring practices. High on the committee's agenda will be the SC-10 Sub-Committee's widely anticipated protocol standards controlling sound systems. Other documents scheduled to be completed at this meeting include listening tests for loudspeakers, methods for archival transfer of audio recordings, interfacing MIDI to sound system, conservation of polarity, and guidelines for authenticating audio recordings.

Details about these projects and the activities of the AESSC, including the schedule of meeting and list of projects, may be found in the **Journal of the Audio Engineering Society**. The meeting will begin on October 4th at the New York Hilton Hotel. This is three days before the opening of the Convention. **Contact the AES at 60 East 42nd Street, New York, NY 10165. 212 661-8528 or fax them at 212 682-0477**.

1 5 0 0
SERIES

WHATEVER YOU DO — DON'T BUY THE WRONG MIXER.

Especially if you're doing multitrack recording — whether digital or analog. Fact is, a mixer that's not specifically configured with the features essential for multitrack recording just isn't a recording mixer. Bottom line is, general purpose mixers make multitrack recording a nightmare.

You see, mixers that aren't designed and engineered for multitrack recording will torture you with the endless hassle of patching and repatching — every time you track, overdub or mixdown. It's frustrating, wastes valuable time and leaves you tangled in cable.

So before you choose a mixer for your studio — be sure it has the features of a dedicated recording mixer.

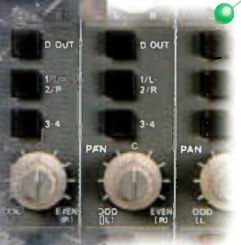
IT'S NOT A RECORDING MIXER IF IT DOESN'T HAVE THESE FEATURES.

SWEEPABLE MIDRANGE EQ

Ask for it. Because when it comes time to tailor your sound, you need the flexibility where the action is — in the midrange. The M1500's sweepable midrange lets you isolate specific mid frequencies allowing you to make the subtle tonal corrections you want.

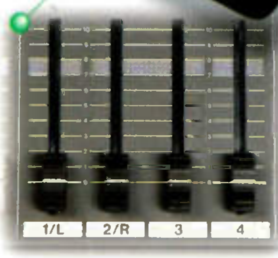
MULTITRACK DECK CONFIGURATION

If you don't have dedicated inputs and outputs for your 8-track deck, where do you plug it in? Without this basic recording configuration you'll be repatching day and night and you won't be able to record on 8 tracks at once. With these inputs, tape monitoring is as simple as pressing a switch. Also, because the TASCAM M1500 is a true 4-buss mixer, you can mix any combination of your input signals to any of the 4 output busses directly to tape.



DIRECT OUT AND GROUP OUT ASSIGNMENT SWITCHES

You gotta have these. Because without them you can't directly send a single input to tape, or record several inputs to one track. But with them, assign your inputs anywhere by pressing a few switches. Best part is, you'll never have to refer to any complex patch diagrams.



ELABORATE MONITORING

In a recording environment you need to hear what's going through your board at all times. With the M1500's comprehensive monitoring matrix you are able to hear any sound source at any time — inputs, tape, AUX sends, anything — it's your choice, just press a switch.

IN-LINE MONITORING

A sure sign of a recording mixer. This lets you monitor your tape tracks at any time without sacrificing an input channel. Just press a switch. With the M1500's dual section not only can you monitor tape tracks, it can be used for additional effects sends, or to double your inputs for virtual tracking at mixdown. And do any of this by flipping a switch.

TRUE TRANSPARENCY AND LOW NOISE

In recording, your signal goes through the mixer several times. And each time it goes through, it is important not to lose or gain anything. Especially an identifiable "mixer sound." Test any mixer for its transparency. Take any signal and bounce it 3 or 4 times on your favorite digital recorder. With the truly transparent M1500, you'd be hard pressed to differentiate between the bounced tracks and the original signal.

At TASCAM, we've been making multitrack recording equipment for more than 20 years. We pack that experience into every mixer we make — and we make more recording mixers than any other company in the world.

For our M1500 Series of recording mixers, the result is an affordable mixing console configured for 8-track recording. A truly transparent mixer that makes tracking, overdubbing, and mixdowns easy. An extraordinarily flexible console loaded with the features and specs you'd expect on consoles costing thousands more.

But the M1500 Series of recording mixers are priced less than many general purpose mixers on the market. They're available in a 16-channel/32-input tabletop version (M1516) and a compact rack mountable 8-channel/16-input version (M1508). So if you're involved in digital or analog 8-track recording, you've just found the best recording console value in the industry.

Get your hands on a true recording mixer today: the TASCAM M1500 Series. There's one waiting for you at your authorized TASCAM dealer. Go ahead — test it and play with it. It's your next recording mixer.

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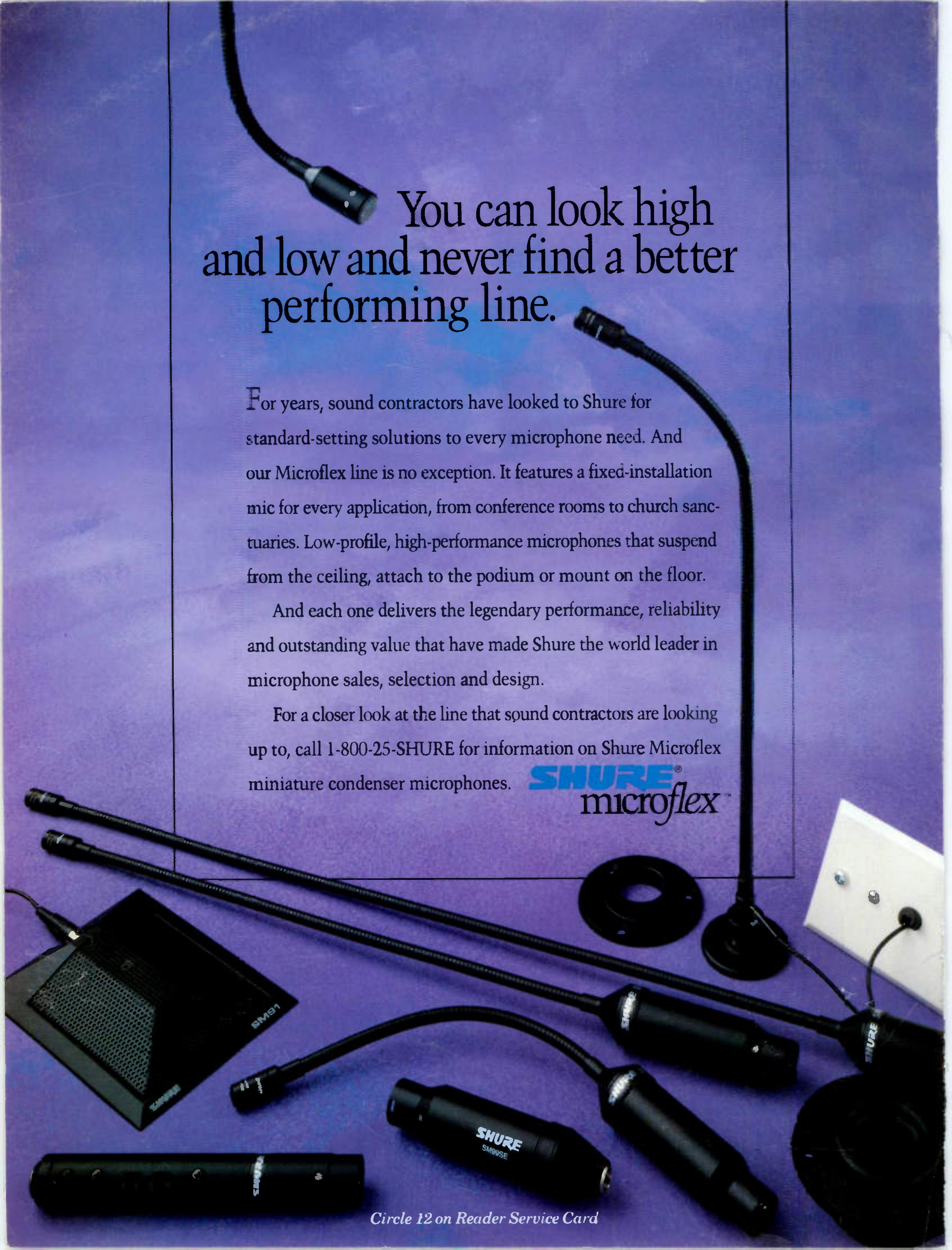
Take advantage of our experience.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