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March/April 1991 Vol. 25, No. 2

db MAGAZINE

serving: recording, broadcast and sound contracting fields

see page 26

see page 30

THE BROADCAST ENGINEER

THE RECORDING ENGINEER

THE SOUND CONTRACTING ENGINEER

THE ELECTRONIC COTTAGE

DEPARTMENTS

ABOUT THE COVER

• The Enterprise was, of course, the mythical space flagship of the Star Trek series. But this very real studio in Los Angeles has not only done post work on the films and TV shows but one of the members of its staff has acted in some of the films. The full story begins on page 22.

CALENDAR

• **The Society of Professional Audio Recording Services** will host a weekend technical conference and interface with manufacturers of digital audio workstations May 18 and 19 in Orlando, FL. Attendees will have opportunities to experience hands-on demonstrations and meet with manufacturers from AMS, DAR, Dyaxis, Lexicon, New England Digital, Otari, Sony Pro Audio, Solid State Logic, Symetrix and Waveframe. For more information, please contact **Shirley Kaye at 800-771-7727.**

• **NHK**, Japan's public broadcasting corporation, will publicly display its major annual technology exhibit outside Japan for the first time at the HDTV World '91 Conference and Exhibition April 15-18 in Las Vegas, NV. The conference is being held in conjunction with the annual **NAB** convention.

The Open House will include prototypes of the latest in consumer and radio television technology, including items never before exhibited in the United States such as fully three-dimensional stereoscopic television, advanced FM multiplex broadcasting and a wall-mounted 33 in. flat screen television.

• **PRODUCTION 91**, a conference designed to bring together an array of sound and image professionals, will be held May 28-30 at Place Bonaventure in Montreal, Quebec, Canada.

The conference will feature over 150 displays and a major symposium on high-definition television. The symposium will focus on the business and creative aspects of HDTV, as well as the technical. Other exhibits will include a forum on performing arts, including special and visual effects, and The First International Competition for New Media Technology which will highlight technological developments in the sound and image industries. The deadline for entries for the competition is March 30.

For more information, please call **Christine Davet at (514) 842-5333 or FAX (514) 842-6717.**

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

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AMY ZIFFER

Synchronization Secrets at VidFilm

VidFilm was established 13 years ago as VCI Duplication, a 10-person video duping facility in Los Angeles. The company relocated to Burbank around 1980, and then again to Glendale in December of 1986. Coincident with the second move came a name change (intended to emphasize their new concentration on film services) and a doubling in size.

IN THE LAST FOUR YEARS, THEY'VE DOUbled size again, to just under 90 employees, added five Rank telecine bays, and, according to Vice President and General Manager Richard Andrews, made significant inroads into film-to-tape mastering for many of the major distributors in Los Angeles.

As the company grew, it started getting requests to do laybacks of foreign language material, which they initially did in their tape room, but as demand grew, the volume couldn't be handled without a specialized environment. In late 1988, it became obvious that VidFilm needed sound editing facilities to handle the burgeoning audio business, and the fall of 1989 saw the opening of their first room.

In May of 1990, John Lawrence, at that time a sales representative for synchronizer manufacturer Adams-Smith, was recruited to head up the nascent sound department. Lawrence has overseen its growth in the past year from that one bare-bones room to two fully-equipped studios in a new building across the street from VidFilm's core operation.

Both rooms are outfitted with Soundcraft 6000 consoles, Westlake or JBL monitors powered by Crown PSA2 amplifiers, Otari and Ampex tape machines with a wide variety of heads, a Magna-Tech mag machine with the requisite complement of head formats to allow them to cope with whatever elements cross their threshold, a selection of specialized outboard equipment, and Adams-Smith AV2600 editors.

In person, Lawrence is an enthusiastic individual, and his manner is permeated with gallows humor—perhaps the result of working all day, every day, to correct other people's problems. He explained that although VidFilm occasionally will do "fixes"—replacing missing sound effects, for instance—the sound department's prime directive is to conform time code. According to Lawrence, the problems VidFilm encounters can be broken down into three groups: mistakes made in the transfer process, NTSC to PAL/SECAM telecine and videotape conversions, and a catch-all "other" category—problems the cause of which can't always be determined.

Figure 1. The VidFilm main room.

MISTRANSFERRED MATERIALS

Since VidFilm relies on the transfer departments of film and television distributors to supply them with elements, they see all the common mistakes. "People think 'time code is time code,'" Lawrence explains. "No, it's not. The speed relationships between film and video are really critical, and there are a number of them. Film is basically 24 frames per second referenced to 60 Hz, but when it runs in telecine, it runs referenced to video at 59.94 Hz, so the actual frame rate is 23.97. But time code isn't always generated referenced to video. There are a lot of variables, and one of the biggest problems we have is people who don't understand the variables and send us wrong combinations of stuff. The most important thing when doing transfers is keeping a chain relationship; any time you break the chain from one element to the next you're up the creek.

"We got this call once from the transfer department at a film studio," Lawrence said. "Their night crew had been working all night

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Amy Ziffer is a well-established freelance writer based in greater Los Angeles.

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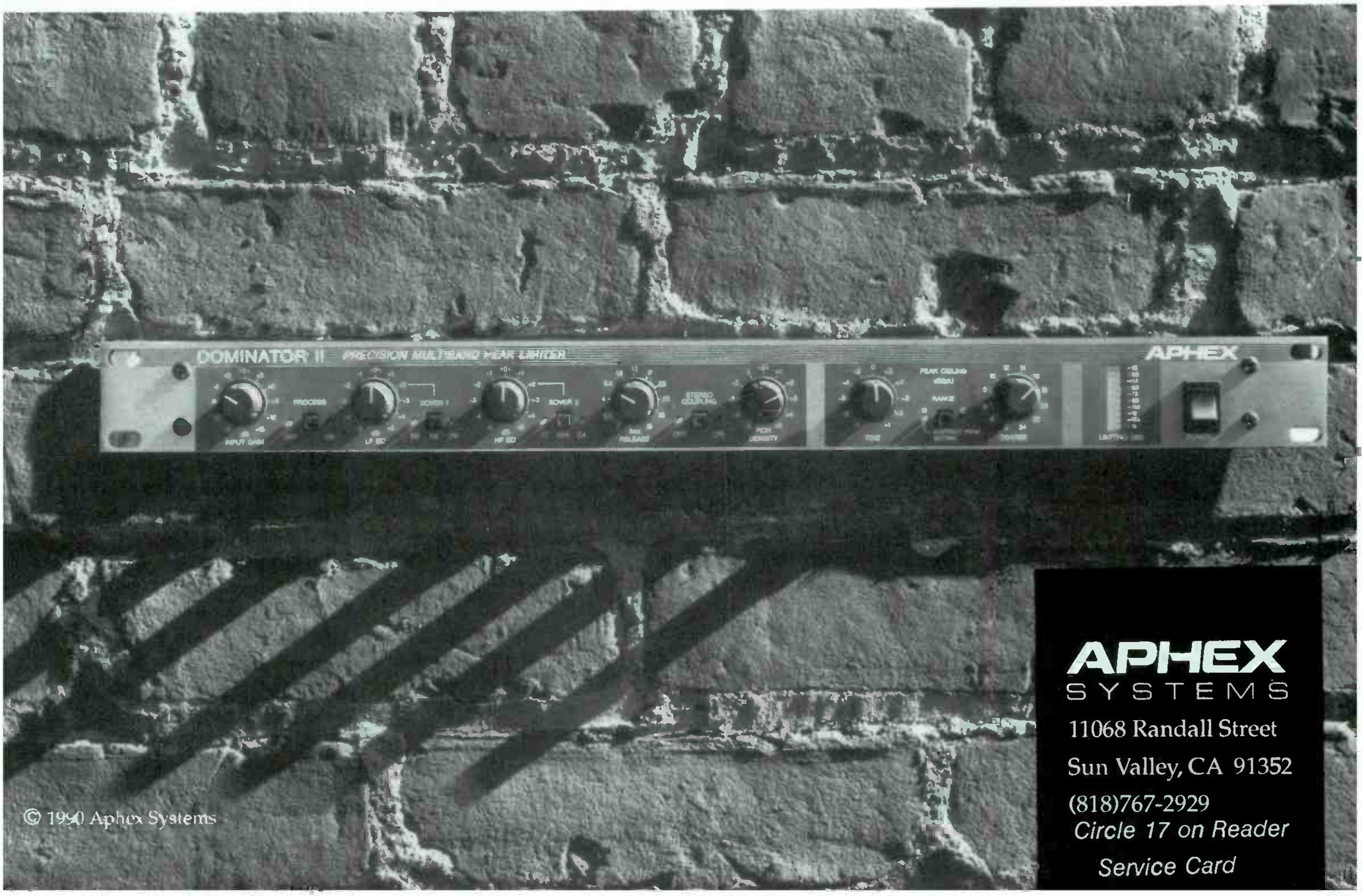

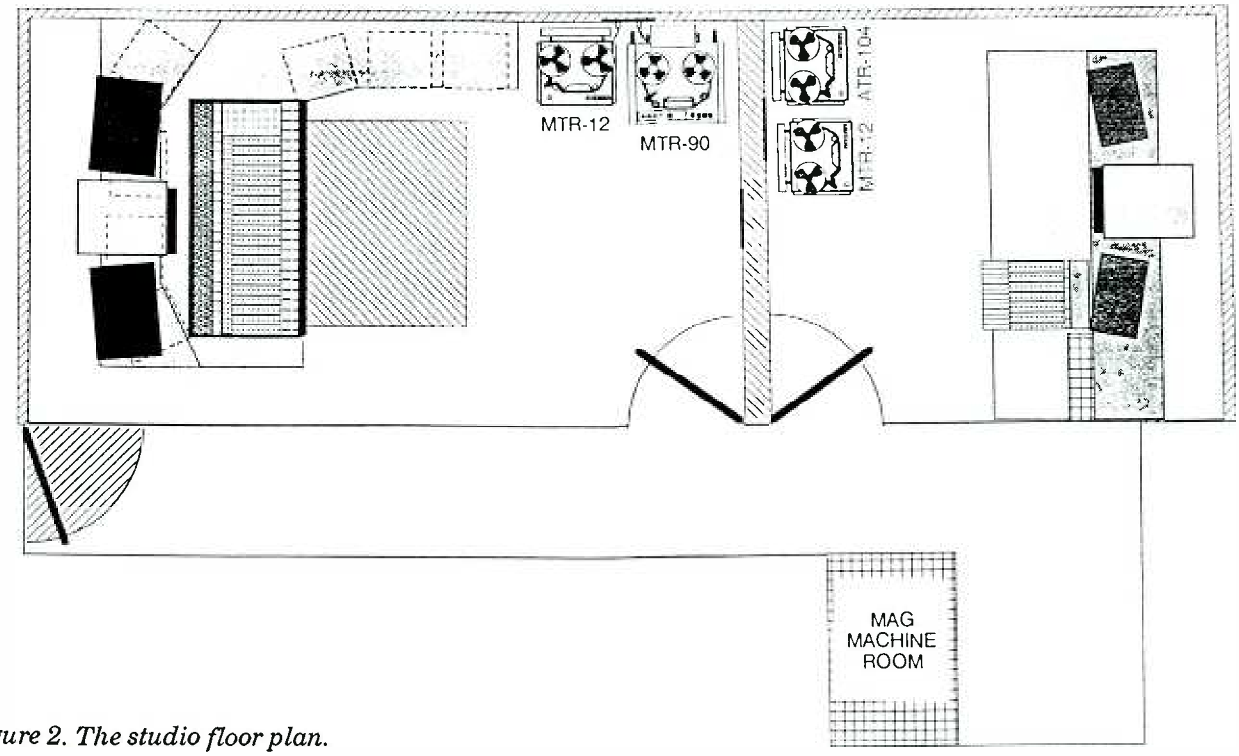

Figure 2. The studio floor plan.

transferring the film at 24 frames per second, referenced to 60 Hz. As they made 4-tracks they laid down audio and striped them with time code, but the time code they laid down was referenced to video. Those are two different speeds. Those things will lock up, but the audio element against picture is going to go like this (he shows a divergence with his hands). You'll lose the difference between 60 Hz, the reference for the film speed, and 59.94 Hz, the reference for the speed of the code. It's like saying on the one hand, 'here's a frame, here's a frame,' and over here [*talking faster*] 'here's a frame, here's a frame, here's a frame.' They will constantly drift out of sync—not out of lock, but out of sync. That particular transfer will never be right for anything," he says.

"However," Lawrence continues, "we have the gear to deal with that: the Adams-Smith 2600. The difference between 60 Hz and 59.94 Hz happens to be exactly .01 percent, so you can enter in plus .01 percent, and it'll run the slave that much faster against the master. What it's doing is building up a big offset table in its memory. It starts with the numbers the same, but a minute later, there's an offset, and a minute later than that the offset is different. It knows in advance what the offsets should be by calculations based on the .01 percent speed difference. So even though the transfer has been made at the wrong speed, we can correct for it. And, the AV can varispeed a machine as fast as the machine can be varispeeded."

NTSC to PAL

NTSC to PAL conversions are the bulk of VidFilm's conforming work; "conversions are what we're all about," asserts Lawrence. One way a show gets to PAL videotape is through creation of a telecine master.

"We get the original NTSC sweetened master," Lawrence explains, "but the audio no longer bears any particular relationship to the show. That master is probably the run length of the original network version. In telecine, when they made the PAL version of that NTSC show, they just ran the film elements at 25 fps—PAL speed. No problem, except that now I get the original NTSC audio element, and how do I make it run 4 percent fast [roughly the difference between 24 fps and 25 fps] and synchronized? I could go to the varispeed and crank it up 4 percent, but that's not going to keep it in sync. How do I *control* it and keep it in sync?

"Here's the way it's always been done," Lawrence explains. "If you have to lay NTSC audio back to a PAL film transfer that's been run fast, you would put up the original NTSC tape, and resolve it so it runs at true speed. As you lay it onto another machine, you stripe that tape with slow PAL code. Regular PAL code is 25 fps referenced to 50 Hz. Regular speed PAL code would lock up with the PAL telecine master that's been created, but unfortunately, it won't run at the *right* speed, because this NTSC tape running at true NTSC speed doesn't match picture any more. Picture is now running 4 percent fast. The actors are actually walking and talking 4 percent faster—you don't notice it, but they are," Lawrence said.

"So if I slow down the code as I lay down the audio—it's actually coming out 4 percent slower as I stripe it—now when I lock it to the PAL telecine master, it's going to cause that code to come up to true PAL speed and the audio will run 4 percent faster. That's how it's always been done, with one of those gearboxes that every transfer department has.

"Here are the problems with that: you lose a generation of audio quality; it takes twice as much time, because every show, before you can lay it back, has to be transferred and it costs more, because I've got to pay somebody to do that. Now we circumvent those problems entirely, because the AV is able to count PAL code against NTSC code and juggle them, and it varispeeds elements to fit—a feature unique to the AV. No transfers.

"When you start with an NTSC videotape, and you want to make a PAL videotape, it's a similar problem. They're two different speeds; one is 30 frames, the other is 25. How do you get this 30 fps show to run at 25 fps? You could just run it slow, I guess, but they don't. They run it through a converter, and everything runs at the same speed, but it overlaps some of the 30 frames to fit into 25. If you scroll it through a cut, you'll see frames superimposed; that's how you can tell it's a conversion. Tape-based shows are all converted that way, because they're not made from film, and there is no PAL telecine master.

"A typical example of this is a sitcom like *Who's The Boss?* We might get an order for 120 episodes going overseas. This is my problem: the original element is NTSC, and it's been converted to PAL. A PAL conversion runs at the exact same speed as its NTSC counterpart. If it's 23 minutes in NTSC, it's 23 minutes in PAL—it's just counted differently. The time code is PAL code, 25 fps instead of 30 fps NTSC. Just for a simple layback like this, you have to take the 24-track machine, put the original audio element on it, resolve it to NTSC to play at true speed, submix the stripes to 4-track, and as you lay it down, stripe it with regular PAL code. The elements run at the same speed, it's just a different count on the PAL machine.

"When I got here," Lawrence said, "we were doing lots of that. Every show that was going to PAL had to be transferred. We were tying up $50,000 worth of machinery, an operator, and our clients' lead times were doubled. A lot of our clients expect their material in 24-48 hours. They've cut a deal to distribute this stuff in Peru, they have air dates, they're ready to go. My big goal when I came here was to eliminate the laydowns and transfers made before laying back. It increases our throughput, cuts lead times, and saves our clients money."

THE DREADFUL OTHER

The situations above cover different speed variances, but they are sort of "knowns," problems that any house with a gearbox, if they know the condition of the masters, can conform. But there are a couple of areas in which that won't help. One is shows that have been varispeeded in telecine.

"We had one show where the varispeed was all over the map," remembers Lawrence. "It was a show known for its cinematic quality. Probably what happened is that the film editors cut it to look the way they wanted. But filmmakers aren't used to cutting to the second, so it went to the network, and the guys from the ad department said 'Uh-uh, it's too long. You can either cut off the end of the scene or you can make it fit.'

"So we get stuff that's been varispeeded all the time, and with ordinary means there'd be little I could do, because the amount of documentation I get at my end is nil. With the 2600, however, I can fix it. This is an example of the type of work we've been able to do here that other places have had to forego.

"To use the 2600, we put the playback machine in varispeed. At the beginning of the picture there's this car honk as part of the M&E track. I mark it in. Now I take it down three minutes. On the audio source maybe it's three minutes, but on the video it's two minutes and 58 seconds. Here's a door slam, and we're out of sync. I mark that point, too. What the system will do for me is interpolate between those two points and calculate whatever it takes to bring them into sync."

Although it sounds as if the 2600 is doing everything, there's room for considerable artistry when time anomalies aren't constant. Lawrence cites an example: "I was working on these Dynastys in French. The show is constructed in a formulaic kind of way: there's a scene, music bumper, scene, music bumper, etc. They didn't seem to care for the music bumpers. They wanted to get to the next scene, so all the music bumpers ran about 8 percent fast, and you'd get back into dialog and it would be 1 or 2 percent fast. It would be a real challenge because the music would always slide under the dialog, so you'd always have to find a place to start a new varispeed amount, to keep the music in sync through the break, and yet have the dialog at the beginning of the next scene in sync. If we had separate elements, that would be no problem, but it's rare to get anything that's not married. That was a difficult project.

"It required not only varispeeding the material, and some extensive cutting around dialog, but we had to use a Lexicon 2400 to keep the music in key at each point where we changed varispeeds," Lawrence said.

WRAP

With all the transfers and conversions of picture done independent of audio, and the lack of understanding of how the two interact, the number of problems that can and do arise are numerous. After almost a year at VidFilm, the surprises are getting "fewer and fewer," according to Lawrence, but they're still coming. Most of the time he and the other engineers don't even bother to try to figure out what happened; they just find a way—any way—to get audio and picture back in sync.

At this point in time, the audio services VidFilm offers are mostly limited to conforming existing audio tracks to picture. However, less than 25 percent of the material they work with will transfer straight to picture. A common situation for VidFilm is to get elements that just don't work with the version of picture they have. There is varispeeded material or different edit versions. Often, though, the elements they've been given are the only elements in existence. "Some of the repairs and editing we perform on shows to make them conform really blur the line between simple 'layback' and actually building a working track," Lawrence says. "We are constantly cutting in and around dialog and music, and fitting effects, to create a finished track that sounds smooth and is in sync.

"At the same time, we are aware that some mixer somewhere mixed that show, that's how they wanted it to sound, and we don't mess with it. But some of the tracks we get are in really bad shape, and we will reprocess them. Our people are qualified enough to know when something doesn't sound right, and we fix it. Often there'll be a 60 cycle hum, or excessive noise. We try to make sure it leaves here sounding as good as it can, but we don't do a lot of EQ shaping or remixing of element levels. Eventually we will offer a full range of audio post services, but right now we want to get as good as we can be at what we're doing. Then we'll move on."

Summing up the sound department's modus operandi, Lawrence laughingly quips, "There's the saying 'You can have it good, fast, or cheap,' but the way we're doing it now with this gear, it's good, fast and cheap"—everything a client could ask for.

That combination of quality, speed and price was a good formula for growth over the past four years, and will undoubtedly continue to be as VidFilm moves through the '90s. db

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Live Sound Reinforcement: Asia-Pacific 1990, Part 2

ED LEARNED

The most challenging aspect of my recent Asia-Pacific trip had to be the difference between the two groups I toured with. The obvious differences in musical style necessitated completely different production requirements and techniques.

During our three month tour of the region, I would be required to reproduce a classic jazz trio's delicate sound with taste, then handle a hard-charging Zydecajun band that loved to rock out! The opportunity to deal with opposite ends of the volume spectrum on one long tour was something I relished. I mentally prepared myself for each band's musical style by listening to as much of their music as I could find. I tried to anticipate the changes I'd need to make in my own working style; each group had radically different ideas concerning the importance of sound reinforcement, staging, visual presentation and work day scheduling.

Jazz history is rich with great trios, most fronted by pianists. Little wonder that Charlie Byrd cites the great piano trios of Erroll Garner and Ahmad Jamal as influences on his trio music. While there were trios that included guitar, the piano was omnipresent. Charlie's seminal trio of 1955 was one of the first *without* a piano, opting for instrumentation of guitar, bass and drums; his 1990 trio used this same setup. This opened up a lot of room for the guitarist, with unparalleled freedom in both the melodic and harmonic domains. From a sonic standpoint, it required the guitar to be perfectly audible at all times, without covering up the subtle dynamics of trio interplay. I had to be sure the guitar could "carry" any room we played, without using overt amplification. I knew the group used no monitor system, which spoke volumes about the type of level they preferred on stage; I would have to apply these same standards of level and balance to the audience area.

Charlie's first performance of the tour took place in Manila, at the Thomas Jefferson Cultural Center. This facility, dedicated to cultural exchanges between the Philippines and the United States, contained a library, several offices, a small art gallery and a conference room/theater. Our concert was held in the theater, a rectangular-shaped room with a low ceiling; its seating capacity was about 175. There was a small speaker system, flush-mounted in the wall that also contained the movie screen, primarily designed to handle audio for movies or video presentations. A small temporary stage was set up in front of the screen for us, so the PA would be behind the trio, and there was nothing I could do about it. In deference to room size and PA placement, I elected to use a single vocal mic, for Charlie's announcements between songs. There would be no music reinforcement save for stage amplification. I didn't know it, but this decision proved crucial in gaining the group's trust.

I was a bit concerned about how the band would react to their soundman nixing a sound system entirely at their first show; it happened to be the smartest move I could have made. Acoustic presentation at this particular venue was common sense; all instruments could be clearly heard anywhere in the room. During our sound check, my job was to confirm that the trio's stage sound created a pleasing blend in the room. The guys were very happy with the results and my approach to the job. They were afraid that I might be another ham-handed engineer whose only technique was to mic everything and turn it up. I gleaned several important tips from our ensuing conversation about sound and soundpeople that I used as guidelines for mixing this trio's music.

NO MONITORS

The group played with no monitors to maintain control of their own stage sound. If Charlie wanted to change his levels during performance, he had sole command of the volume knob; he didn't have to depend on someone else who either might miss the request or overcompensate by making things too loud or too soft. At TJCC concert, where there was no music amplification, the trio also controlled house sound. When Charlie played softly, he made it sound soft. Many soundpeople mix from a basic assumption: if something can't be heard clearly, turn it up until it is clear. How does one reconcile one's desire to make everything perfectly audible all the time versus the band's desire for artistic changes in dynamics? It annoyed Charlie when he tried to play softly, while hearing the soundperson turning his guitar up in the house because level had dropped.

I had to develop a technique for house amplification that would result in audibility for the audience without losing the context of group dynamics. My idea was to let music dictate level: when it was soft, I let it be soft; when it was loud, I let it be loud. I felt Charlie's standard for house spl could be met through creative use of stage sound. When house amplification of instruments was necessary, I wanted the sound coming off stage to comprise 50 percent of what the audience heard. I tried to blend in just enough of the house PA equipment to spread the stage sound throughout the audience area. My goal was to avoid point-source realization if I could: while standing in the center of the audience area while the band played, I wanted the sound to appear as though it was coming from the stage. If I could clearly perceive the sound I heard coming from the PA, then the system was too loud. The group loved it; they had just the right amount of sound coming back at them from the room to "feel" their original dynamics without being drowned out, their first hint of overly loud house sound. Of course, this didn't always work in the largest halls, which required more PA sound for even coverage, but I tried to keep the stage sound/PA sound balance as even as I could.

I believe that personal interaction with the musicians I work with is an essential part of mixing sound for them. After all, the musician makes the music; understanding them lets you understand their musical tastes and sounds. I spent many enjoyable hours getting to know Charlie, Joe Byrd and Chuck Redd. With Wayne Toups & Zydecajun, I had an added advantage: I'd already worked with the group, both in the United States and abroad, and knew most of the guys, although there were two new members. Toups' approach to sound was no mystery: it's appropriate that the group named their first Polygram album "Blast From The Bayou"! Rock-type levels were required; the band desired loud, penetrating monitors onstage, with high-impact sound in the house. Some of the countries we visited were expecting a traditional Cajun group with some amplification; what they got was a rock band with Cajuns in it!

When planning my mix for these guys, I knew a big rhythm section sound was important, but I used my "rule of the bandleader" as a starting point: the name is Wayne Toups & Zydecajun, so thou shalt always hear Wayne Toups! Since Toups was the band's lead vocalist, keeping his vocal clearly audible was an obvious priority. The accordion was another: Toups' prowess on the Cajun diatonic accordion is well-known in Louisiana; now the rest of the United States, and the world, are discovering it, too.

The instrument demanded attention, but Toups often experienced sound problems because many engi-

Figure 1.House tech Sonny Perocho (right) in the CCP Little Theater control room.

neers just don't have a lot of experience working with accordions. It's a new sound to them, so they lock in on it. Level decisions for accordion are assessed by an engineer fixated on this new sound. Invariably, the actual level attained is too low; the accordion gets lost in the mix. The other extreme is even worse: the accordion is raised to screaming level, washing out other instruments and hurting people's ears. Prior experience with Toups gave me a good idea of where to "place" the accordion in the mix for proper audibility and balance. His accordion also had important ensemble value: the band's arrangements were sophisticated, with many unison lines between accordion, guitar and keyboards that required proper voicing.

A weighty drum sound was my next major mix priority. Heavy kick and snare were required in most monitor mixes; the group expected the same in the house. When Toups came out front, he wanted to clearly hear the kick at worst, feel it at best! Reverb was desired for the snare, and occasionally for the toms, to add depth to the kit. I worked closely with drummer Mike Burch and bassist Mark Miller (who also engineered the group's recordings) on different reverbs for the drums, along with some special effects we planned to use on the drums during certain songs. Burch's habit of tuning his kit every day was key to our success; he realized the secret of a good drum sound is good sounding drums!

While my effects rack was rarely used for Charlie's trio, it got a full workout with Toups. One reverb was dedicated to the drums, another to the vocals. Occasionally I would dump instrumental solos into the vocal reverb for effect, or if our venue was especially dead acoustically. I frequently used my DDL on the voices, varying programs often between songs. These ranged from a simple doubling on fast songs to very long echoes, mixed subtly into the background, for ballads. We occasionally used slapback echo, and I created one special "discrete repeat" effect that was used on *Two Step Mamou*; I echoed one word in the verse, at particular times, in tempo with the song's rhythm. That required some fast fader moves; I rehearsed these every day by including the song in our sound check repertoire.

HOUSE SPL

Reaction to house spl was another obvious difference between groups. I had more complaints about volume at our first show with Toups than I'd had during the entire Charlie Byrd tour! The rock nature of Toups' band hadn't been apparent to many local promoters from the advance information they'd received. The inevitable diplomacy required between the band's desires and audience preferences fell on my shoulders. Predictably, the line between too loud and OK was drawn between age groups and their different concert experiences.

My mix point was required to be not more than 75 feet from the stage; typically, c weighted peaks at the mix point never exceeded 108 dB-spl, which is not horribly loud by today's rock standards. The young people present had no problem with it, but typically, these shows draw a large cross-section of the community. A quick explanation of spl readings were enough to cool out most promoters. I usually suggested moving, away from the stage towards the rear of the venue, to irate audience members who complained to me during a show. To my surprise, many

Figure 2. The Charlie Byrd setup at Miriam College.

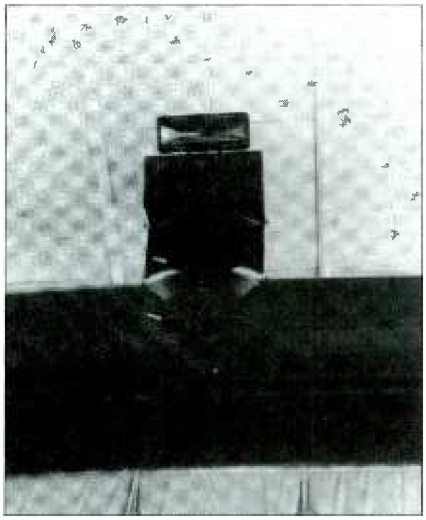


Figure 3. The center cluster at Miriam College.

of these same people came up afterwards, telling me how much they'd enjoyed themselves after taking my advice! Moving back—what a concept.

As I'd anticipated, there were occasions on the Toups tour where the PA equipment provided did not come even close to matching spec. When the sound system could barely handle vocal reinforcement, let alone an amplified band, we used an approach surprisingly similar to what I'd done with Charlie. Instrumental sound for the audience area was generated by the group's stage sound: I would stand in the house while the band played, instructing each musician to turn up or down until we'd achieved the correct balance. In most of these cases, the drums provided our "volume" starting point. We would then add voices, accordion and effects through the PA system, making further adjustments in individual instruments to bring the stage sound into proper balance with the PA sound.

I only amplified drums, guitar, bass and keyboards for solos. The resulting balance sounded wimpy, compared with the fully mic'd and amplified mix we preferred, but it allowed us to deliver an acceptable facsimile of what the band was really about. These circumstances called for more than cooperation from the band; they called for sacrifice.

The levels I required were low enough to cause problems onstage: the guys had problems hearing themselves and each other. It's hard to play when you can't hear what you're doing, yet the group accepted the necessity of these changes with good humor. That kind of professionalism helped us through many tight spots.

PHILIPPINES—MANILA

Aquino International Airport baggage claim is large and spacious, but porters are few inside the customs area. If you've got a lot of gear, be ready to work. The political situation in Manila demands that airport security be tight; prepare yourself for detailed inspections and long lines. Musical equipment gets special attention; a manifest of equipment is attached to your passport, and it will be checked on the way out.

Charlie played his first concert of the tour in Manila, at the TJCC (see above). There were plenty of U.S.-type electrical outlets for power, purportedly providing 110 volts. They actually carried about 105 volts; the receptacles on the side walls were wired out-of-phase from standard U.S. code, while the ones behind the stage were wired correctly. I never did find out what type of system was mounted in the rear wall, but I could tell it was very bassy. With the stage in front of the PA, I found this system very prone to feedback in the 200-250 Hz area. I was very happy to have a tight-pattern mic like the Electro-Voice ND-457 for Charlie's vocal; I would have had a terrible time getting sufficient level with anything else.

Charlie's second Manila concert took place at the Cultural Center of the Philippines, a huge complex of theaters, rehearsal spaces, galleries and work areas dedicated to the arts. We performed in the Little Theater, a small hall seating around 500 people. The acoustics here were excellent: reverb time was no more than 1.3 seconds, and seemed very even. The audience area was steeply raked, so sight lines were superb. The house PA system turned out to be a small E-V bi-amped system mounted in the side walls slightly forward of the stage. House tech Sonny Perocho took a feed from my mix point (at the back of the hall) up to the sound booth through an open control room window; this signal was run at mic level into a Ramsa 18 channel console, which controlled the CCP PA (see *Figure 1*).

There were U.S.-type AC receptacles around that provided 220 volts, wired as a 2 hot, 1 ground system; the two hot lines were supposed to be 110 volts each, but were really 90 and 130. The large difference between hots didn't inspire my confidence in these outlets, so I elected to pull my AC power directly from a drop off-stage left; the ground bar at this box had a good 30 volts on it, so I had to run a ground wire about 125 feet to a backstage bathroom's shower pipe. The voltage was very stable, thanks to CCP's transformer isolated service. The place was packed for the evening show, and the appreciative audience enjoyed a great performance, including some impromptu vocalizing by Charlie.

Both groups performed at Miriam College, located about 90 minutes (average drive time) outside of Manila in Quezon City. The auditorium at this all girls college seated around 800; its tile floor, plaster walls and high ceiling contributed to a very lively reverb time of 2.5 seconds (see *Figure 2*). The venue had no air conditioning, which made for uncomfortably hot and humid conditions inside the poorly ventilated space. AC power came from a drop off-stage right: it furnished 220 volts, with an equipment ground provided by a bar bonded to the metal box. I measured voltage swings of five volts during the day, but things seemed to stabilize after dark.

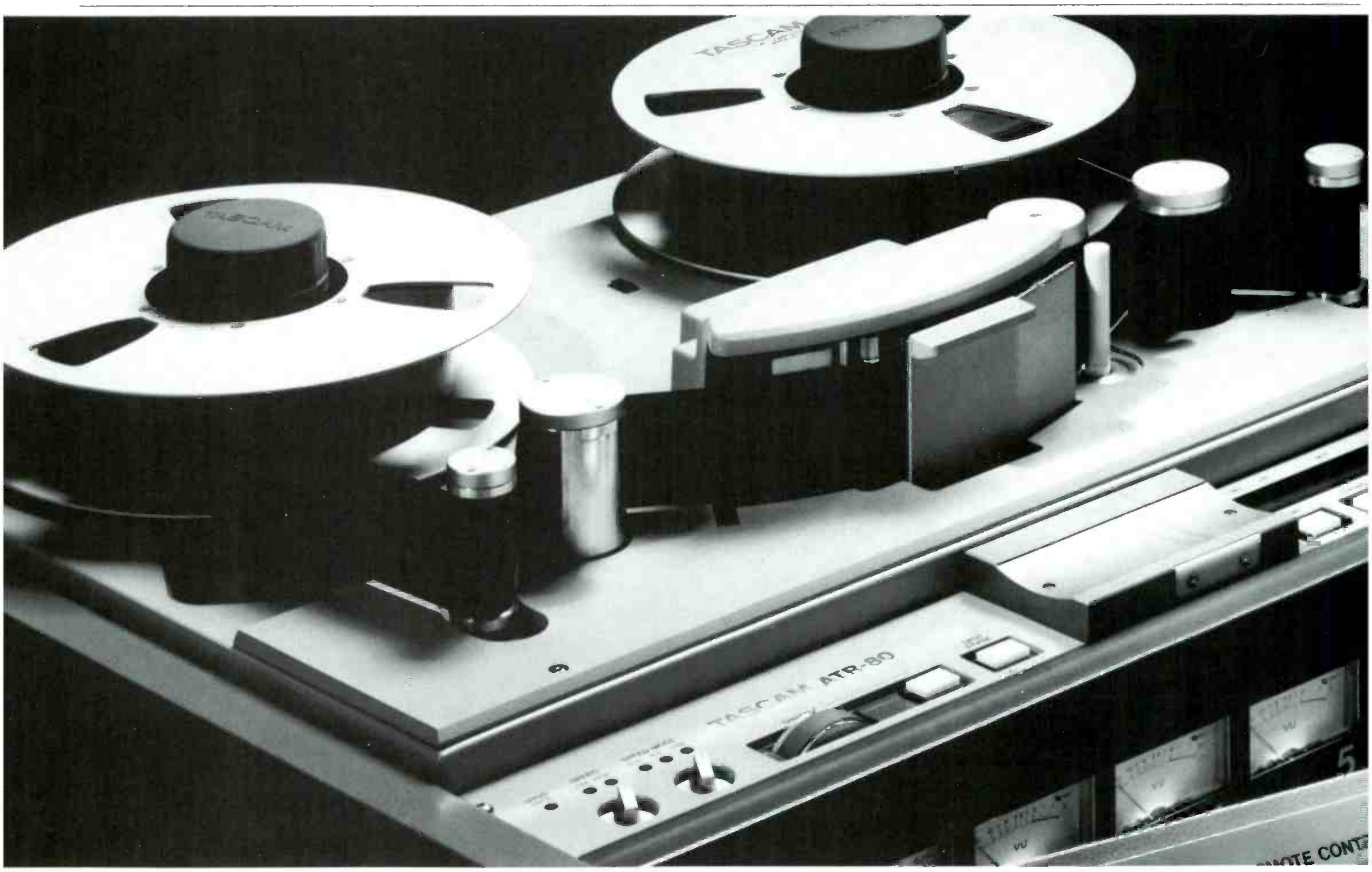
For Charlie's trio, we planned on using the house PA system; I'd been assured this system would meet our spec. I discovered it was a collection of several different E-V horns and a custom bass cabinet, configured as a center cluster flown from the proscenium (see *Figure 3*). This speech-oriented system was in no way appropriate for music. There was no bass response to speak of; some radical EQ curves on my system graphic improved system response from mediocre to fair. I found out rather quickly during sound check that any real bass signal made the PA distort. I tried to balance the band to Joe's bass, hoping his stage sound would be enough to "carry" the room. The live hall actually helped under these circumstances; this PA system wouldn't deliver a lot of spl, so I had to depend on room acoustics to enhance the sound.

The near-capacity crowd was comprised primarily of students; many found Charlie's low-key music interesting, but others decided it was a good time for socializing. I have to go

Figure 4. A Yamaha PM-3000 40-channel console, along with Rane crossovers and graphic equalizers, equipped the house mix point at the Show TEKNIX mix point.

back to a 1987 South Pacific tour to recall an audience that talked more during the performance than this one; my biggest problem all night was getting the band audible over the background chatter! I had one embassy staffer complain about the sound, telling me it was too soft; unfortunately, the PA would only get so loud before distortion occurred. It was a difficult evening.

I returned to Miriam College one and a half months later with Toups. I'd made it *very* clear that under no circumstances would we use their house system again; I insisted that my spec for Toups' PA system be met to the letter. CCP was our co-sponsor for Toups' Manila concerts; Project Coordinators Ariel Santos and Sunshine Domine contracted Show Teknix, a local sound company, to handle our area concerts. To say they did a good job was an understatement; it was the most extensive PA system I'd encountered on any of my overseas trips! A Yamaha PM-3000 40-channel console, along with Rane crossovers and graphic equalizers, equipped the house mix point (see *Figure 4*). System engineer Ed Garcia used a special step-down transformer to power the console, which ran on 120 volts; I powered my effects rack off this same stabilized source. There were also five dbx 160X limiters available for channel patching. Show Teknix used a JBL Concert Series PA system, providing four high packs and four low packs



per side; four Amcron Macrotech 1200 amplifiers powered each side. There were eight custom floor monitors, each with a 15 in. woofer and horn/compression driver, powered by Amcron.

These were controlled via a 32 x 8 modified Soundcraft console. The sound company even provided us a regulated 110-volt grounded AC outlet for our stage amplifiers. That's what I call full service! In deference to their monitor engineer, I elected to use Shure SM-58 mics on all vocals, using my E-V ND-457s on the drums.

The near-capacity crowd enjoyed every minute of an action-packed show; the girls were screaming for Toups and Burch at every opportunity. That same embassy staffer who chastised me with Charlie came up and told me he loved Zydecajun; the pop-oriented production style was more to his liking. Of course, several others complained that the band was too loud, so I guess you could say I'd come full circle at this venue! The same live acoustics that had helped Charlie were a detriment to Toups, although the capacity crowd helped matters by damping the room reverb a bit.

I could fully mix the band with the vocals; with a PA system this nice, the result was the best sound of the tour.

Toups/Zydecajun also played a matinee concert at Philippine Normal College in the school gym (see *Figure 5*). This facility had the

Figure 5. A Wayne Toups sound check at Philippine Normal College Gym.

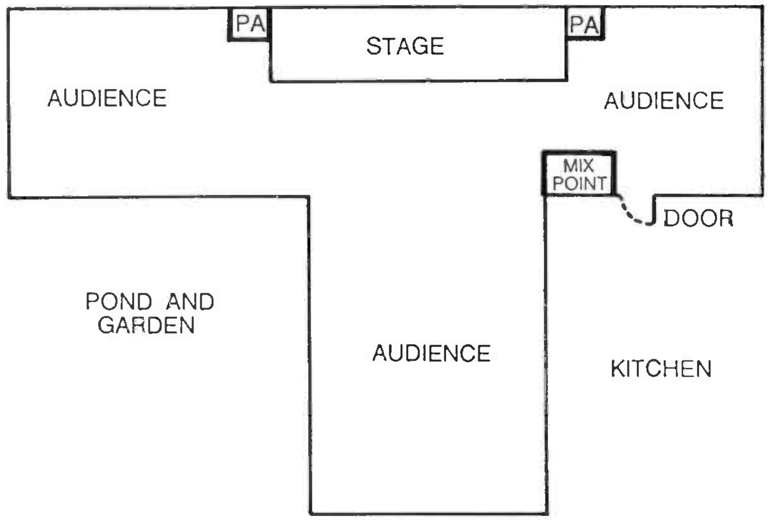


Figure 6.Davao Insular Hotel's "T" shaped show room.

acoustics of, well, a gym. Despite a cover laid over the basketball floor, reverb time was over three seconds, horrible conditions for an amplified group like Toups/Zydecajun. Show Teknix again provided sound reinforcement; with a show together already under our belts, the setup was fast and easy. In deference to this reverberant hell, I laid back on instrument and drum levels in my mix, concentrating on vocal intelligibility.

I modified our effects, shortening reverb times and mixing in less delay; I equalized the effects returns to avoid exciting the room's worst reverb nodes. A quick glance at my faders revealed the mix was considerably softer than at Miriam College the night before, yet it sounded twice as loud because of the room. It was a constant struggle to get the vocals understood. The audience didn't seem to mind; they got on their feet and stayed there! Screaming girls actually fought over the shirt Burch tossed into the audience; dancing in the aisles was the order of the day.

CEBU

Located in the Visayas, the central section of this island nation, Cebu is about a 45 minute flight from Manila. Philippine Airlines operates regular flights to Cebu from both Manila and Davao that can handle a large amount of excess baggage. Automotive access to the terminal at Mactan airport is limited; so is parking. It's important to allow extra time for the handling of equipment and baggage. There are usually porters or other individuals hanging out around the terminal who will assist with cartage for a small fee.

The Charlie Byrd Trio played at the Colegio de la Inmaculada Concepcion. The auditorium's stage had large wings that extended out beyond the proscenium opening; seating capacity was around 400. Abundant hard surfaces contributed to a reverb time of just over two seconds. I tied my transformer into a drop off-stage right; this provided 220 volts, but had no ground source. I ran my ground wire to a bathroom off-stage right, only a few feet from the AC drop.

The PA was provided by a local independent engineer; he used a single Peavey 3020 HT cabinet on each side. Each cabinet contained two 15 in. woofers, two 10 in. woofers, a horn/driver and a tweeter; they could be run tri-amplified or full-range passive. Power amps were a mixture of Peavey and custom, with a Biamp crossover.

I've acquired the habit of always checking my AC power before I plug in and turn on anything, something I did this time. What I didn't expect was an expensive lesson on also checking the other guy's power. When we connected my house feed to the PA crossover and turned it on, smoke and ozone were the result. The crossover was dead; so was my house graphic channel and five snake lines plugged into either the EQ or the console. A quick look at one of these snake lines revealed all conductors were dead shorted; voltage had apparently flowed through it.

A check of the local PA equipment revealed that the chassis of all amps and electronics carried 180 volts. I surmised that this hot-on-ground problem sent voltage down my snake towards the only grounded source—my mix point. Fortunately, the console was OK, and I still had another operational EQ channel. I used my Carver amplifiers to power the PA cabinets full-range, and pulled off the show.

Quite a debate began about our problem: the local engineer blamed my power source, while I suspected his equipment was either faulty or improperly powered (the PA equipment got AC from a different source). Our audience was none the wiser; the concert was co-sponsored by the Guitar Foundation of the Philippines, so the crowd was studded with guitar players and enthusiasts. They understood and appreciated Charlie's artistry on their favorite instrument.

Wayne Toups & Zydecajun performed at the Base Line Restaurant. Their ballroom, located on the second floor above the restaurant, was a popular local club that featured American-influenced pop music. The stage was positioned to "play" the long dimension of this rectangular room. Capacity was around 175, with the tile floor and hard walls creating a reverb time of almost two seconds, quite long for a room this size. AC power came from a box at the end of the room opposite the stage; this provided 240 volts and a good ground. The neutral, however, carried almost 50 volts; I tied my neutral to ground, avoiding this problem. I placed my transformer at the mix point, in the house right rear corner of the room, to keep my AC input short.

Our PA system was the same one I'd used with Charlie; you can bet I didn't plug in anything this time without a thorough investigation! I again measured massive voltage on every amp and crossover chassis. To eliminate the AC power variable, I made the engineer power his gear from the same drop I'd used. Unfortunately, while the voltage dropped somewhat, it didn't go away, remaining dangerously high.

We'd now confirmed there was definitely a dangerous problem with the local equipment; I was again forced to use my own amp to power local speaker cabinets. Unfortunately, every amp channel I had was dedicated to our four monitor mixes, so we had a decision to make.

The band agreed to deal with two mixes, freeing one amplifier for PA power. We combined Toups' mix with the drum/bass mix; the guitar and keyboard mixes were also consolidated.

The small room size actually helped: the band already planned on backing their volume down to prevent overpowering the room, so they didn't need the monitor diversity or power that they normally did. The house mix was almost entirely vocals, accordions and effects.

DAVAO

Located on the large southern island of Mindanao, Davao is about a 90 minute flight from Manila. The airport's arrival and departure areas are spacious and well-organized; not so baggage claim, which is downstairs from the arrival area. There is only one baggage line in an area barely able to handle a small plane, let alone the wide-bodies that Philippine Airlines operates from Manila. Suitcases were enough of a problem, not to mention large instrument cases. After our experience with the Byrd trio, we elected to send all Toups' equipment and luggage from Manila as cargo.

The cargo terminal is immediately adjacent to the passenger terminal. It has room for large cases, no small aisles or stairs to deal with, and a loading area that one can back a truck up to. One other note: I left Davao the last time at 7:30 a.m. on Oct. 4, 1990. The airport was forced to close shortly after we left because of a putsch, led by renegade elements of the Philippine army. That was a close one; talk about getting on the last flight out!

Charlie's concert was at the Davao Insular Hotel. The hotel's compartmented banquet area was converted into one large room which created a T-shaped space. The three defined audience areas, located at the three ends of the T, seated 250 (*see Figure 6*). It was impossible to place speakers on the front corner of the stage without blocking sight lines, so I had to place the PA against the wall at each upstage corner. The hotel's PA system was two-way, comprised of a folded-horn subwoofer and another cabinet with two 10 in. speakers and tweeters, configured in a box that angled each 10 in. away from the other.

There was a custom amp and crossover system housed in a huge rolling DJ rack to power this system. I elected to use only the high-frequency box, since the angle on the two 10 in. speakers gave me the 180 degree coverage I needed to cover the different seating areas—these were powered full-range by one of my Carver amps. The subwoofers were completely by-passed; I didn't need pumping bass, and with the PA against the back wall, these woofers would be behind the band, inviting low-frequency feedback problems. I did, however, use them as speaker stands to elevate the small high-end boxes above head level.

AC power came from a drop located in the kitchen office, 30 feet away through a door adjacent to my mix point. Voltage here was 110 volts, with both a clean neutral and good ground. Using the 120 volt tap on my Variac, I bumped our stage voltage up to the 120 volt level preferred by the Carvers. Balancing the group here was a challenge; there were really three separate acoustic environments, each with a different stage balance due to proximity. Since the PA wouldn't "hold" any real bass, I balanced the band to Joe's bass, counting on his stage volume to "carry" each space.

The rear of each audience area was no more than 75 feet from the stage, so this approach worked well for us. I spent a couple of minutes listening to the group in each "room," striving for an average balance in each. My efforts weren't wasted: every seat was filled, the band played beautifully, and even the trio's most delicate work could be heard everywhere. The hotel's manager dubbed it "the greatest concert ever held here."

The Toups/Zydecajun concert was held at the Central Bank Convention Hall, a huge convention/exhibition space that could accommodate an audience of 3,000 via rows of folding chairs. The sheer size of the room, coupled with a hard floor and metal ceiling, created the acoustics of a cave. The stage was placed against the rear building wall, centered to "play" into the long dimension of the room; sound from the stage took well over three seconds to decay.

I expected a sonic hell, but it was the electricity that proved to be the REAL problem. There were US-type ungrounded outlets on adjacent walls; these supplied 240 volts as two hots, no ground, no neutral. I located a three-phase drop backstage that also had no ground. We ran a ground wire to a bathroom water pipe in the dressing room, backstage about 40 feet from the performing area. Using this ground as a reference, we measured differing voltages on each leg, between 120 and 155 volts; even the "neutral" carried excess voltage. I was fortunate to have two Filipino electrical engineers present, courtesy of our local sponsors, the Davao Variety Club. No one could explain the dirty neutral—our electricians indicated that something was seriously wrong with the building's power system. They agreed we could by-pass the dirty neutral, creating our own from the ground source.

The plan was to use a single hot leg and the 110-volt tap on my Variac to give us the 120 we needed. Unfortunately, our power disappeared when connected: we measured voltage on our hot wire, but when we attached it to the transformer, voltage dropped to zero. My transformer was blamed for this; we waited two hours while another was procured. Unfortunately, the new transformer made no difference and grounding the neutral at the box didn't help either. Our Filipino friends called the Davao electric company; they confirmed an electrical problem, and began to trace the fault with no guarantee it could be rectified by showtime. We paused to consider our options; without power, all our gear was dead in the water.

The PA system here was a component system of four Perkins boxes, each with a single 15 in. woofer, and two radial horn/drivers per side. The power amps were located at the mix point, with miles of speaker cable run back to the stacks. There was a mixer with fixed EQ and two aux sends. There were also two small monitor speakers on stage; I'd planned on using our own monitor system, but the power problem dictated otherwise. In the event that power could not be supplied to us by showtime, we decided to play a tape of the band through the system while the group "lip-synced" the vocals and pantomimed playing their instruments.

Not only did he change all the pre-set house levels, but, in an attempt to get a tape feed, he unplugged half my mics from the snake during sound check!

I was pretty depressed; in my 19 years as an engineer, I'd never had to cancel a live performance for technical reasons. After nine hours on-site dealing with this, it was now 6 p.m., with the show scheduled for 7. The electric company guys returned at 6:15 with the ugliest-looking transformer I'd ever seen. This mass of metal, wires, and tape was a multitap isolation transformer: two hot legs from the drop were used as input; the 120 V output ran to my transformer's input. We still used the same bathroom pipe for my neutral and ground. Hallelujah—it worked! I hurriedly tuned the system and sound checked the group in 15 minutes. The show came off, although it was a struggle to get any kind of intelligibility in this huge reverberant space.

SOUTH KOREA—SEOUL

Kimpo Airport is a very modern facility with spacious baggage claim facilities. Oversize pieces are brought through large doors at either end of the baggage claim area. Customs clearance is a snap, provided you can prove your equipment will leave the country with you. Be forewarned that traffic in and around Seoul is heavy; it's said that every hour is rush hour, so imagine what rush hour is like!

Charlie performed twice in Seoul. Our first performance was an informal 20 minute mini-concert during a reception at the residence of Donald Gregg, the United States Ambassador to Korea. We set up in his living room, using only the drums, guitar amp and bass amp.

We had quite a cross-section of Korean musicians present, from a famous Pansori (traditional operetta) singer to jazz singers to pop musicians. Several got up and performed for us after our brief set.

Their initial shyness gave way to enthusiasm as the trio responded positively to their music. A jam session was soon underway. One of the Korean singers brought along a hi-Z mic and cable; I ran this into the aux input of Charlie's guitar amp, using it to amplify our "guest" vocalists. Maybe not high-tech, but effective for singers in a space this small. That was the only mixing I did; I could really relax and enjoy all the great music that went down.

The second Seoul concert took place at the Seoul Arts Center, a huge multi-venue performing arts complex. We'd held a press conference the previous day at the Recital Hall, and the main performance would be in the Concert Hall. This beautiful theater seated about 2,300 people. Primarily designed for classical music, the acoustics were everything you'd want from a symphony hall. AC power came from a European-style grounded outlet stage right; it provided 225 volts with a clean neutral and good ground.

My PA system was two JBL two-way floor monitors per side, with wall-mounted speaker columns filling in the side and under-balcony areas. The JBL wedges sounded good, but the column speakers didn't have any low frequencies to speak of.

Fortunately, the hall acoustics were so good that we didn't need a whole lot of reinforcement. I worked with the house tech to balance these various systems, minimizing the influence of the column speakers with respect to the wedges. The result was smooth sound everywhere. Unfortunately, when I returned for sound check, a new tech was on duty.

Not only did he change all the preset house levels, but, in an attempt to get a tape feed, he unplugged half my mics from the snake during sound check! Through some fast work, I was able to salvage the sound check, but I'd learned an important lesson about Korean theaters. Technicians can change during the day; insist that the same tech do both setup *and* performance, or forbid any changes after setup.

CHONJU

Our travel throughout the Korean peninsula was via road: the equipment and I travelled in a large cargo van while the trio and our support staff travelled in an embassy carry-all. The drive from Seoul to Chonju took about 3 and½ hours, along excellent roads akin to the American interstate highway system. The trio played at the Cholla Bukdo Student's Hall which seated around 1,000. The house PA system here was based on Altec-Lansing single 15 in. cabinets coupled with multicell horn/drivers: three woofer/horn sets were flown dead center over the proscenium, with one woofer/horn on each side of the stage. I discovered that the down house right horn was dead. Another snag to deal with: my specs called for all system components to be checked and fully operational BEFORE we arrived. We notified our upcoming venues that we expected this to be done.

Power came from a drop in the dressing room, about 40 feet offstage left. This provided 104 volts/leg, which I stepped up to 120 with my Variac. The room had a lot of hard surfaces, but the capacity crowd soaked up a lot of this ambience. My biggest problem was the lack of highs on the far right side of the room. I tried to compensate by using a separate graphic channel for that stack, boosting highs to the full-range woofer. It was marginally successful, but most of our smallish audience never noticed.

KWANGJU

The drive from Chonju took only 1 and 1/2 hours. Our venue, the Namdo Arts Center Auditorium, was on the third floor of the Arts Center building—there was no elevator. I removed everything pertinent from the road cases; the gear was carried up piecemeal, while the cases were stashed under a stairwell on the first floor. The hall seated 500 people and reverb time was a very manageable 1.5 seconds. Power came from a U.S.-type ungrounded outlet just off stage left; voltage was only 105 volts. By this point, I was very happy I'd brought my Variac along. I had to step power up more often in Korea than anywhere else I visited during the tour. My ground was procured from a water pipe in the bathroom just behind the stage, about 35 feet from my electrical outlet.

My interconnection with the house PA created a small buzz; I eliminated it by using an in-line pin #1 lift on my feed cable.

The facility featured a brand new E-V PA system, mounted behind a grill positioned above the proscenium opening. Interfacing our two systems was not trouble-free: I had to run my feed through a direct box and lift the ground to eliminate buzz and hum. The evening's concert was sold out, with people standing three-deep in the rear aisle. This crowd was surprisingly demonstrative, clapping along with Charlie's intermittent vocalizing; they were definitely our best Korean audience.

TAEGU

The drive from Kwangju takes about four hours. Since we had a day off to travel, we elected to stop along the way, visiting the Haeinsa temple complex in Kayasan National Park. We'd completed our journey by late afternoon; our concert was held the following day at the Taegu Culture and Arts Center's auditorium. This venue seated 1,100, with a good portion of this in a large balcony. AC power came from a panel off stage right. This provided 220 volts; the neutral was clean, and the ground came from the panel chassis.

House PA was provided by an extensive distributed system. JBL speakers were mounted in a grill above the proscenium opening and along the side walls; there were ceiling speakers under the balcony and in the rear aisle areas. I could adjust the level of these systems separately for smooth audience coverage. The show sounded excellent; I'd rate the PA system as the best we saw in Korea. Our audience was dotted with United States servicemen from the Taegu base; their enthusiastic applause inspired the more conservative Koreans to join in.

PUSAN

The drive from Taegu took about three hours, some of it over two-lane roads. Pusan, a major port, is the second largest city in Korea; traffic here, while not as relentless as Seoul, was quite heavy during rush hour. We performed at the Pusan Cultural Center concert hall which was remarkably similar to the Seoul Arts Center in construction, although it was smaller, seating only 1,700. Acoustics weren't quite as lively, due to the carpeted floor. The PA system was mounted behind grills on either side of the proscenium opening; it was purportedly a JBL component system. Power came from a drop offstage right, providing 220 volts and a good ground. My interconnection with the house PA created a small buzz; I eliminated it by using an inline pin #1 lift on my feed cable. The near-capacity crowd was treated to an excellent concert. Charlie, ever the perfectionist, was very pleased with the trio's performance, and all three guys expressed their happiness ebulliently. They knew they'd nailed it.

NEXT ISSUE

With either Byrd or Toups, we'll visit Indonesia, Malaysia, Sri Lanka, New Zealand and Fiji; both groups visit Singapore and Thailand. Regular readers of **db Magazine** may recall my past articles about tours of these countries with groups like the Benny Golson All-Stars and the Barrett Sisters (both in 1987). Much has changed in the past three years! I was amazed at how sophisticated local PA equipment had become, especially in some of the smaller cities where little, if any, PA equipment had been available before. The changes are significant; we'll explore the improvements (and some trouble spots) in my next installment. db

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In a facility such as The Enterprise, a "constellation of digital command centers dedicated to the pursuit of ever-greater sonic potential," it is only fitting that the walls should speak of an ancient future past; pillars in bas-relief, ornamental geometric shapes and a warm, pastel ambience reminiscent of soft sculpting breezes on Egyptian desert sands.

PYRAMIDS—BEACONS OF THE GODS, whisper of an era when mortals walked with gilded angels—great beings in flying ships descended from the heavens. The post-modern, Memphis-style design, spacious, triple-tiered control rooms and cutting edge technology evoke a fabled futurism—a rarified atmosphere ripe for creation and long hours of work without worry of fatigue. The light and airy surroundings of The Enterprise bestow an enervating, psychologically positive effect on all whom work within, from songwriter Julian Lennon to producers Beau Hill and Babyface.

"The original concept for The Enterprise was in the tradition of the television series—to push into new frontiers of sound and technology," comments Craig Huxley, founder and CEO of The Enterprise. As a child, he appeared on the original Star Trek television series as Captain Kirk's nephew. Acting, however, was only one mode of creative expression for Huxley. Appearing initially as Craig Hundley (his given moniker), he was recognized at only 14 years of age as an incredible jazz keyboardist and musician. Huxley was a phenomenon, a boy genius with the conceptual vision of DaVinci and the hands of Oscar Peterson. His trio toured as an opening act with several major recording artists on world tours including Bill Cosby and Johnny Mathis. He has also appeared as a guest solo artist with the Milwaukee, Florida and Denver Symphonies.

Huxley's penchant for music led to film scoring and session work for some of the hottest recording artists on the planet.

Indeed, it is Huxley's keyboard wizardry one hears driving the subtle percolating textures of Michael Jackson's *Billy Jean* and the silky, majestic growl of *Thriller*. His keyboards grace the scores of China Beach, Knots Landing, and Dallas.

Huxley wears the hat of player, composer, arranger/programmer, sound designer and producer. His credits for sound design, composition and production of ad spots, television programs, feature films and mini-series are numerous and include the "Genesis Project Sequences" from Star Trek II, III and IV; 2010: A Space Odyssey; Cat People; Firefox; Family Of Spies (for which he won an Emmy); and ad spots for CBS, Panasonic, Computerland, Yamaha Motorcycles and Infiniti.

THE DREAM

"I dreamed of building a facility that would incorporate the spaciousness and high ceilings of a film re-recording stage with all the features of a state-of-the-art recording studio utilizing the latest in technology," Huxley said.

His dream flourished into a multi-suite complex featuring triple-tiered "mini-theater" control suites (the envy of any starship captain), "live" rooms large enough to accommodate a small symphony, and a 30,000 square foot concrete warehouse for "very" live tracking. The facility is also configured with a programming/pre-production room, isolation booths, a central machine room, techno-shop, kitchenettes, client areas, reception area and an administrative wing. The Enterprise's virtual design is such that all four control rooms are interfaceable to any combination of live studios, isolation booths and the pre-production room. All four Enterprise control rooms are equipped with full video projection capability. Mix positions face large view screens rather than studio glass. The control suites are equipped with Augspurger five-channel surround-sound monitoring systems.

Studio A, the Arena, is spectacular—triple-tiered, futuristic, and the most outstanding room for live re-

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cording. Featuring a Neve VR 72-channel console, the room has been used on a frequent basis by Beau Hill, producer for bands Warrant and Winger. Also triple-tiered, Studio B, the Bridge, is an exceptional mixdown and film re-recording stage which has played host to numerous feature films, TV movies and mini-series including this year's Godfather III and Huxley's own Family of Spies. Both the Bridge and Studio C, the Cabin, feature brand new 80-channel Solid State Logic 4080 G Series consoles, the first of their kind available in America. The new SSL consoles are equipped with sensational new features and software updates unavailable anywhere else in the United States to date.

New England Digital Synclavier and Post Pro direct-to-disk systems fan out all over the complex. All front end electronics and controllers tie line to a central machine room housing NED towers and direct-to-disk drives.

EQUIPMENT

Each of the studios is equipped with dedicated:

Studer A820s
Neve Stereo 33609
AMS DMX Stereo
Lexicon 480L
Lexicon PCM 70
Massenburg 8200 equalizer
Eventide H3000SEs
Roland 3000 Delay
dbx 160x compressors
UREI 1176 compressors
Yamaha SPX90
dbx 900 rack with 902S and 904S
Drawmer gates
Pultec Equalizer
Aphex II Aural Exciter
TC 2290, CD player with 8X oversampling
Yamaha NS10 nearfields
Otari MTR-12 two and four-track stereo tape machines
Augspurger monitors with Class A amps.

OPTIONAL EQUIPMENT

Optionally available on just a moment's notice are:

Mitsubishi X880 Digital 32-track and HS X86 digital 2-track tape machines
Sony 3324A and 3348 digital tape machines
Otari MTR 90 MKII 24, 16, or 1 in. 8-track tape machines
Dolby SR
Lynx module
Sony 1 in. BVH 3000
Dolby SP24
AMS DMX Stereo (6.5)
AMS RMX
Publison Infernal 90
Cyclosonic Panners
Fairchild 670
Neve Limiters
Yamaha Rev 5
Quantec XLC
NED Super Synclavier 9600 with optical gigabytes and 64 voices
NED 16-track direct-to-disk recorder
Emulator II
Roland D-550
Korg M1
Yamaha TX 802
Linn 9000+

—and a host of additional sound modules and effects.

A unique feature of The Enterprise is the techno-shop where the staff's diverse group of sound designer/programmer/engineers experiment with new technologies, hybrid instruments, and totally unheard-of methods of music and effects generation. This rarified atmosphere has given birth to Huxley's brainchildren—the "Blaster Beam" and the "Flexitonal Tubulon." The Tubulon is an aluminum-based instrument tuned to a 53-tone octave, capable of very subtle nuances of pitch including a neutral 3rd, neither major nor minor. Its timbre is so beguiling that it was used to identify "Bowman" on the 2010 soundtrack produced by Huxley at The Enterprise. The "Blaster Beam"— possessed of a powerful sliding low end, has been used extensively in all of the Star Trek movies for a variety of timbres and effects. Both the "Beam" and the "Tubulon" have made appearances on Huxley's Quantum Mechanix CD released on the Enterprise's Sonic Atmospheres Label

While The Enterprise was initially conceived of for film scoring and mixing, its cutting edge reputation spread quickly throughout Los Angeles...

Another favorite instrument in Huxley's avant-garde collection is the 19-tone clavichord designed by Erv Wilson. Huxley and crew sample these instruments and process the results in a variety of ways. They've recently completed and released a CD of exotic sounds for Synclavier and E-mu libraries called "Exotica I." The sounds are hybrid samples of Huxley's exotic, futuristic instruments melded to sounds and effects generated and sampled from ancient Japanese, Peruvian and African instruments. The combination and juxtaposition of electronic, metallic, futuristic samples, and raw, organic sounds is a familiar theme at The Enterprise.

"When composer Elliot Goldenthal was in working on Drug Store Cowboy, he would compose and record basic tracks and lots of exotic percussion, then bring solo players in, roll the picture and describe the scene emotionally," said staff mixer/NED programmer Joel Iwataki. "He'd set up a framework and then let the players improvise over the recorded changes. One player had an Australian Aborigine instrument called a Didgridu—a hollow wooden drone instrument with a long tube. The player used circular breathing techniques to get these incredible overtones. It was amazing. Over the top of that, Elliot tracked a bop sax player. We ended up with this incredible layering of tones and textures totally appropriate for the film. Elliot selected and featured different solo instruments at various points in the film mixdown. It turned out great," he said.

Goldenthal is one of an eclectic group of talented composers working in a variety of visual media at The Enterprise. The group includes Jerrold Immel (Knots Landing), Paul Chihara (China Beach, Across Delancey Street), David Shire (2010, The Great L.A. Quake), Stewart Copeland and Huxley himself. "Once we decided to build The Enterprise, it took two years and we did have some hair-raising moments," laughed Huxley. "During the construction of the second wing, there was a room full of unique, custom furniture that had been designed and built for The Enterprise by local artist Cameron Aston. The room also contained a collection of instruments, an antique clavichord 4 and a Blaster Beam. There was a bursting sound when wet cement came pouring into the room from over the roof.

525 Post Production: A Jaunt Through Digitaland in Hollywood

FRED JONES

It was one of those Day-For-Night kind of afternoons we get so often in Hollywood. A beautiful Technicolor sunset (the kind you usually see only in the movies) was just beginning to set in the West, as the sounds of war in the Middle East came over the radio.

As I walked into the more than thirty-foot-high art deco lobby of what was once, appropriately enough, a part of Technicolor in the late 1930s, I couldn't help thinking that it was an interesting juxtaposition to have such a beautiful old building being used as an audio and video editing facility on the cutting edge of technology.

After being served a Coke in a glass with ice by the friendly person whose job that is, I was introduced to Tom Davis, director of Audio at 525 Post Production. You may not recognize him by name, but you will most likely know his work.

Figure 1. 525 Post Production Studio A.

After about a thirty minute tour of what is arguably one of the finest video editing facilities in the business with every format known to man and all the digital video toys you could ever need (and then some), we settled into the audio portion of the facility.

Davis has worked in the entertainment business in Los Angeles since 1975. He started behind the camera, worked his way through various positions in film, video and music production, and eventually wound up being a video editor. His first love of audio won out, however, and in 1986, he became co-founder of POST LOGIC. In just under four years, Post Logic has become one of the largest and best equipped independent facilities in the industry.

Fred Jones has been active in Los Angeles as both a major studio owner and a prolific writer of pro-audio articles.

BACKGROUND

Davis has worked on such diverse projects as "The Rolling Stones 25x5" documentary, "The Tracy Ullman Show," many Chrysler-Plymouth commercials, promos for all four major networks and rock videos for Madonna, Van Halen, Fleetwood Mac, Bon Jovi, Bobby Brown, Linda Ronstadt, Eurythmics, Phil Collins and Ziggy Marley.

In March of 1990, Davis needed a new challenge and became director of Audio for 525 Post Production, putting the once picture-only oriented company into the audio for video business.

Prior to his coming aboard, Steve Hendricks and Kelvin Duckett (the founders of 525 Post) in June of 1987 believed there was room for a very high end facility that would specialize in music videos and cater to special effects clients. "We felt there was a real need in the marketplace for a facility that had all the technical expertise, as well as the creature comforts," Hendricks said.

Duckett had been working in Europe as an editor doing most of the top music videos at a place called Visions. "I came over to freelance in L.A.," says Duckett, "and couldn't find a place that really equalled where I was working in London. Steve had been working at The Post Group and I learned he also wanted something better, too." The result was the founding of 525 Post.

"Although we started as primarily a music video facility," said Hendricks, "our mix of business is now about 35 percent music video, 30 percent broadcast, 25 percent commercials and 10 percent high-end industrial and home video."

When Davis was approached for the Audio Director position, he told Hendricks and Duckett they needed a "Rolls Royce" kind of room that would match up to the rest of the facility. The owners' attitude was, "You've got it! Get the best people and the best equipment and here's the budget to do it right!"

Davis' background in the music business told him to go for the highest quality of sound possible. "I needed equipment that top musical talent would feel comfortable seeing when they came in, since we would be doing a good number of rock videos in the room. I also wanted the most flexibility to be able to do any type of project that came in the door, with full automation capabilities in the console and a monitor system that could handle normal TV. mixing levels as well as high SPL levels, and still sound good."

To accomplish his goal, Davis brought in Vincent Van Haaff of Waterland Construction. "Vincent has built a large number of rooms in the music business, such as A&M records, and I needed a room that rock 'n rollers liked the sound of, as well as the commercial clients," Davis said. "We felt that it was very important to get the monitor system right. It is a custom design using TAD components and has the ability to produce 3,000 watts of non-distorted power if necessary.

GETTING THE BEST

"Since a Virtual console is at least a few years away, let's get the best analog desk we can find," Davis said. His choice was a Neve VRP 48 channel with Flying Faders. "It's a console that has proven itself many times in the music industry, plus I think it's the best sounding analog console there is, and when you add in the automation, you have a console whose functionality is unbeatable. Besides, good analog sounds better than bad digital," he said.

For synchronization purposes, an Adams-Smith 2600 A/V was chosen primarily for its ability to emulate both CMX and Grass Valley editors, and accept their edit lists directly from the edit session from any room in the facility. "Also, the unit's keyboard is similar to a CMX editor, so it is really familiar to me," Davis said. If necessary, he can take control of any of 525's video decks anywhere they may be located via a custom computer switching network.

"Aside from the fact that I am familiar with it, I chose the 2600 A/V because it saves me a lot of time due to the fact that it can do linear type preview editing, automatic record edits, and it's the best synchronizer to use for the type of heavy machine manipulation that we sometimes have to do here," Davis said.

There is also a custom-designed Macintosh-based sound effect data base system that allows Davis to rapidly find and insert into the project he is working on any effect in his library from various sources as diverse as Compact Disc (using a Sony CDK-006 60 disk changer) to Emulator III samples.

STUDIO A

Studio A has an extensive array of outboard gear that includes a Lexicon 480L, an Eventide H3000 (which he used in conjunction with the 2600 A/V's Supersync feature) for time compression, and practically every other item needed to fix any kind of problem that might occur, all in a rack situated behind the mixing position. This rack also acts as a cleverly designed client table/desk.

Additionally, the studio is equipped with the latest analog tape machines, such as an Otari MTR-100 with Dolby A and SR noise reduction systems, several four-tracks, two-tracks and DAT machines. All of these are located in a room just off to the right hand side of the control room. The Digital Dynamics ProDisk-464, however, handles the majority of the studio's multi-track recording duties. "This unit is one of the reasons that Studio A at 525 really stands out among other rooms around town," Davis said.

One of the things he is insistent about is not using this new technology in a way that scares the clients. "If I'm working with someone who is really into the technology, I share it with them. Often I don't even tell the client I'm using (the ProDisk-464). Other times I will use the analog multi-track, if that's what the client wants."

The rate card amount includes the mixing engineer (Davis is currently the only mixer at 525), second engineer and all the equipment available in the audio department. The rate even stays the same whether you are using the analog, digital or even both multi-track machines. The advantage to this, according to Davis, is that the engineer can make the decision of what equipment is best suited (both audibly and time efficient) to the client's needs.

Quite often on a project, a week will go by and the analog machines are not even fired up. Davis works with the Digital Dynamics System the whole week. He backs up the material from the ProDisk-464 on to 8mm exabite.

"I have six hard drives and six back-up tapes, so for large projects that have a lot of tracks, I can backup real fast. Plus they're cheap! Maybe five bucks a tape, that makes it $30, as opposed to a roll of 2 in. tape at $140!" he says.

Many engineers and facilities have concerns with hard disk recorders due to long backup times, because they may have to be done in real time. On the ProDisk-464, the backup 'exabite' process itself is one-third real time. Davis says it can be even faster if you are running multiple drives as he is.

He qualifies this, however, by stating that if a project has a running time over 15 minutes, it doesn't make sense to run the whole thing in the ProDisk system.

"If it is over 15 minutes long, I'll lay it down on the analog multi-track from the one inch (if I think there will be some serious cutting and splicing going on, I'll go to the ProDisk), then do all the pre-building of the sound effects, music and voice-over on the ProDisk." He then locks up the multi-track and the ProDisk together on the Neve VRP 48 for an automated mix to the final format.

In discussing some problems that the ProDisk can solve better than when dealing in an analog situation, the first instance that came to Davis' mind was as follows:

"We were about ten hours into a pre-building session. Everyone was tired and I was in the final stages of the mix when a new guy came in and said, 'Wouldn't that sound better if the voice-over came in about a half a second sooner and was paced a little faster? Would that be much of a problem to fix?' "

As you well know, if you're in analog this "little problem" means you have to strip the VO off the 24 track, edit it on a 2 track, then re-sync it back to the 24 track. Then before starting the mix over again, the client might decide he doesn't like it, so you have to go back to where you originally were, and if you've pulled everything off, this could really become a nightmare!

"Using the ProDisk," says Davis, "I can adjust any single track in relation to any other track, so if they want to shift something a half-second, it's real easy. But I can also adjust *within* a track, so if they want to change the pacing, I just slip and slide it a little in just a few seconds (instead of hours) and go on with the mix.

"And if they don't like the changes," he says, "I can simply recall the original version, and I haven't left my chair, called in the second, or even disturbed the automation for the mix!"

Davis continued, "A really exciting part of using the ProDisk-464 is that your clients grow with you,"

Take, for example, the scenario above. You've been in session ten hours and everyone's tired. If you are dealing with an experienced client, they will know how long it will take to adjust that voice-over a half a second and may decide not to go for it because of the hassle factor. Now that Davis' clients are getting used to these advantages, they have the freedom to experiment and play with the production more.

Another sterling example of the system's versatility came when a client wanted to make two different audio versions of the same commercial. There were two producers coming from two different creative directions at once. The visuals and some of the elements were the same, but they wanted to have different music, effects and voice-over in various places throughout the spots. In fact, the composer of the music was in the control room playing a Fairlight to the track while the producers were making decisions and changes on the spot.

"I simply duplicated the project in the ProDisk-464 and built two spots simultaneously," said Davis. "At the end, I had rough automation built up that I held in the RAM of the Neve, so it made mixing the two versions a piece of cake."

Davis has many mag-oriented clients that particularly like working with this system because they can slip any sound within the track. Which is one step beyond mag in which you would slip the whole track.

When 525 did a trailer (preview/coming attraction) for a home video release, the producer cut the spot together with a video tape copy of the film with a composite mix track. Normally, you would do a rough cut with this track and then order up individual elements to make the spot. However, the deadlines were such that the spot had to be done with the composite mix track. This meant the elements came in on a four-track with time code that meant nothing. The producer knew where all the pieces were on the four track, so Davis loaded them as individual sections into the ProDisk and gave each piece a name, not paying any attention to the code. It was like putting together an audio jigsaw puzzle. Using the 3⁄4 in. rough cut as a reference, he re-synced everything up by grabbing time code on the fly from VITC, and in no time at all, he had built this spot from elements that would have taken days to put together manually.

After all these examples, I asked Davis if there was anything the ProDisk wouldn't do. He laughed and replied, "Yeah, it won't get me a Coke and a glass with ice in it. But all I have to do is call up front and THEY will bring me one!"

Isn't technology wonderful? db

Historical PERSPECTIVES

Rudy Vallee, a popular crooner during the 1930s and 1940s, performs with a Shure Brothers Model 55C microphone, a variation of the Model 55 mic which was produced in the late 1930s. The mic is significant because it was the first single cartridge uni-directional mic ever made. It's worthy of note that Shure still makes and sells a version of this mic.

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JOHN BARILLA

Natural and Organic Recording at Sunset Ridge

"I joined the Navy to see the world, and they sent me to Chelsea!" Not Chelsea across the sea in England, but Chelsea down the interstate highway in Massachusetts—about an hour's drive from the family farm in Hampton Falls, NH. So recalls Christopher Biggi, owner of Sunset Ridge Recording, of the place where he first got interested in capturing music on tape.

CHELSEA, HOMEPORT FOR THE UNITED States Navy, is a working class community separated from Boston by the Mystic river. Its bleak skyline of crooked, grey, three-deck apartment houses, peopled by struggling families, tough-skinned dock workers and aspiring musicians, exudes an aura of gritty New England romance. It is much the same today as it was during the late 1960s when Biggi was first stationed there, with one notable exception: back then there was a sense of musical community—a creative crescendo—that has not been heard since, and Biggi, armed with a venerable Wollensak monaural tape recorder, was there to preserve that moment in musical history.

Combat recording—that's what he calls it. Being in the right place at the right time, having the mic open, levels set, the record button pushed without missing a beat, and coming back to the nightclub/coffeehouse battlefield again and again to apprehend that single instance of musical magic from amongst the chaos of the times.

This was Biggi's passion and he did it again and again until he got good at it, and he carries that passion with him today.

By the early 1980s, Biggi was ready to explore the world of multitrack recording. What spurred the change? Certainly, live music died a slow death in the '70s. The wind that had fanned the creative fever in Boston a decade earlier had died down; there was no more war to protest, no sexual taboo left to be broken, and the casualties of the drug culture were too noticeable to be dismissed.

Figure 1. The Sunset Ridge control room. Tascam M600 console and GA30 EQ with Auratone and Westlake Audio speakers.

There was no musical frontier, except, of course, for technology. Music was now manufactured, rather than captured in studios. Biggi gradually transformed the basement of his house on Sunset Ridge into a recording studio, starting initially with a 4-track cassette, then 8- and 16-track open reel formats from various manufacturers until arriving at his present system of strictly Tascam recorders and mixing console.

His first 8-track and 16-track were from another major manufacturer. "I was using that stuff basically for the economy of it," Biggi says. Despite the cost-effectiveness of those products, he found himself dissatisfied with the overall quality. According to Biggi, "the low end, the whole bottom end, was just lacking." On the advice of studio installation consultant Bob Morrissette, he switched over to a complete Tascam system and was able to find the sound he desired. Some of Biggi's clients were saying, "You switched from 16 track to 16 track? Why bother?" to which he replied, "I was working on a project when I made the switch—shooting for the same sound in both sessions, and it was such a dramatic difference in headroom and everything."

The current list of equipment at Sunset Ridge includes a Tascam M-600 series console, a Tascam MS-16 1 in. multi-track and a Tascam 42B 2-track. Biggi is particularly bullish on the M-600 console for several reasons. For one, he is pleased with the warmth of the console which is a major factor in his overall sound.

Figure 2. A Tascam 42B 2-track and MS-16 16-track are to the right of the console.

Second, the 600 series can be flexibly configured, and the 32 input/32 monitor board with 8 independent sends gave him a flexibility he had not experienced before. Third, he had the last eight input modules replaced with stereo modules (two channels ganged together on one fader with symmetrical EQ). These modules have proven themselves valuable by providing convenient stereo control over all playback machines, stereo keyboards and effects returns.

Likewise, the MS-16 solved more than one of Biggi's recording problems. It wasn't only tightening up of the bass sound which sold him on the machine, but also the increased headroom and dynamic range afforded by the larger tape format and dbx Type I noise reduction. There were also some unforseen, but beneficial side effects of the new system— the kinds of things that cannot be written into spec sheets, nor are they immediately noticeable, but display themselves in rather subtle ways. For example, Biggi notes: "Once I got this board and the MS-16, I noticed that it compresses the tracks to a degree, 'cause when I go into mixes, I compress a lot less than I used to."

(*Ed. Note:* The dbx does indeed compress tracks upon recording and reverses this process on playback by expanding the signal. Theoretically, the result should be an exact replica of the input signal, but with up to 30 dB less noise than if recorded without dbx. The results, however, are not without controversy. Some people feel there is a residual compression that can be heard on playback. Some, like Biggi, enjoy this effect and use it creatively; others find it somewhat disconcerting.)

The Tascam 42B was formerly the main mixdown machine at Sunset Ridge, but Biggi has found his Panasonic SV-3500 PCM DAT recorder to be the most frequently used. Still, the 42B gets put to good use in rock 'n roll and other sessions where a good dose of analog warmth is required as part of the overall sound. "The hardest thing about digital recording," says Biggi, "is to not have it sound too trebly. You have to be careful. I'm pleased with what I get, but a lot of people make that mistake. If a rock band wants a real hot type of sound, I mix initially to digital, then lay it onto the half-track for tape saturation. That's the only way."

BUILDING A BUSINESS

Getting the MS-16 and the M-600 console seemed to put Sunset Ridge over the edge from being a nice home studio to being a professional studio. Even though it was located in rural New Hampshire, word spread and business took an upturn. "Once I went to the MS-16, I was doing quite well," Biggi said. "As far as business goes, it's basically word of mouth—which is sort of a built-in quality control. Anyway, it's worked out that way. If there's something I don't like, I really don't do it very long which is a wonderful position to be in. It's really ideal. I figure that my sanity is just as important as anything else.

"I do a lot of individual productions, some in the pop-type line—you know, classic demos, but I'm also doing folky type projects where people are just putting out their own tapes and CDs," he says. "And I think a lot more people are drifting towards that now, anyway, 'cause they're tired of beating their heads against doors that aren't opening for them."

As an authentic witness to the last great era of folk music some twenty years ago, Biggi is in good position to catch the next wave of contemporary folk music as it begins to crest.

(*Ed. Note:* So called "custom" CD projects are becoming popular options for unsigned artists. As prices for small quantities have dropped, retail prices have remained high, so the number of copies which should be sold in order to break even is rather low. Many gigging artists can do that without even getting airplay.)

Although Sunset Ridge is located in a rural area, it draws a large part of its business from the metropolitan Boston area. Seeking respite from the fast-paced city life, Biggi says, "(people) can come up here and relax; it makes quite a difference, I think." In addition to providing an excellent recording environment, Biggi usually finds himself involved on the production side as well, since his clients develop a close working relationship with him, trusting and even soliciting his ideas and concepts. The reason for this unusual bonding between Biggi and his clients is simply because most of them are repeat clients having worked steadily with him through the years. "A lot of my people developed along with me, as far as what I've been able to do. So most of it is generally a co-production situation," he says.

While many producers today get involved in MIDI programming, Biggi does not. The equipment is there for those who want to use it, but this is certainly not the focus of his business. As an authentic witness to the last great era of folk music

Figure 3. Christopher Biggi caught in a relaxed moment. Note the two racks of equipment at easy reach.

some twenty years ago, Biggi is in good position to catch the next wave of contemporary folk music as it begins to crest.

"I've found up here in New Hampshire, it's really still pretty much a traditional type recording 'cause I do a lot of live stuff and people want to all be there and play their instruments at the same time," he says.

Occasionally a client will come in with pre-programmed MIDI tracks, but many of those tracks are provisional. Bass and drums, for example, are usually dropped out and replaced with real instruments.

STUDIO DESIGN

In building Sunset Ridge, Biggi came across the usual set of problems in working out the sound of his control room. Since he was on a budget, the hi-tech solution (calling in an acoustical consultant to design the control room and build it to specification, then shoot the room and design traps for specific frequencies, etc.) was simply not an option. With the pragmatic posture of a rural New Englander, he took some basic acoustical principles, applied them in a non-technical way, and messed around until coming up with an acceptable sonic environment.

First, Biggi had enough sense to leave well enough alone in the geometry of the room. An irregular shaped foundation wall jutted into the control room area making it an unusual asymmetrical space. The initial temptation was to frame it out to make it look cosmetically even, but he resisted the impulse realizing that having unequal non-parallel walls would hinder the buildup of deleterious standing waves. Biggi strategically placed Sonex in various sections of the room, moving the pieces around experimentally until achieving the desired balance between dry and reflected sound.

Biggi says it is important to make music in an environment free from substance abuse. He knows the dangers of this personally, and he also knows the blessings of being free from their bondage.

Biggi tells of using this same trial and error technique in building bass traps for the room after discovering an annoying low-end boominess. "I had talked to someone who was doing some major installs—we were talking about the cylinders in the back of the room...so I just make a frame, and take a piece of masonite and bend it, and it really works dramatically." (See *Figure 4.*) Biggi did not attempt to tune these traps in any way; he just built them, then moved them around until getting the desired effect. This seemed to take care of the major room resonances. Any remaining wrinkles in the response of the room were ironed out with a slight amount of EQ, utilizing a Tascam GA-30 stereo graphic. Once again, trial and error was the only technique utilized, and the results appear to be quite good.

Although Biggi is confident of the room's sound, he is careful to test the mix in several environments when mixing. This involves playing a dub of the provisional mix in his listening room upstairs in the house (which features the audiophile Snell Model A-1 speakers), in his office (which is deliberately decked with some generic sounding bookshelf speakers), and of course, out in his van. The re-

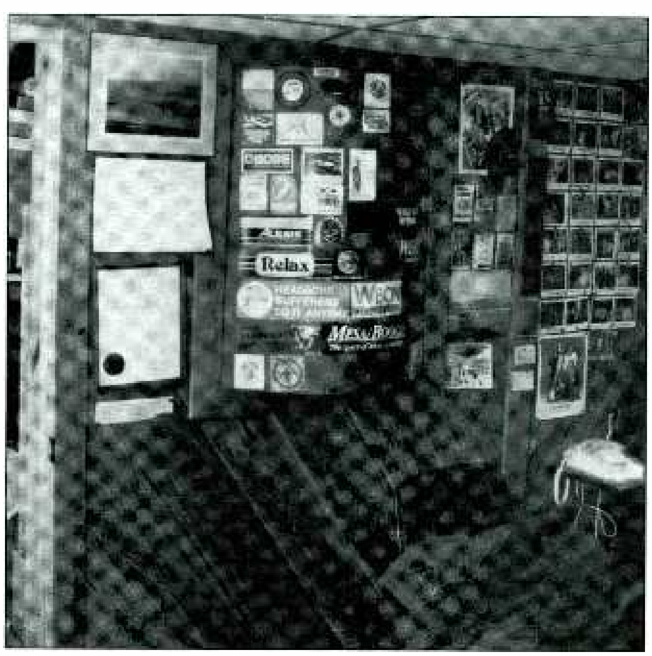


Figure 4. One of Biggi's "strange baffles" with decals and bumper stickers affixed. Note the angled wood finish below.

laxed atmosphere at Sunset Ridge enables clients to do these all important tests before making the final mix.

A DRUG-FREE ENVIRONMENT

Biggi says it is important to make music in an environment free from substance abuse. He knows the dangers of this personally, and he also knows the blessings of being free from their bondage. This single restriction is one he compassionately enforces, for his clients' own good. "What I'm getting," he says, "is a lot of people who have worked at other studios and they've had substance abuse problems and don't want to get trapped into it again. I'm a recovering alcoholic myself. I haven't had a drink in 11 years now. So many of the sessions elsewhere, you know you can't get away from it. And that's sort of another feature of up here: you've got to come here to work."

SOUND BUSINESS PRINCIPLES

Despite the national economic downturn, business at Sunset Ridge has steadily increased without advertising—strictly by word of mouth. Biggi believes if you provide a quality service, you have to hang in there long enough, and the business will start happening—although this process may require several years. "There are studios up here that are dropping like flies, and of course I feel like a vulture when I go get some of their equipment. It's tough. What I see in the economy is that people get in trouble when they expand too much," he says.

NATURAL AND ORGANIC RECORDING

Biggi's typical recording session seems to have more in common with classic 1950's techniques—where all parts (vocals and instruments) are performed simultaneously, live to tape—than 1990's style recording where every piece of outboard gear is strapped onto a single track to make it sound like something other than what it is. Frankly, it's refreshing to see someone faithfully reproducing acoustical sounds or doing subtle processing which enhances the music, rather than drawing attention to himself as a technical wizard. This sort of egoless engineering virtually transparent to the music is rare today. It's a kind of quiet brilliance whose highest compliment is that it sounds natural and organic: like you're right there in the room jamming with the musicians.

In an age where MIDI programming holds sway over the shape of pop music production, it's nice to visit with a guy who never forgot the lost art of mic placement.

The one outstanding difference between Biggi's recording style and authentic 1950's recording is that Biggi most frequently records to multi-track and then mixes. Nonetheless, that freshness, the interactivity of players and singers all being in the same room, responding to the same chord at the same time—all of that, remains. Even if the vocals or mandolin solo are not keepers and need to be replaced, most of the tracks were shared experiences and the energy level carries over to any subsequent overdubs.

IN THE STUDIO

The vocal booth and studio space it subdivides have walls covered with pecky cypress from the Florida Everglades. Not normally used in recording studio interiors, this hardwood is riddled with wormholes from life in the swamp. It is a hauntingly beautiful wood, golden with a slight greenish overcast. Beauty aside, Biggi is happiest about the warm sound it seems to generate. During sessions he often finds himself repositioning players, singers and microphones closer or further from the cypress walls in order to adjust the sound—closer to the walls bringing about a brighter sound.

The vocal booth gets lots of use during these live multi-track sessions. In order to keep the personal interaction intact without massive leakage between mics, the bass is usually taken direct, with electric guitars plugged into a Mesa Boogie amp whose internal speaker has been bypassed in favor of an extension cabinet in another remote room. What's actually being mic'd in the studio is the drum kit (which is hidden behind its own baffles), and whatever acoustic instruments are being played openly in the room.

GOOD TECHNIQUE

Apparently, one of Biggi's major interests is getting a natural and spacious sound on acoustical instruments. One of his favorite techniques, especially on acoustical guitars that have a high quality internal pickup, is to take that direct signal, panned centrally and slightly delayed, and blend it with the output of two critically placed mics, panned hard left and hard right. The mics of choice for Biggi are two AKG C-414 BULS mics—one placed slightly below the "O"-hole of the guitar, and the other placed slightly above the fingerboard (near where it intersects the body). By manipulating the relative levels of the three signals, a wide variety of guitar sounds can be achieved.

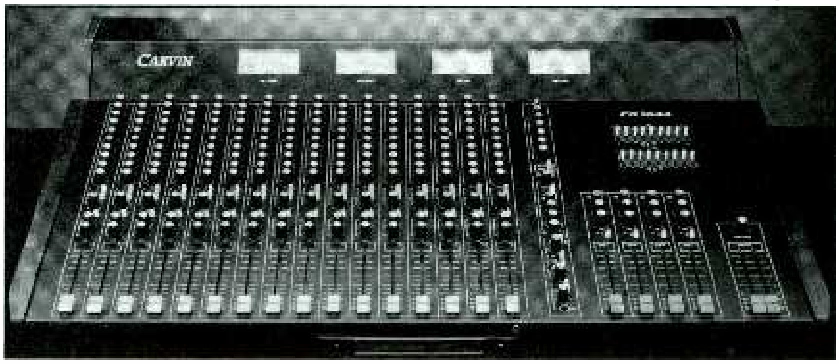
FINAL THOUGHTS

In an age where MIDI programming holds sway over the shape of pop music production, it's nice to visit with a guy who never forgot the lost art of mic placement. From his vantage point in New Hampshire, Biggi has theorized that the immense popularity of MIDI programming in urban areas has something to do with the space limitations of city life. With MIDI, it's easier to get a big sound in a small space, and where space is not a problem, the attractiveness diminishes. Or perhaps it's a limitation of time; it's quicker to use a generic sound from a sampler then to spend a couple of hours moving around mics to find the unique sound of an acoustical instrument. Well, that's life in the fast lane, but at Sunset Ridge, there seems to be no problem with space or time. Even if you carry the tension of urban life into the studio with you, it will soon subside. As an antidote, Biggi often prescribes a short walk just before dusk along the 160 acre expanse of his farm. There and then, you can watch that big, red ball slip into obscurity, and for a moment, there will be peace in the world. db

LAB REPORT

LEN FELDMAN

Carvin's FX1644 Live Sound and Recording Mixer



GENERAL INFORMATION

• The FX1644 is one of the FX44 series of Carvin mixers. The number "16" represents the number of input channels that have been configured into this particular model. Mixers in this series are intended for use in concert, broadcast, theater and recording systems. All the input channels of the FX44 series mixers utilize low-noise differential microphone preamps, separate mic and line gain controls and ultra low noise op-amps. Each input channel features four-band equalization. Each channel also has two monitor sends and four effect sends. Carvin has also added a built-in reverb system for the FX44 consoles. All FX44 series consoles, including the FX1644, feature two built-in nine-band graphic equalizers ahead of the two-track or mono outputs.

Each channel of the FX1644 features a separate printed circuit card. These channel cards are connected with modular connectors and can be removed and replaced easily and quickly, should the need ever arise. Unlike most mixers in which you must select either mic or high level line input for each channel, the FX1644 allows mic and line mixing at each channel, effectively doubling the available inputs.

Channel assignments can be made to a two-track output or to four sub-groups. A pre-fader listen (PFL) system is available for solo auditioning of selected channels or groups. The mixer also offers control room monitoring of 2-track, Monaural or effects return as well as headphone monitoring of control room selection. There is an outlet available for connection of a 12-volt miniature lamp which Carvin can supply as an optional accessory. Phantom power (+48 volts DC) for condenser mic powering is also available from the console. While FX44 series are available with built-in 500 Watt MOSFET power amplifiers, the particular sample we tested did not incorporate any power amplifiers. A description of the input and master controls of this mixer will give you a better idea of how flexible and complete its facilities are.

INPUT CHANNEL CONTROL LAYOUT

Each of the sixteen input channel modules contains an active four-band equalizer. Rotary controls at the top of each channel module alter response at center frequencies of 100 Hz, 500 Hz, 2 kHz and 10 kHz. Six rotary controls for the programmable buses come next. Internal mini DIP switches permit custom bus setups. As preset at the factory, setups are as follows: *Monitor 1 and 2 in set for pre fader and pose EQ; Effects 1 through 4 in* set for post fader and post EQ. Three channel assignment buttons and a pan pot come next. Channel signals can be assigned to any of the four sub-groups or they can bypass the sub-groups and be assigned directly to the two track mix. The pan pot allows panning across a sub-group pair or across the two track mix. Pressing the PFL switch located below the pan pot will illuminate an associated LED and allow you to listen to the channel through the control room monitors or phones before turning up the channel fader.

Instead of having to choose between mic and line inputs with a mic/line switch, the FX1644 allows you to use both inputs simultaneously. Independent mic and line gain controls, located on each channel module, allow you to set the level of each input, thereby effectively doubling the number of available inputs. Theoretically, you could use as many as 16 line level inputs as well as 16 mic inputs all at the same time, if the need ever arises. For easy level setting, these rotary mic and line level controls are located just above the channel fader and peak indicator for each channel.

MASTER EFFECTS STRIP CONTROL LAYOUT

To the right of the 16 channel modules is a single, master effects strip. At the far end of this strip is a BNC connector that will accept any 12 volt mini-lamp. Below the

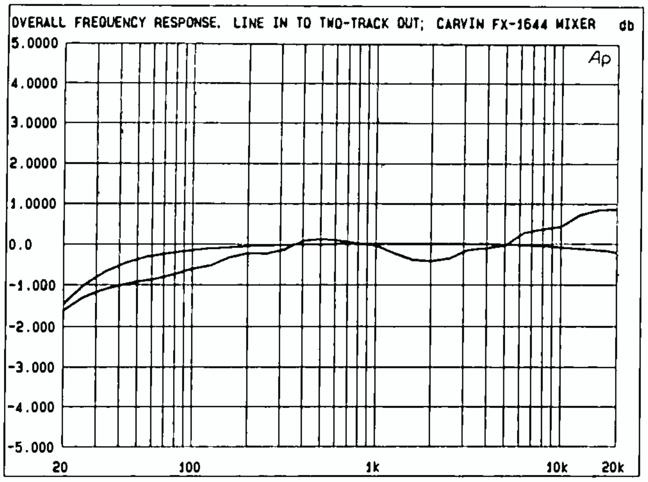


Figure 1. Frequency response. Flatter response was obtained with the master equalizer in the ***off*** *position.*

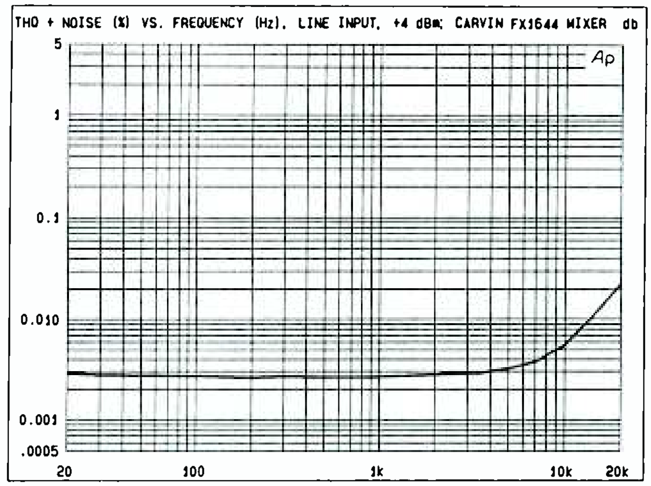


Figure 2. Harmonic distortion plus noise versus frequencvy with +4 dBm applied to the line input and output adjusted for unity gain.

lamp connector are six master send controls that set the overall signal level at the Monitor and Effects outputs. Two meter selection switches come next. Depressing the *Sub-Group 1-2* switch permits monitoring *Mon 1-2*, while depressing the *Sub Group 3-4* switch permits metering the two-track L/R outputs. The 48 volt phantom power switch comes next, and below it are four Effects Returns. *Stereo Return A* can be assigned to the sub-groups and two-track. *Stereo returns B* and *C* feed the two track mix. *Mono return D* can be panned across the two-track mix. The internal reverb system normally feeds effects return *D*. It is automatically defeated when an outboard device is plugged into the *D* return jack on the mixer's rear panel.

Next comes a level control and on/off switch that permits playback of two-track (stereo) sources without using up input channels. Two separate RCA-type phono jacks on the rear panel are available for this purpose. Control room group and talk-back group controls complete the master effects strip layout. With the control room group you can select either two-track, Mon 1-2, or effects return A for monitoring in the control room or via the phones. The talkback group of controls lets you talk back to the L/R mains or Mon 1-2 through a small built-in condenser mic located along the lower edge of this strip.

To the right of the master effects strip, near the upper end of the console panel, there are two nine-band graphic equalizers that allow final fine tuning of left and right two-track outputs or the mono output. They can be bypassed by means of a pushbutton switch nearby.

SUB GROUP CONTROL LAYOUT

Each of the four sub-group strips contains a bus/tape switch that selects the signal feeding the sub-group's 2 track and Mon 1 *sends*. A *Sub Monitor 1 in* level control sends the sub-group signal back to the *Monitor 1 Send*. Two-track *send* and *pan* controls send the sub-group signal to the 2-track mix. A PFL switch interrupts the control room mix and substitutes the sub-group signal for solo auditioning. Finally, a sub-group fader, conveniently located near the operator's hand, sets the level of the group signal at the *Sub Out* and *Tape Out* jacks—two RCA phono jacks and two phone jacks per sub-group.

MASTER SECTION CONTROLS

Two track master faders set the overall level at the two track outputs, which consist of two RCA phono jacks and two balanced XLR connectors. A mono master rotary control directly above the master faders sets the overall level at the mono output, available from a balanced XLR connector.

REAR PANEL LAYOUT

Each channel has two inputs: one for low level mic and one for high level line. Inputs are via a balanced XLR connector and phone jack. Each channel has its own patch jack, allowing you to patch outboard signal processors into the channel. Connections are made through a stereo phone jack (tip and ring). A *Direct Out* is available by partially inserting a ¼ in. phone plug into the Patch jack. The detachable power cord is connected at the rear panel, and nearby is the rocker-type power on/off switch for the entire console. A fuseholder contains a 1 ampere fast-blow fuse and Carvin kindly supplies a spare fuse, "just in case." A tiny slide switch nearby selects 120 volt or 240 volt operation.

Four well-lit VU meters are mounted at a comfortable viewing angle at the far end of the console's front panel. A stereo headphone jack is found up front, at the right end of the short vertical wall of the console.

LABORATORY MEASUREMENTS

There aren't all that many measurements that can be made for a console of this type. Our chief concern was to determine if the console's input channels and sub-group modules have adequate bandwidth, low distortion and a high signal-to-noise ratio. All of these requirements were easily met by the Carvin FX1644, as our tests show.

Figure 1 is a plot of overall frequency response of the console, using the line level inputs. The flatter response curve, which is essentially flat out to 20 kHz and down 1.4 dB at 20 Hz, was plotted with the nine-band equalizer out of the signal path.

When the nine band equalizer was activated, with its slider controls carefully set to their mid-points, additional deviation from flat response was observed. Under those conditions, response was off by +0.9 dB at 20 kHz and by −1.6 dB at 20 Hz. We used a rather expanded vertical scale to illustrate this difference, so don't let the

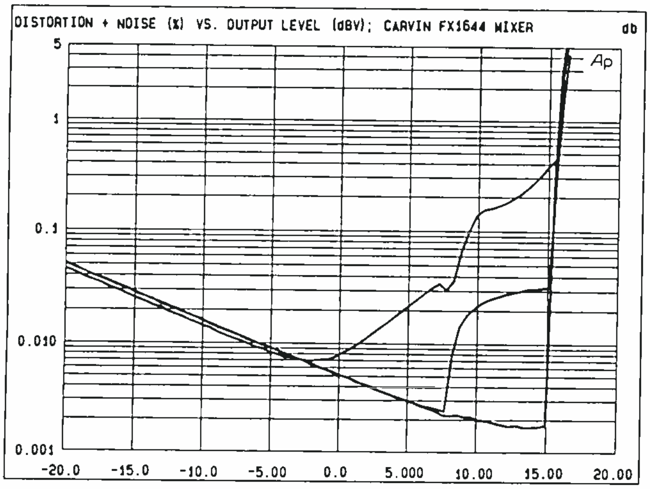


Figure 3. THD + noise versus output level, nominal channnel and master output settings. Middle curve is for 1 kHz, poorest high-level curve is for 20 kHz..

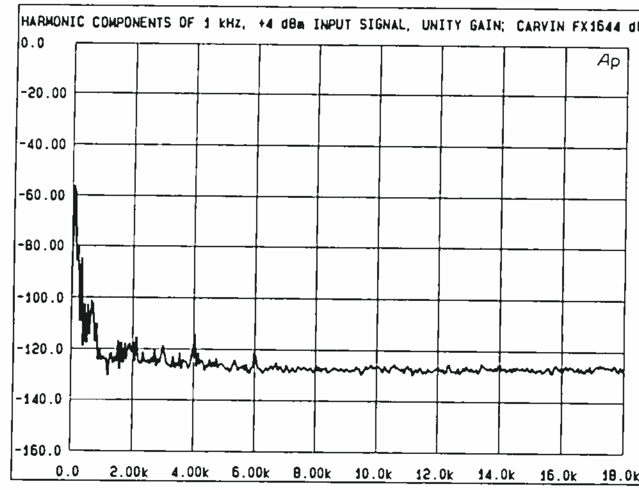


Figure 4. Spectrum analysis of harmonics of +4 dBm 1 kHz signal applied to line input, with gain set for +4 dBm output. Averaged result of 16 acqusitions, to reduce displayed noise.

wiggles in the response curve discourage you. Had we used the more usual calibration of, say, 20 dB for the vertical scale of the graph of *Figure 1*, both curves would have looked almost equally flat.

With a signal of +4 dBm applied to an input and with channel fader and master fader controls adjusted for unity gain (+4 dBm output), we plotted harmonic distortion plus noise as a function of signal frequency. Results, as shown in *Figure 2*, were so good you might easily mistake them for the THD-plus-noise readings of a top quality high-fidelity audio component. At mid- and low-frequencies, THD-plus-noise measured an insignificantly low 0.0026 percent. A slight rise in THD was observed at higher frequencies, but even at 20 kHz, THD plus noise was only 0.021 percent.

Figure 3 shows how harmonic distortion plus noise varied as a function of output level for three test frequencies (1 kHz, 100 Hz and 20 kHz). Up to an output level of +10 dBv, distortion plus noise remained well below Carvin's published ratings, with a reading of 0.002 percent at 100 Hz and 0.022 percent at 1 kHz. For a 20 kHz signal at that same output, THD plus noise was somewhat higher, measuring approximately 0.14 percent. In order to isolate the actual harmonic distortion components from the residual random noise, we used the spectrum analysis capabilities of our test instruments to display those distortion components generated by the console circuitry when a +4 dBm 1 kHz signal is applied to an input, with faders adjusted for unity gain. The results, shown in *Figure 4*, are averaged over 16 successive sweeps, in order to cancel much of the random noise and allow the coherent, harmonic components to stand out. What minute harmonic components there were are all at a level of between –116 and –121 dB below reference level. Taking the square root of the sum of the squares of those components visible at 2 kHz, we calculated an actual THD figure of only 0.00127 percent!

With fader controls set to minimum, signal-to-noise ratio for the FX1644 measured –93 dB below 0 dBv. A spectrum analysis of this residual noise, shown in *Figure 5 (A)*, reveals that most of it is not even random noise, but rather minute contributions of hum and buzz attributable to the built-in power supply of the console and visible in *Figure 5 (A)* at 60 Hz, 180 Hz (the third harmonic of the power supply frequency) and at 300 Hz. Much the same situation held true for the signal-to-noise ratio observed when faders were set to their nominal inputs (both channel and master faders set to nominal *0 dB*

Figure 5 (A). Spectrum analysis of residual noise, referred to 0 dBV output, with all faders set to minimum.

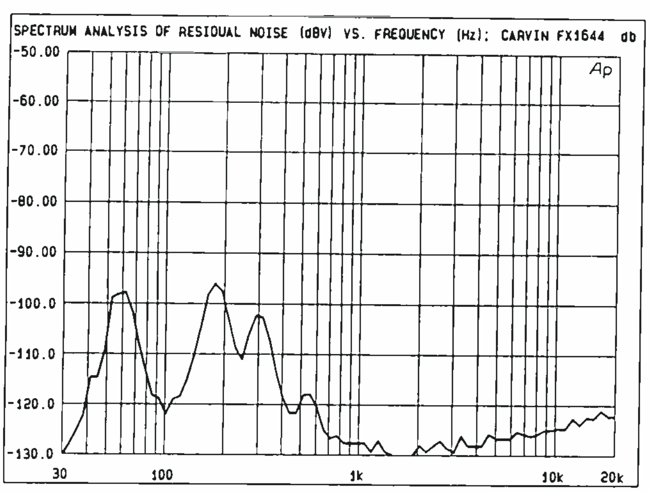
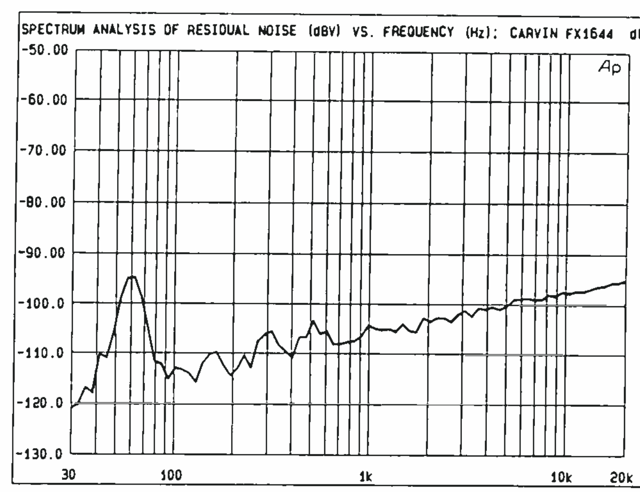


Figure 5 (B). Spectrum analysis of residual noise, referred to 0 dBV output, with all faders set to nonimal settings.



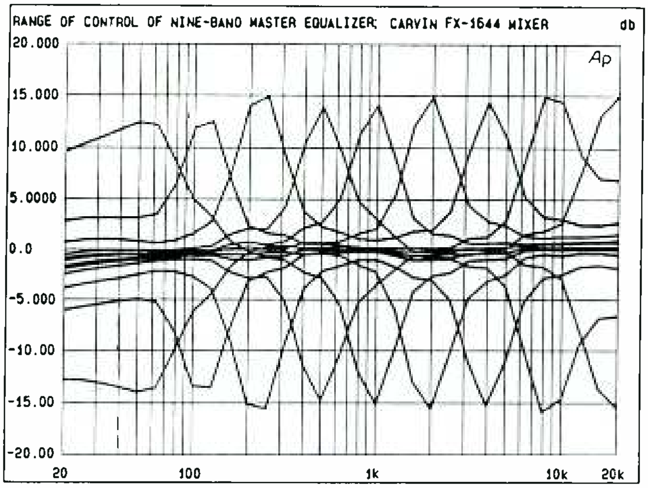


Figure 6. Multiple response curves obtained by setting each slider control of the 9-band equalizer to its maximum boost and maximum cut positions.

marks). In that case, the signal-to-noise ratio was an impressively high –91.4 dB and the corresponding spectrum analysis sweep is shown in *Figure 5 (B)*.

Finally, we checked out the action of the nine-band equalizer. For the multiple sweeps shown in *Figure 6* we adjusted each of the nine controls to maximum and minimum settings. Typically, most of the sliders provided a range close to 15 dB as opposed to the 12 dB specified by Carvin. Center frequencies corresponded very closely to the 63 Hz, 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz, 8 kHz and 16 kHz specified in Carvin's published specifications. Those specifications, incidentally, are very complete. Many, but not all of the published specifications for the FX1644 will be found in our *Vital Statistics* table at the conclusion of this report.

CONCLUSIONS

In our experiments with the FX1644, on the lab bench, we were particularly impressed by the overall layout of this console. While we did not have an opportunity to use it at an actual concert or recording event, it was easy to see that operators at such events would have no trouble at all familiarizing themselves with its features and learning to take full advantage of them. Whether your application is live mixing at a concert or in any sound reinforcement project, or for a mixdown to a master 2-track tape, the FX1644 should prove to be an ideal mixer-console.

We should note that if 16 channels of line and mic input are more than you require, Carvin has other models in the FX44 series that may be more suitable, such as their FX844 (8 channels worth of inputs), or their FX1244 (with 12 channels of mic and line inputs). On the other hand, if your project requires even more than 16 channels of mic and line inputs, Carvin also has available their Model FX2444 which, as you've probably guessed, features a full 24 input channels. db

VITAL STATISTICS

STATISTICS	MFR'S CLAIM	db MEASURED
Frequency Response	20 Hz to 20 kHz,1 dB	0.7 dB*
Total Harmonic Distortion		
+10 dBv,20 Hz-20 kHz	0.1%	0.14%
Typical 1 kHz THD	0.03%	0.022%
Output Noise		
Faders at minimum	–92 dBv	–93 dBv
Master faders nominal	–88 dBv	–91.4 dBv
Maximum Gain		
Mic to 2-Track Out	70 dB	Confirmed
Line to 2-Track Out	32 dB	32.5 dB
Effects Ret. to 2-Track	26 dB	Confirmed
Crosstalk, Adjacent Ch.	–60 dB @ 1 kHz	–63 dB
Peak Warning Level	+14 dBv	+13.5 dBv
Channel EQ Range		
Hi Band	12 dB @ 10 kHz	14 dB
Hi Mid Band	12 dB @ 2 kHz	Confirmed
Mid Band	12 dB @ 500 Hz	12.5 dB
Low Band	12 dB @ 100 Hz	Confirmed
Nominal Input Range		
Mic	–70 to –10 dBv	Confirmed
Line	–20 to +10 dBv	Confirmed
Maximum Input Range		
Mic	+10 dBv	+12 dBv
Line	+30 dBv	+32 dBv
Power Requirements	120/240 VAC, 50-60 Hz	Confirmed
Dimensions (WxHxD, inches)	N/A	37-3/4 X 7-3/8 X 26
Weight	70 lbs	Confirmed
Price	$1,699.00	

(* Measured with 9-band equalizer bypassed.)

HANDS ON

JOHN BARILLA

Aphex' New Dominator II

• The Dominator II is the newest addition to the arsenal of killer Aphex products. A company with a reputation for well- engineered devices, Aphex seems to have refined the concept of peak limiting to a fine degree with their new entry. The Dominator II, an advancement on the original Studio Dominator introduced circa 1985, is a true multi-band peak limiter with a couple of unique features for which Aphex has gained a patent. It is a flexible device with extremely interactive controls, hence, it is geared towards the professional recording engineer. Still, it is an intuitive device and sufficiently forgiving in its control settings that even a novice would be able to use it effectively.

There are two versions of the Dominator II: Model 720 which is geared towards recording, mastering and live sound operations and lists for $1350, and Model 723 which is customized for broadcast applications, has a few more features and costs a few more bucks. Both have servo balanced inputs and outputs making the Dominator II easy to interface with both balanced and unbalanced equipment. The unit reviewed in this hands-on report is a Model 720.

SO WHAT MAKES IT TICK?

A very commendable feature of all Aphex products is the depth and utility of their user manuals. Besides the expected quick set-up procedures, Aphex provides plenty of technical information to satisfy a curious soul on what makes the Dominator II tick; so let's explore this dimension before going on to a functional and sonic evaluation.

Aphex bills Dominator II as a "Precision Multiband Peak Limiter." It's a good idea to stop here and recall exactly what a limiter is, why a peak limiter is a unique type of limiter, why multiband limiting results in improved performance, and what "precision" means in the context of all the above.

Let's start with a textbook definition of a limiter. According to "The New Recording Studio Handbook" (Elar Publishing 1989), a limiter is "a compressor, whose output level remains constant (after threshold), regardless of its input level." An ideal limiter is clearly then a special type of compressor—not your usual garden variety which gently suppresses a signal in proportion to how far it has exceeded that imaginary line called the "threshold." Instead, a limiter should act more like a fence beyond which a sound should (theoretically) not travel — though it may move with impunity if it stays below the threshold.

In other words, the ideal limiter would do absolutely nothing until something crosses over the line, at which point it would prevent the sound from increasing in loudness. However, very few limiters (particularly the compressor/limiter combination type units) come remotely close to this theoretical ideal, and of those that do, many pay a high price in sonic quality to get the job done.

The problem is what engineers call "overshoot" and it seems to be most noticeable when a limiter is trying to deal with complex program material—such as a stereo mix. Some elements of the mix (such as a boomy bass drum) have stronger peaks than others. While the limiter is busy responding to the strong peak, it may overzealously suppress a weaker sound which follows closely on the heels of the stronger sound. Another factor in overshoot is if the limiter is not fast enough to respond to a peak, a piece of that sound may actually exceed the threshold for an instant. On the other hand, if the limiter is too fast, it may actually follow the individual waveforms too closely (rather than the overall envelope of the sound), thereby causing harmonic distortion. The result of this overshoot phenomenon is a sound that can be described as brittle, harsh and strangulated.

The classic solution to the problem (found usually in pricy mastering and broadcast limiters) is dividing the incoming signal into two or more frequency ranges and limiting each range separately. That way (in the context of a stereo mix for example), the effects of limiting upon a boomy bass drum would have some independence from the effect of limiting upon a vocal or a cymbal; hence, greater control. The limiter in this case does not have to be overburdened with work, since the work is divided among two or more frequency sensitive limiters whose output is later summed together.

While multiband limiting is quite an improvement over the common wideband method, it is not perfect. When the outputs of the frequency sensitive limiters are summed, they may piggyback in some unpredictable ways, resulting in overshoot. Tightening down the thresholds or lengthening the attack time may suppress some of the overshoot, but the result may be a loss of dynamics.

THE APHEX SOLUTION

To deal with this problem, Aphex follows the output of the limiter with a clipper. A clipper works differently than a limiter: it does not lower the volume of signal relative to a threshold. Rather, it simply lops off the tops of an incoming waveform, creating an absolute ceiling beyond which the amplitude of the waveform cannot transgress. It is as though the signal ran into a brick wall. The clipper, of course, is not without side effects either. If overdriven by too intense a signal, it will output audible distortion. If, however, it is driven moderately, it will add a certain harmonic warmth to the sound which is quite desirable.

As you can probably surmise by now, the real genius of Dominator's design lies not merely in the use of multiband limiting or in the use of a clipper, but rather the combination of these two elements in this interactive way. When controls are properly set (for a given program or track), the multiband limiter does its best to keep the sound from overshooting the threshold, but if some of the sound does elude the limiter, it gets caught and positively squashed by the clipper. These two lines of defense offer such tight control over the output that Aphex can legitimately claim the Dominator II is capable of zero overshoot; no signal can ever supersede the peak ceiling set by the user.

The unique feature of Dominator II is the *density* control which essentially varies the relationship of the limiter's threshold relative to the clipper's threshold. When the limiter's threshold is set low, most of the work in suppressing the signal is done by the limiter. When the limiter's threshold is set high relative to the clipper's threshold, a lot of the work is transferred to the clipper section. There are also lots of possibilities in between these two extreme settings, giving the user a variety of dynamics and harmonic warmth.

THE FRONT PANEL

From left to right on the front panel the user is first given an *input gain* control which allows adjustment of input gain over a 30 dB range (+ or –15dB beyond unity). Right next to it is a *process off/on* switch which allows for a true hardwire bypass of the unit. Moving along to the right are two more rotary pots which control the frequency selective gain to individual limiters (whose outputs are later summed). For each control there is a selective crossover switch: one for a low frequency crossover at either 100 Hz or 210 Hz, and another for a high frequency crossover at 1.7kHz or 3.4kHz. Both low and high frequency EQ is adjustable over a + or –5dB range.

The limit threshold is interactively controlled by several sources, two of which appear on the front panel, and one which is internal. The internal control is called an *ALT* (Automatic Limit Threshold). Based on reference information it receives from various sources, the *ALT* automatically adjusts the threshold of limiting for the individual limiters; it senses the summed output and lowers the threshold of limiting if it concludes that the output might drive the clipper stage (which is downstream) into audible distortion.

Another parameter adjusted by the *ALT* is the attack time of the limiters. A slower attack time will result in some portion of the transients eluding the limiter and barging into the clipper; a shorter attack time means the limiter is working harder to take care of the transients all by itself. The "density" control sets both attack time and limiter threshold through the *ALT*. As mentioned before, density is some complex measure of the limiter threshold as it relates to the strength of the waveforms going into the clipper. When the density control is full clockwise (at +5), the clipper has the predominant influence over the sound; at full counterclockwise (–5), the sound will predominantly be affected by the limiter, and the midpoints offer a blend of limiting and clipping. (It's understandable how Aphex could get a patent on this extremely useful feature).

Release time is also adjustable by a front panel control. Short release times from 150 ms to long release times out to seven seconds are possible. This setting also influences the proportion of time a signal spends in the clipper versus the limiter and the overall strength and dynamic of the signal. It is also an important tool useful in maximizing a given setting while avoiding distortion from either the clipper or limiter.

Finally, there are three peak ceiling controls: a *coarse* control that allows the output to be adjusted over a 24 dB range, a *fine* control which allows fine tuning of the output in 0. 2 dB steps, and a switch which shifts the entire gain structure by 10dB, giving a 34 dB range of control over the output. This allows for precision settings into a tape machine or other device, allowing one to really squeak the absolute maximum headroom out of any device fed by the Dominator II. The peak ceiling controls also set the threshold of clipping and work with the input gain settings to give varying degrees of drive to the clipper.

TAKING IT OUT FOR A RIDE

As you can see, Dominator II is a flexible, interactive, thoroughly professional device, but even if you don't understand all the concepts behind it, it's intuitively laid out so it can be easily and quickly tuned by using heuristic methods (such as "let's see what happens if I twiddle this knob!"). While intelligent use is most profitable, you can easily stumble onto a sweet setting.

I tried Dominator II on several sources. First, I tried it on drums. I was able to squash snare and kick drum into a very tight dynamic range without losing the impact of the transients (as would be the case with a normal wideband limiter). The individual limiter EQ controls were extremely useful in shaping the overall sound. I was able to increase the loudness of certain "hot" frequencies in the snare drum without risking a harsh sound. It seems that driving the clipper section hard with drums increases both the warmth and brilliance of the drum by adding some pleasant harmonic distortion. It reminded me of that certain magic delivered by the old Pultec tube equalizers—only without the noise. (The unit is extremely clean adding absolutely no noise when the input is run at unity gain setting.) A similar test was done while recording vocal tracks. The unit enabled me to record vocals with a very tight dynamic, allowing me to take advantage of every fraction of a dB of headroom on my multi-track recorder, and also making subsequent placement of those tracks in the mix a breeze—without the need for any compression.

The acid test, of course, is program material. I reprocessed some previously recorded stereo masters through Dominator II. It enabled me to play mastering engineer right in my studio. Set with long release times, Dominator II gave me rock-steady levels, even on programs with questionable left-right balance. The EQ also gave me the option of accentuating or not accentuating certain areas of the mix without adding any noise.

Dominator II is a serious professional tool with a unique flair. I highly recommend it. db

BRUCE BARTLETT AND JENNY BARTLETT

RECORDING TECHNIQUES

Mastering to DAT

• More and more, engineers and musicians are mastering to DAT instead of open-reel tape. "Sixty percent of the master tapes we receive are DATs," reports Ann Shepard, studio manager of Northeastern Digital Recording, a CD manufacturer in Shrewsbury, MA.

DAT mastering is becoming popular because it has several advantages over open reel. For starters, DAT sound quality is very high for the price—potentially as good as a compact disc. What you put in, you get out, virtually without any added hiss, distortion, or wow and flutter. When a DAT machine is used to record your stereo mixes, the DAT stereo master sounds as clean as your multi-track tapes.

What's more, you don't have to clean, align, and calibrate the DAT recorder before each session as you do with an open-reel recorder. You don't even need to record test tones at various frequencies because the frequency response is always flat—even when your tape is played on someone else's machine. All DAT tapes and recorders are compatible, so a tape made on one machine should play on any other.

There are marketing advantages, too. You can advertise that your studio offers digital mastering, and that "digital" buzzword attracts clients.

Some DAT recorders are small, light and easy to carry, which is a boon for on-location recording. Because a single DAT tape runs for two hours, you can usually record an entire live concert on a single DAT cassette. No more frantic tape swapping!

DRAWBACKS

A major drawback of DAT is that you can't re-sequence the tape by splicing it. Instead, you must copy from one DAT machine to another, changing the order of selections during the copying process.

Tight edits are impossible unless you use a digital-audio editor, which is expensive. This device, however, can be rented at some studios. One such editor is the Sonic System by Sonic Solutions. It is a hard-disk-based editing system with software that runs on a Macintosh II. You play your tape, record it on hard disk, use a mouse and computer screen to edit the selections, and record a new edited digital tape off the hard disk.

There are also other problems with mastering to DAT machines. In some units, the quality of the analog electronics and D/A converters is relatively poor (due to budget constraints), and the archival ability of DAT's tape is not clearly established.

Yet another drawback: you can't leader the tape to provide silent spaces between songs. On an open-reel machine you can splice in a few inches of leader tape—blank plastic or paper tape—just before a song starts and just after it ends. This eliminates studio noises that occurred just before or after each song, and puts a few seconds of silence between songs. But you can't splice leader into the tape.

There are ways around the leadering problem, and we'll cover them here, along with other tips on how to master to DAT effectively.

DAT CONTROLS

Before getting into mastering techniques, let's review the controls that must be set correctly in the mastering process.

Start ID: A Start ID is a subcode—a piece of information written on tape that is independent of the audio signal. The Start ID marks the beginning of each selection, and can be written manually or automatically. Automatic Start IDs are put on tape whenever enough signal is applied, after a silence of three or more seconds. You'll want to record manual Start IDs after recording all your mixes onto a digital audio tape.

Like other subcodes, Start IDs can be recorded or erased without altering the audio program. You can enter them during recording or playback. If the cassette's safety tab is set to prevent accidental erasure, you can't record or erase subcodes.

Sampling Frequency: Consumer DAT machines record only at a 48 kHz sampling frequency. A CD requires a sampling rate of 44.1 kHz. If you have a consumer DAT deck and want to do a digital-to-digital transfer to CD, the mastering engineer must use a sampling-rate converter. Most CD manufacturers can convert a 48K tape to 44.1K, but check with them first. The mastering engineer can avoid sampling-rate conversion if the DAT's analog outputs are used.

Which method provides better sound quality: using the DAT analog output signal, or staying digital and

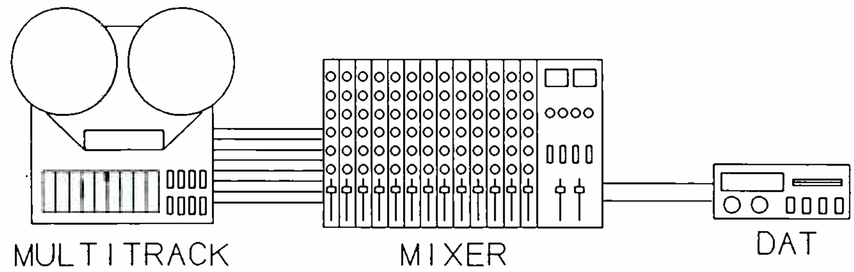


Figure 1. Mastering connections for mastering to DAT.

using a sampling-rate converter? According to Dr. Toby Mountain, a mastering engineer at Northeastern Digital Recording, the conversion process is not totally benign. It sometimes adds a little noise or modulates the volume. The conversion process is not easy, and requires a top-quality converter run by an experienced engineer for best results. For most jobs, Mountain obtains the audio signal from the analog output of the DAT machine. This signal is then transferred to a Sony 1630 digital processor for editing before the CD is cut. Mountain uses only a high-quality professional DAT machine to play clients' tapes.

If you have a professional DAT recorder, you can set the sampling frequency to 44.1, 48, or 32 kHz. A rate of 44.1 kHz is preferred if your tapes will be duplicated on compact disc, because no sample-rate conversion is necessary.

Emphasis: This active pre-emphasis control boosts high frequencies during recording and attenuates them during playback for lower noise.

Copy Inhibit: This prevents dubbing any copy-protected material. A DAT deck identifies data that has been recorded with a copy-inhibit flag in its subcode, and will not digitally copy that recording. Be sure copy inhibit is *off* if you want to duplicate your DAT master digitally!

CASSETTE OR CD DUPLICATION?

Once you set the DAT's controls properly, mastering on DAT is simple. *Figure 1* shows the connections.

First, decide whether you want your tape to be duplicated on compact disc. If so, put long silent spaces between tracks—say, 20 seconds—for easier editing by the CD mastering house.

Mountain strongly recommends that you have the CD mastering house edit your tape. Here, "edit" means to insert short silences between tracks. An editing job costs more than a one-to-one transfer, but is well worth it for professional results.

The copy-inhibit flag (one of the status bits in the digital bitstream) is picked up from the CD you're copying—it is not generated by your DAT recorder.

According to Mountain, editing charges for an album are typically only $400 to $500, a small fraction of the total project costs. The mastering engineer can re-sequence your tracks if necessary.

In contrast, cassette or vinyl LP duplicators seldom edit your tapes. They want a finished, edited and sequenced tape, ready for a simple transfer.

DAT MASTERING FOR CASSETTE OR LP DUPLICATION

If you plan to have your DAT tape duplicated only on cassette or LP, follow this procedure to record your mixes onto DAT:

1. First, record a 30-second 1 kHz tone at the head of the tape at –15 dB; record 20 seconds of silence, then hit *stop*. This tone will be used by the cassette duplicator to set levels and match channel levels.

NOTE: No tone is needed for a CD duplicator; instead, the peak program level is used for level setting.

2. On your multi-track tape, find the first song you want to appear on the master tape. Play the multi-track tape several times to practice the mix.

3. The DAT has a peak-reading recording-level meter. Set the DAT machine in record and pause modes, and set the recording level so the DAT meter peaks around –3 dB maximum. Unlike with an analog tape deck, "0" on the DAT meter is absolute maximum recording level. If your peaks are reaching 0, that level is too high.

4. Cue the multi-track tape to the beginning of the count-off for the song. (The count-off is the recording of a musician counting the tempo of the tune.)

5. Set the DAT tape counter to "00:00."

6. Play the multi-track tape; listen for the count-off and tap the *pause* button on the DAT recorder to start recording just after the count-off ends. For example, suppose the count-off goes "1, 2, 3, 4, 1, 2, (rest) (rest)," then the song begins. Hit the *pause* button in the rest just after the second "2," and record the mix.

Note that the *pause* button is a toggle: You tap it once to stop tape motion, and tap it again to release it and start tape motion.

You may need to plan the count-offs at your recording sessions to make sure there's plenty of silence before each song. This permits easier DAT cueing. If the song tempo is very fast, count off "1, 2, 3, 4, (rest) (rest) (rest) (rest)."

7. When the song is done or fades into silence, hit *pause* on the DAT machine to stop recording.

8. Now add a silent space between songs. If you want 3 seconds of silence, press *pause* to record, wait 3 seconds and press *stop*. This process is imprecise, so you may need to experiment.

9. Cue up the multi-track tape to the next song you want to mix, and go to step 2. Repeat this procedure

for all the songs you want to mix to DAT.

If you have to re-record a mix, follow this procedure:

1. Find and play the end of the last mix you recorded on DAT.

2. As soon as the song stops, hit *pause*.

3. Hit *record* and *play* while in pause mode.

4. For a 3-second space between songs, hit *pause*, wait 3 seconds, and hit *pause*.

5. Play the multi-track tape. Just after the count-off, hit *pause* on the DAT machine to release it and start recording the mix.

Using *pause* and *stop* sometimes makes slight noises on the DAT tape, but you'll have to live with them. Some DAT machines are quieter than others in this regard. (Note that many CD mastering plants will reject a master DAT tape if it has such noises. Later in this article we'll offer special tips on DAT mastering for CD duplication.)

If the multi-track tape has noises just before the music starts, you'll have to leader the multi-track tape. That is, mark the multi-track tape with a grease pencil at the playback head gap at the point where the music starts. Put the tape in a splicing block and cut it just to the right of the mark. This eliminates the noises. Then splice the leader tape to the multi-track tape at the beginning of the song. This method may be easier than trying to punch in and out of pause mode after a count-off.

In some studio productions, the multi-track recording process is eliminated in favor of mixing everything live to the DAT machine.

As an alternative, patch a noise gate between your mixer and the DAT machine. Set the noise-gate threshold so it removes noises before the music, but gates on reliably when the music starts. Once the music is in progress, bypass the noise gate or set its threshold to minimum (if this can be done inaudibly). Turn the gate back on just before the end of the tune.

```
                        TAPE LOG

Title: Escape to Air Island

Artist: Gog McGog

Engineer: Surb Teltrab

Producer: Bart Brucelet

Sampling rate: 44.1 kHz

All cuts are indexed.

SIDE A

00:00  Start of tape, tape fully rewound.

00:10  1 kHz tone at -15 dB, both channels, 30 sec.

1. Tender Lovin' Care         Start 00:58   Stop 03:47

2. Maybe We Need a New Dog    Start 03:51   Stop 07:02

3. Crunch Crunch              Start 07:06   Stop 10:21

SIDE B

4. But You Said               Start 10:25   Stop 14:01

5. Mooncake                   Start 14:04   Stop 17:34

6. Throbbing Pygmy            Start 17:38   Stop 19:11

Mastering notes: Peak program level is at 08:14.  Boost level

2 dB only on Crunch Crunch.  Equalize Mooncake +3 dB at 100 Hz.
```

Figure 2. A sample tape log.

DAT MASTERING FOR CD DUPLICATION

If you have a consumer DAT machine, it probably includes a copy-inhibit chip called SCMS. This chip lets you make a digital-to-digital recording of a commercial DAT or CD, but prevents regenerations from that copy. That is, you can't make subsequent digital copies of the copy.

Suppose you intend your DAT tape to be duplicated digitally on compact disc. The SCMS copy-protection scheme doesn't prevent this. Here's the story:

The copy-inhibit flag (one of the status bits in the digital bitstream) is picked up from the CD you're copying—it is not generated by your DAT recorder. It does not appear on recordings you make from your mixer output. This means any DAT recording you make of your own material can be duplicated digitally by a CD mastering house. Mastering to DAT for CD duplication is relatively simple because you let the CD mastering house do the editing between tracks. You'll leave a big space between tracks for easier editing. Here's what to do:

1. Don't bother recording a calibration tone. According to Mountain, this tone is unnecessary. The CD mastering engineer looks for the peak program level on your DAT tape to set the recording level, rather than using a tone. On your tape log, note the tape-counter time at the loudest part of your program.

2. Record about 20 seconds of silence at the head of the DAT tape.

3. On your multi-track tape, find the song you want to mix. Play the

multi-track tape several times to practice the mix.

4. Set the DAT machine in record and pause mode, play the multi-track tape and set the DAT recording level to peak around –3 dB maximum.

5. Cue the multi-track tape to the beginning of the count-off for the song.

6. Set the tape counter to "00:00."

7. Hit *pause* to start recording on the DAT machine; wait a few seconds for the DAT machine to get up to speed, and play the multi-track tape. You'll be recording the count-off as well as the song.

8. When the song is done or fades into silence, stop the multi-track machine and record about 20 seconds of silence on the DAT tape before starting the next tune.

RECORDING LIVE-TO-DAT IN THE STUDIO

In some studio productions, the multi-track recording process is eliminated in favor of mixing everything live to the DAT machine.

To do this, follow the same procedure as you did with the multi-track mixdown, except hit *pause* (to start recording) just after the live count-off. When the song is done (have a musician signal you), hit *pause*.

This method isn't always convenient. You might find it easier to record the session non-stop, then copy from one DAT to another, editing and re-sequencing as you go. If possible, copy digital-to-digital for best sound quality.

RECORDING LIVE-TO-DAT ON LOCATION

In this situation, a good procedure is to record the event non-stop. If you recorded a concert, you can edit the tape later to remove long applause and noises between songs. You do this by dubbing desired segments from one DAT to another.

DUPLICATING YOUR DAT MASTER

Once you have the DAT master tape completed, make a safety copy or run two DATs in parallel during the mastering session.

You'll probably want to duplicate your master tape on DAT, compact disc, LP, or cassette. For small runs of cassettes— say, for demo tapes—you can copy the DAT master several times on a stereo cassette deck. For larger runs, you'll need an outside duplicator.

If a compact disc will be cut from your DAT tape, the CD mastering engineer will first transfer your tape to a digital workstation or a Sony 1630 digital processor for editing.

Since DAT is a new technology, not all duplication houses have DAT machines to master from. Check to see whether the duplicating house can handle DAT tapes. If not, you might take your DAT machine to their plant for them to use, or pay them to rent one.

It's very important to document everything on your DAT tape.

First rewind the tape to the beginning, set the counter to 00:00, hit *play*, and note the start and stop times of each selection or track. You might want to manually punch in an index point (Start ID) at the beginning of each track.

Now you're ready to make a tape log describing the DAT master. Be sure to include this information:

Title of production, artist, engineer, producer, date.

Sampling rate and indexing status.

A note describing the calibration tone, if any.

Counter time of the loudest part of the program.

Start/stop times and title for each selection.

A note about which tracks go on side A and which on side B.

Notes on desired EQ and level changes (some mastering houses do not perform this function, and require a ready-to-cut tape).

Figure 2 shows a sample tape log for a DAT master tape intended for cassette duplication.

If a compact disc will be cut from your DAT tape, the CD mastering engineer will first transfer your tape to a digital workstation or a Sony 1630 digital processor for editing. At this stage, PQ subcodes are added, telling the CD player where to locate each track on the disc.

If your DAT machine has SMPTE time code (most don't), a compact disc can be mastered directly from your DAT tape.

1. Start with an edited, sequenced DAT master tape.

2. Stripe the DAT tape with SMPTE time code.

3. Play the DAT tape and note the SMPTE times when each selection starts and stops. When the CD is mastered from your tape, the mastering engineer will enter the SMPTE start/stop times as PQ subcodes in the CD laser mastering machine. Then the engineer will play your DAT tape and cut the CD.

As this article is written, only the Fostex D-20 DAT machine records SMPTE time code (*Figure 3*). It has SMPTE in and out connectors, and records the SMPTE time code in the DAT subcode area. *[Ed Note: Sony 7000 Series and JVC DS-DT900N DAT machines are now available and also have SMPTE in and out, it is recorded on all the above machines in a now-standard way.]* By following the suggestions given here, you should be able to create top-quality masters on DAT tape, and prepare your DAT masters for duplication. Your stereo master tapes will sound just as clean as your multi-track tapes, and that's always a thrill!

ACKNOWLEDGMENTS

Many thanks to Ann Shepard and Dr. Toby Mountain of Northeastern Digital Recording for providing CD duplication information. db

Getting the Right Sound at Dixieland Jazz Festivals

SHELLEY HERMAN

Since the advent of the first Sacramento Dixieland Jubilee sixteen years ago, there has been a resurgence of what is now called "traditional jazz."

THE OLD DIXIELAND JAZZ CLUBS HAVE disappeared, but the bands still play on our turntables and in our memories: The Firehouse Five Plus Two, Turk Murphy, Red Nichols and His Five Pennies, Rosie McHargue, The Yerba Buena Jazz Band (the band is gone, but the yerba buena still seems to be with us), Pete Daily and His Chicagoans, and Bob Crosby and the Bobcats. Suddenly, however, the music has been rediscovered, along with many fine new bands from all over the country.

The renaissance of this music has given birth to Dixieland Jazz festivals everywhere, and each one needs sound reinforcement. The festivals sometimes have only one venue, two or three bands and they last for one afternoon.

Some of the larger festivals may have as many as fifty venues with over one hundred bands, and last for three to five days. The venues may vary in size from fewer than one hundred to over two thousand. The bands have anywhere from three or four musicians to 1940's-style big bands with twenty to thirty musicians.

The resurgence of Dixieland Jazz offers an opportunity for those in the sound reinforcement business to capture some of this market and hear some pretty good music at the same time. The problem is, most of the sound reinforcement companies and their operators were raised on rock 'n roll and need to be educated in the "new" old music. To inject a personal note; if I do not like a particular type of music, I will not accept a job mixing or recording it.

I do not think I will do a good job and will not give the customer his money's worth. The same criteria must be used in selecting festival sound operators; if an operator puts the music down as something unacceptable, he or she doesn't belong on the gig.

PROBLEMS

As a lifelong Dixieland Jazz fan, I have gone to many festivals, and heard some pretty bad sound. At several recent multi-venue festivals, many of the most common problems were in evidence.

The most obvious was the lack in sound consistency from venue to venue. These inconsistencies were as a result, in small part, of differing room acoustics, but largely due to the varying levels of competence of the people who designed and installed the sound systems at each site, and the lack of competence of some of the people who operated the systems.

EQUIPMENT PROBLEMS

In several venues, the loudspeakers were on the stage essentially at ground (floor) level. Placement of loudspeakers at ground level is unsatisfactory. They must be raised above the heads of the audience. Unless this is done, the coverage will be severely compromised and the people in front will be deafened while the people in back will hear the sound filtered through all the bodies in front of them. This is basic sound engineering!

In another venue the sound system was good, but there were not enough microphones; only three plus the piano, insufficient for a seven-to-ten-piece band.

Many problems were caused by low quality sound systems that were just not good enough. Sometimes the systems looked like the supplier cleaned out his junkroom just to have enough equipment to cover every venue. I saw two Bose speakers being driven by half a Crown D60 with no Bose equalizer. It sounded terrible.

OPERATOR PROBLEMS

When a popular 1940s vocal group performed at a particular show, the operator had no idea what the group was supposed to sound like, because I heard no blend, and no increase in

Shelley A. Herman is an audio engineer who has spent over forty years in the entertainment technology business. He is the owner of Acoustronics Sound Co., Hollywood, CA, a full-service sound engineering organization. He is recognized as an expert in antique and period sound equipment and is on the advisory board of Orion Research, publishers of the Professional Audio "Blue Book." He is one of the founders, and executive director of SAVET, an organization devoted to Saving the Artifacts Vital to Entertainment Technology. Herman's professional affiliations include: Audio Engineering Society, The Hollywood Sapphire Group, Pacific Pioneer Broadcasters and The National Academy of Recording Arts and Sciences, where he has had a Grammy nomination. His personal interests include Barbershop and Dixieland music, and his collection of early microphones and mechanical sound equipment. In 1949, Herman was on the sound crew at Hollywood's Fairfax High School, and there found Dixieland Jazz. His friends were Dixieland fans, and listening to Frank Bull's "Strictly From Dixie" radio program was a daily ritual. Imagine how thrilled he was when there was a special assembly with Kid Ory as the star, and he was running the sound. Since then he has been there when Dixieland music was played, and in the past few years has attended and participated in many Dixieland Jazz festivals on the West Coast.

volume when solos were sung. The operator was using "reverb," which was inappropriate because the act is a re-creation of a 1940's sound when there was no artificial reverberation. I am old enough to remember the group in its heyday, and what I heard bore very little resemblance to what I heard in my youth.

During a concert featuring female vocalists from every band, the operators apparently did not understand the music being performed. They did not balance the instruments and vocalists correctly. They did not have the vocal mics equalized properly, so when a soloist changed mics, it sounded like a different person was singing. The operators were inattentive and often absent from the board when a performer needed a mic turned on or a level changed.

Another room had an acceptable sound system, but there was no operator, although one was badly needed. Apparently, the festival organizers thought they could just turn on the sound system and let the bands fend for themselves. As a matter of fact, unattended systems are one of the ongoing problems at every festival. Many musicians get quite irate about that problem.

At one outdoor venue, I entered the tent and the unattended sound system sounded all muffled with no volume. After walking around and looking at the loudspeakers, I found that the wooden weather protection covers were still on the front of the loudspeakers. I removed them to the cheers of the crowd (and the band). This was in the middle of the afternoon on the second day of the festival!

PHILOSOPHY

Sound operators must understand that Dixieland music is an acoustic instrument ensemble sound. Operators must also understand that unlike rock music, Dixieland Jazz uses sound reinforcement systems. The sound system is there to help people in the back of the room hear the sound being produced naturally onstage, and help balance the vocalist and the softer instruments such as piano, banjo, clarinet and tuba with the louder instruments, such as horns and drums. The overall sound pressure level (SPL) must be kept much lower than rock 'n roll, because the music is acoustic and not meant to be loud. The Dixieland Jazz audience is older and will complain vociferously if the sound is too loud.

Most of the basic precepts of sound reinforcement apply to these festivals just as they would anywhere else. Edsel Murphy's law is always present, so for any venue that has an audience of more than 100 people, a full-time operator must be on duty whenever an audience is in the room. To do otherwise is unprofessional, foolish, and in case of an emergency, dangerous. If someone were injured as a result of a sound operator leaving his station in a crowded room, the operator, supplier, or festival producer, might be held liable.

The festival producer must understand that it is his responsibility to ensure that provision be made for full-time operators for all venues over 100 seats. The producer must also realize that sound operators are professionals and should be treated with the same respect and have the same perks as the musicians playing in the festival.

BAND STRUCTURE

The "traditional Dixieland Jazz band" has a front line that includes trumpet, trombone, reeds and banjo and/or guitar. The second row includes the piano, drums and the bass, which may be electric, stand up, bass saxophone, tuba, or sousaphone. Some bands may have additional instruments. Often, there is a vocalist from the band or a singer that comes on just for selected numbers, or both.

MIC PLACEMENT

It is not necessary to use mics for the trumpet, trombone, or drums in a small venue. Mics will always be necessary for the vocalist(s), piano, clarinet and banjo. Unamplified bass instruments must be amplified with a mic in the correct place. When placing a mic for a sousaphone, use caution that the high position of the mic does not put it in front of an air conditioning register or in the near field of overhead loudspeakers. If there is an amplified bass, there is always enough leakage into the other mics to fill the room with bass. Festival bass amplifiers are normally rented, very old, and hum, spit and crackle, so beware! Some bands need a mic on a boom for the piano player who may sing or speak. The solo vocalist must have a handheld mic with a pop screen and long cable, because he often walks out into the audience. If a wireless mic can be supplied, all the better. In larger venues, or when simultaneously recording, it may be necessary to use mics for the trumpet, trombone and drums. Do so with caution and a lot of padding: those folks play loud!

Piano Mics. The piano sound should be picked up from the string side of the sounding board, because the piano is a percussion instrument and the percussive sound of the hammers hitting the strings is part of the desired sound. When the mic is placed on the side of the sounding board opposite the strings, the percussive sound is lost. If the mic is incorrectly placed, most mixers will turn the piano mic up louder in an attempt to get the percussive sound, unbalancing the music. Direct pickups such as the C-Ducer tend to lose the desired sound.

The best way to pick up sound on an upright piano is with a directional boundary mic (Crown PCC 160, PZM, or Shure or Audio Technica equivalent) placed underneath the keyboard, with the lower front cover board off of the piano. If a standard dynamic mic must be used, it should be on a short stand by the player's legs, or in front of his face.

A grand piano should have a boundary mic attached to the inside of the lid. The lid should be closed or on its short stick. If the lid is raised to the high stick position, the mic on the lid will pick up all the other instruments plus the sound coming back off the rear wall. If a regular mic is used, it belongs on a boom, inside the top of the piano. Again, never put a mic under or behind the sounding board.

MIXING DIXIE

Unlike rock 'n roll, with Dixieland music, the sound mixer is not a part of the creative process. His primary job is to ensure that the music produced on the stage is heard in the far reaches of a large room. Many mixers used to rock 'n roll will find it difficult to accept the concept of not being able to express a personal opinion of how the music should sound. The sound mixer must take the time to listen to some of this music played with no reinforcement so they can understand how the music is supposed to sound.

MUSICAL FORMAT

The usual format of a Dixieland song starts with an ensemble performance of the whole song. Each musician then takes a turn playing a solo or lead part, not everyone every time, and not always in the same order. The mixer must concentrate and watch the musicians and leader for cues. After all the solos, vocals, etc., the whole band usually goes back to the beginning of the song, plays it through, and ends with a big finish. It is up to the sound operator to keep the soloists "out front" during their solo, then return to a good ensemble mix. A mixer used to Bluegrass or old style country music will have no trouble with Dixieland.

MONITORS

In a medium to large venue, monitor loudspeakers are often necessary. The drummer must hear the piano, banjo, reeds and vocals. The piano player must hear the reeds, banjo and vocals. The clarinet and banjo players must hear themselves. The vocalist must hear the piano and himself. As there is usually only one monitor channel, it becomes obvious that the foldback monitors must contain a good mix of piano, vocal, reeds and vocals. Since there is no separate monitor or mixer, putting a monitor speaker near the console will make this easier.

SOUND DIRECTOR

It is often necessary to use more than one company to supply the necessary equipment for festival sound. Even smaller festivals may get their systems from several sources. When several sources and venues are involved, it is necessary to have a Sound Director for the festival. The Sound Director's job is to specify, coordinate and supervise the sound for the festival. The person selected should be an experienced professional sound engineer who fully understands the type of music to be played and is familiar with the various performers that will be using the facilities. It is important the festival producer not use the contractor or supplier as the Sound Director until he is confident the person is sufficiently knowledgeable and quality oriented. The Sound Director's duties are as follows:

1. **Specifications**

Develop specifications for each venue, describing:

a. minimum quantity of mics and stands required;
b. where loudspeakers, cables and consoles should be placed;
c. quantity and placement of stage monitors;
d. minimum and maximum SPL required in each section of the venue;
e. desired performance standards, including coverage, SPL, gain before feedback, and system noise;
f. cable runs, sight lines, talent entrance areas and other production parameters as related to sound;
g. coordinate with the lighting people to avoid dimmer noise in the sound system;
h. schedule set up crews to install and check out all sound systems before the festival, and remove and load out the equipment as each venue closes. It is important to schedule a sufficient quantity of personnel so injury to crew members does not occur;
i. schedule operators at every venue, ensuring that there is always an operator on duty. Sound operators should not mix more than five hours without a two hour break. The sound levels generated by these bands (80 to 100dB SPL) cause a temporary threshold shift, and can severely affect hearing and judgement;
j. designate a specific person to put the mics up every day, turn on the system, and confirm that it is working before the first band arrives;
k. designate a specific person to remove the mics to safe storage after the last band of each day, and check that the entire system is shut down safely for the night.

2. **Quality Control**

Ensure that the sound systems and operators provide consistent, high quality sound at each venue:

a. examine every sound system at each venue before the festival opening to certify that the supplier adheres to the specifications;
b. determine daily that no degradation of the system has occurred;
c. constantly circulate throughout the venues during the festival to make sure that all systems work properly, and that the operators perform correctly;
d. use instruments (SPL, RTA, etc.) to measure adherence to the standards.

3. **Operator Training**

The Sound Director, in concert with the festival producer, should establish a training program for sound operators. This program should include, but not be limited to:

a. well-written instructions on operation of the equipment;
b. examples of what traditional jazz music is supposed to sound like, including recordings of the correct and incorrect ways to mix traditional jazz;
c. establish a certificate or special badge to be awarded to those mixers (both professional and amateur) that have invested the time and effort to learn, and have demonstrated their proficiency.

For larger festivals, the Sound Director's job may be too much for one person, so there might be a Sound Director's Committee to share the load.

Using the Sound Director approach will notify sound system suppliers and operators that performance standards must be met, ensuring consistency and high quality sound at each venue. Conversely, the festival should budget enough money to properly outfit every venue with a first class sound system and operator, and not squeeze the suppliers so hard financially that they are unable to provide the quality and quantity of equipment necessary.

OPERATORS

Most festivals do not have the funds to pay professional operators standard wages for the twelve to sixteen hours a day over the three to five day period of the festival. Therefore, the festivals, especially the smaller ones, must rely on donated time and volunteers to operate some of the systems. A good training program for those volunteers, as outlined above, will raise the quality of sound and the enjoyment of the audience and the musicians. This training program will also help professional mixers, who may have never mixed anything but rock 'n roll, learn to handle traditional jazz.

It is obvious that all aspects of jazz festival sound reinforcement cannot be covered in a single article, but for a sound company that wants to expand its horizons, this is a field with potential. The festivals are usually repeated yearly, and the musicians and audience tell their friends where they heard the good sound. db

AD•VENTURES

BRIAN BATTLES

• A few years ago, there was no way for a small recording studio to compete head-to-head with the big-city facilities. They could afford the "name" talent, the hardware and the technological depth to bang out classy, big-budget masterpieces, while those of us operating small "neighborhood" studios had to rely mainly on inventive copywriting, gimmicky recording and low-priced musicians. A do-it-yourself commercial production company faced daunting obstacles and required intense determination to attract clientele.

HIGH-TECH IN YOUR HOUSE

We've come a long way in a short time. Yesterday's state-of-the-art professional iron is today's home gadget—and tomorrow's antique. Digital audio tape (DAT) is packaged as a consumer product, MIDI instruments are sold in department stores, and the prices of computers for editing, programming and management are plunging. Professional video production houses suffered a loss of business from private and industrial projects as the home-video boom of a decade ago unleashed affordable camcorders that replaced separate VCRs and cameras. Now low-priced video editing/effects consoles are sold in discount retail stores. This trend pushed some video houses out of business and forced others to look at different markets and seek out fancier ways of shooting, directing and applying digital graphics.

Audio recording is in a similar phase. That same decade ago, the capabilities of a New England Digital "Tapeless Studio" or Synclavier were far out of reach for most of us. Today's hard-disk-based audio recording/editing systems sell for not much more than yesterday's multitrack tape decks. MIDI gear interconnects to $500 desktop IBM PC clones. Even the software can be downloaded from telephone BBSs (bulletin-board systems). Musicians and aspiring producers can grab their VISA cards, fill a rack or two with gizmos, and solicit commercial production business from local establishments that would have sneered at seasoned local professionals a few years ago.

TIME FOR AN IMAGE

If you're serious about turning to the radio commercial-production industry as an area of specialization, the field is wide open. You do, however, have to study and prepare. Not just in becoming an expert at operating your studio, but at understanding marketing. Look around you and see what need you can fill.

Now that the holiday season is over, it's time to prepare for the spring/summer advertising season. Depending on where you live, local businesses adopt strategies to sell their products and services. While retail establishments everywhere try to take advantage of end-of-school, wedding, travel and income-tax time, there are regional differences. For instance, in the North, tanning salons, travel agents and hardware/home centers are gearing up for the time of year when people begin to spend time outdoors again. Everyone is looking forward to Valentine's Day, Mother's Day, Father's Day, June weddings and anniversaries, school graduation and the beginning of summer. Lawn and garden centers are busy, automobile dealers have to be creative to sell the no-longer-fresh models and entrepreneurs have to fight the post-holiday, pre-summer doldrums. Now's the time to let your creative flair stand out!

Face it, anybody can sell products just before Christmas. Bathing suits move themselves off the racks in June. School clothes fly out of the stores in August. But what can you offer to a slowed-down business to boost them up at this time of year?

Image advertising is always a safe bet. In an image campaign, it's not a particular offer that's promoted so much as the sponsor's name. Brand recognition and retention must be carefully seeded at times between major sales events. When it comes time to pick out a child's birthday gift, do you want him to remember Toys 'R Us, Kay Bee Toy and Hobby, The Toy Works or Kiddie Kastle? Coca-Cola doesn't beat your brains out with ads night and day because they expect you to drop everything and run out to buy a can of soda pop whenever you hear their ads. This is the kind of advertising designed to get the sponsor into the consumer's mind.

The techniques for image advertising lend themselves to a realm of creative ideas. Jingles, comedy skits, characters and signature effects can be manipulated like putty to build an unforgettable image.

DO YOUR HOMEWORK

Follow the types of spots your prospective client has been using in the recent past and take note of specific ways to improve on them. Listen to the radio and note who advertises on which stations and what days and times their commercials run most often. This will give you clues in determining the sponsor's target audience, so you'll know what approaches to try.

Follow your prospective client's competition, too. If you're trying to win an account producing commercials for a local fast-food joint, don't mimic McDonald's; on the other hand, analyze the elements of what McDonald's does, because if they use an idea consistently, it's because it works.

FIGHT FIRE WITH FIRE

Today's recording studio technology has reached a point where electronics and MIDI permit the smallest "basement" studio to generate productions largely indistinguishable from the big guns in New York City and Los Angeles. Now that you're armed with the necessary tools, play the role of an artist by manipulating your tools to dazzle the local business owner and land some lucrative contracts. db

db BUYER'S GUIDE

On the pages that follow, we present this issue's Buyer's Guide on Consoles and Mixers. 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.

ALLEN & HEATH USA
SR SERIES
Designed for live sound applications, they are also usable for 2 and 4 track recording applications. All feature 4 band EQ, 4 aux sends, and stereo and mono outputs. Available in 8, 12 and 16 input configurations with 4 sub group versions available in 16 and 24 input configurations.
Price: $1,800.00 to $4,400.00
SC SERIES
Modular series of consoles for live sound and 4 and 8 track recording applications. Available in 16, 24 and 32 input configurations with a variety of interchangeable modules including Fixed EQ, Sweep EQ, Stereo Line Input and Talkback/Oscillator Modules. General modules available in live sound/recording and sound contracting matrix versions.
Price: $4,800.00 to $9,370.00
SCEPTER SERIES
This is a 12 channel, 20 input rack mixer designed for contracting and live sound applications, with additional uses within the music and recording industries. Extensive use of internal jumpers make this mixer configurable for individual custom installation needs. Also available in a monitor version with 12 input channels and 10 outputs, 6 with sweep EQ and 1:1 mic splitter system.
Price: $2,700.00; Monitor: $2,800.00
VISION SERIES
Sound contracting mixers in 8, 12 and 16 input configuration. Two additional stereo returns provided. Rear mounted connectors, slanted front panel and optional Transformer inputs and outputs simplify installation, dynamic signal preset indicators and multiple sample points peak indicators.
Price: $1,600.00 to $2,600.00
SPECTRUM SERIES
Professional recording series for personal recording applications. Available in 16 Tape In/Output configuration with 40 inputs, 24 Tape In/Output configuration with 48 inputs, and 32 Tape In/Output configuration with 56 inputs. Automated mute system with self-contained mute sequencer. VU or Bar Graph metering.
Price: $8,900.00 to $14,900.00
SABER SERIES
These are 16 bus recording and sound reinforcement consoles available in 4 frame sizes with up to 24, 32, 40 or 48 input channel configurations. Available with 16 channel tape monitoring, 24 channel tape monitoring or standard and matrix live sound versions. Includes microprocessor controlled mute system.
Price: $16,500.00 to $33,000.00
SIGMA SERIES
In line recording consoles with dual input capabilities. Features include 8 aux buses, 4 band EQ plus variable frequency high pass filter, MIDI mute system, Solo in place plus PFL and Stereo AFL on all input channels. Frame size capacities are 32, 44 and 56 dual input channels. Producers desk and patch bay options available.
Price: $32,900.00 to $70,800.00

ALTEC LANSING CORPORATION
DYNACORD MCX SERIES 4800
They are available in 16, 24 or 32 input channels with 8 submixes down to 2 channels. Each input channel is provided with an XLR input for mic (balanced) line input jack, insert, tape inputs and outputs. The mic input is also provided with a switchable +48 v phantom voltage and -20 dB attenuator. The separate 19 inch power supply unit is included.
Price: MCX 16.8.2: $17,998.00
MCX 24.8.2: $21,998.00
MCX 32.8.2: $27,998.00
DYNACORD SERIES MCX 2400 MIXERS
They are available with 16 or 24 input channels and 4 subgroups. Each input channel features mic/line inputs, phantom power (+48 V), inserts, gain control, 3 band EQ (with parametric mids), 4 aux paths, panorama control, PFL, subgroup routing and channel fader, tape in/out tape-to-monitor, and headphones-preamp.
Price: MCX 16.4.2: $5,798.00
MCX 24.4.2: $$7,198.00
DYNACORD SERIES MCX 2400 MIXERS
These correspond exactly to the MCX 2400 series except without the subgroups and channel routing. This makes the MCX 16.2 and 24.2 mixers an inexpensive alternative for live and home recording.
Price: MCX 16.2: $3,798.00
MCX 24.2: $7,198.00

AMEK/TAC U.S. OPERATIONS

TAC B2

A new video post-production console available in three frame sizes, 8 to 28 mono or stereo inputs. Features include six discrete aux sends, individual routing to the four subgroups and stereo bus, and monitoring of 2-track machines. Available with AFV parallel and serial interfaces.

Price: starts at $3,950.00

TAC BULLET

Designed for production, post-production and multi-track recording, and available in three frame sizes offering from 10 to 30 inputs, four or eight subgroups. Consoles can be rack-mounting, built into larger control desks or used free-standing. Options include AFV parallel and serial interfaces.

Price: starts at $4,500.00

AMEK B2520

Developed from the G2520 multi-track console, the B2520 is designed for stereo broadcast production and features 24 buses, 24 to 48 track monitoring and eight auxiliary buses, plus eight stereo subgroups. Plasma metering and VCA digital grouping are standard.

Price: starts at $120,000.00

AMEK CLASSIC

Flexible broadcast and production console system, designed with stereo broadcasting as a principle objective. Numerous options include multi-track monitoring modules, metering (VU, PPM, plasma), dynamics, automation (VCA, moving fader).

Price: starts at $77,000.00

AMEK MOZART

A 32 bus multi-track and post-production console with AMEK Supertrue automation. Chassis sizes up to 80 inputs available. Options include the MZ15-RN in-line input module designed by Rupert Neve. Software dynamics available Spring 1991.

Price: starts at $108,000.00

MEDICI EQUALIZER

A self-powered 2U rack-mounting unit embodying two identical audio channels each with 4-band EQ, two variable filters and side-chain, designed by Rupert Neve. Unique features include "Warmth" and "Sheen" controls.

Price: $6,018.00

TAC SR6000

A mid-priced TV production, monitoring and sound reinforcement console with up to 34 independently mixable outputs and VCA grouping on both inputs and outputs.

Price: Not available

AMEK BROADCAST CONSOLE

A new broadcast console with many features for stereo production such as image width controls, previously only available on the larger Amek Classic.

Price: Not available

ARX SYSTEMS —See our ad on page 2

DI-6

This unit features six active DI Boxes and a six channel line mixer in a one rack unit package. Other features include multiple balanced outputs, individual ground lifts, Ch and master volume, clip LEDs and a headphone output.

Price: $549.00

AUDIO-TECHNICA U.S., Inc.

AT4462

A true stereo field mixer, this unit features two stereo and two center channel pannable inputs, stereo output, a pre-fader "cue" for each input, a unique MODU-COMM™ IFB circuit, and slate tone with internal slate mic.

Price: $1,395.00

AUDIO TEKNOLOGY INC.

ATI PARAGON

The 40 channel Paragon has 16 aux sends per channel, full four-band parametric EQ on channels and submasters, 16 sub masters, mix master, 8 dual stereo effects returns (16 more inputs), and four mono and four stereo matrix outputs. Consoles can be configured from 24 to 64 channels.

Price: From $55,550.00 to $130,000.00

64 CHANNEL PARAGON

The new console has all the standard features which include full dynamics processing (gates and compressors) on each channel, 8 dual stereo effects returns, 16 masters (16 mono or 8 stereo), 16 Aux sends per channel, 4 mono and 4 stereo matrix outputs, and 9 VCA masters.

AUDITRONICS, INC.

1900 IFB/MIX-MINUS SYSTEM

This unit features expansion beyond 8 outputs to 16, 24 or more; any number of IFB control locations in a single system, confidence module to provide continuous feeds, talkback module for two-way set-up conversation, and field upgrades for inputs, outputs and IFB controls.

Price: Available upon request

210 SERIES

Available in mainframe configurations of 6, 12, 18 and 24 input channels. Each console may be configured with any combination of mic and/or Stereo Line input modules. Each input console includes dual input selection, VCA fader control, dual stero outputs and other functions. Standard console features include all Output Amplifiers, Control Room Monitor, Headset Amplifier with Equalizer.

Price: Available upon request

310 SERIES

Developed for broadcast applications, and includes 4 aux send and returns, 4 or 8 output submastering, 4 group master faders, VCA fader control, cue, stereo solo, stereo monitoring, phantom mic power and a complete metering package.

Price: Available upon request

400 SERIES

Mono and stereo input modules available with optional equalizers. Each input module has hi/lo pass filters, 2 aux buses, VCA level control with P&G faders, patch insert switch, LED indicators for machine control status, stereo solo, cue, etc.

Price: Available upon request

900 SERIES

This design lets it rapidly access multiple high level stereo audio sources in a minimum of space with a low cost-per-input ratio. The console can directly communicate with the stereo audio x-points in your house audio/video router. Additional features are digital control, preview and program buses, memory storage of set-ups and input alpha-numeric read-outs.

Price: Available upon request

DESTINY 2000

Features four types of input modules, three standard stereo output buses, rack mounting for analog electronics, console expansion and upgrades are via plug-in PC cards. Each input has three pushbutton selectable Mic/Line inputs, separate Music/Voice buses for selective signal processing, built-in "smart" Telco with four simultaneous telephone inputs, optional two studio operation, digitally-controlled signal processors and tape machine control logic.

Price: Available upon request

CARVIN —See our ad on page 8

FX844 is a sound reinforcement/recording console with 8 in, 4 out capability. It has 250 watt/channel at 4 ohm amplification.

Price: $1,149.00 sold directly only.

FX1244 is as above but with 12 by 4 configuration.

Price: $1,499.00 sold direct only.

FX1644 is as above but has 16 by 4 configuration but no power amplification.

Price: $1,699.00 sold direct only.

FX2444 is as above but has 24 by 4 configuration.

Price: $2,299.00 sold direct only.

MX1688 is a recording consoles with 16 inputs and 8 outputs.

Price: $2,995.00 sold direct only.

MX2488 is similar to the MX1688 but offers 24 in by 8 outputs.

Price: $3,995.00 sold direct only.

CREST AUDIO INC.

GAMBLE SERIES EX MONITOR CONSOLE

Configured with 48 input channels with 16 mono mix buses that feed a 16 X16 matrix which feeds 16 output channels, each channel includes 4 band full parametric EQ, 24 dB per octave sweepable low cut filter, stereo aux level and pan, 20 segment LED meter and 8 programmable scene mutes. Has 20 segment LED meter.

Price: $74,900.00

GAMBLE HOUSE CONSOLE

The standard console consists of 56 input channels, 8 stereo subgroups, 8 stereo matrixes and a comprehensive internal patchbay. Features on each input channel include a 4 band full parametric EQ, 24 dB per octave sweepable low cut filter, 10 aux sends, 8 programmable scene mutes and a 20 segment LED meter.

Price: $74,900.00

DDA

DCM 224V

In line design, 56 channels/24 bus with additional 4 stereo subgroups and matrix section. Central control and automation of 24 I/O module switch functions, moving faders and console status storage in any of 30 snapshot memories.

Price: $183,000.00

AMR 24

A traditional 24 bus split design with separate group and monitor paths has 36 or 44 input frames with 24 or 32 track monitoring options. Has ability to monitor/mix two 24 track machines simultaneously with 4 band EQ and aux VCAs on moving-faders option.

Price: $71,500.00

DMR-12

Has split design, 56 channels/12 bus routing with switch access to 24 tracks. I/O modules feature mic and two line inputs, 112 line inputs available on mixdown. Identical 4 band EQ as input and output module. Midi-Mute automation.

Price: $45,500.00

DDA PROFILE

Has split concept/56 channel, 24 bus with 136 inputs available in mixdown, 4 band fully parametric EQ on all modules, 8 switch functions per channel. VCA Fader Automation with Dynamic control, 10 aux buses with direct out facility.

Price: $90,000.00

D-SERIES

The console can be configured for 16 or 24 track monitoring. Available in 1640 input versions, and has 4 band EQ with sweep mids, 8 aux sends, 8 group outs with either 16 or 24 track monitor returns (including corresponding 16 or 24 meters plus stereo l-r), all XLR inputs and outputs are electronically balanced, low noise and crosstalk.

Price: $27,425.00

Q-SERIES

The console is available in 16-40 input versions. Includes 4 band EQ with sweep mids, 8 aux/FB sends with 3 band EQ, 8 group outputs plus stereo Left and Right, and 8 x 4 matrix, 8 aux returns with 3 band EQ, mute group version also includes (8) mute groups and 20 segment LED meters on all inputs.

Price: $29,590.00

ELECTRO-VOICE

BK-842

Features mic and line inputs on each channel, 3-band EQ with sweepable midrange from 300 Hz to 5 kHz, 3 aux sends per channel, P.F.L., servo-balanced low-impedance outputs, stereo record outputs and separate tape/CD inputs with level controls, mechanical VU meters with integral peak LEDs and humbucking ground design to eliminate noise from external sources. Convertible for rack mount or tabletop operation, 60 mm faders.

Price: $1,200.00

BK1242

Features mic and line inputs on each channel, 3-band EQ with sweepable midrange from 300 Hz to 5 kHz, 3 aux sends per channel, P.F.L., servo-balanced low-impedance outputs, stereo record outputs and separate tape/CD inputs with level controls, mechanical VU meters with integral peak LEDs and humbucking ground, 100 mm faders.

Price: $1,450.00

BK-1642

Has mic and line inputs on each channel, 3-band EQ with sweepable midrange from 300Hz to 5kHz, 3 aux sends per channel, P.F.L., servo-balanced low-impedance outputs, stereo record outputs and separate tape/CD inputs with level controls, mechanical VU meters with integral peak LEDs and humbucking ground design, 100 mm faders.

Price: $1,750 00

BK-2442

It features mic and line inputs on each channel, 3-band EQ with sweepable midrange from 300Hz to 5kHz, 3 aux sends per channel, P.F.L., servo-balanced low-impedance outputs, stereo record outputs and separate tape/CD inputs with level controls, mechanical VU meters with integral peak LEDs and humbucking ground design to eliminate noise from external sources, and has 100 mm faders.

Price: $2,400.00

FOSTEX CORPORATION OF AMERICA —See our ad on page 29

812

A 12 X 8 X 2 broadcast or recording console, contains a MIDI Mute adaptor (Model 8200) which gives the board the ability to have MIDI Note information automate the muting on the board.

Price: $1,795.00

820

A 20 X 8 X 2 broadcast or recording console, contains a MIDI Mute adapter (Model 8201) which also permits the board to have MIDI Note information automate the muting on the board.

Price: $2,995.00

454

A 8 X 4 X 2 portable board with in-line monitoring, dual parametric mid and low EQ, shelving high EQ, input and output solo, 2 effects send.

Price: $995.00

FURMAN SOUND, INC.

MODELS MM-4A AND MM-8A

They feature four inputs, mono ((MM-4A) or stereo (MM-8A) outputs, pan pots on each MM-8A input, effects bus with send and return jacks, stereo auxiliary inputs with RCA jacks and level control, low cut buttons on each input are –3dB at 100 Hz; master fader; headphone amp with front panel jack and volume control. "-B" Models: Balanced ins with both phone and XLR connectors, mic/line switches; "-BP" Models: Same as "-B", plus 48 V phantom powering on all inputs and phantom power switch.

Prices: MM-4A: $339.00
MM-4AB: $379.00
MM-4ABP: $415.00
MM-8A: $399.00
MM-8AB: $445.00
MM-8ABP: $475.00

MODELS DJM-8 AND DJM-308

They have eight stereo inputs (2 phono, 6 line) feed four input faders; bypass-able crossfader with "Beat Sync" LEDs; talkover mic input with 3-band EQ; Dim/Mute button; Cue button/LED on each fader; headphones can receive stereo cue/program blend or "split mono"; master and two aux zone faders; patch points and switch for external processor; stereo VU meters; output 3-band EQ with proprietary subharmonic synthesizer; extra outputs for tape dubs (with or without talkover); lighting system sync and mono subwoofer. Model DJM-308 has built-in stereo power amp with 150 watts/channel.

Prices: DJM-8: $749.00
DJM-308: $1,199.00

INDUSTRIAL RESEARCH PRODUCTS, A KNOWLES DIVISION

DE-4013

Modular inputs, Level-Matic, phantom power, remote status, buffered preamps and aux inputs, 18 lbs.

Price: $2,955.00

DE-4018

As above, has EQ.

Price: $ 3,650.00

DE-4014

4 inputs, phantom power, 6 lbs.

Price: $1,545.00

DE-4016

As DE-4014 above, has remote.

Price: $1,950.00

DE-4024

As DE-4014 above with remote and phantom power.

Price: $2,112.00

INTERNATIONAL MUSIC COMPANY

MINIMIX

This is a 16 channel rack mountable console with mic and line inputs, two band EQ, 4 aux sends (2 pre and 2 post fade—internally switchable), PFL, pan, mute, peak level indicators and high quality 100mm faders.

Price: Available upon request

MULTIMIX

Has 16 channels, rack-mountable, and can be used in a 16:4:2:1, 12:4:2:1, or 16:2:1 configuration, featuring 16 mic/line inputs, 4 subgroups, 3-band EQ with defeat, direct outputs on all channels, 2 aux sends, 100mm faders, and 48 V phantom power, for recording, broadcast or sound reinforcement.

Price: Available upon request

OMNIMIX

Has a 20:8:8:2 rack mountable console featuring 12 mic, line and tape inputs, each with phantom power, 20 dB pad mic input, 3 band EQ with sweepable mid and defeat switch, 4 aux sends, 8 sub-groups, muting, PFL, peak indicators and 100mm Alps faders, 4 stereo inputs each with 2 band EQ, gain, pan, PFL, muting and 8 aux returns, optional meter bridge available.

Price: Available upon request.

SOUNDMIX

This is a semi modular console available as a 24:4:2:1 or 16:4:2:1. Designed for recording, broadcast or sound reinforcement, features include: 16 or 24 mic/line inputs, 4 subgroups, 4 aux sends, 4 aux returns, 4 band EQ, 100mm faders (Alps or Noble), 48 volt phantom power, direct outputs on all channels and subgroup;s, insert points on all channels, sub-groups, and masters, PFL system and 12 segment LED display on sub-groups and masters.

Price: Available upon request

DATUM

A modular console available in three types—studio recording, sound reinforcement, and stage monitoring, with four frame sizes for each type. Three choices of input and output modules, 4 or 8 bus, up to 48 inputs on most configurations sweepable EQ, 8 aux sends, stereo tape inputs, EQ and aux send on all tape monitors, and FX return module provides two mono returns and one stereo.

Price: Available on request

CONCEPT 200 AND 400 SERIES

They are sound reinforcement and recording consoles. Available from 16:8:2 with an 8:8 matrix to 56:24:48:2, these consoles feature patch bays, Alps or P & G faders, 3-way speaker select, 2 types of EQ available (6-band with sweep filter or 4-band sweepable), 8 or 12 aux sends using dual concentric or 4/6 sends switchable to 8/12, programmable mutes, A & B 2-track returns, DC controlled subgroups.

Price: Available upon request

LEXICON, INC.

OPUS DIGITAL AUDIO PRODUCTION SYSTEM

Has standard configuration of 12 channels of digital mixing, each with 4 aux sends, patch points, phase invert, mute and solo functions, and 190 track-minutes of audio storage. Also has record capacity of up to 790 track-minutes at 48khz sampling rate, 99 tracks, simultaneous multi-format audio (analog and digital) I/O, and true multi-track editing. Primary features of the OPUS Automix automation package include full recall of all console knobs and switches, absolute or trim updating, and joining with auto-takover and auto-release.

Price: Starts at $170,000.00

MIDAS

XLE

Available from 24-40 input versions, a 16 input extender frame is also available, designed for use on front of house or monitors, the XL3 has 18 sends to 18 outputs, 8 mute groups controlling inputs as well as outputs, 8 VCA masters, as well as 2 VCA Grand Masters for overall control of 16 outputs.

Price: $79,900.00

NEVE

VR series console is an automated "flying fader" multi-track recording console for the music, video post and film industries, available in 36, 48, 60 or 72 inputs, including the Formant Spectrum Equalizer, mic/dynamics unit and 8 mono/4 stereo aux Additional benefits include a centrally positioned monitor path status indication to enable rapid console status checks and choice of metering options.

Price: Available upon request

8232 console for TV production, post-production and multi-track recording has 32 mic/line input channels with 24 mixing buses and optional stereo reverb returns. Each channel features the Formant Spectrum Equalizer, 4 mono aux sends and 1 stereo cue send, and is "flying-fader automated."

Price:Available upon request

VPR series are multi-track systems for video post and film recording with total storage and recall of all console settings. Dolby matrix monitoring on switchable 4 or 8 track buses up to 48 tracks, 4 or 8 track to stereo or mono TV feeds, separate feeds for music and effects, inset switching for Dolby DS4 matrix with solo interrupt.

Price: Available upon request

6DTC-2 DIGITAL TRANSFER CONSOLE

New equalization consists of two pairs of selections for LF and HF, with a full range of peak/shelf responses in both sections, mid-range band coverage between 1000 Hz and 3150 Hz with up to seven different Q values. A/B store systems facilitate comparisons between different EQ settings, new Neve A/D and D/A converters include anti-aliasing and anti-imaging filters resulting in signal-to-noise figures exceeding 102 dB.

Price: Not available

N.I.H. LABS

PM160 is a16 input (balanced and unbalanced) reinforcement console, two internal stereo power amplifiers—400W x 2 at 4 ohms. Built in 99 program digital effects processor, graphic equalizer for stereo and mono, talkback mic, 12-point LED meters, dimensions 32 ¼ W X 21 ¾ D X 6 ½ H, 70 lbs.

Price: $2,195.00

PM80 is an 8 input (balanced and unbalanced) reinforcement console, two internal stereo power amps—180 W x 2 at 4 ohms, built in digital delay, graphic EQ for stereo output, 3 EQ positions for each input channel, 12-point LED meters, dimensions 20 ½ W X 21 ¾ D X 6 ½ H, 35 lbs.

Price: $995.00

NUMARK ELECTRONICS CORPORATION

DM1075

Features six stereo inputs and accommodates 3 turntables and 3 line level sources. Other features include an assignable crossfader; pushbutton-activated input selection provides trouble-free cueing and source monitoring; master level controls; stereo/mono switch; record outputs and low-noise; and wide band frequency performance.

Price: $190.00

DM1175

Accommodates 3 turntables, 5 stereo line sources and 2 mics; separate 6 band equalizers for the left and right channels and assignable crossfader.

Price: $280.00

PANASONIC-RAMSA

WR-5852

A 52/8/8/2 sound reinforcement console with 52 inputs, mono,monitor and submix, 8 groups, 8 aux, l-r output, 4-band EQ, 293 lbs.

Price: $36,300.00

WR-C900

A 32 input theater sound reinforcement console with true l-c-r panning 4-band seep EQ, 4 groups, 4 aux, 191 lbs.

Price: $36,000.00

WR-584OF

A 40 input stage monitor console, has 18 aux, 4-band sweep EQ, and monitor, mono, submix inputs, 293 lbs.

Price: $38,500.00

WR-8616

A 32-input compact fully modular production console, has 16 + 16 mono, stereo, tape inputs, 4 groups, 4 aux, l-r output, 3-band sweep EQ, 114 lbs.

Price: $12,000.00

WR-T820B

A 20 + 20-input recording console, has mono, tape and line inputs, 8 groups,4 aux and l-r output and can have up to 48 separate inputs for mixdown, 115 lbs.

Price:$8,950.00

WR-S216

A sound-reinforcement mixer with 16 inputs of mono, line, stereo, 3-band mid-sweep EQ, and 3 groups and 3 aux outs, 47 lbs.

Price: $2,850.00

WR-33

An 8-input sound-reinforcement console, user friendly with 2 groups and 2 aux outputs, 2-band EQ, rack mount option, 22 lbs.

Price: $1,290.00

WR-M10A

A multi-purpose rack mixer with 4 + 2 inputs of mono, stereo, phono, 2 groups and 1 aux out, 2-band EQ, built-in compression and auto-mute, 15 lbs.

Price: $900.00

PEAVEY ELECTRONICS CORPORATION —See our ad on Cover III

Mark VIII sound reinforcement console is available in 24 and 36 channels. It also has the following features: 8 submasters, 8 aux sends, 4-band sweepable EQ, 8 outputs with LED output level indication, 3-band EQ on aux returns, stereo L/R outputs, totally modular design.

Price: $9,999.99

SR 2400 SOUND REINFORCEMENT MIXING CONSOLE

Has 24 input channels, four submasters, six aux sends, 3 band EQ with sweepable mid on each channel, balanced and unbalanced inputs, stereo mix capability, 100mm long-throw faders, Master (sum of L and R) output.

Price: $2,599.99

SRC 1600 SOUND REINFORCEMENT MIXING CONSOLE

Has 16 input channels, four submasters, six aux sends, 3 band EQ with sweepable mid on each channel, balanced and unbalanced inputs, stereo aux capability, 100mm long-throw faders, Master (sum of L and R) output.

Price: $1,999.99

D-IIIB SOUND REINFORCEMENT MIXING CONSOLE

Available in 12 and 16 input versions with two submasters, six aux sends, 3 band EQ with sweepable mid on each channel, balanced and unbalanced inputs, stereo mix capability, 100mm long-throw faders, Master (mono) output, transformer balanced outputs on L, R, Aux A, Aux B.

Prices: 12 Channel: $1,399.99

16 Channel: $1,699.99

UNITY 1000 SOUND REINFORCEMENT MIXING CONSOLE

Available in 8 and 12 channel versions with low Z and high Z inputs, 3 band EQ on each channel, monitor send, post effects send, stereo effects return, 60mm faders, tape out and tape in with level controls (RCA jacks).

Prices: 8 Channel: $449.00

12 Channel: $599.00

UNITY 2000 SOUND REINFORCEMENT MIXING CONSOLE

Has 12 and 16 channel versions, low Z and high Z inputs, 3 band EQ on each channel, two monitor sends, two post effects send, stereo effects return, 60mm faders, tape out and tape in with level controls (RCA jacks).

Price: 12 Channel: $699.99

16 Channel: $899.99

PEAVEY AUDIO MEDIA RESEARCH 1242 RECORDING CONSOLE

Has 12 input channels, 3 band EQ with sweepable mid each channel, two aux effects sends, Mic/line and Tape/line switches, stereo monitor output, one mono effects return, one stereo effects return, eight input monitor mix section.

Price: $1,399.99

PEAVEY AUDIO MEDIA RESEARCH PRODUCTION SERIES RECORDING CONSOLES

Available in 24, 32 and 36 input channel versions, -4dBm, split configuration, four band sweepable EQ, 8 sends, 8, 16 or 24 bus configurations, two monitors per bus (i.e. 24 bus can monitor 48 tracks w/2 band EQ), eight assignable effects returns. VU versions optional.

Prices: 24x8: $899.99
32x16: $12,999.99
36x24: $15,999.99

ROSS SYSTEMS

RX SERIES

These mixing consoles are available in 8, 12 or 16 channel, and each unit can be expanded by 4 channels with the EX4 expander. Each unit features balanced mic inputs, line inputs, channel gain trim with LED headroom indicator, 3 band EQ, monitor bus and effects loop, channel inserts, assignable bar graphs for master out, XLR and 1/4-inch output jacks. Designed for on stage mixing of drums or keyboards.

Price: $599.95 to $999.95

SOLID STATE LOGIC

SL 4000 G series master studio system is a multi-track music recording and mixing system available with 24 to 72 input/output channel, full dynamics processing and G Series EQ. Also G Series Studio Computer and Total Recall.

Price: Available upon request

SL 5000 M series audio production system is a stereo broadcast, on-air, continuity and post-production system based on a modular cassette structure. Up to 96 channels, accepts G Series Studio Computer, Total Recall and Instant Reset.

Price: On request.

SL 5000 M series film post-production system is a modular, cassette based film console system, available in configurations for ADR/Foley, premixing, music scoring, final mix, video post-production, and multi-operator film dubbing. Up to 96 channels, optional moving fader automation, accepts G Series Studio Computer, Total Recall and Instant Reset.

Price: Available upon request

SL 6000 G series stereo video system is a stereo music, video and teleproduction system with 24 to 72 input/output channels, full dynamics processing and G Series EQ. Also G Series Studio Computer and Total Recall automation.

Price: Available upon request

SAMSON TECHNOLOGIES CORPORATION

PC MIDI SERIES

The "in-line" concept enables many inputs with EQ and Aux to be available during mixdown. MIDI muting of all inputs, monitors aux masters and optional effects return modules can be programmed utilizing the onboard muting computer or be written as individual note numbers to a MIDI sequencer, comes standard to meter 24 bus returns to the console.

Price: From $8,300.00 to $16,200.00

QUARTZ SERIES

A 24 bus in-line console with computer mute automation of channels, tape monitors and aux masters. Up to 100 mute patches may be stored internally and triggered from a MIDI sequencer enabling mute playbacks to be locked to timecode. Available in 32 or 48 channels, both with TT patchbay as standard, features include a four-band sweepable EQ and six auxiliary sends, options include effects return modules, Penny and Giles faders, stereo input modules and additional tie lines.

Price: From $43,000.00 to $55,000.00

IN-LINE SERIES

The 48 and 36 input versions are provided with 32 buses, crosstalk between channels or monitors is typically better than –85 dB @ 10 kHz, dual line inputs are provided in addition to a mic input on each channel, four band fully parametric EQ may be assigned to or split between the monitor path and/or the channel path, eight auxiliaries are provided on each channel, a comprehensive TT jackfield is supplied as standard.

Price: From $64,000.00 to $80,000.00

MRX SERIES

A traditional split console design available in 24, 32 and 40, 26 with TT Patchbay and 34 with TT patchbay frame sizes. Full 16 track monitoring is provided with full metering, input channel features a four band EQ, 6 aux sends and illuminated mute and solo buttons, fader reverse on the lower 8 eight monitor returns along with 3 band EQ allows for more inputs on re-mix.

Price: From $10,000.00 to $18,000.00

SHURE BROTHERS, INC. —See our ad on Cover 1V

FP51

This is a compact, portable gated memory compressor combined with a four-input, one-output mic mixer, designed for applications in broadcasting, recording and sound reinforcement, the FP51 integrates all the features of professional compressors and mixers in a single unit, dimensions: 3 1/8 X 12 7/32 X 9 1/32, weight: 6 lbs., 1 oz.

Price: $965.00

FP42

A four-input, two-output, compact, self-contained stereo mixer for applications in broadcasting, recording and sound reinforcement, features wide, flat frequency response, low distortion and high output level, durable construction provides operation under all temperature and humidity conditions, dimensions: 3 1/8 X 12 7/32 X 9 1/16, weight: 6 lbs., 8 oz.

Price: $990.00

FP32

A portable stereo mixer for electronic news gathering, electronic field production or film production. This very small, three-input, two-output mixer provides all the necessary features for professional broadcast remotes. dimensions: 2 5/16 X 7 1/4 X 6, weight: 2.5 lbs.

Price: $1,395.00

FP31

This is a portable, three-input, two-output electronic news gathering, electronic field production, or film production mixer that provides the quality and features needed for remotes. Wide, flat response, extremely low distortion and up to +18 dBm output level provide studio quality performance in a portable mixer, dimensions: 1 7/8 X 6 5/16 X 5 5/16, weight: 2.2 lbs.

Price: $1,025.00

M267

A mic mixer-remote amplifier specifically designed for professional applications. The performance and features of this console make it an ideal choice for studio, remote, or sound reinforcement use, and as an add-on mixer for expanding existing facilities. Also ideally suited for use with audio and video tape recorders to provide multiple mic inputs, dimensions: 2 $\frac{23}{32}$ X 12 $\frac{5}{32}$ x 9, weight: 5 lbs., 2 oz.

Price: $520.00

M268

This is a portable mixer designed for use with sound reinforcement, tape recording and A/V systems. Its operational characteristics, compact size and functional capabilities make it a fine choice as a primary or add-on mixer in any sound system, dimensions 2 $\frac{23}{32}$ X 12 $\frac{5}{32}$ X 9, weight: 4 lbs., 1 oz.

Price: $315.00

SONY COMMUNICATIONS PRODUCTS COMPANY, PROFESSIONAL AUDIO DIVISION

MXP-3056 VF

This audio recording/remixing console is intended for use in recording studios. It has 56 channels which allows for interfacing with the Sony PCM-3348 digital audio multi-track recorder. Each input/output module features modular equalizers and mic/line pre-amplifiers. The Audio Group Master (AGM) function allows for audio grouping on the ACN bus and conventional in-line operation.

Price: up to $100,000.00 depending on configuration.

MXP-3036 VF

Designed with a vacuum fluorescent (VF) light meter that displays various selectable scales including VU, BBC Peak, Din Peak, Nordic Peak and a d.c. scale. This d.c. scale indicates fader position in the automated version of the MXP-3036 VF. The automated version includes Version 2.0 software and optional wild faders that permit a user to increase the number of effects in a mix.

Price: up to $110,000.00 depending on configuration.

MXP-3000

This series is a modular 24-bus console primarily intended for music recording applications. Available in 20-, 36- and 56- input frame sizes with many options including five types of equalizers, four input configurations, and automation choices.

Price: $55,000.00 to $200,000.00 depending on size and configuration.

MXP-2900

The audio consoles is a modular audio-for-video system available from 8 to 36 inputs in four frame sizes. Extensive video interface options are available, mono and stereo modules, built-in compressor/limiters, and extensive routing and communication capabilities.

Price: $15,000.00 to $45,000.00 depending on size and configuration.

MXP-290

An 8-input mixer designed primarily to be used in conjunction with a video editor in a post-production environment, it offers microphone, balanced, and unbalanced inputs on each input channel and balanced outputs, an internal audio edit preview function also included.

Price: $3,819.00

MXP210

This has all of the features of the MXP-290, except those related to video editor interface, the MXP-210, like the MXP-290, offers excellent audio performance in a rack-mountable mixer.

Price: $1,992.00

MX-P61VU

A 12-channel audio mixer. It is equipped with 12 mic/line inputs and 4 line outputs. Features include built-in 1 kHz test tone for precise level setting, high-cut and low-cut filters for convenient bandwidth limiting and a.c./d.c. operation.

Price: $10,675.00

SOUNDCRAFT

MODEL 3200

Up to 36 inputs with full 32 bus routing and advanced EQ circuit; identical facilities for EQ on the inputs and group modules; individual monitor modules with global multi-track A/B switching and tape machine record/ready switches.

Price: Starting at $80,000.00

DELTA CONSOLES

Delta, Delta 8-mono, stereo input modules with various equalizer configurations, 6 aux sends, 4 to 8 bus available with 8, 16 track tape monitor returns. Delta Monitor-10 mono and 1 stereo monitor mix for stage monitor and production foldback. Up to forty input frames available.

Price: From $3,900.00 to $20,000.00

VENUE

Eight bus live; 16, 24, 32 or 40 channel available; 8 plus 2 receive matrix section with meterbridge and center master section standard in larger frames. Inputs include 6 aux sends, 4 band EQ, phase reverse, patented padless mic preamp, 4 mute groups.

Price: $11,000.00 to $24,000.00

SAPPHYRE

Multi-track/production console with in-line monitoring and film/post features. Channels have 4 band EQ, integral noise gate for signal control. Signal routing includes 6 aux sends, track buses available as additional aux sends. Mono and Stereo inputs available, full metering standard.

SAC CONSOLES

SAC100, SAC200, SAC2000

These are for on-air/production, mono, stereo inputs available with or without EQ; aux, sub group buses, external logic controls. Models vary in signal routing complexity but SAC range has similar electronic interface and specifications.

Price: $5,000.00 to $20,000.00

6000

A 16 or 24 bus recording console. Each input has 6 independent sends and 4 band EQ with 2 sweepable mids. Features include PFL and true solo in place, low crosstalk routing matrix, silent electronic muting. Available in 16-56 input versions. Automation package optional.

Price: From $12,000.00 to $52,000.00

SPIRIT CONSOLES

Live consoles—8, 16, 24 inputs with 3 band EQ, two sweep controls, high pass filter, 4 aux sends, separate stereo/mono bus assignment allow various routing possibilities, studio consoles—16, 24 inputs with in-line monitoring for up to 56 inputs with EQ on all paths.

Price: From $1,295.00 to $5,650.00

DELTA AVE

Audio production, audio-follow-video capability, up to 16 inputs, mono or stereo, external control via parallel, GPI or serial, supports ESAM 1, ESAM 2, GVG 100, AMX 100, other protocols both in eavesdrop and reply mode.

Price: From $6,000.00 to $20,000.00

STUDER REVOX AMERICA, INC.

A779

A compact mixing console with six mono and six stereo inputs with EQ, 1 aux bus, and stereo master output, ideally suited for video edit suites, mobile and remote applications.

Price: $3,995.00

961/962

This is for remote recording and broadcast production applications, features include up to 16 inputs, 4 master outputs, 2 aux inputs, an optional editor interface is available, 3-band EQ on each input and compressor/limiter on outputs, metering options are available.

Price: $12,500.00 and up

963

A general recording console suitable for music and broadcast applications, available with 16 to 56 inputs, up to 8 subgroups, 2-4 masters, 3-band EQ on each input, compressor limiter on outputs, alternate input modules, metering, monitor mixers, and machine remotes are available.

Price: $35,000.00 and up

900

This series of consoles can be configured for post production, on-air TV broadcasts, multi-track recording, or other production tasks. Features include 12 to 60 inputs with 3- or 4- band EQ, mono or stereo inputs and stereo master. Moving-fader automation is available. Outputs have compressors/limiters. Consoles can be customized.

Price: $50,000.00 and up

SUNN/FENDER

MX 4212

It has 12 input channels, left, right, main and monitor outputs, phantom power, 3 band EQ per input, monitor, effects and aux buses, cue bus, extensive patching capabilities, spring reverb, separate in/out patching jacks per input.

Price: $1,499.99

MX 4216

Features 16 input channels, left, right, main and monitor outputs, phantom power, 3 band EQ per input, monitor, effects and aux buses, cue bus, extensive patching capabilities, spring reverb, and separate in/out patching jacks per input.

Price: $1,799.99

4116

Features 16 inputs, left, right, main and monitor outputs, VU meters, phantom power, 2 buses—monitor and effects, 3 band EQ per input, and headphone bus.

Price: $1,249.99

RMX 4110

Has 10 inputs, left, right and mono outputs, 3 band EQ, 3 buses, 3 stereo returns, RCA tape inputs, and 6 rack spaces.

Price: $899.99

PSM-8

It has 8 line level inputs, left and right outputs, 2 band EQ per input, effects bus, and headphone bus.

Price: $419.99

SR 6520

It delivers 520 watts at 2 ohms and 350 watts at 4 ohms, features include 2 band EQ per input 9 band graphic EQ, effects and monitor buses, high and low impedance inputs; spring reverb, and carpet cabinet, SR 6520 has 6 inputs, SR 8520 has 8 inputs.

Price: SR 6520—$789.99; SR 8520—$879.99

PX 2108

Has 8 input channels, left, right, main and monitor outputs, 2 X 250 watts into 4 ohms, two 10 band graphic EQs, 3 band EQ per input, extensive patching capabilities, spring reverb, monitor, effects and aux buses, and phantom power.

Price: $1,649.99

PX 2112

It has 12 input channels; left, right, main and monitor outputs; 2 X 250 watts @ 4 ohms; two 10 band graphic EQs; 3 band EQ per input; extensive patching capabilities; spring reverb; monitor, effects and aux buses; and phantom power.

Price: $1,899.99

TASCAM, TEAC CORPORATION OF AMERICA —See our ad on page 14-17

M-700

It is configured with a 40 channel I/O layout, 32 groups and quad dual stereo mix buses, 12 aux sends—2 stereo and 8 mono, 4 band equalization with switchable peak/shelving high and low bands, full function high and lo-mid parametric plus variable hi-pass filter.

Price: $69,999.00

M-600

Features 32 input positions, 16 group outs, mono modules offering 4 band EQ with sweep high and low mids and shelving high and low, stereo modules offer 3 band shelving EQ, both modules offer 8 aux, a single module with 4 band EQ or dual with 3 band shelving are available, and up to 32 stereo or mono ins and 32 monitors.

Price: From $9,999.00

M-3700 SERIES
Has VCA fader and mute automation built in, no external computer or monitor required, two automation modes available, dynamically automates all fader and mute moves in real-time and Snapshot which records all changes into 99 event files, internal disk drive and SMPTE reader/generator.
Price: From $11,999.00
M-3500
It offers a choice of 24 or 32 inputs and 8 group outs, in-line monitor and "Flip" function allow doubling of available sources at mixdown, 4 band EQ with dual sweepable mids, 6 aux sends with 4 available for main or monitor, AFL, PFL and Phantom power, also available in 8 stereo/24 mono configuration.
Price: From $7,499.00
M-2500 SERIES
This console is available in 16 or 24 input configurations with 8 group outs and stereo out, built-in MIDI Mute function stores 99 snapshots and may alternately be operated by external sequencer, three-band EQ features shelving high and sweep mid and low bands, four aux with two available on in-line monitors.
Price: From $2,999.00

M-1016/M-1024
These consoles are configured with 4 stereo and either 8 or 16 mono inputs, stereo inputs are ¼ in. while mono channels are XLR and ¼ in., three band EQ with sweep midrange, 2 pre or post aux sends and 4 effects sends, 4 mono and 2 stereo effects returns, PFL and headphone capabilities, rack-mountable.
Price: From $1,299.00
M-200 SERIES
These are mix and recording consoles with a choice of 8, 16 or 24 input channels, 3-band EQ, 2 aux sends, 2 stereo effect returns, PFL and full monitoring.
Price: From $1,199.00
MM 1
This is a keyboard mixer with 4 stereo and 12 mono inputs, the output is stereo, trim control has 40 dB of gain, additional features include 2-band EQ, 4 effects sends and 4 stereo returns, MIDI muting may be controlled by internal 99 scens memory or external sequencer, rack-mountable.
Price: $1,099.00

TRIDENT AUDIO USA
VECTOR 432
In-line broadcast, post-production, film, music console with 32 to 72 inputs, 32 multi-track/4 stereo/8 aux outputs, high/low pass filter, 4-band parametric EQ, 12 automated switches per input, optional stereo input/dynamics modules, integral/remote patchbay, VCA/Moving-fader automation/machine control.
Price: $130,000.00 to $350,000.00
MACHINE CONTROL SYSTEM
PC-based machine control for music and post-production. Master, slave, and MIDI (virtual) transport control via SMPTE and bar-beats. Featuring sophisticated cue list, multiple cycle, automated drop-in, and off-line functions, vector on-board QWERTY keyboard integrates machine control and fader automation. Includes synchronizer and computer.
Price: $24,950.00
SERIES 80C
Has 32 to 48 input music/post-production console with 24 multi-track/5 aux outputs/4 band equalizer, 24 track dual monitor (48 track capable) with full EQ/aux sends/long faders, 60 inputs on remix of 32 frame, integral patchbay, VCA/moving fader automation.
Price: $75,000.00 to $170,000.00
TRIDENT 24
Has 28 to 52 input music/post-production consoles with 24 multi-track/8 auxiliary outputs, 4 band mid sweep EQ/variable high pass filter, 24 monitor channels-2 band EQ/long faders, 4 effects returns, optional stereo inputs, integral patchbay for 28.36 frames.
Price: $24,950.00 to $39,750.00
TRIDENT 16
Features 16 to 56 input music/post-production console with 16 multi-track/8 aux outputs, 4 band mid sweep equalizer/variable high pass filter, 16 monitor channels-2 band EQ/long faders, 4 effects returns, optional stereo inputs, solo-in-place/auto muting bus.
Price: $15,000.00 to $35,500.00

WASHBURN
SOUNDTECH ST 324
A 32 input console with 4 effects/buses,1 stereo tape out, 4 group outs, 1 main stereo out, 2 monitor outs, 100 lbs.
Price: $4,749.90
SOUNDTECH ST 162
16 inputs 3 aux/effects buses, 1 stereo out, 1 monitor out, provides phantom power and reverb.
Price: $1,299.90
SOUNDTECH ST 122
Similar to ST 162 above has 12 inputs.
Price: $1,099.90
SOUNDTECH PC 800
As above, has 8 inputs, has 150W X 2 power amps, 45.2 lbs.
Price: $1,399.90
SOUNDTECH PC 1200
12 inputs, 3 buses, provides phantom power, 250W X 2 power amps, 56.2 lbs.
Price: $1,899.90

WHEATSTONE CORPORATION

TV-600 TV AUDIO PRODUCTION

The console has 8-56 mono or stereo inputs, 5 band tunable EQ, 4 stereo sends, 8 group buses configures stereo or mono, 2 stereo buses, 1 mono bus, muting and machine logic, Bus-Minus IFB system and VCA Grouping option.

Price: Available upon request

SP-6

The TV and radio audio production console features 8-56 mono or stereo inputs, 3 band tunable EQ, 4 mono sends, 8 group buses configured stereo or mono, stereo master bus, 2 mono masters, extensive logic features facilitate fast production and direct machine logic interface.

Price: Available upon request

SP-5

This 8-Bus audio production console has 8-56 mono or stereo inputs, 3 band tunable EQ, 4 mono sends, 8 group buses configured stereo or mono, stereo master bus, mono sum master; muting and tally logic.

Price: Available upon request

A-500

Radio "On-Air" console features two stereo buses, hidden mix-minus bus, 16 mono mic and stereo line input modules, comprehensive telephone, intercom and other special modules, console logic directly inter-faces with external machines without interface black boxes.

Price: Available upon request

3224

This 24 bus multi-track recording console features in-line configuration up to 56 inputs, 24 track metering, comprehensive soloing including tape, group, pre and post fader, 3 band tunable EQ, 4 aux buses, integral jackfield option, built-in talkback mic, built-in oscillator.

Price: Available upon request

MTX-1080

Up to 56+ inputs are featured in this sound reinforcement console which also has 8 aux buses each pre or post, 8 group buses, 8+ matrix mixers, stereo and mono masters, 5 band parametric EQ, 8 programmable mutes, solo and talkback.

Price: Available upon request

YAMAHA PRO AUDIO PRODUCTS

PM3000-24/32/40C

Available in 24, 32 or 40 inputs, 8 group buses, 8 aux buses (each Pre/Off/Post) and separate stereo bus, VCA assignable grouping with 8 submasters with automation interface 8 bus muting master system with safety override. XLR inputs are differentially balanced with 34dB trim and 5 position pad for optimizing gain structure.

Prices: PM3000-24: $34,000.00
PM3000-32: $39,000.00
PM3000-40C: $46,000.00

PM2800M-32/40

Available in 32 or 40 inputs with 8 group buses, 4 aux buses. 8 master mute groups with mute assign switches, 4 matrix mixes with level for all 8 channels, stereo L & R level and master. XLR inputs are differentially balanced with 34dB trim and 3-position pad for optimizing gain structure.

Prices: PM2800M-32, PM2800M-40: See your dealer

PM1800-16/24/32/40C

Available in 16, 24, 32 or 40 inputs with 8 primary mixing buses, 4 band sweepable EQ, 6 mute groups (can interface with 3000), 8 X 4 Mix Matrix and 4 stereo auxiliary returns.

Prices: PM1800-16: $14,000.00
PM1800-24: $17,000.00
PM1800-32: $20,000.00
PM1800-40C: $24,000.00

DMP7D DIGITAL MIXING PROCESSOR

All digital mixing and signal processing with digital inputs and outputs, 3 on-board Digital Signal Processors. Digital 3-band Parametric EQ on each channel. Preset memories: 32 internal, 67 external via supplied RAM Cartridge. Motorized multi-function faders, digital stereo output, compressor, 4 bar-graph meters and LCD parameter read-out.

Price: $5,995.00

DMP11 DIGITAL MIXING PROCESSOR

All digital mixing and signal processing with analog inputs/outputs, 2 on-board DSPs. Digital 3-band Parametric EQ on each channel, preset memories: 32 internal, 67 external via supplied RAM Cartridge, digital stereo output, compressor. MIDI control of preset changes and parameter manipulations, 4 bar-graph meters and LCD parameter read-out.

Price: see your dealer

MJ100 MULTI-SOURCE MIXER

Two main inputs sources, each with four rear-panel line-level stereo inputs, separate input selector switches for each stereo input on the two main sources, Cross Fader cross-fades between two source inputs, built-in digital delay system, independent faders for microphone and instrument level setting, 5-band graphic equalizer on the stereo bus.

Price: $350.00

M406 PROFESSIONAL SOUND MIXER

Six channel, with 3-band EQ and 6-position input level controls, high gain (84 dB) for full output, stereo program output with left and right master controls, echo/effects send bus with master send control, two effects inputs, each with level and pan control, right VU meter and headphone output switchable to monitor program or echo output, rack mountable.

Price: $1,295.00

MC1204/1604/2404/2408M

Available in 12, 16 or 24 inputs—4 program mix buses, 2 effects buses, 2 foldback buses and a cue bus, each input features a pad, gain control and peak LED for precise gain matching, 4 band EQ with the two mid-bands featuring quasi-parametric control, foldback 1 and 2, and ECHO 1 and 2 strappable pre/post EQ.

Prices: MC1204: $2,495.00
MC1604: $2,995.00
MC2024: $3,995.00
MC2408M Stage Monitor: $3,995.00

ADDRESSES

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5 Connair Road
Orange, CT 06477

AMEK/TAC U.S. Operations
10815 Burbank Boulevard
North Hollywood, CA 91601

Altec Lansing Corporation
P.O. Box 26105
Oklahoma City, OK 73128

ARX Systems
28271 Bond Way
Siverado, CA 92676

Audio Teknology Inc.
7556 Southwest Bridgeport Road
Durham, OR 97224

Audio-Technica U.S., Inc.
1221 Commerce Drive
Stow, OH 44224

Auditronics, Inc.
3750 Old Getwell Road
Memphis, TN 38118

Carvin
1155 Industrial Avenue
Escondido, CA 92025

Crest Audio Inc.
150 Florence Avenue
Hawthorne, NJ 07506

DDA
200 Sea Lane
Farmingdale, NY 11735

Electro-Voice
600 Cecil Street
Buchanan, MI 49107

Fostex Corporation of America
15431 Blackburn Avenue
Norwalk, CA 90650

Furman Sound, Inc.
30 Rich Street
Greenbrae, CA 94004

Industrial Research Products
321 Bond Street
Elk Grove Village, IL 60007

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Promoting Your Studio: Part 2

• In Part 1 of *Promoting Your Studio* (see the January/February 1991 issue of **db Magazine**), I discussed in a general way the two kinds of promotional avenues—advertising and public relations—and why both are necessary for promoting a service-oriented business such as a recording studio or production company. The article then turned its focus to just one of those avenues—advertising—and I developed some guidelines for using print ads (space ads, classified ads and direct mail) to help promote a studio business. That leaves us with one remaining avenue to examine: public relations.

As pointed out in the previous installment, while advertising has an inherent and definite cost, public relations (PR) is in the realm of things you can get, relatively speaking, for free. That factor alone should make this field worthy of some methodical study, especially for those studios running on a tight margin. Whoever has access to a word processor or typewriter, and has the budget for a sheet of paper, envelope and stamp can begin reaping the benefits of a public relations campaign. Of course, PR is not always so devoid of cost. Some techniques involve the staging of special events which can sometimes involve considerable out of pocket expenses, and unlike advertising (whose benefits often can be felt immediately), public relations may require a period of time before benefits begin to accrue. Hence, PR has to be seen as a long-term promotional policy.

THE TOOLS OF PUBLIC RELATIONS

Like any other discipline, public relations has its arsenal of tools, and these tools were created to achieve the espoused goals of the discipline. The goals of public relations are to mold the opinions and attitudes of a particular "public" relative to a particular person, business or organization so that a positive image might be attained and maintained over time. In plain English, this means you, as a studio business person , must educate your pool of potential clients to the quality and value of the services you provide. People form opinions based on available information which comes to them either through word of mouth or the media.

Good PR has the aura of genuine news. Unlike advertising with its obvious self-serving purposes, PR must at least have the appearance of providing information in the public interest. Quite often it is only a veneer which differentiates PR from advertising, but the point to remember is that PR must be received by the audience as news, rather than unmitigated hype. People must feel that they made their own decision about your service, based solely on the available information—without any coercion whatsoever. To attain this goal, public relations practitioners have developed several very effective tools, three of which will be examined in the course of this article: the news release, the publicity package and the promotional event.

THE NEWS RELEASE

As previously mentioned, PR must be regarded as news in order to be effective. From the vast amount of events that occur during a period of time, only a few are selected for coverage by the press. Editors of newspapers, magazines and other publications are always on the lookout for relevant stories geared towards their readership. Many stories are pursued solely on the basis of a news release that was sent to them from an outside source. A news release then, can be defined as a document designed to influence the media to write favorably about a particular subject—in this case, your recording studio.

There are many ways a news release can find its way into print. The simplest and most consistently available publicity can be had in the local music and entertainment journals—in the columns that are called something like "Studio Notes" or "What's Happening," and so on. Every city or region has a newspaper or magazine of this kind with a similar section reserved for studio hype. The publication usually won't give you more than one paragraph per issue, but most successful and also wanna-be successful studios are consistently represented in these columns. The purpose is to tell in a news-like manner who is recording at your studio. You needn't worry if it's just a local act; it will probably get printed anyway. The bottom line here is simply getting your name in print—irrespective of your clientele's importance.

The most desirable outcome of a news release is, of course, to have some sort of story written about your studio—ideally, a feature story replete with interview and photo essay. Obviously, your news release would have to be geared a little differently to attain this end. It might have to be almost a complete story in itself—or at least the outline of a story with a strong "hook" that would be interesting to both the editor and readership. To really make an impression, it should probably also contain a very attractive photograph. A story is often given preference because it contains a great photo, so don't skimp on the film.

It is difficult to say what the content of a news release should be, since this might vary from journal to journal. What's best is simply to study the particular publication you are intending to pitch to and write down the essentials of a story appropriate to the editorial slant of that magazine. In other words, scope out the magazine; read back issues and find out what issues the magazine deals with and use this as a guideline for the content of your news release.

The form of a news release is much easier to specify; it follows a format that never changes—regardless of content. *Figure 1* shows the basic elements of a news release. At the top in large type is a banner stating boldly that the piece of paper is an official news release — not to be confused with any other papers on the editor's desk. Directly below, tight to the left margin, is the name and address of the company sending the release, and below that, the all-important name of the contact person at that company. Next follows the release date, which is usually stated as being *For Immediate Release* since you want to get across a sense of timeliness (or even urgency) in establishing a need to print the information. The actual story itself must start with a hook-laden headline: it must draw you into the text by arousing your interest. Next (since this is "news"), the city of origin must be identified.

Finally, we reach the body of the story itself. The first line is very important. In true journalistic fashion, it should give a summary statement of the entire article, — the story at a glance. Then, in the subsequent paragraphs, the story should be fleshed out in greater detail, being certain to answer the usual journalistic questions: who, what, where, when and how. Another important axiom to keep in mind is that "news makes news." This means that if a particular issue is currently in the public eye and you have something being recorded at your studio that will relate to it, there is a much greater chance it will be published.

A classic case might be if a peace activist were recording anti-war songs during the current Middle Eastern conflict; but the example does not always have to be that dramatic. Perhaps someone is recording keyboard tracks using a hot new synthesizer or the artist was formerly lead vocalist for a popular band or the bald-headed keyboard player is really a brain surgeon by day; if you dig hard enough, you can find a newsworthy item about many of your clients, so start digging!

THE PUBLICITY PACKAGE

While the news release is a more or less expendable tool of public relations (since it refers to a one-time news event), we also have need of a relatively permanent tool—something we can use over and over to provide information and affect public opinion. Here is where a publicity package (also called a press kit) comes in handy. The elements found in a publicity package can be almost anything, but there are some important items that should be included.

First of all, the term "publicity package" implies that there is some sort of container to hold multiple pieces in place; — that's the essence

NEWS RELEASE

Company Name
Address

CONTACT: Individual's Name
Telephone Number

FOR IMMEDIATE RELEASE

HEADLINE ______________________

City of Origin. Body of text ....

Figure 1. The news release.

of a package or a kit. When someone wants to see the profile of your studio at a glance, he can reach for this package and skip to whatever documents he's interested in. Some studios photocopy or print whatever they feel is relevant on standard 8.5 in. by 11 in. paper and put all the sheets into a clear plastic binder. This looks very neat and assures that none of the sheets will get lost. Others prefer to use a pocket binder or portfolio because they are available in attractive colors and you can store lots of odd sized materials within.

One of the major items in a publicity package is usually a single page statement of purpose for your studio business detailing the kinds of services you provide. Also commonly included might be a short biographical sketch of the owner(s)/operator(s) of the studio telling of their education, previous experience and achievements. (If you've only been in business a short time, the above two items could be written as a single document.) Another good item to include is a list of client references. (Don't feel insecure if your list does not contain someone famous; even a list of unknown local clients will establish that you have a legitimate business). Additionally, you might consider including copies of any testimonial letters from satisfied clients, a good photograph of your facility and reprints of any previous press coverage you may have. A rate card (detailing a range of prices for various services) is also a very desirable item to include. One other option might be to include a sample tape featuring short excerpts from your best projects.

By now you may be thinking that this could be a fairly expensive proposition on a large scale. When all is said and done, it might cost you a couple of bucks a package (or more if you include a tape). Obviously, don't use a publicity package as a tool for mass mailing; instead, reserve it for contacts who are potentially serious clients or for those highly visible media people (such as magazine editors) who might be able to give you some publicity. If you make even one such contact, the time, effort and money spent in putting together these packages will seem rather insignificant.

THE PROMOTIONAL EVENT

One of the most effective ways to gain publicity is by staging or becoming involved in a promotional event. A classic example of this is the production company that offers its services to a regional charity. The idea is that an original theme song will be written and produced by the company and released to radio stations as part of a campaign to increase support for the charitable organization. These special events are difficult, but not impossible to pull together; today it may be easier than ever to get involved in such special events, since all charitable and community organizations are suffering from a decline in funding. The offer of free audio production can seem very attractive to an organization seeking to shave budgetary corners, and the publicity a studio can glean from such an association can pay off big time.

An event that requires working with an outside organization also requires much planning and coordination. However, there is one event you can run without anybody's help: an open house. An open house is really a fun way to introduce your services to a new group of people and at the same time, show appreciation to your old clients for their patronage. It's simply a party you throw at your studio. Just provide an urbane selection of wines and cheeses, some mood lighting and, of course, a preprogrammed tape of all your best productions.

This can really help people get turned on to your studio. It's a great opportunity to schmooze with old friends and network out to some new clients. The open house is really a powerful public relations tool: your audience is handpicked and the mere fact that you are a gracious host makes them more receptive to using your services in the future db

MARK MAULUCCI

The Guitarist In The MIDI Studio

• This month's missive, though aimed at guitarists, is really for those who operate or work in a MIDI studio. I define a MIDI studio as one where all the re- cording is done directly. That is, the sound sources (usually all electronic) are patched directly into the mixing console or tape recorder and all the gear shares the same space. This is opposed to a regular recording studio where there is a room (or rooms) devoted to the mic'ing of drums, amplifiers, vocals, and so on, away from the control room. The sound of these rooms is paramount, just as it's important the control room be just that—a controlled environment where the only sound heard is that coming out of the playback monitors.

The MIDI studio is a control room only. Usually the haven of keyboardists, the MIDI studio features a rack of keyboards, digital and analog sound modules, signal processing, line mixers and at the heart of it all: a sequencer (whether it be a computer or a dedicated MIDI sequencer).

Often times, the MIDI studio is not even a multi-track recording studio. Many keyboard and soundtrack composers do their composing, arranging, tweaking, processing and mixing strictly in the MIDI domain. If using a sequencer to control performance data, these composers don't need to overdub or commit to tape until their MIDI masterpiece is completed. They often have 16, 32, 64 or up to 256 tracks of sequencing information at their disposal! After programming the entire performance and mix to their satisfaction, they can record direct to stereo, hence never losing a generation. With the advent of hard disk recording and DAT, composers don't have to leave the digital domain at all.

It's hard to imagine life without MIDI. Like in-door plumbing, I've taken MIDI completely for granted. I can still remember the days before MIDI, though. Just ten years ago, I was walking the songwriter's treadmill: write a song, save up some money, gather musicians, rehearse the song, procure studio time, record song, spend money, pitch song and so on and so on. Now whenever the mood strikes, all I need to do is pry myself off the couch, stumble into my studio and voila—instant gratification. What if it's 2:00 a.m. when inspiration hits? No problem. The instruments are direct, so I can listen on headphones. Family, roommates, neighbors even, are none the wiser. Why, one could own and operate a MIDI studio in a high rise apartment building. Believe me, it's being done!

So, no more dealing with the schedules and high rates of studios and musicians, right? MIDI is a means to an end but it's not *the* end. Our end is to make music and many times our music needs vocals, acoustic and electric guitars and other acoustic instruments such as saxophone, for instance. Except in rare instances, sampled guitars and saxes don't make it for me. The point of this article is how to integrate our organic stuff with technology so they become a balanced whole.

IN SEARCH OF TONE

Guitarists are often a strange breed. I can say this as I am a guitarist. Unlike many of our keyboardist comrades who salivate over the latest technology, we guitarists often reserve our saliva glands for old things—guitars and amps from a bygone era. As a performing artist, I use a 34-year-old Fender Stratocaster through a variety of pedal effects and a combination of older tube amps. Primitive you say? I concur but I love the tone.

When I'm in my studio, however, it's a different story. It's a direct studio with my MIDI gear to the left, my multi-track and console in the center and my effects rack to the right. It's all very convenient. My amps are invariably over at different drummers' houses. Being somewhat lazy, I don't want to be always schlepping gear in and out of my studio in addition to gigs. Plus, being in a direct studio, I avoid mic'ing whenever possible. I plug into a Boss GL-100 pre-amp ninety percent of the time which is then routed in stereo to my patch bay. That way, I can easily patch in compression or noise reduction if necessary. From there, it goes directly to the console.

The GL-100 has some neat features: two switchable channels with gain and master volumes, equalization and level controls, nine tone modes from ultra-clean to Godzilla distortion, stereo effects loop, hi and lo Z outputs, as well as buffered and monitor outputs, and a speaker simulation feature that electronically emulates the tonal characteristics of a reactive load. The thing has knobs and sounds remarkably like a mic'd guitar amp.

There is a plethora of guitar pre-amps and effects processors on the market from the likes of Korg, Yamaha, Roland, Chandler, Groove Tube, Tom Scholz, ADA, Applied Research and Technology and ZOOM, among others. Many have similar features and some seem to do everything except make coffee in the morning. Budget and sound preferences are the prime determinants of choice here. The abundance of this sort of product indicates the demand in the marketplace by today's guitarist.

I often find myself compromising when working in the electronic studio. I'm just not set up to mic stacks, and my neighbors wouldn't approve anyway. If I must have an amp sound, I'll mic a 12 watt Fender Princeton reverb and stick it in the closet, whatever the song needs.

We really are creating illusion here folks, much in the same way that film makers do. Most of the great exterior space scenes in the Star Wars saga were done with miniatures and computer graphics. Heck, there isn't even sound in outer space.

Again, to achieve balance in my opuses, I mic vocals, acoustic guitars and the occasional wind instrument. I've been inviting other musicians over to play as well. After seven years, I'm tired of programming all the parts myself! But more on that next issue.

SEQUENCING FOR THE FAINT OF HEART

Sequencing is an area that often bamboozles guitarists. Most of the sequencing done by MIDI types is of the linear variety. That is, one chooses a track and records the performance from beginning to end much like the approach to multitrack recording.

Then the user goes back to edit, change or correct any event (notes, volume, bender information, any MIDI control parameters) on that track. The programmer then chooses another track and builds the performance as one might overdub on a sound recording.

This can go on and on depending on how many tracks the sequencer has, its merging and unmerging capabilities, the number of MIDI channels it will transmit on (multiples of 16) and the number of sound sources at his or her disposal and their multitimbral properties. Of course, this all occurs in the digital domain. It's not sound, but performance data only. A space age player-piano if you will.

For some reason, I've never been that comfortable with linear sequencing. Over the years, I've learned to become quite adept at programming rhythm machines. Being primarily a pop songwriter, I tend to think in terms of patterns. You know, intro, verse, chorus, bridge and so on. That's how I like to sequence to this day. I write patterns, edit them and string 'em together. Many sequencers can't do this. I recommend the exploration of pattern derived sequencers or software to those would-be programmers out there who have been intimidated by linear sequencing. It might be just the avenue you've been waiting for to enter into the fray.

Next time: Drummers and bands in the MIDI studio. db

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JIM PAUL

Winter NAMM Roundup

NAMM Review

• The 1991 Winter NAMM Show will go down in history for several reasons. First, despite the wheeling and dealing on the floor, there was an undercurrent of concern about the Persian Gulf war which started the day before the show. Second, NAMM officially announced the cancellation of the summer show, which was a great relief to many of the people I spoke with. Third, this was a show to top all shows! In a word, it was **huge**! Nearly 800 exhibitors, wall-to-wall attendees, and not much talk of recession all added up to a very successful event. Here is a taste of the goings-on at NAMM.

CONCERT EVENTS

One of the best parts of any NAMM show are the fantastic concert events, and this year was no exception. On Friday night, St. Louis Music sponsored a huge free food and drink party for about 2,000 people with four great bands performing. Featured were *Boxtown Bandits*, *Tom Borton Band*, *EYES*, and *Ken Hensley Jam*.

Saturday was the toughest night because one was faced with several excellent choices. How about guitar wunderkind *Steve Vai* (does he really have eight fingers on each hand?) in his first ever smokin' solo performance? Or you could choose *Eddie Van Halen* with *Steve Morse*, *Albert Lee* and *Steve Luthiker* in a crash-bash rockin' guitar shootout. For the more eclectic among us, there was the infamous *Spinal Tap* in its original form, hacking away at the frontiers of musical weirdness. And for percussion heads, there was *Chad Wackerman* and his buddies Wack-ing away in a drum-workshop/concert/jam session.

Sunday was easy and in my mind the high point of all the aforementioned concerts. Valley Arts USA presented an "Intimate Evening with Larry Carlton" and it was even better than it was billed. The room only held about 600 people and the sound was fantastic. Carlton was absolutely stunning in his ensemble numbers but especially in his incredible solo guitar pieces. Larry, we're so glad you're back!

NEW EXCITING PRODUCTS

Yamaha Ultra-Miniature MIDI Studio

In what may have been the most innovative and exciting product on the floor, Yamaha introduced a complete MIDI studio about the size of a VHS video tape. This little box contains an 8 track, 6,000 note sequencer; an AWM sample player with 30 multi-timbral sounds, 26 note polyphony, and 27 sampled drum sounds. In addition there are 76 ROM-based songs, each with 4 parts consisting of a rhythm part, a bass line, and two complete harmony or chordal lines. Each song may be ported into the sequencer to then be manipulated by the user! And it sounds good. This most amazing little box is only $399.00 list and is shipping now. Bravo!

QSC Creates a Whole New Amp Line

When QSC introduced the EX-4000, many people were impressed by its price/performance ratio. When they cut its price by $1,000.00 even some of the sceptics were converted. Now they have introduced a whole new EX line modeled after the EX-4,000, with high-end specs, rugged design, and very reasonable prices. One true convenience feature to the gigging musician is that the EX amps will operate on a 15 amp circuit with no fear of power loss or blown breakers, and no more searches for 20 or 30 amp circuits.

Kurzweil Returns!

The music industry reacted with sadness in 1990 when Kurzweil filed bankruptcy, watching another American musical company going down. Enter Young Chang Inc. of Korea with new ideas, manufacturing and marketing expertise, and lots of capital and before you can say Lazarus, Kurzweil is resurrected as a division of Young Chang. They reportedly kept most of the engineering staff , and replaced the marketing staff. Kurzweil's big announcement was the K-2000, a new super-hot keyboard billed as 10 times more powerful as the venerable (if bulky) Kurzweil 250. The working prototype sounded incredible, and with a little redesign of the outside, they may have the next big keyboard winner. Available June, 91.

Tascam Automates, Integrates, and Invigorates

With full production of their smash 3700 automated console at hand, Tascam refuses to sit on its aspirations. They unveiled their new 2500 series consoles which contain many of the features of the 3500 and sell for a lot less. By integrating functions, scaling back slightly, and replacing the in-line monitor faders with pots, they cut the cost and still offer 4 aux sends, MIDI muting, a three band/2 sweep EQ section, phantom power, 2 stereo effect returns (plus 2 mono), and even more. The 16 input M-

2516 list at $2,999.00 and the 24 input M-2524 at $3,999.00. Remember that these are 16 X 8 X 6 and 24 X 8 X 24 boards! This effectively doubles your inputs at mixdown time.

Roland: A Step Back Into The Future

With the number of sample playing keyboards up to about a zillion, leave it to Roland to come up with a new way to do an old thing (or is it an old way to do a new thing? Oh well...) Enter the JD-800, Roland's new keyboard with lots of knobs and sliders that control gizmos called envelope generators and doohickeys called filters and widgets called LFOs. Reminds me a lot of a synthesizer. With a sleek *stealth* look and sounds to match, this may be another big winner for Roland (remember the D-50?)! Also Roland introduced lots of other good stuff like a digital single-ended noise eliminator, a SMPTE/MIDI interface box, a less expensive version of the S-770 sampler, and an entry level synth—the JX-1.

Passport Sequencer With Digital Audio Recording

Continuing the trend toward tape-less studios, Passport Designs showed off a new product called Audiotrax which incorporates MIDI sequencing with 2 tracks of digital audio *without* any extra hardware cards! While the fidelity is limited (8 bit sampling at 32 kHz), it is fine for quickie demos or scratch vocals. The vocal demo I heard sounded quite good with effects and music to mask the audio graininess.

Has Karaoke's Time Arrived?

There were more companies hawking *Karaoke* or sing along systems than ever before this year. Crowd interest however, seemed to be low. This led me to wonder if the systems which are popular in Japan (reportedly a 3 billion dollar per year industry) are really catching on in the U.S.A. I cornered a rep from Pioneer's LaserKaraoke division and he expressed a lot of enthusiasm for the future, saying that he sees Karaoke systems in homes and clubs doubling or tripling in the next five years. Personally I still don't know of one location in Southern California which has a commercial system in place. Maybe I'm just not enough of a party animal!

Sony Squares Off in the Signal Processing Arena

Correctly assessing a still growing market for high quality affordable signal processors, Sony introduced two new boxes, the MU-L021 compressor limiter, and the MU-E041 parametric equalizer. They also were debuting a digital delay, the DPS-D7 digital hyper delay, but they didn't have one to show. Anyway, while I was wondering who thought up those names, I checked out the units Sony did have. They are attractive, functional and easy to use. The EQ is 4 band with centers from 20-320 Hz, 80-1.3 kHz, 310-5 kHz, and 1.2 kHz-20 kHz. The comp/limiter has standard controls such as attack, release, threshold, and compression ratio, plus a gate threshold. The compressor lists at $799.00 and the EQ at $699.00, making them competitive with comparable models from other companies.

Alesis Announces a Digital 8 Track

With all the companies working on inexpensive digital multi-tracks (Roland, Yamaha, Korg, Digidesigns, etc.), it should not surprize anyone that Alesis would throw their hat into the fray. They did, after all, wow us with the MidiVerb, the HR-16, the QuadraVerb and the 1622 Mixer. This unit, dubbed the ADAT will use a ½inch S-VHS tape, variable sampling frequency from 42.76 to 50.85, 64 times oversampling, +4 or −10 inputs and outputs, and comes with a remote control unit. Up to 16 units may be chained together for up to 128 tracks. If Alesis can bring it in on schedule (4th quarter 1991), on budget ($3,995.00), and on target (see specs above), they may have another killer product. Good luck guys!

JBL/Soundcraft Gives Power to the People

You can't get away from these people. JBL speakers, Soundcraft consoles and UREI signal processors are everywhere. In the sound reinforcement area however, some have complained that JBL reserved the best stuff for the large touring sound companies. Well, not so anymore. With the introduction of the SR-4700 series of loudspeakers and the new Spirit consoles, it is possible for the small user to have an awesome sounding reinforcement system. The 4700s contain the same components as their 4800 professional series, but in a less expensive box, and they sound great! The Spirit console, which is available in a recording or reinforcement version, incorporates much of the technology that made the Soundcraft name famous, but now it's very affordable. Hurray! hurray!

Casio Rap Box

Casio has been known for creative little consumer keyboards with good sounds and inexpensive prices. Their newest product definitely follows that tradition with built-in rap rhythms, a built in microphone, and digital pitch-shifting! There's even a pad for adding DJ scratching as well. You haven't lived till you've heard the Beverly Hillbillies song done in rap style, and pitch-shifted down an octave!

Now MAC Can Drive DAT

When Panasonic introduced the SV-3700 at the recent AES, everyone went wow. But for the R&D guys at Panasonic, wow isn't enough. They went ahead and came up with the $2,100.00 SV-3900 which is a 3700 but with one very important difference. The 3900 uses a serial interface which allows the user to daisy-chain up to 128 of them together and best of all it_s totally controlled by a computer. Panasonic is developing little control routines in C (a computer programming language), which can be used to incorporate machine control into virtually any software program. Now your sequencer can even start and stop your DAT via MIDI. Pretty soon we engineers won't have to do anything in a mix except hit the GO button and go eat lunch!

Sennheiser Tinkers With A Legend!

Virtually no studio is without a couple of Sennheiser MD-421 mics for kick, toms, guitars, and even vocals. So imagine my shock when they told me at their booth that they had a new model 422 mic to upgrade (not replace) the 421! After they calmed me down, they explained that the only real differences are better, off-axis rejection, a slightly hotter output, tunable low end, and more rugged construction. I felt much better and an A/B comparison showed it really sounds good! I was tempted to stomp on it to check out the ruggedness of its construction, but was discouraged by the staff from doing this. Oh well, I can say it still has the 421 sound plus a more sleek look.

Sennheiser's new mics were shown by Michael Lieske

The along-awaited E-III expander has arrived.

Sennheiser also introduced the new Blackfire mic with superior off-axis rejection and a sliding wind screen which has the effect of tuning the amount of proximity effect the mic will produce. We'll see if it catches on!

E-mu Keeps Moving Forward

With its cool little piano boxes, Proteus I and II, the E-max,and the Emulator III, E-mu has kept busy the last few years. This year they were happy to announce the long awaited expander of the E-III. This will take the machine into realms where it's never gone before with 32 megabytes of RAM, on-board DSP, 64 times oversampling Delta/Sigma A/D converters and AES/EBU digital I/O. The company also introduced its Procussion sound module with over 1,000 sounds based on the Emulator III library. With that many sounds to choose from, anyone should be able to find that killer snare!

Crown Intros Macro-Tech Amp

Crown International was showing off its new Macro-Tech 3600VZ amp rated at 1800 watts/channel at 2 ohms, 1655 into 4 ohms, and 1165 into 8 ohms. Using technology from their Macro-Reference studio amp, Crown has packed high specs into a 3 ½-in. high package. 105 dB S/N ratio, 20-20 k response ⅒ of a dB, and a damping factor greater than 1000. It all adds up to an impressive package.

THE 1ST ANNUAL JIM PAUL CLG (CREATIVE LITTLE GUY) AWARD

Every year, the big guys in this industry get all the press, all the attention, and often all the sales. This is not unusual because one can't compare international giants like Sony with the tiny Acme Guitar Pick Company. However, it is often the little guy who comes up with that really innovative product that makes people say "Gee, why didn't I think of that?" So, in order to find a worthy recipient, I spent time at the outside corners of the NAMM show, at those little one space booths that many people skip and I came up with a two- way tie for the *1st Annual Creative Little Guy Award.*

JBL\Soundcraft reps frame this new Spirit console

Spins International Inc., Montoursville, PA

This company produces a guitar/bass strap which is like a double shoulder harness and distributes the weight of the instrument much more evenly for those long gigs. But the really cool thing which caught my eye is that the guitar and its cable attach at a rear pivot point, allowing it to be spun while still playing. It looks great and is a very cool marketing gimmick, as well as a innovative idea. Congratulations to owner Ken Wittman.

Y Square LTD, Inc., San Clemente CA.

This company manufactures a wonderful product for keyboardists and anyone who uses MIDI control—a fiber optic MIDI cable! This allows the user to run very long cable runs without fear of EMI, RFI or data errors. The company sells 50 to 200 foot cables, but it has been reported that this type of cable will transmit data reliably over *several miles*! So here is a small entrepreneurial company taking advantage of technology developed by large corporations such as the phone company to create a very hip little product for the music industry. Good show!

With nearly 800 exhibitors, it was impossible to get around to everyone, and then print space is always limited, but I hope this sampling gives you a feel for this huge and exciting show. With all the new products and the enthusiasm on the show floor, it would be safe to say that the music-technology business is alive and well in 1991!

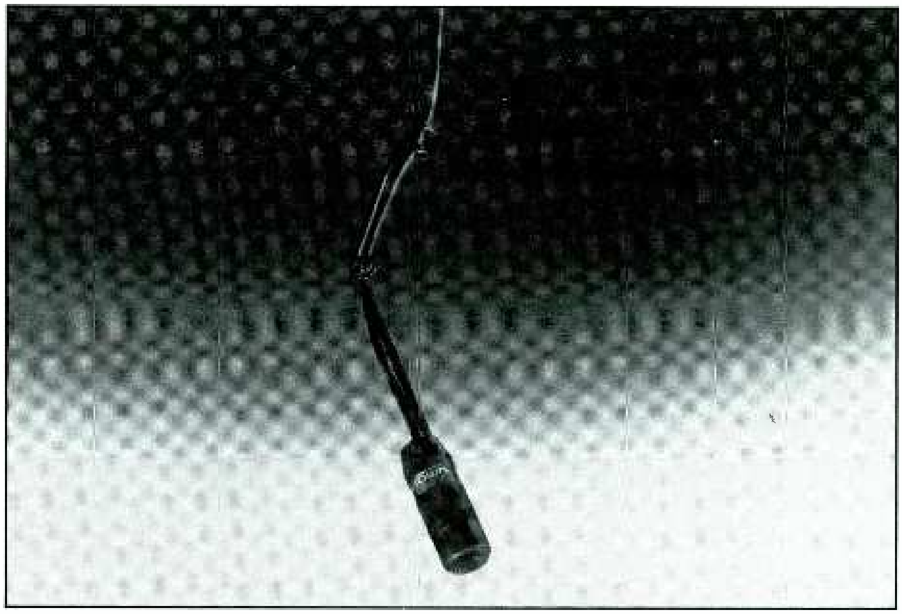
NEW PRODUCTS

MINI MIC

● Specially designed for hanging applications, the CM-30 microphone is a miniature, supercardioid electret condenser unit ideally suited for use above a choir, orchestra, or stage. Operable with 12-48V of phantom power, the CM-30 incorporates electronics which attach to a rectangular or circular electrical plate for quick installation into any ceiling electrical box. Featuring a sensitivity of 13.5 mV/Pa and an impedance of 150 ohms (balanced), the mic is also equipped with a strain relief which can be used to securely clamp the unit's cable at the desired length. The electronics/plate assembly can additionally be mounted to any electrical handy box.

Manufacturer: Crown International
Price: $215.00

Circle 70 on Reader Service Card



NOISE REDUCTION

● The RX-NR4 Quad Noise Reduction System is ideal for use in sound reinforcement, multi-track recording, instrument processing and other signal processing applications. The RX-NR4 is a single rack space unit featuring four discrete noise reduction circuits, an expander function for expanding the strength of the input signal within a range of 1:1 to 1:8, reduction in out switches, a sweepable low pass filter for noise reduction or full bandwidth noise gate operation. The controls featured on the RX-NR4 are threshold, ratio, decay and filter adjust, with bypass switches for each noise reduction circuit.

Manufacturer: Ross Systems
Price: $249.95

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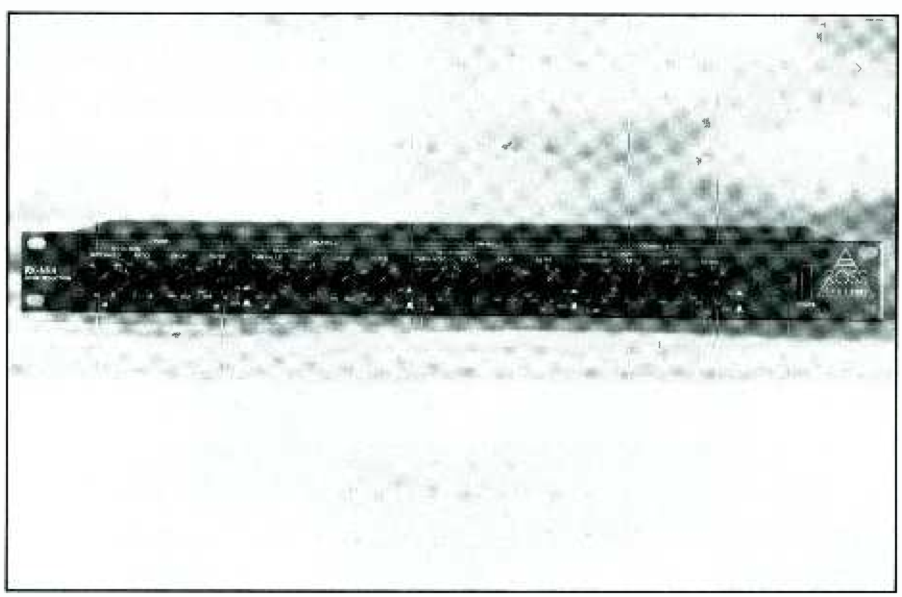
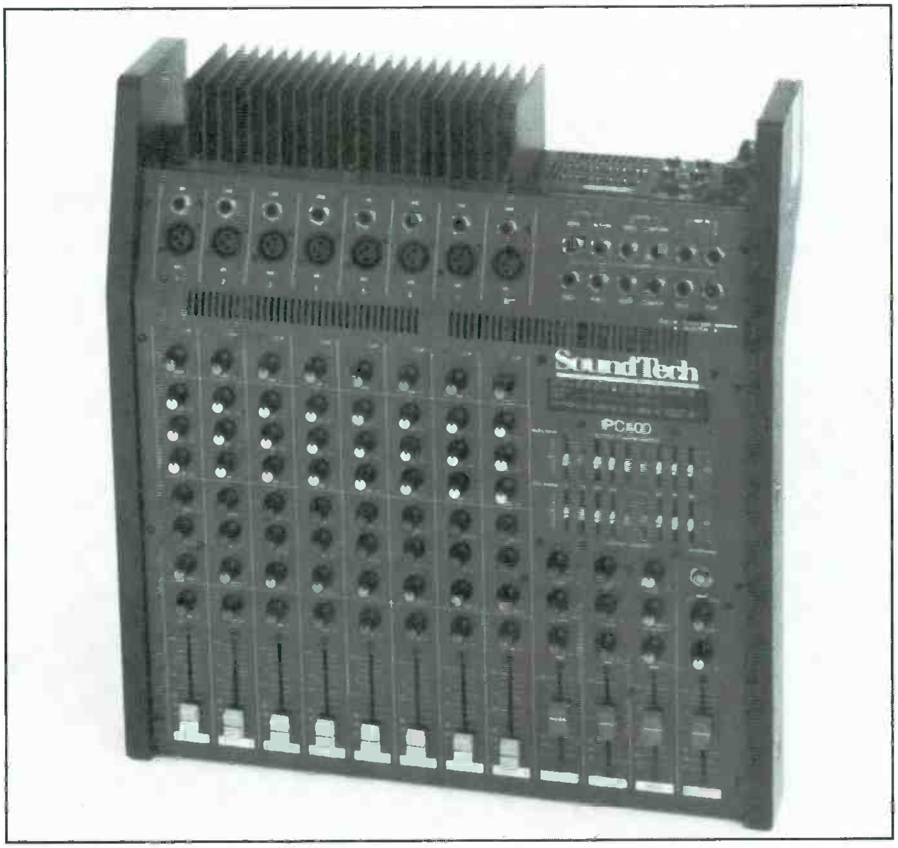


MIXER/AMPS

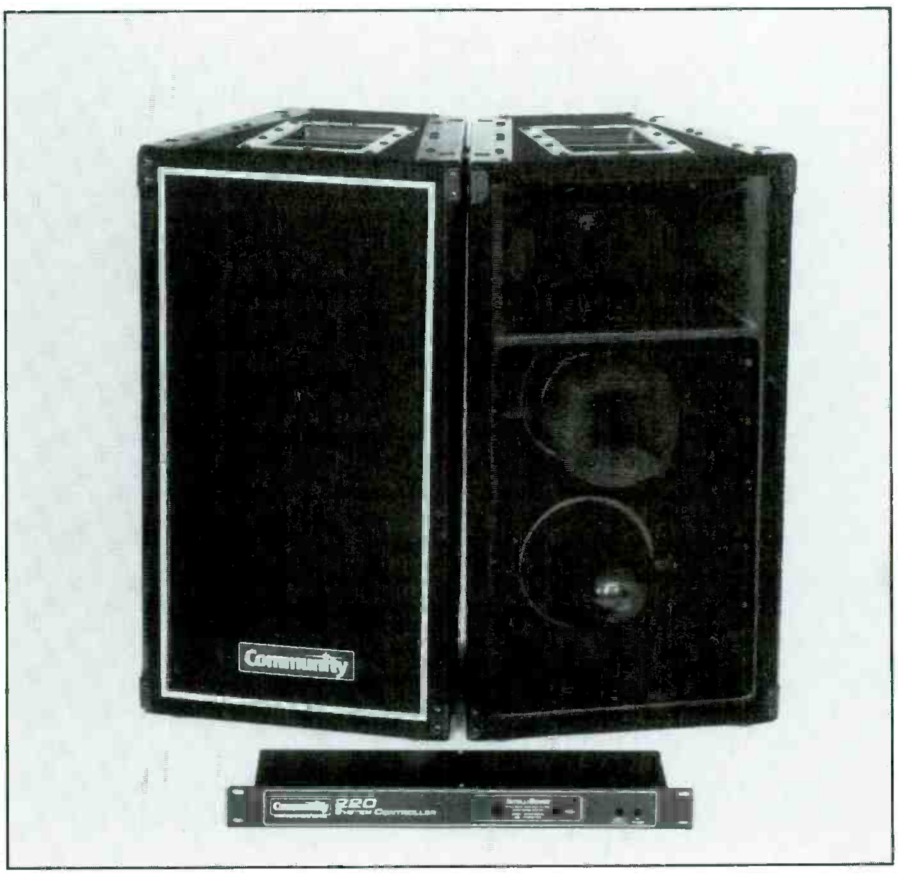
• The PC800 and PC1200 mixers boast 8 and 12 channels and stereo power ratings of 150 and 250 watts per side respectively. Each mixer can be used in either an 8 (12) X 2 stereo format or in an 8 (12) X 2 X 1 mono configuration. In the stereo setting, the built-in power amp serves to amplify the left and right main outputs. In mono, the power amp sends power to the monitor speakers and the main systems, allowing the PC series to be used in a variety of applications. Each power amplifier sections also has a 9 band graphic equalizer that is switchable between stereo or main/monitor settings. The units also have selectable metering and an extensive patch bay that allows expansion of the system and integration into larger systems. The PC800 and PC1200 are convection cooled for totally quiet operation, making them ideal for use in areas with very low ambient noise such as churches.
Manufacturer: SoundTech
Price: PC800—$1,399.90,
PC1200—$1,899.90



Circle 72 on Reader Service Card

FULL-RANGE POWER

• The RS220 loudspeaker is operable from 100 Hz to 18 kHz and posts power handling figures of 200 watts pink noise/500 watts program, while maximum output is greater than 127 dB at 1 meter. Drivers incorporated in the RS220's three-way proprietary Wavefront Coherent design start at the low end with twin 8 in. ferro-fluid cooled dual-spider woofers. Proceeding up the frequency spectrum to the midrange, each RS220 pack an M-200 compression driver, which employs one-piece Mylar diaphragms and feature 2 in. exit throats. High frequencies are directed to another compression driver equipped with a titanium diaphragm and a 1 in. exit area. Crossover points are at 800 Hz and 3 kHz, while the loudspeaker produces a 65 degree nominal pattern in the vertical plane. Housed in an enclosure identical to the RS220 which is available with the same options, the VBS210 subwoofer contains a pair of 10 in. dual-spider drivers which operate between 60 Hz and 150 Hz when used with the 220 System Controller.
Manufacturer: Community Light and Sound
Price: RS220—$995.00, VBS210—$595.00, 220 System Controller—$690.00



Circle 73 on Reader Service Card

ACOUSTC FEEDBACK ELIMINATOR

• The FBX Feedback Exterminator is a microprocessor-controlled filtering device which automatically finds and eliminates feedback almost as soon as it occurs. The FBX quickly senses whether a PA system is experiencing feedback and determines its pitch. Once feedback occurs, the FBX assigns one of six very narrow notch filters to the resonating frequency and quickly eliminates the feedback. The FBX utilizes six narrow-band, independent, parametric notch filters that operate in two stages—three Fixed and three Dynamic. The FBX sets the three Fixed filters to control the strongest feedback frequencies. The three Dynamic filters control intermittent feedback that may develop during a program. They are automatically reassigned new frequencies as feedback occurs.

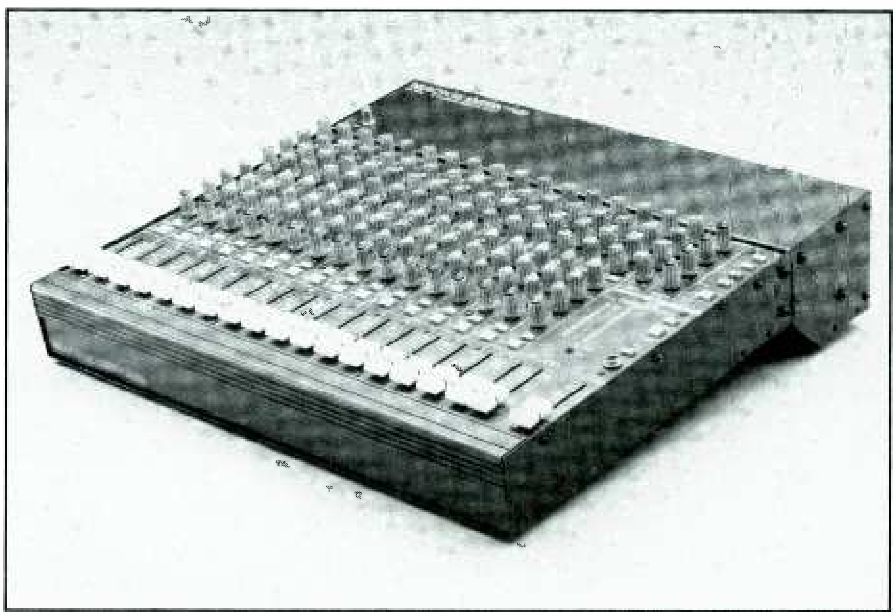
Manufacturer: Sabine Musical Manufacturing Co.
Price: 499.95

Circle 74 on Reader Service Card

VERSATILE MIXER

• The CR-1604 features seven AUX sends per channel and four stereo or eight mono effects returns, balanced main stereo outputs, eight direct outputs, studio-quality + 48-volt phantom-powered MIC preamps and channel inserts on inputs. It is said to have a working signal-to-noise ratio of 90dB along with 112dB dynamic range. The CR-1604 can be used as either a rack-mount or table-top mixer via a unique convertible configuration. All inputs and outputs are contained in a rotatable connector pod. When this pod is turned 90 degrees to the back, the CR-1604 becomes a seven-space rack-mount mixer with all connections except headphones on the rear. When the pod is mounted flush to the front of the mixer, the CR-1604 becomes a sloped tabletop console with easy-access to all inputs and outputs. Other features include built-in power supply, in-place stereo solo, UnityPlus™ gain structure, double-redundant parallel-wired faders, environmentally-sealed rotary controls and BNC connector for gooseneck lamp.

Manufacturer: Mackie Designs Inc.
Price: $1,099.00



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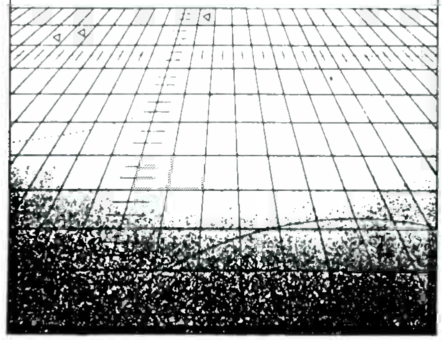

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IN MEMORY, Robert Trabue Davis: November 3, 1935-January 23, 1991

We are deeply saddened to announce the death of Robert Trabue Davis. Bob was truly a person who, through his dignity, humor, intelligence and gentleness of spirit, made this world a better place for all who knew him. In addition to his contributions to the audio industry, Bob will also be remembered for his gift of music.

Bob began studying piano at the age of five, and by the fourth grade, he had switched his interest to the clarinet. Within two years he had so advanced his skills that he began taking lessons at the Music Department of the University of Kentucky. In the summer of his eleventh year, Bob went to the Interlochen Music Academy in Michigan, and studied with Keith Stein of the Chicago Symphony.

He continued his dedication to music throughout his junior and senior high school years. Every other weekend, he travelled 160 miles from his home in Lexington, KY, to the Cincinnati Conservatory of Music to study with the principal clarinetist of the Cincinnati Symphony.

Bob attended Michigan State University for two years before transferring to the University of Kentucky, majoring in broadcasting and communications, with a minor in music. After receiving his B.A. in 1959, Bob did his post-graduate work at the University of Kentucky and received a Master's in Musicology. Upon graduation, he joined the faculty of the Music Department of the University of Kentucky. Although he truly loved the study of music, he often said in jest, "the only thing a Master's in Musicology is good for is teaching others to be Musicologists."

He joined the Lexington Philharmonic as principal Clarinetist, played in the faculty woodwind quintet, and gave recitals. During this period, he became interested in vocal music, which was to become his lifelong passion.

He left the University and went into broadcasting as operations manager and announcer at radio station WSAC, Fort Knox. His vocal training enabled him to work on-air without a regional accent.

After his short career in radio, he reentered the audio visual industry with a commercial sound contractor in Lexington, KY. His father was a well-known audio/visual dealer in the Southeast, and Bob had worked with his father in his youth. From there, he joined Technical Service Corporation, a sound contracting firm in Louisville as vice president of sales.

While in Louisville, he joined the Louisville Woodwind Quintet, the Kentucky Opera Association, and the Louisville Bach Society. His operatic roles included both Jenik and Vasek in "The Bartered Bride" by Semerana, Edguardo in "Lucia," Rinuncio in Puccini's "Gianni Schicchi," and Faust in Gounod's "Faust." In addition to opera, he enjoyed singing and performing in musical comedy, and he sang the lead roles of Tommy Albright in "Brigadoon," Billy Bigelow in "Carousel," and El Gallo in "The Fantasticks." He also pursued his love for sacred music with performances of the St. John and the St. Paul Passions by J.S. Bach.

He moved to Detroit in 1971 and joined Industrial Communication Company as a sales engineer and sound system designer. Here, he began the practice of including his resume with his proposals so the client could judge the qualifications and character of the designer. He also gained a reputation as the man who installed more multi-cell horns in churches than anyone else before him. Using his knowledge on architectural acoustics and sound system design, combined with his teaching background, he began to lecture at the University of Detroit and the Lawrence Institute of Technology. He continued his performing career with the Detroit Cantata Academy.

Leaving Detroit in 1975 for Anaheim, CA, he joined Altec Lansing Corporation as manager of Market Development and Training. In two short years, he was promoted to director of Systems/Application Engineering, and in 1979, became vice president for Market Development. It was during this period that the dynamic products such as the first digital delay lines, Incremental Power amplifiers, constant-directivity horns and large-scale touring sound systems were developed at Altec under his supervision. He worked on system design for sound systems for the 1976 Montreal Olympics, The Moscow World Trade Centre, Sun Devil Stadium, Rupp Arena for the University of Kentucky, and the 1984 Winter Olympics in Sarajevo, Yugoslavia.

When Altec moved its operations to Oklahoma in 1982, Bob stayed in Southern California and became a system consultant, working with companies like Countryman Associates, Inc. In 1983, he joined Yamaha Music Corporation as a product specialist for commercial audio products in the Combo division. Six months later, he was promoted to national sales manager for the Professional Audio Products division. In 1988, he became manager of Promotions and Advertising.

He was a member of the Audio Engineering Society, Acoustical Society of America, and the Society of Motion Picture and Television Engineers..

Whether you knew Bob as a profoundly dignified, spotlessly attired, articulate and witty personage at trade shows and professional meetings, or if you were fortunate enough to know the humorist and relaxed Bob in pressed blue jeans and plaid shirt who would rattle off the tongue twisting lines of Gilbert and Sullivan's "Modern Major-General" to test speakers and sound systems, or if you were blessed to hear him sing, he will be missed. Not only have we lost someone who made a great contribution to the audio community, we have lost a friend who taught each of us a lot about being human.

A Robert Trabue Davis Scholarship Fund has been established to benefit the Central Kentucky Youth Orchestras, 161 North Mill Street, Lexington, KY, 40507.