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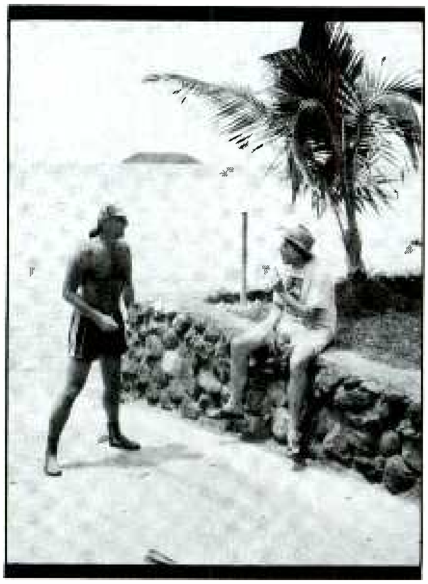

See page 7

THE SOUND CONTRACTING ENGINEER

THE RECORDING ENGINEER

AUDIO FOR THE CHURCH

THE ELECTRONIC COTTAGE

THE BROADCAST ENGINEER

ABOUT THE COVER

● Wayne Toups and Zydecajun turn it up in New Zealand. In Ed Learned's final installment on his tour through the Asia-Pacific region, he profiles Toups/Zydecajun and the Charlie Byrd Trio, and offers the troubleshooting methods he used at each venue. *Photo by Laura Bucknell.*

LETTERS

Dear Editor:

Since I have been receiving your magazine, I have enjoyed the articles and benefitted from them. I have been running my own business now since 1973 and to date, I must say that your magazine is one of the better ones that I read and receive. The articles have good tips and are full of valuable information.

The article *Promoting Your Studio: Part I by John Barilla* has to be one of the best I have ever read. Step-by-step information is given on promoting a studio and I'm sure that any studio or any business will find this article helpful. I really liked the section on Classified Ads where John pointed out to beware of production companies that will put music to song lyrics for $60.00; this is hype and we all know that.

Also, John pointed out in the section on Direct Mail that this is a very good way to advertise. It works because you can really get your services across in a letter with samples, photos of your facility, and let's not forget services that you offer. If you were to say everything that you can say in a letter in a classified ad, the cost would be very expensive.

Keep up the great work, and I really look forward to your next issue of **db Magazine**.

Keith A. Gutschwager, President

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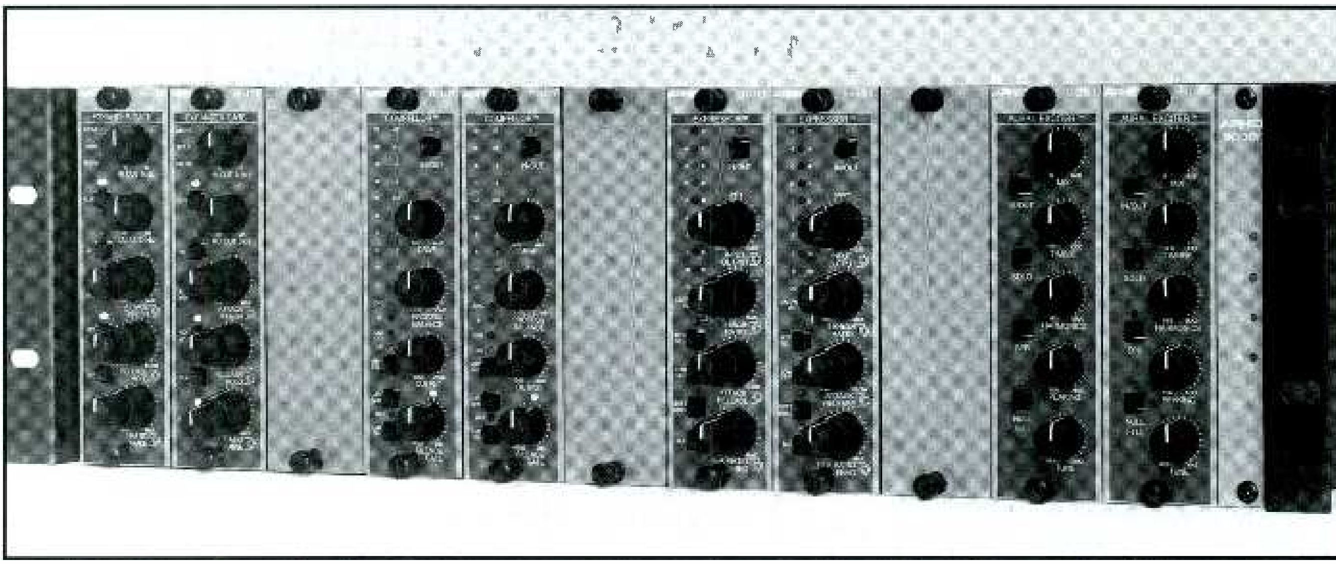
Thanks for the kind comments to Senior Editor John Barilla and to us.We will continue to strive to make each issue better than the one before.And kkep those cards and letters coming with the good and the bad.

Space Saver...



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CALENDAR

● Curt Taipale and guest instructors like Blair McNair and Tom Blake will host five-day intensive study workshops on sound reinforcement in St. Louis, MO, beginning *July 8; Orlando, FL, Aug. 19th; and Chicago, IL, Sept. 23.*

The workshops involve thirty hours of lecture and demonstrations, assigned reading and study assignments. The $295.00 registration fee includes all materials and a copy of Yamaha's Sound Reinforcement Handbook. For more information and a $45.00 savings on registration, please contact *Soundcheck Workshops at 1471 Colgate Drive, St. Charles, MO, 63303, or call (314) 946-4360.*

● Sony Corporation has announced the theme for its 1991 International "Sony Design Vision" competition, "Audio Lifestyle."

The design must incorporate an AM/FM radio, a compact disc player and be AC powered. It can also include video or other non-audio components as long as the design is primarily an audio system.The competition is open to anyone who is a student as of July 31, 1991 or who will be graduating in 1991. Entries must be original works that have not been previously presented in public. Works may be submitted by an individual or a group. The entry must be postmarked by July 31, 1991 and arrive at regional headquarters by August 15, 1991. *For more information, contact Marie Berardino at CRN, Inc. at (516) 487-5904.*

● The Sony Institute of Applied Video Technology is offering courses in post production and editing at its Hollywood, CA, and Savannah, GA, campuses.

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See the ELAR Booksource ad beginning on page 38

A/B Roll Editing is geared towards control track editors who want to shift into computerized time-code editing. The $895.00 workshop is scheduled for July 23-25 in Hollywood.

The agenda for **Mastering the Technology of Recording and Post Production** will include time code, edit control systems, signal paths, system timing, phasing and trouble-shooting, hands-on with test equipment and more. The $895.00 workshop at the Savannah campus is from Aug. 5-7.

Production Audio for Video: Fundamentals of Field Audio is an in-depth look at production audio-for-video with an emphasis on preproduction planning. This course is $695.00, and will take place from Aug. 19-21 at the Hollywood campus.

Post-Production Audio for Video will take place Aug. 21-23 in Hollywood. Fee is $895.00. This workshop will offer an in-depth examination of the post production audio-for-video process.

For registration and more information about these and other workshops, please call (213) 462-1987. Enrollment is limited.

● The National Association of Broadcasters will hold its Radio 1991 Convention September 11-14 at the San Francisco Moscone Convention Center. Sessions in sales and marketing, programming, management and technology management will begin Wednesday, Sept. 11.

Featured keynote speakers include Keith Reinhard, chairman/CEO, DDB Needham Worldwide, Inc., for sales and marketing seminars, Mario Cuomo, governor of New York State, for management sessions, and Quincy Jones, producer/director, for programming workshops.

To register for Radio 1991, please call (800) 342-2460, or for program information, please call (202) 429-5420.

ED LEARNED

Live Sound Reinforcement: Asia-Pacific 1990, Part IV

In this final installment of Ed Learned's Asia-Pacific 1990 tour he begins in Indonesia—Jakarta.

One thing you can count on when bringing equipment into Indonesia is a complete inspection of everything. Usually a notation will be made on a passport (mine in this case) linking that person's exit to the exit of said equipment. Baggage claim is medium-sized for an international airport; a new airport is being built, so things should be less congested in the future. No porters are allowed into the customs area, so the initial labor is on you.

Charlie Byrd gave two performances in the Jakarta area. The first, held in the Juliana Disco/Club at the Jakarta Hilton, was to benefit a local children's charity. No problem with endless room reverberation here: much of the room was carpeted, the walls were padded, and the ceiling had heavy acoustic treatment. Decay time was less than a half second. Our local sponsors expected us to use the disco system as our PA. This system was comprised of Meyer 500A cabinets and folded-horn sub-woofers. Front and rear ceiling cupolas contained four 500A cabinets each; the sub-woofers, placed on the ground surrounding the two dance floors, were carpeted and disguised as tables or counter space. Meyer MS-500A amplifiers provided power. The stage placement and seating arrangements, vis a vis speaker placement, called for some creative sound decisions (see *Figure 1*).

My first concern was the center seating area, which wasn't covered by direct sound from either of the disco cupolas. I elected to place a single E-V S-200 against the wall, splayed to cover this area and the entry hall only. Both the S-200 and the rear disco cupola were delayed, via my Yamaha DDL-3 multi-tap delay, to insure time coherency between these widely-spaced systems. Another consideration was the front disco cupola: the east-west-north-south speaker placement meant that at least one house PA cabinet would fire directly onto the stage, something the trio wouldn't like. The Meyer speakers were run in pairs; I suggested we unplug the single cabinet pointing at the stage. We also unplugged the sub-woofers and moved them to the side; I didn't need that kind of low end, and our sponsors needed the room for more seating. Power came from a UK-type grounded outlet on the wall stage left; 220 volts were provided. It's a good thing we moved the sub-woofers: the place was packed, with standees anywhere there was extra room. Charlie surprised everyone (including me) by opening the second half of our show with a set of Spanish waltzes, played acoustically with no accompaniment. In deference to the dead room, I used a large hall reverb on the guitar to create the sonic illusion of a warm space. The crowd of 350 listened in reverent silence, mesmerized by this beautiful music.

The second concert took place at the Gedung Kesenian (Palace of Fine Arts). This European-style opera house was built in 1820, and had recently been renovated. It held around 900 people, including three levels of wrap-around balconies. Acoustics were quite nice: reverb time was around 1.75 seconds, with even decay across all frequencies. The house PA system was located behind grills above the proscenium opening; I never was able to find out exactly what they had up there. There was no real bass response, but for what I planned to do with this group at this venue, that wasn't a problem. There was a background buzz in the system which disappeared when the house lights were turned off. AC power came from a drop far off stage right. I tied in directly, procuring 230 volts, a good ground and clean neutral. As per usual in Jakarta, voltage did hop around during the day, beginning a steady climb during the early evening. I actually had to turn down my transformer output to compensate for this rise.

Our audience was studded with jazz fans; some had come from as far away as Bandung. The group did not let them down. Drummer Chuck Redd delighted the crowd by throwing away his sticks in mid-solo, opting to play with his hands and fingers to create a hypnotic tom-tom dialogue. The Ellington medley was also well-received, as Charlie threw in several additional favorites as a bonus. After several encores, it took a farewell announcement from Charlie to end the show; the enthusiastic crowd just didn't want the trio to stop!

SURABAYA

Located near the eastern end of Java, Surabaya is one of Indonesia's major ports. Garuda Airlines operates a wide-body service to Surabaya that can easily handle heavy cases. The airport is small, but well laid out. It's a jumping-off point for Bali, so you can usually

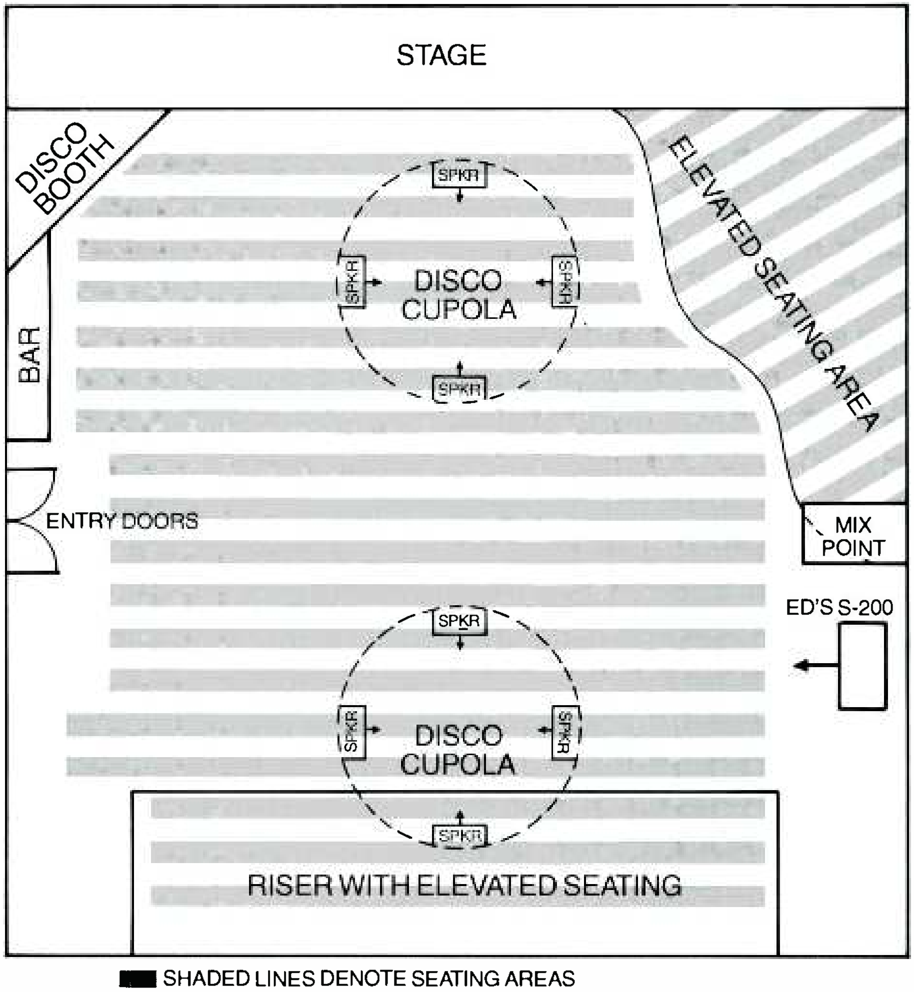


Figure 1. Jakarta. The stage placement and seating arrangements, vis a vis speaker placement, called for some creative sound decisions.

find someone who speaks English if you get in a jam.

Charlie's concert here was held in the ballroom of the Hyatt Regency hotel. I'd visited this venue with Jackson; reverb time was well over two seconds, but this time I didn't have to deal with a loud band. Power came from wall outlets, UK-type receptacles that supplied either five or fifteen amps of 220 volt, grounded power. The hotel's own voltage stabilization system insured that our AC line levels stayed right in the pocket. The PA system was provided by a local band: two single 15 in. bass bins, two double 12 in. mid cabinets, and two high frequency sections (one radial horn/driver and two piezo tweeters) per side. This was tri-amped with a Rane crossover into custom-built power amps. The high end was rippingly loud, as the crossover levels were all wide open. I backed this down to appropriate levels, balancing the system as much as I could with crossover controls. This was new to the local sound guys, who had "balanced" the system with a radical EQ curve on their graphic. The ballroom could hold 500 people and we had close to that for our evening concert.

MEDAN

We traveled to Medan from Surabaya, which meant traveling through Jakarta. Medan is a city of over one million people, located at the northern end of Sumatra. Medan's airport is not very large or sophisticated; Garuda's flight service here includes a few wide bodies, but more often utilizes smaller planes. We were again forced to send the gear out on an earlier flight to insure its timely arrival in Medan. We also emigrated from Medan. Since the departure area here is smaller, it's important to allow extra time for equipment processing and check-in, especially if you have a lot of stuff.

The University of North Sumatra sponsored our Medan concert, which was held in their brand-new auditorium. Seating 2,500, this huge venue featured a large seating area that required real 180 degree sound coverage (see *Figure 2).* The plethora of hard surfaces in this huge space contributed to a reverb time of over three seconds—not a place you'd want to mix a loud band. Charlie's trio was the first foreign musical group to ever perform here, so whatever I did would inevitably set standards. The university's PA system was a conglomeration of Peavey equipment: two different cabinets powered full-range by CS-800 and CS-400 amplifiers. Definitely not state-of-the-art, but all I had to work with.

My biggest problem was how to cover this extremely wide seating area. I used the four most similar cabinets for side and straight-ahead coverage, fed from my EQ. Two small Peavey wedges were provided for stage use; we didn't need stage monitors, so I decided to use them for center fill (see *Figure 3*). I split the PA feed so I could send the same signal to both the large cluster and the small center fill wedges. The sound guys had brought a ten-band equalizer for monitor EQ which I used on the improvised center-fill system to match levels and tone with the main system.

The crowd of 600 people that showed up for our concert looked awfully small in this huge space, but made up in enthusiasm what they lacked in numbers. My work on the PA was appreciated by the building's director, who told me we had the best sound of any musical group to date here. I didn't really think the PA system or what I'd done with it was all that outstanding, but what I thought really didn't matter in the final analysis; you do what you have to do to pull off the show to the best of your ability, with whatever gear you have.

MALAYSIA—PENANG

Bayan Lepas International Airport is about twelve miles from the city center. It's a small facility with all the modern appointments. Baggage claim is a level above the tarmac, so equipment comes up a conveyor and dumps onto circular carousels. Larger pieces have to be brought up from below by trac-

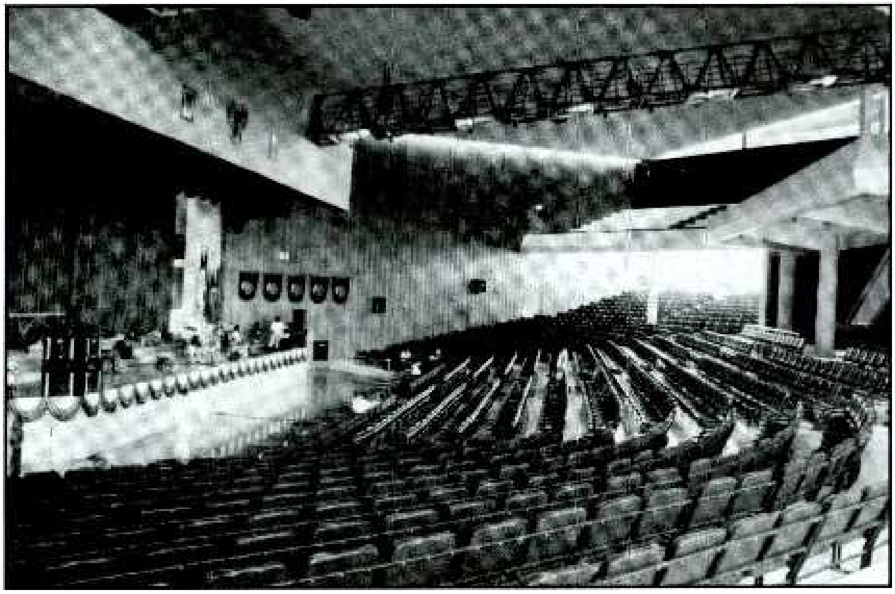


Figure 2. The University of North Sumatra Auditorium at Medan, Indonesia.

tor/flatbed baggage vehicles. We were able to back this right up to our truck for easy loading. I had a bad experience here on my last trip with truck licensing; make sure your equipment transport has the proper registration for handling cargo.

Charlie performed in the ballroom of the Shangri-La Hotel, located in the heart of Georgetown. This small room was very lively, with a reverb time of just over two seconds. The stage was set up so that we "played" to the short dimension of the room. While this helped minimize the room's reflection of sound, it did mean we had to cover a wide audience area with sound. The PA, provided by a local AV company, was a three-way component system based on two single 15 in. sub-woofers, a double 10 in. horn-loaded midrange cabinet and a 900 CD horn/driver per side, powered by two Peavey CS-800 and two CS-400 power amps. Power could be obtained from either the stage right or left rear walls; these were UK-style receptacles that supplied 230 volts. All had good grounds, and while I did observe fluctuations of five volts in either direction, they seemed to be trends, not spikes or drops. Because the stage was positioned against the rear wall, we couldn't place any speakers on the downstage corners of the stage without seriously blocking sight lines. With the PA speakers behind the band, I had a potential feedback problem to deal with. Fortunately, the room was so lively that I only had to amplify Charlie's guitar and vocal mic. Careful positioning of the mics with respect to the PA gave me the gain-before-feedback I needed without resorting to radical EQ.

KUALA LUMPUR

Subang International Airport is a large, well-maintained facility about twelve miles from Kuala Lumpur. The journey into town can take over an hour during rush hour, so leave early for your morning flights. Baggage claim is spacious, porters abound; customs clearance of equipment, both inbound and outbound, is painless IF you have a manifest and return ticket. The domestic terminal is very close to the international arrival area, which is handy if you have a quick domestic connection.

Our stay in Kuala Lumpur was very short: flying in Wednesday, Sept. 5, 1990, we performed that night, flying out the next day. The "concert" took place at the residence of Paul Cleveland, United States ambassador to Malaysia: the stage was a covered patio at the rear of the house, "playing" to a backyard audience area through open folding doors. We had a seating capacity of 175: rows of folding chairs, located next to the garden, were covered by a large tent. There was a UK-type outlet, with functional ground, on the patio which supplied 230 volts; power levels never fluctuated more than a couple of volts. My PA system was four Ross two-way cabinets, powered full-range from two Ross amplifiers. Each cabinet had a 15 in. woofer and a small horn/driver.

Acoustically, I had to deal with two different spaces; the audience was outdoors, the trio was indoors. Our whole concept of stage sound reinforcing the music wouldn't work here: at the level the trio played, hardly any sound escaped the patio area into the backyard tent. The first few rows of seats might hear the group marginally, but that was it. I positioned one pair of speakers at the stage end of the tent; the others were positioned outside the tent about halfway back, pointed in to cover both rear seating and the food/drink tables.

A diplomatic reception/concert dictates attention to volume: people expect a concert quality mix, but don't want loud levels. I had to amplify everything here to get it audible inside the tent, but I didn't want to blast the front rows in an attempt to cover the rear. I felt a distributed system, using a delayed back-fill, was the only way to go. It proved a smart decision: the tent was packed, with every seat full and standing room in the back. Thanks to the rear fill, I could do things like keep Charlie's solo acoustic pieces at the level where they sounded acoustic, knowing they could still be heard in the back. I was also moved to use reverb extensively, in an attempt to replace the "ambience" of the concert hall missing at this garden party. It was another love-in; the crowd just didn't want the band to stop! Our host had to intercede after the third encore request, ending the show in deference to our morning departure the following day. It was a satisfying evening for me, too: I finally had an opportunity to fully "mix" Charlie's music, and I savored every minute of it! I must admit a certain pleasure when the ultimate critic, Charlie, told me I'd done an excellent job.

SRI LANKA—COLOMBO

Colombo International Airport is about an hour outside of town; there are no "freeways" in Sri Lanka, and if a pothole doesn't get you, an oblivious truck driver

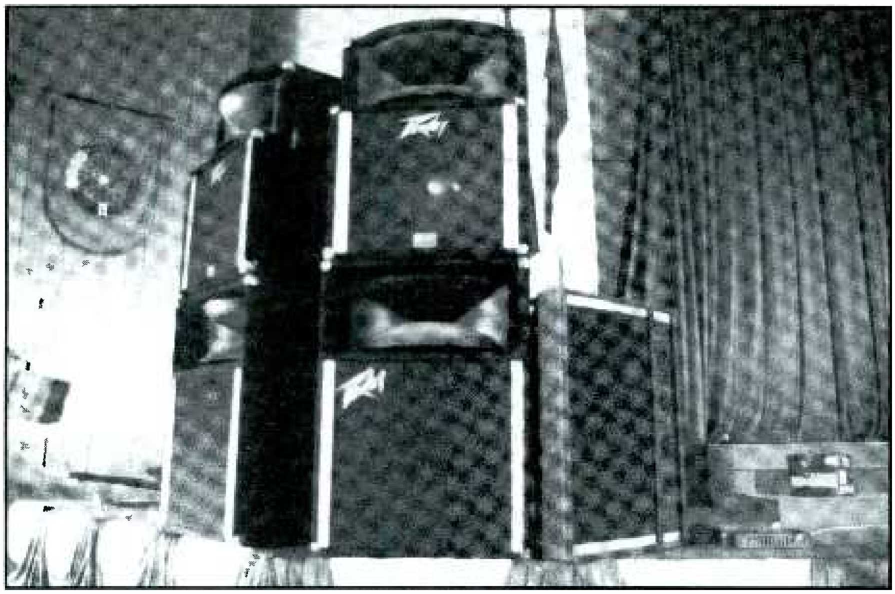


Figure 3. The speaker cluster used as center fill at Medan, Indonesia.

might. Be careful, and allow some extra transit time. Baggage claim is of decent size, but the real bottleneck is in the customs area; clearance here is much more difficult going in then going out.

Wayne Toups & Zydecajun played a single concert in Colombo. I visited our venue, the Lionel Wendt Memorial Theater, the day before our concert to check out the facility. The theater was small, seating only 500 people. The concrete floor, plaster walls and hard ceiling created a very live environment. Reverb time was over two seconds, and had that harsh "concrete" sound. I was very concerned about sound quality here, especially when the building manager told me the hall "had such good acoustics you don't need a sound system." My worst fears were confirmed: there WAS no sound system, and none had been arranged. With CAA (Cultural Affairs assistant) Ranjith Sandanayake and Berty Fonseka, assistant director of Sri Lanka's Department of Cultural Affairs, in tow, we raced through Colombo on a search for sound equipment. There really wasn't much stuff available; most systems were small, geared for vocal-only situations in hotel lounges or for program music playback. The best system we could find was a pair of Roland two-way cabinets, borrowed from John de Silva Auditorium (where I'd played with jazz saxophonist Chico Freeman in 1983). These contained a single 15 in. woofer and a small horn with a 1 in. driver; they could be run full-range or bi-amped. There was also a Roland power amp, but it put out only 150 watts/channel, nowhere near enough power for us. I decided to use only the cabinets and run them with one of my Carvers.

The band's cooperation was essential to successful shows in Sri Lanka. We would have to cut the monitor mixes from four to two, in order to free up an amp for PA use. We'd already done this once on our Asia-Pacific tour (see Cebu, **db Magazine**, March/April 1991), and everyone felt confident they could cope with it better this time. It turned out to be more difficult than we anticipated. The Roland cabinets could only handle 175 watts, and the Carvers could deliver 450 watts to each.

I knew we had to use this system in Kandy as well, so I couldn't afford to blow it up at the first gig. I used a channel of my dbx 166 for main system limiting, and also turned down the Carver outputs a couple of notches.

This seemed to keep the PA in a safe operating area. Power came from a drop off stage left that provided 220 volts. I copped my ground from a water pipe only a few feet from the drop. Voltage flipped around quite a bit during the day; the fluctuations smoothed out at night, when overall voltage would rise up to ten volts higher than during the day.

My audio fears were confirmed during sound check: the band easily overpowered the room AND the PA. My mix consisted of vocals and accordion only; I had to insist that everyone radically reduce their stage volume to give the PA a chance to be heard. Things got so bad on stage that both Pate and bassist Mark Miller complained they couldn't hear themselves play. The live acoustics and insufficient PA made this a necessity which the band grudgingly accepted. Their spirits improved immensely at showtime, as a capacity crowd responded enthusiastically to the band's music. It sounded wimpy to us, but must have been awesome to Sri Lankan ears: I saw some older audience members with fingers in their ears and big smiles on their faces as they watched the band. The younger generation wasted no time in turning the front aisle into a dance floor.

We were scheduled to tape a performance for Rupavahini, the national television network, the following day. To make things easier, Toups & Zydecajun elected to lip-synch the performance, using five songs from the band's "Blast From The Bayou" recording as program. A stage had been constructed for us at one end of Studio C; we set up all of our equipment to insure a realistic "look," although we didn't actually power anything up. The band did require a high powered monitor system, in order to better hear and "feel" the playback for their pantomimed performance. There were two JBL 4412 studio monitors on rolling carts for this purpose, but these couldn't generate the kind of level the group desired. We ended up using our own monitor system, run as a single mix and fed from the studio's audio console. I got AC power from a wall on the stage right side of the studio which was an Indian-style five amp grounded receptacle (see *Figure 1* in the May-June issue) that supplied 230 volts. It took around five hours to tape the half hour show, as the Sri Lankan director elected to rehearse camera moves several times for each song.

KANDY

The drive to Kandy took about five hours, giving us a closer look at the interior of this lush, tropical island. We made good time on the

Figure 4. Wayne Toups in performance at the Gluepot, Ponsonby, Aukland, New Zealand. Photo by Laura Bucknell. (This is also seen in its dramatic color on our front cover.)

open road, but the numerous small towns and villages along the way slowed us back down. We soon began the winding climb into the mountains, which, depending on the vehicle, can slow you down even more! Kandy, at an elevation of over 2,000 feet, is substantially cooler and drier than Colombo, bringing welcome relief from the sticky, tropical heat of the coast. It gets cool at night, and we were glad for the sweatshirts we'd brought along.

Toups' concert here was at the auditorium of the Kandy Public Library. This second floor facility did not have an elevator, so equipment was carried up a rear staircase. This long rectangular room seated 300 and the stage "played" to the long dimension of the room.

The floor was tile, but the walls were padded and the ceiling had an acoustic tile treatment, so reverb time was a surprisingly low 1.5 seconds. This really worked to our advantage; the room soaked up enough stage sound so the group could play at more comfortable levels (for them) without blowing the PA away.

The band visited a local girls school while I was setting up; they enjoyed a demonstration of Kandyan dancing, and took the opportunity to invite everyone to the show. This helped pack the house for our evening concert; the girls demonstrated they were also adept at Western dancing, as the front and side aisles became dance floors. Toups and drummer

Mike Burch prefaced our usual encore with an acoustic improvisation featuring accordion, Kandyan drum and Sri Lankan ankle bells. It was the hit of the evening.

NEW ZEALAND—AUCKLAND

Auckland is the usual port of entry for groups coming from the United States, although Christchurch also handles international flights. Many American groups have visited New Zealand over the years, so bands passing through are commonplace for customs officials; clearance of gear in either direction is painless. There are no porters allowed in the secured international arrival area, so make sure to grab luggage carts before they're all gone. The airport is about a forty minute drive from downtown. My past tours of New Zealand made it clear that quality PA and monitor systems were available in most major cities; there are several PA companies that cater to large entertainment events. We elected to ship our sound equipment to Fiji, the next destination after New Zealand, and contract complete sound systems locally. We would travel with microphones, transformer and the band's stage gear only.

Toups & Zydecajun gave two performances on consecutive evenings at the Gluepot, a local club, as part of the Ponsonby Blues & Jazz Festival. The room, which seated around 600, was rectangular, with a proscenium stage "playing" to the long dimension of the room. Stage power was obtained from an Australian-type receptacle (see *Figure 1* in the May-June issue) behind the stage right PA stack; this provided 225 volts, with a good ground. The club's PA system was a combination of JBL folded horn sub-woofers and Renkus-Heinz mid-highs, powered by custom amplifiers generating 5,000 watts of total system power. Three monitor mixes, run from the house desk, operated five wedges and a drum monitor, with a total power of 2,500 watts. I assigned two wedges on a mix to Toups, three wedges on a mix to guitar, keys and bass. Effects included an SPX 90 and a Roland 555 space echo. The low ceiling and carpeting reduced reverb time to around a second; sound check went smoothly.

I was ready for a great night, but received a nasty surprise as one of the four submasters on the house desk died as the band hit the stage. I had to do some frantic scrambling during the first two songs to re-mix the group on three subs. We had about two-thirds of a house, but they went absolutely nuts over the group, clapping, screaming and dancing like crazy! Our show the following night was pandemonium: the word was out about Toups & Zydecajun, and the place was packed; the club had to turn people away at the door. A local blues duo, the Ebeling Brothers, opened for us, and I mixed them as well. The reaction of this crazy crowd was unbelievable; people were dancing anywhere they could find space, even on top of tables! "Zydecajun Train" saw the audience form a train and snake through the crowded bar, while Toups' touching vocal on "Tell It Like It Is" inspired slow dancing and swaying from side-to-side (see *Figure 4*). Four encores were required, including incendiary versions of Stevie Wonder's "Superstition" and the old R & B classic "Standing On Shaky Ground," both featuring Pate on lead vocals.

CHRISTCHURCH

Our venue here was the Hagley Room of the Carlton Hotel: the stage was set to "play" the short dimension of this rectangular multi-purpose room. Capacity was officially 450, although it could handle (illegally) 750 standees.

Quite a few visiting blues bands had played this room, so our sponsor, the Christchurch Town Hall, felt we'd have a built-in audience. Stage placement was complicated by a permanent disco booth, located upstage center; we had to set up around this, which meant moving the drums to the upstage right corner of the stage.

I pulled our stage power from an outlet just stage left of the disco booth. This Aussie-type outlet supplied 225 volts, with a functional ground.

The PA system here was a JBL concert series system: one sub-woofer and one mid/high cabinet per side, powered by Carver PM-1.5 amps. The hall was really too wide for coverage by a single cabinet, but that's all we had; I elected to cover the area in front of the band, leaving things soft on the sides. I mixed monitors from the house; the Soundtracs 32-channel console had four monitor sends, which I assigned in our usual manner. There were only four matched wedges, each with a JBL E120 and bullet tweeter. These were designated: two for Toups, one for guitar, one for keys. There was a large monitor with two 15-in. speakers and a horn/driver, for drums; I asked for, and got, a different wedge for Miller. An SPX-90 and Roland SDE-1000 were provided for effects. Word-of-mouth reports on the group had obviously reached Christchurch; the place was packed for our evening show, and the audience was ready to dance the night away. The party atmosphere continued in the bar after the show, as both band members and fans mingled over food, drink and good times.

Figure 5. The author (right) programs effects and systems engineer Andy Craig looks on. This is at James Caberet, Wellington, NZ. Photo by Laura Bucknell.

WELLINGTON

Our performance in New Zealand's capital city was held at the James Cabaret, a sparkling new club with a capacity of 500. Reverb time here was just under two seconds, due to all the chrome, glass and tile surfaces. Power for our stage gear came off the upstage wall; an Australian-type grounded receptacle supplied 220 volts of stable power. My PA system was provided by Oak Park Productions: per side, there were two single 18 in. JBL W-bin sub-woofers and two AP-30 (two 15 in., two 10 in., one 900 CD horn/2 in. driver) cabinets. System engineer Andy Craig gave me a Yamaha 2404 house console, Brooke-Siren crossovers and JBL 5547A graphics for house drive. Two dbx 161 and one dbx 166 limiters were available for submaster or channel patching; Yamaha REV-7 and SPX-90 reverbs and a D1500 delay were available for effects (see *Figure 5*). Monitors were mixed from the stage on a custom 24 X 8 console and JBL 5547A equalizers; five bi-amped wedges with JBL 15 in. woofers and horn/1 in. driver were provided. A drum fill cabinet, with two 15 in. woofers and the same horn/driver combination, satisfied our drum monitoring needs.

Oak Park's crew was very professional: the band's monitor needs were handled with alacrity and courtesy. Sound check progressed quickly, so we were able to enjoy a relaxed afternoon in Wellington. The concert? Let's say I'm glad the fire marshall didn't show up! Seven hundred people were crammed into the club; you couldn't have shoe-horned anyone else in. It took five minutes to walk the 75 foot from the mix point to the dressing room. The audience was ready to *bon temps roulet*, and the band did not disappoint; it was another mass dance party. Promoter Graeme Nesbitt was impressed by this response from the supposedly blase' Wellington audience. Plans are now being made to bring the band back for an extended New Zealand tour in the near future.

FIJI

Nadi International Airport is the sole point of entry for international air travel to Fiji. Nadi town doesn't really offer too much in the way of sights; it's primarily a jumping-off point for travel to Fiji's numerous off-shore resorts. There are numerous hotels to overnight in before beginning travel around the country, which is what we did. The baggage claim area is large; customs clearance for equipment depends on when you arrive. Late arrivals are the best for minimum hassle. Ex-

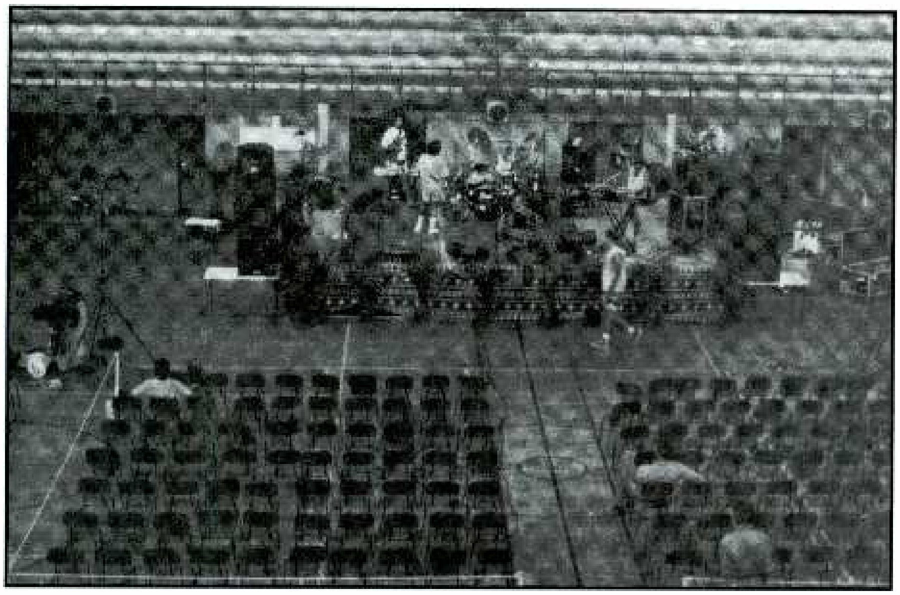


Figure 6. Sound check at the Suva National Gym, Suva, Fiji. Photo by Laura Bucknell.

pect a more thorough inspection on the way in than on the way out.

SUVA

The drive from Nadi to Suva takes about three and a half hours on the Queen's Road; it is a beautiful drive, especially where the road skirts the Coral Coast. You can fly from Nadi to Suva (Air Pacific operates daily flights), but forget it if you have large baggage or equipment. The planes are twin prop Fokker Friendships that can barely handle people, let alone large amounts of baggage; that meant ground travel for Toups & Zydecajun was a given.

Our concert was at the Suva National Gymnasium, which had the acoustics of one: reverb time was a horrible three and a half seconds—and this with the portable stage "playing" to the short dimension of the room! The PA system, which we would use throughout Fiji, was rented from a local music store. It was comprised of one Peavey SP-4 and one SP-1 per side, powered by two Yamaha 2250 power amps. That was not a lot of PA for this huge facility, which seated 3,000 (see *Figure 6*). Power came from a receptacle in the hall behind the stage, supplying 240 volts on an Aussie-type receptacle, and the ground was good.

I had to ask the group to cut back on their stage volume in the interest of clarity; it helped somewhat, but only a capacity crowd could have saved us. We weren't that lucky—a little under 1,000 people showed up. Fijian crowds are reluctant to try something new, so that type of turnout was considered a success by our sponsors. We even had a few dancers, which Fijians told us was quite extraordinary.

SIGATOKA

The de facto capital of the Coral Coast, Sigatoka itself is not very big; Toups' concert here was the first ever by a visiting American group. No one really knew what to expect, so a small venue was selected to be safe. The Angel Theater, in the center of Sigatoka across from the Town Council building, held about 800 people, including a small balcony. Reverb time was around two seconds and power came from an Aussie-style outlet stage left, by the access stairs to the stage. It supplied 230 volts, with a good ground. The equipment had to be carried up a wide flight of stairs to this second-floor facility.

Sigatoka was bustling on this major market day; the sound of the band at its sound check attracted the curious, who told their friends. When Toups & Zydecajun hit the stage for our afternoon concert, every one of the 800 seats was full, with standees filling the rear aisle.

...the Girmit Center, could hold up to 800 people in rows of folding chairs. Reverb time was almost two and one half seconds, due to the plethora of concrete and brick surfaces.

By now, we were well aware of the passiveness of Fijian audiences, but this one certainly didn't fit the stereotype. People were dancing up a storm everywhere, guys were throwing their hats into the air, and girls were rushing the stage to give flowers and kisses to the band! It was a resounding success, evidenced by the fact that the mayor of Sigatoka, who had a front-row seat, invited us all to dinner at the Fijian Resort the following night. He and the Fijian's general manager lobbied hard for an outdoor concert in Sigatoka next time!

LAUTOKA

This was the last concert for Toups & Zydecajun; for me, it marked the end of my three month tour of the region. Lautoka is often called "Sugar City"; it is home to the biggest sugar refining plant in Fiji. Our venue, the Girmit Center, could hold up to 800 people in rows of folding chairs. Reverb time was almost two and one half seconds, due to the plethora of concrete and brick surfaces. Power came from Aussie-style outlets on the stage left wall which provided 225 volts, but only two out of the four outlets on this wall had functional grounds.

Unlike the last two venues, there was no stage lighting to speak of; the band performed under the dim glow of fluorescent fixtures. About 500 people showed up, mostly young people who'd heard the music and were curious as to what was going on! Without question, they were our most conservative audience; the audience did clap and move to the music, but no one danced, even when Toups and Pate jumped into the audience and tried to stir things up. db

BRENT HARSHBARGER

Audio For The Church: Wireless Mic Systems

● Churches have gone great guns in the past few years purchasing wireless devices such as microphones and hard of hearing systems. The wireless technology uses either infrared or radio frequency (RF) transmitters and receivers. Infrared, which is usually used with hard of hearing systems, has good sound quality and does not interfere with mics using RF. The only drawback to infrared is that it is strictly line-of-sight technology. In other words, the transmitter and the receiver cannot have anything blocking the path in between them, or the signal will go away. There are many successful installations of these type systems in large and small venues. It's well worth checking these systems out if you're in the market for a hard of hearing system, but for this segment, the topic will be based on mic systems.

Many people may think wireless mics are all created equally, but this is not so. RF products have not typically been a strong suit for many audio operators and technicians, so if you are not familiar with the technology, it's the same all around. As a result, many churches are using what I call second class wireless mics. I say this because the audio quality is very poor, can reduce intelligibility, and has a less than dynamic performance from singers.

The buzz words of wireless systems are diversity, companding, range and dynamic range. We will discuss all of these to some degree, but first a little background on RF.

BASIC RF

Most wireless systems today are in the VHF (Very High Frequency) spectrum, which consists of the frequencies ranging from 30 to 300 MHz. Wireless mics are generally in the 160 MHz to 220 MHz range, in round figures. With the recent increase in sales of wireless systems, it becomes more difficult to select unoccupied frequencies or harmonic interference from wireless mics close in frequency. As a result, UHF (Ultra High Frequency) systems are rapidly gaining popularity.

RF consists of an information signal, in this case audio, and a carrier signal, which is a particular frequency in the range mentioned above. These two frequencies are combined together (or modulated) to become a "complete" RF signal ready to be transmitted and intercepted by a receiver and demodulated, just like a commercial FM radio station. In fact, the frequencies are mini FM radio stations.

Like sound system design, the mic is the first link in the system, and its ability to reproduce is the key to the rest of the system sounding good. Therefore, selection should be carefully thought out. Will the mic be used for speech or a multi-purpose application? Will it be used in average or high sound pressure level areas? These questions need to be asked any time you select a mic. Questions concerning sensitivity and frequency response also need to be asked.

TRANSMITTERS

The transmitter section is next, and in the case of a wireless system, I consider it a part of the mic, because purchasing a good mic with a poor transmitter will give the same or worse results as selecting a poor mic with a good transmitter.

Transmitters come in two different styles. The handheld transmitter has the transmitting electronics and antenna in or at the end of the barrel of the mic. The lavalier, also referred to as a body mic, has its antenna hanging from the transmitter, and on inexpensive units, the antenna shares the cable with the mic.

Regardless of the style of wireless system used, audio quality is what counts. The selectivity of the transmitter is of great importance, especially when multiple wireless systems are used.

Wireless transmitters, like radio stations, use compression to gain the best audio signal-to-noise ratio before being modulated and transmitted. Although today most systems use companders (a device not only used as a compressor, but also does downward expansion—commonly called an expander) to get the best signal-to-noise ratio, whether the input signal is strong or weak, it will expand or compress the signal as needed. This audio processing is good when proper design and components are used. If not, it can greatly degrade speech intelligibility.

Two good tests to perform are to take a wired mic, which is the same mic on the head of the transmitter, and compare the quality between them. The one which sounds closest to the wired mic is the one to be selected for the next test: take your keys and shake them in front of the mic. If you hear funny sounds, or the audio quality and noise floor changes dramatically, try another wireless system that is a little more expensive.

THE RECEIVER

The receiver is similar to the receiver in your home or car radio. The signal comes in via the antenna and is then demodulated, meaning the RF carrier frequency is separated from the audio signal. This simple single antenna system is referred to as a non-diversity system.

Diversity systems, in my opinion, are the only option for purchasing a wireless system. Nevertheless, I will give you the facts so you can decide which system is best for you. Non-diversity systems are less expensive compared to their diversity counterpart, and if you will be using the system in the same location at all times (both transmitter and receiver, without audible drop-outs or swishing sounds), then chances are that a non-diversity will work best and still leave money for new cables and connectors.

What does diversity do? A diversity system is technique which increases the quality of RF systems in adverse conditions. Remember that the audio signal is merged with a carrier signal, the radio frequency section, so therefore, if the carrier signal is degraded, then the audio signal will also suffer. RF is not excluded from the effects of wave cancellations, which are commonly called multi-path cancellations. These multi-paths are the cause of drop outs and other noises, hence the reason for the creation of diversity.

DIVERSITY SYSTEMS

There are three types of diversity systems: multi-antenna, phase and multi-receiver.

Multi-antenna diversity systems are not used as much today as they once were, so phase diversity and dual receiver diversity will be discussed in greater detail. First, a brief explanation of multi-antenna diversity, or antenna diversity. Antenna diversity uses three antennas together, each isolated by an isolation amplifier and coupled to a summing network. Simply, the amplitude and phase are summed together to make a single signal.

Most of the problems people have when using wireless systems are self-inflicted because they do not read the owner's manual

The theory is that if two antennas are 180 degrees out of phase of each other, the third antenna signal will remain balancing out, making a strong signal.

Phase diversity is very popular. It uses two antennas, but actually only has one receiver. Phase diversity looks at both antennas and uses a summing technique similar to the multi-antenna system, but it also uses a special electronic circuit, instead of a third antenna, to switch the phase of the signal from an antenna when the control circuitry in the receiver detects a drop in signal level.

Multi-receiver diversity is the best approach to diversity systems, but the best in most cases, as in this one, is always more expensive. Multi-receiver diversity systems employ two receivers and antennas in a single unit, which uses a switch circuit similar in principle to the one used in the phase diversity system. Multi-receiver systems' switch circuitry is on the output instead of the RF output like the phase diversity, therefore, it is switching audio quality instead of RF quality.

For those who may be saying, "if you have poor RF then the audio will be poor as well," let me ask this: if the RF in one antenna drops enough for the antenna to switch, and then goes to the other antenna only to find the RF worse, wouldn't it be better for it to switch if the audio quality is degraded instead of just a drop in RF? The cost between multi-receiver and phase diversity is not night and day, but it is enough for someone to look harder at the differences in the two systems. Many people have found that the disadvantages of phase diversity versus multi-antenna diversity are distinguished by price, making phase diversity very popular in the church community, since the boards are not technical, and purchase is based on price.

Range is one of the first things that people ask about when looking to buy a wireless mic, but as you can see from the topics discussed so far, it is not high on my priority list for a quality mic. Most of today's wireless systems work about the same range, and besides, where you use the mic greatly determines the range of the system. It's really a variable.

Most of the problems people have when using wireless systems are self-inflicted because they do not read the owner's manual. I'm as guilty as anyone, but when using a new type of equipment, the best place for information is in the owner's manual. Wireless systems are more than mics used without wires; they are mini-radio stations using RF technology, which have a new set of rules to be learned, and by learning them, you will have better luck with your existing system, or have a better understanding of what you should be purchasing for your church. db

TERRI FIYALKO

SPARS Shootout— The Last Word in Digital Audio Workstations

During the weekend of May 18-19, 1991, the Society of Professional Audio Recording Services presented their Business / Technical Conference at the Orlando Penta Hotel in Florida showcasing ten digital audio workstations (DAWs) developed by their membership manufacturers. This was the third such "shoot-out," occurring annually since its success in Chicago in 1989, and Nashville in 1990.

HOSTED BY SPARS EXECUTIVE Director, Shirley Kaye, and current President, Pete Caldwell, the one hundred and fifty registrants, members of the press and contributing manufacturer representatives, were treated to a plethora of sophisticated software and hardware features, demonstrated with the aid of two large projection screens focusing on keyboard/ mouse movements and the computer screen display.

The impressive array of DAWs, strategically lining the walls of the banquet hall, were easily accessible for a one-on-one trial at various times throughout the two-day seminar. Altogether, it was a mind-boggling amount of information to be absorbed in such a short time period. But, thanks to exceptional planning, the conference was flawless and experienced no technical difficulties. SPARS and staffers responsible for the slick technical set-up (including Total Audio Visual, Meyer's Speakers, Sony, and Joe Neil from Atlanta's Doppler Studios acting as consultant) should be highly commended for a four-star effort to keep us apprised of the advantages and latest developments of this ever-growing technology.

With the vast amount of computer-based editors emerging on the market, it comes as no surprise that facilities and editors have embraced the technology with hesitance. It has little to do with the never-ending debate of sonic superiority between digital and analog, nor the question of whether manipulating audio in the digital domain is more time and cost-effective than traditional analog methods. Rather, the issues are a fear of investing a substantial amount of money in a system that will quickly be outdated, the stability and support of the manufacturers, and the ability to interchange data between the systems.

If you are shopping, trying to keep up with what's available this week, or want to put your hands on the system controller yourself, and you don't want to deal with the massive insanity abundant at the trade conventions, the place to be is the SPARS Business/Technical Conference "shoot-out." Manufacturers join together to tout the latest software versions and give us a peek into upcoming developments.

The privilege of writing this article invokes a powerful need to express a candid opinion to those of you hesitant with this technology. As an editor, I have spent a vast amount of time, and more projects than I care to remember, editing on a DAW. You would have to drag me back to analog editing methods kicking, screaming, biting and fighting. All the systems I have seen have tremendous editing attributes, but there is one out there that will suit your individual needs more than others.

Yes, there is always room for improvement. But in order for these improvements to come about, the systems must be used and abused by those of us in the production world. Our feedback to the manufacturers is invaluable, and our support of their efforts vital. Even in their infant form, these systems are ready to make you money by saving time, and increasing your creativity by their powerful features. Okay, so they aren't as interchangeable as a two-inch tape yet. But if you have need to get data from one type of system to another, let's not forget about traditional methods. (Remember digital multitracks?) Hurdles have never stopped production before, and because they exist is no reason not to enter the race.

For those interested in trying out a DAW, all the manufacturers will try to arrange a trial/demo with you on a case-by-case basis, regardless of your location in the country. Whether you are a small studio handling local advertising spots or a large post production facility, I urge you to contact the sales representatives and set up a viable demo.

You may breathe a sigh of relief because I'm done with my opinions. I have kept my synopsis of the conference brief with a minimum of details. Instead, I prefer to let you hear the views of five professionals who attended the show.

DAY ONE:

It was a shaky start for most of us. Eight-thirty came early and it was a long day ahead. But thanks to ample coffee and a sample of the palate-pleasing food we were treated to for the duration, we settled in for a day of hour-long demos prepared by Otari, Studer/Dyaxis, AMS, New England Digital, Solid State Logic, WaveFrame, Roland, Akai, Digidesign and Lexicon.

All demos were quite polished and well-presented with a main orator addressing the group and an assistant performing on the system. Otari built a thirty-second spot on the DDR-10 (which utilizes a Sound Tools front end), and proudly showed its latest addition to their line, the Pro Disk 464. One of the digital audio workstations "old-timers," AMS AudioFile, walked us through the menus, gave examples of editing techniques, and filled us in on the upcoming ADR software. Studer Dyaxis' Gerry Kirby and Matt Ward did a smooth job demonstrating features in their latest software. Although the Dyaxis system is also no newcomer, the benefit of the merge with Studer is apparent.

New England Digital's demo of the Post Pro started out just the way they started out—with music—created on a Synclavier. Highlights included their new software for ADR and DSP developments. Roland, although a new entry in DAWs, was present and turning a lot of heads with the DM-80 multitrack hard disk recorder. Although the software wasn't completed for demonstration, menu screens, graphics and a well-informed staff gave us a preview of the near future.

Digidesign concentrated on showing Pro Tools, DEC automation and announcing their new products, Pro Store One, Pro Store Optical and a CD recorder. Suz Howell's performance on Pro Tools made you feel the system could be mastered in moments.

Akai's Ron Franklin was equally slick on the DD-1000 utilizing MO (Magneto Optical) drive. Akai also premiered its newest sampler, the S-1100, with an impressive demonstration of its time compression and expansion capability.

Certainly Lexicon's Opus is one of the most aesthetically beautiful workstations. Celebrating its twentieth year in marketing digital products, Lexicon is no newcomer to binary technology. For fear of being redundant on the editing features of Opus, they concentrated on the digital console's extensive automation.

We were all entertained by an unusual and deliciously risky approach taken by Solid State Logic for their demo of ScreenSound. Just prior to lunch, we were informed that they had no idea what their demo material would be. During the break, Solid State Logic recorded random material from the Central Florida cable system, selecting a thirty-second spot promoting vacationing in Tampa. Running on the basis that all audio for this spot had been destroyed, they assumed the position of the "lucky" post production house selected to lay in music, voice-overs and sound effects—all within thirty minutes. (An ambitious undertaking at best.) Although they did not finish the entire spot in the twenty-eight minutes utilized, we all derived a working knowledge on using the system to this purpose, and quite a few laughs creating sound effects from the library available.

Figure 1. The event achieved a good-sized attendance for each of the presentations.

It was unfortunate that WaveFrame's demonstration of the Audio Frame had to abort early due to some temperamental behavior caused by damage sustained to the mainframe during shipping. As the demonstrator and a loyal user, Michael Bard of Newton-Bard Music in Portland, OR, should be commended for his personal testimonial and performance as a "song-and-dance" man put "on the spot."

By the time the last demonstration ended, we were all in dire need of a break. Following wining and dining, the manufacturers met with enthusiastic attendees for more individualized attention.

DAY TWO:

By this time the attendees were alert and armed with questions for the panel comprised of John Carey, Otari; Curtis Chan, Roland; John Gluck, AMS; Tore Nordahl and Dave Bowman, Studer Dyaxis; Piers Plaskitt, SSL; Mack Leatherby, Lexicon; Ron Franklin, Akai; Ted Pine, NED; Steve Krampf and Peter Gotcher, Digidesign; and Bill Hughes, WaveFrame.

Following opening statements by moderator Dick Trump, each manufacturer gave us insight to where their product was headed in long term development, and pledges of cooperation between all for making data exchange a reality. Tom Scott of Skywalker Ranch in California put forth his opinions, both pro and con, on setting standards. His was a sensitive position, wearing the two hats of a facility and software developer.

In answer to the widespread concern over data interchange, The Audio Engineering Society formed a committee to act as liaison between the various manufacturers. Bill Hogan, chairman of this committee, was present to brief us on the technical aspects involved in creating these standards.

Then it was the attendees turn to speak. A variety of questions were fired at the panel, but the most recurring seemed to involve measurements of the systems to guarantee the fidelity of digital sound. (Isn't this where we started in the early eighties?) Although it was a stimulating, and occasionally heated, discussion, it was left unresolved. After all, such a subjective opinion can only be formed with common source material in a familiar environment. Sounds like a good reason to try one out in your own studio, no?

FROM THE "MAN ON THE STREET"

Jay Palmer, chief engineer

20th Century Fox Film Corporation

Los Angeles, CA

db: What kind of work do you do in your facility?

JP: We do audio mixing, primarily for film and television.

db: Prelay work, too?

Figure 2. The NED demo. Note the large screens up front which were available for each demo.

JP: No, we're looking to get into that.

db: Do you have any DAWs?

JP: No, we've been watching this the last few years. We use an independent, Harry Snodgrass, who uses a Digidesign.

db: Are you here shopping or for education, to see where they are now?

JP: Both.

db: What did you think of the conference?

JP: It seemed that all the manufacturers (that) were serious about doing business, serious about supporting the professional industry, were here, and (were) represented quite well. It was very good.

db: Did anything stand out for you personally?

JP: The presentation of the New England Digital system was very impressive. The Roland system, being brand new, holds promise. I was pretty impressed with that for a newcomer. Akai's system was also impressive for a fairly new system. The different ranges of all the systems was quite interesting. If you could combine all the aspects of all the systems, you would have the perfect system. But in fact, they all have their own personalities, their own approaches. It was very interesting to see how they all worked.

db: In respect to the conference, did you think that this was a good way to see the systems? Any improvements?

JP: It was pretty much an assault the first day. Personally, I would rather see (SPARS) split up the presentation into two days. You find that by the time the afternoon rolls around, you've pretty much been overloaded—data overload.

db: Are there any improvements in the technology that you personally would like to see?

JP: We're all headed to the tapeless studio, the tapeless editing, the networked sound department, the editorial networked to the stages. Some people (manufacturers) have actually played with this a little bit. But, until we can get megatrack capability, until we can actually have an unlimited amount of tracks so that we can supply dubbing stages, theaters, video sweetening houses with 96, 120, 156 tracks simultaneously, and yet have them be on line with an editing room—we're not there yet. And if we can feed all that to a digital console, then this is what it's all about. Hopefully, all these manufacturers can stay financially sound enough to get to that point. The competition is the tough part for all of them.

db: Do you think that setting standards could prohibit the growth of what you want it to be—a studio of megatracks?

JP: No. (The manufacturers) have to play ball with one another. The reason film editing is still around is because it's the only standard in the world. Period. Any film editor can go to any country in the world, cut a piece of film, put it into their mix—whether it's sound or picture—and work. That's why film hasn't really gone away—because it is the only standard. And to imagine that there's anything else that is a standard is not dealing with the actual fact.

Ron Lagerlof, operations manager

Skywalker Sound

San Rafael, CA

db: Although most know Skywalker Sound, how would you describe your facility?

RL: Primarily film post production, sound design and mixing.

db: Obviously this is not your first exposure to DAWs. Why are you here?

RL: We have a dual purpose here. We're developing software for the (New England Digital) Post Pro SD. The SoundDroid software allows the operator to work in more of a film style cue sheet like they are used to working on moviola and 35mm film. It gives them more of an operator interface that they can relate to so they are not so intimidated by the hardware. We use Synclaviers extensively for sound design. We're definitely not new to digital technology, but we also have an interest in getting more into that so we can operate our facilities more efficiently. Instead of using forty to fifty editors on a film, maybe we can do it with ten or so.

db: So you wanted to see what others are doing? A collection of ideas?

RL: Exactly.

db: What did you think of the overall presentation of the conference?

RL: For our purposes, I thought it was run very well, very efficiently. There wasn't a whole lot of "mud slinging" amongst the competition. It was all generally an open forum which is good for this sort of thing. What needs to come out of it are more discussions about interchange and compatibility between the systems.

db: Did any of the systems or presentations stand out to you personally?

RL: Several did for different reasons. I don't see any of the systems that are particularly THE system to use for every aspect of the process in the way that we work. But I see several of them that can be used as a way of getting digital technology into the way we work, helping expedite things more efficiently. In that sense, there were quite a few of them that were really interesting—seeing the Akai thing with the removeable medium. A rack of those could maybe take the place of ten or twelve dubbers. Sound Tools for sound effects libraries, putting sound effects that we normally put on 1/4 in. tape and archiving them to CD might be another way of merging this into our business. But I don't think that there's a single workstation that can do everything that we can start doing feature film work on from beginning to end.

db: Do you have any opinions on changes and improvements overall in the technology? And do you think that standards could prohibit this growth?

RL: I don't think standards will necessarily prohibit any sort of growth for this industry. I think it can only help. We have a use for this in our own facility, disk file interchange. It's just now becoming somewhat feasible to do the big jobs on a digital workstation. It's not totally affordable, but it's coming down. It's getting to the point where it will be affordable. When it becomes affordable, and it does allow a platform to do a full motion picture, for instance, that's the time (for us) to start really getting serious about buying into it. We still do a lot of work on 35mm film with Dolby SR and it sounds great. There's a lot to be said for that.

Les Bateman, technical director/chief engineer
Sounds Interchange Ltd.
Toronto, Canada

db: What kind of work does Sounds Interchange do?

LB: We do basically everything from platinum albums to television productions, a lot of commercial work, documentaries, soundtracks for film. Everything to do with recording. We do everything from the beginning to end at our place.

db: Do you have any DAWs?

LB: Yes, we have a DDR-10, two ScreenSounds and the networking system for it.

db: So why are you here? Shopping? Catching up?

LB: I didn't go to the NAB because of the new facility we're building. I came down here to give (the manufacturers) a few shots about some of this wonderful stuff that they build that still doesn't do everything that it's supposed to do.

db: What did you think of the conference?

LB: I've been researching this for the last couple of years and I've had access to almost all the boxes here, except the Pro Disk 464. That's about the only one that I haven't tried at our facility—and a pile of the other ones. But if somebody was shopping for any of that type of equipment, it would have been well worth their money to have come down here and see what's really going on. I've already bought, but I also learned a lot about systems here. They're all right there, and there aren't fifty thousand people standing around every booth. It was real easy walking around to any of the booths and finding out all the information firsthand. Still, it wouldn't duplicate making (the manufacturers) actually bring you the system. I don't buy anything anymore unless they bring it to me and we try it, for real, in our own environment.

Figure 3. This was the SSL demo. They used live impromptu voice-over talent.

db: Did any of the demos or systems stand out to you?

LB: I was impressed with the Pro Disk 464. I told (Otari) to bring me one of those to try. I was impressed with the new (New England Digital) EditView software and the autoconform because I actually got to see it working. I don't have that stuff on our Synclavier. I'm going to try that out. We never used the Synclavier as a post production tool other than doing music tracks.

db: Do you have any opinions about changes and improvements overall in the technology? Do you think standards would prohibit that?

LB: One of the things that they have to look at, (or) what I've noticed is all the digital recording mediums, whether DASH format or hard disk recorders, when you transfer into the digital domain from an analog source, because of the delays from the converters, it's not in sync to the original tracks. You can't get it perfectly, phase coherently in sync with the stuff you're putting on your digital multi-track tape or your hard disk editing system. So to me, that's a problem and nobody's really addressed that. The other thing that drives me nuts is the AES/EBU standard; it only does two channels at a time. Since that's the standard, you can't blame the hard disk recorders (manufacturers). There's no (digital multi-track) tape machine made in the world that you can take the stuff that you put on your workstation and transfer (more than two channels at a time) to your digital multi-track. It doesn't exist yet. I think somebody missed the boat on all that. Studer has gone as far as to actually have the connector on the back of their 48-track machine for the MADI interface. I really think that they should get off the pot. To keep everything in the digital domain makes a lot of sense. You need some type of multiplex digital way of sending the signal back and forth. Right now, it's almost burning the clock. If you want to make a clone of the 48-track (unless they're sitting side-by-side), you have to hook up the 96 cords it takes and transfer to analog to get it back into the digital machine. So you're not even making a proper clone.

Gary French, "Head Audio Guy"
Pyramid Teleproductions
Dallas, TX

db: What kind of work does Pyramid do?

GF: We're a TV post production facility. We do industrial, sports TV programming and a lot of commercial spots.

db: Do you have any DAWs now?

GF: No. This is our first serious look, as in we're going to do one, and we need to make a decision.

db: How did you like the conference? Did you get the information you needed?

GF: Yes. I think it was presented well. It gave (the manufacturers) time to do a good presentation, more than you'd do at a normal trade show. There was more time, because of fewer people, to sit down and talk with the manufacturers individually, to answer more questions and get better demos without the pressure of 150 people standing around trying to get in, and getting distracted. The big people and smaller people have been divided a little bit more, as far as function of their machines, what they'll do. You can discount people you don't want to talk to a bit easier now. I got the information I was looking for. Enough to narrow it down to the three to get some hands-on, inhouse demos, which is the next step.

db: Were there any particular demonstrations that stood out to you?

GF: There were three that stood out. What you get at a demonstration versus what is really there, and hopefully you're not getting "vapor-ware," as they call it—what they tell me, is it all there?

If I'm going to get an autoconform thing, is it in the box, or do I need to get another box so it can do it. I feel pretty secure with the presentations, but like anybody these days, you need to look back and see how the company is behind them.

db: Are there any changes overall in the technology you'd like to see? Do you think standards will inhibit this change?

GF: They have this (problem of) one manufacturer to another translation of files and information that isn't there. That would be important. As a facility, not an in-house production facility, we're dealing with a lot of people who travel around to different other facilities. To be able to take work done at one place and transfer into our own system would be awfully nice. I'm looking for more external machine control, which one company did come up with. That's important to me.

db: Anything you would like to say about the conference?

GF: I think you get a lot more done, a lot more accomplished in specific areas at the smaller conferences. I'd like to see that happen with other pieces of equipment, other than workstations. Audio analog editors, shoot-outs that way, where you have all the people for that piece of equipment, and that's all they talk about. It works out a lot better, it's more relaxed, and you get a lot more information. I'd like to see more of those.

Andy DeGanahl, general manager/chief engineer
PARC Studios
Altamonte Springs, FL

db: What kind of work does PARC do?

AD: It's primarily an audio studio. We don't do a lot of audio for video. Our major clients include Disney World. We did the assembly and mixing for the Superbowl halftime show this year. We do a lot of their major extravaganza, their Christmas and Easter TV shows. We do all their preproduction for that, a lot of overdubs. We also are the support studio for PARC Records, which is the primary reason for our existence. We never intended to be an outside client studio.

db: Are you here shopping, or catching up on what's happening?

AD: It's a combination of longterm looking at what's coming up, and trying to stay current on what's available. At the present time I have no plans for purchase of any of that type of gear, basically because most of that market is very soft here in Orlando.

db: How do you feel about the conference itself?

AD: It was much more in depth, and presented much more information than I thought would be possible in the amount of time they had allotted. I thought all the manufacturers did a very good job of showing up the features of their machine, and the way they related to the audience was very good. The audience was made up people that do this every day, and half of the people in there were just trying to see what was going on, like me. I think the compromise between the in-depth technical features of each of the systems versus the ease of operation was presented very well by most of the manufacturers.

db: Did anything stand out to you personally?

AD: I think I was most impressed with SSL's "risk everything" presentation. The SSL ScreenSound seemed to be the most together thing for a real world operation. I was also very impressed with Roland's potential, if what they say is going to happen, does happen. I thought they looked like they might be a real good, cost-effective alternative for digital multi-track work. Digidesign also seemed to be a capable presentation. I was very impressed with that.

db: Do you have opinions on changes and improvements in the technology, and do you think standards will prohibit that growth.

AD: I think the people who are into the MO (magneto optical) are a step ahead—eliminating the difficulties of archiving and back-up, or by putting it in the background of the software. Even then I would feel uncomfortable leaving the system on, without an operator, to do archiving in the middle of the night. Even done in the background, it's still prone to problems with power failures, especially in the Central Florida area where the power is unstable because of the lightning. The magneto optical looks to me like the next important thing just because you don't have to back up, and that's great. What would I like to see the manufacturers do? Since I don't do this every day, I'm not familiar enough with actual specifics. It looks to me like everybody's doing their homework, and headed in just about the same direction. I was very impressed with about everything that they did.

The discussion of standardization was interesting. It seems like we have two problems. We have simple sounds, sound files themselves that should be able to be transferred fairly simply. When you get into problems with sequences and play lists, all the manufacturers have their own system for loading those files. I think (John Gluck) from AMS raised a valid point: to even go to the MADI format for multi-track digital interface, they have to slow down their machine because that specification still isn't good enough. I think I would be leery of having a standard come in too soon and we have the same problem we have with NTSC TV—where the standard for color TV in the United States is so far below the technology we can handle now, but we're stuck with it. I'd like

to see the standards hang on for awhile until something shakes out, but not go as far as the digital multi-tracks where we have two major competing, incompatible technologies going on.

Although the ten manufacturers present at the SPARS conference represent a healthy cross section, there are, at last count, thirty-five to forty manufacturers of software dedicated to editing audio. Some are larger companies that are not SPARS members, some are small. Below is a guide to twenty-eight manufacturers, including the SPARS presenters. Our apologies for not finding the remaining ten or so manufacturers. db

THE MANUFACTURERS

AKAI/IMC (DD-1000)
1316 East Lancaster
Fort Worth, TX 76102
Tel. (817) 336-5114
Fax (817) 870-1271

AKG (DSE-7000)
1525 Alvarado Street
San Leandro, CA 95477
Tel. (415) 351-3500
Fax (415) 351-0500

Alpha Audio (DR-2)
2049 West Broad Street
Richmond, VA 23220
Tel. (804) 358-3852
Fax (804) 358-9496

AMS (Audio File)
7 Parklawn Drive
Bethel, CT 06801
Tel. (203) 792-4997
Fax (203) 792-7863

AQ Design, Inc./Transcom Digital
908 Broadway
Suite 1400
New York, NY 10010
Tel. (212) 529-1000
Fax (212) 529-1944

Creation Technologies (Next Computer)
2925 Fromme Road
North Vancouver, British Columbia
Canada V7K 2C6
Fax (604) 980-6711

Digidesign (Sound Tools)
1360 Willow Road
Suite 101
Menlo Park, CA 94025
Tel. (415) 688-0600
Fax (415) 327-0777

Digital Audio Labs
6311 Wayzata Boulevard
Suite 330
Saint Louis Park, MN 55416
Tel. (612) 559-6104
Fax (612) 544-5573

Digital Audio Research Limited (Sound Station II)
2 Silverglade Business Park—Leatherhead Road
Chessington, Surrey KT9 2QL, United Kingdom
Tel. 44-37-274-2848
Fax 44-37-274-3532

Digital Dynamics (SEE OTARI)

Doremi Labs Inc. (DAWN)
4927 North Glen Arden Avenue
Covina, CA 91724
Tel. (818) 966-2454
Fax (213) 622-1225

Ediflex Systems (Audiflex/Optiflex)
1225 Grand Central Avenue
Glendale, CA 91201
Tel. (818) 502-9100
Fax (818) 502-0052

Fairlight
30 Bay Street, Broadway
New South Wales 2007
Australia
Tel. 61 (2) 212-6111
Fax 61 (2) 281-5503

Hybrid Arts (ADAP II)
8522 National Boulevard
Culver City, CA 90232
Tel. (213) 841-0340
Fax (213) 841-0346

J.L. Cooper Electronics
13478 Beach Avenue
Marina Del Ray, CA 90292
Tel. (213) 306-4131
Fax (213) 822-2252

Lexicon (Opus)
100 Beaver Street
Waltham, MA 02154
Tel. (617) 891-6790
Fax (617) 891-0340

Micro Technology Unlimited
156 Windchime Court
Raleigh, NC 27615
Tel. (919) 870-0344
Fax (919) 870-7163

New England Digital (Post Pro SD)
Rivermill Commercial Center
Lebanon, NH 03766
Tel. (603) 448-5897
Fax (603) 448-3684

Opcode Systems (StudioVision, Cue)
3641 Haven, Suite A
Menlo Park, CA 94025
Tel. (415) 369-8131
Fax (415) 369-1747

Otari Corporation (DDR-10, Pro Disk-464)
378 Vintage Park Drive
Foster City, CA 94404
Tel. (415) 341-5900
Fax (415) 341-7200

Pine Link Inc. (Microsound)
11684 Ventura Boulevard, Suite 910
Studio City, CA 91604
Tel. (818) 760-4539
Fax (818) 505-0341

Roland Corporation, Pro Audio/Video Group (DM-80)
7200 Dominion Circle
Los Angeles, CA 90040
Tel. (213) 685-5141
Fax (213) 722-0911

Solid State Logic (Screen Sound)
320 West 46th Street, Second Floor
New York, NY 10036
Tel. (212) 315-1111
Fax (212) 315-0251

Sonic Solutions (Sonic System, No Noise)
1902 Van Ness Avenue, Suite 300
San Francisco, CA 94109
Tel. (415) 394-8100
Fax (415) 394-8099

Soundtracker Pty, Ltd.
169 Bank Street
South Melbourne, Victoria3205
Australia

Studer Dyaxis
1425 Elm Hill Pike
Nashville, TN 37210
Tel. (615) 254-5651
Fax (615) 256-7619

Turtle Beach (56K DRC)
1600 Pennsylvania Avenue, Unit 33
York, PA 17405
Tel. (717) 843-6916
Fax (717) 854-8319

WaveFrame (Audioframe, Cyberframe)
2511 55th Street
Boulder, CO 80301
Tel. (303) 447-1572
Fax (303) 447-2351

TEK

TEXT

BOB THURMOND

Basic Sound System Performance Measurements

The system just doesn't sound as good as it should. The equipment seems fine, and the installation looks OK. In fact, there are no obvious problems at all, but the sound quality is still less than you expected. What do you do now?

IN SUCH A SITUATION, YOU MAY HAVE three choices:

1. Walk away from it;
2. Mess around with it;
3. Get help.

Let's think about these choices. The first one may be the wisest, unless this is your own system which you just finished installing. If not, maybe you can convince the owner that this is the best that can be done, but how long will it be until someone else proves you wrong? How long do you intend to stay in this business? Even if it is someone else's job, will the owner really believe you when you tell him a new system will solve everything? What if it doesn't? This has never happened to you? Just wait—it's a real thrill.

The second choice is probably the most common, since most people are quite confident they can find a problem if they just look long enough. But what if the problem is one you have never encountered before, or is so obscure you simply do not recognize it? What if there are several problems, all interacting with each other, so that you cannot tell which is doing what, or you just overlook some of them? Such problems are very common in complex systems like these. Again, you run the risk of failing to solve the problems, because you did not recognize or understand them well enough.

The third choice has, by far, the best chance of succeeding and making you look (sound?) good. It may also carry the highest immediate cost, both to your budget and ego. In fact, this latter hurdle may be the most difficult to overcome; no one likes to think that they may not know as much as they should. Think of such help as an investment, just like many others you must make.

HELP

This can come from many sources; these pages and others, formal courses and good old experience. For specific problems and projects, you need something a bit more tangible and immediate. Possibilities include:

1. An expert;
2. Appropriate test equipment;
3. All of the above.

While it might be possible to hire a full-time expert who can supply all the information you need, it is unlikely that you could either find or afford such a person. If you are lucky, you might find someone with some knowledge in some of the areas you need, for a reasonable salary. However, this still leaves some gaps to fill in. Maybe what you need is an occasional "gun for hire," more commonly called a consultant. Some are actually quite willing to work on such an occasional and limited basis; ask around. Most people prefer the second option, however. The prospect of becoming an instant expert just by buying the right hardware is very appealing. After all, isn't this exactly what your competition flaunts in your face? Ours is a hardware industry; you need the toys to play the game.

There may be a few tiny problems with this approach, however, such as:

1. The toys are expensive;
2. They are not easy to learn to use;
3. They do not assure you of winning the game.

In fact, there are several important system performance characteristics which even the expensive, trendy devices cannot measure. Perhaps this is not the right answer, either. That leaves the combination approach, as you have already guessed. Somehow, we must find a combination of hardware and software (expertise) which will tell us everything we need to know. This is a tall order! Many consultants do not carry that many guns.

TELL ME WHAT YOU WANT

By the way, exactly what do we need to know about the performance of a sound system? Naturally, there is considerable difference of opinion on this matter, but there is also some agreement. For example, there is compelling evidence that the most important audible characteristic of any sound system, by far, is its frequency response. Other obviously important characteristics are loudness, freedom from distortion and noise, and reliability. Furthermore, there are some hidden implications here. In particular, the frequency response is perceived by many different listeners at the same time. Naturally, they should all hear very nearly the same thing, but few sound professionals have ever really checked to see how far this requirement has been met (or missed). The same considerations also apply to other characteristics, such as loudness. All of this means that audience coverage uniformity emerges as a very important, but widely overlooked, characteristic.

So, we have more to measure than we first thought. Maybe this is

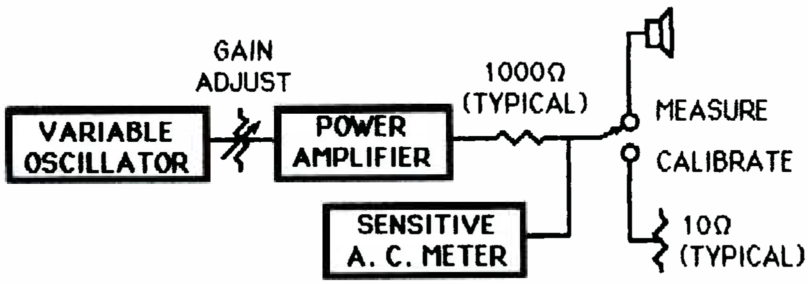


Figure 1. A simple apparatus for measuring the impedance of a loudspeaker line at all frequencies.

why a system equalization often turns out poorly; something important was left unmeasured and completely overlooked. This is ominous; could it be the situation is much more complex than we had expected and that measuring it adequately will be just too expensive? Yes and no! It is true that the factors affecting the sound heard over a system are more complex than most people realize, but they can be measured and understood. Expensive instruments are not required, either; inexpensive devices will work just fine if you only have the right ones and know how to use them.

WHERE TO BEGIN

Since the frequency response is the most important system characteristic, shouldn't we measure it first? That is what many people do; in fact, they often do little else. But that is the mistake: many problems with a system can affect its response. These all need to be found and eliminated before the final response can be measured accurately. In fact, it is wisest to make sure the system is functioning as well as it possibly can first, and measure its response last.

The logical first step is a thorough visual inspection of the system, looking for wiring and mounting errors. Are all the loudspeaker drivers very close together and in the same plane? This is very important for good coverage uniformity, and is a common problem. Many "arrayed" full-range units have severe problems in this respect, as later tests can show.

The next step would probably be to listen carefully to the system, to detect any distortion, noise or malfunction. A simple audio oscillator can be very helpful here, especially in setting gain controls for the best signal levels and for troubleshooting. This is also the time to see that all microphones and loudspeakers are wired in the same polarity (not phase). Assume nothing—a unit may be miswired internally. An inexpensive polarity checker is very handy here, but there are ways to test polarity without a checker. For example, you can put two mics side by side, open both to the same level, and listen to what they are picking up together. If the sound is OK, so is the polarity; otherwise, it is not.

IMPEDANCE PROGRESS

Probably the most common system wiring error is a loudspeaker load with an impedance too low for the power amplifier to handle adequately. If it is a little too low, the result is increased distortion and chance of overload; a greater error can result in the sudden failure of drivers and/or amplifiers. Again, do not assume anything, especially that the manufacturer's rating is correct. A loudspeaker's impedance varies greatly with frequency, and is often considerably lower than the rating. Furthermore, as loudspeakers age, their impedance typically drops even lower.

There are impedance meters which are not expensive, but they measure at only one or two frequencies, which is not good enough. It is much better to measure the impedance over the full frequency range. No inexpensive instrument exists to do this, but you can rig one up very easily.

Figure 1 shows how to do it. The oscillator feeds the line under test through a resistor about 100 times the nominal value of the line impedance. Under these conditions, the voltage across this line will vary according to the impedance. We first calibrate the meter by putting a known value in place of the line and setting the levels so that the meter reading corresponds to the resistor value. For example, if we use a 10 ohm resistor, we might set the levels so that the meter reads 10 mV. A little experimentation will be needed to get everything right. Check to see that the reading is the same at all frequencies. Then substitute the line for the known resistor and read the level on the meter. A reading of 5 mV means 5 ohms, 20 mV is 20 ohms, and so on. Then we vary the oscillator over the audio range and watch the meter readings, especially for low values (a high impedance is seldom a problem). The actual readings may be a lot lower than you expect.

Transformers, such as those used in 70V lines, are especially bad at low frequencies. The impedance of any such line should always be measured carefully. High-level crossovers are another problem area; if the driver impedance changes strongly near the crossover frequency (usually because of a resonance), the crossover characteristics will most likely be affected. An impedance which rises with higher frequencies indicates an inductive load (which may be bad at low frequencies), while one which falls indicates a capacitive load (maybe bad at high frequencies). Numerous peaks (resonances) are another sign of trouble. Each model of loudspeaker has its characteristic impedance curve, just like a response curve. Any variation from this norm implies a similar variation in response, probably indicating a faulty unit. This is another reason to run such tests.

Any load, not just loudspeakers, can be checked in this way, but the most problems, by far, are found in loudspeaker lines. For sure, the total load on each power amplifier in a system should be measured, and it would be wise to check each individual driver. In this way, many unsuspected problems can be located before causing trouble.

GET COVERED

After we are sure all the components of a system are working properly, individually and together, then we can see how well they are doing their job of delivering sound to the audience. The parameters here are frequency response and sound level, and our first concern is that these be as uniform as possible over the listening area. How can we say what the response or level of a system is if these characteristics are different everywhere?

Measurements of sound in a room require some special devices and techniques. A single frequency (as from an oscillator) will not tell us much, because the zillions (by actual count) of resonances in a room cause the sound level to change greatly with even slight changes in frequency or location. Measured this way, uniformity is nonexistent!

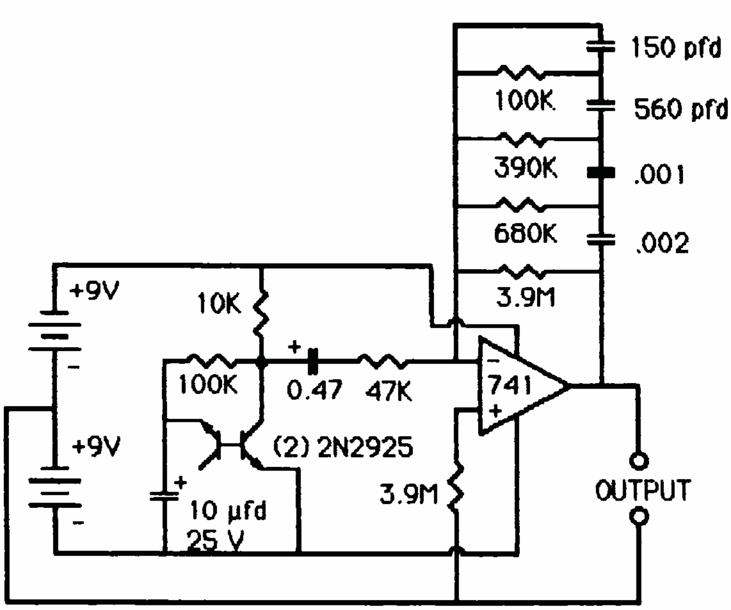
We are usually unaware of this because we hear groups of frequencies rather than individual ones, which averages out the level variations. Rather than fight nature, we should take our measurements in the same way. That means we need a signal source which produces many frequencies at once; the most commonly used source of this type is random noise.

Our ears work on a nearly logarithmic frequency scale, which means that we hear (at higher levels) nearly the same loudness in bandwidths which are the same percentages of their center frequency. If we measure random electronic (white) noise in this way, its level increases toward higher frequencies, at a rate of 3 dB per octave. Since it would be much more pleasing to both our ears and our measurements if the spectrum were flat, we usually feed the noise through a "pink" filter which attenuates the higher frequencies at a rate of 3 dB per octave.

Inexpensive pink noise generators are available commercially, or you can build one quite easily. *Figure 2* shows the schematic for a generator which is within 1 dB of ideal flatness between 50 and 15 kHz, and which rolls off gradually outside of these limits. The noise generator portion was derived from a circuit which Walter Jung published in the February 1971 issue of **db Magazine**. Other circuits are somewhat better, but this one is very simple and inexpensive, and quite satisfactory if you understand its limitations. Specifically, its output level varies some with temperature, so use it in a fairly stable environment. The first transistor should be selected for highest output without sputter. I have never found another type which works as well as the 2N2925 here, but, since they are readily available for about 25 cents each, that is no problem.

In a real pinch, you can even use an FM radio tuned off station. The noise spectrum is white with the highest frequencies rolled off by the de-emphasis circuit, which results in a very broad hump centered about 2 kHz. This is not too terrible for a rough coverage check, as the hump falls close to the frequency at which our ears are most sensitive to coverage irregularities.

Figure 2. An inexpensive pink-noise generator.



Of course, we also need some way to measure the sound levels which the noise produces over the system. Such a device is cleverly called a sound level meter (SLM). A model which is adequate for basic measurements is available from Radio Shack for under $40. Other devices are more accurate and do more, and cost more. The additional feature which is highly desirable, but not absolutely essential, is octave or third-octave filters built in. Consult the guide in the November/December 1990 issue of **db Magazine** to see what is available.

START HERE

Let us assume you have only the simplest equipment and let's see what we can do with it. Feed noise over the system at a moderate level and set your SLM to A-weighting. This setting rolls off the lower frequencies rather strongly, similar to your hearing at lower sound levels. It also lets the meter read mostly the frequencies where coverage uniformity is most important. Set the sensitivity to get an on-scale reading at a slow meter speed. Slowly walk through the listening area, holding the meter away from you and not getting between it and the loudspeakers, or the meter between you and the loudspeakers. Walk from side to side and from front to rear, along several different paths, and note how the reading changes. With such a broad frequency range, the reading should be nearly the same at all locations, within 3 or 4 dB. If it varies more than that, your coverage is not as uniform as it should be, which is a very common problem. Even if the reading is consistent, you still may have coverage problems. If you are in a reverberant room, for example, then you may be reading only reverberant sound, which has almost the same level everywhere, while the direct sound from the loudspeakers varies by an unknown amount.

While making your traverses, listen carefully for any changes in the noise. Sudden changes across a limited distance, especially if they are repetitive, are probably caused by a poor overlap between adjacent loudspeakers. Your ears are more sensitive to such irregularities than this type of instrument. On the other hand, your ears will probably not notice a gradual change in level, which the meter will clearly indicate. Listen also to the nature of the sound source; it should be well-defined and stable. Problems here are further indication of poor loudspeaker layout.

PIECE BY PIECE

There are other techniques which will tell us much more about the coverage. One of the most useful techniques is to feed noise over only one loudspeaker at a time, and measure its coverage. Find the point where the level is the highest, then see where the level drops off 3 dB from this, and, for a better understanding, mark such points on a floor plan. Connect these points, and you will have drawn the -3 dB contour, which is the limit of really good coverage. For even more information, do the same for the -6 dB points. Then repeat all of this for each loudspeaker, at least roughly.

Now examine each coverage area. Individually, each should cover a portion of the audience evenly and not spill much outside this area, or onto walls or unoccupied floor. Together, they should fit neatly to cover all the listening area and little else, with little mutual overlap. Furthermore, the actual level within each area should be the same as in the others.

Tedious? Yes. Revealing? Absolutely! Disturbing? Quite possibly. Certainly this procedure will tell you what is right and wrong about the coverage, and may well reveal that the coverage is not what you expected. It will also tell you, of course, how well your improvement efforts are working. It may show, for example, that no matter what you do with your loudspeakers, you cannot cover all of the audience area evenly. Bad news: you need different loudspeakers. This may, however, explain a few things.

Likewise, this technique is not limited to high-frequency coverage. Coverage of low-frequency units can be measured simply by turning off everything else and using the C-weighting on the meter. However, since this weighting is almost flat, the meter will now pick up a great deal of low-frequency ambient noise. Check the level of this first with the system off, then set the system level 10 or 20 dB higher. Be careful not to overload anything!

At low frequencies, level variations occur much more gradually because of the longer wavelengths involved. For the same reason, low-frequency loudspeakers are almost non-directional, so changes in their location and aiming have less effect. It is often impossible to get really good coverage in this region.

BAND TOGETHER

Even more can be learned by using octave band filters. Such a limited frequency range reveals level variations much more easily and quickly. Additionally, full-range units can be tested with all drivers on, because the filters can discriminate between them.

It is convenient, but not essential, to have the filters built into the SLM. In fact, nearly the same results can be obtained with an octave band graphic equalizer. Simply wire it into the noise signal path, then turn one filter all the way up and the others all the way down. This will produce an octave band of noise, good enough for coverage measurements. Uniformity should be checked in several bands—typically at 4 kHz and an octave or so below each crossover frequency.

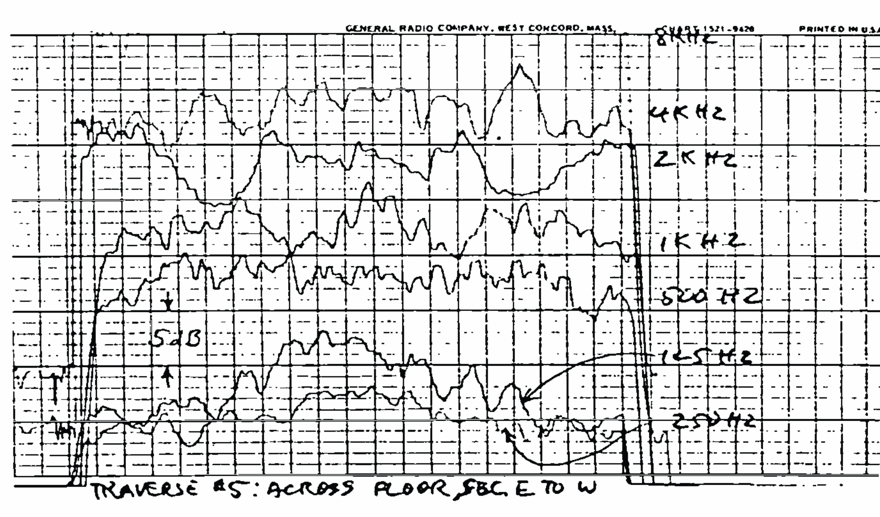
Filters are also very useful for setting levels, both between the various drivers in a full-range unit, and between various units covering the same frequency range. An equalizer may be a bit less accurate in this application, however, as the various filters may put out slightly different levels at their maximum settings. If these levels are measured, however, and adjusted to be the same, then there should be no problem.

THICKER PLOT

Extensive readings taken by these techniques and carefully plotted will show a great deal more about actual loudspeaker coverage than you ever suspected, if you are willing to go to the trouble. There is a way to get even more information and save considerable time and trouble at the same time, but, of course, it is not free. It involves tape recording the sound during the traverses, rather than reading the levels directly, then playing back the recording through filters into a chart recorder. The resulting graphs, such as those in *Figure 3*, are much more informative than any direct readings could be.

For example, the graphs shown are plotted from a traverse across the floor of a large arena, which had a well-designed central loudspeaker cluster. The 125 Hz band shows a rise in level near the center of the floor, because that area is closer to the cluster and the low-frequency units were not directional enough at this frequency to offset this effect. At 250 and 500 Hz, their directional characteristics are adequate to provide very uniform cov-

Figure 3. Plots of actual loudspeaker coverage uniformity.



erage. The high frequencies show moderate irregularities about halfway to each side, because this is the transition region between horns. At any given location, the level may be up at one frequency and down at another. Readings taken over a broader frequency range would not reveal such irregularities, even though they are clearly audible, which illustrates the value of this particular technique. Actually, the coverage shown is quite good, yet its shortcomings are clearly revealed by this measurement technique.

The filters needed for this method are no problem, as we have already seen. The recorder used was a Sony Walkman Professional; any good-quality recorder without a limiter will do. The chart recorder? Ah, that is the problem; they all seem to cost $2,000 or more. That is really not bad for what they can do, but if your budget is not quite ready for that, there is an alternative. Some consultants are willing to take your recording and plot it out in this way for a reasonable fee. Ask around if you are interested.

ANOTHER DIMENSION

This set of plots has also given us some information on another matter. In some locations, the sound level was up in one frequency band and down in another. This, of course, amounts to a response irregularity, which changes with location. In other words, the response may not be uniform over the listening area, even if the overall level is. Coverage uniformity comes in two dimensions: overall level and frequency response. Just when you thought you were beginning to get a handle on all of this, Murphy strikes again.

Unfortunately, response irregularities are even more pervasive and difficult to control than level irregularities. Anyone who has ever carried a real-time analyzer around in a listening area while pink noise was playing over a system has seen the truth of this. The response is literally different in every seat.

This has several annoying implications. First, there is no location where an "average" system response can be measured. Second, the only way to determine the actual average system response is to measure the response in a number of locations and then average the results together. Third, the response in any given location will be different from this average, perhaps considerably so.

All this means that the system frequency response is difficult to measure. Forget about just taking a reading with a real-time analyzer; that will tell you something about the response at one location, which will be different at all others. If we could take many such readings at different locations and average all of them together, we might have something useful, but how do we go about such a thing? Furthermore, how do we go about it if we do not have a real-time analyzer and do not want to spend the money for one?

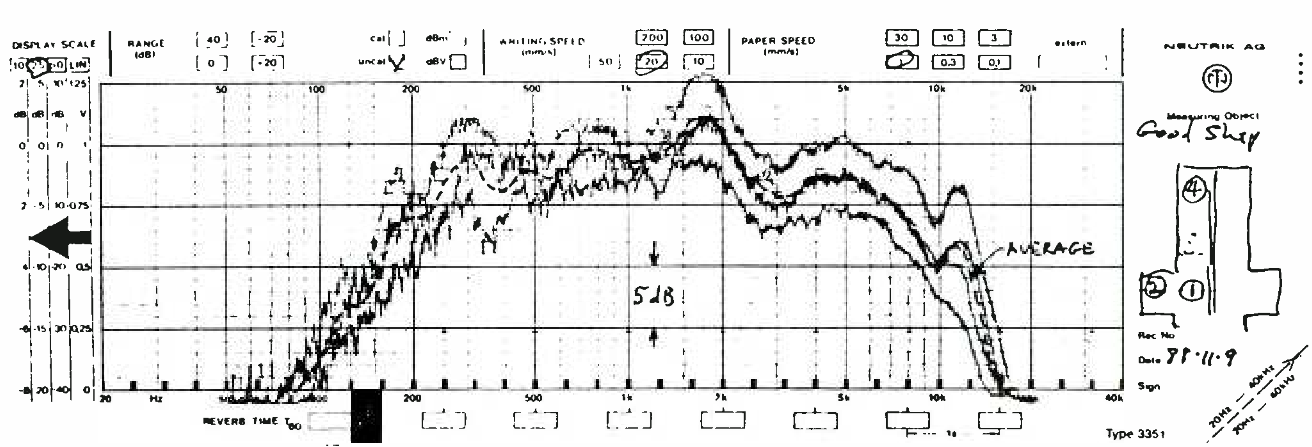
THE RIGHT RESPONSE

As before, there really are ways to make accurate and meaningful measurements of the system frequency response with a minimum of equipment, if you are willing to go to the trouble. How do you think it was done before there were any real-time analyzers? (Yes, such measurements were actually made for many years before then, but only by a few dedicated professionals.)

The minimum test equipment needed is a pink-noise generator and a filter set. The same equalizer you use for response correction can be used for measurement, if you are careful. It can be any bandwidth—octave, two-thirds or third-octave—of the graphic (boost and cut) type. As always, the narrower the bandwidth, the better the results. Rauland-Borg used to manufacture a device designed just for this application, which worked very well (trust me; I designed it). It is no longer available, but you can approximate its function with any graphic equalizer.

The trick, once again, is to set all the equalizer bands to minimum, except for one, which is set to maximum. The danger in this is that the various filters in the equalizer may not be matched well enough to produce all the same level. Check this first by feeding the noise generator through the equalizer to a VU meter. It will be easier to read the meter, especially at lower frequencies, if you temporarily slow down its response speed by connecting a large (at least 1000 fd) capacitor across its terminals.

Figure 4. Plots of actual system frequency response, measured by continuous sweep.



Set all the bands to minimum except one, which is set at maximum, and read the resulting level. Try starting with the lowest band, and adjust overall gains (not the band) until you get a "0" reading on the meter. Visually average the meter fluctuations over a few seconds. Then reduce this band to minimum, turn up the next one, and read the level without changing any other gains. If this reading is higher than before (usual), reduce the filter setting until you get the same reading, and note (mark) the resulting setting. Repeat this for all of the filters, and you will have them calibrated to the same level. Such filters will not be as good as those in most analyzers, but they will be quite sufficient for our purposes.

THE SETUP

Set up the system by feeding the pink noise into the power amplifiers, set to produce a moderately high sound level over the loudspeakers (about 80 dB). Place the main system mic (if there is one; otherwise use a flat-response omnidirectional type) in the seating area, at ear level and pointed toward the nearest loudspeaker. Feed its output into the slow-responding VU meter, with gains set to produce a mid-scale reading. Turn off the noise generator temporarily to be sure that the background (ambient) noise level is much lower. Insert the filter set (equalizer) into the system either after the noise generator or before the meter. Theoretically, the results will be the same either way, but there are practical differences. With the filter after the noise generator, you will hear each noise band individually, which may be useful for detecting overload, distortion, rattles, and so on, in the amplifiers or loudspeakers. Filtering before the meter will help discriminate against ambient noise if it is too high.

Run the lowest band on the equalizer up to maximum, with all others at minimum, and read the level on the meter (if it is not too low). Plot this as a point on frequency graph paper. Then turn this band down, run the next one up to its calibrated level, and take and plot the new reading. Repeat this for each band in turn, then connect the points on the graph for clarity. Next, move the mic to an entirely different location in the listening area and take another set of readings. The plots will be clearer if each is made in a different color. Take readings at six to twelve locations scattered throughout the listening area.

This may take a couple of hours, but there is little other cost, and the information obtained is highly valuable.

ON THE AVERAGE

Examine the resulting plots to see how closely they track each other. If they all fall within a 5 dB window, you have either an incredibly good loudspeaker system or a very reverberant room, which gave you misleading readings. It is normal to have greater variations at lower frequencies because of the lack of directional control in loudspeakers in that region. Greater variations at high frequencies are also common, and indicate inadequate coverage. One or more plots which are consistently higher or lower than most indicate poor level matching. Wide variations over the mid-range strongly indicate an inadequate loudspeaker design.

Draw an average through all the plots. This, finally, is the overall system response. Examine its shape; it is not unusual for the level to drop off rather sharply near the frequency extremes, but large irregularities elsewhere are a sign of trouble. Crossovers are a common problem area; consider these carefully. All in all, your first carefully measured system response may look considerably worse than you expected.

This method will give you as much information about your system response as will a real-time analyzer with built-in memories. In fact, it can do even better, because the number of plots is not limited by the memories, and because they are produced in hard copy. The results are certainly nothing to be ashamed of, but there are ways to do even better. One way would be to use a bandwidth narrower than third-octave, but such devices are rare and expensive. Another way is to analyze and equalize, using filters which can be set to any frequency, rather than being limited to fixed points. Such filters are called parametrics, and they are readily available at reasonable prices. It is even possible to build the necessary sweepable filter easily and inexpensively, if you are interested. As before, the only real price for more information and better results is more time and trouble expended.

A CLEAN SWEEP

In this case, the procedure is to set up the system, with the equalizer installed as before, and set the filter bandwidth to one-third octave. Then the filter center frequency is slowly swept, and the resulting level variations are read on the meter. The exact frequencies of peaks and dips can be found and plotted, as well as enough other points to establish the true shape of the response curve. After several such curves are plotted and averaged to get the overall system response, the needed corrections can be seen very clearly.

All of this is very helpful, but tedious. Fortunately, a recording swept analyzer, which consists of a tunable filter coupled to a chart recorder, can do all of this for you.

Just seeing which band of an equalizer the feedback falls in is not good enough, since the bands are too broad and overlap too much

Currently, only one manufacturer (Neutrik) makes a version well-suited to this application, and it costs about $3,000, but another manufacturer (GenRad) used to make a similar device which is still widely available on the used equipment market. Both are capable of making several types of useful measurements, including reverberation time. *Figure 4* shows an example of such an automated response plot.

As before, it is possible to record the noise at various locations, then send the recording to someone who has the equipment to analyze it and plot out the frequency response. The tricky part here is that the response of the recorder probably is not flat, but a test recording, consisting of pink noise fed directly into the recorder, can be analyzed to compensate for such shortcomings. It can work, if you have time to wait for the results.

THE HOWLING

The final system measurement is of the feedback frequencies, which, like many other characteristics, are often not measured at all. In fact, it is not necessary to measure the feedback frequencies unless feedback is a real problem in a particular system, and a special effort must be made to control it. Then, more information may be helpful.

Just seeing which band of an equalizer the feedback falls in is not good enough, since the bands are too broad and overlap too much. If you have an oscillator with an accurate tuning indicator, you can tune it to the same pitch as the feedback and read the scale, but that is usually not close enough, either. A better method is to read the frequency directly with an inexpensive frequency counter. There are even ways to measure all the frequencies at which a system can feed back, without actually making it do so, but these require a bit of specialized equipment and techniques which almost no one seems to know about. Furthermore, while it is possible to perform this analysis via a recording, such a technique would be difficult and probably impractical.

NOW WE KNOW

However, we have seen that many very useful performance tests can be carried out easily and inexpensively. In fact, these tests provide a great deal of information on exactly why the system sounds the way it does, much of which cannot be obtained at all with highly sophisticated and expensive test equipment. While the most basic tests can be tedious, they provide excellent results for a minimal equipment cost. Moreover, even better results can be obtained much more quickly and easily for a relatively modest investment. Also, if you have only occasional need for full system testing, or would rather not try to handle it all yourself, there are consultants who can provide such services.

We have not yet said as much as we need to about how to interpret the results of these tests, and have said almost nothing about how to correct the problems they reveal. All of that would fill another article, as it probably should. If you just cannot wait, more information is available from the referenced papers, or you can contact the author directly at (512) 837-7252.

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JAMES MASON

Book Review

THE SONGWRITER'S WORKSHOP

● Take a select group of music industry veterans, each with vast experience in different disciplines—songwriting, audio engineering, MIDI programming and the publishing business—and put them together with the mandate: create a source of information and direction for anyone serious about songwriting (novice or adept). What do you get? The Songwriter's Workshop.

The Songwriter's Workshop is like attending a private seminar; you are transported to a convention on songwriting without the air fare, hotel bills, and the crowds.

The purpose of The Songwriter's Workshop is to teach you how to improve the quality of your songs, properly use the available technology to create high quality demos, and market your songs more effectively.

The Songwriter's Workshop consists of an eighty-six page paperbound book and two audio cassettes which let you hear many of the techniques and processes described.

There are four sections to The Songwriter's Workshop. Each is a separate workshop led by an industry professional. The first section on songwriting is written by Janis Ian. The second, written by John Barilla, is on recording your demo. James Becher has written the third on MIDI, and the fourth section, on marketing your songs, is written by Teri Muench.

From its opening line, Ian's section "On Songwriting" is extremely well-written and insightful. Muench's section on "The Art of Pitching Songs" is also packed with helpful information. The focus of this review, however, in keeping with the direction of **db Magazine**, will be the "Making Demos" section by Barilla and Becher's section "Understanding MIDI."

If you are a regular reader of **db Magazine**, you are well aware of the ongoing changes the computer, home studios and MIDI have had on this industry. The past seven years have seen numerous audio-related devices, with remarkable capacity, come onto the market. Porta-Studios, Digital Audio Tape, Sequencers, Multi-Timbral Synthesizers and Digital Signal Processors are of such low cost and high quality that in many ways, the home studio of today can outperform the million dollar studio of the early eighties.

The universe of songwriter demos no longer need contain such disparity between have and have-nots. The slick, expensive, record company style demo, or the cheap I'm-just-starting-out sounding productions now become artistic choices rather than economic limitations. At least this is true for many more people than ever before. That's the good news. The price that comes along with this, at least until artificially intelligent operating systems replace audio engineers, is the need to understand how to properly, and artfully, use all of these wonderful devices. As wonderful as they are, though, they won't jump out of the box, properly install themselves and begin making hit records. (That particular device is being held captive by Narada Michael Walden.)

I'm sure many songwriters who would love to use these technologies in their demos or perhaps build and operate a home studio are hesitant due to a lack of technical insight and experience. If this feeling sounds familiar, it is for you that the two "technical" sections of The Songwriter's Workshop were written.

Barilla's section on recording your demo is rich with the type of nuts and bolts information a songwriter will need to build a home studio and engineer/produce a home demo. He covers the gamut from the most inexpensive to a mid-priced home studio.

This section also includes an introduction to the use of audio signal processors. Here the audio cassette comes in very handy. The tape takes you through the production of a demo and, in addition to showing you song format, demonstrates the use of commonly-used effects and techniques. A songwriter can learn the basics of "at home engineering" by simply trying the demonstration examples on their own demos. The tapes and the text are well-conceived and helpful, but the total novice might find the pace a bit overwhelming.

The same can be said of the pace of Becher's section "Understanding MIDI." This section is not unlike other introductions to MIDI books on the market. In a craftsman-like manner, you are taken on a tour of the world according to MIDI. However, the included tapes set this section apart from other books of this type. The tapes allow you to hear various techniques used to create layers of sound using MIDI devices. This elevates Becher's treaties beyond the average MIDI primer. You are not just given another dry unfulfilling description of MIDI's potential. Useful techniques are clearly demonstrated.

Overall, given the information contained within and the twenty-five dollar price tag, I'd say The Songwriter's Workshop is a great value. The tapes and text allow you to examine expert advice over and over (until you get it right). The sections not included in this review are also very good. Even the experts will find something of use here. This is one seminar well worth attending, and I recommend it highly.

James Mason is the Chairman of the Recording Arts Department at The Center For the Media Arts and has had many years of experience as a recording engineer..

LARRY ZIDE

Sound Reinforcement in The Park, Part II

In this concluding part, David Robb of Jaffe Acoustics continues in his explanation of what, how, which, and when, went into the new system used last and this year in New York City's Central Park.

db: We were going to start describing the actual components that are in each system.

DR: We did get to that. Let me just get back to that speaker tower just in a broad sense. After we've gotten out of the electronics part of it, the third piece of that tower are the speaker units. There are three separate subsystems of speakers all enclosed in one box. This was a design that we came up with in working with Ken Forsythe, the head of engineering at Eastern Acoustic Works. The main part of this speaker system is the segment that faces toward the audience. This provides the bulk of the reinforcement of whatever is happening on the stage. The system consists of a three-way amplified system, the low end of that being two 12 in. speakers, and the mid range having two 7 in. speakers. The hi end is an interesting device kind of invented by Eastern Acoustic Works which consists of a typical compression driver, but not loaded onto a horn. It is more of a freestanding driver that has a bit of a raveguide to it. The client and we here at Jaffe were very concerned about anything that would smack of being a horn because it never seems to sound very good for particularly classical-type music. That's a triamplified system there which is the main reinforcement. there is also a suplimentary subwoofer which is a single 18 in. speaker...

db: All in the same box?

DR: There is actually a separate chamber, but the entire package is in one box. But in a separate chamber we have a subwoofer, a single 18, and it faces straight down. The box is held up in the air and there's nothing directly underneath it. There's actually a window between the top of the electronics rack and the bottom of the speaker and that was part of the design philosophy because these things were going to be sitting out in the middle of the field. The audience needed to be able to still see the stage and the big "blobs" there. I don't know if we've completely succeeded, but we tried to make it so the average person could see over the electronics and under the speaker and still be able to look right through there and not have the impression of this big thing. Between the top of the electronics and the bottom of the speaker there's a large amount of space, so we're able to put the subwoofer down far. This helps us out quite a bit. It's truly a subwoofer; it only operates up to 100 cycles. Not only did we not have to make the cabinet bigger to make the 18 face forward, we had been able to tuck this up in behind, and it had very little directional characteristics being below 100 cycles. It also seemed to just work better to aim straight down and kind of roll out in all directions on the ground and to, well, that's kind of a visual thing...

db: What it actually did, because I did hear them there, was virtually create a room sound outdoors.

DR: That's part of it. That's the subwoofer being really omni and being able to meld easily with an adjacent tower. It really worked out well for us that way. If you listen to one of these boxes standing alone, I initially thought the sub part of it might be just a little light in volume. I was wondering if we needed just a bit more there. But what happened was as soon as we spread these things out in a configuration on the field (wherever you stand you're within probably direct shot of one reinforcement system that has hit you in the face, and you might have a couple of other ones on the sides that are way down in volume because you're out of the pattern), the low end from the middle of four subwoofers and that level came strong and right back up. It was very impressive, I thought. So that subwoofer 'is section two.

The third subsection is an interesting part of this. Jaffe Acoustics, about twenty years ago, came up with an electronics system for the Ravina concert facility for the Chicago Symphony out in Chicago. Because this was a pavilion in an outdoor situation and you could mix classical music and project it out to people, you can add reverberation and do whatever you want to it, but it still remains a very two dimensional thing. You can get very good sound doing it that way, but it is nothing like the concert hall experience. The concert hall experience, in addition to having the reverberation and multiple arrivals and all, has another sense of being in the middle of the sound field and not just experiencing a two dimensional thing as it comes at you. but actually that third dimension. I'm not really talking about quad or anything. In a concert hall you have reflections off the ceilings and walls and everywhere.

db: Which you do not have in the anechoic chamber we call the outdoors.

DR: Right. That big old field. You might get something that bounces off a bird but... So, regardless of the quality of sound you start out with, it can be good sound, but it is not like the concert-hall experience that we were really trying to take to that field. So this third part of our system is to assist in experiencing those reflections. What we've done is we have a smaller system of limited bandwidth, limited volume, and different delay time that will allow signal to arrive at the listener from the sides, from the rear, and in effect, put them in the middle of an experience. It has worked extremely well. Typically, when you're at any listening position, you have some relationship to four of the speaker towers. Usually there's one between you and the orchestra and that's what's providing your main reinforcement. Then there's probably one on either side of you from which you're getting some combination of ambient sounds. You're hearing a bit of the forward firing system and you're hearing a bit of the rearward firing system from both. This is actually the third subsystem which is aimed back toward the stage. Then you have the one that's somewhere behind you where you're only hearing the rear-firing. All four of those different systems have different time relationships to where you are. They're all calculated to give that experience of where you might be sitting in a concert hall. The rear wall might be seventy-five feet behind you, and you're getting reflections from that and the side walls, so all this stuff is figured out. The way that it's done is by manipulation of the digital delays which are in each one of the speaker towers.

The product we chose is made by Audio Digital and is actually kind of a hybrid unit that is located somewhere between their ADD Series and their newer PAD Series. We needed some of the functions of both of those systems and so Audio Digital made a sort of customized version of that. With the outputs of the delay, we have one output that is just a propagation delay, that matches up the sound source at the tower and delays it back so it arrives at the same time as any source from the stage that can actually reach that far out. At a certain point out in the audience, the stage sound is so minimal the wine bottle clinking next to you will completely mask it. We got into a number of interesting discussions and experiments with the Haas-Precedence Effect as to whether we actually wanted to try to hold the image back to the stage, or whether there was actually going to be any sound image back there that we had to connect with. We ended up with a combination of some of the nearer things that could hear the stage (we actually did defer to the stage as the source of sound), but as we got farther out, we didn't defer to the second ring of speakers as being the source because we could no longer hear the stage. We let *them* become the source. We didn't keep delaying back all the way as we went out and I think we made the right choice.

Once you get out into the field, the broadcast system is done in mono. There's just one source being broadcast out there, so everything is done with the ambient speakers. Everything else we're doing there is all generated from one single signal; there's no additional things added to it. We just manipulated there at each tower. I'll give you a little map of how we set the speakers out there. You'll see that all the speakers lie on concentric rings that run out from the center of the stage and this is how we choose our delay times. Now the delays are set. Every one of these towers is identical because as they're put up and down every day, we don't want to have to say, "Okay, this is Tower No. 3 and it goes way out over there" when it might be much easier to just put it right down here in front of you.

So, every one of these towers is identical, every one of the delays is identical, and the delays have sixteen memories that are committed into each one of them, What we can do is, depending on where they get placed, we can call up one of those memories which will then call up the associated three delay times required to operate that tower. At the very beginning of the day when the speaker positions are mapped out, each one gets a colored flag which corresponds to the ring number on which it is out from the stage so that when somebody drops down the speaker at a yellow color or that's on ring 3, when they erected the tower, they opened a little door in the back and they just punched ring 3. They just dial that ring and it calls up the associated delays. (See *Figures 1* and *2*)

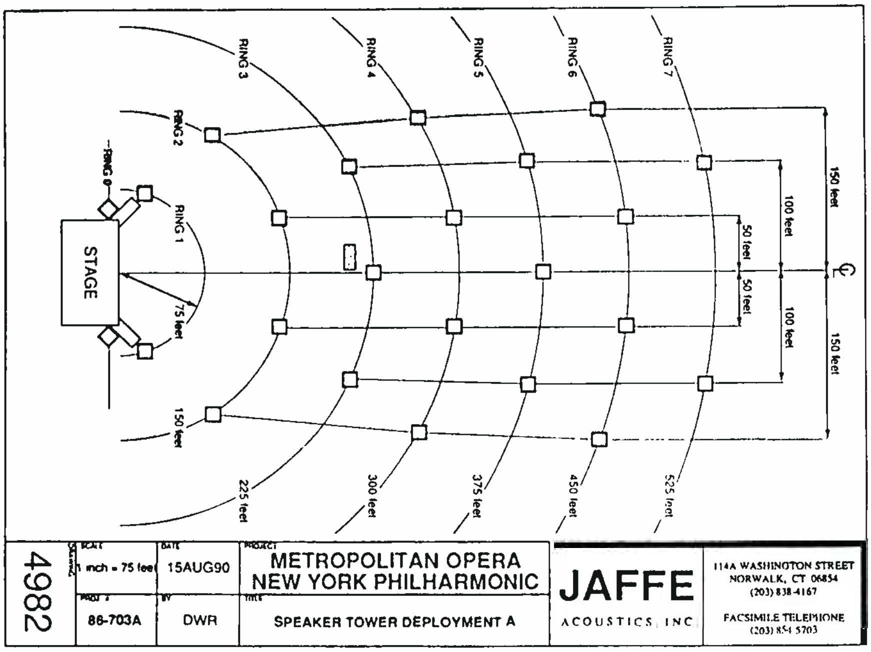


Figure 1. Speaker deployment alternatives. This is a seven-ring vertical spread of speaker towers.

db: My question from this diagram, which shows seven rings which make for a very vertical delay, is would there be situations where you might have five rings more horizontally spread?

DR: Yes there would. There are other ways to do this. Actually, Central Park is the most demanding of all the twenty-four places that the Metropolitan Opera is going to play and that first diagram I showed you was for Central Park, a narrow and very long venue. But the rings work. The general idea is to follow the rings and many of venues they play may use a third to a half of the total number of speakers.

db: But what I'm saying is you don't necessarily have to have seven rings...

DR: Oh no. That was chosen as the distance that would allow us to deal with ring 1 in Central Park. That and being budget driven. I would love to see thirty-two of these towers but that's much more.

db: Well, could you do all of Central Park's various venues?

DR: Much more. I would love to see some more and maybe there will be some time in the future, but now everything else is in place. All the auxiliary and control and broadcast systems are in place, so at this point, if they wanted another tower, it's just a matter of building another tower. You don't have to add anything else. So if they decide that they want to do that at some point in time, then it would be a fairly simple matter to add on to this.

db: Have we said enough about the amplifiers?

DR: No. Not the electronics. The initial design from this office called for operating this system directly off the batteries. We actually had a series configuration of the batteries where we took all six of them, put them end to end, and then referenced the middle point between the third and fourth battery and called that audio ground. And then at that point we could get plus or minus twelve, plus or minus twenty-four, or plus or minus thirty-six to operate all the various electronics. We had conversations with the different manufacturers to see if they could provide systems that would operate like that. We made requirements that the contractor had to modify certain things to conform to this whole package. During the bidding process, two of the three contractors came back to us with their own idea about how to do this and it involved a system called *ENERGENIUS*, a backup. It's thought of, not exactly as a UPS, like an uninterruptible power source, but something you could use when power goes down or in an outdoor application.

Maryland Sound has used these devices before and had a lot of luck with them; there's just no AC available at the place that you need to have audio. So they suggested that perhaps we could go with one of these units that would just take our six batteries in parallel giving that romping 600 amp hours at 12 volts and just convert it all to 120 volt AC. Just run a bunch of the electronics off that. Since both of the contractors came up with this and had experience with it, we decided to allow that in the package. However, during the course of the instruction, we ran into a couple of problems with this system where they were just not compatible with various other pieces in the electronic chain. If it worked, we started getting some noise and some switch sound in the power supplies and we ultimately had to abandon parallel.We were going to use Crest amplifiers. Crest initially agreed to develop an amplifier that could run off DC specifically for this job, because there are other applications for it down the line. So Crest was working fast and furiously on coming up with something that would work. They were interested in the project and wanted to help develop this thing, but time became a problem. They weren't able to finish their design before we had to choose something. We went with *ENERGENIUS* at that point. I then switched over to conventional Crest amplifiers, but they and some of the other electronics did not work quietly enough with the *ENERGENIUS* device. We had to rethink the whole thing and well into the project we changed the whole idea and went to an amplifier that was designed to run off 12 volts. Linear Sound makes it for both the marine and automobile markets. Maryland Sound put together a mock-up, brought it up to New York City for me, and I did some listening tests. Maryland Sound did a great deal of measurement and at that point, that's when they started running down the power supplies to see what the amplifier sounded like as they went from the typical 14 volts (that you get out of a 12 volt battery) down.

That was the process by which we finally went there, so we made that choice about a month after. The Linear Power amps sound fine and they're doing well. For the future, if this facility should ever be used by other types of programming or popular music, I'm not saying it will be, but if for some reason it should be during the course of its life, I think we would need just a bit more headroom in the subwoofer amplifier. Everything else has quite a bit of headroom there, but Linear Power makes another amplifier that would increase that and we would have to swap it out at that point. It has more than enough for what it's designed for right now.

db: Where did you go for other components of the systems?

DR: For the crossovers we went with Brooke/Siren. The FDS 360 has a very well-respected crossover that has many options, but it is not a *tuneable* frequency crossover. You must decide your frequencies ahead of time by cards you plug in. In addition to just frequency points, you can also get various slopes and asymmetrical crossover points; you can get anything you want from the manufacturer. They also have a nice little equalization card that plugs in that allows you to take four channels of fully parametric equalizer. You have to choose what you want: frequency centered bandwidth and gain, or a kind of boost. They will build a fixed frequency version of this thing that then plugs in so you can assign either in groups or singly to the outputs.

db: This is one of the components you have that still has to run off 110V, though.

DR: Yes, that's correct. That and the digital delay are the two things that run off the 110V. I've had a lot of experience with Brooke/Siren; I find them to be very good and probably the main reason we used it is Eastern Acoustic Works deals with them directly. As a part of their design process, Eastern Acoustic Works will provide all of the crossover, slope and equalization information to Brooke/Siren, so they make up a custom set of filters and equalizers that go with particular products Eastern Acoustic Works manufactures. They've done that for a number of their other products and they did that for this one as well. Eastern Acoustic Works called in all the information to Brooke/Siren, the cards were made up specially just for these speaker boxes, and that's the way it came out.

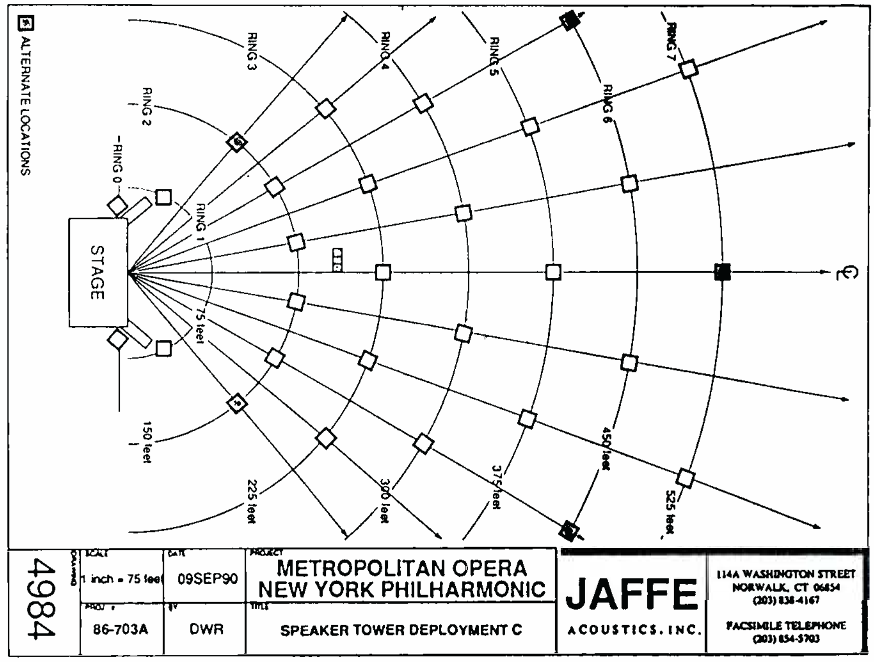


Figure 2. A wider angle than the deployment of Figure 1. Note that seven rings of speakers are still being used.

db: Were there any special precautions made to handle the vagaries of weather they must endure?

DR: Yes, weatherproof is the word of the day here. We hope we are successful. The speaker system itself is interesting. All three of the subsections are covered first. For vandalism, they're covered with a perforated metal grille so they can sustain a bottle thrown at it or a brick or something with little chance of trouble. As a side here, one of the things that I know the last system in the park had a lot of trouble with: they did waterproofing treatments of their different transducer cones and the frequency response of the transducers suffered pretty severely because of it. They made up for it, they changed things, they equalized, they did whatever they had to do to make up for it, but I know it was a real problem.

I was hopeful that we could stay away from actually treating the transducers themselves. What we needed was something between the perforated metal grille and the transducer that was going to be as acoustically transparent as possible, yet would not allow any rain to fall in or be blown in and again, Eastern Acoustic Works came up with something I think that's almost obvious, yet ingenious at the same time. Eastern Acoustic Works came up with a material that is like your typical window screen, a metal screen, only it's extremely fine. The material is a very thin stainless steel strand which is meshed just like a window screen, and there are 10,000 holes per square inch in this material. All the measurements and listening tests we've done are indistinguishable as if there was nothing between you and the speaker. We cannot hear anything. What is claimed is that the holes in the material are so small that water drops can't go through the screen, but water vapor *can* go through it so that is apparently what is happening with this screen in front of each of the speakers. The sound waves are going through it without being affected at all, yet raindrops and particles of rain don't go through. They hit it)and they just run down. Since it's made of stainless steel it's also not going to rust.

At the bottom of each one of the perf metal grilles there's rubber gasketing there so as the rain runs down, it hits the bottom and hits rubber and runs back out of the cabinet. It seemed to be very good during the trial run stuff which was done for several weeks up in Monticello, New York out in the middle of a big field. We were repeatedly hit with some very heavy rains and had no problems at all. We had a little bit of wood swelling, but that was to be expected; no evidence of any kind of degradation due to weather.

db: Finally, we get to the console and the stage.

DR: Yeah, we started talking way back that there were, like, six areas of this thing. We've got to number three. Number four is a stage monitoring system which is a signal is generated from the console at the house mix position, but the amplifiers, equalizers and speakers are all located in a rack on the stage so it makes it easier to deal with. The monitor system has not been requested as of yet by the client, but we felt it was something that we wanted to add because we have several different, I mean, typically in an orchestra there is no monitoring system, but there are several things we might want to do there. Once in the operatics situation with a soloist, we will almost certainly use a spot monitor for the featured performers.

The other thing that we're considering, (just as the audience is receiving ambient information in their various locations and the shell we designed), which you'll see sitting on the stage underneath the roof, is to be like a typical orchestra shell which allows the sound not only to be focused outward for the audience, but to reflect back onto the stage in certain areas so one side of the orchestra can hear what the other side is doing. We're thinking that perhaps for the Philharmonic, we may augment what the shell is doing by taking four or six monitors around the rear of the orchestra and aiming the speaker away from the orchestra into the shells and walls. This will kind of augment that kind of reverberant sound typically found within an orchestra shell which would have ceiling pieces where reflections could come back down.

db: Well probably one of the biggest problems that a large group like the New York Philharmonic would have is the ability to hear themselves.

DR: It's true. That's where Chris Jaffe started all of his work in the mid or early 60s. I guess it was designing orchestra shells. He's pretty widely regarded as one of the people that really knows what the heck an orchestra shell can and should do in the orchestra and the audience. And so that is part of our design here: to try to allow the orchestra to hear themselves as best they can and if in an outdoor situation we need to augment that with some speakers, and I'm not talking about where some people might imagine, "Okay, I want floor monitors sitting right there two feet away from me with a trombone blatting out of it." We're talking about just kind of enhancing, kind of building up the system underneath the roof.

There's a patch bay connecting everything. There are also multiple press feeds. We could give separate stereo feeds to eight different people at the same time.

Okay, that was system four, then we have the fifth thing, which is a full intercom communication system both wired and wireless that connect the control booths to the stage, stage manager, and any remote locations that need to be done. The last, and as a part of that, the client also asked for some huge bullhorns that some of the very largest ones we ended up specifying are from TOA Electronics. They're big old things that sling over your shoulder, you carry around and it's got a little package. That's just to give labor directions and stuff during the setup and teardown, or to be able to shout at somebody a couple of hundred feet away when you can't get them on an intercom or whatever. So far, Peter Wexler, the director, has had a great time while walking around with these bullhorns.

The last system is a performer dressing room system. The performers have several trailers that are provided for them as dressing rooms at every one of the venues, and we also wanted to provide them with program material and also an actual mix of the event. The musicians typically won't be in there while they're out on the stage playing, but I don't know whether family and friends are, or whether certain musicians are waiting in the wings for their piece.

db: Is there a standardized console and standardized requirements for the console?

DR: Yes. I can talk about the house mix area. We've got three chunks of equipment out there: a console, a supply and audio filtering of the AC power rack, and then we have electronic outboard devices. The console itself is the Ramsa C900 Series which we've been looking at a lot for theatrical use because it is a true, left-center-right console, which, if you have a left and right stack and a center cluster in a theater, there are various opportunities for discrete left-center-right mixing. And that's what this console was designed for.

Obviously we will not be using those functions in the park for this system because we're mainly a mono system. So the left-center-right was not much of a concern for this job. However, it is the latest in a series of Ramsa consoles which are gaining good reviews from most people in the sound reinforcement industry as to having a very transparent sound. It has a very nice-sounding mic preamplifier and equalization section. The EQ section is not as elaborate as many of the four-band fully parametric equalizers that are on reinforcement consoles these days, but in classical application, a lot of times we don't even use equalization at all. The less we do to manipulate the signal, I think the better it's going to work for us.

So that's why we chose that particular console. The left-center-right of it had actually nothing to do with this application. We had a fixed amount of space within the control booths because there the equipment is all fixed in the booth, and then the forklift comes out, picks up the booth and goes and puts it on a truck. We had to think about space. Like I said, the second piece of gear out there is a rack which holds the power supply and a spare power supply for that console, and also because there was only one AC power service delivered to the entire control booth which we were sharing with digital information and the lights and the large power drawers for projection.

We chose to have a localized power conditioner for all the audio stuff called a Juicegoose which is a ferro-resident transformer we've had a lot of luck with; I like that a lot. Then the third unit was a whole double rack full of outboard gear that holds all the transmitters for the primary transmitter with its backup, and the secondary frequency transmitter with its backup, various equalizers and intercom. We also have a digital reverb out there by Lexicon—all the little tricks of the trade.

db: So if any job requires special equipment, it can be brought in right there and plugged in.

DR: Absolutely. There's a patch bay connecting everything. There are also multiple press feeds. We could give separate stereo feeds to eight different people at the same time.

db: So they're having directional information even if it's not from the speakers. What have we missed? What have we not talked about?

I think we've given most of the credit where credit is due.

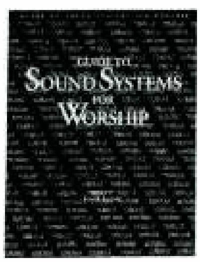
DR: The architects for the entire project are FTL, a New York City firm.

db: Just the letters *FTL*?

DR: Yes, FTL stands for Future Tents Limited where Tents is Tents because one of their main thrusts is that they design things such as the tensile fabric structure that is the roof to this whole thing. To the uninitiated, it kind of looks like a circus tent. It's collapsible, yet extremely rigid and strong. Chris Jaffe, it's obviously his real ideas right from go—his ideas about the ambient sound being able to surround the audience people and provide that concert experience, the orchestra shell design—all of that comes from Chris, and I think I also mentioned Chuck McGregor who worked here at Jaffe in the mid-80s who put together the initial packaging of this thing. For the last two years right through the completion stage here, it's been my baby—I've done everything.

I think that's probably about it.

GUIDE TO SOUND SYSTEMS FOR WORSHIP *by Jon F. Eiche*



This book is written to assist in the design, purchase, and operation of a sound system. It provides the basic information on sound systems that is most needed by ministers, members of Boards of Trustees and worship and music committees, interested members of congregations, and even employees of musical instrument dealers that sell sound systems. To be of greatest value to all, it is written to be both nondenominational and "non-brand-name."

183pp. Paper
$24.95 #1-791

SOUND SYSTEM ENGINEERING, SECOND EDITION *by Don and Carolyn Davis*

Like the first edition, this comprehensive text provides readers with useful information for the day-to-day work of designing sound systems. This updated version contains in-depth coverage that carefully examines acoustic gain, clarity of sound, and required electrical power.

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STEREO MICROPHONE TECHNIQUES
by Bruce Bartlett

This book is extremely timely for sound engineers and video or audio producers. Also, as Digital Audio Tape (DAT) production becomes less costly to use in the field, all electronic media will be trying to achieve the highest level sound production possible. This book tells how to position the correct microphones in the proper locations in order to record optimal quality stereo sound. The many illustrations and clear organization easily explain the theory behind stereo mic'ing methods, and describe specific techniques, including comparative evaluations. In addition, it offers suggestions on session procedures and stereo troubleshooting as well as recent developments in binaural and transaural stereo and stereo boundary arrays.

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CONCERT SOUND AND LIGHTING SYSTEMS *by John Vasey*

This book shows how to set up, maintain, and operate sound and lighting equipment for the performance of amplified music or any kind of touring production. An excellent reference and/or guide to procedure, the book provides descriptions of all the components that make up a system, explanations of how they all work together, and photographs and illustrations that show specific equipment and proper stage setup.

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$27.95 #4-791

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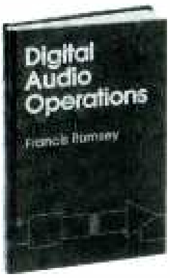
This brand-new second edition has been updated to include the latest in MIDI, cinema sound, tranformers and compact discs. Readers learn the new developments in audio electronics, circuits, and equipment. There is also an in-depth examination of disc, magnetic, and digital recording and playback.

1,400pp. Hardcover
$99.95 #5-791

DIGITAL AUDIO OPERATIONS *by Francis Rumsey*



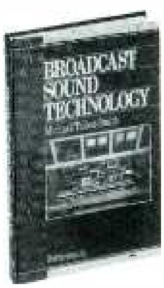
Leaving the higher levels of theory to other digital audio texts, this handbook emphasizes principles for the studio and those aspects of digital audio appropriate for day-to-day sound engineering operations. It describes the sampling process, error correction, editing systems and different recording options. This book is written to help producers and engineers in the studio get the best possible results from the high quality standard equipment in use today.

256pp. Hardcover

$39.95 #6-791

BROADCAST SOUND TECHNOLOGY *by Michael Talbot-Smith*

This is an introduction to the technical aspects of sound in radio and television. It examines in detail the main items in the broadcast chain: studio acoustics, microphones, loudspeakers, mixing consoles, recording and replay (analog and digital), and the principles of stereo. It offers a technically solid treatment of audio principles and broadcast hardware.



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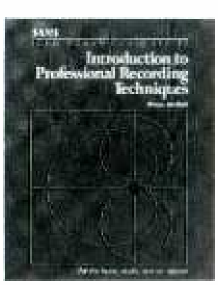

Geared primarily for the aspiring professional, this book provides a comprehesive discussion of recording, engineering and production techniques. Special coverage of microphones, microphone techniques, sampling, sequencing, and MIDI is also included.

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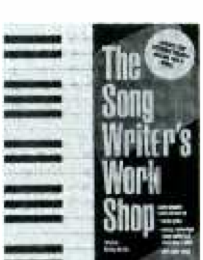

This book and cassette meets the songwriter on the level that he practices his craft—through sound. Beginning with an idea for a song, it travels through inspiration and creativity, writing lyrics, making a demo, understanding MIDI and how to pitch songs to the industry. Each lesson is to be learned through reading and also hearing the lesson, and is taught by experts such as John Barilla.

96pp. and 2 cassettes, Paper

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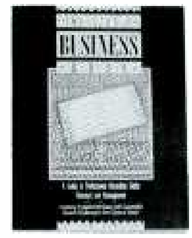
THE STUDIO BUSINESS HANDBOOK: A GUIDE TO PROFESSIONAL RECORDING STUDIO BUSINESS AND MANAGEMENT *by Jim Mandell*



Here is a comprehensive survey on the state of recording studio business and management in the nineties that includes startup and equipment cost comparisons from low budget to world class operations; equipment purchasing strategies; rate-setting factors; actual examples of pre-session contracts; how different studios handle billing, credit applications, payment guarantees, conflicts and collections; how to write publicity releases that will get into print; what to avoid in advertising;

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This handbook covers the entire range of sound recording and emphasizes the technology of the field as well as the aesthetic aspects of actual recording. It features a sequence of six chapters covering the basic tools of the trade, from the acoustical recording environment through the reproducing environment. Eargle details the actual production decisions which are made in classical and popular recording. Also covered are the physical and operational principles of tape, disc, and digital recording, as well as the economic and physical aspects of the low-cost studio.

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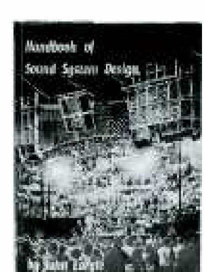

In one complete up-to-date book here are all the practical as well as theoretical aspects of sound reinforcement. The detailed chapters include information on electrical fundamentals, acoustical fundamentals and psychoacoustical aspects; high-, low-, and mid-frequency systems; microphones in sound reinforcement and system architecture; central loudspeaker arrays, distributed systems, speech reinforcement and paging systems; system intelligibility; high-level sound reproduction, a theater sound overview, and sections on live music reinforcement, line arrays and sound columns.

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This new edition has been accepted as the long-awaited replacement to the original book published in 1976. The new edition is not "old wine in a new bottle." The revision has been done by Professor Kefauver. He is the Coordinator of the Recording Arts and Sciences Department and Director of Recording at the prestigious Peabody Conservatory of Music. The book is used by most of the recording schools and universities here and abroad. This book contains all the basics for the recording studio engineer, as well as more advanced information covering MIDI, Automated Consoles, SMPTE Time Code and Digital Audio. This book has been and remains the "bible" of the audio industry.

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LAB REPORT

LEN FELDMAN

Lab Report: Shure Model FP410 Portable Automatic Mixer

GENERAL INFORMATION

● The Shure FP410 Mixer attempts to solve a problem in multiple microphone use that has plagued audio professionals ever since multiple mics were used at a single event. If you use several mics at once, with each turned up while a single talker speaks, audio quality is degraded in many ways. For one thing, there's more background noise picked up as well as more reverberation. Furthermore, sound from the talker arrives at the various mics at different times. When these mic signals are mixed, they produce an effect known as comb-filtering (cancellation at some frequencies, reinforcement at others, all across the audio spectrum). Amplified or recorded sounds then often sound hollow, diffuse and thin. Until now the only solution was to keep opening and closing mic channels manually, and we all know what problems that can cause, such as when a speaker suddenly wants to say something at a conference or seminar and finds that his mic is not turned up.

Shure has come up with an innovative and technically sophisticated solution in this little four input mic/line mixer. The FP410 reduces comb filtering by automatically keeping unused mics attenuated by about –13 dB (not completely off—to make mic activation sound smoother). When a given mic is needed, the FP410 raises it from its attenuated state to a preset activated state in less than four milliseconds. Shure calls this a "NOMA" circuit, and the acronym stands for *Number of Open Mics Attenuator*. This circuit not only solves the problem of comb-filtering, but actually increases gain settings possible before the onset of feedback. Normally, to avoid feedback, many PA systems are operated 5 at 10 dB below the point where the system begins to howl. This safety margin is reduced every time another mic is opened. The NOMA circuit comes to the rescue once again. As more mics are activated, the overall gain of the mixer is lowered.

Other features of this little mixer include transformer-balanced inputs and outputs, using XLR connectors, switchable 14 volt or 48 volt phantom powering, linking capability for up to 25 mixers and an AC operating voltage range of from 80 to 132 volts AC (or, overseas, from 160 to 264 volts AC). LED indicators show input levels, output levels and limiter action. A defeatable *Last Mic Lock-On* circuit keeps the last

Figure 1(A). Frequency response of the line input with and without the low-cut filter activated.

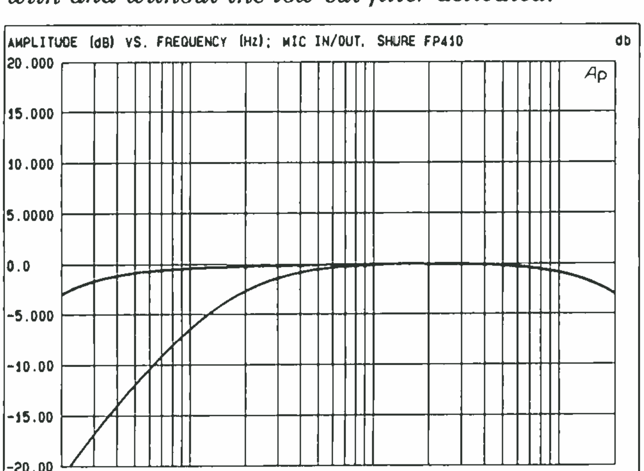
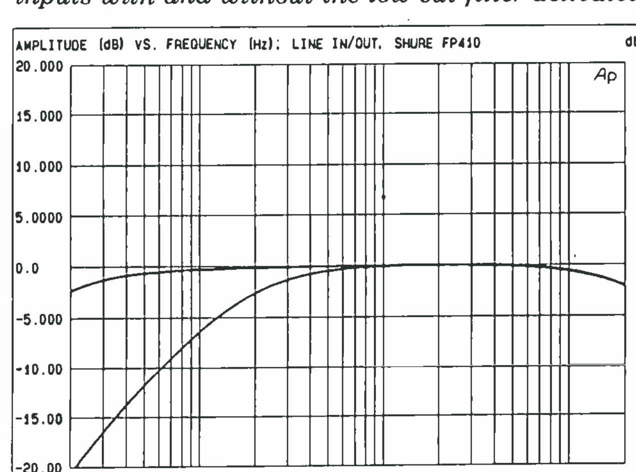


Figure 1(B). Frequency response of the microphone inputs with and without the low-cut filter activated.



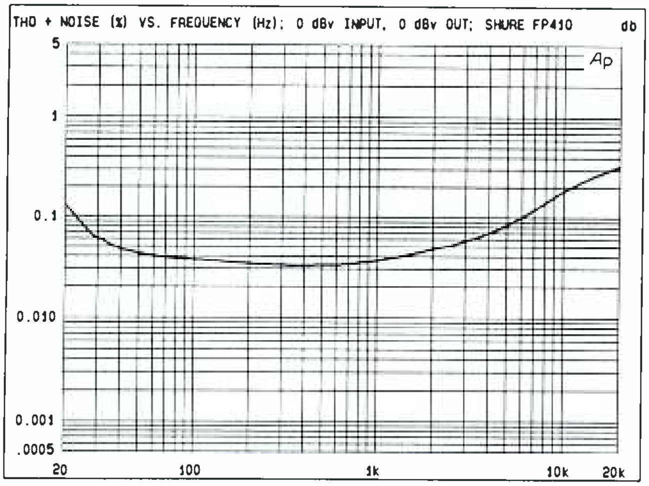
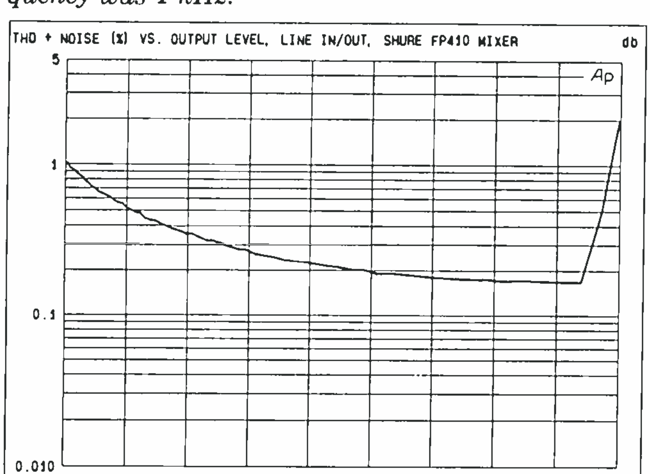


Figure 2. Harmonic distortion plus noise versus frequency with line in/out at 0 dBu input. Controls adjusted for 0 dBu output.

used mic activated to maintain a consistent background ambience. The mixer can be operated using the NOMA (automatic) feature or you can operate it manually on those rare occasions when manual operation would be preferable. For portable operation, the FP410 can be powered by two standard 9 volt batteries. The LED VU meters can show peak or VU readings, as you choose.

When you open up the battery compartment, you also gain access to several modifiable functions changed by altering the settings of several dip switches. These include being able to vary the amount of automatic off-attenuation of non used mics from –13 dB to infinite attenuation, changing the *hold* time (the time a mic stays on after the talker stops talking) from 0.4 seconds to 1 second, defeating the *Last Mic Lock-On* feature described earlier, changing limiter threshold (from factory set +16 dBm to either +8, +4 or 0 dBm, changing VU meter calibration—from +4 dBm = 0 VU to +8 dBm = 0 VU), and changing phantom power voltage from 14 to 48 volts. Many additional modifiable functions are listed in the operating manual, but these additional functions should be modified only by

Figure 3. Harmonic distortion plus noise versus voltage output. Measurement made with channel and output controls at maximum. Test signal frequency was 1 kHz.



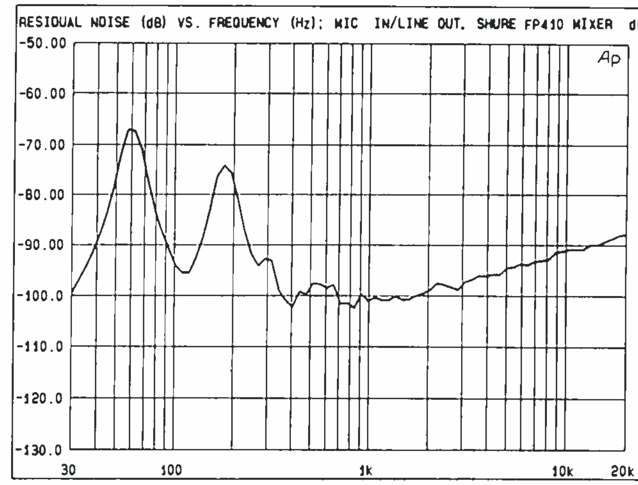


Figure 4(A). Spectrum analysis of residual noise. Mic/line out referenced to 5 mV input, controls set 500 mV out.

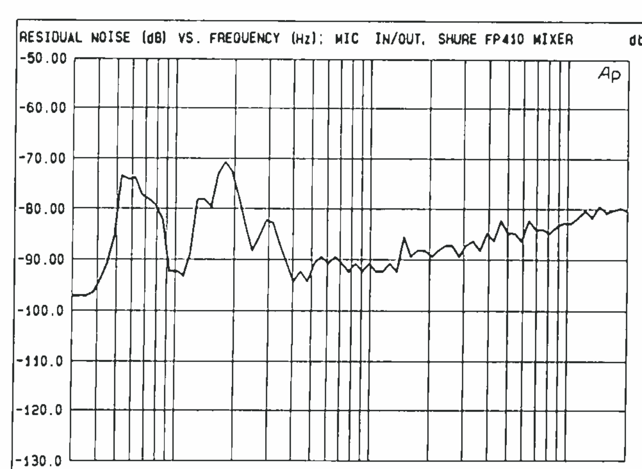
a qualified technician as they involve more than just changing the setting of a dip switch.

CONTROL LAYOUT

Four input level controls are positioned along the left portion of the front panel of the FP410. Each is augmented by one green and one red LED to indicate normal or high input levels. Also adjacent to each input level control is a tiny slide switch that activates a low-cut filter having a 6 dB per octave roll-off below 170 Hz. The filter is useful in reducing undesirable noise such as wind noise.

The master gain control is located centrally on the panel, and if you pull its knob, a 1 kHz test tone is activated. Output level Peak/VU LED metering is arranged to the right of the master gain control in the form of 11 LEDs (7 green, 1 yellow and 3 red), and immediately to their right is a switch that selects VU or Peak mode for the meter. Three switches below the LEDs include a momentary *Battery Test* switch that operates in conjunction with the green LEDs, a *Manual/Auto* switch, and a limiter *On/Off* switch with its associated red LED indicator. Further to the right are

Figure 4(B). Spectrum analysis of residual noise. Mic-In/Mic-Out referenced to 5 mV input with controls set for 5 mV out.



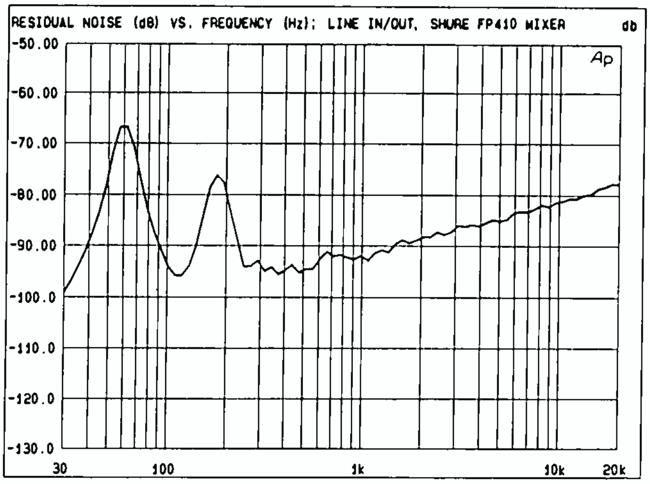


Figure 4(C). Spectrum analysis of residual noise Line In / Out. Referenced to 500 mV input, with controls set for unity gain.

¼-in. and 3.5 mm phone jacks and a headphone rotary level control. Pulling on the monitor switch applies signals from the rear panel monitor-in jack to the phones rather than the mixer output signal. A power on/off slide switch and a power on indicator LED at the right end of the panel completes the front-panel layout.

The rear panel is equipped with a power cord receptacle (line cord is supplied separately). Two 3-pin male XLR output connectors for connection to one or two amplifiers, recorders or other mixers come next, each of which can be switched to either line level or low-impedance mic level using adjacent slide switches. Eight-pin *Link* in/out connectors permit virtually an unlimited number of FP410 mixers to be stacked when more inputs are required. Miniature 3.5 mm *Tape Out* and *Monitor In* jacks are positioned on the rear panel above the *Link* connectors, as is the phantom power on/off slide switch. The remainder of the rear panel is given over to the four input 3-pin female XLR connectors, each of which has its own adjacent mic/line slide switch. Along the right side panel of the unit is the battery compartment cover which is easily removed to gain access to the previously described dip switches, and for battery installation and replacement.

TEST RESULTS

Very few measurements were required to confirm the excellent performance of this little mixer. *Figures 1(A)* and *1(B)* show the frequency response of a typical input with and without the low-cut filter turned on. Response in the mic-in/mic-out mode (see *Figure 1(A)* was virtually identical to that obtained for operation in the line-in/line-out mode (see *Figure 1(B)* with response down around 2 dB at 20 kHz for the line mode, and about 3 dB at 20 kHz for the mic in/out mode.

Total harmonic distortion at 0 dBV in and out was well below the Shure specification of 0.5 percent at any frequency, as is shown in *Figure 2*. At mid-frequencies, THD + noise was between 0.03 and 0.04 percent, and even at 20 kHz, THD plus noise remained a low 0.31 percent. *Figure 3* is a plot of THD plus noise versus output level, using the line-in/line-out mode. Input and output control were set to maximum and inputs signals were varied so as to produce outputs ranging from 1 volt to 10 volts. Evidence of clipping occurred at about 9.3 volts.

We measured signal-to-noise ratios in several different ways. As specified by Shure, we turned down both input and master controls and obtained a S/N reading of 98 dBV. With the channel control still fully counterclockwise and the master control opened clockwise, S/N decreased to a still very acceptable 73 dBV. Other S/N measurements involved adjusting channel and output controls to specific points. For example, with a channel control set for mic input and line outputs adjusted for 500 mV out when a 5 mV input is applied, A-weighted S/N measured 81.15 dB. A spectrum analysis of the noise content under these conditions is shown in *Figure 4(A)*. For mic in/out settings, with controls adjusted to deliver 5 mV output for 5 mV input, S/N measured 74.13 dB. An analysis of noise content under these conditions is shown in *Figure 4(B)*. Finally, with 500 mV applied to an input and controls adjusted for 500 mV output (in the line-in/line-out mode), S/N measured 72.68 dB. Spectrum analysis of this condition is shown in *Figure 4(C)*. Note that in all three figures, the major components contributing to the overall S/N figure were power supply frequency components (60 Hz, 180 Hz, etc.) rather than more annoying random noise.

CONCLUSIONS

While we were not in a position to check out the effectiveness of the NOMA circuit in an actual field test, we were able to confirm the action of this circuitry by connecting as many as three different mics to the unit in our lab and observing the action of this circuitry as we moved from mic to mic. Any doubts that we may have had when reading about this feature were quickly dispelled. With a 4 millisecond turn-on time, not even the first syllable of a talker's speech is likely to be lost. Furthermore, the smoothness of the action is such that you will not even realize that the mic is being voice activated. We attribute that, in part at least, to the fact that unused mics are not fully turned off, but only attenuated by about 13 dB. That seems to be enough to overcome the problems associated with too many mics being active at once in a PA environment, yet not enough to cause the kinds of switching anomalies normally associated with voice activated switching circuits.

All in all, we strongly feel that this technology is well worth incorporating in a whole class of mixers—not just in this handy portable relatively low-cost unit. Incidentally, Shure wisely printed five basic steps for optimal FP410 performance right on the top cover of the unit. We followed the instructions, and so should you if you end up owning this cleverly-designed mixer. Start by setting the mixer to manual and then adjusting each input until its red LED flashes occasionally. Only then do you switch to the automatic mode and adjust the master volume control to suit your needs. That explains why the manual mode is there in the first place—considering the fact the whole reason for this mixer is that most of the time you'll want to use the automatic mode. db

VITAL STATISTICS

SPECIFICATIONS	MFR'S CLAIM	db Measured
Frequency Response (25 Hz–20 kHz)		
Mic-Mic Mode	+0.5, –2.0 dB	+0, –3.0 dB
Line-Line Mode	+0.5, –2.0 dB	+0, –2.0 dB
Input Clipping Level		
Mic/Line/Monitor (dBV)	–15/+26/+21	–14/+26/+22
Output Clipping Level		
Mic/Line/Tape/Phones	–31/+19/–2/+6	–30/+20/–2/+6
Output Noise		
Master & Inputs CCW	–88 dBV	–98 dBV
Master CW, Inputs CCW	–71 dBV	–73 dBV
Hum and Noise (A-weighted)		
Mic In/Out, re: 5 mV	N/A	–74.13 dB
Line In/Out, re: 500 m	N/A	–72.68 dB
Mic In/Line Out	N/A	–81.15 dB
Channel Activation		
Attack Time	4 msec	Confirmed
Hold Time	0.4 or 1 sec	Confirmed
Decay Time	0.5 sec	Confirmed
Low-Cut Rolloff	<170 Hz	Confirmed
Limiter Threshold (dBm)	+16,+8,+4 or 0	Confirmed
Power Requirements	100-120V 50/60 Hz, 8W	Confirmed
Battery Life	Approx. 12 Hrs.	N/A
Dimensions (HxWxD, inches)	1-¾ x 14-½ x 8-¼	Confirmed
Net Weight	5 lb. (2.27 kg)	Confirmed
Price:	$1595.00	

SOUND REINFORCEMENT

ROBYN GATELY

Reflections on Building a Career

● It recently occurred to me that twenty years have passed since I did my first paid show. Admittedly, I was in high school and only got $15, but it was the start of something that has supported family and friends for many years. As a result, I have given some thought as to what it takes to prosper in this industry.

Even after twenty years, I still love music. I love the feeling of working with it, of putting my hands on it and shaping it. I love adding my input to something and making 20,000 people go 'YEAH!'

A good sound mixer can take a band with no sound check, and inside of a minute make the place rocking, and within a song and a half, make you think you've got the record at 110 dB. Of course this takes experience, but it is the kind of pressure a great live mixer loves, and with my current act, it's the kind I get all the time. When you have taken a band from ground zero to the Stratosphere in one song, you can be sure the cheers at the end of the song are for you, too; and you can be sure that a shiver still goes down your back, even after twenty years.

You have to listen to and like *all* kinds of music if you want to survive in this business. I couldn't have survived this long if I wasn't intimately involved with many kinds of music: Rock, Blues, Jazz, Folk, Anything. Yet, this has led to so many highlights in my life that I sometimes can't believe the history of my career myself. Sometimes when I tell people some of the things that have happened to me, I think I really have lived everyone's teenage dream of the perfect life.

LONG TERM MEMORIES

I am constantly surprised at how many irreplaceable friends and memories I have. You can't work with every single star in this business, but you can work with too many to count! Years ago, James Brown liked the job a co-worker and I did so much that he brought Foster and I out on stage, introduced us to the audience, then had us bring 'our women' (as he called them) out, and had us all take a bow. Afterwards he danced with 'the women' on stage, and then finally, as Snagglepuss would say, it was time to "Exit Stage Left." (Of course, the inside joke on television is that Snagglepuss goes to the Audience Left after such a statement, moving in the opposite direction from the way he really should.)

When the show was over, James called me into the dressing room and tipped everyone $100. Discussing the night afterwards, we all figured that no night could ever be as magic as that. Boy, were we ever wrong!

Instead, my life has been a series of peaks as wonderful as that night. Could there be one single peak in a life when just a few of the headlines include: helping Duran Duran get ready for Live Aid, doing enough shows with Sammy Davis, Jr. that he offered business advice, many magic nights with James Brown, taking one of the world's premier singers, Judy Collins, everywhere from the Tonight Show, Disney Channel and Showtime to arenas with 15,000 people. And along the way, I've mixed acts like B.B. King, Wynton Marsalis, and many, many others to crowds as large as 65,000. Then this past summer, I was asked by The City of New York to be the "official" city sound engineer, making me the first soundman ever appointed as the result of a Supreme Court decision.

ADVICE

So, what advice could I possibly give someone else who wants to live a dream like mine? *Don't Give Up.* I am constantly surprised by how many good people in our business disappear. To most people, the music business is not about making millions of dollars. It's about creating something that affects you *deeply, spiritually*. It is impossible to effectively relate the feeling of 20,000 people cheering your work. There is no other business where thousands of people will immediately evaluate your work and render their decision in no uncertain terms.

I am amazed at how many people drift into our business instead of actually preparing for it. The entertainment industry is a hard business to break into. It requires the same sort of planning that comes with becoming a doctor or lawyer. An awful lot of kids see someone on MTV making the big bucks at age 21 and think that is how the music industry works; you get discovered and you're a star (for about 15 minutes). Instead, the music industry is really comprised of many individuals who slowly built their careers from part-time to full-time; people who spent years in the doldrums waiting to show they have what it takes.

What does it take? Mainly it takes talent, knowledge, and "Balls With A Capital B." The first two should be obvious, but how will you react when you need to run out in front of 15,000 people to fix something? Will you be able to speak intelligently and confidently when dealing with your first multi-platinum performer?

It's hard to remember that people are looking to you for answers when you've never done something before. It takes a lot of "Balls With A Capital B" to answer confidently and knowledgeably about something you are just really doing for the first time. I remember my first "real" job in this business. I saw an ad in a newspaper for a soundman to work with a now world-famous act. I figured, 'what is a manager going to want out of a 20-year-old kid?' Nothing, unless you are giving him *answers*.

Now, I knew that every soundman in the world was going after this job; I knew that everyone was going to say, "I've seen XXXXX, I really want to work for him. I think he's going to be a star." So, I walked in, sat down and said, "I've seen this show a dozen times and this, this and this is wrong with the show, this is wrong with the staging, this is wrong with the lighting, and this, this and this is wrong with the sound." Within three minutes, the manager had sent all other applicants home. Why? Because, even though he didn't agree with me on most points, I had opinions and was willing to share my ideas to try and make a better show.

RULES

So, after this much time, I can offer some advice on how to survive and avoid the pitfalls of the business. Thousands of small rules won't help you, but a few general ones work all the time:

Get as much sleep as you can on the road and stop eating all those cheeseburgers! Exercise.

Force yourself to take time for your family, because you never know what's going to happen.

Be a Boy Scout—Be Prepared! Not just with every tool in the business, but mentally.

Try to be nice. This sounds pretty basic, but things can get pretty frazzled out on the road. Smile. Try to tell a joke a day.

(One important note: There is a certain ego trip that goes along with putting your hands on 50,000 watts. Sure, you have enough power to move people to ecstasy, but you also have enough power to hurt people. There is a certain responsibility to what you are doing. It becomes imperative that you learn quite a bit about system design before you just turn it up so it Rocks The House. The cabinets that are closest to the audience can really hurt some people, so it becomes important that you learn about system design and measurement before you think that having a great sounding mix is the ultimate answer. And don't forget those are your ears you're assaulting, too.)

There is no way of knowing whether you'll survive in our business. There is only one guaranteed reward—personal satisfaction. But if you stick with it, plan your moves and make sure it's obvious you like what you're doing, one day you're going to realize that your "hobby" has turned into the most fun career you could ever imagine. db

The Supreme Court decision in The City Of New York versus Rock Against Racism is the only sound decision to have ever gone to the Supreme Court. The outcome of the decision is that The City Of New York received permission from The Supreme Court *to run any sound system that is on the city's property.*

The case is a perfect example of where being nice is important, because if the original soundman had done so, the case never would have gone as far, nor cost the money it did.)

Hot Tips: Designing Vocals: Part II

● Just as a coin has two sides, designing vocals also has two sides: the technical side and the performance side. They are really indispensable to each other, so it is a fruitless exercise to speculate on which factor is most important. Sometimes a great vocal performance can be mired in a bad recording—perhaps a harsh EQ, uncontrolled dynamics, noise or distortion. It can be subtle enough that it will not be immediately noticeable, but such things will quietly turn a listener off. It is like looking at a beautiful mountain view from a chalet with dirty windows: it really spoils the impact! Hopefully, in **Designing Vocals: Part I** we explored the technical factors one needs to keep in mind in order to (as it were) "keep the windows clean." Technical factors can sometimes be complex, but they are pretty much cut and dried: it's just learning various procedures and how to use certain devices. Black boxes and acoustics don't change much; once you establish good recording habits, you are pretty much set for life.

However, the other side of the coin—the performance side—is much more elusive. Here we are not dealing with the predictable flow of electrons through wires, across capacitors or within the maze of integrated circuits; we are dealing instead with real flesh and blood human beings, each individual having a unique combination of intellect, emotions, hang-ups and varying degrees of talent. I don't know of any books written on this subject, and I think I understand why: it is complex. It's much easier to generate another volume on how to push up faders than to deal with the intricacies of the human personality in the recording studio. Yet (like it or not), the reputation of your studio is inextricably tied to how you handle the human factors. If you as engineer and/or producer cannot nurture a good vocal performance from your clients, your pristine sound will never get noticed. To extend the imagery we used earlier: it's like looking at the scene of a squalid, urban slum through a pane of squeaky clean glass. No one will ever comment on the glass, unless the person has a highly developed sense of irony. The bottom line is that both technical and performance factors need to be considered when designing vocals; and of the two, eliciting a good performance is probably worthy of greater attention because it is tougher to learn how to do.

PERFORMANCE ENGINEERING

I freely admit to having fabricated the term "performance engineering." It doesn't exist in the dictionary, nor can you get a degree in it from any known university. I just stumbled on this descriptive term one night while recording a female vocalist. She was having a rough time singing a particular passage in a song; the difficulty was making the transition between her lower range (the "full" voice) and her upper range (the "head" voice). She tried and tried, listening back to each pass for signs of improvement, but the tape was cruel in revealing the uncertainty in her voice. She had an otherwise lovely voice, but the entire track was in jeopardy because of this one blemish. Realizing that she had all the right stuff, but was somehow not putting it all together, I took her aside to encourage her. Straining for an illustration to explain her dilemma, I told her to consider how she would go about building a bridge between her lower and upper registers. I explained how there was a process involved, decisions to be made, matter and energy to be manipulated to achieve the goal. If she were building an actual bridge over a body of water, she would have to decide exactly where to build the bridge, where to sink the piers on which the bridge would stand, how long the span would be, how high, and so on. Some formulas would have to be consulted, some numbers punched in a computer and a working model built and tested, a few numbers and materials shifted around—but eventually, a bridge would be built. The dimensions, procedures and materials might be up for discussion, but the bottom line was that the bridge could be constructed; she simply had to figure out how.

That's the way an engineer would look at it. An engineer would take the known and apply it to the unknown in a creative way; an engineer would manipulate matter and energy to achieve a desired goal. That's the root function of anyone that calls himself an engineer—from the graduate electronics specialist who designs microprocessors to the bulldozer operator who moves around tons of earth. Almost nothing is impossible to an engineer; he just figures out a way to do it. This young lady learned a valuable lesson that night. She realized she had the tools to do much more than she ever imagined. By taking a few minutes to study her range and the phrasing of the song, she figured out where her lower range started to fade and where her upper range crossed over; she decided on which notes to take in full voice and which to take in head voice. She rolled up her sleeves and built a bridge, and in a couple of passes she was blowing through the song smoothly—like a seasoned professional. That's what I mean by performance engineering. And this "can-do" kind of attitude needs to be applied to every aspect of vocal design. Here are some other aspects of performance engineering you need to consider.

SETTING THE MOOD

The old proverb holds true here: "An ounce of prevention is worth a pound of cure." It is most definitely preferable to set the mood in advance than to try and fix it later on—after it has gone sour. Vocalists generally need a stable, pleasant environment in order to relax and open up in front of a microphone. It needs to be neutral—not emotionally charged—to be able to transform itself (via the imagination) into anything he or she wants it to be; and believe it or not, it's not all that hard to achieve. This area gets fudged a lot, particularly on the level of the home studio, but a few little considerations can make all the difference in the world. For example, proper ventilation and temperature control is a must. How can the vocalist sing a sophisticated love song when he is sweating profusely and the studio smells like a locker room?

Each of the senses feeds the artist with information on whether to be relaxed or uptight. It's no secret that being uptight contributes to muscle tension, muscle tension affects the vocal chords, and so on. Hence, the more elements in the environment under conscious control, the better. Here are some of the elements you might want to think about:

COLORS

They really ought to be neutral and natural, not drawing attention to themselves. If the walls are screaming with super-saturated colors, it can be distracting to a vocalist trying to imagine a particular setting.

LIGHTING

This is another key factor. Indirect lighting works best, and it can cover a multitude of eyesores in your studio. It really doesn't take much to install some track lights; even a couple of clip-on lamps will do the trick. A subtly-colored bulb projected on a neutral wall can make even the humblest basement studio seem like a well-lit stage.

SMELLS

Smells are an extremely important part of the environment. Physiological psychologists tell us that the senses of smell are mapped onto a very primitive part of the brain (the rhinencephalon). Smell does not register on the cortex (the outer part of the brain associated with consciousness), but rather on the inner part which is associated with instinct. Subtle smells can put people on an emotional trip that takes them back in time to wonderful or traumatic experiences. Once again, neutrality and naturalness are the watchwords. Keep the air circulating and even spruce it up with a negative ionizer (it turns oxygen molecules into ozone by applying an electrical charge to the air) or perhaps some good quality air freshener will do the trick. I may seem a bit of a hippy, but I always burn just a little incense in the studio (*not* the control room) before the client arrives. It's so subtle as not to be obvious, but I always get lots of compliments on what a pleasant, relaxing environment I have in my studio. I am sure tuning up the air has something to do with it.

CONTROLLING THE VOCAL SESSION

There are a few techniques that can really make the vocal session a success. Remember that you, as producer or engineer, play a major role in the vocal. It is a team effort. A vocalist may sing well on stage or in the shower, but he may clam up in the studio. Since you get part of the glory (if the recording is successful) or part of the blame (if it's terrible), you have a vested interest in making even the most naive, unprofessional vocalist sound great. Whether the vocalist be amateur or pro, here are some things to consider:

PREPARATION

This should be done if at all possible. If there is a day or so between the cutting of the tracks and the vocal session, give the vocalist a work tape to practice with. That way, he'll have made lots of creative decisions at home, and when he comes to sing, he won't be drained; he'll be able to focus more energy on performance.

WARMING UP

Warming up the vocal chords is extremely important, since they are muscles. The warm-up can be simply letting the vocalist rundown the tune either live in the control room or in the studio on headphones. I prefer a few passes live in the control room, because it gives an opportunity to hear the performance candidly—without facing the mic. A low-key critique can be made at this time and it will not be perceived as threatening or judgmental, because it is admittedly informal. After a few passes and some exchange of ideas, the vocalist can then proceed to the mic knowing that his/her approach is "in the ballpark."

THE HEADPHONE MIX

It must be conducive to singing accurately. It must be adjusted on the basis of feedback from the vocalist or perhaps on the basis of your own perception of what needs to dominate the mix. Always remember that the purpose of the headphone mix is not to sound pretty—like a final mix, but rather to facilitate a good performance. That may require temporarily abandoning aesthetics in favor of utility. For example, if a vocalist seems to have trouble keeping time, then perhaps it would be best to sneak the drums in gradually, louder and louder, until the rhythm problems are rectified. This should be done as unceremoniously as possible; simply raise the drums a little louder every time you rewind the tape for another pass. Chances are, the artist will not notice the difference, but will imagine that he is "getting into it" a little more with each run through the song.

Another factor to consider is the overall level of the vocal in the context of the mix. Sometimes the artist will hear himself too predominantly in the mix, but instead of asking you to turn his level down, he may simply sing in an extremely reserved fashion. If the song is an up-tempo rocker, this can render a rather insipid performance. It's advisable in this case to pull the vocal level down and slightly under in the mix, so that the vocalist will need to strain a bit to hear himself. The exact converse may also be true, at times. The important point is to sense the problems in performance and experiment to see if they can be remedied through a modified headphone mix.

BREAKS

Breaks are a necessary part of the creative process. A wise producer/engineer knows when to call for a break: precisely when the vocalist needs one, but doesn't want to ask. Usually it's best to administer a break with a brief word of encouragement: "Sounds like you're just about ready to lay down a track. Why don't we take a short break, then come back and knock it off?" If the vocalist asks for a break, it might connote a fear of failure, so she will probably just keep hacking away at it until her voice is gone. As with any creative process, a lot is achieved during a refractory period. Writers and painters sometimes speak of taking a nap and waking with solutions to their creative problems. Getting away from it all—even for a few minutes at the water fountain—is sometimes all that is needed to regain perspective on a performance that is missing the mark. And remembering also that the voice is muscle, it can rebound rather quickly if given a few minutes rest.

DYNAMICS

They must be built into a performance if it is to sound realistic and inspired. Unfortunately, the necessities of recording pop music on tape (limiters, compressors, etc.) can sometimes iron out too much of the natural dynamics—although settings can be adjusted to minimize this problem. Perhaps the biggest enemy of natural dynamics is the very process of multi-track recording. Say, for example, a song has three verses. After listening back to the entire song, it becomes clear that the vocal performance on verse one is not up to par. So, it is decided to re-record verse one. By now the vocalist is hot, so verse one should be easy to cut —and it is. But an interesting phenomenon happens. Because the vocalist is now in high gear, verse one now sounds too animated relative to the verses that follow. (I'm not talking about volume. That could be remedied by moving a fader. I am referring instead to the *je ne sais quoi* of vocal performance which is not fixable by technological means.) If a song is going to have a crescendo (an increased intensity), it will generally occur towards the end of the song—definitely not at the beginning. So in this case, in order to assure natural continuity between verses, one of two things must be done: either verse one must be performed again in a more subdued mood, or else verses two and three will have to be performed again with increased intensity beyond the standard set in verse one. While it is true that some types of music are deliberately uni-dynamical (such as techno-pop and dance music), most other kinds of pop music productions will benefit from an objective scrutiny of the dynamics and continuity of vocal performance.

In the next installment of **Designing Vocals,** we will review some hot tips on treating vocals in the final mix. db

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db BUYER'S GUIDE

Microphone (including Wireless) Microphones, Recording Tape and Tape Accessories

On the pages that follow you will find a Guide to Microphones in chart form. This is followed by a Guide to Wireless Mics in paragraph form. This is, in turn, followed by paragraph-form Guides to Recording Tape and Tape Accessories. Manufacturers' addresses conclude the Guides.

MICROPHONES

Model	Type	Patterns	Frequency Response dB, ±dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
AMS INDUSTRIES, INC.												
ST 250	Cond.	Var.	20-20k 3	100	27	135	6.5 2.5	8 oz.	black	XLR	$3800.00	Based on a four-capsule array capable of remote manipulation of both polar patterns and stereo image.
MK IV	Cond.	Var.	20-20k 3	100	27	140 at	9.5 2.5	18 oz.	black	XLR	$5850.00	Has a processing unit that can electronically and remotely position the mic for live recording and post production.
AUDIO-TECHNICA U.S INC.												
ATM35	Cond.	Card.	30-20k 3	200	44	145 1	1.39 0.47	5.4		XLR	$280.00	Comes with AT8418 instrument mount, with built-in foam windscreen and "rubber grip" spring clamp.
ATM6 1HE		Dyn card.	Hyper 3	50-18k		600	55 1.98	7	9.7	XLR	$250.00	Hi-Energy design provides high sensitivity superior internal shock mounting. High output with low handling noise.
AT825	Ster.	X/Y Stere	30-20k 3	200	46	126 1	8.43 2.44	8.5	low reflec. matte	XLR	$95.00	Equipped with a pair of closely matched UniPoint miniature cardioid elements. Switchable low-cut filter, windscreen, two-way power.
AT804	Dyn	Omni	50-15k 3	600	53		5.94 1.42	7.5	low reflec. matte	XLR	$95.00	Hand-held omnidirectional designed for interviews/sportscasting. Rugged construction for field applications.
AT4031	Cond.	Card.	30-20k 3	200	46	145 1	6.3 0.83	4.9	black paint	XLR	$330.00	Pressure gradient capacitor, low self-noise, high SPL capability, 9-52V phantom.
AT4053	Cond.	Hyper card.	20-20k 3	250	36	144 1	6.1 0.83	4.2	black chrome	XLR	$640.00	Direct-couple design, low self-noise, high SPL capability. Uniform polar pattern at all frequencies. Interchangeable element capsules.
AT4071	Cond	Lobar	30-20k 3	250	25	127 1	15.55 0.83	5.8	black alum.	XLR	$950.00	Line and gradient offering high output and very low noise floor. Direct-coupled design provides distortion-free output.
ATM25	Dyn.	Hyper card.	30-15k	600	57	150 1	4.65 1.5	13		XLR	$250.00	Large diaphragm high SPL capability, low-end response. Suited for instrument mic'ing.

Model	Type	Patterns	Frequency Response dB, ±dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features

BEYER DYNAMIC

Model	Type	Patterns	Frequency Response dB, ±dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
170H	Dyn.	Hyper card.							black		$1,295.00	Features TGX 480 head, gain before feedback. Noiseless muted switch.
170P	Cond.	Omni/ Hyper card.							black		$1,295.00	Has miniature design, variable mic sensitivity switching.
170G									black		$1,195.00	Circuitry designed for sonic transmission of single and dual coil humbucking style pick-ups without coloration.
MC740	Cond.	Card.	30-20k 3	150	46	140 1	6.5 3	13	black or nickle	XLR	$799.95	Three position low frequency roll-off and switchable, 10dB attenuator.
TG- X180	Dyn	Hyper card.	40-16k 3	290	54	140 1	6 2.5	4.5	black ano- dizod	XLR	$159.00	Uses a special diaphragm for clear, accurate sound at maximum SPL.
TG- X280D	Dyn	Hyper card.	30-16k	290	54	140 1	6 2.5	4.5	black ano- dizod	XLR	$199.00	Suited for vocal, instrument applications. High volume before feedback is required.
TG- X480	Dyn	Hyper card.	40-18k 3	280	50	140 1	8 3	6	black ano- dizod	XLR	$269.00	Has large diaphragm, deep sound with pronounced proximity effect.
TG-X580	Dyn	Hyper card.	30-18k 3	280	50	140 1	8 3	6	black ano- dizod	XLR	$349.00	Large, sensitive hostaphan diaphragm has extended frequency/fast transient response. Internal shock mount reduces handling noise.

BRUEL & KJAER INSTRUMENTS

Model	Type	Patterns	Frequency Response dB, ±dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
4006	Cond.	Omni	20-20k 2	30	33 6	135 1	165 16	150 g	Black	XLR	$1,490.00	High-sensitivity, low-noise mics suited to orchestral work, general purpose recordings, vocals and interviewing.
4007	Cond.	Omni	20-40k 2	30	45 6	148 1	165 16	150 g	Black	XLR	$1,490.00	Handles extremely high pressure levels. Ideal for close placement to high-intensity sound sources.
4011	Cond.	Card.	40-20k 1.5	180	35 6	124 1	175 19	165 g	Black	XLR	$1,590.00	Suited for live and studio recordings. Max SPL 158 dB before clipping.
3530	Cond.	Omni	20-20k 2	30	33 6	135 1	165 16	150 g	Black	XLR	$4,815.00	Includes two 4006s, stereobar, two noise cones for total omnidirectional response.

CARVIN CORPORATION

Model	Type	Patterns	Frequency Response dB, ±dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
CM-68	Dyn.	Card.	45-15k	600	74		6.7 2.1	10	Grey enaml	XLR	$99.00	Vocal hand-held mic
CM-67	Dyn.	Card.	40-15k	250	77		6.5 1.5	10	Grey enaml	XLR	$99.00	Studio or concert instrument mic'ing.
CM-90	Cond.	Card.	30-20k	600	71		7.5 1.03	6.2	Grey enaml	XLR	$139.00	Flat response and wide frequency range.

CROWN INTERNATIONAL —See our ad on Cover III

Model	Type	Patterns	Frequency Response dB, ±dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
PZM-11	Cond.	Hemi- omni	80-20k 3	240	47	120 3	4.5 2.8	2.5	off white	barr strip	$67.00	Appropriate for security and surveillance applications.
PCC- 190	Cond.	Super card.	50-18k 3	150	31	120 3	7.5 3.2	12	black	XLR	$329.00	Features a flush-mounted touch-on/ touch-off membrane switch and high intensity LED indicator.
CM-230	Tri- cond.	Card.	80-15k 3	200	52	150 3	7.5 1.8	7	black	XLR	$795.00	Has three independent condenser mic capsules in standard CM 200 mic housing for audio pool feed, house PA feed, backup capsule.

COUNTRYMAN ASSOICATES INC.

Model	Type	Patterns	Frequency Response dB, ±dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
ISO- MAX 4	Elec.	Hyper card.	70-15k 3	270	50	145	18 0.75	4.7	black	XLR	$359.00	Podium mic with electronic vibration isolation.
ISO- MAX 2	Elec.	Omni/ Hyper/ Bi	20-17k 3	200	-40 -50 -60	130 140 150	0.75 0.25	0.1	black or white	XLR	$231.00	Ideal choir mic.
EMW	Elec.	Omni	20-17k 3	200	-50	145	0.50 0.25	.05	black/ grey/ white	XLR	$210.00	Highly water-resistant lavalier.
ISOMAX Head	Elec.	Hyper card.	20-17k	200	-60	150	0.50 0.25	0.7	black	XLR	$254.00	Broadcast quality headworn mic.

ELECTRO-VOICE

Model	Type	Patterns	Frequency Response dB, ±dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
RE 27N/D	Dyn	Card.	45-20k	150	-51 Bal.	135	217	25 54.	satin nickel	Bal. XLR	$750.00	Produces studio quality sound without phantom power, proximity effect.

Model	Type	Patterns	Frequency Response dB, ±dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
RE 38N/D	Dyn	Card.	25-20k	150 Bal.	-53	135	122	134 109	satin nickel	Bal. XLR	$552.00	The 16 position EQ switch allows the mic to be used in any situation.
N/D857	Dyn	Super card.	25-22k	150 Bal.	-50	135	188	7.9 57	Matte black	Bal. XLR	$450.00	Has off-axis rejection for use with high monitor levels.
N/D 757A	Dyn	Super card.	25-22k	150 Bal.	-51	134	181	7.7 52	matte black	Bal. XLR	$330.00	Reacts to high transients like a studio condenser. Captures full sound for any condenser.
N/D 408A	Dyn	Super card.	30-22k	150 Bal.	-51	134	115 72	6.7	matte black	Bal. XLR	$258.00	Pivoting yoke allows optimum pickup of any source.
N/D 308A	Dyn	Card.	40-20k	150 Bal.	-53	132	115 72	6.7	matte black	Bal. XLR	$222.00	Pivoting yoke and cardioid pattern make this mic ideal for multiple horns, kick drums, background vocals.
CP212	Cond.	Card.	50-20k	150 Bal.	-43	117	294 10.5	6.4	matte black	Bal. XLR	$298.00	Yields flat frequency response for more natural sound in speaking applications. Also available in 18 in. version.
RE 45N/D	Dyn	Cardi	50-15k	600 Bal.	-50	135	292 48	7.5	Matte black	Bal. XLR	$514.00	Hand-held shotgun can be used for ENG and EFP. First hand-held shotgun w/N/Dym Aligned Technology and lobe free off-axis rejection.
FOSTEX CORPORATION OF AMERICA												
M77RP	Rib.	Card.	40-18k	250	56		147 2	2 6.8	black alum.	XLR	$460.00	Excellent mic for kick and snare. Has three position EQ switch.
M88RP	Rib.	Bi	40-18k	600	58	55- 2	5.5 1.75	2	black alum.	XLR	$650.00	Bi-directional mic can be used to produce a subtle ambience. Has useful deep notch. Response to 90 degrees off axis.
M20RP	Rib.	M/S	40-18k	600	54	148 2	6.87 2.75	25	black alum.	XLR	$695.00	Trimmed-down less expensive version of M22RP.
M11RP	Rib.	Uni	40-18k	600	54	148 2	5.5 1.5 2.75	20	black zinc	XLR	$595.00	Smoothest cardioid pattern in the RP series. Offers flat frequency response.
MILAB												
VIP 50	Cond.	Mult. (5) pos.	40-20k 2.5	200	14 mV/Pa	112 1	170 mm	400	black	XLR	$1,495.00	
DC96B	Cond.	Card.	40-20k 2.5	200	6 mV/Pa	118 1	145 mm	200 g	black	XLR	$795.00	
VM44	Cond.	Card.	40-20k 2.5	200	14.5 mV/Pa	128 1	126 mm	130 g	black	XLR	$615.00	
LSR 2000	Cond.	Card.	40-20k 2.5	200	7 mV/Pa	133 1	185 mm	290 g	black	XLR	$675.00	
LC 25	Cond.	Card.	40-20k 2.5	200	5 mV/Pa	128 1	185 mm	290 g	black	XLR	$575.00	
MP 30	Cond.	Hemi	40-20k 2.5	200	9 mV/Pa	110 1	25 mm	66 g	black	XLR	$385.00	
D37	Dyn	Card.	50-20k 2.5	250	1.3 mV/Pa		190 mm	290 g	black	XLR	$330.00	
NEUMANN USA												
KM 100	Cond.		20-20k	50	38.4	148 0.5	3.6 0.8	2.8	Matte black	XLR- 3pin	$795.00- $995.00	Interchangeable pattern, capsule and output stage can be separated to allow interconnection of accessories such as goosenecks.
TLM 170	Cond.		20-20k	100	41.9	150 0.5	6 2.4	22	Satin/ matte black	XLR- 3pin	$2,295.00	Multi-pattern mic employs FET 100 technology and 10dB attenuation switch, integral shock-mount. Includes windscreen.
U 89i	Cond.		20-20k	150	41.9	134 0.5	7.3 1.8	14	Satin/ matte black	XLR- 3pin	$2,195.00	Multi-pattern studio mic, handles high sound pressure levels. Bass roll-off switch.
U 87A	Cond.		20-20k	200	31.1	117 0.5	7.9 2.2	18	satin/ matte black	XLR- 3pin	$2,375.00	Multi-pattern studio mic with 10dB attenuation, low frequency roll-off and pattern switches.
SM 69 fet	Cond.		20-20k	200	34.4	110 0.5	10.2 1.9	16	satin/ matte black	DIN 12	$4,800.00	Dual capsule stereo m/s and x/y condenser mic capsules can be rotated to vary stereo effect.
RSM 191S	Cond.		20-20k	50	32.8	134 0.5	8.3 1.2	6	matte black	DIN 7	$3,150.00	MS stereo shotgun condenser mic includes MTX 191 matrix and accessories. High pass filter built into MTX 191.
KMR 81I	Cond.		20-20k	150	34.9	128 0.5	8.9 0.8	5.1	satin/ matte black	XLR- 3pin	$1,275.00	Short shotgun condenser mic, 12dB side attenuation, base roll-off switch, 10dB attenuation switch, includes windscreen.
KMR 82i	Cond.		20-20k	150	33.6	128 0.5	15.5 0.8	8.8	satin/ matte black	XLR- 3pin	$2,395.00	Shotgun condeser mic, higher directional efficiency than KMR 81i, high/low frequency roll-off switches, includes windscreen.

RAMSA

Model	Type	Patterns	Frequency Response dB, ±dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
WM-S1	Cond.	Card.	50-18k	600	-42 dBu	148	13 33 16		flat black	XLR	$210.00	For cymbals, hi-hats, strings; has phantom-power, transient/frequency response tailored to avoid proximity effect. Has mini boom arm and soft rubber clamp.
WM-S2	Cond.	Card.	120-15k	250	-56 dBu	138	13 33 16		flat black	XLR	$170.00	Same features; ideal for brass, toms, percussion; can be battery/phantom-powered.
WM-S5	Cond.	Card.	70-16k	600	-52 dBu	158	13 33x 16		Flat black	XLR	$280.00	Same features, mic designed for toms, snares and other high SPL sources.
WM-S10	Cond.	Card.	120-15k	250	-56 dBu	138	13 33 90		flat black	XLR	$220.00	Is headset-mounted, vocal mic for keyboardists, drummers. Same features as above.
MD 530	Dyn	Super card.	40-16k	350	57.1		7.1 1.7	9.9	black alum.	XLR	$395.00	Ideal for vocal applications. Adjustable sound inlet basket makes proximity effect variable.
MKE-2	Elect	Omni	40-20k	200 or 1000	45.8	126 1	0.43 0.23	be-low 0.1	black flesh ABS	XLR/ pig tail	$285.00	Popular for broadcasters, church, theater applications when used w/wireless transmitter; used as clip on mic for instruments.
MD 4	Dyn	Card.	30-17k	200	54	175	8 1.5	13	Black ABS	XLR	$569.00	Handles high SPLs. Five position bass roll-off provides equalization up to 1000 Hz.
MD-431	Dyn	Super card.	40-16k	250	57	120 1	7.88 1.25	9	black alum.	XLR	$479.00	High gain before feedback, handles high SPL, triple layer steel mesh grill, shock-mounted capsule, magnetic reed on/off switch.
MKH-40	R.F. cond.	Card.	40-20k	150	32	142 0.5	6 1	3.6	black alum.	XLR	$1,195.00	Low distortion, high output, switchable 10dB pad and bass attenuation, very transparent response.
MKE 4032	Elect	Super card.	70-20k	200	46	140 1	8.1 1.9	7.5	black alum.	XLR	$639.00	Phantom or AA battery operation, built-in blast filter and shock mounting, handles high SPL, has high sensitivity.
MKE 300	Elect	Super card.	150-17k	200	35.9		9.1 2	2.1	black	3.5 mm	$225.00	Video mic ideal for enhancing audio input on camcorders. Has integrated windscreen.

SHURE BROTHERS, INC. —See our ad on Cover IV

Model	Type	Patterns	Frequency Response dB, ±dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
BETA 58	Dyn	Super card.	50-16k	150/71.5 290			6.4 2	9.3	blue die cast	XLR	$266.00	Incorporates neodymium magnet, three stage directional tuning network, advanced shock isolation system and humbucking coil.
BETA 57	Dyn	Super card.	50-16k	150/71 290			6.18 1.5	9.2	blue die cast	XLR	$258.00	Same, but designed for musical instrument mic'ing.
VP88	Cond.	Card.	40-20k	150/66 100		129 1	11.44 1.56	15	satin black alum.	5pin XLR	$995.00	Features built-in left-right stereo matrix, adjustable side level switch, switchable low-cut filter, battery/phantom power.
SM7	Dyn	Card.	40-16k	150/57 150			5.84 7.53 3.78	27	gray alum. steel	XLR	$550.00	Permits choice of four different response curves. Ideal for music recording and studio announcing. Yoke assembly allows exact positioning.

SONY PRO AUDIO

Model	Type	Patterns	Frequency Response dB, ±dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
C-48	Cond.	Multi	30-16k	150	39	128 1	2.2 9.1	20	Satin nickel	XLR	$1,050.00	Selectable patterns, 10dB pad, lo-cut switch 9V battery or phantom power. Vibration-proof structure.
C-535P	Cond.	Card.	30-16k	200	40	138 1	0.8 6.1	4.9	Black alum.	XLR	$495.00	Slim-line design with 10dB pad. Rejects SCR, TV and other electronic noise. Excellent transient response.
ECM-MS5	Elec cond.	Stereo card.	70-20k	150	40	130 1	1.9 8.4	7.6	Alum.	XLR -5	$1,250.	Three capsule design for M-S recording. Built-in M-S matrix field-rugged construction, 12-48V phantom powered.
F-730	Dyn	Card.	50-11k	300	60		1.7 6.5	8.8	Black alum.	XLR	$120.00	For vocal recording, offers extra punch in low range.

TELEX COMMUNICATIONS

Model	Type	Patterns	Frequency Response dB, ±dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
TE 10	Cond.	Card.	30-20k	150-200		75	140	7.4	matte black	XLR	$143.00	Element is suspended by flexible "fingers" which isolate it from shocks.
TD 11	Dyn	Card.	50-16k	100-250		77		9.2	matte black	XLR	$132.00	Has tight cardioid pickup pattern.
ELM 22S	Elec.	Omni	20-20k	5k	-56	142	0.2 0.3	0.5	enaml black	XLR	$175.00	Requires power supply.]
ELM 33S	Elec.	Uni	20-20k	5k	-56	142	0.2 0.75	0.5	enaml black	XLR	$175.00	Same

WIRELESS MICS

AUDIO TECHNICA U.S.

ATW1031 UniPak and ATW1032 comprise an automatic diversity wireless microphone system. Features include a rack-mounted receiver with adjustable output and squelch. Transmitters are crystal-controlled and are available as body pack with instrument, hand-held or mic input. Ten VHF frequencies are available.

Price: $650.00 and up depending on mic element

ELECTRO-VOICE, INC.

The MS-1000, comprised of an N/D757A transducer with a 50mW transmitter, utilizes DNX audio processing which provides signal-to-noise ratio -105 dB minimum and 108 dB A-weighted. It has an audio adjustment range of 30 dB and a RF-on and MIC-on switch to eliminate the potential for speaker popping.

Price:$1,380.00

NADY SYSTEMS, INC.

The 101 VHF Wireless System features companding circuitry for quality sound, with dynamic range of 120 dB. System includes receiver and choice of lavalier or hand-held transmitters.

Prices: Lavalier—$279.95

Hand-held—$299.95

The 201 VHF Wireless System's companding circuitry allows sound quality equal to hardware mics, with no hiss or overload distortion.

Prices: Lavalier—$409.95

Hand-held—$429.95

The 650 VHF Wireless System can allow ten 650 systems to operate on the same stage, all with clean, natural sound and good dynamic range.

Prices:Lavalier: $639.95

Hand-held: $659.95

The 750 VHF Wireless System features two complete True Diversity receivers operating on two different frequencies in one compact rack-mount unit. Up to five 750 systems can be used simultaneously.

Prices: (with two hand-held mics) $1,499.95

The 1200 VHF Wireless System features optional user changeable mic elements and custom frequencies. Up to twenty systems can be operated simultaneously.

Price for hand-held transmitter with Shure SM-58 or NDYM 757 mic element: $1,699.95

New from Nady, the four-channel 301 UHF and the ten-channel 950 UHF frequency synthesis wireless mic systems both feature True Diversity reception, 120 dB dynamic range and more.

Prices: 301 UHF—$600.00, 950 UHF—$1,500.00

SAMSON TECHNOLOGIES CORPORATION

The VLP series is a non-diversity set up in either hand-held, instrument, lavalier and now a headset.

Price: starts at $119.95

The Stage 2, non-diversity and Stage 22, true diversity, feature dbx noise reduction and your choice of popular mic elelments.

Price: starts at $319.95

The new Concert 2 wireless series is a fully rack-mountable true diversity system wiht a balanced output, internal power supply and removeable antennas for remote mounting.

Price: $499.95

The Super TD wireless offers cavity tuned design, galls pc board, dual power supply and detaches on the body pac.

Price: $1,099.95

Broadcast STD is the only system with ten selectable frequencies at both transmitter and receiver to insure trouble-free operation under adverse conditions.

Price: Starts at $1,794.95

The MR-1 is an internally-powered receiver (9 V battery) that's the size of a body pac. This receiver can be attached to the side of a video camera and is compatible with Stage, Concert 2, Super TD and Broadcast transmitters.

Price: starts at $519.95

The UHF series has many of the same features found on our other professional wireless systems with the advantage of operation in the 938 MHz to 952 MHz (just below microwave) to insure an interference-free performance.

Price: starts at $1,995.00

SENNHEISER —See our ad on page 2

The VHF 1H features the SKM4031-90 hand-held mic transmitter and EK2012-90 miniature receiver on VHF carrier frequency for ENG/EFP applications.

Price: $3,245.00

The VHF 1B features the SK2010-90 body pack transmitter with MKE2-2-R RD lavalier mic and ED2012-90 miniature receiver on VHF carrier frequency for ENG/EFP applications.

Price: $3,920.00

The VHF 2H features the SKM4031-90 hand-held mic transmitter and EM2003-90 diversity receiver with ground plane antennae on VHF carrier frequency.

Price: $3,651.00

The VHF 2B features a SK2012-90 body pack transmitter with MKE2-2R RD lavalier mic and EM2003-90 diversity receiver with groud plane antennae on VHF carrier frequency.

Price: $4,326.00

The UHF 2H features the SKM4031-TV hand-held mic transmitter and EM2003-90 diversity receiver on UHF carrier frequency.

Price: $5,835.00

The UHF 2B features theSK2012-TV body pack transmitter with MKE2-2-R RD lavalier mic and EM2003-TV diversity receiver on UHF carrier frequency.

Price: $6,615.00

The UHF 2EH features the SKM4031-TV hand-held mic transmitter and battery-operated EM2003-TV diversity receiver on UHF carrier frequency, supplied in canvas carrying bag.

Price: $6,525.00

The UHF 2EB features the SK2012-TV body pack transmitter with MKE2-2-R RD lavalier mic and EM2003-90 diversity receiver operating on UFH carrier frequency, supplied in canvas carrying bag.

Price: $7,305.00

SHURE BROTHERS, INC. —See our ad on Cover IV

The Shure Wireless System, Model LS12, consists of one each of the L1 Body-Pack Transmitter, L3 Receiver and WA300 Instrument Cable. Receiver features include Power On, RF Signal and, Audio Peak Indicators. Squelch is adjustable by recessed control on rear panel.

Price: $445.00

The Shure Diversity Wireless System, Model LS14/839, features the L1 Body-Pack Transmitter and the 839W omnidirectional lavalier mic. The L4 diversity receiver uses the MARCAD system that monitors both RF signals and combines them for increased gain, improved reception and freedom from dropout.

Price: $495.00

The Shure hand-held Wireless System, Model LS23/58, consists of one L3 receiver and one L2/58 hand-held transmitter. The L2 utilizes Surface Mount Technology and an internal loop antenna. The cartridge used assures the same frequency response and polar pattern of a cabled mic without the cable.

Price: $532.00

The Shure hand-held Diversity Wireless System, Model LS24/96, features the L4 MARCAD Diversity receiver and L2/96 transmitter. The L2 features a black ARMO-DUR plastic case, and the mic cartridges are interchangeable for when a SM58 dynamic head is needed.

The L2, using the SM96 electret condenser cartridge, gives the same frequency response and symmetrical cardioid pattern as the cabled version.

Price: $707.00

The hand-held Diversity Wireless System, model LS24/Beta 58, features the L4 MARCAD Diversity receiver and L2/Beta 58 transmitter. The L2 hand-held uses the Beta 58 cartridge, assuring the same response and supercardioid pattern as the cabled Beta 58 mic.

Price: $748.50

SONY PRO AUDIO

Sony has a large variety of both non- and full-diversity tuners, synthesized transmitters and mic heads for every type of professional application.

Prices: available upon request

The 800 Series is designed to be operated in the 794MHz to 806MHz UHF band to keep external noise and interference to a minimum. The series' Phase Locked Loop synthesized system controls the transmitting and receiving frequency. Multiple-channel operation at the same location is also available.

Price: not available

TELEX COMMUNICATIONS, INC.

The HT-400 is a two channel wireless mic with integral transmitter and antenna. Includes on/off swithces for audio and mic and a, low-battery warning LED. The HT-400 series includes models with a Telex TE-10 condenser, a Shure SM-87 condenser and a Shure SM-58 dynamic head configuration. All of the above heads are interchangeable.

Prices: from $1,020.00 to $1,250.00 depending on head

The HT-100 is a single channel wireless mic with integral transmitter and antenna. Includes on/off switches for audio and mic. The HT-100 series includes models with Telex TE-10 condenser, Telex TD-11 dynamic, Shure SM-58 dynamic and Shure SM-87 condenser heads.

Prices: from $370.00 to $620.00

AUDIO TAPE

BASF CORPORATION

PEM 469 studio mastering tape features high output and low noise for a wide dynamic range and standard bias for compatibility with many tapes and machines. Low print-through minimizes the effects of pre- and post echo and excellent slitting provides consistent phase stability and superior winding characteristics.

PEM 468 features extremely low print-through, high output and low noise for a wide dynamic range. Excellent slitting for consistent pahse stability, and batch number and web position printed on the back coating for permanent tape-type identification.

PEM 369 features 1 mil. thickness, 3,600 feet for extended record time. Extremely low print and excellent slitting for superior winding and phase stability.

PEM 526 and Chrome Loop Master 921 are designed for high-speed cassette duplication, and provide excellent durability under the stress of bin-loop applications. Also features high-frequency signal retention after numerous passes and extremely low print-through.

PE 649/949/1249 have premium iron oxide, high output, low noise, standard IEC Bias I cassette tape. Features extended headroom in low and high frequencies for the most critical and demanding music duplication.

PE 619I/919I has low noise, high output, extended high end response iron oxide cassette tape for bias compatibility with Standard I designation.

PE 668/968 is a ferro-cobalt, high coercivity, IEC II bias tape. The tape has high output and very low noise. Ideal for demanding music duplication.

TDK ELECTRONICS: PROFESSIONAL PRODUCTS DIVISION

The Acoustic Master (AM) is a normal-bias ferric oxide formulation with reduced noise levels and wider dynamic range. Has jam-proof cassette mechanism with five screw construction and transparent shell. Available in 30-, 46-, 60-, 90- and 120-minute lengths.

The Sound Master (SM) is a high-bias cassette for studio and demo recordings. Uses SA tape pancake for low noise and wide dynamic range. A dual-layer anti-resonance cassette mechanism ensures phase accuracy and low modulation noise. Available in 10-, 20-, 30- and 60- minute lengths.

Instant Starting Acoustic Master (AL) is a normal-bias tape in an instant starting cassette. New pure-grained ferric oxide formulation for reduced noise levels and wider dynamic range. Housed in a jam-proof cassette mechanism. Available in 30-, 60- and 90-minute lengths.

The Duplication Master (ZM) has a normal-bias tape formulation, pure grained ferric oxide particles achieve reduced noise levels with wide dynamic range. Housed in jam-proof cassette mechanism. Available in 30-, 46-, 60-, 90- and 120-minute lengths.

The Medical Master has a normal-bias formulation for slow speed medical recording applications. Pure grained ferric oxide particle plus high-dispersion binder and advanced coating technology yield extremely low drop-out rate. Available in 60-, 90- and 120-minute lengths.

The Digital Audio Tape (DA-R) is a super Finavinx pure metal particle digital audio tape housed in a high-precision sealed cartridge. Available in 60-, 90- and 120-minute lengths.

3M

The 250 Audio Mastering Tape incorporates a 1.5 mil thick back-coated polyester backing. Delivers high output/low noise performance with the widest possible dynamic range of analog mastering tapes. Ideal for high quality music mastering. Available in ¼ in., ½ in., 1 in., and 2 in. widths.

The 806 Audio Mastering Tape has a 1.5 mil thick backcoated polyester backing. Developed to give good compromise in print-through and maximum output level characteristics. Available in ¼ in., ½ in., 1 in., and 2 in. widths.

The 808 Audio Mastering Tape has a back-coated polyester base 1.5 mil thick. Has extremely low print-through characteristics. Ideal for speech, sound effects, and other applications where low print through is required. Available in ¼ in. width only.

AUD Digital Audio Cassettes are engineered to deliver state-of-the-art performance in the production of CDs, record albums and cassettes. Available in 30, 60, and 75 minute lengths, these units incorporate anti-static system.

AVX Audio Cassettes are normal bias (IEC Type I), optimized for heavy duty usage and high speed duplication. Available individually boxed or bulk packaged in 20, 30, 46, 60, 90 and 120 minute lengths.

IRC Audio Cassettes are "instant record", normal bias (IEC Type I). A magnetically-coated leader is employed so that recordings can be started at the very beginning of tape. Available in both boxed and bulk packages, in lengths of 30, 60, and 90 minutes.

SX Audio Cassettes are chrome bias (IEC Type II) for high quality recording applications. Available in album boxes, 30, 46, and 90 minute lengths.

3M 996 audio mastering tape is the first analog mastering tape capable of recording at operating level +9 with virtuallly no distortion. Provides widest dynamic range of any analog mastering tape and offers improved print with almost no increase in tape noise. Available in ¼ in., ½ in., 1 in., and 2 in.

3M 226 audio mastering tape is designed for music mastering on a budget. Delivers high output/low noise performance on 1.5 mil polyester backing. Available in ¼ in., ½ in., 1 in., and 2 in.

3M DAT is designed for the demanding professional user who needs low error rates and a dependable product. Labels allow for plenty of space for documenting the project. Available in 46, 60, 90 and 120 minute lenghts.

3M 186 audio tape meets the needs for program distribution, storage and everyday use. Designed as a low noise tape for cost-effective use in standard bias settings. Available in ¼ in. width.

TAPE RECORDERS

Cassette Recorders

FOSTEX CORPORATION OF AMERICA

The X-26 is a 6-channel/4-track mixer/recorder. It is a 2-head machine running at 1-⅞ in./sec. with a frequency response of 40 Hz to 12.5 kHz, a flutter rate of 0.15 percent, and a S/N ratio of 58 dB (A-weighted).

Price: $449.00

TASCAM, TEAC CORPORATION OF AMERICA —See our ad on pages 20-21

The 102 stereo cassette recorder has two heads, two motors with Dolby HX Pro; wow & flutter: ±0.45 percent; frequency response of 25 Hz to 18 kHz; Unwtd. S/N ratio of less than 80 dB with Dolby C.

Price: $299.00

The 103 stereo cassette recorder has three heads, two motors with Dolby HX Pro; wow & flutter: ±0.045 percent; frequency response of 20 Hz to 20 kHz; Unwtd. S/N ratio of less than 80 dB with Dolby C.

Price: $499.00

The 202WR dual stereo cassette recorder each has two heads, two motors with Dolby HX Pro; wow & flutter: ±0.06 percent; frequency response of 30 Hz to 18 kHz; Unwtd. S/N ratio of less than 79 dB with Dolby C.

Price: $499.00

The 112/112B stereo cassette recorder has three heads, two motors with Dolby HX Pro. B versio offers XLR in/out wow & flutter: ±0.04 percent; frequency response of 25 Hz to 18 kHz; THD @ 400Hz of 1 percent; Unwtd. S/N ratio of less than 78 dB with Dolby C.

Price: $679.00

The 112R auto reverse stereo cassette recorder has four heads, two motors; wow & flutter: ±0.03 percent; frequency response of 25 Hz to 19 kHz; THD @ 400 Hz of 1 percent; Unwtd. S/N ratio of less than 80m dB with Dolby C.

Price: $329.00

The 122 MKII stereo cassette recorder with ¼ in. mic inputs has three heads; three motors; wow & flutter: ±0.04 percent; frequency response of 25 Hz to 19 kHz; THD @ 400 Hz of 1 percent; Unwtd. S/N ratio of less than 78 dB with Dolby C.

Price: $1,099.00

The 238 8-track cassette recorder has a speed of 3 3/4 in./sec.. It has wow & flutter, ±0.04 percent; frequency response of 30 Hz to 16 kHz; THD @ 400 Hz of 0.8 percent IHF. S/N ratio of less than 93 dB with dbx.

Price: $1,799.00

UHER OF AMERICA

The CR1600 has 4 track stereo, electronic drive control for auto-reverse operation in record or playback mode, 2 speeds: 15⁄16, 1-7⁄8 in./sec., 3 heads, 2 channel, 2 VU meter, Dolby B, switchable ALC, solenoid controlled, fully remote controlled, built-in voice activation system, memory pulse facility, record time—6 hours, frequency response—20-19 kHz., wow & flutter less than 0.2 percent, S/N 64dB.

Price: $2,080.00

The CR1601 portable cassette has 4 track monaural, 3 speeds: 15⁄32, 15⁄16, 1-7⁄8 in./sec., 3 heads, 2 channel, 1 VU meter, switchable ALC, solenoid controlled, fully remote controlled, built-in voice activation system, memory pulse facility, record time—8 hours, frequency response 20-19 kHz, wow & flutter less than 0.2 percent, S/N 50 dB.

Price: $2,080.00

DAT Recorders

FOSTEX CORPORATION OF AMERICA

D20 is a 2-channel/2-track DAT recorder. It is a 4-head machine running at 8.15 mm/s with a frequency response of 20 Hz to 20 kHz and a S/N ratio of 90 dB. THD is 0.05 percent at 1 kHz.

Price: $8,000.00

SONY PRO AUDIO

The PCM-2500 is a 2-channel, 2-head DAT recorder whose running speed is 8.15 mm/s. It has a S/N ratio of 90 dB and 0.05 percent THD. Frequency response is 2 Hz to 22 kHz. It is driven by a servo type motor, has LED record indicators and measures 17x4x16.6 in., weight: 24 lbs., 11 oz.

Price: $3,550.00

PCM-2000 is a 2-channel, 2-head DAT recorder whose running speed is 8.15 mm/s. It has a S/N ratio of 90 dB and 0.07 percent THD. Frequency response is 2 Hz to 22 kHz. It is driven by a servo type motor, has LCD record indicators and measures 8.4x3x7.6 in., weight: 8 lbs., 13 oz.

Price: $5,000.00

TCD-D10 is a 2 channel, 2-head DAT recorder whose running speed is 8.15 mm/s. It has a S/N ratio of 85 dB and 0.08 percent THD. Frequency response is 20 Hz to 20 kHz. It is driven by a servo type motor, has LCD record indicators and measures 10x2.3x7.6 in., weight: 4 lbs., 7 oz.

Price: $3,550.00

STUDER REVOX AMERICA, INC.

The A721 cassette/R-DAT recorder features two channel, four motors, three heads, speed: 1 7⁄8 in. per second, DC direct drive spooling motors, distortion less than 1 percent, frequency response: 20 Hz, 20 kHz +2/-3 dB, line input: balanced and floating, minimum 10 kOhm, LCD, THD percentage: less than 1 percent at O VU, less than 72 dB, 19 in. rack mount.

Price: $2,795.00

The D740 CD Recorder features converter technology: bitstream in differential mode, sampling frequency: 44.1 kHz, digital inputs: optical, cinch and XLR, analog inputs: cinch and XLR, frequency response: (record and playback)) 20 Hz, 20 kHz ±0.2 dB, total harmonic distortion plus noise: (record and playback): less than 0.008 percent (20 Hz to 20 kHz), less than 0.005 percent (1 kHz) (playback only): less than 0.006 percent (20 Hz to 20 kHz).

Price: to be announced

TASCAM, TEAC CORPORATION OF AMERICA —See our ad on pages 20-21

The DA-30 DAT Recorder with AES/EBU and consumer digital I/O performs to the following specifications: wow & flutter: ±0.001 percent; frequency response: 1 Hz to 22 kHz; Unwtd. S/N ratio of less than 94 dB.

Price: $1,499.00

YAMAHA CORPORATION OF AMERICA

The DTR2 includes four sets of Input/Output connections; digital I/O (coaxial and optical), RCA-type phono unbalanced analog connections and balanced XLR (+4 dB) connectors. A front-panel switch selects analog or digital inputs. Another front panel switch selects either 44.1 kHz or 48 kHz sampling frequency.

Price: $1,495.00

The YPDR601 CD Recorder allows a full TOC to be written to disc either before or after recording; allows recording of audio data to disc to be interrrupted using standard pauses functions; direct input/output connections can be either analog or digital. Analog connections are made via balanced XLRs, while digital connections can be made via AES/EBU, or SDIF-2 data formats.

Price: not available

Open Reel Recorders

FOSTEX CORPORATION OF AMERICA

The R8 is an 8-channel/8-track reel-to-reel recorder. It is a 2-head machine running ¼ in. tape at 15 in./sec. with a frequency response of 40 Hz to 18 kHz, a flutter rate of 0.06 percent and a S/N ratio of 78 dB. THD is 1 percent at 1 kHz.

Price: $2,800.00

The E2 is a 2-channel/2-track recorder. It is a 3-head machine running ¼ in. tape at 15 in./sec. with a frequency respone of 30 Hz to 20 kHz, a flutter rate of 0.05 percent, and a S/N ratio of 74 dB. THD is 1 percent at 1 kHz.

Price: $3,795.00

E8 is an 8 channel/8 track reel-to-reel recorder. It is a 2-head machine runniing ¼ in. tape at 15 in./sec. with a frequency response of 40 Hz to 18 kHz, a flutter ratio of 0.05 percent, and a S/N ratio of 80 dB. THD is 1 percent at 1 kHz.

Price: $4,495.00

G-16S is a 16 channel/16 track reel-to-reel recorder. It is a 2-head machine running ½ in. tap at 15 in./sec. with a frequency respone of 40 Hz to 18 kHz, a flutter rate of 0.05 percent WTD and a S/N ratio of 80 dB with built-in Dolby S NR. THD is 1 percent at 1 kHz.

Price: $8,500.00

G-24S is a 24-channel/24-track reel-to-reel recorder. It is a 2-head machine running 1 in. tape at 15 in./sec. with a frequency respone of 40-18 kHz, a flutter ratio of 0.05 percent, WTD, and a S/N ratio of 80 dB with built-in Dolby S NR. THD is 1 percent at 1 kHz.

Price: $14,500.00

MITSUBISHI DIGITAL PRO AUDIO

The X-86HS has four heads, is DC servo controlled with a pinch roller drive with constant tension servo system; line input impedance of 10k ohms; line output impedance suitable for 200-600 ohm load; frequency response of 20 Hz to 40 kHz; dynamic range over 90 dB; and wow and flutter. Additional options available.

Price: available upon request

The X-86 and X-86C are two-channel ready; have four heads; are DC servo controlled; has pinch roller drive with constant tension servo system; line input impedance 10k ohms; line output impedance suitable for 200 ohm load; frequency response of 20 Hz to 20 kHz; dynamic range over 90 dB; and wow and flutter. Additional options available.

Price: available upon request

The X-880 has 32 channels; is DC servo controlled; has a closed loop servo, pinch rollerless capstan drive system; line input impedance of 10k ohms; line output impedance suitable for 200 ohm load; frequency response of 20 Hz to 20 kHz; dynamic range of over 90 dB; and wow and flutter. Additional options available.

Price: available upon request

NEVE

The X-86E razor-edit digital recorder has one channel time code; line input level of +4 dBm nominal, +19 dBm max.; line input impedance of 10k ohms; line ouput level of +4dBm nominal, +24 dBm max.; line output impedance suitable for 200 ohm loads; frequency response of 20 Hz to 20 kHz; dynamic range over 90 dB; less than 0.05 percent distortion; wow and flutter.

Price: available upon request

OTARI CORPORATION

MTR-15 is a ¼ in. or ½ in. 2-track mastering tape recorder with 12.5 in. reel size, quartz PLL controlled brushless, direct-drive capstan motor, DC closed-loop bi-directional servo reel motor, wow and flutter max. 0.04 percent at 30 in./sec., frequency response @ 30 in./sec. 40 Hz-28 kHz +2 dB, signal to noise ration of 72 dB.

Price: $12,670.00

DTR-900II is a 1 in. 32-channel digital tape recorder with 14 in. reel size, 4 heads, ±0.041 percent wow and flutter, frequency response 20Hz-18kHz ±0.5dB, 9600 Hz PLL capstan motor, two servo ½ HP DC reel motors.

Price: $150,000.00

MTR-100A is a 2 in. 24-track with automatic alignment, 14 in. max. reel size, quartz PLL DC brush-type bi-directional reel motor, wow and flutter @ 30 in./sec. less than 0.04 percent. Frequency response @ 30 in./sec. is 50 Hz-25 Hz, S/N ratio at 1040 nWb/m is 70dB, input impedance is more than 10 k ohms (20 Hz-20 kHz).

Price: $59,950.00

MX-55N is a 2-channel ¼ in. compact recorder with one DC servo-controlled capstan, 2 induction reel motors, 4 heads, 10.5 in. max. reel size, wow and flutter less than 0.06 percent @ 15 in./sec., frequency response 30-22 kHz ±2 dB @ 15in./sec.. Mic input impedance is 10 k ohms, unweighted S/N 69 dB @ 1040 nWb/m.

Price: $4,295.00

MX-70 features DC servo reel motors and brushless DC capstan motor, crystal referenced, three heads, max. reel size 10.5 in., wow and flutter less than 0.04 percent, frequency response +2/-3 dB, 50-22 kHz, unweighted S/N ratio 70 dB @ three precent third harmonic distortion.

Price: (8-track) $17,200.00 or (16-track) $421,650.00

MX-80 is a 2 in. 24-track model, 10.5 in. tape width, 9600 Hz PLL capstan motor, microprocessor-controlled, 2 servo 1/3 HP DC reel motors, 3 heads, S/N ratio greater than 67 dB, frequency response 60 Hz-22 kHz, ±2 dB.

Price: $31,950.00

MK-IV-8 is a ½ in. 8-channel recorder with 10.5 in. reel size, DC servo-controlled capstan motor, two induction reel motors; 3 heads; wow and flutter 0.06 percent @ 15 in./sec.; frequency response 40 Hz-20 kHz +2/-3 dB.

Price: $46,295.00

MTR-90III is a 14 in. reel size; 9600 Hz PLL capstan motor, 2 servo ½ HP DC reel motors, wow and flutter @ 30 in./sec. is less than 0.04 percent, frequency response @ 250 nWb/m 50 Hz-22 kHz (30 in./sec.), S/N ratio @ 1240 nWb/m is 72 dB (30 in./sec. AES), 3 heads.

Price: (for 16/24 model) $39,350.00
(for 24-track model) $42,950.00

SONY PRO AUDIO

APR-24 is a 24-channel analog recorder utilizing 2 in. tape. It features amorphous steel heads and "DC constant tension design." Frequency response at 30 in./sec. is 48 Hz—25 kHz (+0.75/-3.0 dB). S/N is 70 dB at 30 in./sec. and 66 dB at 15 in./sec. includes remote control with stand.

Price: $45,500.00

APR-5000 is a 2-channel analog recorder which can be purchased in various configurations (including an IEC center-track time-code version). The 5002W version features a 50 Hz-28 kHz (+0.75/-3.0 dB) frequency resonse and a S/N ratio of 65 dB. Other versions feature 9-pin serial interface.
Prices: from $8,875.00 to $11,950.00

STUDER REVOX AMERICA, INC.

The PR99 MK 3 features ¼ in. two channel, three motors, two channels, AC servo capstan, three heads, 10.5 in. max. reel size, W & F less than 0.1 percent DIN 45507, distortion less than 1 percent, S/N (A weighted) 66 dB, response 30 Hz, 22 kHz +2/-3 dB.
Price: $3,495.00

The C270 series features ¼ in. two channel or ¼ in. four channel, ½ in. eight channel three motor (servo controlled), DC capstan, three heads, 10.5 in. max reel size, W & F less than 0.05 percent, distortion less than 1 percent, S/N 64 dB, response 30 Hz, 22 kHz ±2 dB, optional SMPTE time-code center track.
Price: $3,995.00

The A807 features ¼ in. two channel, ½ in. four channel, three motor, servo controlled DC capstan, three heads, 10.5 in. max. reel size, W & F less than 0.05 percent, distortion less than 1 percent, S/N 66 dB, response 30 Hz, 20 kHz ±2 dB, optional: balanced mic. SMPTE time-code center track.
Price: $6,985.00

The A812 features ¼ in. two channel, three motor, servo DC capstan, three heads, 12.5 in. max. reel size, W & F less than 0.04 percent, distortion less than 1 percent, S/N 70 dB, response 30 Hz, 20 kHz ±2 dB, optional SMPTE time-code center track.
Price: $14,700.00

The A820 features two track recorders, ¼ in., ½ in., three motor, servo DC capstan, three heads, 14 in. max. reel size, W & F less than 0.03 percent, distortion less than 1 percent, S/N 70 dB, response 30 Hz, 20 kHz ±2 dB, optional SMPTE time-code center track.
Price: $18,900.00

The A827-MCH features 2 in. 24 channel, three motor, servo DC capstan, three heads, 14 in. max. reel size, W & F less than 0.03 percent, distortion less than 1 percent, S/N 70 dB, response 30 Hz, 20 kHz ±2 dB.
Price: (includes remote locator) $44,900.00

The A820-MCH features 2 in. 24 channel, three motor, servo DC capstan, three heads, 14 in. max. reel size, W & F less than 0.03 percent, distortion less than 1 percent, S/N 70 dB, response 30 Hz, 20 kHz ±2 dB, automatic alignment, internal noise reduction.
Price: (includes remote and locator) $76,900.00

TASCAM, TEAC CORPORATION OF AMERICA —See our ad on pages 20-21

The ATR-80/24 and 32 2 in. track recorders have three heads; three motors; wow & flutter: ±0.05 percent @ 30 in./sec.; frequency response of 45 Hz to 25 kHz; THD @ 1 kHz of 0.5 percent; Unwtd. S/N ratio of less than 67 dB.
Price: $34,999.00

The ATR-60/16 1 in. 16 track recorder has three heads; three motors; wow & flutter: ±0.08 percent @ 30 in./sec.; frequency response of 40 Hz to 22 kHz; THD @ 1 kHz of 0.8 percent. A wtd. S/N ratio of less than 71 dB.
Price: $15,999.00

The MSR-24 1 in. 24 track recorder has two heads; three motors; wow & flutter: ±0.06 percent @ 15 in./sec.; frequency response of 40 Hz to 20 kHz; THD @ 1 kHz of 0.8 percent; A wtd. S/N ratio of less than 65 dB.
Price: $12,499.00

The MSR-16 ½ in. 16 track recorder has two heads; three motors; wow & flutter: ±0.06 percent @ 15 in./sec.; frequency response of 40 Hz to 20 kHz; THD @ 1 kHz of 0.8 percent; A wtd. S/N ratio of less than 65 dB.
Price: $7,499.00

The TSR-8 ½ in. 8 track recorder has two heads; three motors; wow & flutter: ±0.08 percent @ 15 in./sec.; frequency response of 40 Hz to 20 kHz; THD @ 1 kHz of 0.8 percent; A wtd. S/N ratio of less than 68dB.
Price: $3,499.00

The BR-20 and BR-20T ¼ in. two track recorder has three heads; three motors; wow & flutter: ±0.06 percent @ 15 in./sec.; frequency response of 35 Hz to 22 kHz; THD @ 1 kHz of 0.8 percent. A wtd. S/N ratio of less than 72 dB.
Prices: $2,999.00 and 3,999.00 respectively.

The DA-800 ½ in. 24 track DASH recorder has three heads; three motors; wow & flutter: unmeasurable; frequency response of 20 Hz to 20 kHz; THD @ 1 kHz of 0.05 percent.
Price: $99,000.00

The ATR-60/2T ¼ in. 2 track recorder with center track TC has three heads; three motors; wow & flutter: ±0.05 percent @ 15 in./sec.; frequency response of 40 Hz to 22 kHz; THD @ 1 kHz of 0.6 percent; A wtd. S/N ratio of less than 72 dB.
Price: $6,999.00

UHER OF AMERICA

The 4000 report monitor AV has 2 track monaural, 4 speeds: 15⁄16, 1-⅞, 3-¾, 7-½ in./sec., 3 heads, 5 in. reel, 1 channel, belt drive, 1 VU meter, frequency response 20-25 kHz., wow & flutter less than 0.2 percent, S/N 64 dB, mic inputs.
Price: $1,927.00

The 4200 report monitor has 2 track stereo, 4 speeds: 15⁄16, 1-⅞, 3-¾, 7-½ in./sec., 3 heads, 5 in. reel, 1 channel, belt drive, 2 VU meters, frequency response 20-25 kHz, 200 Ohms, LED functions indicators, switchable ALC.
Price: $2,066.00

The 4400 report monitor has 4 track stereo, 4 speeds: 15⁄16, 1-⅞, 3-¾, 7-½ in./sec., 3 heads, 5 in. reel, 2 channels, belt drive, 2 VU meters, frequency response 20-25 kHz, wow & flutter les~s than 0.2 percent, S/N 62 dB, mic inputs.
Price: $2,066.00

The 6000 report universal has 2 track monaural, 4 speeds: 3-¾, 1-⅞, 15⁄16, 15⁄32 in./sec., 3 heads, 5 in. reel, 1 channel, solenoid controlled, belt drive, 1 VU meter, built-in voice activation system, memory pulse facility, fully remote controlled, frequency response 20-22,000 Hz, wow & flutter less than 0.2 percent, S/N 62 dB.
Price: $2,513.00

TAPE AND RECORDER ACCESSORIES

JRF MAGNETIC SCIENCES **—See our ad on page 2**
T-Bar head adjustable mount has conventional bottom mount head assemblies converted to a top mount T-Bar which offers azimuth and wrap adjustments. The T-Bar mount will reduce mechanical head alignment time.

Price: (includes installation and head reconditioning) $250.00 to $450.00

PLX magnetic heads are available for tape machines such as Teac ½ in. 80-8, 1 in. 85-16; Otari ½ in. (4 and 8 track) MX5050, ½ in. (4 track) MTR10/12; Ampex 350, 440 ¼ in. (mono and 2 track) and MM1100/1200 (24 track); Studer A80 and A800 (24 track); Mincom M56 and M79 (16 and 24 track); MCI/Sony ½ in. (2 track) 2 in. (24 track).

Price: available upon request

Time-code retro-fit kits are available for many ¼ in. tape machines. Features include center track time code record and playback capability; adjustable to zero sub frame accuracy; FM playback; optional FM and mono pilot playback (MK 1 Kit) available; fully functional at 7 ½, 15, 30 in./sec.

Price: ranges from $1,795.00 to $3,200.00

POLYLINE CORPORATION
Leader tapes, splicing tapes, special tapes (hold-down, cleaning), splicing blocks, Mylar splicing tabs, metal foil tabs, empty boxes, cassette loading supplies, labels, index cards, vinyl albums for audio cassettes, corrugated shippers for albums.

Prices: Bulk prices available on all products

SAKI MAGNETICS, INC.
We manufacture long life ferrite heads for all high speed reel-to-reel duplicators, in-cassette duplicators and long life ferrite heads for most pro-audio recorders from Ampex, MCI, Sony, Otari and Studer. We also manufacture 24 track metal heads for machines by Ampex, MCI, Mincom, Otari and Studer.

XEDIT CORPORATION
The EDITALL Professional Splicing Products are available in over 30 standard configurations covering every tape size and format. The S3-D ¼ in. and S3.5D ½ in. are deluxe, highly finished three cut blocks.

Prices: S3-D—$60.00

S3.5D—$75.00

The S-2 series are compact blocks that will easily fit onto many machines. They are available in ¼ in. and ½ in. models.

Prices: S-2—$42.00

S2.5-OT—$55.00

The S2-OT, S2.5-OT and the S3-OT are Editall replacements for Otari supplied blocks.

Prices: $42.00, $55.00, $60.00, respectively.

The MD-25 is an exact retrofit for the block used on the Mitsubishi ¼ in. Digital equipment. It is however a "hybrid" curved trough block with a precisely fitted flat splice point.

Price: $160.00

The SA-2 is a two-inch splicing block, offering the flexibility of three cutting angles and precise tape control.

Price: $180.00

EDITABS are precision die-cut editing tabs designed to eliminate any contamination of the adhesive by finger contact and are self-aligning. They are available in all tape sizes from audio cassette through one inch including 8mm video/DAT. The CL6-250 are ½ in. Editabs for audio and video splicing and repair.

Prices: CL6-250—$26.50

$73.50 for 1000/box

ADDRESSES

AMS Industries, Inc.
7 Parklawn Drive
Bethel, CT 06801

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

BASF Corporation
19 Crosby Drive
Bedford, MA 01730

Beyer Dynamic
5-05 Burns Avenue
Hicksville, NY 11801

Bruel & Kjaer Instruments
185 Forest Street
Marlborough, MA 01752

Carvin Corporation
1155 Industrial Avenue
Escondido, CA 92025

Crown International
1718 West Mishawaka Road
Elkhart, IN 46517

Countryman Associates
417 Stamford
Redwood City, CA 94063

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

Fostex Corporation of America
15431 Blackburn Avenue
Norwalk, CA 90650

JRF Magnetic Sciences
249 Kennedy Road
P.C. Box 121
Greendell, NJ 07839

Milab
200 Sea Lane
Farmingdale, NY 11735

Mitsubishi Pro Audio/Neve
Berkshire Industrial Park
Bethel, CT 06801

Nady Systems, Inc.
6701 Bay Street
Emeryville, CA 94608

Neumann (USA)
6 Vista Drive
Old Lyme, CT 06371

Otari Corporation
378 Vintage Park Drive
Foster City, CA 94404

Polyline Corporation
1233 Rand Road
Des Plaines, IL 60016

Ramsa
6550 Katella Avenue
Cypress, CA 90630

Saki Magnetics, Inc.
26600 Agoura Road
Calabasas, CA 91302

Samson Technologies Corporation
P.O. Box 9068
Hicksville, NY 11802-3815

Sennheiser Electronic Corporation
6 Vista Drive
P.O. Box 987
Old Lyme, CT 06371

Shure Brothers, Inc.
222 Hartrey Avenue
Evanston, IL 60202

Sony Pro Audio
3 Paragon Drive
Montvale, NJ 07645

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

Tascam, Teac Corporation of America
7733 Telegraph Road
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3M Center
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UHER of America
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Xedit Corporation
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Queens Village, NY 11429

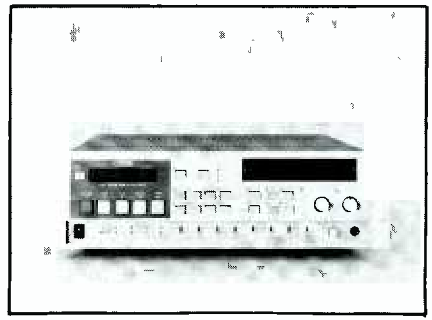
Yamaha Corporation of America
Post Office Box 6600
Buena Park, CA 90622

NEW PRODUCTS

ANALOG AND DIGITAL IMPROVEMENTS

● The G16S, a ½-in. sixteen track recorder and the G24S, a 1-in. twenty-four track recorder, offer, in addition to Dolby S noise reduction, auto locate, auto play, auto return, zone limiting, ten memory locations for cue point and a detachable control panel that becomes a full-feature remote control. As a plug-in option, Model 8330 contains a SMPTE generator/reader with full-chase synchronizer and sophisticated MIDI control.

• Now included with all D20 recorders is the 8310 system expansion board. With this addition, the D20 delivers VTR emulation with a wide variety of editors, and offers not only different machine emulation modes, but also multiple editor specific modes. All VTR emulations and editor selections are done from the D20 front panel. You can edit over and over again at the same time-code address on the tape and not destroy the subcode data. Also new is the ability to jam sync SMPTE time code against the A (Absolute) time recorded on most DATs.



Owners of older D20s without the 8310 or the latest software updates may have their D20s updated at no charge for parts or labor. Please contact Duke Gee at Fostex for details.

Manufacturer: Fostex Corporation of America
Prices: G16S is $8,500.00
G24S is $14,500.00
Model 8330 is $1,000.00

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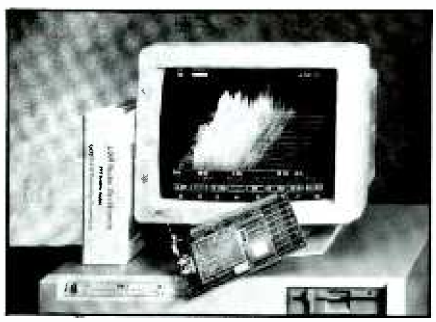
COMPUTER AUDIO ANALYSER

● The FFT software module allows viewing events in the audio spectrum in realtime. The VGA compatible graphics display plots frequency, amplitude and time in two and three dimensions. A dynamic range of up to 180 dB reveals the contour of any noise floor. You can capture FFT data to disk for later replay and analysis. A peak hold memory references stored data to elapsed time or SMPTE time code.



Differential memory functions allow source independent measurement. The DSP Workbench achieves its best performance on an IBM PC/AT running at 20 MHz or faster with color monitor and VGA graphics card. The basic package includes an ISA Bus plug-in card with a 50 MHZ AT&T DSP32C processor, software and comprehensive manual. Versions available for analog or digital input. SMPTE time code reader board is optional.

Manufacturer: Gotham Technology Corporation
Price: $3,500.00

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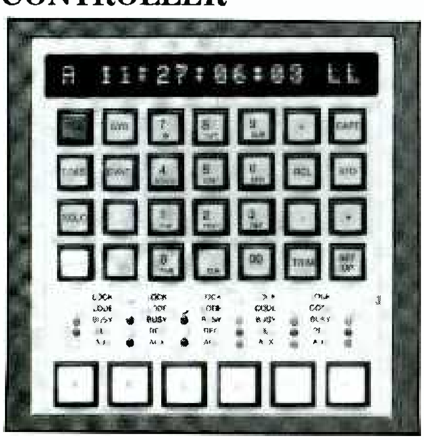
TIME-CODE CONTROLLER



● The Console Control Unit, a miniature keypad, mounts directly into standard consoles. The CCU operates the System Supervisor multiple machine controller which interfaces to standard console automation software with no changes or updates required. Utilizing Lynx Time Code Modules, the CCU controls up to six analog or digital audio tape recorders, VTRs or sprocketed film transports. Any transport may be designated as the master without switching cables. The CCU system offers variable speed control of the master for pitch changes of an entire synchronized machine group. A jog/shuttle wheel is also available. Other features include a sixteen character alphanumeric display, readout of machine time code location and offsets, status lights for each machine indicating lock, code, record, busy and aux, group and solo functions.

Manufacturer: TimeLine
Price: $1,500.00

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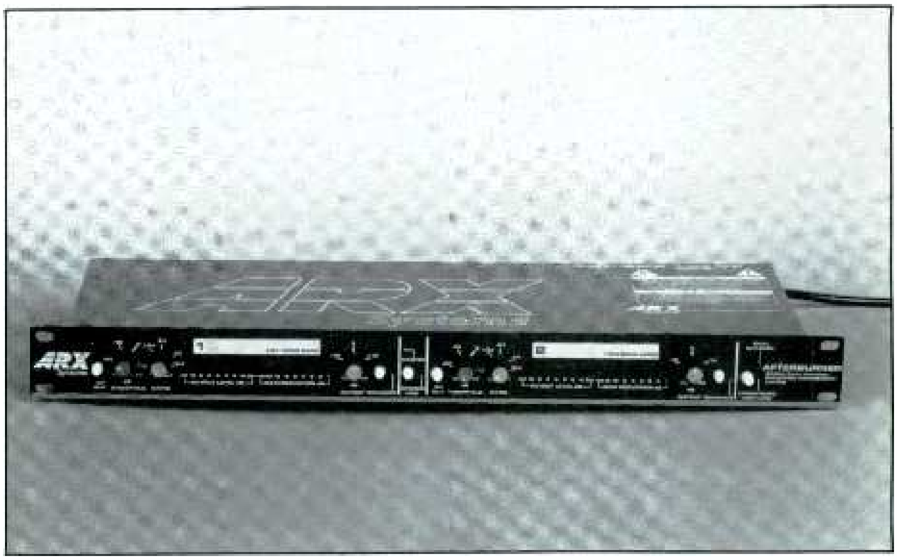
COMPRESSOR/LIMITER

● "Afterburner" dual channel enhanced compressor limiter feature balanced inputs/outputs on jack and XLRs and detector loop insert. Front panel controls include variable threshold, ratio, output gain, hardwire bypass, enhance in/out, stereo link switch and dual channel/mono two band mode. The "Afterburner" can be configured as a mono dual band low/high compressor limiter which allows separate dynamics control of low and high frequencies. The "Enhance" feature allows for frequency correction to preserve the spectral balance of the audio system.
Manufacturer: ARX Systems
Price: $495.00 (retail)



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WIRELESSS MIC ACCESSORY

● The L11 is one-third smaller than its predecessor, the L1, and it features several improvements, including: a surface-mount circuitry design for mechanical stability and forty to fifty percent longer battery life, low noise preamplifier stage for quiet operation, durable ¼-wave trailing antenna for maximum signal radiation, noiseless mic mute switch, adjustable 40 dB input sensitivity range control and special shielding for increased protection from audio/RF interaction. The L11's cleaner output signal also allows a larger number of L Series

Wireless systems to operate simultaneously.
Manufacturer: Shure Brothers, Inc.
Price: from $360.00 to $620.00
Circle 78 on Reader Service Card

PRODUCTION PACKAGE

● The Stereo Field Production Package consists of the AT4462 Stereo Field Production Mixer, the AT825 One Point X/Y Stereo Field Recording Mic and the AT804 Omnidirectional Moving Coil Dynamic Mic. The AT4462 mixer features two stereo inputs and two pannable "mono" inputs. Mic inputs are configured with a 20 dB pad and a lo-cut filter. Additional features are also included. The AT825 stereo condenser mic is ideal for stereo field recording. It is equipped with switchable lo-cut filter, windscreen, two-way power and a pair of closely matched UniPoint miniature condenser cardioid elements. The AT804 hand-held, omnidirectional dynamic mic features high output and smooth response.
Manufacturer: Audio-Technica U.S., Inc.
Prices: The AT4462 is $1,395.00; The AT825 is $399.00; The AT804 is $158.00
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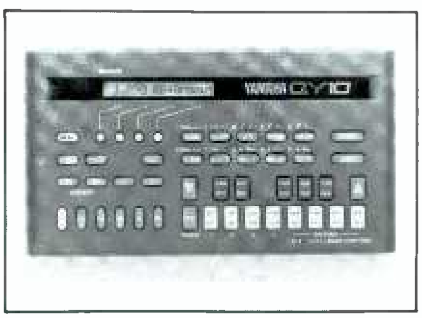
TINY BUT POWERFUL



● The QY10 provides options of twenty-nine sampled instruments and twenty-six sampled drum sounds. Up to eight different sounds and twenty-eight individual notes may be played at once. The rhythm section has seventy-six preset backing patterns with memory for twenty-four more of your own. These patterns contain drum, bass and chord combinations of 1-8 measures, many with strings or brass parts. The sequencer/recording portion offers eight songs and eight tracks. Realtime and steptime recording are available, with extensive editing of even the slightest nuance of what was recorded. MIDI IN and OUT connections also allow the QY10 to be part of a larger MIDI system when desired. Other features include stereo line outputs, LCD display, stereo headphone jack with volume and DC power input.
Manufacturer: Yamaha Corporation of America
Price: $399.00
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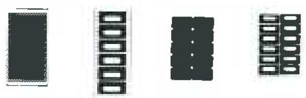

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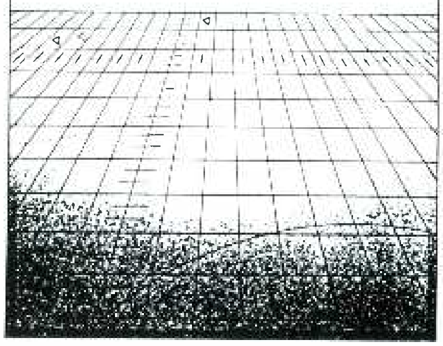

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PEOPLE, PLACES & HAPPENINGS

• **Mark IV Audio President Robert Pabst** announced the formation of the **Mark IV Audio Broadcast/Production Group**. The group will be responsible for coordinating all Mark IV Audio marketing and product management efforts in television and radio broadcast markets, as well as film and television production, multitrack recording and other related markets in the United States.

• **Brooklyn Recording Studios**, which has an emphasis on vintage analog recording gear, has opened in Los Angeles. Bill Dooley, formerly chief engineer and general manager of **The Record Plant,** heads the facility. Some of the vintage equipment includes an assortment of mics from **Telefunken, Neumann, Sennheiser, AKG, Electro-Voice** and **Shure Brothers**.

• **The Record Plant** (Los Angeles)itself has been purchased by **Summa Music President Rick Stevens**. The facility, which opened June 10, will focus on music recording, with a special emphasis on a service-oriented management style.

• **Audio-Technica U.S., Inc.** is now distributing all of its professional audio products through Canada.

• **Sennheiser** has announced its first **Infrared Theater** Installation solely intended to assist the blind at the **Theatre National De Chaillot**. The system, which utilizes Sennheiser's infrared transmitting and receiving equipment, allows the blind to visualize and imagine the stage by relying on their senses. Sennheiser's HDI 2 Infrared Assistive Listening headsets are used in conjunction with the SI 1013 Infrared Transmitter and SZI 1219 Infrared Emitter Panel to complete the Audiovision system.

• **Channel 4 Television** in London, England, has become the world's first television company to order the **CEDAR** (Computer Enhanced Digital Audio Restoration) Sound Restoration and Production System. The system is designed to address all classes of degradation—broadband noise, scratches and pops, thumps, buzzes and hums—and removes unwanted noise from both new, or old and damaged, audio recordings. In other equipment purchases and acquisitions...**The Olympic Stadium** in Athens, Greece, has updated its sound system and installed 70 **Gauss** 4081 compression drivers which are mounted on five-meter steel columns strategically placed throughout the 80,000-seat stadium...**Altec Lansing** VIR and VIT (Vari-Intense) equipment has been installed in the **Hofheinz Pavilion** at the **University of Houston**; **Grace Baptist Church** in Belleville, MI; **St. Patrick's Church** in Victor, NY; and **Northville High School** in Northville, MI. The technology employs a new design principle for horns...**QSC Model 1400** amplifiers are now in place at the **United Cinemas International** theater chain. **Sound Associates**, one of the largest suppliers of cinema equipment in the U.K., installed the equipment...A second **Adams-Smith** 2600 E-A/V audio for video editor has been delivered to London's **Tele-Cine Cell Group**, a film-video, post, duplication and computer animation company. A 2600 rack with the compact controller has been purchased by noted U.K. sound mixer **John Wood**. Other purchases of the 2600 system have been made by **Howard Schwartz Recording** in New York City and **John Lawrence** of **VidFilm Services** in Glendale, CA.

ARX Systems has been delivering equipment throughout Australia and other parts of the world: in Adelaide, Australia, the **Festival Centre Complex** has updated their audio systems with an assortment of ARX equipment; the new **Performing Arts Centre** in Seoul, Korea, will be outfitted with a complete ARX electronics and loudspeaker system. To date, the installation will be the single largest ARX installation in Korea; in Traralgon, Australia, **Strangelane Productions** has purchased a complete ARX KA115/212 touring system; a complete 912 based sound system for touring through the Northern Territory of Australia has been delivered to the **Groote Island Land Council**; and other ARX Systems purchases have been made by the **Australian Broadcasting Commission; CBS Television** in New York; and the **Hessisches Staats Theatre** in Weisbaden in Frankfurt, Germany...Eleven **University Sound** UW-30 speakers were installed in **The Tokyo Metropolitan Gymnasium** in Japan, and the underwater speakers were also installed in the **Rose Bowl Aquatics Center** in California. Other University Sound installations include an 1802 preamplifier/mixer in the **Cambria County War Memorial Arena** in Johnston, PA; and a new sound system at **Fenway Park** in Boston, MA, that includes 250 CDP horns.

• **Michael Price** has been appointed vice president of engineering at **E-mu Systems, Inc.**... **Studer Revox America, Inc.** has appointed **Thomas M. Spain** to national sales manager, **Revox Division...Mark Lierly** has been named national sales manager at **SoundTech... Andre Macaluso,** vice president and general manager of **Audio Plus Video International/International Post (APVI/IP)**, has been named group vice president for VSC. In his position, Macaluso will remain responsible for APVI/IP and will also oversee the operations of **Atlantic Satellite Communications, Inc.** and **Waterfront Communications Corporation.**

• **Ampex Recording Media Corporation** appointed **David H. Davies** as vice president, Development Engineering, where he will be responsible for product and process development programs. db


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