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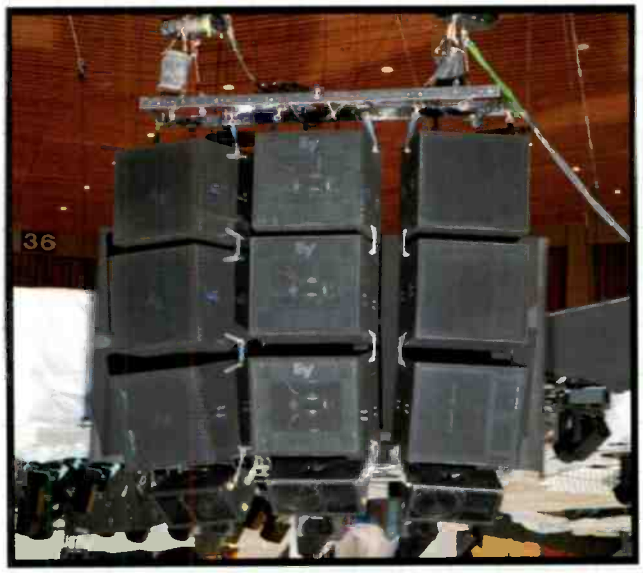
Sound On The Stage

- A Talk With Maryland Sound
- The Gyuto Monks in Performance
- UB-40 in Israel
- Winter Special Olympics in Reno

Hot Tips for Small Studios

Hoffner, Gately, Battles, and Bartlett



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MAGAZINE

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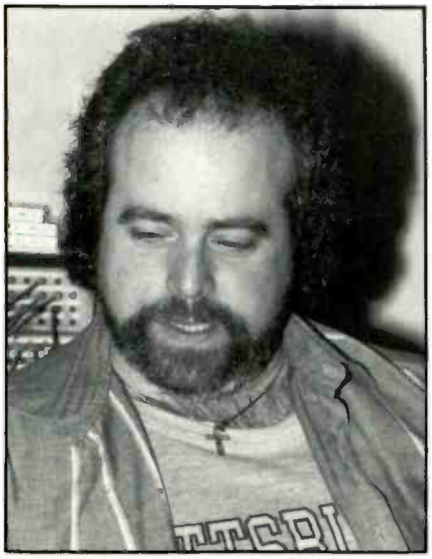

see page 85

About the Cover

● To keep with this issue's theme of Live Sound, we show a Shure Beta 58 microphone, Carvin's MX2488 sound-reinforcement console, a Soundcraftsmen 300X4 multi-channel power amplifier, and our speaker system comes from Electro-Voice and is their Manifold System MT-4.

JULY/AUG 1989 VOLUME 23 NO.4

The sound contracting engineer

The broadcast engineer

THE ELECTRONIC COTTAGE

db

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Letters

The Editor:

I certainly was taken aback to read Drew Daniel's letter (**db** September/October 1988, p. 48) that I had "pedantically insisted that (he) should have used five decimal places in (his) work where (he) used only four, and that (he) should have striven for 0.1 percent accuracy instead of 5 percent" in my comments after his presentation on Theile/Small parameter measurements during the 70th Convention of the AES in New York in 1981. I couldn't recall saying anything of the sort, and to have made such a comment would indeed appear asinine to say the least. However, as this was seven years ago, I obtained from the AES a copy of the archival recording of Drew's paper and the subsequent questions to refresh my memory. Here verbatim is the relevant portion of what was actually said during the question period.

SPL: "I have what really amounts to partly a question and partly a comment, and that is concerning the free-air resonance, the f_s, of the driver. My understanding is that that is not strictly speaking the resonance frequency of the loudspeaker hanging in free space, but is in actual fact the resonance frequency of the loudspeaker mounted on a baffle which approximates, in the same way the box does, the separation between the front and back of the driver, but does not have the compliance of the air in the box. In other words, in many cases it would be the speaker mounted in an infinite baffle-a true plane."

DD: "What we did in practice was to hang the speaker in free air as close as possible..."

SPL: "That is not what I am getting at. The loudspeaker should, strictly speaking, not be measured that way. That is my understanding. Perhaps our chairman would be able to correct me. In practice I believe the difference is very small, and I don't think that is very significant. I believe strictly speaking the loudspeaker should probably be mounted on a large plane baffle."

DD: "Strictly speaking you're probably right, but as I say, the cumulative errors in this type of testing can be as high as 10 percent when you've got it all accumulated, and 10 percent is less than a dB, and if you can tolerate that kind of ripple as in a commercial system or anything like that, it doesn't pay to waste the time in manhours to get every last decimal point."

What I had been getting at was the fact that the air-load mass should be kept as near as possible to that experienced by the driver in use, when measuring f_s, and this may not be the case when fs is measured on a totally unbaffled driver, as done by Daniels. Indeed, Daniels himself says on page 3 of his preprint (#1802): "The air load on the cone of the drivers is important," and during his talk he said: "These air loads are critical...the presence of a person within 2 feet or so can make a big difference to a large woofer." He largely missed my point, but chairman Don Keele understood it, and later in the question period commented:

Keele: "Normally speaking, when you're doing a box design, you want a free-air resonance with the total moving mass which includes all the associated air-load masses on the front and the rear, and so you want to put it in some kind of a baffle that's not enclosed but yet simulates the air-load masses on front and rear...Now we've found, for the most part, from a practical standpoint we don't measure it on a typical baffle at JBL. Just measuring it in free air and forgetting about it gets you close enough."...(and again later) "Normally for low-efficiency heavy drivers the air-load mass is a small portion. Where you do get in trouble is with high-efficiency, low-mass systems, especially with large diaphragms, so you can get in trouble that way".

But where did I insist on the ridiculous five decimal places? I didn't. I never said or implied anything of the sort. I hope that I am not that far removed from reality!

Stanley P. Lipshitz
Audio Research Group
University of Waterloo
Ontario, Canada

Ed note: We have known Stanley Lipshitz for many a year now and we know that he is not removed from reality.

The Editor,

We were very interested in the article by Corey Davidson in the March/April issue "On tour with the Dixie Dregs", which talked about how Ken Hirsch had modified his Omnicraft GT4 noisegate.

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The IEQ with Smartcurve™ is a programmable, high performance graphic equalizer that includes a video output. For those who wish to enjoy the video output of the IEQ, ART makes the IEQ Video Monitor.* Smartcurve™ proprietary software developed by ART gives you instant actual frequency response as easy as the push of a button.
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- THD - ≤ .009% @ 1kHz, 0dBM typical
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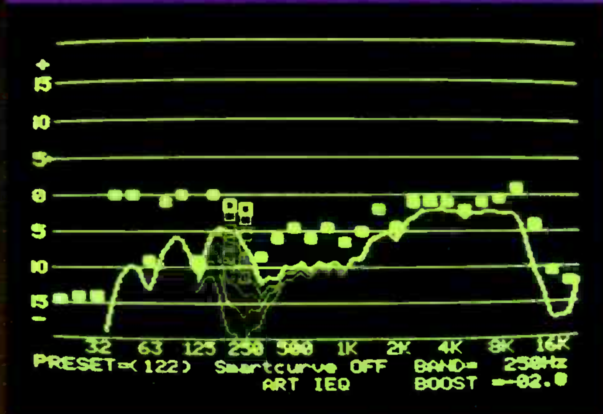
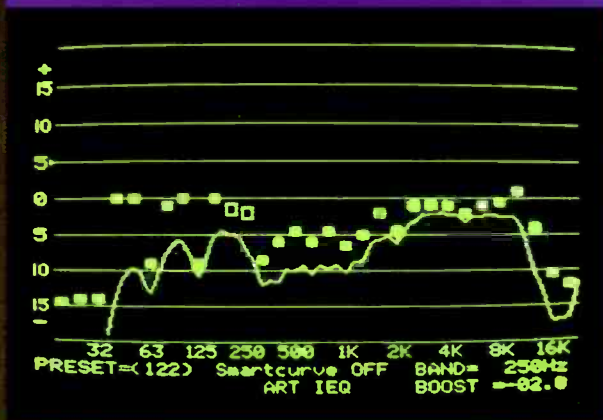


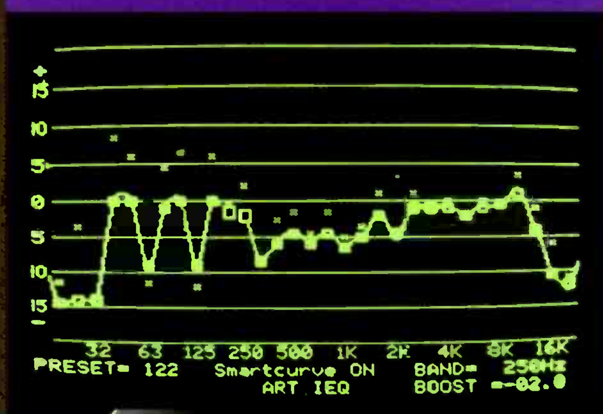
1 See the Sound

This is a video output of the IEQ as the unit is being adjusted. The sliders can be moved ± 15dB in 1/2dB steps to get the exact response you need. With the simple push of a button complex equalization can be done in seconds with incredible accuracy.



2 Hear the Sound

The power of the IEQ readily becomes apparent as the video display plots the frequency response due to the slider settings. The IEQ offers high quality constant "Q" equalization. The video graphic display shows the correlation between the sliders and the frequency response.



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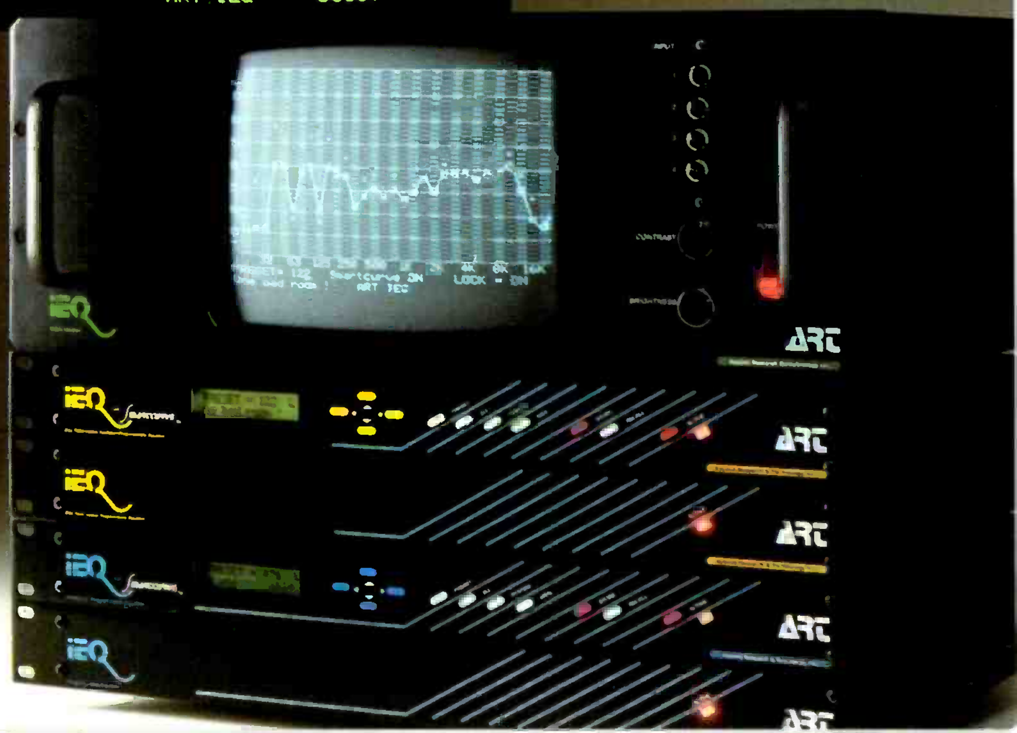

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Circle 17 on Reader Service Card

We bought out Omnicraft about three years ago, and by that time—as a result of comments from users—the GT4 had been updated to the GT4A which included the modification similar to that described by Ken.

The control is called 'range' and operates as described in the article, *i.e.* controlling the amount of gating without in any way affecting the threshold point at which gating first occurs.

To echo Ken Hirsch, the Omnicraft is still one of the lowest cost gates on the market, at 4-channels for $395; it is also the quietest due to its optical circuitry.

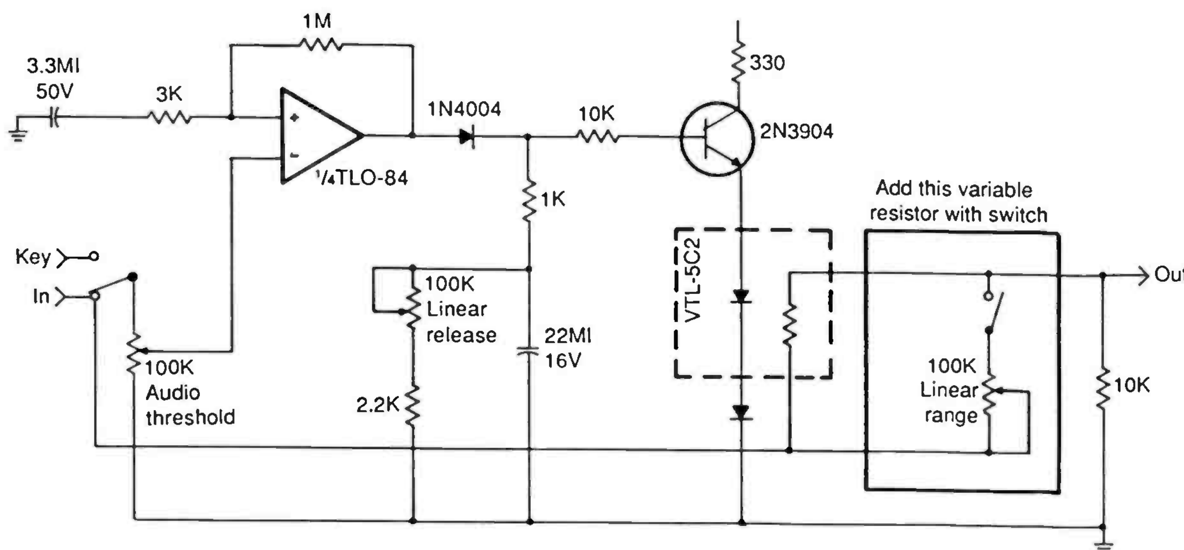
For the benefit of those people who still have the old GT4, we do not, by the way, recommend doing the modification as shown on the circuit diagram.

That was *Figure 5* of the article, the d.c. conditions of the drive transistor and the optical switch will not remain stable with this technique. A correct modification is shown below.

Janet Page Walton

Publicity Manager

C-T Audio Marketing, Inc.



This is the recommended rewiring schematic.

Calendar

• **Pro Audio Asia '89**, the international trade exhibition for professionals in the broadcast, recording, public address, installation, contracting, and duplicating industries, will take place from July 6-8 1989 at the Hong Kong Exhibition Centre in Hong Kong.

• **Synergetic Audio Concepts** of Norman, Indiana, announces a series of special three day classes to be held at the Syn-Aud-Con farm in Southern Indiana. These classes feature the opportunity for gaining direct learning experiences for specific objectives in fundamental audio systems measurement and analysis.

The Farm in Southern Indiana:, July 21-23, August 24-26, September 22-24, and October 5-7.

New York area (Secaucus, NJ): October 17-18

DC area (Rockville, MD): October 26-27

Orlando, FL: November 15-16

For additional information, call or write:

Synergetic Audio Concepts

RR #1, Box 267

Norman, IN 47264

(812) 995-8212

• **The Festival International de Jazz de Montreal's** tenth birthday celebration will take place from June 30th to July 9th at the Montreal Forum in Quebec, Canada. Scheduled to appear are bluesman B.B. King, jazzman George Benson and many others.

• The **Institute of Audio Research** is offering short courses in July and August. These include: Producing for Recording Engineers, Advanced Studio Maintenance and Live Sound Reinforcement, to name a few. For more info, call **(212) 677-7580** and ask for **Angelo Cherta** or **Paul Zeman.**

• The **AES 87th Convention** will take place on October 18th through the 21st in the Hilton in New York City. This convention offers exhibitors the time and forum to meet thousands of audio professionals who are eager to learn and experience the newest technology, and to make product decisions.

For additional info, write or call:

Christopher Plunkett

Audio Engineering Society

60 E. 42nd Street

New York, NY 10165

(212) 661-8528

• To those of our readers in far flung places, **Ed Learned** will be in the following areas with a 4-piece blue-grass western and country band called B.C.H.& F. Look him up if he's in your area and tell him you saw it in **db** magazine.

June21 Auckland, New Zealand25 Napier, New Zealand26/27 Wellington, New Zealand28Dunedin, New ZealandJuly 01 Tonga04 Western Samoa09 Fiji13 Honiara, Solomon Islands19 Port Moresby, Papua New Guinea21 Ponhapei, Federated State of Micronesia24 Majuro, Republic of the Marshalls

BRUCE BARTLETT

Sounds and Surprises at the Winter 1989 Special Olympics

The shoosh of skiis and skates, the clatter of hockey sticks, and the music of great performers filled the air at the Fourth International Winter Special Olympics Games (IWSOG). Held in the Reno/Lake Tahoe area from April 1 through 8, 1989, this competition was the world's largest Winter amateur sporting event.

It celebrated the skill and courage of athletes with mental retardation. The Games help the athletes develop not only physical fitness, but also a sense of self-esteem and accomplishment.

Over 600,000 volunteers are involved in Special Olympics programs worldwide, including hundreds who worked hard to provide quality audio for the Games.

Nearly 1,400 athletes from all 50 states and 17 countries competed in five sports:

- Alpine skiing at Squaw Valley (home of the 1960 Winter Olympic Games)
- Nordic skiing at Royal Gorge (site of the National Nordic Skiing Championships for three years)
- Figure skating at Lawlor Events Center in Reno
- Speed skating and floor hockey at the Reno-Sparks Convention Center.

Events during the week-long competition began with the Opening Ceremonies at Squaw Valley, complete with a parade of athletes, torch lighting, and entertainment by major stars. Cultural exchanges were held throughout the Games with educational clinics, family activities, international dances, musical shows and receptions. Closing Ceremonies featured a variety ice show and a large-screen slide presentation of the athletes in action.

All these events, plus the sports announcements and awards, required sound systems. Let's look at what went on behind the scenes to make it all possible.

PRE-PLANNING

Tom Durell, a free-lance audio engineer, worked with Crown International to prepare the audio systems for the Special Olympics. Tom was a sound engineer for Liberty Weekend and the 1984 Summer Olympics in Los Angeles. He also designs or operates sound systems for TV shows, TV commercials, and movies. He also did audio for Disney for 15 years.

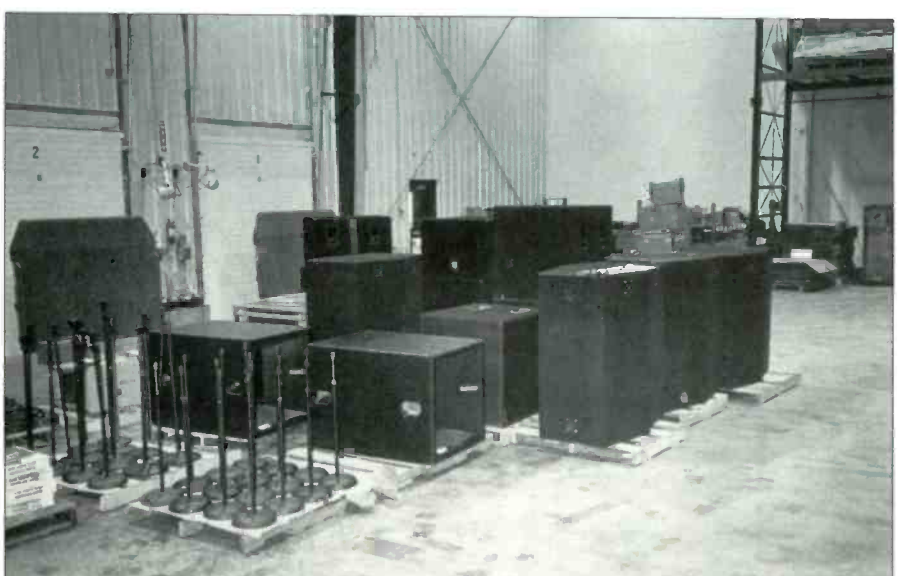
Working closely with Tom was Crown's Bill Raventos, Product Director/Microphones. Crown International (of Elkhart, Indiana) is a manufacturer of professional power amplifiers, microphones, TDS analyzers, and related electronics.

The International Special Olympics required several elaborate sound systems because much of the program included musical entertainment. In addition to the Opening and Closing Ceremonies, concerts kept the spectators occupied and entertained. In all, over 100 audio systems were installed, operated, and torn down in 7 days!

For the majority of the systems, a special computer program called AVAIL was used, with an operating system provided by THEOS. This software allowed the system designers to assign sound systems and technicians to venues. Data entered into Avail included the names of the available technicians, their capabilities, the equipment, and the sound jobs that had to be done.

Crew personnel were trucked in and out from a central donated warehouse containing the sound equipment. Supervisors made sure that each sound system got to the right place in time for setup, and that there were people to operate them. When each system was finished being used, the trucks were sent to pick up the system and bring it back to the warehouse (*Figure 1*).

Figure 1. Audio equipment, destined for the Winter Olympics was staged in a warehouse.



You can't build a better mousetrap unless you know your mice.

For years, those of you who wished to enjoy the many benefits of processed speaker systems had to endure one major drawback: loss of sound quality when the system switched into the protect mode.

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We also rigged these systems for flight—with optional three-point harnesses equipped with steel-reinforced pan-fitting hardware, perfect for safely arraying speakers in multi-cabinet clusters.

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BROADCAST COVERAGE

Turner Broadcasting System provided TV coverage of all events, including the concerts. On April 9, TBS aired a two-hour program on the Games. Award-winning sports producer Tony Verna, executive director of TBS Inc.'s telecast of the 1990 Goodwill games, was producer and director. Video One of Hollywood did the audio and video recording.

Figure 2. The opening ceremonies at Squaw Valley.

OPENING CEREMONIES SYSTEM

The Games were kicked off by the Opening Ceremonies held at Squaw Valley on April 2 (in a snow/sleet storm!). It featured the Parade of Athletes, the Torch Lighting Ceremony and entertainment by world-famous celebrities. Olympic gold medalist Bruce Jenner emceed, and music was provided by the Grammy-award-winning group, Take 6. Eunice Kennedy Shriver, chair-person and founder of Special Olympics International, announced the official opening (*Figure 2*).

The sound system employed several Community Light & Sound RS880 and VBS415 speaker systems with 880EQ Dynamic Equalizers, Crown MacroTech 2400 amplifiers and microphones, and a Yamaha PM3000C console.

Unfortunately, heavy sleet fell continuously. "It was miserable!" reported a sound technician. "Eight hours of *Grapes of Wrath*. People were wringing out their gloves. We looked like drowned rats."

The ground was so muddy that one crew had to improvise scaffolding for the mixing position, putting planks over the mud and plywood on top. Winds gusted up to 50 m.p.h. Care was taken to protect the equipment during moving, setup and teardown. Plastic sheets were strung over the console and speakers, including scrims set over the horn mouths.

SYSTEMS AT LAWLOR EVENTS CENTER AND RENO-SPARKS CONVENTION CENTER

Lawlor Events Center is Reno's new 12,000-seat facility located near downtown. The audio for the ice-skating arena used the house central speaker cluster, including twelve E-V dual 15" woofers, JBL horns and drivers, and Altec Manta Ray horns. Some of the hooks holding up the cluster had pulled out, so the speakers needed to be strapped in place. A Crown CM-310 Differoid was the announce microphone for the sporting events. The Reno-Sparks Convention Center used a similar setup.

Crown volunteers, including Joel Lewitz (of Paoletti & Lewitz, noted consultants from San Francisco) spent hours debugging and upgrading the entire system. This was necessary to get the system back in spec after years of maladjustment and lack of preventive maintenance.

LUNCHTIME ENTERTAINMENT SYSTEMS

To provide musical entertainment during lunch and dances, small-to-moderate systems were set up. The small system at Squaw Valley included the following equipment:

- Yamaha MC1602 mixer
- Two Community CS70 house speakers
- Community CS38M monitor speaker (*Figure 3*)

The rack included:

- Tascam 112 cassette deck (for background music)

Figure 3. Even at lunchtime, an entertainment system had to be set up.

●Yamaha SPX90 multi-effects processor
●Rane AC22 active crossover
●Two White 1/3-octave graphic equalizers
●Rane DC24 Dynamic Controller
●Two Crown MacroTech 1200 power amplifiers (one used).

A typical act was a singer/guitarist or small group. We used a Whirlwind DI for the acoustic-guitar pickup and a Crown CM-310 Differoid Microphone for vocals (to avoid feedback in close quarters). For a slap echo, we patched into the performer's effects. The special athletes in the audience were totally into the music—a joy to see.

A larger system was used in the lunch room at the ice- skating arena. It included the following equipment:
●Community CS70 house speakers
●Community CS38M floor monitors
●Yamaha 1604 MC Series mixing console
●Yamaha SPX90 multi-effects processor
●White graphic EQ
●Crown MacroTech 1200 power amps (one for the house, one for the monitors as seen in *Figure 4*).

We created an input list for each act and kept it near the snake box; this helped us keep a handle on things during the chaos of mic'ing each act. I mixed the show between bites of pizza. It was an enjoyable gig: nothing failed; the mixer was effective and easy to operate, and I didn't have to fight the speakers to correct their response.

Three acts played during lunch, and most of the kids danced with tremendous enthusiasm. For many of the bands, the audience response was the best they ever had.

Later, during the strike, we wrapped the cables with Cord Lok™ Velcro™ strips—a much-appreciated convenience.

AWARDS SYSTEMS

Located at all the sports areas, each of these systems included four Community CS35 speakers on Ultimate Support Systems stands, an HME wireless microphone with a CM-200 head, and a rack with a mixer, cassette deck, graphic EQ, complimiter, and power amps. This can be seen in *Figure 5*.

Virtually every athlete who competed got an award. The award ceremonies ran continuously, with a person announcing the names of the people standing on the graduated steps. After the awards were presented, a cassette tape of the Olympic theme was played, the athletes went off and the next group came on.

Figure 4. The lunch time entertainment system.

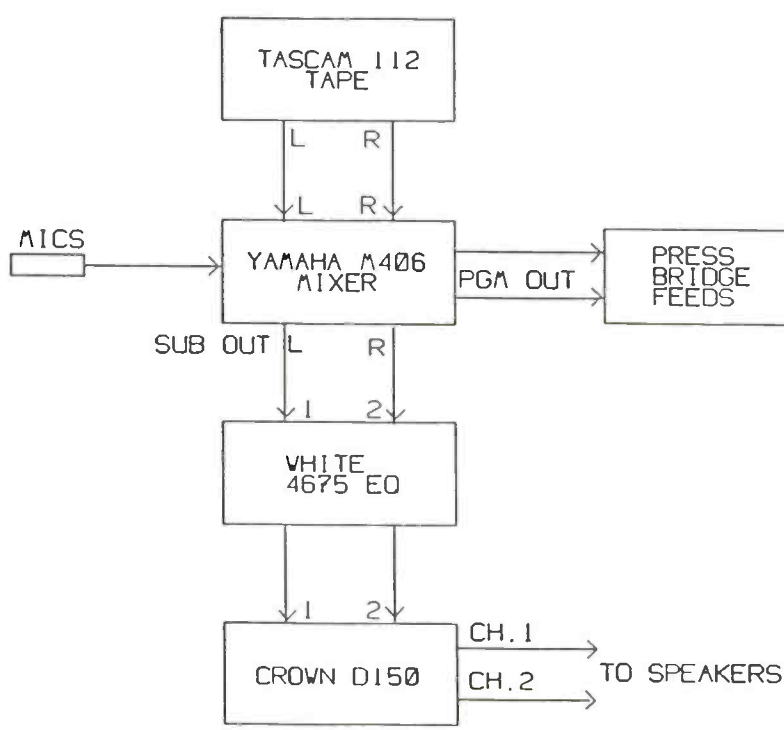
BEACH FEST SYSTEM

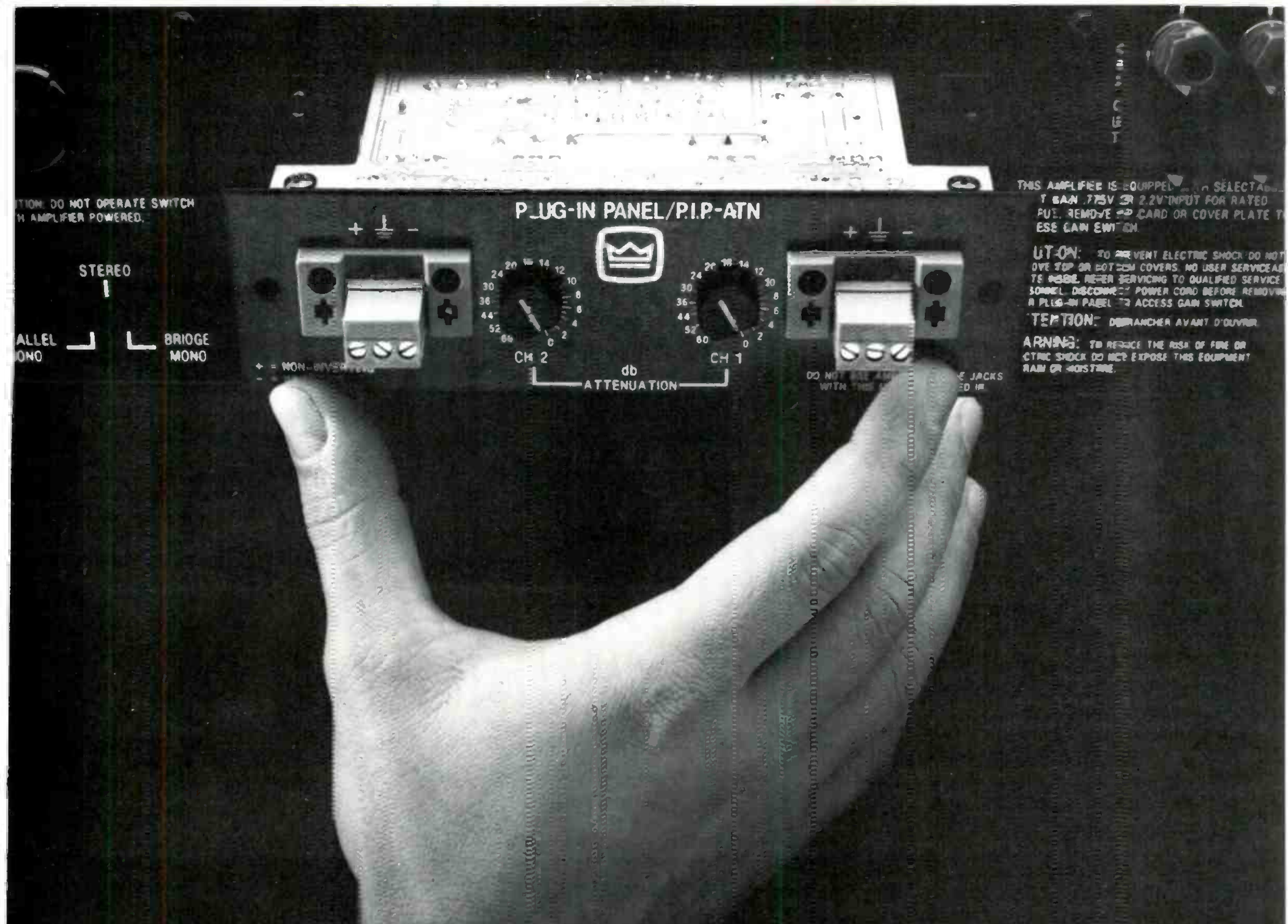
Red Lobster sponsored a Beach Fest, a day-long party with a DJ and games on a huge simulated beach: 1400 tons of sand dumped on a parking lot, complete with volleyball nets and lifeguard posts. Red Lobster tours this event around the country.

A 60's revival concert was held there the evening of April 4, and it required a major concert sound system.

Five Crown MacroTech 2400 power amplifiers powered the main

Figure 5. A block diagram of the rack system (courtesy of Ken Kuespert.)





These days, it seems like everyone's putting in a plug for Crown.

And we don't just mean celebrity endorsements, either. We're talking about Crown's exclusive Plug-In Panel (P.I.P.)™ technology—yet another reason why so many big names insist on touring with Crown Macro-Tech™ amplifiers.

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Electronic crossovers. 32-step precision attenuation (shown above). *And now, for the first time ever, total remote control and monitoring of up to 2,000 amplifiers from one central computer (IQ System 2000™).*

Only Crown gives you all these expandable options. Plus a list of standard features that really wowed them at the Patent Office.

Like Crown's exclusive Multi-Mode® (A, AB + B) and Grounded Bridge Circuitry for earth-shaking power with almost no distortion. Our patented IOC® indicator that flashes if any distortion equals or exceeds 0.05%. Unique ODEP™ circuitry that can actually *prevent* overload and overheating damage. And, of course, front panel LEDs that tell you the operating status of each amplifier at a glance.

All this *plus* your choice of stereo, bridge mono or parallel mono modes. And a transferable No-Fault Warranty* that only *the most reliable amp in the industry* could afford to offer: 3 years parts and labor, free round-trip shipping, *plus* a 3-year extendible option.

With this much going for it, it's no wonder Crown Macro-Tech is the most talked-about amp on the touring circuit today. Call us today for more information. Because chances are, *you'll* be putting in a plug for Crown one day, too.

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speakers. There were 10 channels of power with one 2- ohm speaker per channel. MacroTech 1200's powered the monitors, making the total system power nearly 20,000 watts.

The main speakers, five per side, were Community RS880 systems with dynamic equalization. These new systems have horn-loaded dual 15" woofers, a horn midrange and horn tweeter, internally wired with a passive crossover. The voice coils are cooled by ferro-fluid. A passive equalizer ahead of the power amp provides some low- and high-frequency boost, so that the speakers uniformly cover the range from 50 Hz to 20 kHz. In addition, two Community VBS415 subwoofers were used per side. Each contains four 18" cones to cover the range from 25 to 50 Hz. Stage-floor monitors were Community CS38M (*Figure 6*).

The system block diagram was unusual (*Figure 7*). At the output of the power amp was a sensing line that provided feedback to the dynamic equalizer. To protect the drivers, the dynamic equalizer provided 6 dB of compression (from 60 V to 30 V) and the passive EQ flattened out if the system went into compression.

The quality of these speaker systems was apparent when I looked at

Figure 6. Community Light & Sound's VBS415 Subwoofers installed at the Beach Fest.

the graphic EQ settings chosen by the engineers: the EQ was nearly flat.

The sound system was debugged by Tom Durell, Community's John Wiggins, Ken Kuespert, and Bill Porter. All of these folks were volunteers who worked on the Crown Audio Group Team, many paying their own travel expenses to be there. Ken Kuespert is Director of Field Operations for TPC Production Services and a free-lance audio engineer; he was our hum-fixing genius. Billy Porter is the legendary recording engineer who mixed hundreds of hit records for such stars as Elvis Presley, Roy Orbison, Johnny Tillotson, and the Everly Brothers.

When the system was first powered up, it had hum. One of the Community 880EQ Dynamic Equalizer pro-

Figure 7. A block diagram of the Beach Fest System.

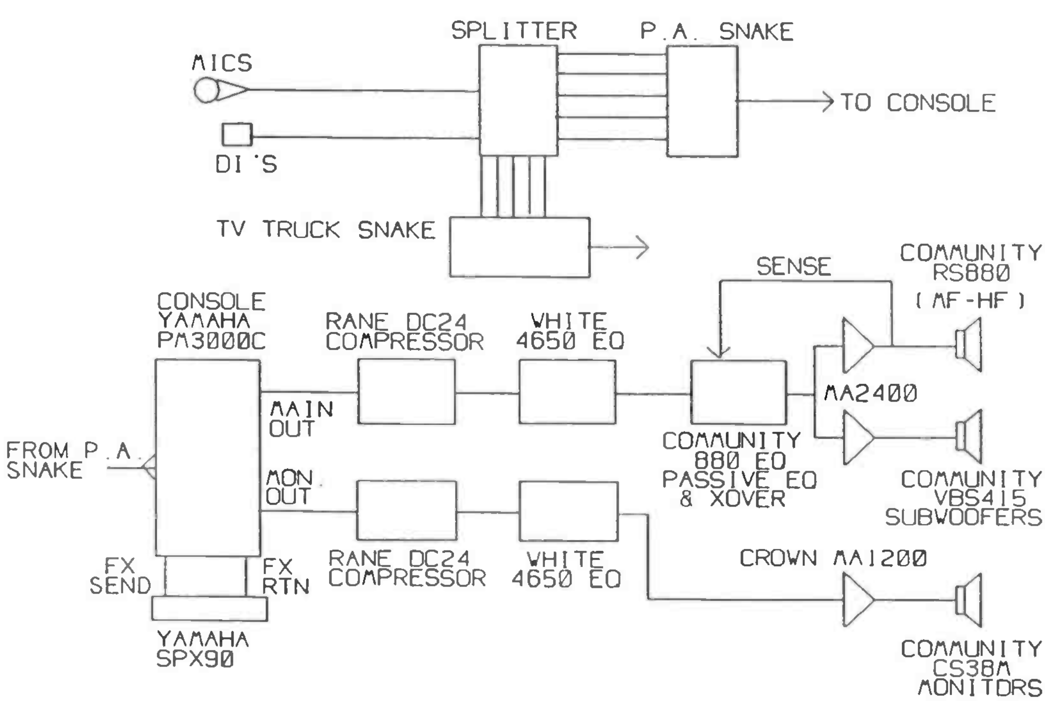

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Figure 8. Bill Porter, wearing a hat, is seen setting up the stage microphones for the Beach Fest.

totypes was unbalanced (production models will be balanced). The unbalanced signal was suspected of causing hum, so a Rane SP15 studio parametric was used in an attempt to balance it. But the Rane had a differential output which was thought to cause hum. When the Rane was bypassed, it became unbalanced. A DI box was added for balancing, but it seemed to introduce distortion in the line-level signal. Finally a White 4650 graphic EQ was used to balance the signal, and the hum stopped.

This is a good example of creative engineering enabled by skilled people working with available equipment.

The system also sounded distorted. One cause was a compact disk used for testing; the CD itself was distorted! In addition, the amplifiers and speakers were being overdriven in an attempt to get enough volume outdoors. Two more speakers and amplifier channels were installed, which resulted in a clean signal with adequate volume.

As for stage mic'ing, the drum set was covered with two Crown GLM-100 omnis (kick and rack toms) and two GLM-200 hypercardioids (floor tom and snare). Main vocals and guitar amps used Crown CM-200 cardioid condensers, while keyboards and bass were picked up direct with Whirlwind IMP-2 DI boxes.

Vocals by the drummer and keyboardist were handled by Crown CM-310 Differoids, which are noise-cancelling stage-vocal mics. When these were soloed over headphones, almost no leakage was audible.

Bill Porter put gaffer's tape over each mic/XLR junction (*Figure 8*) to keep the connector from falling out. When we ran mic cables, a microphone split was provided for TV coverage.

Although the Yamaha PM3000C console performed flawlessly, it had perhaps too many features for this application, creating some confusion. We were unable to set up monitor (aux) mixes over headphones, so Bill had to rely on the musicians' suggestions.

The evening concert at the "beach" featured such early 60's stars as Bobby Vee, Tommy Rowe, Johnny Tillotson, Dell Shannon, Bryan Hyland, and Merry Clayton (from *Dirty Dancing*)—all well backed up by the Rockin' Ricochets. Also appearing (*Figure 9*) were TV celebrities Wayne Pryor, Stephanie Nichols and Tina Yothers.

The mix was done by Billy Porter, and there was no better choice for the job. He ran the audio in mono except for some panning on the vocals. The total mix was slightly compressed by a Rane DC24 Dynamic Controller to add punch, while a Yamaha SPX90 Multi-Effects Processor provided reverb.

Figure 9. The complete, Red Lobster sponsored, Beach Fest concert system.

Bill set the eight submasters at equal levels for a reference point, and set the level of each subgroup with individual channel faders. Then it was easier to mix the show with the submasters. He let me do the drum mix while he attended other matters. Bill cautioned me not to change the gain trims because this affected the monitor mix.

The man is *good.* He quickly created a warm, rich sound with a perfect balance—a great party mix! The crowd danced in conga lines, bunny hopped and tossed balls high into the air. At the end of a tune, Bill mischievously wiggled the guitar fader to add tremolo. "They're havin' a party; why can't we?"

It was a great pleasure to work with this gentleman and watch a master of his craft. *(See the sidebar on Porter's techniques).*

WINDHAM HILL CONCERT

Three Windham Hill acts played in the Pioneer Theater in downtown Reno. One Community CS-60 and one CS-70 speaker was placed at each side of the stage on the stage floor, powered by one Crown Macro-Tech 1200. Richard Johnson of HSA mixed the show. The audio crew reported no major problems except that a slow-working piano tuner forced them to limit the sound check to two hours.

CHARLIE DANIELS BAND CONCERT

This major concert was held at the Reno Livestock Events Center, a cavernous rodeo arena. The sound system used the same speakers and amplifiers as the Beach Fest system. Nine Community RS880 speaker systems were hung on a truss in an arc over center stage. Community CS-70 speakers provided rear fill: six were placed at stage rear aiming at the audience. Community VBS415 subwoofers were on the ground in front of the stage. A Ramsa S840 Ser-

Figure 10. The Community Light & Sound RS888 array set up for the Charlie Daniels concert.

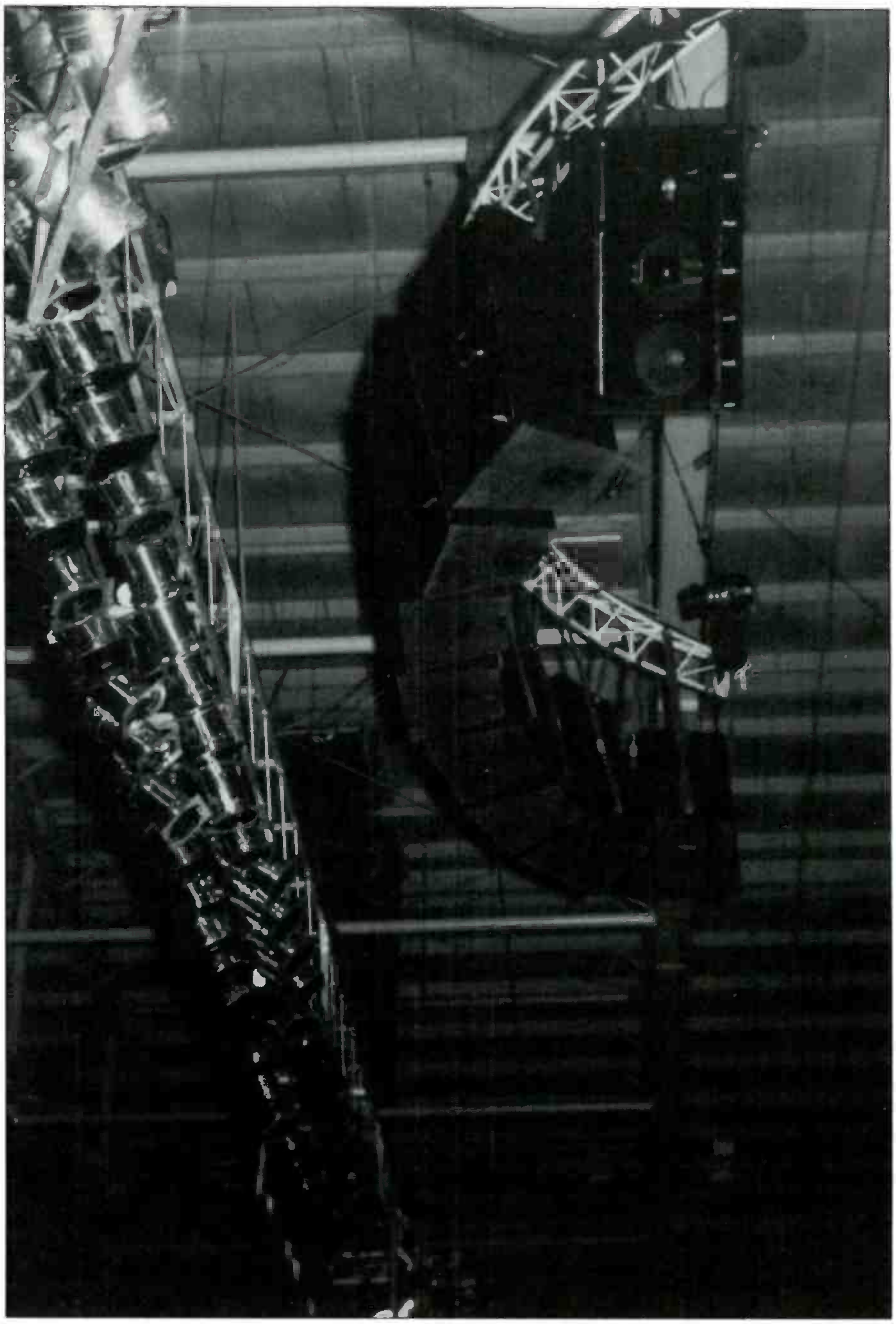
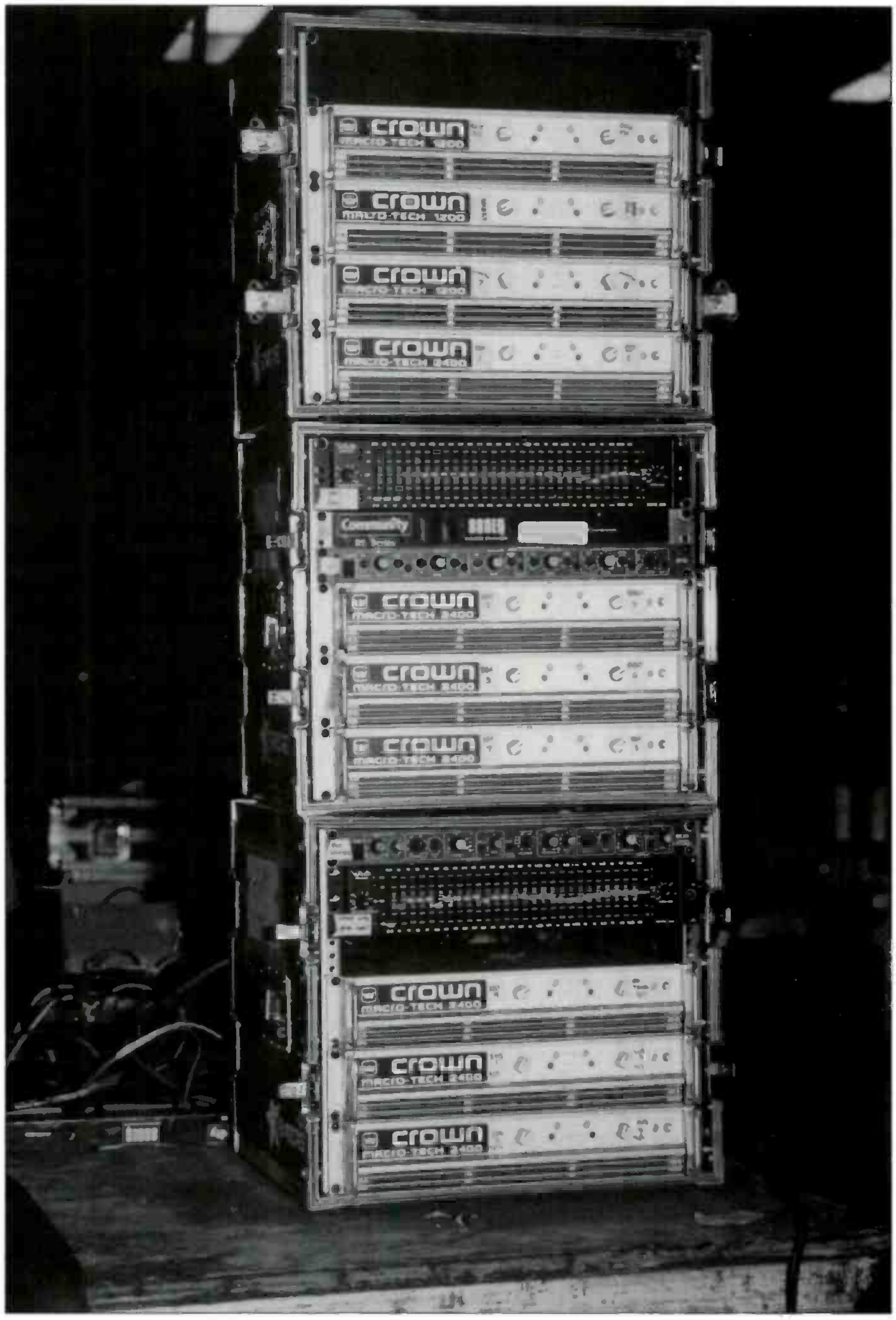


Figure 11. The power\processor rack for the Charlie Daniels concert.



ies house console was installed about midway into the audience on one side as seen in *Figure 10* and *Figure11*.

Figure 12 shows the stage setup. The show used a separate house mixer, monitor mixer, and recording mixer in a remote truck. House and monitor mixes (*Figure 13*) were done by Daniels' personnel.

Bill Raventos presented Charlie Daniels with a Crown Select Series CM-200 wood-handle microphone, engraved with Charlie's name. Daniels sang through this mic (*Figure 14*) during the show.

When first powered up, the system had hum. The cause turned out to be an unbalanced cable. A Community Dynamic Equalizer prototype was assumed to be unbalanced, so it was used with an unbalanced cable. But the unit actually was balanced tip, ring and sleeve. Changing to a balanced cable and lifting the ground cleaned up the hum. Ken Kuespert suggested that we add 1:1 isolation transformers with ground lifts to the rack, as well as distribution amps for feeds.

Finally, it was a few minutes to concert time. Cowboy hats were everywhere. Important lookin' dudes stood on stage alongside lanky sound techs and beautiful women. In the audience, a tidal wave of raised arms and hoots circled around the arena.

The house lights dimmed, cameras flashed, and the stage lights came up to cheers. The kick drum sounded like cannon shots, and the bass thundered throughout the enormous arena. The sound and the music were so exciting that folks jitterbugged and two-stepped behind the bleachers (Figure 15).

It was a high-intensity rock-type mix, and the vocals were fairly intelligible despite the room reverb. As Crown's Tony Satariano says, "A good rock mix should hit you in the face!" The sound level was uniform all around the arena.

After the concert, a huge cherry picker was employed to work on the truss. It was lowered with winches onto the stage, where we removed the mounting straps and packed the truck driven up to the stage. Billy

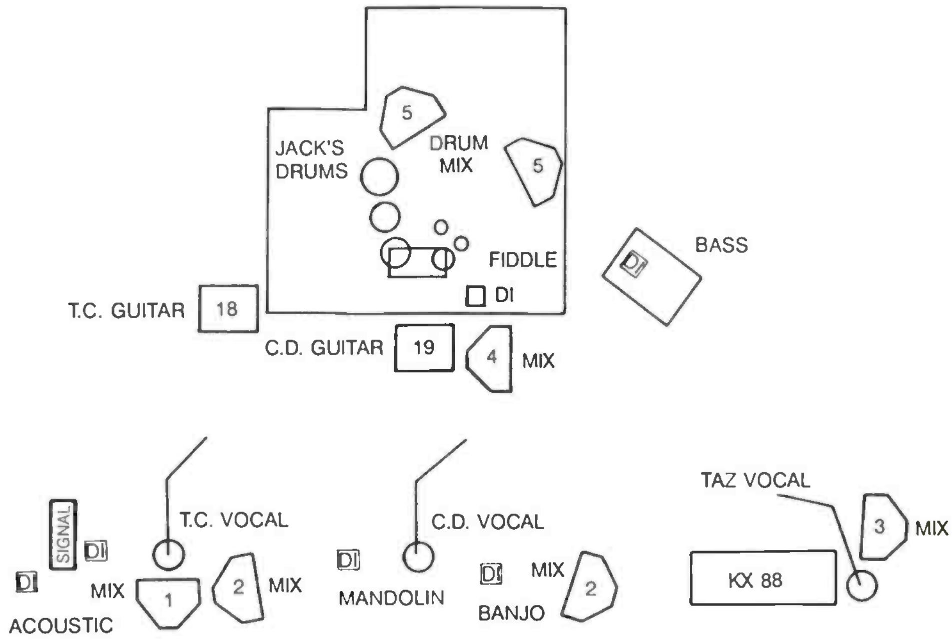


Figure 12. The stage set up for the Charlie Daniels concert.

Porter pitched in and helped us with this unglamorous type of labor all week long.

CLOSING CEREMONIES

Held at the Lawlor Events Center arena, this ceremony featured a variety ice show, a large-screen slide presentation capturing the highlights of the Games and a video-taped message from President Bush. The evening ended with a knockout fireworks display and laser light-show.

Equipment in the sound booth included a Yamaha 2404 console, Tascam 34B 4-track deck, Tascam 112 cassette deck, White 1/3 octave graphic equalizer, and a Rane Dynamic Processor. Crown CM-200 microphones were used for the MC, announcers, and celebrities at the lectern. Performers on the ice spoke through HME wireless mics with CM-200 heads, and the dual antennas of the receivers were gaffer-taped high on a wall. (Wireless handhelds with CM-310 Differoid heads were used elsewhere in the Games.)

Communications among the house mixer, lighting, and A/V personnel were done on HME headsets, both wireless (System 8100 and 8101) and cabled (BH720 and BH721). HME RP series power stations provided two functions: powering the cabled intercoms and assigning headset stations to channels.

The cockpit crew at the house mixing position included Tom Durell (director) who followed the script and gave cues to Billy Porter (mixer) and Crown's Joe Wisler (tape op). (Wisler formerly was head of audio for PTL.)

Before the show, Durell equalized the cluster with pink noise and a Cetec Ivie realtime analyzer, then finished tweaking by ear. After equalization, the cluster sounded more like music and less like speakers. Basically, the EQ was a cut in the ranges of 150 Hz and 3 kHz, and a high- frequency boost up to 12.5 kHz.

Musical selections for the show were provided to Bill Porter on several cassettes. To put it politely, they varied widely in sound quality. As Crown's Tony Satariano said, "You know a tape is bad if you can't tell whether it's Barbara Streisand or Bette Midler singing!" The cassettes were dubbed onto a ¼" program tape in the proper sequence, with leadered cues.

Last-minute changes in the ice show during rehearsal kept the engineers on their toes. A few hours before the show, the order of musical selections was re-sequenced. "Welcome to the world of live production," said Bill. Lee Taggert of Star Sound had to edit the program tape on the spot, right up to showtime! At one point Lee exclaimed, "I've got 8 inches of tape on a 7-inch reel!"

The edited dub of the Olympic Theme was too short, so we scrambled to find the master. Bill asked, "Where's the master tape?" "Here it is," said Lee, handing Bill a tangle of edited outtakes.

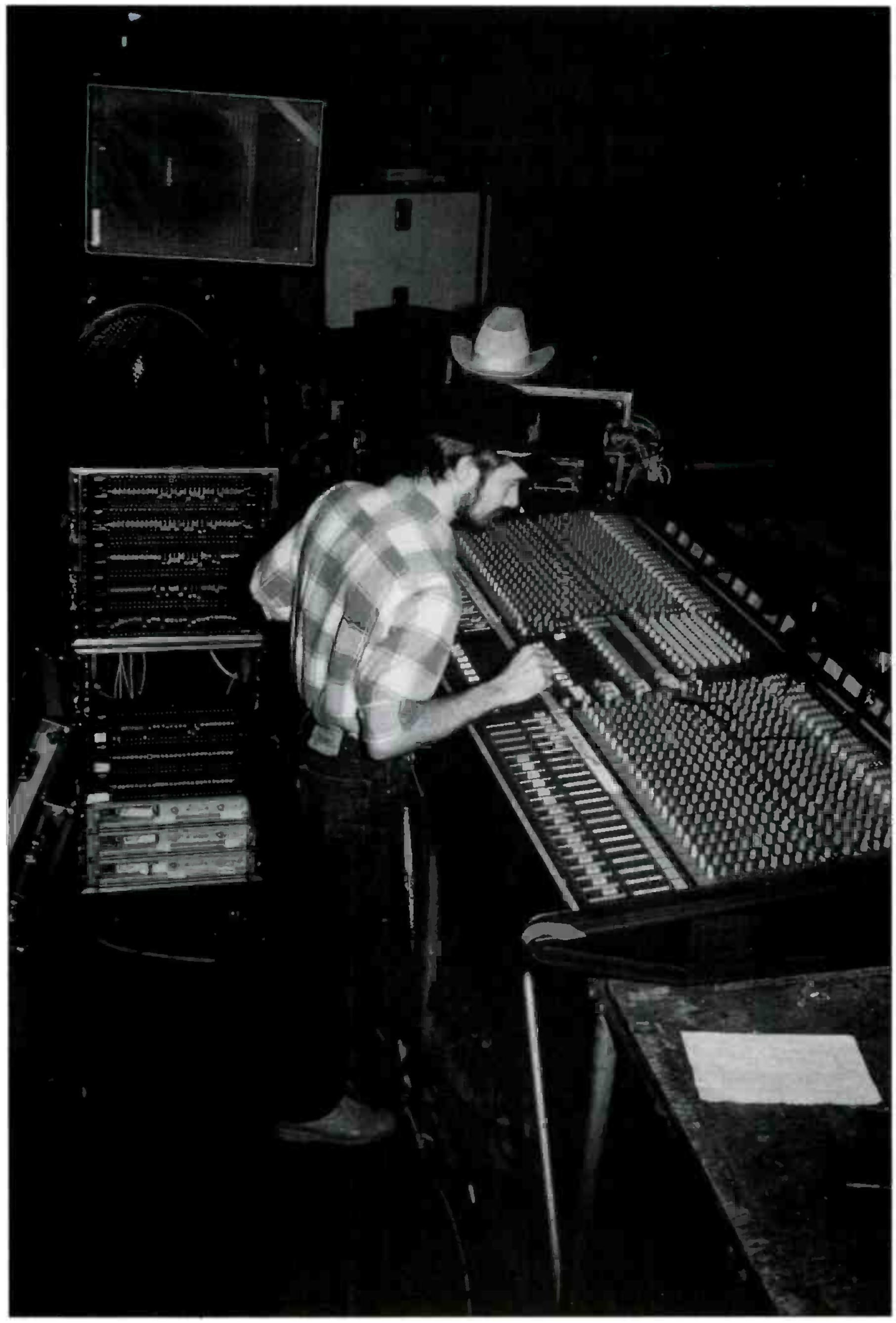
The sound-reinforcement console sent a stereo feed to the TV truck on mixer groups 1 and 2, while the house mix was sent mono on groups 3 and 4. The TV truck took some splits as well. Because of the 400-amp lighting dimmers in the walls, the TV feed had a little hum.

A wireless-mic receiver rack was miswired. "Its got user's disease," quipped Durell.

Finally the show began. It went smoothly for the most part. Joe Wisler performed tape cues with 4-track tapes, 2- track tapes, and cassettes brought in at the last minute. Bob Ramsey ably handled lighting cues, but his main function during the Games was to crack up the crew.

Bob did lots of hard work. He was phenomenal at rigging sound and miraculous with the lights for Clos-

Figure 13. The monitor mix position for the Charlie Daniels concert.



ing Ceremonies (he made something out of nothing).

After the spectacular finale of fireworks and lasers, the audio gang hugged each other and began teardown.

CONCLUSION

It was gratifying to experience the cooperation among all the workers. A bass player lost his guitar cord, but a musician from another band donated his. Star Sound of Reno was very helpful, working with us as if they were part of the same crew. All week long people solved problems for each other. There was a great spirit of friendship and teamwork.

Coordinating and running the 100 sound jobs—with excellent sound quality—added to the success of the Special Olympics. It happened because of the careful planning of Bill Raventos and Tom Durell, and the hard work and long hours of hundreds of volunteers.

The theme of the 1989 Special Olympics was "Uniting the World", and indeed it happened for a week in Reno/Tahoe. Seeing the athletes' profound joy as they were cheered and as they hugged their coaches made it all worthwhile.

THE CONTRIBUTORS

Much appreciated was the willingness and generosity of the following companies who donated equipment for the Games:

AEI Music Network: satellite downlink background-music systems.

Atlas/Soundolier: mic stands.

Belden: cable.

Community Light & Sound—loudspeakers.

HME: wireless mics and intercoms.

Ivie Electronics: RTA and pink-noise generator.

Micro-Innovatons: AVAIL computer software.

Figure 14. Charlie Daniels performing into a Crown CM-200 Select Series microphone.

Figure 15. The Charlie Daniels band on stage.

Figure 16 This ice show was part of the closing ceremonies.

Pro Co Sound Co.: cable.

Rane Corporation: signal processors.

Ritron: walkie-talkies.

Shuford Mills: gaffer's tape.

Starcase Co.: road cases.

Teac/Tascam: processional cassette and reel-to-reel machines.

Theos Software Corporation: computer operating system.

Tripp Lite Inc.: power strips.

Ultimate Support Systems: speaker stands and mic stands.

Yamaha (Pro Audio Division): mixing consoles, mixers, and effects.

Yamaha (Power Equipment Division): power generators.

Also appreciated were the following individuals who donated their labor and skills for the project: Larry Driskill and Glen Meyer of Cetec Ivie, Joel Lewitz of Paoletti/Lewitz, Richard Johnson of Heather Sound, Bob Ramsey, Billy Porter, John Wiggins of Community Light & Sound, Randy Opela of HME, Ken Kuespert, Crown employees and Crown sales representatives. db

Bill Porter On Recording And Mixing

BRUCE BARTLETT

• Bill Porter teaches a recording class and has many valuable ideas on audio techniques which he shared with us.

I asked Bill if he preferred recording or live mixing, He alternates in his preference, but said that, "With live mixing, you're part of the performance. It's exciting because you have only one chance to get it right." Tom Durell agreed that live mixing was exhilarating. Bill noted that when he mixed the same act every night, he liked to vary the rhythm mix to keep himself interested.

His early recordings—many were gold records—were done live-to-2-track with all the musicians in the same room! Although the era was before noise reduction, his mixes were very clean because he aligned and calibrated his tape machines every day.

To achieve isolation in the studio, he beat on a tom- tom as he walked around the room to find dead spots (nulls in the standing-wave pattern). He put X's on the floor there and put musicians in the same spots.

He prefers to use as few mics as necessary, placed 1-½ to 3 feet from each sound source (when possible) to let the full timbre develop. It's better to achieve the desired tone quality by microphone selection and placement, he says, rather than by equalization. On stage, of course, the mics must be placed very close to prevent feedback. In that case you're forced to equalize to make the timbre more natural.

Once Bill played his students a recording that compared several studio mics on a grand piano. It was a blind listening test. Although some of the mics were expensive Neumanns and AKGs, the listeners preferred Shure SM-57s! Sometimes an *inexpensive* mic sounds best.

When mixing, his attention constantly scans the inputs, almost like a computer. He listens to each instrument in turn to make sure it is audible. He also listens to the balance of the bass with the drums, the balance of that combination with the piano, the balance of the rhythm instruments with the vocal, and so on.

At live concerts, Bill often looks at the stage to see what's happening. He adjusts the faders so that he can hear it. If he sees the drummer hit a cymbal crash, or the lead guitar take a solo, they better be in the mix. Bill tries to emphasize the main frequencies of each instrument, making each one occupy a different part of the audible spectrum so that they don't step on each other.

When asked, "What makes a recording sound commercial?" Bill replied that it has to have personality. This comes from the creative ideas of the musicians and the engineer. It's not good to preplan a tune so much that you lose spontaneous input from the creative people involved.

Bill feels very strongly that many younger engineers try to impress themselves and other engineers with subtle sonic effects that only they can hear. They overdo the sonic production and get away from the music. The purpose of engineering and production is not to show off sonically, but rather to convey and enhance the musician's message.

COMMUNITY'S RS880 FULL-RANGE LOUDSPEAKER SYSTEM

• First unveiled as a prototype in January of this year, Community's new RS880 full-range loudspeaker system is poised for a full-blown market introduction, following successful field trials at the Winter Special Olympics in April and an outdoor concert featuring the Charlie Daniels Band at Delaware's Brandywine Racetrack. Complemented by Community's proprietary 880EQ and VBS415 sub-woofer system, it is the latest player in the Chester, Pennsylvania-based loudspeaker manufacturer's professionally-oriented RS Series. More specifically, it was designed by Community engineers to meet the growing needs of contractors seeking a reliable device capable of meeting the needs of touring sound, fixed installations, and other flying arrays.

Outwardly, the 3-way RS880 is housed in a rugged trapezoidal enclosure built from extensively-braced plywood covered with black carpet. Measuring 49 1/2 inches high by 30 inches across, it is 22 1/2 inches deep with a pitch of 22 degrees. Relatively light at 170 lbs., various options including Penn Fab D-rings and internal steel trussing make the enclosure highly suitable for flying arrays. Adding even further to the RS880's versatility is the availability of two high frequency coaxial horns: a 90 x 40 and a 60 x 40. For applications where it will be exposed to the elements, a laminated fiberglass exterior can be ordered, and to protect the drivers from sharp blows and punctures, Community has provided an optional high-strength, steel-reinforced foam grille.

Four individual transducers comprise the RS880's 3-way design. At the low-end, two of Community's newly-developed 15- inch "Triple Spider" cast-frame loudspeakers manage frequencies up to 450 Hz. Horn-loaded for better projection, these LF drivers are additionally outfitted with ferro-fluid cooled voice coils. Mounted above the top 15-inch driver, Community's M200 2- inch compression driver takes charge of the midband frequencies. At 3 kHz, signals are directed by the RS880's sophisticated passive crossover into a 1-inch titanium diaphragm driver, which is coaxially mounted in the bass horn section.

The entire RS880 system was developed using Techron TEF technology to provide Community's trademark Wavefront Coherent performance, which insures that each transducer is optimally located to virtually eliminate the need for signal delays or other complex processing to achieve proper phase characteristics. With an operating range of 45Hz - 18kHz, sensitivity measures 107 dB (1 watt 1 meter), and power handling is rated at 400 watts pink noise/1000 watts program. Maximum output exceeds 131 dB, and impedance is 8 ohms nominal and 4 ohms minimum.

A dedicated dynamic equalization device for the RS880, the 880EQ is outfitted with feedback loop sensing to maximize dynamic range. Incorporating balanced inputs and outputs as well as ground lift switches, the 880EQ progressively reduces equalization near peak operating conditions to maximize level capabilities and enhance long-term average power handling of the RS880 loudspeaker system. It also has outputs for the VBS415 sub-woofer, which is a vented device housing four specially- developed cast frame, ferro-fluid cooled, long excursion 15-inch cone drivers that cover the frequency spectrum between 25 - 50 Hz. When used in conjunction with an RS880, the VBS415 receives its own amplification, making the complete system bi-amped. While it has been suggested that the RS880 could be tri-amped, Community strongly advises against it, as the precise nature of the passive crossover's phase characteristics would be adversely effected.

One of many who worked with Crown International to provide quality audio for the Fourth International Winter Special Olympic Games (IWSOG) held in the Reno/Lake Tahoe area in early April, Community v.p. of sales and marketing John T. Wiggins utilized the RS880 in a host of venues ranging from rain-soaked outdoor pavilions to the cavernous Reno Livestock Events Center, which is normally used for rodeos. For a '60s revival concert (which was billed as a "Beach Fest" and featured 1400 tons of sand dumped onto a parking lot to create an ersatz beach), the RS880s were stacked five-to-a-side on the main stage and were powered by five of Crown's Macro-Tech amplifiers. With Bobby Vee, Tommy Rowe, Del Shannon, and a host of other entertainers crossing the stage during the all-day event, the legendary Bill Porter handled mixing chores.

Following the Beach Fest concert, the same RS880 loudspeakers were hung on a truss in an arc over the main stage of the Reno Livestock Events Center for the Charlie Daniels Band concert. Although skeptical at first about the loudspeakers' ability to handle the SPLs required to fill the vast structure in an intelligible, low-distortion fashion, the RS880s performed so admirably that Daniels' sound technicians requested that they be used for their Brandywine Racetrack performance held about a month later just north of Wilmington, Delaware.

Editorial

In our continuing efforts to make **db Magazine** better for its readers, we have already re-designed our covers to make them more informational about what's in each issue. Now we have also addressed the pages themselves.

But not with re-design. Rather, a new type style is being used beginning with this issue. We have extensively tested it against what went before with each pair of eyes agreeing that the new type is clearer and easier to read. What's your opinion?

However, the medium is not the message. The message in this issue is **Sound Reinforcement**, particularly that element that is used in live sound. Careful planning indeed must be used when a reinforcement system is being set up—probably for only one use.

In our lead story on the *1989 Winter Special Olympics*, author Bruce Barlett does indeed detail the planning, and the changes of plans nature sometimes forces, that go into such an endeavor. It must also be understood that all the equipment used in Reno was donated, and with the budgets for men and materials also coming from the respective companies. All in all it was a fully professional and most worthy endeavor for an equally worthy cause.

Ed Learned, our peripatetic sound reinforcement contributor has also written a detailed interview of Jim Risgin of Maryland Sound—one of the major forces operating in live sound reinforcement today.

Sometimes a live-sound engineer finds himself with an artist or group of artists that do not conform to what is considered normal. Such is the case when an engineer for the Grateful Dead found himself responsible for a tour by a group of Tibeten monks (living in India) who had no previous experience (and little present concern) for the problems of providing sound for an audience. The tour of the *Gyuto Monks* was a stunning success. Read why in this issue.

When the British group *UB40* toured Israel recently it was fortunate that our resident sound man Ron Altman was nearby to get us pictures and text on what the group brought with them as well as what they were able to find and use in the country.

There's more of course. In *Computers, Humidity and Sound* author Wiliam Graham gives us a computer program for sound reinforcement sessions that must take into account the relative humidity of the venue.

Live music broadcasting is the subject covered by our new Senior Editor John Barilla in his interview with *Phil Antonucci*, who produces the music scores for Madison Square Garden sports events.

All our columnists are here as well, of course. That's *Bartlett, Barilla, Battles, Gately*, and *Hoffner*. All this editorial has also meant not everything planned was able to get into this issue. Most prominent is the omission of **Tape Recorders** from our **Buying Guides** for this issue. Tape Recorders will be in our September/October issue along with scheduled **Signal Processing Equipment**. And in that September/October issue, watch for a new columm that we're sure will be of considerable interest to all of you. L.Z.

An Interview with Jim Risgin of Maryland Sound

Successful sound companies stay that way by continually procuring top talent. In a short span of time, Jim Risgin has made the transition from local sound jobs to one of the country's largest sound companies, Maryland Sound Industries.

In this interview with engineer Ed Learned, Jim traces his career and discusses his two most recent MSI clients, Miami Sound Machine and Steve Miller.

BACK TO THE BEGINNING

db—How did you get interested in sound reinforcement?

JR—I got hooked into live productions doing lighting for theater and theater production in high school. I went in from there to start doing sound in bars for local bands. I really enjoyed doing it, and started to get a lot of jobs. It became a sort of natural progression to do it for a living.

db—The first time we met was at Lansing Sound Studios. At what point was that in your audio career?

JR—That was really at the beginning of my professional career, around 1982. It was a good place to be at the time. It was a small studio and I had free run of the place. I could try out different things and experiment with different ideas. It was a great experience, even though it didn't pay worth beans. I was able to find out what worked and what didn't work.

db—We later worked together with Ashford and Simpson, in the summer of 1985. I mixed front-of-house, you mixed monitors. That was your first tour, wasn't it?

JR—Certainly was.

db—I remember that I had to keep reminding myself that it was your first tour because your composure was excellent. I knew you had the mental elements required to deal with doing monitors at the major league level. You know all about that now (both laugh)!

JR—That's the thing about doing monitors; it's the most mental gig out there. If you're mixing front-of-house, you're mixing for yourself. Monitor mixing is not a question of doing what you want, it's doing what the artist wants. You have to put yourself in their shoes—or, a lot of times, five different people's shoes. It's really a high-pressure, big strain gig.

db—Before, during and after Ashford and Simpson, you were working for Sounds Good Audio of Lansing, MI. How did you hook up with Maryland Sound?

JR—MSI subcontracted Sounds Good to do two weeks with Frankie Valli. Usually, MSI would send a monitor man out with Frankie, but it couldn't be arranged. As SGA's monitor guy, I went out and did monitors. Frankie liked me and wanted me to come out and do the whole tour. I could only do that working for MSI, so, basically, I switched companies.

db—Frankie Valli became your first account for MSI. What was he like to work for?

JR—Frankie is a perfectionist. He knows what he wants. If you can give him what he wants, he loves you.

db—Obviously, you were able to do that.

JR—Yes...And if you can't, it's see you later, bye (both laugh). He's a proving ground for a lot of our engineers. Frankie always gets the best from Maryland Sound. He gets our top engineers and will continue to because of his relationship with the owner. Once you've proved yourself with Frankie, you've done well in the company's eyes.

db—How long did you work with Frankie?

JR—I was with him from June, 1987 to June, 1988. At that point, I joined Miami Sound Machine on the *Let It Loose* tour.

db—When you prepare for a tour, what are some of your major concerns?

JR—It depends if you join the tour in progress or start at the beginning with rehearsals and the whole nine yards. With Miami, I jumped in on the middle of the tour.

Big Ed Learned (as he is known around these offices) is a frequent contributor to these pages on sound reinforcement subjects, paticularly writing about his travels around the world with the USIA.

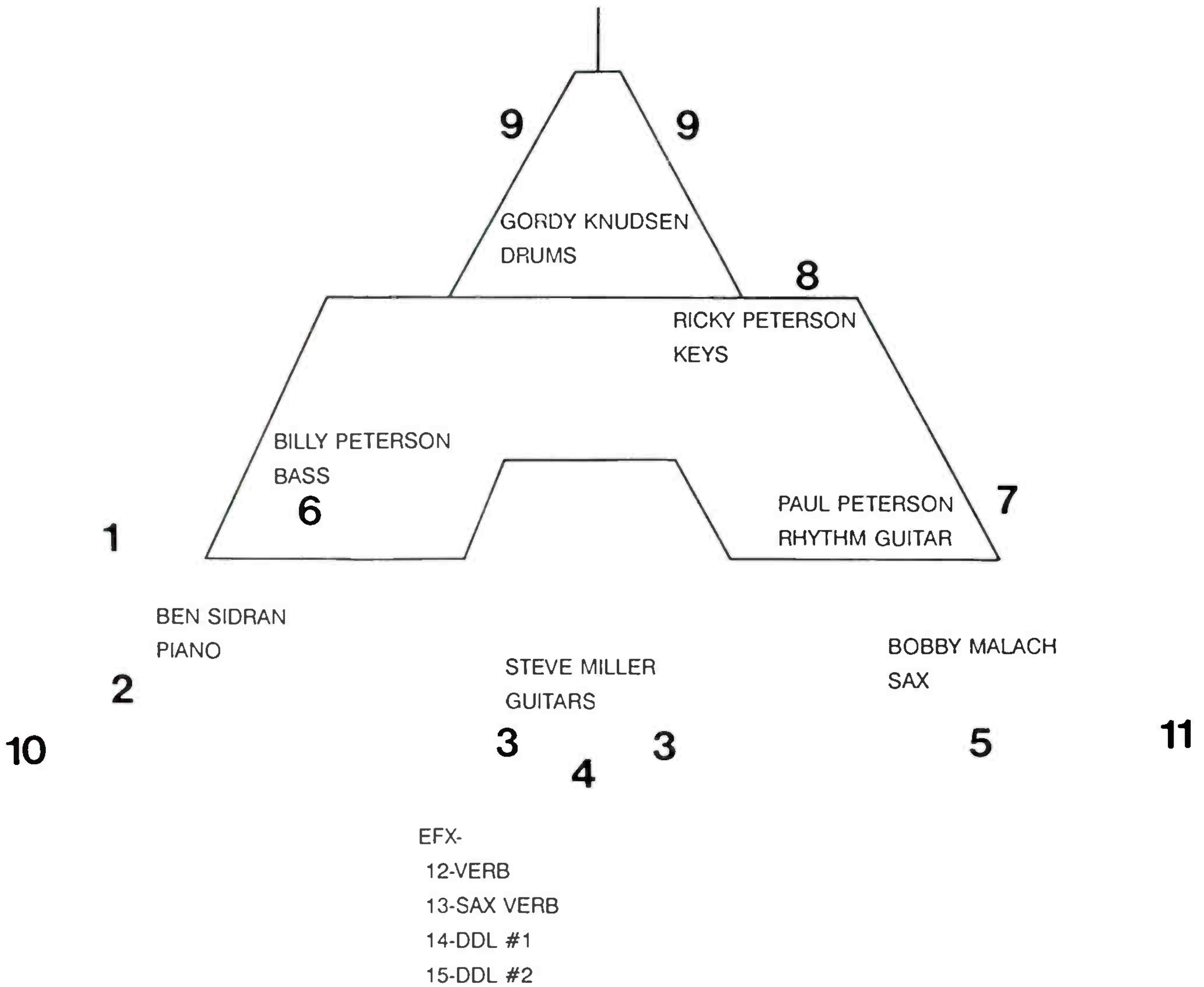


Figure 1. The layout for the Steve Miller Born To Be Blue tour 1988.

db—What was that like?

JR—It was a pain. I was going into a situation where I was changing from a 32x10 to a 44x18 situation on the day I arrived.

db—Was this increase at the request of the group?

JR—I had done two one-offs with them before, and I sort of knew how their show went. I had feelings of what their engineer was doing wrong. The increase in size was my own initiative.

PROBLEMS, PROBLEMS

db—By increasing the size of the monitor kit, you were attempting to address specific problems. What exactly did you perceive and how did you attack these problems?

JR—With Miami Sound Machine being such a physically active band, with people constantly changing position and lots of choreography, I had to regionalize a lot of the mixes. For instance, the previous engineer used stereo slide fills and one mix downstage, with four wedges on it. I increased this to three mixes down stage. That way Gloria could still have her vocal emphasized where she was, and yet when the other members of the band came downstage, I could still highlight them when they were away from their areas.

db—How many mixes did you use for Miami?

JR—It ended up being 14 mixes, a headphone mix and three effects mixes.

db—Could you briefly describe each of these mixes and what you were using for hardware on each?

JR—There were the downstage mixes: stage left, stage center and stage right. Stage center was two wedges, left and right were either one or two wedges, depending on if I had flying side fills. I used stereo side fills, flown over half the time, although there were stupid gigs where I couldn't fly anything.

db—How about the upstage area?

JR—I used percussion, keyboards stage right, drums, keyboards stage left, guitar, bass, trumpet, trombone, upstage side fills and a headphone mix for the drummer.

db—You mentioned three effects sends. What devices were you using and to which artist were they assigned?

JR—I used a Yamaha SPX-90 for reverb on Gloria, another SPX for reverb on percussion and a Lexicon PCM-60 for reverb on drums.

db—I noticed that you had both downstage and upstage side fills. Were they both using the same type of cabinet?

JR—Actually, they weren't. Downstage was an MSI 3-way box, what we call an MA-3. It had 2 TAD 1601 15" woofers, 2 JBL 10" woofers and a TAD 4001 2" horn driver. I had two of these per side. Upstage we had this ramp setup I needed to cover, so I flew a pair of Meyer UPA cabinets, one per side, from the lighting truss. It's a nice, compact, beautiful little box that's great for things like that.

db—What did the downstage side fill mix have in it?

JR—Mostly vocals. Whenever a band member would come downstage, I would bump up that instrument in either the left or right downstage side fill, whichever was closer. The last six feet of the stage were Gloria's. She just wanted to hear herself and a little bit of the drum machine. It was basically heavy vocals downstage, with the rest of the stuff upstage.

db—As she moved around, or as others came downstage, what sort of differences evolved in the downstage wedges?

JR—All three mixes were heavy on the vocals. As far as the band, everybody was running around on stage. Bass, guitar, both keyboard players had portable KX-5s and the horn players were wireless. So if someone wandered downstage, I would "follow" them wherever they went, turning them up in the nearest monitor.

db—You really had to keep your eyes open.

JR— I was mixing for 10 people and was constantly busy. I could have used eyes in the back of my head and an extra pair of hands.

db—Well, you must have done fine because you still have the gig.

JR—I still have the gig and I'm trying to move out and do front-of-house (both laugh). But I'll be happy to do monitors for them if that's what they want. I have no problem with that.

db—What were the upstage mixes like?

JR—Most of those were very drum heavy. The drum machine was the click track and everybody followed that, including the drummer. It was controlled by the keyboard player, so it had to be heavy in all the back mixes.

db—What were you using as a drum monitor?

JR—I was using a 3 Meyer UPA and a 1 MSI MS-10B cabinet. The MS-10B is two TAD 15" woofers in a ported box. I also had the headphone mix.

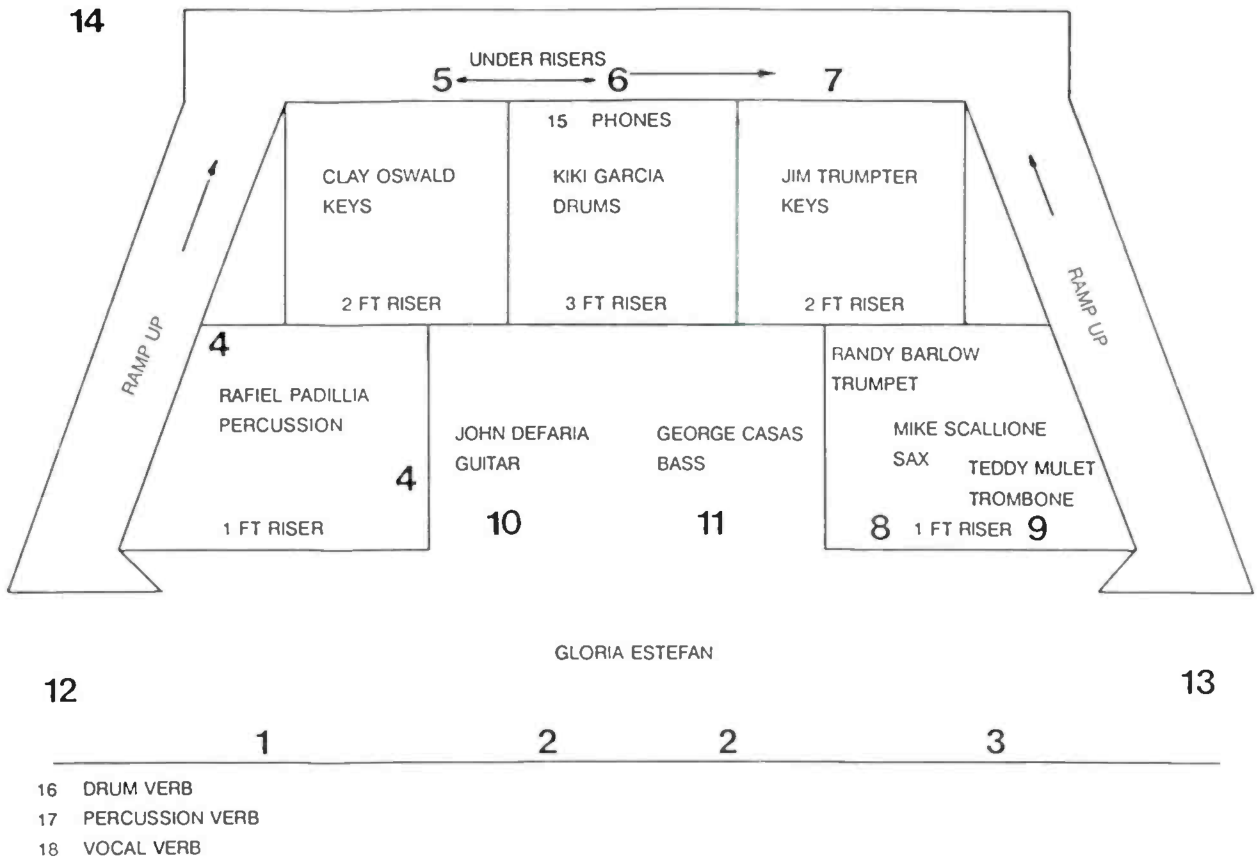


Figure 2. The layout for Gloria Estafan.

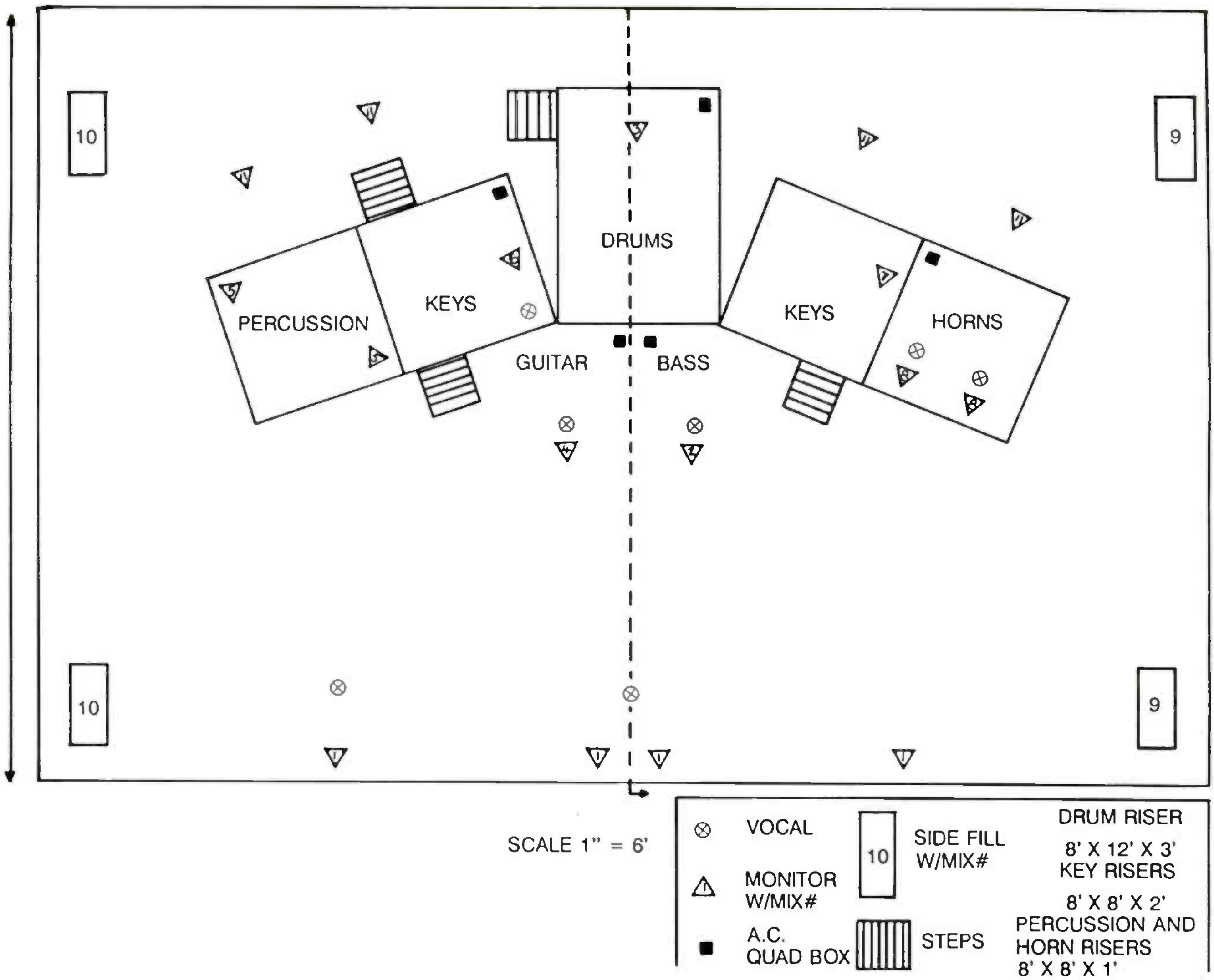


Figure 3. Stage layout for the Miami Sound Machine.

db—What about your wedges? Were you using the same type of cabinet everywhere?

JR—They were identical. I used the MSI 2x2 wedge, which contains two JBL 2202 woofers and a JBL 2445 2" driver.

db—What do you particularly like about that cabinet?

JR—It's a nice cabinet, but what really makes it great is our digital crossover system, what we call a digital switcher. We have dedicated monitor racks with a 16-pair multipin input. Each rack will power up to eight cabinets, with an SAE P50 on highs and a Crest 3501 on lows. Each wedge has its own crossover and a dedicated switcher card, which is a digital 1-16 select, so you can dial up any wedge on any mix, with no patching involved. It's an MSI standard feature.

db—You can basically place your wedges anywhere on stage that you want and decide what mix each is on by dialing it up. That's very hip.

JR—It's way hip. Another thing about the switcher racks is the fact that there's built-in time alignment and EQ on each crossover. We even have dbx-type limiting incorporated for protection. It's a very flat, high fidelity, high SPL system. We also have another type of wedge, a 2x15, which has two TAD 1601 or 1603 15" woofers, depending on use, and a TAD 2" driver. They can be used on the same switcher.

db—Do you have to throw some kind of switch to change from 2x12 to 2x15?

JR—No. It's labeled 2x12 or 2x15 on the output.

db—You mentioned that the 2x15 wedges had two different types of 15" speakers. Can you give us some insight on the differences and where each type might be used?

JR—The 1601 is a high-power 15", with heavier paper, and delivers more low end. The other one is similar, only maybe it doesn't have quite as much bass. I use the 1601 for bass players, stuff like that. We have a new switcher rack for higher-power wedges, called a Hi-Po switcher. Right now, that has a Crest 3501 for highs and a Crest 7001 for lows. We also have a high power 2x12 wedge, with JBL 2404 woofers instead of 2400s. The 2404 is like 6 dB less efficient, but has a tighter low end. So there are a lot of differences between our wedges if you get right down to it, but it's the basic crossover-switcher concept that makes it so cool. It makes doing monitors easy.

db—That's good because it sounds like you have enough things to worry about.

JR—You got that right (both laugh). It lets you forget the technicalities and concentrate on mixing.

I'll also be using flown side fills this time—I didn't do that last time. I'll use a different 3-way cabinet, with 12" woofers replacing the 10" ones, so I can get a little more meat on the guitar and vocals.

ENGINEERING FOR STEVE MILLER

db—You're currently working as monitor engineer for Steve Miller. How many mixes are you running for him?

JR—We're off the road right now but we'll be going back out May 30. Before, I ran 12 mixes. This time I'll do 13 or 14 because we'll have Norton Buffalo with us on this leg of the tour.

db—Super bad harp player. You'll hear some great harp playing out of him.

JR—I just got the stage plot. I haven't really finalized anything yet. I know I'll be using 2x15 wedges more with this show than with Miami.

db—Why the difference in approach?

JR—First off, there wasn't any room om Miami's stage for a box of that size. The double 15 is an unbelievably smooth sounding box. The clarity in the midrange is great and it has more warmth and definition at higher volumes than the double 12 box does. Don't ask me why (laughs). I'll use these mostly for vocals. Steve, who is very picky about how his voice sounds, will respond better to this type of sound. I have better control with the double 15.

db—So you'll use a pair of these downstage for Steve?

JR—I think so but to be honest, I'm not sure what I'm going to do. I think I'm going to run those two wedges in stereo because Steve was talking about wanting more of an image in the band vocals in his wedges.

db—So you can pan the vocals around in his wedges to achieve that.

JR—Exactly. I think I'll also put a Meyer UM (Ultra-Monitor) in the center between the two double 15s just for his guitar. It's a very narrow dispersion, very directional box, so I can control stage volume with it.

STEVE MILLER BORN TO BE BLUE TOUR 1988 (FALL)

MIX#	POSITION	COMPONENT
1	SR PIANO	1-MEYER UM
	BEN SIDRAN	
2	SR KEYS	
	BEN SIDRAN	1-MSI 2X12
3	STEVE MILLER	2-MSI 2X15
4	" " GUITAR	1-MEYER UM
5	BOBBY MALACH	
	SAX SL 1-MSI 2X12	
6	BILLY PETERSON	
	BASS SR	1-MSI 2X12
7	PAUL PETERSON	
	GUITAR SL	1-MSI 2X12
8	RICKY PETERSON	
	KEYS SL	1-MSI 2X12
9	GORDY KNUDSEN	2-MSI HIGHPOWER
	DRUMS 2X12	
10	FILLS SR	1-MSI MS 3W
11	FILLS SL	1- " "
12	VERB—LEX 200	MSI MS 3W
13	SAX VERB—SRV 3000	2-JBL 15'S
14	DDL #1—REV 7	2-JBL 12'S
15	DDL #2 —SPX 90	1-TAD 4001 2"

Steve does like to hear his guitar quite loud.

db—That's real smart. With the guitar right in his face, he'll be happy without blasting guitar all over the stage, causing all sorts of bleed problems.

JR—I'll also be using flown side fills this time—I didn't do that last time. I'll use a different 3-way cabinet, with 12" woofers replacing the 10" ones, so I can get a little more meat on the guitar and vocals.

db—What will you put in the side fills?

JR—Mostly just Steve's vocals and guitar. Sometimes I might add a bit of drums and/or piano, depending on Steve's moods. His opinion of what he needs changes quite a bit.

db—We pretty much have the downstage stuff covered. How about the rest of the band?

JR—We have Ben Sidran, Steve's lifelong buddy and pal, on keyboards. Ben has done a lot of jazz work, so he's a different player to have in a rock band. His touch on the piano is very light, and he has different techniques. Because his touch is so light, I have to do some different things. I actually gave him two mixes—a 2x12 for the band mix and a Meyer UM just for the piano. The UM has the clarity and directionality I need on the piano. The band mix has pretty much everything else in it, along with his DX-7.

db—Do you have both these wedges together, or do you spread them?

JR—I spread them, with one on either side. I talked him into doing this so I can get better separation between the piano and everything else. Say I put the band wedge on his upstage side, the piano wedge on the downstage side. If he ever needs more piano, he knows where to look for it, rather than having to pick it out of a full band mix. I can also run the piano at a lower level this way.

One of the things we used to talk about was the level of drum monitors and how they could affect drum sound.

db—Let's continue on with the other mixes.

JR—Everybody else just has double 12 wedges. The drummer gets a pair of Hi-Po double 12 wedges, the one with the 2404s. Other than that it's a pretty standard MSI monitor rig.

db—One of the things we used to talk about was the level of drum monitors and how they could affect drum sound. I notice that your approach takes this into account since you are into smaller drum monitors. You're not into the pile up walls of stuff and knock the drummer off his stool philosophy.

JR—The problem I have with a huge monitor is that it does blow into every microphone out there. But more importantly, it doesn't help create the space of the drum kit. If someone wants a lot of low end, I'll give them a subwoofer behind them, but I really want a point source, a wedge, on either side of the drummer. If the cabinet is just on one side, it's not natural; it becomes an amplification of the drums. By surrounding the drummer, you bring the sound of the drums back. The wedges become part of actual drum sound.

db—The net result is that you have a workable volume and the drummer "hears" more naturally.

JR—If I'm not looking for it to be point source from the monitor; to me, that's a real disturbing thing. When you hit the drums, you hear them all around you rather than from one particular spot. That's my philosophy in monitors: I create a stage sound. I don't create an isolated environment. If you have a guitar and bass player right next to each other, and the bass player wants something in his wedge, you consider how it will affect the guitar player next to him. The guitar player will hear part of what he might want from the nearby bass wedge.

db—Well, let's be real—that is what's actually happening on stage. They will hear each other's wedge.

When you get that unified sound, the band picks up on it, the house picks up on it.

MONITORS AND MUSICIANS

JR—Monitor engineers for the longest time have done just what the musician asked for. They created seven or eight or nine different worlds on stage. But it's not nine headphone mixes out there; everything interacts. You have to create a unified stage sound. If you can't create a unified stage sound, how is the band supposed to create a unified stage sound?

db—The answer is they don't.

JR—If you can do it, they can, and the whole show's happening. Mixing monitors isn't turn it on and leave it anymore. I'm mixing—I'm doing stuff before the artist even asks for it. Songs change and you have to be sensitive to style changes. Steve Miller has gone from being guitar army rock and roll to doing jazz covers and he does them all in one show. You have to change the sound as the song dictates or as the band wants. I plan out a lot of this stuff with the band, but a lot of it is unspoken. For instance, during sound check the artist will start asking for a lot of things because the house is empty and acoustics are different. You have to tune in on the show.

db—That could, of course, cause problems with famous artists who are used to getting their every wish granted. How do you smooth those ruffled feathers?

JR—Steve used to ask me for stuff during sound check and I wouldn't give it to him. I'd say "let's just wait. If it doesn't sound right during the show, I'll give it to you. But you know it's going to change." So we worked out a trust between each other. I sort of knew what he wanted, and he knew what I wanted. The same thing with Gloria. If the artist doesn't trust

STEVE MILLER BORN TO BE BLUE TOUR SUMMER 1989

MIX #	LOCATION	COMPONENT
1	BEN SIRAN KEYS SR	1-2X12
2	BILLY PETERSON BASS SR	1-2X12
3	STEVE MILLER L	1-2X15
4	" " R	1-2X15
5	" " CENTER	MEYER UM
6	NORTON BUFFALO HARP	1-2X12
7	NORTON US KEYS	1-2X12
8	KEITH ALLEN GUITAR SL	1-2X12
9	BOBBY MALACH SAX SL	1-2X12
10	GORDY KNUDSEN DRUMS USC	2-HENLEY MON (1-TAD 1603 15") 1-TAD 4001 2"
11	FILLS SR	2 MEYER MSL 3 FLOWN
12	" SL	2 MEYER MSL 3 FLOWN
13	VERB	SPX 90
14	SAX VERB	SRV 3000
15	DDL	SUPER PRIME TIME
16	CHORUS	SPX 90

the monitor engineer, he's not going to feel comfortable on stage.

db—That's very true. It gets to a point where a certain look can tell you exactly what they need.

JR—Not even a look. You can see they're moving differently, and you know they're not comfortable out there. You really have to pay attention to what's going on. You have to become one with the artist.

db—How important is the monitor system to overall house sound?

JR—Monitors are the whole bottom line. I have a friend who does Neil Young's monitors and he showed me. You create the stage volume. When you get that unified sound, the band picks up on it, the house picks up on it. You create the sound of what's going on. I didn't really realize it until I did house for Frankie Valli. It was an enlightening experience doing house for a band I'd done before as a monitor guy, and working with a monitor guy I really respected. You learn the interaction between house and monitors.

db—As you move out front, what do you see as the major difference between the two, and how should both engineers work together?

JR—I see that I preached to every house guy I worked with (both laugh). The house guys these days don't mix with the stage. We're in the sound reinforcement business, not the sound replacement business. Turn off the PA and listen to what the band wants. Listen to how the band hears. Listen to what the band sounds like coming off that stage. Then, turn on the PA, add to that original sound, create what the artist wants. You don't create what you want to hear, but what the sound is.

HOUSE FIGHTING MONITORS

db—Could you give a specific example of house fighting monitors?

JR—You have a snare drum on stage that has this tight, compact, cutting kind of sound, and the house puts this huge gated reverb on it that makes it gush all over the room. It messes up the stage sound because the artist doesn't want to hear that, and there's no way a monitor system can keep up with a house system. It ruins the vibe. You have to know what the artist wants.

The first ten mixes are single knobs, the next eight mixes are on four dual concentric pots.

db—From my personal experience, I would always get together with the band beforehand to discuss effects, voicing, etc., and not take a lot of liberties of my own. Do you think that happens today?

JR—I think it's the most important thing today. With monitors being mixed the way they are, the artist has more control. If these ideas are not being carried out in the house...It's like the house says, "I can make it sound like anything," instead of, "I can make it sound like this group."

db—Or sound like the glamour sound of the day.

JR—Glamour sound doesn't mean anything. When I mixed sound for Frankie Valli, I was using two reverbs and a delay. You just don't need

HOUSE	MONITOR	INSTRUMENT	MIC/DI	TYPE OF STAND
1	1	Lead Vocal	431	Straight
2	2	Lead Wireless	431	Straight
3	3	Gtr Vocal	M88/58	Boom
4	4	Bass Vocal	M88/58	Boom
5	5	Keys Vocal SR	M88/58	Boom
6	6	Keys Vocal DS	M88/58	Boom
7	7	Trumpet Vocal	M88/58	Straight
8	8	Trombone Vocal	M88/58	Straight
9	9	Key Mix L SR	DI	
10	10	Key Mix R SR	DI	
11	11	Emulator II SR	DI	
12	12	Emulator II SL	DI	
13	13	Key Mix L SL	DI	
14	14	Key Mix R SL	DI	
15	15	Gtr L	57/M88	Boom
16	16	Gtr R	57/M88	Boom
17	17	Acoustic Gtr SR	DI	
18	18	Acoustic Gtr SL	DI	
19	19	Bass	DI	
20	20	Kick	441/421	Boom
21	21	Snare	56	
22	22	Hat	451	
23	23	Elec. Toms L	DI	
24	24	Electronic R	DI	
25	—	OH L	460	
26	—	OH R	460	
27	25	SP 12	DI	
28	26	Hi Conga SR	421/M88	Boom
29	27	Lo/Mid Congas SR	2-421 "Y"	Boom
30	28	Bongos SR	441	Boom
31	—	Timbales SR	421	Boom
32	29	Woodblocks SR	KM84/452	Boom
33	30	Toys SR	KM84/452	Boom
34	31	Trumpet	DI	
35	32	Sax	421	Boom
36	33	Trombone	DI	
37		SDE 3000 RET		
38		949 RET		
39		SPX90 RET		
40		REV 7 L RET		
41		REV 7 R RET		
42		Lexicon 200 L RET		
43		Lexicon 200 R RET		

MIAMI SOUND MACINE INPUT LIST

dealing with stereo mixes and effects. It's a clean console, well laid out, and it has a great grounding scheme and is extremely quiet.

db—What about the EQ?

JR—It's a 4-band peaking deal, but it doesn't have a separate bandwidth control. It's really a quasi-parametric: the more you boost or cut, the narrower the bandwidth gets. It really sounds sweet. I like it better than a Yamaha PM-3000 EQ.

PROGRAMMABLE MUTES

db—How do you feel about programmable mutes on a monitor console?

JR—Some people love 'em. For me, no. If you start turning vocal mics off, the bleed that was coming through those mics goes away too, so level and tone changes in every monitor those mics are assigned to. I don't like that. You shouldn't have to do that. If you pick the right mic/monitor combination, you should be OK. Same thing with noisegates. If I use one, I set it so it's only about a 3-6 dB

all of this stuff—well, sometimes you do—but there are people out there who don't know how to use the technology. They just have to have it because it's trendy.

db—What people forget about effects is the definition. It's an enhancement, not a be-all-and-end-all reason for sound being reproduced.

JR—Standing on stage, you really learn when someone is doing that, when they are trying to replace instead of enhance. It screws up the stage sound—the artist looks at me and says, "Why does my voice sound so thick? Why is it distant?" I've got four 12s and two horns against his 50,000 watt PA. What can I do about it? We have to work together. It's unfortunate, but sometimes I run into people like that. I've also had the good fortune of working with great people—present company included.

db—What is your preference in monitor consoles right now?

JR—The Ramsa S840 is my current favorite—that's a 40x18 console. The first ten mixes are single knobs, the next eight mixes are on four dual concentric pots. These are internally selectable, so there can be eight discreet mixes or four stereo mixes, which is nice when you start

GLORIA ESTEFAN AND THE MIAMI SOUND MACHINE

MIX #	PERSON	COMPONENTS
1	DSR GLORIA	1-MSI 2X12
2	DSC GLORIA	2-MSI 2X12
3	DSL GLORIA	1-MSI 2X12
4	SR PERCUSSION	
	RAFIEL PADILLIA	2-MSI 2X12
5	SR KEYS	
	CLAY OSWALD	1-MSI 2X12
6	KIKI GARCIA	2-MEYER UPA AND
	DRUMS	1-MSI MS10B 2X15 SUB
7	SL KEYS	
	JIM TRUMPTER	1-MSI 2X12
8	HORNS SL	
9	RANDY BARLOW	
	TRUMPET	1-MSI 2X12
	MIKE SCALLIONE	
	SAX PEC MIX	
	TEDDY MULET	
	TROMBONE	
10	GUITAR SR	
	JOHN DEFARIA	1-MSI 2X12
11	BASS SL	
	GEORGE CASAS	1-MSI 2X12
12	FILLS SR	2-MSI MA3 CABINET
		-2 TAD 1603 15"
		-2 JBL 10"
13	FILLS SL	2-MSI MA3
14	US FILLS	2-UPA'S FLOWN ON
	LIGHTING	
15	DRUM HEADPHONE MIX MONO	
16	DRUM VERB	PCM 60
17	PERC VERB	SPX 90
18	VOCAL VERB	SPX 90

gain reduction, just enough to remove the overtones. I don't like a gate that turns the mic on and off.

db—Do you use limiters at all?

JR—I don't really use them a lot on monitors because it affects the way the artist uses the microphone. In the house, it's another story. But for monitors they each have their own personal mic technique...

db—They are their own limiter.

JR—Right. If you start using a limiter on top of them, it can really start to mess with the way they're hearing things. I do have compressors in the crossovers, but those are strictly for protection. The artist is used to hearing the volume increase and decrease consistently with his performance. I want to leave that alone. db

LARRY OPPENHEIMER

Sound Reinforcement for the Gyuto Monks

Providing high-quality sound reinforcement for a blasting hard core band in an arena is certainly a different task than reinforcing a solo cellist in Carnegie Hall. Nonetheless, both of these situations are relatively well understood and the problems they present well defined. What happens when the need arises to reinforce a sound source so totally foreign (literally and figuratively) in nature that there is no precedent to draw from? This was the challenge presented to Grateful Dead sound engineer Dan Healy and Ultra Sound engineers/owners Don Pearson and Howard Danchik when Grateful Dead percussionist Mickey Hart convinced a group of Tibetan monks with an unearthly chanting style to do a "concert tour" of the U.S. late last year.

In recent years, Hart has become almost as well known for his interest in eclectic world musics as for his role in the Dead. Hart has masterminded and expedited many remarkable recordings and performances by the likes of Babatunde Olatunji and the Diga Rhythm Band, and is currently assembling a series of recordings of world music for the Rykodisc CD label, as well as overseeing for the Smithsonian Institution the transfer to CD of the entire catalog of recordings from the old Folkways label.

Since 1968, Hart had listened to a tape of the prayer chants of the Gyuto monks to "come down" from Grateful Dead performances, without any idea of who or what he was listening to. In 1985, Hart found himself experiencing the monks' chants first hand at a ritual they performed in Amherst, Massachusetts.

THE GYUTO MONKS

The Gyuto monks are a Buddhist sect which lived in the Lhasa region for over 500 years, until 1959, when brutal Chinese suppression of a Tibetan uprising forced a small group of survivors (less than a hundred) to relocate to the north of India, where they remain today. Their chanting bears little resemblance to the better known prayer chants of European Gregorian monks. What distinguishes it is the ability of the Gyuto monks to sing multiple tones simultaneously. A single monk is capable of singing two and sometimes three notes at once in a chordal structure, with the fundamental pitch being below the traditional bass range—in the vicinity of 70 Hz.

I flashed that if it was possible to use the more modern conventional technology of headset microphones, maybe we could put all that together and amplify it in a form that represented the aesthetic, religious, communicative portions of their performance...

The monks' chants fascinated Hart, who took them into a modern recording studio and, with Healy engineering, produced an album of their prayer. In November 1988, twenty-one of the now approximately 200 monks flew to San Francisco and did a little more than a dozen "performances" of a prayer ceremony: seven on the West Coast, six on the East Coast, and a final performance in the San Rafael, where the Dead are based. The nature of the monks' singing and the fact that they were conducting a ritual and not performing for an audience threw a mean curveball at Healy, Pearson and Danchik. From a technical stand point, only one substantial problem arose in doing sound reinforcement for them, but it was so extreme that it presented a significant challenge.

"For years, Mickey and I have been intrigued with ethnic music, and we've encountered a variety of different situations (for recording or reinforcement).In this particular one, because the singing register is so extremely low, the sound pressure level coming out of each and every monk is incredibly low, at least in comparison to what we know as rock and roll standards, electric music. Collectively, they barely make it to a level where you can even address using a sound system. We're talking about levels below the level of a thousand people being polite in the audience, which is as quiet as they can be: they're still breathing and sneezing and coughing. If you take an SPL meter, 70 dB is really remarkably low. To put it in street perspective, if you're walking down the street, cars going by would be in the neighborhood of 20 dB louder than what you could expect from a monk's voice. You have to understand that it's very low level.

MIC'ING THE MONKS

"A year and a half ago we did an album with the monks, and we played around in the studio with a lot of different mic'ing techniques. We found out that, even in the studio, the ambience was immensely delicate. We tried to conventionally close-mic each and every monk's voice. It worked, but it wasn't real smooth. So in the studio we wound up using more collective mic'ing and we chose a real good room: Studio A at Fantasy Studios (in Berkeley, CA). We played around with all the rooms there, but the object wasn't so much to make a value judgment on the rooms. It was

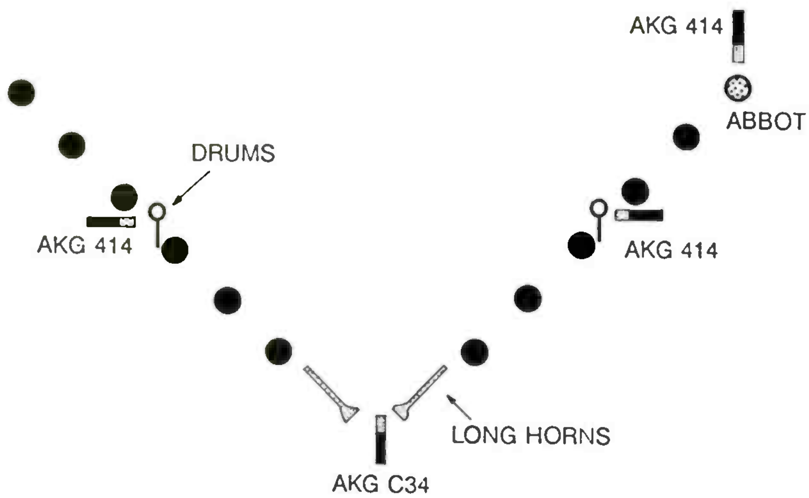


Figure 1. The microphones and their placements.

more a matter of finding a place that had the physical space to set everybody up and also doesn't trash the sound out too much; it captures what you need. The concept of digitally capturing some thing with an extremely subtle dynamic range without running into signal-to-noise ratio as a prohibitive factor was something that could only happen as a product of the times.

"To be able to create a situation where you can amplify it for an audience...we searched and searched and went through a lot of changes. I flashed that the headset microphone concept might be worth a try, but the problem with that was that we weren't sure if the monks could adapt to the concept of putting something on, and then be able to poke it up in front of their mouths and get the sound. Philosophically, when they're out there performing, they're not really oriented towards a basis of technological needs. They're manifesting their art and their religion, joy, and love. They have never really had an opportunity to be able to interface with what we know as the real world in terms of sound and reproduction.

GETTING THE REAL SOUND

"We did some shows with them a year and a half ago when the album came out, and we tried to get the monks to grok the concept of singing into microphones, but they have a certain amount of motion (when they sing), and the SPL is so small that even moving away from your microphone a little bit means that you're lost, especially in terms of real-time performance. I flashed that if it was possible to use the more

modern conventional technology of headset microphones, maybe we could put all that together and amplify it in a form that represented the aesthetic, religious, communicative portions of their performance, and, at the same time, be able to pull that off mechanically for large audiences. They could sit on the stage without any sound system and chant and, yes, the audience would hear it and there would be something. But when we hung out with the monks and had the experience of being able to walk in amongst them and hear the abundance and richness of the overtones and harmonics, the first flash Mickey and I had was, 'How can we make this possible for the audience?' We played around with a lot of different techniques; we tried every different headset that was available. The AKGs had the right price, the right groove, and they worked." After hearing a description of the project and the problem, AKG agreed to sell Healy the 25 C410 headset microphones needed at factory cost. "Several other manufacturers offered us the same deal, but I chose the AKGs, I'll tell you frankly, because they worked the best." The monks used the C410s for all 27 performances on the tour and commented that, after some awkwardness during the first few concerts, they became quite accustomed to wearing them and did not find them distracting.

In addition to the headsets, spot microphones were used to pick up the instruments that the monks played at various points in the performance: an AKG C34 for the pair of long trumpets played by the monks in front, and Neumann U87s for the drums (which were mounted on poles approximately four feet high and played with strikers that resembled an oversize shepherds staff) and the abbott, who played ceremonial bells and other small instruments. According to Healy, the placement of the spot mics was carefully plotted to maintain phase coherency in the overall onstage pickup.

In New York, a ritual was performed at St. John the Divine, a famous Manhattan cathedral.

On the West Coast dates, the sound system was provided by Ultra Sound, the San Rafael-based sound company that provides the Grateful Dead's unparalleled reinforcement. For the Gyuto monks, Ultra Sound provided a Yamaha PM180032-input console (usually used by Grateful Dead technician Bob Bralove to do an onstage submix of Mickey Hart's instruments), Meyer Sound CP10 parametric equalizers, Crest 4000 amplifiers, and Meyer Sound MSL3, UPA1$_a$, and 650R2 speaker systems (with their accompanying electronics). The number of speakers and amplifiers varied with the venue, of course. Although Healy, Pearson and Danchik were the primary architects of the system used, it was actually Healy's son Michael that mixed the West Coast shows, receiving assistance for the first few shows from his father and, for the Los Angeles concerts, from Grateful Dead monitor mixer Harry Popick.

The East Coast dates, because of such pragmatic considerations as time and budget, used the house sound systems installed in the venues, with one significant exception.

ST. JOHN THE DIVINE PERFORMANCE

In New York, a ritual was performed at St. John the Divine, a famous Manhattan cathedral. At this concert, Hart and composers Philip Glass and Kitaro performed first, the conclusion of their performance cross fading into the monks. St. John the Divine has architecture typical of large cathedrals: it is much longer than it is wide; in fact it is on the order of a quarter of a mile long! For most church music, the low direct-to-reverberant ratio often experienced in the back of a cathedral is quite acceptable. In order to provide the clarity necessary to convey the monks' chanting style, however, the installed house sound system was used to supplement a system brought in by UltraSound's East Coast branch. Three stereo pairs of delay stacks (using Electro Voice speakers) lining the walls filled the cathedral well enough that very little volume was needed from any individual stack.

The performance at St. John the Divine was recorded for an upcoming CD release as part of the Rykodisc series Hart is producing. The board mix was digitized by a Sony 1630 processor and recorded onto time-coded videotape, while the pickup from a Neumann RSM190 M-S microphone in the house was similarly recorded through another 1630. Back in Berkeley, recording engineer Tom Flye decoded the M-S signal, synchronized the two decks, and transferred the audio to four tracks of a Mitsubishi X850 digital multi-track tape recorder. Following the East Coast swing, the monks returned to Northern California for a final concert (put together on very short notice) at the intimate, 2,000 seat Veterans Memorial Auditorium at Marin Civic Center in San Rafael. This concert differed from the rest in several ways. Unfortunately, the power supply for the console had gotten buried deep in the bowels of an equipment semi truck after the last Grateful Dead show. Thus, a 24-channel Yamaha 2404 was borrowed at the last minute from Leo's Audio and Music Technologies in Oakland. Additionally, AKG 414s were used for the fill microphones in place of the C34s and U87s. But the most significant difference was the use of a multi-channel sound system.

Tom Flye used a combination of traditional close-mic'ing (one mic for each monk plus fill mics for the instruments) and four stereo pairs at increasing distances.

Aside from the front speaker stacks, there were two UPA1s on either side of the hall which received independent feeds. Marin Veterans Auditorium was, coincidentally, the first place that the elder Healy (who mixed the monks there) attempted to use a multi-channel sound system with the Dead.

"My intention is not to generate six different mixes, but to generate a stereo mix that I can move through graduated auxiliary feeds to pull the spatiality out from the stage, to warp it out into the room. I'm not trying to make a statement about rear speakers and the conventional quad concept; I'm more into the groove of being able to use the rear sets of speakers to pull the mix farther out into the room. In this model I'm just moving it back in the room through amplitude." (Healy uses a system providing amplitude and time-based cues in his current multi-channel work with the Dead.)

The monks' rituals have also been recorded along the way onto a Sony PCM2500 Professional DAT recorder. At Zellerbach Auditorium in Berkeley, California, UltraSound's "MS Box" was employed. This system combines an M-S decoder, calibrated digital delays, and customized mixing and monitoring facilities to allow an optimum recording created from a mix of direct feeds from the console with an M-S microphone picking up the sound in the house. At Marin Veterans, Healy used three stereo pairs of microphones in a small-scale implementation of his multiple-image stereophony concept, which was used in recording the Grateful Dead's last two live albums: *Reckoning* and *Dead Set*.

SKYWALKER RANCH

Further, Hart took the monks to the newly created soundstage at Skywalker Ranch, filmmaker George Lucas' Northern California technological palace, to do more recording for the Rykodisc CD. Engineer Tom Flye used a combination of traditional close-mic'ing (one mic for each monk plus fill mics for the instruments) and four stereo pairs at increasing distances. Because of the large number of mics needed, he used a wide variety of mics, including: Neumann U67s, a modified U47, modified U87s, and TLM170s, and AKG 414s, C12s, and C12s. For the stereo pairs he used a Neumann SM69, and a pair of U49s, a pair of AKG C12s, and a pair of B&K 4000 series omnis (for the farthest pair). All these mics were run directly into an Otari DTR900 digital multi-track tape recorder. Two recordings were made at this session, one with the close mics and one without.

The monks' ritual, conducted on an altar featuring tapestries and statues, filled the Marin Veterans Auditorium with their ethereal chanting. The multi-channel sound system quite effectively allowed the audience to experience the acoustic interference artifacts that so thrilled Healy and Hart. After their conclusion, the audience drifted away, still entranced. And the monks, sated with experiences from America, prepared to return to their seclusion in India. db

WILLIAM R. GRAHAM

Computers, Humidity and Sound

• Have you ever gone to a lot of trouble equalizing a PA system to get the best sound possible, and then, when the crowd arrives, find that it sounds completely different? You made allowances for the absorption of the people, you made adjustments for the distances, in fact you took into account all the factors you could think of, but it just wasn't right. What went wrong?

Perhaps you weren't aware of the great effect that humidity, or a change of humidity, can have on sound in large rooms or spaces. In a recent paper (1) the effects of humidity on sound have been documented in much more detail than in the past. Referring to *Figure 1*, you will see that at 15 percent relative humidity (RH) that 8 kHz is attenuated 5.5 dB per 100 feet and 12.5 kHz is attenuated 7.6 dB. At 20 percent RH the 8 kHz is down to 5.61 while the 12.5 kHz attenuation has risen to 8.9 dB. Quite a difference for a 5 percent change in humidity! Add to that a change in temperature (which often occurs with a change in humidity) and you will see why this article can be of real help to you.

Earlier work on the subject of humidity and the transmission of sound can be found in the Master Handbook of Acoustics (2), and in Sound System Engineering (3), but they deal with a limited number of frequencies or humidity points. The real problem seems to be that by the time the audience has settled and you note a change in humidity, or in the middle of an outdoor concert cool moist air rolls in, there is no time to interpolate rather general data, make a lot of calculations, or start to spread charts out on top of your mixer. The solution to the problem is the small computer program presented here. Just enter the temperature and humidity and the screen will immediately show the attenuation per 100 feet at each frequency of your equalizer.

A second feature of the program is the ability to calculate the phase relationship of a sound wave reaching a distant point directly and by reflection (at a frequency of your choice), to calculate the change in phase with a change in temperature and/or humidity, and to determine the speed of sound at that temperature and humidity. The speed of sound is important, of course, when calculating equalization for room resonances. These printouts are shown in *Table 1* and *Table 2*.

The program, as presented here, was written for the Atari 130XE computer, but in everything but the printer routine, it can easily be converted to any personal computer. I suggest you just add your favorite printer routine in place of lines 5500 to 5570. You should also note that the Atari uses the command "GRAPHICS 0" to clear the screen.

```
10      KCEL=1.0814-03:TEMPFAC=0.18
15      KA=1.0645:KB=154000:KC=0.185:KD=.09:KE=3.31
20      DIM ANS$(1),HOLD$(1)
30      GOSUB 5501:REM ** SET PRINTER
100     GRAPHICS 0:? :? "DO YOU WISH TO WORK IN:"
105     ? "      C-CELSIUS, OR"
110     ? "      F-FARHENHEIT ";:INPUT ANS$
115     HOLD$=ANS$
120     GRAPHICS 0:REM ** CLEAR SCREEN
125     ?:?"ENTER TEMPERATURE ";:INPUT TEMPIN
130     ?:?"ENTER HUMIDITY ";:INPUT RH
135     IF RH=0 THEN RH=.005
140     TEMP=TEMPIN
145     IF ANS$="F" THEN GOSUB 5001
150     HUMFAC=RH*KCEL
160     FOR N=1 TO TEMP-1
165     HUMFAC=HUMFAC*KA
170     NEXT N
175     HUMFAC=(INT(HUMFAC*1000+0.5))/1000
```

ABOUT THE AUTHOR:

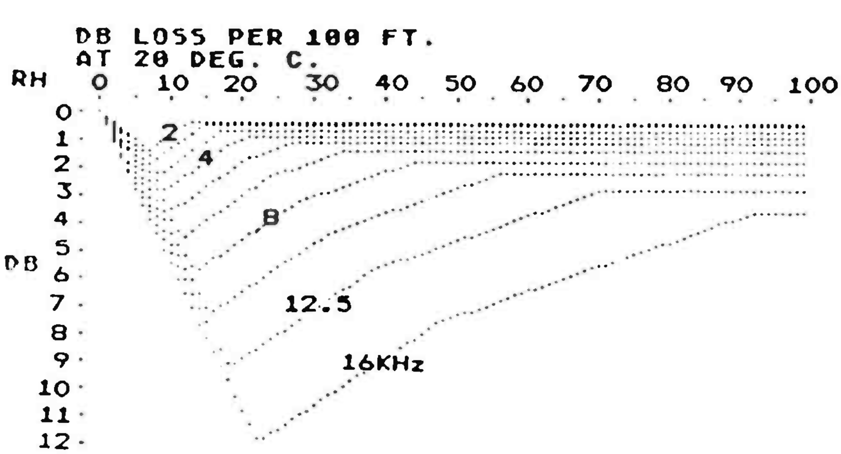
Bill Graham is a member of the Audio Engineering Society and the Society of Motion Picture and Television Engineers. He has many years experience in Canadian Broadcasting and is presently the Audio Visual Engineering Consultant at Mutual Life of Canada.

```
200   REM ** COMBINE TEMP & RH
205   VELFAC=HUMFAC+(TEMPFAC*TEMP)
210   VELFAC=(INT(VELFAC*1000+0.5))/1000
250   REM ** CONVERT VELFAC TO VELOCITY
255   VELSOUND=1087.42:REM ** AT 0 DEG C
260   VELSOUND =VELSOUND*((VELFAC/100)+1)
265   VELSOUND=INT((VELSOUND*100))/100
300   ANS$="X":GRAPHICS 0:?:?"DO YOU WISH TO DETERMINE:"
305   ?"      P-PHASE CANCELLATION, OR"
310   ?"      A-ATTENUATION ";:INPUT ANS$
315   IF ANS$="A" THEN GO TO 1001
400   GRAPHICS 0
405   ?:?"ENTER DIRECT PATH DISTANCE";:INPUT D1
410   ?:?"ENTER REFLECTED PATH DISTANCE";:INPUT D2
415   ?:?"ENTER FREQUENCY IN HZ";:INPUT FR
450   WL=VELSOUND/FR
455   CYC1=D1/WL
460   CYC2=D2/WL
465   PH1=CYC1-INT(CYC1)
470   PH2=CYC2-INT(CYC2)
475   IF PH2>=PH1 THEN PHA=PH2-PH1
480   IF PH1>PH2 THEN PHA=PH1-PH2
485   PHA=INT(PHA*360)
500   GRAPHICS 0
505   ?:?"      PHASE CANCELLATION RESULTS"
510   ?:?"AT A TEMPERATURE OF ";:?TEMPIN;:?" DEG. ";:?HOLD$
515   ?"AND A HUMIDITY OF ";:?RH;:?"% RH"
520   ?"THE VEL. OF SOUND IS ";:?VELSOUND;:?" FT./SEC."
525   ?:?"THE DIRECT PATH IS ";:?D1;:?" FEET, AND"
530   ?"THE REFLECTED PATH IS ";:?D2;:?" FEET"
535   ?:?"PHASE DIFFERENCE IS ";:?PHA;:?" DEG."
540   ?"FOR THE FREQUENCY ";:?FR;:?" HZ."
545   ?:?:?:?:?:?"....CALCULATIONS ARE COMPLETED"
550   ANS$="X":?"DO YOU WISH TO:"
555   ?"        B-BEGIN AGAIN, DO"
560   ?"   P-PHASE CANCELLATION, OR"
565   ?"   O-OUTPUT THIS SCREEN TO PRINTER";
570   INPUT ANS$
575   IF ANS$="B" THEN GRAPHICS 0:GOTO 120
580   IF ANS$="P" THEN GOTO 400
585   IF ANS$="O" THEN GOSUB 5525
590   IF ANS$<>"B" AND ANS$<>"P" AND ANS$<>"O" THEN ?"THAT
      CHOICE IS NOT POSSIBLE":?:ANS$="X":GOTO 570
595   ANS$="X":GOTO 570
1000  REM ** ATTENUATION OF SOUND
1001  RESTORE:GRAPHICS O
1005  ?:?"    DB LOSS PER 100 FT.":?:?"AT ";:?TEMPIN;:?" DEG. ";:?HOLD$;:?" AND ";:?RH;:?"% RH"
1010  ?:?"FREQ        LOSS"
1015  FOR N=0 TO 9
1020  READ FR
1025  PKRH=((1+HUMFAC)*(FR/1000))+5
1030  PKATT=FR/(KB-(FR/2))
1040  PKDBL=(10^((VELFAC/100)+2))*PKATT
1045  IF RH>PKRH AND RH<=PKRH*2 THEN DBL=PKDBL-((RH-
      PKRH)*KC)
1050  RHA=(RH-(PKRH*2))*KD
1055  IF RH>PKRH*2 THEN DBHOLD=PKDBL-((PKRH*2)-PKRH)*KC:DBL=DBHOLD-RHA
1060  IF DBL<PKDBL/KE THEN DBL=PKDBL/KE
1065  IF RH<=PKRH THEN DBL=PKDBL/(PKRH/RH)
1070  DBL=(INT(DBL*100))/100
1075  ?FR;
1080  IF FR<10000 THEN ?" ";
1085  ?"       ";:?DBL
```

```
1090    NEXT N
1095    ?:?:?"CALCULATIONS ARE COMPLETED......"
1100    ANS$="X":?"DO YOU WISH TO:"
1105    ?"          B-BEGIN AGAIN, OR"
1110    ?"          O-OUTPUT THIS SCREEN TO PRINTER";
1120    INPUT ANS$
1125    IF ANS$="B" THEN GOTO 120
1130    IF ANS$="O" THEN GOSUB 5525
1135    IF ANS$<>"B" AND ANS$<>"O" THEN ?"THAT CHOICE IS NOT POSSIBLE":GOTO 1100
1140    ANS$="X":GOTO 1120
5000    REM ** CONVERT FARHENHEIT TO CELSIUS
5001    TEMP=TEMPIN-32
5005    TEMP=TEMP/1.8
5010    RETURN
5500    REM ** SCREEN PRINT ROUTINE **
5501    CLOSE #4:CLOSE #5
5505    OPEN #5,4,0,"S:"
5510    OPEN #4,8,0,"P:"
5515    RETURN
5520    REM ** TEXT FROM SCREEN
5525    FOR Y=0 TO 16
5530    FOR X=0 TO 39
5535    POSITION X,Y
5540    GET #5,G
5545    ?#4;CHR$(G);
5550    NEXT X
5555    ?#4
5560    NEXT Y
5565    POSITION 0,23
5570    RETURN
7000    REM ** FREQUENCIES FOR ATTENUATION CALCULULATIONS
7001    DATA 2000,2500,3200,4000,5000,6300,8000,10000,12500,16000
7005    REM ** THESE FREQ. MAY BE ALTERED TO SUIT YOUR NEEDS
7010    REM ** FREQ. BELOW 2000 ARE INSIGNIFICANT FOR ATTENUATION CALCULATIONS
```

LINES 10 to 30: set up the program. The various factors used in the program are lines 10 and 15. In this way they are easy to locate and change as new work is done on this subject. Line 30 calls up a routine to set the printer.

Figure 1.



LINES 100 to 145: put the first menu on the screen and store the choice, then put up the screen calling for temperature and humidity. Then, if Fahrenheit temperature is being entered, it calls up the subroutine at line 5000 to convert these values to Celsius.

LINES 150 to 175: create a humidity factor for the temperature given.

LINES 200 to 210: combine temperature and humidity into a velocity factor.

LINES 250 to 265: take the velocity of sound at 0 degrees Celsius and re-calculate for the factors given.

LINES 300 to 315: here the program branches to calculate Phase Cancellation or Attenuation.

LINES 400 to 415: input the data required for Phase Cancellation

LINES 450 to 485: calculate the wavelength and phase angle.

LINES 500 to 565: create the screen of information calculated.

Table 1.

DB LOSS PER 100 FT.

AT 20 DEG. C AND 45% RH

FREQ	LOSS
2000	0.43
2500	0.53
3200	0.69
4000	0.86
5000	1.08
6300	1.37
8000	1.75
10000	3.18
12500	5.07
16000	7.98

Table 2.

PHASE CANCELLATION RESULTS

AT A TEMPERATURE OF 68 DEG. F
AND A HUMIDITY OF 45% RH
THE VEL. OF SOUND IS 1128.3 FT./SEC.

THE DIRECT PATH IS 40 FEET, AND
THE REFLECTED PATH IS 135 FEET

PHASE DIFFERENCE IS 9 DEG.
FOR THE FREQUENCY 10000 HZ.

LINES 570 to 595: provide the options printed on the bottom of the last screen.

LINES 1000 to 1010: print the screen header when the Attenuation option has been chosen in line 310.

LINES 1015 to 1090: calculate and print to the screen, in turn, the frequency and dB loss at that frequency for the temperature and humidity entered. The frequencies are read from line 7001, and may be changed to suit user needs.

LINES 1095 to 1110: complete the screen with options.

LINES 1120 to 1140: provide the option chosen.

This program certainly has been useful to me and I hope it provides you with an additional way to make your PA work better and easier.

Who knows, the "wet bulb thermometer" may become a standard tool for audio enineers.

BIBLIOGRAPHY

1 D. A. Bohn, "Environmental Effects on the Speed of Sound", J. Audio Engineering Society, vol. 36, num. 4, pp. 223-231 (1988).

2 F.A. Everest, "The Master Handbook of Acoustics", TAB Books Inc., pp. 66-68 (after Knudson, Haris).

3 D. and C. Davis, "Sound System Engineering" (1975) Howard W. Sams and Company Inc., pp. 52-55.

ANDRE WALTON

Noise Gates—Back to Basics

What is a noise gate? What does it do? Our author looks at these questions, as well as others, in his tutorial.

Originally, the function of a noise gate was, quite simply, to eliminate unwanted noise, and the only really important criterion was that the electronics were designed so that the gate itself had no inherent pops, crackles or hum that might replace the external noise being filtered out.

Today's sound processing market offers a great variety of highly sophisticated—and highly expensive—noise gates that are capable of an enormous range of complex operations, but for the vast majority of day-to-day applications, a fairly basic noise gate is still all that is necessary.

APPLICATIONS

A simple noise gate is an immensely useful device for a multitude of "cooking" sound processing applications. Essentially, it listens to the sounds going on and when these fall below a certain level, it immediately turns off the signal source, turning it on again when the sound again reaches the pre-set level. This means that unwanted noise—echo, audience fidgets, tape noise, amplifier hum, or any other disturbance—can be easily and simply eliminated.

Andre Walton is president of C-T Audio Marketing of Boynton Beach, Florida..

It is a facility that can be particularly useful where sound equipment is less than perfect; a noisy amplifier from an electric guitar, for instance, will generally be completely drowned out by the guitar itself when it is being played, and is objectionable only when the guitar is quiet. A noise gate will come into operation as soon as the guitar falls silent, shutting down the random background noise from the amplifier. A basic noise gate like the Omnicraft GT4A will perform this task perfectly, and, even in an extreme situation where the hum is loud enough by itself to trigger the gate, this can be dealt with by conditioning the Key input. To do this, use a "Y" adapter or a patchbay mult to split the noisy signal into two paths before it enters the gate. Filter one path to remove the hum and patch that signal into the Key input. Patch the remaining signal back into the Input and the gate can then be operated normally without being triggered by hum.

During a live performance, a noise gate can be used to dramatically increase the sound level before feedback; this is done by gating each input channel of the PA mixer during the performance, which automatically turns each mic off when not in use. Using the GT4A, this is inserted into the input channel after the mic preamp—a provision that is available on most professional mixing desks.

A similar technique can be applied in the studio in order to reduce leakage between tracks of a multi-channel recording. During multi-channel dub-downs, each track can be gated to turn off when there is no content. This can make a substantial reduction in apparent tape hiss, especially in production work where there may be long periods of silence—on tracks such as sound effects, for instance.

NOISE GATE DESIGN

All noise gates consist of some form of attenuation device that is activated when the signal to be gated exceeds a certain threshold level. Traditionally this attenuation device is a Voltage Controlled Amplifier, but this has a tendency to add noise of its own. Omnicraft gates employ a unique means of overcoming this inherent noise problem: an optical switch that is capable of rapidly attenuating the signal to be gated does not pass through any active components, but instead sees only a resistor network. As a result, the GT4A is far quieter and less intrusive than other, similar devices on the market, a virtue which is enhanced by careful tailoring of the frequency response of the gating circuit, and the speed at which the device turns on.

BASIC PLUS—USEFUL ADDITIONS

Range is one of the most useful additions to a noise gate. On a basic noise gate there are only two positions—on and off—that is, either you have sound or you have no sound. With a range control, you can select the attenuation level to leave a degree of background noise, which often gives a much more realistic and "live" feeling to the soundtrack than absolute silence. One application in particular is the cleaning up of old dialogue tapes, which tend to sound sterile and unnatural if the random noise is totally removed. A certain degree of ambient noise can either be left on using the Range control, or by mixing in some of your own studio cleanroom "presence" after the signal goes through the noise gate.

KEYING

We have already mentioned one use for the Key function—another feature that is often added to the basic noise-gate design. Essentially, the Key inputs enable the signal going through the noise gate to be keyed on and off from a different source, but this simple description belies the wide range of uses and special effects that can be achieved with this function. In its most simple use, one instrument can be keyed by another, which is ideal for ensuring that no other instrument in the group beats the lead instrument to the starting post.

In a more esoteric environment, tones can be keyed by voices to give "computerized" or "outer space" effects. An excellent in-house substitute for footsteps on a gravelled surface is a continuous track of cellophane being crumpled that is keyed on and off by footsteps in the studio.

Echo returns can be keyed by echo sends to alter the decay time of an echo chamber. Key inputs can be filtered or conditioned so that instruments are heard only when certain notes are played—the possibilities for imaginative keying are endless! db

Sound Reinforcement in Central America and the Caribbean, Part III

The Don Cherry Quartet held a workshop Wednesday that was attended by many professional musicians. Some of us enjoyed a concert that night by Trinidadian pan legend Len "Boogsie" Sharpe; the things he did with pans were beyond belief!

Thursday was a free day, enjoyed in many different ways. Carlos took off for the neighboring island of Tobago, which he described as "an island paradise." Bob and I took advantage of Mike Boothman's hospitality, and took a brief tour around the Port of Spain area. I also phoned the Dominican Republic, and spoke to CAO Lezetta Moyer. I informed her of our sound needs in support of my cable, as the monitor situation in Trinidad indicated that something might have gotten lost in the translation.

On Friday, December 11, it was time to pull up stakes and make our move to the Dominican Republic. The 5 am band departure from the hotel was bad, but mine was worse: I left at 4:30 with Bob Dance to handle the task of clearing our band gear and medical stuff through customs. Our BWIA flight left just after 7 am; it was another "island hopper" that deposited us in Puerto Rico at 11 am. Toney managed to squeeze in a fluid change for Ed and chase our baggage during the two hour San Juan layover. Our Eastern flight from San Juan deposited us at the Las Americas Airport in Santo Domingo promptly at 2:25 pm. We had a fairly long walk to the terminal, and I could see a wave of reporters and TV cameras waiting just outside immigration. We wondered who the celebrities were, at least until we got to the door. The big greeting was for us (*Figure 1*)!

CAO Lezetta Moyer let the media have their way with us, then supervised an orderly departure from a very disorderly arrival area. The coastline views on our way into town were lovely, and preparations for our arrival at the Hotel Lina made registration quick and easy. Lezetta gave a short briefing and supervised the delivery of Ed's last dialysis stockpile. Our tired group opted to relax away the rest of our day.

Figure 1. Greeted at Las Americas Airport, Santa Domingo, Dominican Republic. From left to right: Ed Blackwell, Bob Stewart, CAO Lezetta Moyer, author Ed Learned, Carlos Ward, and Don Cherry.

CHECKING THE VENUE

USIS radio/TV tech Tony Perez picked me up early Saturday for my morning inspection of the National Theater, located a short distance from our hotel. This theater had a capacity of 1700, and was a completely modern facility with all the conveniences. The soft carpet and acoustical treatment of the walls kept reverb time to a very modest 1 second despite the large interior dimensions. The a.c. power was available from US-type receptacles offstage right or left; the grounds were functional, and I measured a very stable 115 volts, thanks to the theater's own transformer. Lezetta had hired Sistemas Profesionales, S.A., a sound company from Santo Domingo, to provide sound for all our concert dates in the DR. I introduced myself to engineers Miguel Gonzalez del Rey and John Risk, who already had my stage layout and mic list; Lezetta had obviously done her homework!

Per side, the PA was comprised of 2 single 15-inch horn-loaded bass bins, 2 single 12-inch horn-loaded mids, 4 Altec metal multi-cell horns, 2 with JBL 2425 drivers, 2 with Peavey 22A drivers, and 4 Foster tweeters. The system was run 4-way via an Ashley XR-88 stereo 4-way crossover. Four custom-built amplifiers powered this system: the larger amp produced 600 watts/channel at 4 ohms, the smaller 100 watts/channel at 8 ohms. There were 2 different monitor designs: 2 had a 15-inch woofer, exponential horn with JBL driver, and 1 piezo tweeter, 2 had a 15-inch woofer and 1 piezo tweeter. Both styles were passively crossed over, and each was powered by one side of a Peavey CS-400. I assigned the 3-way wedges to Don and Carlos, the 2-ways to Bob and Ed. Front-of-house gear included a Carvin 1644 16-channel console with internal graphic EQ (used for the house system), Yamaha GQ 2031 stereo 1/3-octave graphic EQ (monitors), dbx 166 compressor/limiter, DigiTech 3.6 Sec delay, and an Alesis Microverb.

Their mic selection was excellent, so I used an E-V RE-20 on bass drum (I preferred it to my 421), a Shure SM-58 on the snare, and 2 E-V PL-10s for overheads. A Sunn bass head with on-board graphic EQ and 2 single 15-inch folded horns were provided for Bob's tuba. When I first fired up the PA with taped music, I notices a large difference between the two sides. Closer inspection revealed that one bass cabinet house left was out-of-phase with the rest, and two horns house right weren't working. This surprised Miguel, who had checked each component individually before turning the system over to me. We traced the problem to an intermittent cable (isn't it always the case), and after some system tuning, I was pleased with the result. I returned to the hotel just after noon, and spent the day relaxing at poolside.

WAITING IN THE DARK

When I returned to my room to prepare for my 6:15 pm departure, I noticed that only a few of my room lights were working, and my electric razor didn't work at all. The desk informed me that the power was off; they were running designated lights off an emergency generator. Toney confirmed my worst fear: the power was dead in this entire section of Santo Domingo, which included the National Theater. This would effectively kill our concert unless things were restored. I went over to the hall with Tony Perez; Toney and Lezetta stayed at the hotel with the group, awaiting further developments.

The theater was pitch dark when we arrived, with only hallway emergency lights functional. Miguel and John had disconnected their a.c. to protect against spikes if and when power was restored. Our concert was scheduled for 8:30 pm, so we sat in the dark and talked to while away the time. They asked me quite a few questions about sound contracting in the U.S. At 7:45 pm, the lights came back on; John quickly restored a.c. while Miguel checked the system. The band arrived at 8 pm, and the doors were held while I conducted a rapid sound check. The situation did not hurt our attendance: we had a complete sellout, with standing room in the back and even some people sitting in the aisles.

They were completely enthralled by the quartet, and responded with the most vigorous applause of the tour. The band responded to this encouragement with their most brilliant performance to date. Ed's infectious African rhythms on *Leto* had the crowd clapping along, and a romantic rendition of Carlos Ward's *First Love* brought over 1600 people to a respectful hush. Perhaps the biggest response was reserved for Bob Stewart. He played a magnificent solo on his own *Nonet*: it was at times a swing/funk poly-rhythm, other times a series of long tones, doubled by the delay into a shimmering wave. I took full advantage of the stereo PA to do some wet/dry panning, so the tuba seemed to whip across the audience's perspective. The crowd was mesmerized, and tumultuous applause made it clear what they thought about his exceptional playing. Many people stayed after the show for autographs and conversation, and some even accompanied us to a local nightclub, where we en-

Figure 2. Sistemas Profesionales house left PA stack at the jam session concert. I used this PA throughout the D.R. The amp rack and the hardwood reflector at the Institute are also visible.

Figure 3.The author mixing the jam session.

joyed a performance by one of the hottest local bands.

After a restful Sunday-by-the-beach, I spent most of Monday preparing for perhaps the most interesting show on the whole tour. Lezetta had scheduled a jam session concert at the Instituto Cultural Dominico-Americano. The idea was for the quartet and local Dominican musicians to jam together in a concert setting. The actual structure of the event was open to debate. Don suggested a 2-part show: part 1 would feature the Don Cherry Quartet with different combinations of local musicians; part 2 would feature the Dominican All-Stars with different combinations of Don's band. Quartet compositions would be featured in the first half, Dominican compositions in the second half. I was over at the Institute by 9 am Monday to meet some of the musicians and see to the PA arrangements. The auditorium was a square room with an extremely high ceiling. Capacity was 600, including two balconies set high on the house left wall; hardwood reflectors covered the same area everywhere else. These hard wall surfaces, along with the tile floor and high ceiling spelled doom for clarity: reverb time was a nasty 2-1/2 seconds full of colorations.

Figure 4. The set up at the Steel Shed, Bridgetown, Barbados.

Power was located on the three stage walls; US-type receptacles provided 125 volt a.c. The wiring scheme of these plugs was out-of-phase to normal US code. John and Miguel had the PA already up and working (*Figure 2*); I re-tuned the system, concentrating on a tighter low end and attenuated the 1.8-2 kHz area to reduce brash response enhanced by the room. With two percussionists, electric bass, 2 electric guitars, 2 horns, and a keyboard mix added to our normal set-up, we had no problem filling up available stage space or empty console inputs.

We held a rehearsal/sound check late that afternoon; I tried to impress the electrified players with the importance of low stage volume when playing with acoustic instruments. For instance, it's hard to amplify a doussin' gouni when there is someone playing electric guitar too loudly in the immediate vicinity! A cruel but simple truth for electric musicians in this room: if you could hear yourself really well, you were probably too loud.

JAM SESSION

Monday's 8:30 pm jam-concert audience surpassed anyone's expectations. The institute auditorium was crammed full; Tony Perez even set up TV monitors outside in the institute's courtyard so several hundred fans with no seats could watch and listen. This mass of humanity helped dry the room reverb up to the point where I could take off my handcuffs!

Don's opening announcement concerned the legacy of jam sessions in jazz history, and his wish that both participants and audience be open to whatever music would be made. The quartet played a few songs alone, then were joined as various Dominicans walked out to improvise accompaniment. Some of the best combinations were vocal, doussin' gouni, electric guitar, bongos, and hand percussion during Don's blues, Ed and several hand drummers in a furious percussion duel, and Carlos leading a 3-piece sax section. The All-

Stars segment featured a completely different musical direction: their music was heavy jazz-rock fare, laced with Latin rhythms. Don and the group proved to be equally adept at this style as well. Bob joined with the electric bass in some serious electronic jousting; Carlos and Don played furious horn solos over the bubbling rhythm section. The crowd went absolutely nuts!

It was a tremendous kick for me to mix two such totally different "bands" in one concert; I loved it (*Figure 3*). The "All-Stars with Quartet" jam was reminiscent of the Miles Davis bands I'd worked with in the mid 70s. I received a real compliment from John and Miguel: they said their PA had never sounded that good before. Lezetta mentioned that it was the first time she'd ever been able to hear instruments clearly in that room without pain. Considering the adverse acoustics, this was high praise indeed.

We left Santo Domingo late Tuesday morning for Santiago, located about 100 miles to the northwest. The 2-1/2 hour drive gave us a leisurely look at the country's interior. After a late lunch, Tony Perez took me over to the Madre y Maestra Catholic University, site of our 8:30 pm concert. The theater seated 500; its carpeted floors and thickly upholstered seats soaked up a lot of sound. Reverb time was just over 1 second, a veritable paradise over the last venue.

John and Miguel had my layout memorized by now: the stage was completely set up when I arrived. Sound check was quick and easy, and we even had an hour to relax before the show. Some of our new fans had followed us to Santiago, helping to swell our audience close to capacity. Ed changed gears with a succinct drum solo that featured slow tempos fashioned out of mallet work on the toms and open snare. Carlos later scorched everyone with his solo's intensity and facility on *Pettiford's Bridge*. I taped this performance for Lezetta to use in her library archives or for later re-broadcast.

We drove directly to Las Americas Airport Wednesday morning to depart for our final Caribbean destination. Lezetta adroitly handled customs and emigration formalities, so our exit was quick and hassle-free. We again endured a 3-hour layover in San Juan before catching an American Airlines non-stop flight to Bridgetown, Barbados; none of us missed the island hopper.

Figure 5. The PA stack, house left, at the Steel Shed, Barbados.

We arrived around 10 pm, and were met by CAO Dr. Ambrocio Lopez and CAS Susan Walker. Barbados customs officials gave us a cursory inspection, then sent us on our way. The drive to our hotel, the Rockley Resort, took about a ½-hour; we spent some of that time twisting through heavily populated city areas. The wide-open spaces we'd seen in Trinidad and Jamaica were not in evidence around Bridgetown. I had a chance to chat with Susan about the PA situation, and discovered I'd be using in-house systems at two of our three concerts. She had also arranged to rent extra monitors, as none of our venues had more than two.

THE STEEL SHED

Thursday, December 17, marked our first Barbadian performance. It was a 10 am charity performance for the handicapped children of Barbados, and was held at the Steel Shed. Its name was descriptive of its construction: the concrete floor and

Figure 6. The mix point at the Steel Shed, Barbados.

Figure 7. Don plays his dussin' gouni at the Steel Shed, Barbados. Note the microphone placement with respect to the hole in the body. By the way, we are told, he doesn't usually play sitting down!

steel roof created yet another acoustical torture chamber, with reverb time well over 2 seconds (*Figure 4*). Power was available offstage right or left on US-type ungrounded receptacles; I measured 118 volts. I didn't have a lot of time to set up, so I elected to go with my front line mics and the two DI boxes; drums would go acoustic. The house PA system included 2 H/H Electronic 15-inch woofers and 1 H/H Electronic radial horn per side (*Figure 5*). House power amps, graphic EQ, and a 20-input Soundcraft 1S console were located in a mixing booth located high in the rear wall overlooking the stage (*Figure 6*). I found by experimentation that if something sounded clear at the mix point, it was too loud in the audience area. A murky-sounding mix was therefore a good mix.

My monitor augment of 2 Peavey wedges, powered by a Peavey CS-400 amp, was designated for Don and Carlos. House wedges were used for Bob and Ed. A Peavey TKO65 bass amp was provided for Bob's tuba. We had our smallest audience of the tour: only 30 people showed up. The many children in attendance were completely captivated by Don's doussin' gouni (*Figure 7*); he played a "talking 'bout Africa" blues especially for them.

Friday's "concert" was actually a mini-performance at the home of Paul Russo, US Ambassador to Barbados. The band would perform a short 30-minute set for the invited guests. The "stage" was a small landing overlooking the Russo's pool and courtyard. Two Yamaha cabinets and an 8-channel powered mixer had been provided for PA. The major parameter here was to remain unobtrusive: the situation did not require much sound reinforcement. I placed both speakers on a single tripod and used a single-point stack, splaying the two cabinets to cover the courtyard area completely. We used no monitors, and I put up only trumpet, sax, and vocal mics. The highlight of the evening's short set was a rollicking version of *Surinam*, which induced even the most staid party-goers to get up and move.

Figure 8. The bands last concert on the tour, at the Frank Collymore Hall, Bridgetown, Barbados.

LAST STOP THIS TIME

Saturday, December 19, our last scheduled performance day of the tour, was spent at the Frank Collymore Hall, a beautiful new concert facility seating around 500. The stage thrust out into the room and was surrounded by the audience on three sides. Stage a.c. was supplied either from receptacles located on the rear-stage wall, or from floor pockets. US-type supplied grounded 120 volts, while UK-type receptacles supplied grounded 240 volts.

The PA system was flown from the ceiling just over the lip of the stage; three separate two-way clusters pointed to each of the three main audience areas. Two E-V passive 2-way floor monitors were provided for stage use; UREI power amps and graphics provided power and EQ, respectively, for both systems. A Biamp 16-channel console was located in a mix booth built into the rear center wall. I used the house wedges for Bob and Ed, with my augment handling the front line.

When I first turned the house system on, 2 of the 3 high-frequency sections didn't work. The fault was in the extensive patchbay located in the mix booth; eventually, all the horns were made to work, but the system lacked any real impact. A loud transient, such as one of Bob's elephant calls, would easily send the PA into distortion. There was again a serious discrepancy between apparent volume at the mix point and volume in the seating area. I tried to get a feel for the relationship between headphone cuing of the house mix and audience level; this gave me a frame of reference for where my mix was really at vis-a-vis the audience's perception. This seemed to work well, although maybe I needn't have worried: attendance again was terrible. We had less than ¼ of a house, but this included some rabid Don Cherry fans (*Figure 8*). In response to their requests, Don included pieces of Ornette's *Lonely Woman* and *Peace* in several of his solos. Carlos contributed a lively flute solo on Don's composition *Guinea*, and Bob displayed the ability to hum and play at the same time on his *Nonet* solo.

Some of us enjoyed the opportunity to sleep on Sunday. Toney and Susan, however, made an early morning trip to the airport to clear the equipment and luggage in advance of the group's 6 pm departure. Our entire entourage rendezvoused for a last group lunch at the Rockley just after noon; there was much reminiscing and good-natured kidding about our many shared experiences on this wonderful tour. Toney was toasted by all of us; his quiet diligence and attention to detail was one of the secrets responsible for our lack of logistical difficulties. I bid farewell to our group in the hotel lobby around 3:30; I'd elected to stay in Barbados a few extra days to enjoy some R&R on this lovely island. I'd learned a lot about music from these guys; we parted company with a healthy respect for each other's abilities. The best benefit of all was in acquiring five new friends. db

UB40 Tours Israel

Israel, for four days in April, hosted the Reggae band UB40. They arrived the night before the first concert was scheduled. Most of the requirements for the concert were to be supplied by the local sound company, Betty Bam. Betty Bam, one of the largest sound and light companies in the country, has done many shows previously. Some of the groups which they have provided PA for in the last two years have been Bob Dylan/Tom Petty and the Heartbreakers, Joan Armatrading, Bryan Adams, Chick Corea, James Taylor, Jethro Tull, Santana and Joan Baez to mention just a few.

One aspect of the concert was that all were a bit anxious as to how the new Turbo Sound System would work. Though the system had been in Israel since late January, the concert with UB40 was the first big concert series by outside artists. The system presently consists of sixteen TMS-3s and four TSW 124 subwoofers. By the time this article is published, there will be eight more TMS-3s and four TSW 124s.

Miron Reuther, president of Betty Bam, cited a number of reasons for the Turbo System. The first is that Turbo Sound is one of the leading speaker manufacturers to date, and he was looking for a new modern modular speaker enclosure. The TMS-3 is exactly what he wanted. It packs very neatly in trucks with detachable wheel carts. The second, and maybe even more important reason, is that Betty Bam and the English sound company Britannia Row have set up an Israeli company, Row Productions, Israel in which Betty Bam and B.R. are partners. At present, B.R. England has in their inventory 240 TMS-3s and 60 TSW-3 subs. If these aren't enough reasons, Turbo Sound is an English company and therefore is physically closer than American companies. Let's not forget that the "turbo's" sound is really great and it's also an essentially

Ron Altman is a sound engineer living in Israel.

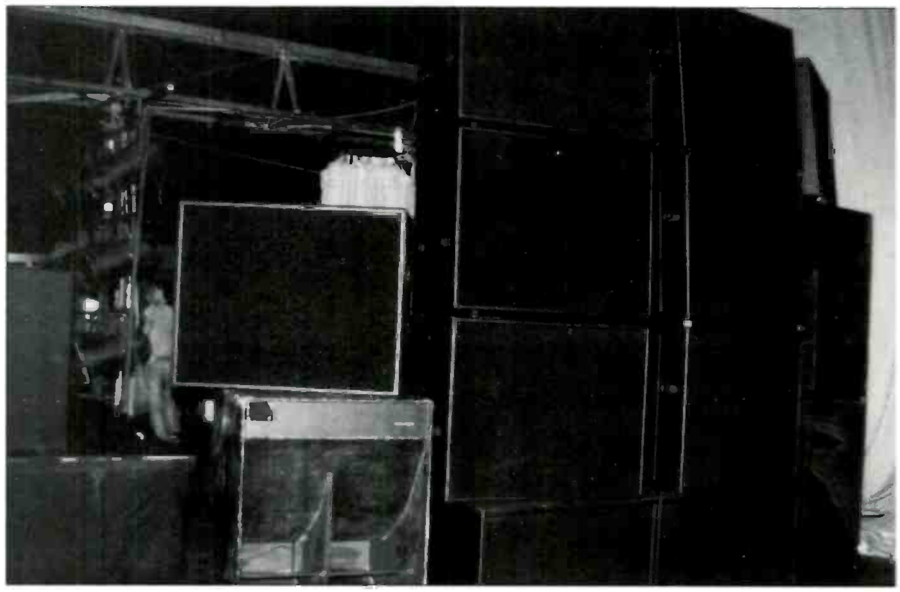
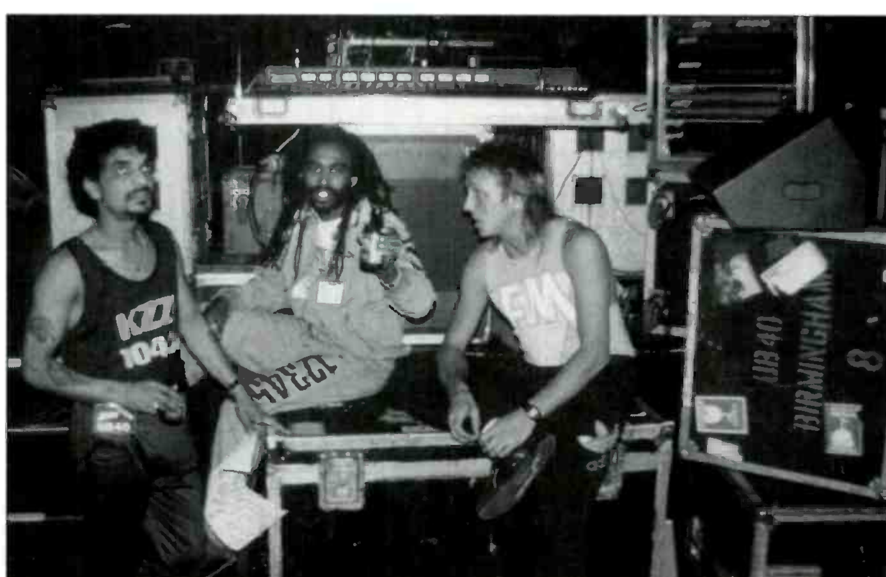


Figure 1. Speaker Stacks at Cinerama (right side. Note that the TMS-3 cabinets are sitting on the "house" Martins.

Figure 2. From left to right, UB40s drum technician, keyboard technician, and monitor engineer Vince.



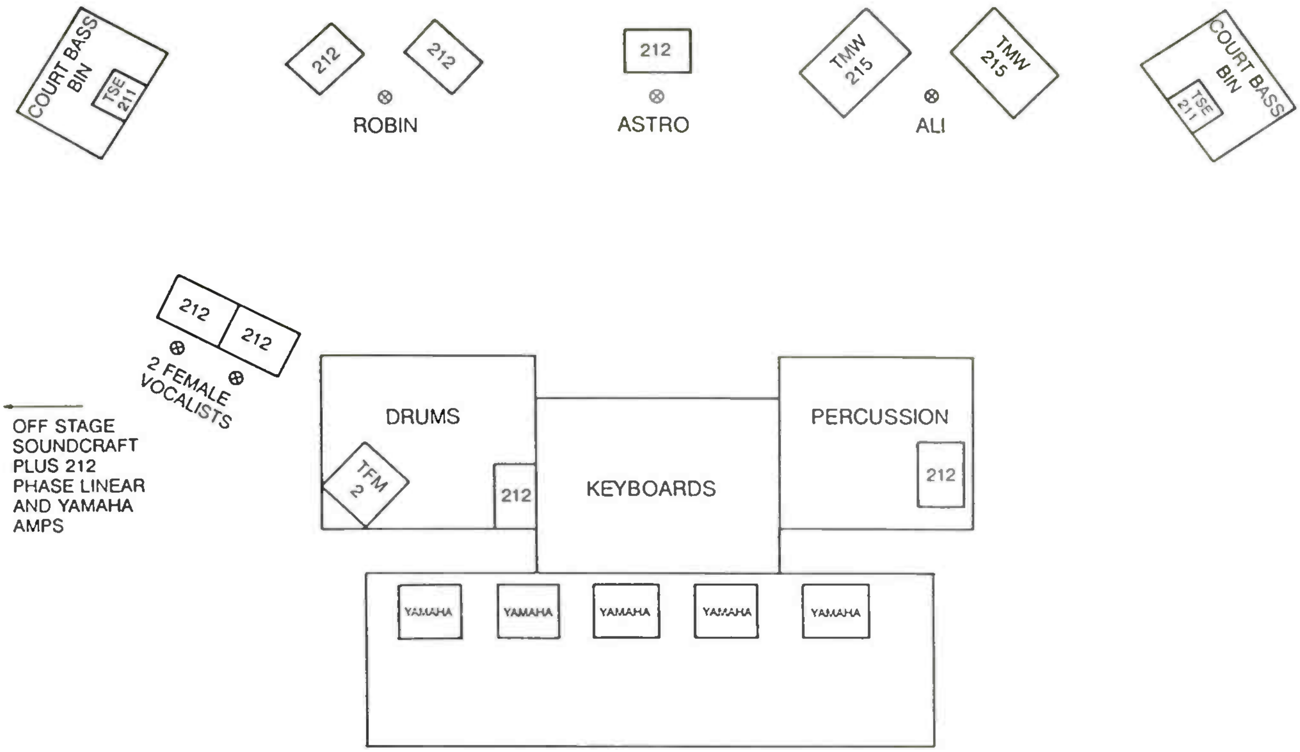


Figure 3. The stage setup at Cinerama

good-looking speaker cabinet. Miron also has a good supply of Crest Amplifiers, 8001s and 4001s.

SHOW AT THE CINERAMA

As UB40 spent the night sleeping in Tel Aviv, the engineers and crew of Betty Bam were getting ready for the first evening's show at the Cinerama. The Cinerama is the largest disco in the Middle East with a capacity of well over 3000. The hall is circular and has a domed ceiling, though it's hard to see with all the effects hanging from it.

After all was loaded and it had arrived OK at the Cinerama, the task faced was to take down the house PA. The rider specified that the Turbo Sound System was to be used throughout the tour. The house people at the Cinerama were very cooperative and even let Betty Bam use some of the "Martin" Bass Bins as support for the TMS-3 cabinets (see *Figure 3*).

In addition to the house PA, Betty Bam also supplied the monitor system. It was comprised of side fills, five front-of-stage monitors, percussion, drums, keyboards, rhythm section, and background vocalists.

The side fills were made up of one Court bass bin and one Turbo TSE 211 high pack with an active BSS crossover. The front vocals were three TMW-212s—one for the lead vocals, two for rhythm guitar/vocals and two TMW 215s. The drum monitor was a TFM-2 with BBS crossover, bi-amped, percussion, keyboards and background vocals; each received a TMW 212. On the back riser were five Yamaha two-way monitors. The power for the side fills was Phase Linear power amps. The rest was Yamaha 2200 for rhythm and vocals.

The monitor console was a Soundcraft 24x8. Besides the normal band

Figure 4. Stage left view.

Figure 5. (Left) An Israeli technician worked the equalization.

Figure 6. (Right) viewed from left to right, Steve "Gonzo" Smith, Chris the tour manager, and Gerry Parchment.

gear carried by UB40, they also tour with their own house console which is a TAC SR9000 desk modified for the road. Also included in the F.O.H. cases were their effects, gates and limiters. Some of the gear found in the racks was 480 Lexicon, SPX90 and Rev 5, an AM5 RMX 16 and an AMS DMX 80 Reverberator, eight channels of dbx RM 160s and 16 channels of Drawmer 201 gates.

IS TWO BETTER THAN ONE?

Behind this massive array of equipment you will find two very competent engineers totally opposite one another. The "old" man is Steve "Gonzo" Smith, who's been in the business eighteen years, nine of them with UB40. He started out as their monitor engineer and after a while moved to the front of the house. Next to him you will find Gerry Parchment, a twenty-three-year-old who, for the last four years, has been the UB40 studio engineer and during this world tour worked side by side with "Gonzo" doing their live sound as well.

The rhythm and lead guitars' wireless mics ran off stage where the respective technicians cared for the effects

When I asked them why two, they answered that UB40 tries to get as close to the album as possible and there is just too much work for one sound engineer to handle by himself. As I talked to Gerry, he also had high praise for the Israeli crew. Sound checks happened as planned and all was done on time. He mentioned that there were certain advantages to non-union crews. Gerry also thought that the TSW 124s sounded really great.

Monitor engineer Vince, besides having to deal with nine monitor mixes, was also responsible for the two vocal wireless Shure SM 58 mics and the three wireless mics for the rhythm section, Shure SM98s. The rhythm and lead guitars' wireless mics ran off stage where the respective technicians cared for the effects, mutings, etc., of the guitars.

With all the guitar amps and the wireless mics off stage, downstage was really nice and clean. The low profile TMW 212 helped matters even more. It's really important for the artists to have a feel of the audience and for the audience to be able to see the artists they are paying to see.

So now we are set for the first night of the Cinerama. I was sort of hoping that something would go wrong as it makes for a more interesting article. Most unfortunately for me, nothing did. Originally, three performances were planned, in Tel Aviv, Haifa and Jerusalem. Due to great demand, an additional concert was added in Tel Aviv. Gonzo and Gerry had only praise for what happened in Israel, they even liked the catering.

As UB40's tour was a success, let's hope there will be many more throughout the summer season. db

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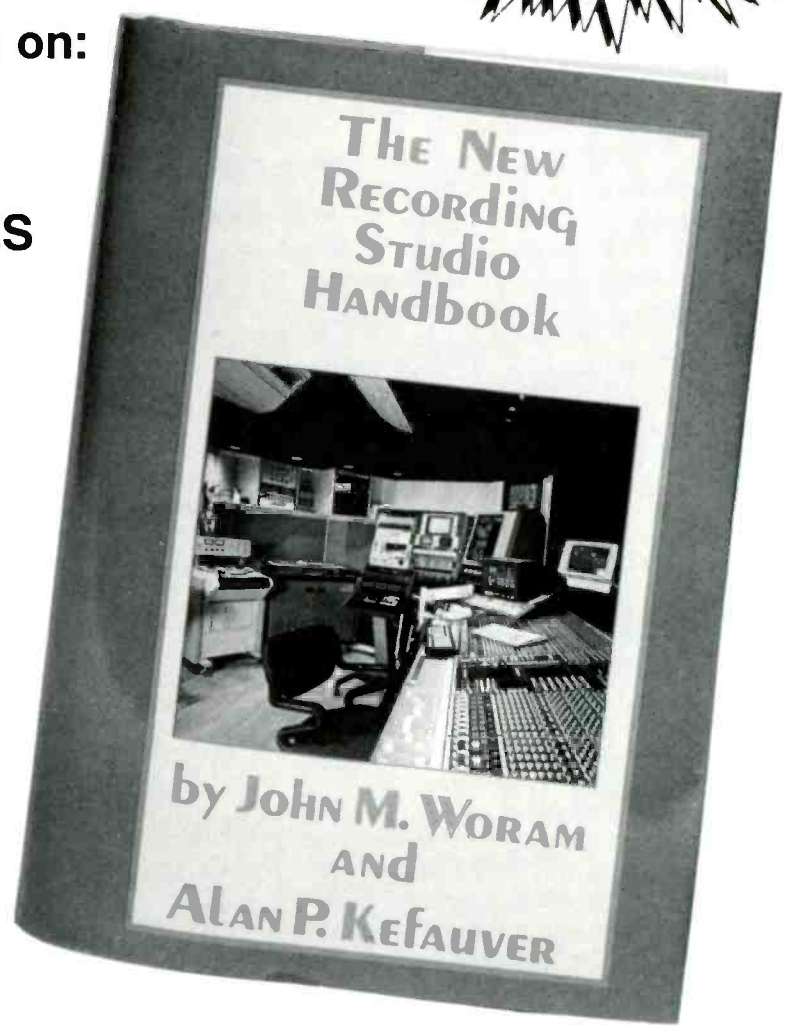

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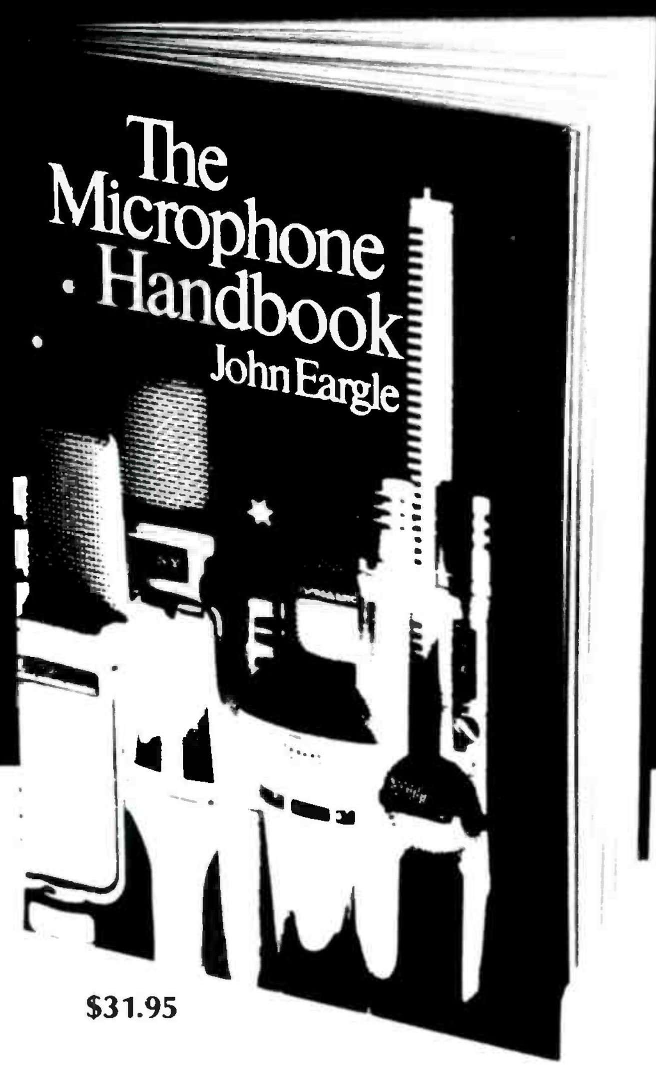

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ROBYN GATELY

The Sound Contracting Engineer

Guest Soundmen

• One of the fastest growing professions in live sound is engineering for acts that do not carry a sound system. The problem of creating a consistent "sound" while performing on a different system every night is difficult. Add to this the concept that what sounds good to one person may be barely adequate to another and you can see that the visiting soundman must not only be a well-rounded engineer, but a diplomat as well.

Being both a sound company owner and Judy Collins' soundman, I tend to experience both sides of the coin. As a result, I have seen where soundman either know how to relate to this job or don't.

Quite often the tone of the entire show is established by when and how the "guest mixer" advances the show. By advancing the show, I mean an initial conversation about two weeks prior to the date of the engagement. The point of this conversation is both to establish with the sound company that they can fulfill the needs of your act, and to establish *pleasant* working relations long before you meet. It is always helpful to speak to the actual engineer who will be working your performance.

In most cases, you will have already established some minimum criteria by including a *RIDER* in your contract. A rider is a written list of the needs of your act. Riders usually include sound requirements such as mic, console and speaker system needs, as well as information on the number of monitors, monitor mixes if required, etc. Also in the rider would be the lighting, food and dressing room requirements. The problem with riders is that most bands have made them "wish lists". They ask for the world. If they get half of it, then things are fine. What this creates is a situation where, sometimes, production budgets are pushed needlessly out of line. In some cases, promoters pass on acts because the production expenses make the show unprofitable.

THE INITIAL CONVERSATION

In the initial conversation, try to touch base on the highlights of your requirements. Do not attempt to give a mic list or any other list over the phone—the post office or fax machine is built for this.

I find it extremely annoying when someone calls me with a 40 channel mic list with mic brand and position, and the type of stand. This means a list of 120 items. People who do this have no consideration for my time and don't seem very organized. It certainly does not make me look forward to working with this person.

In the initial conversation it is important to *listen* as well as to talk. Sometimes, the sound company can really help you do your job if you aren't bull-headed. I recently had an interesting experience with someone who refused to listen during the initial conversation. An orchestra I work with sometimes brings in "pop" performers to headline the evening. The reason for this is that orchestras are under tight budgetary constraints due to the nature of their work. (In fact, during 1987, six of 38 full-time orchestras in this country went out of business!) As a result, they bring in pop performers to bring in badly needed funds; but can't pay for outlandish production.

My normal approach is to mic the performer's needs, add about 10 mics to "fill in" the strings, and maybe one or two solo mics. No more than 24 mics are used. (The reason for the 10 mics on the string is because pop mixing tends to be more "string" heavy than symphonic work. In addition, orchestras do not want everything mic'ed because they don't want to be loud. They want to add a pop singer to the orchestra.)

My agreement with this particular orchestra is to provide up to 24 channels and a sound system for their pop dates.After explaining the orchestra's artistic and budgetary constraints (including the 10 mics on the strings), I was given a list of 55 mics, with 12 on the horns, etc.

When I explained the financial considerations, I was told this was the minimum he could deal with. The orchestra was shocked by the proposed additions and informed me that the maximum number of mics allowed was 30. The soundman then read me a list of what he could deal with—41 mics. When I explained reality to him, he proceeded to tell me I was jeopardizing his show. He berated me for not following his "commands."

To be honest, I stopped taking his phone calls because I knew he was not considering anyone's viewpoint but his own. If he had listened the first time, he could have avoided a showdown with the orchestra and kept me out of the middle. As it was, I waited for him to arrive the day of the show, took him in another room and in my politest voice, read him the riot act. He behaved perfectly all weekend.

COMMUNICATION, POLITENESS AND RESPECT

I find it difficult to believe that anyone in this business can survive very long without a good amount of communication. However, beyond communication and politeness, the most important aspect of being a guest mixer is respect. It is assumed that the sound company will show you a certain amount of respect because you are the artist's representative; however, I continually see guest soundmen who have no respect for the sound companies position. It doesn't take a whole lot of intelligence to figure out that if you're going to play in outer Mongolia, that "Outer Mongolia Sound Inc." will probably not have every item you want.

If you want ABC equipment, and the best they have is XYZ, then you need to bring what is essential to you or deal with XYZ. The concept that you can't do without is not the fault of the sound company, but more likely your unwillingness to adapt to the reality of traveling without a PA.

In my travels, I may want an esoteric mic for a particular use, but if the only sound company in town has junk mics, then I'm going to deal with their mics or bring my own. Telling them that they must obtain what I need or else is not only unrealistic, but highly unprofessional.

The most important job of the guest mixer is to make the show happen and create a pleasant working environment for all of the parties involved. Over the years, I've encountered literally hundreds of shows where the promoter has already branded the guest soundman as a "turkey" (or worse) merely because of his attitude on the phone, and, as a result, has instructed me to give the guest soundman the absolute minimum, with the promoter's excuse always being that he can get away with anything by claiming "budgetary considerations."

ESTABLISHING WORKING RELATIONS

Assuming that you have established a pleasant working relationship prior to the gig, the entire tone of the day will be established in the first hour that you are there. Upon arrival you want to establish contact with the sound companies engineers. In this first contact it is important that you come off as being the friendliest soundman who ever lived. It is important that you establish a working relationship that makes the sound company feel like they are working with you, not for you. If they are working with you, they will go out of their way to try to give you their best; if they are working for you, they will only do what is absolutely necessary. This is a crucial difference because even with the best equipment, attitude is the one part of the equation that changes a great gig to a mediocre or poor one.

In this initial contact, you want to establish that the basics that you need are there. The sound company is probably going to tell you about a couple of substitutions that they had to make. The worst thing you could say is, "Is that the best you can do?", because now you have cast doubt on everyones competence (including your own). If the sound company can fulfill all of your requests within their budget constraints, they usually will.

It is at this point that you encounter my favorite situation. It is called "reality strikes". You, as guest mixer, now prove how well your parents brought you up by what your reaction is. You would be amazed at how many people throw temper tantrums, as opposed to being gracious and just dealing with it.

After your initial contact, you want to hear the system—first the house and then the monitors. It is best to hear the system utilizing a compact disc by the artist you are working for. Try to hear the system without any graphic EQ and work with the house soundman to tweak the system to it's maximum capabilities without the EQ. Some systems can be made to sound amazing without the EQ, others require a decent amount of EQ to make them sound right.

At this point, there are a few basic rules you should be aware of as a guest. It is expected that you are going to have to make adjustments to a console's input gain, EQ, sends and fader level. At the same time, do not touch the master fader setting, limiter or crossover settings without permission or sound company personnel to assist you. I know of too many soundmen (who thought they knew it all), who have destroyed parts of the PA because they have had no respect for the equipment. As a result, the act has ended up shelling out unnecessary dollars. Being a guest soundman is not permission to do whatever you want.

"MAKING DO"

No matter how much experience you have, you have been brought in by the act because they trust you to get their "sound" *with whatever is available*. If what's needed is not available, then your job becomes one of "making do." It is more important that the artist gets through the gig and gets paid, then that you get star treatment.

After you have checked the house and monitor systems and made sure that they sound right, your first job is to make sure that your vocal mics are put up and checked out immediately. If this is done, then if your act arrives early at least they can get started.

Clearly, it is the sound companies job to layout and explain their equipment so that you may understand its operation. If there are parts that you do not understand, don't be afraid to ask questions, but don't become a pest. A gig is not the place to try and learn massive amounts of information.

The sound company will usually set up the mics for you, but the more assistance you provide, the better the day will go. Don't go beyond the limits of your knowledge, but if you can help wire up the stage, you can speed the day up immensely.

As you mix your soundcheck and show, look for help from the sound personnel regarding your mix, use of their board, and limitations of the system. It is of considerable importance that you get into the audience to hear your mix at least once, in order to make the necessary adjustments. It is better to make it sound "good" all over, than to make it sound "great" in one place.

At the end of the show make sure you thank the engineers for their assistance. (If you are just starting out and have learned a lot, it wouldn't hurt to give them a hand packing up.) But, no matter what your experience may be, the important achievement is that the audience was entertained, and that the artist was able to achieve his best because you did your job correctly. That's the real satisfaction in this business. Don't ever forget that. db

Broadcast Audio

Broadcast Television Stereo: The BTSC System

• Broadcast television stereo in the United States has just celebrated its fifth birthday. After this relatively short period, about 99 percent of the United States television viewing audience is within the coverage area of at least one television signal. About 25 percent of U.S. television households have stereo television reception capability, attesting to the success of this expansion of the television audio service.

THE BTSC SYSTEM

How does the BTSC system work, and how does it differ from the FM stereo transmission system that has been used in the United States for over 25 years? Standard monophonic television sound transmission employs frequency modulation of the aural carrier with a single audio signal whose frequency range is 50 Hz to 15 kHz. The aural carrier is modulated to a peak deviation of ±25 kHz. The aural carrier is located 250 kHz below the top edge of the television channel, and about 300 kHz above the upper edge of the visual signal. This leaves considerable unoccupied bandwidth beyond that taken up by monophonic deviation. The BTSC system uses this spectrum space to add subcarriers to the aural service that may provide stereophonic sound, an additional program-related or non-program-related channel, and a professional channel for private use by the li-

Figure 1. BTSC system aural baseband.

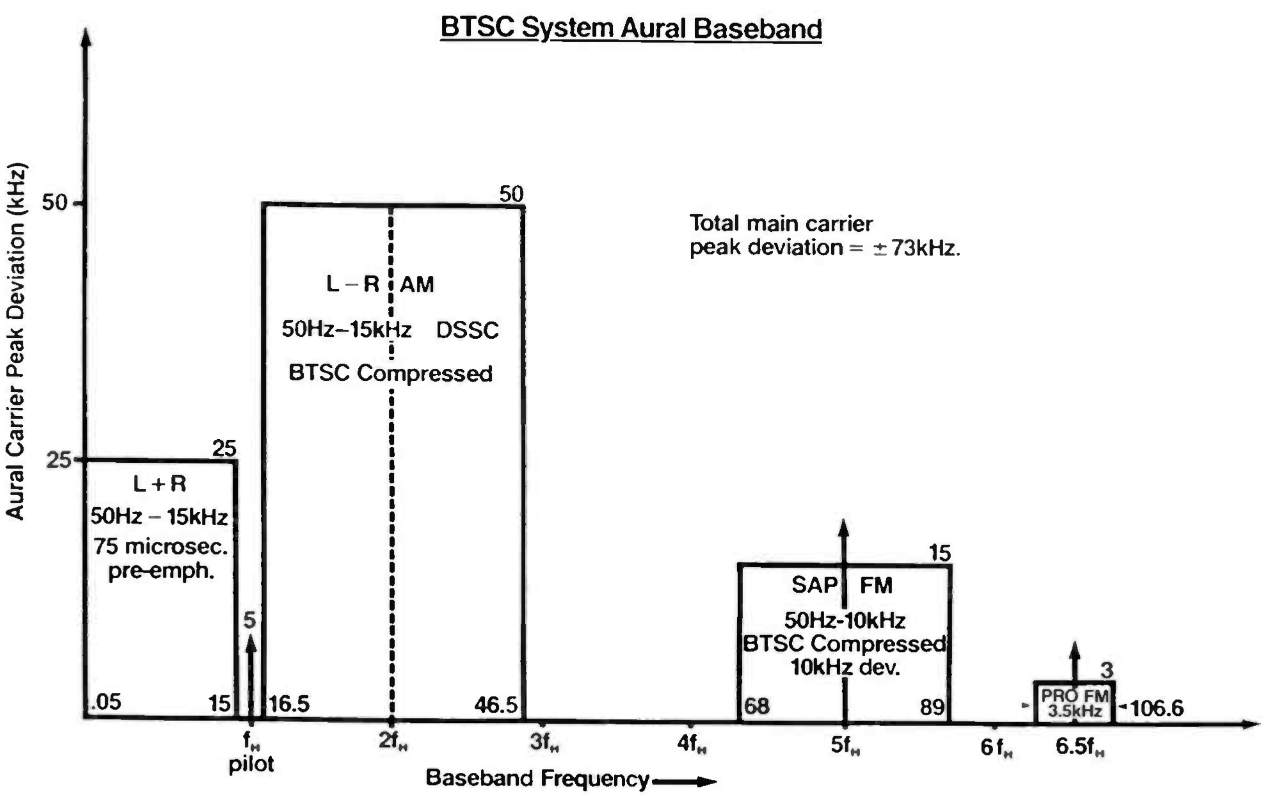
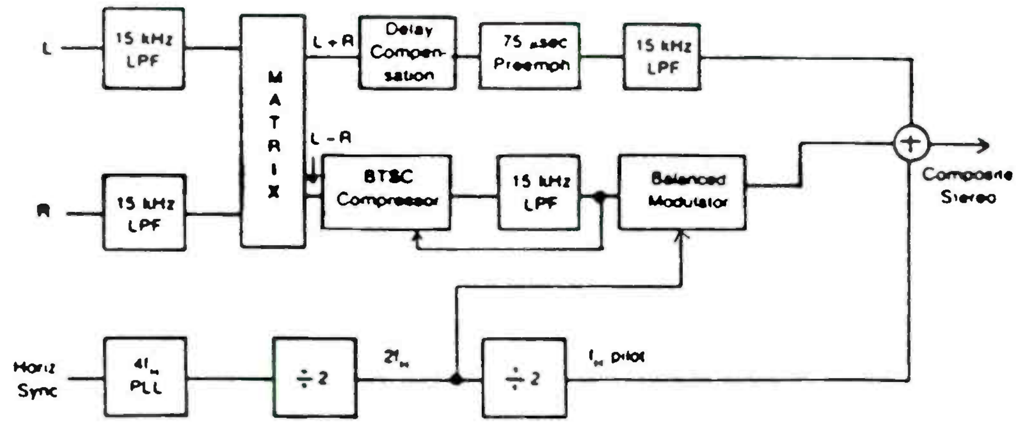


Figure 1

Figure 2. A representative BTSC stereo generator.



censee. All these extra services are realized without exacting any penalty in monophonic performance or coverage.

Figure 1 details the use of the aural spectrum by all elements of the BTSC system. The 50 Hz to 15 kHz mono sum or L+R signal incorporates 75 microsecond pre-emphasis and deviates the aural carrier ±25 kHz. These are the operating parameters of the standard monophonic television sound signal, which insures compatibility with monophonic receivers. For stereo operation, a pilot tone of 15.734 kHz is transmitted. The pilot is at the same frequency as the television horizontal scan rate fH, and is locked to the horizontal synchronizing signal to prevent beat frequencies between pilot and synchronizing signals. The pilot tone is a sine wave which deviates the aural carrier ±5kHz. The pilot serves two functions. It lights the stereo indicator on the receiver to show that stereo is being received, and it synchronizes the receiver's stereo demodulator with the transmitter's modulator to enable proper reconstruction of the left and right channels.

Figure 2 is a functional block diagram of a representative stereo generator. The generator adds the left and right channel stereo signals to obtain their sum, the L+R signal, which modulates the main channel. The right channel is also subtracted from the left channel, yielding the L-R or difference signal, which after processing modulates the stereo subchannel, a double sideband, suppressed carrier, amplitude-modulated subcarrier centered at $2f_H$ or 31.468 kHz. Processing includes compressing the difference signal according to the BTSC noise reduction algorithm, and then doubling its voltage. The stereo difference channel may deviate the aural carrier up to ±50 kHz peak. In practice, sum-and-difference interleaving may be expected to keep this deviation below its maximum, and in fact the total deviation of the aural carrier by both sum and difference signals is limited to ±50 kHz, a limit which is not exceeded by most program material. It is recognized that in practice certain types of stereo signals, notably those in which only one stereo channel is driven, may cause this limit to be exceeded for brief periods.

TV VS. FM

Television stereo and FM stereo differ in some important respects. The most immediately apparent difference is that the pilot and stereo subcarrier frequencies are 15.734 kHz and 31.468 kHz respectively, as opposed to FM radio's 19 kHz and 38 kHz. These frequencies are multiples of fH and enable their signals to be locked to the horizontal scan reference.

The TV stereo subcarriers represent additional modulation of the aural carrier, allowing multichannel operation with no degradation of the monophonic signal. In the FM stereo system, monophonic transmission uses the full ±75 kHz carrier deviation, while the additional carriers required for stereophonic operation must fit into the same occupied bandwidth.

This means that when stereo is being transmitted, the monophonic channel's modulation must be reduced to accommodate the stereo subcarriers. This, plus the fact that the stereo subcarrier is amplitude modulated and located at a higher frequency than the main channel, with its attendant triangular increase in noise, results in a sacrifice in signal-to-noise ratio of at least 20 dB, and a concomitant reduction in coverage area, when stereo is transmitted as opposed to mono.

The fact that no monophonic modulation is given up for TV stereo transmission, combined with subchannel noise reduction, yields stereo-to-noise ratio and coverage for television audio that are essentially identical to those achieved in mono.

The most radical difference between television and FM stereo is the presence of noise reduction in the stereo subchannel. It is a companding system, in which the stereo difference signal is compressed in a specific way before transmission, and expanded in exactly complementary fashion at the receiving end. The encoding process involves broadband compression and a combination of fixed and variable pre-emphasis, the amount of variable pre-emphasis being dependent upon frequency content of the signal present.

The net result of noise reduction encoding is a significant increase in the average signal level and the average high-frequency content of the difference channel signal through the transmission medium, which upon expansion produces a much more favorable signal-to-noise ratio.

The BTSC multichannel system is capable of providing two additional channels, although neither is implemented to nearly the degree stereo is. There is a subcarrier centered at $5f_H$ or 78.670 kHz called the second audio program or SAP channel. The SAP channel may be used for any program-related or non-program-related transmission, and is most often used, when implemented, to provide a simultaneous translation of the program audio in a second language.

The SAP subcarrier is frequency modulated with a maximum deviation of ±10 kHz, and its deviation of the main aural carrier is ±15 kHz. The SAP subcarrier is locked to the horizontal sync frequency as are the stereo subcarriers, and it is processed using the same noise reduction as the stereo difference channel.

Its modulating frequency range is 50 Hz to 10 kHz. The SAP channel's noise reduction system and high injection level assure that it has a quite acceptable signal-to-noise ratio, even at an appreciable distance from the transmitter.

THE PROFESSIONAL CHANNEL

There is also provision for a professional channel, to be used by the broadcaster for such applications as telemetry or signalling. The professional channel subcarrier is centered at $6.5f_H$ or 102.27 kHz, is frequency modulated with a maximum deviation of ±3.5 kHz and deviates the main aural carrier ±3 kHz.

The audio frequency range of the professional channel is 50 Hz to 3.5 kHz with 150 microsecond pre-emphasis for voice, or 0 to 1.5 kHz with no pre-emphasis for data. This is obviously not a high fidelity channel, and its high carrier frequency and low injection level limit its useful range. Very few television stations have implemented the professional channel.

The BTSC system provides high quality stereophonic sound to the television viewer, and is capable of furnishing audio channels in addition to stereo.

The number of stereo television signals and the number of stereo television households attest to the success and viability of the stereo service on television.

If the growth and acceptance of stereo television continues as it has since 1984, it will soon become the rule rather than the exception throughout the U.S. television industry. This is already the case at NBC, where the only block of monophonic programming left is the daytime schedule, and the daytime stereo barrier has been broken there with a new soap opera in stereo! db

JOHN BARILLA

Hot Tips For The Home Studio

• You are now patched into a listening-post for electronic cottageers. In each issue of db, you will receive a potpourri of practical information conjured up by home studio operators like yourself. From production techniques to money saving DIY projects to studio problem-solving, this column is designed to be your bulletin board. So keep those creative ideas coming in. If your tip is a hot one, we will definitely print it. Send your savory studio secrets to me, John Barilla, at P.O. Box 585, Wheatley Heights, NY 11798. (Include a paragraph about who you are and what you do, so we can tell the readers something about your background).

1. DIY STUDIO MICROPHONE WINDSCREEN

Remember the "We Are The World Video"? Millions of naive viewers were left wondering why people were singing into those weird paddle-shaped objects—which unbeknownst to them, were state- of the-art windscreens. Well, Rick Bieder, engineer and owner of RBS Recording in Massapequa, took one look and decided that he wanted one for his studio.

When he found out the price tag for these units though, he decided he could probably fabricate one himself for a fraction of the cost. He did and it worked beautifully.

Before we share the details of this simple, money-saving project, let me tell you something about Rick Bieder. Rick is an electronic cottageer par excellence. As producer/engineer with Alpha/Omega Productions, he has been turning out a steady stream of dance music for well known labels like Emergency and Profile Records. And it's all been recorded right in the basement of his house. If you frequent discos, you've probably heard his recent opus for Profile, a tune called "Baby Be Mine" by Desiree.

Back to the windscreen. This type of stand-alone windscreen is of value to the engineer for several reasons:

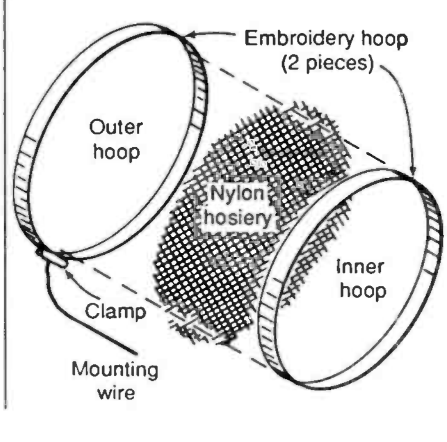
a) It augments the standard microphone pop filter. Those potent "p" sounds ("plosives") as they are called by linguists) move a lot of air and a second diffuser placed between the vocalist and microphone can greatly reduce the impact upon the mic capsule.

b) It helps establish a fixed distance between the vocalist and the microphone, thereby evening out recording levels and minimizing the proximity effect. (Proximity effect is the sometimes unwanted buildup of bass frequencies that occurs when a vocalist gets too close to the microphone). Basically this simple gadget can do a lot to help you get a more professional vocal sound. So here's Rick Bieder's idea on how to fabricate one inexpensively.

Go to your local fabric store and purchase an embroidery hoop. Made from either plastic or wood, these devices are made from two concentric rings with a locking clamp. They are generally used to hold a section of fabric taut while sewing a design. Simply cut a section off the leg of some nylon pantyhose (sheer fine mesh is best). Cut it just large enough to stretch over the inner ring of the hoop. Hold it taut on a flat surface while you press the outer ring over the inner ring and tighten the clamp.

Mounting the unit can be achieved in various ways. Bieder's original brainstorm led him to unscrew a mic clip exposing the inner threads. Normally there is a notch across the top of the inner barrel. He inserted the clamp from the hoop in the notch and found that it mated perfectly. Personally, I think he got lucky here. I made one of these gizmos myself and had to find another way to mount it. I simply got some stiff wire, bent it in an appropriate shape (like a gooseneck) inserted one end in the clamp of the embroidery ring and epoxied it in a stationary position. The other end I simply bent around the mic stand extension and taped it into place. In any case, you'll have a very effective windscreen worth about $50 on the market, for less

Figure 1. A home-made proximity-effect-reduction screen.



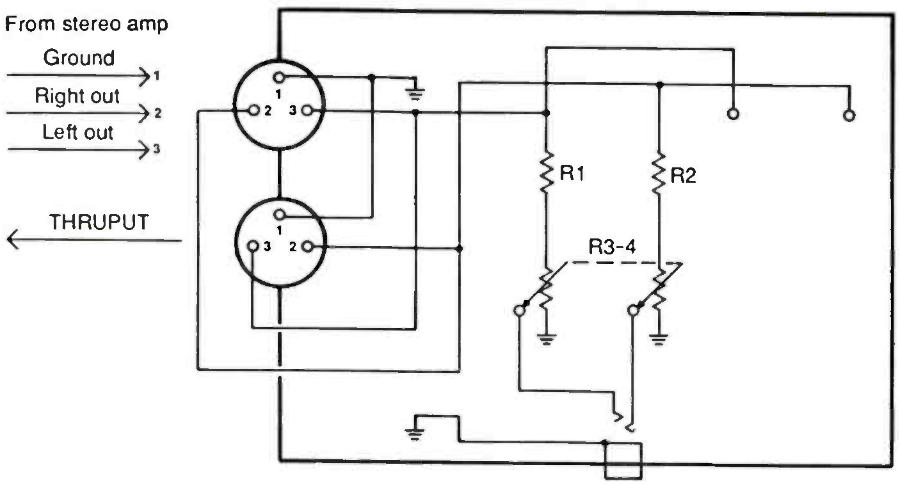


Figure 2. An easily built, high-quality headphone amplifier.

than $5 worth of parts (pantyhose included).

2. THE PARAMETRIC POP REMOVER

Speaking of pops...What can you do when you discover that some high energy "plosives" have been recorded on tape. Somehow it eluded you while you were tracking, but now it's mixdown time and it's too late to redo the vocals. Much to my chagrin, this situation happened to me recently. But much to my credit, I successfully resolved the problem, and I share that solution with you now.

Here was my logic. I listened closely to the critical "p" sounds from the track. They sounded like a controlled thud- a miniature explosion (ergo the term, "plosive"). From the microphone's point of view, a short blast of low frequency (therefore long wavelength) energy temporarily overshadowed the normal harmonic structure of the voice. If I could find the exact offending frequency, perhaps I could notch it into oblivion. It was certainly worth a shot, so I patched the vocal into a parametric equalizer.

To find the frequency range of greatest offense, I set the Q control of the parametric to home-in on a fairly narrow bandwidth, boosted the gain all the way, and carefully swept the frequency control from 40 to about 200 Hz until I found a point of resonance. (Needless to say, I had to loop the tape machine around the offending plosive so that I heard "puh...puh ...puh..." over and over again.) Once I located the offending region, I turned up the Q (narrowed the bandwidth) until I had about the most pronounced (i.e., offensive) sound I could get. Having found the "sore spot", I then turned the gain in the opposite direction till the "puh" was virtually gone.

I wish I could say that was the end of the story, but it wasn't that simple. It got rid of the plosives alright, but the rest of the vocal track now sounded cold and small - not what I had intended at all. If only I could turn the parametric EQ on for just the plosives, but off for the rest of the tune. Why not? There were only four bad pops in the first verse. Everything else was copacetic. With a little coordination, I knew it could be done. I tried punching the EQ in and out with the switch, but 15 dB of cut proved to be too noticeable of a change. The "hard-edge" of the switching action was a giveaway. What was needed was a little ramp time, a more natural cross-fade to fool the ear. I was about to get my day in court. Would the concept work? I grabbed the gain pot on the parametric. Holding at zero for the majority of the verse, I awaited the first plosive, ready to plunge the gain to – 15 dB a split second before the attack and ride it back up immediately after it had passed. Happily, with a little practice, it worked flawlessly. In summation, my advice is this. Avoid pops like you would the plague. Build Rick Bieder's homemade windscreen, and always monitor your vocals (twice!) in solo. But if perchance, having done all diligence, there is later discovered some unexplainable pops on tape—take heart! Contrary to conventional wisdom, there still may be some recourse if you possess a parametric and fast reflexes.

3. BECHER'S NOISE REDUCTION AUGMENTATION SYSTEM

Jim Becher, a composer and itinerant studio musician from East Northport, NY tells of a rather un-orthodox technique for enhancing the signal-to-noise ratio of his studio. Becher operates a rather elaborate MIDI oriented studio utilizing scads of keyboards monitored through two cascaded 16 channel mixers, and controlled by an Atari computer. For "live" analog tracks or "layers" of sound from his bank of synthesizers he records on a Tascam Porta-1 four-track cassette recorder.

Having gotten used to the purity of the direct synthesized sounds, Becher experimented until he found a way to get the optimum quality from his Porta-1. Obviously, he is a graduate of the Seat-Of-The-Pants School of Engineering (of which I am also an alumnus), because his methodology is certainly non-conventional. Nonetheless, judging from the superior quality of his tapes, one would have to agree that he is doing something right.

His system involves two rather simple steps, each of which work independently to enhance the sound, but which because of the built-in dbx noise reduction in the Porta-1, seem to work synergistically as well.

Step one simply involves taking advantage of the varispeed option on the Porta-1. Normal speed on the unit is standard 1 7⁄8 in./sec. cassette speed. Knowing that frequency response on playback is related to the speed at which the program is recorded, Becher opted to record his tracks with the varispeed cranked all the way up (+15%). While sacrificing the liberty to increase tape speed later on, Becher found an audible improvement in high frequency response.

But just how much brighter was the sound at this slightly elevated speed? (The Tascam user manual stated that at normal speed the machine was rated at + or –3 dB out to 12 k, which means in plain English that the high end response is a little quirky). Becher decided he might be able to quantify his subjective assessment by recording high frequency tones at zero VU (dbx out) for both

JOHN BARILL/

Hot Tips For Akai 12-track Users

•In the early 1970s a noted jazz musician, who had done much to popularize the Fender Rhodes electric piano, was asked in a magazine interview what he thought of the instrument. The interviewer, who I'm sure expected a word of unqualified praise for the instrument, received instead a rather cryptic answer. The musician's reply went something like this: "The Fender Rhodes? Oh yeah, nice idea. Too bad they never finished it."

Somehow I never forgot that quote. Back then I too had been an itinerant musician, lugging my Fender Rhodes from gig to gig, up and down the east coast. I identified with that musician's ambivalence towards what was then considered to be a wonderfully expressive but quirky new instrument.

I'll also never forget how many hours I spent adjusting the tines—the steel rods that vibrated over a magnetic pickup to create the unique sound—only to have one break in the middle of my best solo. That, of course, was not an unusual event, but a periodic occurrence that Rhodes players just learned to put up with for the sheer joy of playing such a responsive instrument. It seemed a valid trade-off.

What has all this got to do with the Akai 12-12 and 12-14? Well, some fifteen years after my love affair with the Fender Rhodes piano, my current passion is for this marvelous 12-channel, 12-track mixer/tape recorder. But my feelings for the two units are quite equivalent: Overwhelming amounts of love, pride, and loyalty, alloyed with sufficient annoyance to make me want to requote that famous jazz musician: "The Akai 12-track? Oh yeah, nice idea. Too bad they never finished it."

But lest you think that I'm about to launch out on some criticism, let me state this out front: While the Akai has its quirks, all of them are surmountable and definitely worth the effort. It's difficult to imagine how you could purchase a more professional sounding multi-track for anything in this price range. For under $6,000 (actual market price) you can have 12 tracks of very high quality analog audio (94 dB S/N ratio with built-in dbx Type I noise reduction). Since they have a *hidden* time code track, *all* twelve tracks can be utilized for your program. And two Akai 12-14s can be linked together with something like a Fostex 4030/4035 Synchronizer to give you, at about $14,000, probably the world's least-expensive, self-contained 24-track recording studio. All in all—judged by any standard—the Akai 12-track is still an incredibly sweet-smelling deal. But the rose does have a few thorns.

Another Akai user I know had a conversation with one of the Japanese engineers who designed the unit, and confronted him about these *thorns*. The engineer was rather cavalier about the issue, saying that one had to be "brave" to operate the Akai 12. In other words, it was not a tool for the timid. You had to be a real aficionado to get the most from it. I have found that to be absolutely true. After operating my Akai 12-12 for four years, I have learned through perseverance, trial-and-error, and consultation just how to get the Akai

EDITORIAL CORRECTION!

We apologize for a puzzling bit of text in our July/ August issue in John Barilla's *Hot Tips For The Home Studio*. Regarding item #4, "DIY Headphone Distribution Box," the resistor designations in the text do not agree with the designations in the circuit diagram. Originally, they did agree, but the diagram was a re-labeled in order to make it more readable. Unfortunately, the text was not changed to comply. The circuit diagram *is* correct, but the text should read as follows (third column, second paragraph):

"The variable resistor (potentiometer) R3-4 controls the level of the sound to each set of headphones. For convenience, the left and right sides are ganged together on a concentric shaft so that both sides go up or down in volume by turning a single knob. R1 and R2 are thrown in just to make sure there is always sufficient resistance to prevent the amp from shorting to ground—even when the variable wiper R3-4 is all the way up, hence functioning as a straight wire."

normal speed and elevated speeds and noting the difference in playback level. Feeding in a 12 k tone he found a 3 dB improvement at the higher speed, and at 10 k there was a greater than 4 dB improvement. Convinced now that his ears are telling the sonic truth he routinely records all his productions on maximum varispeed thereby superseding the machine's published specs.

Not satisfied with that small (but noticeable) improvement, Becher decided to try to diminish the residual high frequency hiss by using a primitive noise reduction technique he "stumbled" upon years ago as a high school student experimenting with electronic music. He had a rather low-quality tape recorder whose specs should have been rated as noise-to-signal ratio, rather than signal-to noise. Serendipitously, he found that if he radically boosted the high-end on his mixer while recording, and reduced it by a similar amount upon playback, he could significantly reduce the apparent noise on tape, while maintaining the fidelity of his music. Of course the audio cognescenti among us understand that Jim Becher's adolescent discovery was nothing more than a variation of the standard pre-emphasis/ de-emphasis curve utilized in the circuitry of all tape recorders.

Nonetheless, his application of this technique—"piggy-backed" as an addition to internal equalization is interesting, and according to Becher, quite effective at reducing high frequency hiss. Fact is on his side, for as he records his high frequency-loaded sounds, the residual tape noise remains the same. Another words, there is significantly more high-end sound on tape then required, but the noise floor—(much of which is made up of similar frequencies)—remains a constant factor. When these high frequencies are rolled-off during mixdown, a lot of the residual noise is reduced along with it.

Since Becher uses the inboard EQ of the Porta-1 for this (which is a high frequency shelf from 10 k on up), whatever record emphasis he chooses can be accurately "undone" on play back by mirror imaging the gain control. (All aesthetic EQ decisions are made from another console situated downstream from the Porta-1.)

Becher also reports that when he increases the high frequency on input, the VU meters indicate decreasing levels with dbx noise reduction on, thereby enabling him to record at a hotter level. On this issue, I am convinced that there is a synergistic effect going on between Becher's initial high-end boost cascaded into dbx's own pre-emphasis circuit. The high-end pre-emphasis of dbx is designed to selectively key the compression circuit which follows. Becher's idea of adding an extra dollop of highs apparently hyper-compresses the signal, allowing him to record at unusually high levels without any audible distortion on tape.

While this method may be a little cumbersome for most of us, it works well for Becher. Frankly, you've got to send kudos to a guy who gets more from his equipment than the manufacturers ever intended!

4. DIY HEADPHONE DISTRIBUTION BOX

Many electronic cottages are sophisticated when it comes to MIDI implementation, but oftentimes they are lacking in basic audio "nuts 'n' bolts". Things like headphone systems seem to get shortchanged in the average home studio. Perhaps it's the last thing anyone thinks of, so it wasn't figured into the budget. Or perhaps the "MIDI generation"—(those amongst us who grew up on synths and sequencers and missed the rush of recording in a first class pro studio)—never felt the need of performing more than one track at a time.

But sooner or later, there comes a time when live tracks become a desirable commodity. When the necessity hits, it's kind of funny to see how people scramble to jury-rig a headphone system. Out come the Y-cords, long extensions, and adaptors, along with noise, hum, and shorts. One headphone goes out, and maybe even the whole system goes down. The next day, they go out and buy a distribution box.

Can't say I blame them for not wanting to part with their money though. You could buy another synth module for what you'd pay for a couple of these boxes. And when you realize what simple little devices they are, you'll probably want to spend a rainy afternoon building one for yourself and pocketing the change.

I received an idea for an easy-to-build headphone distribution box from Steve Kureczco. Steve is a seasoned recording and maintenance engineer, who has also taught studio electronics and done design work for various manufacturers. He has his own electronic cottage facility in Babylon, NY—a studio he has provocatively named, Ground Zero.

In *Figure 2.* you will see a wiring diagram for a simple, inexpensive distribution box. What you are looking at in *Figure 2* is simply a voltage divider. What makes it a distribution box is just the way that it's hooked up. Each side of the circuit (left or right stereo) contains only two resistors: one regular and one variable. The variable resistor (potentiometer) R2, controls the level of the sound to each set of headphones. For convenience, the left and right sides are ganged together on a concentric shaft so that both sides go up or down in volume by turning a single knob. R1 is thrown in just to make sure there is always sufficient resistance to prevent the amp from shorting to ground—even when the variable wiper R2 is all the way up, hence functioning as a straight wire. The diagram describes one stereo headphone output, but you can stick about four of them in one box by wiring the additional channels in parallel. With the XLR "thruput" you will also be able to cascade another box if you require four more outputs. Wire this stuff up in a metal box (a cast aluminum box is easiest to work with) and you're ready to rock!.

P.S. Well, nobody caught me (that I know of), but I'm gonna 'fess up anyway. Remember HOT TIP #1 from the May-June issue? You know, "Getting A Warmer Sound From Your Microphone?". It's a great technique. I use it a lot. There's just one problem. I said you could do it on a Yamaha SPX90, but the truth is you can't. The "level or percentage of first reflection" is a parameter that's not available on the SPX90. However, it is available on its cousin, the REV7 (probably the Yamaha's new REV5 as well). I use both units so often, I got them mixed-up. Sorry about that. db

Ad Ventures

BRIAN BATTLES

• "Truth in advertising" is a common expression. But what about "truth in creating advertising?" As in any high-pressure situation, the combination of creativity and subjective judgement can make it hard for some people to accept your innovative ideas. When deadlines, errors, or misunderstandings cause problems, a true professional draws from an astounding repertoire of explanations and excuses. After several years of experience in radio commercial production, I've noted that creativity is nowhere more conspicuous that in the list presented below. Here is a semi-scholarly compilation of The Most Often-Repeated Excuses in Radio Commercial Production.

Commercial Producers to Clients:

1) We'll fix that in the mix.

2) This is just a demo; the finished product will be better.

3) Everybody says that.

4) You just heard it once; listeners will hear it all the time.

5) I stole this idea from a Clio winner.

6) They must have gotten a bad dub.

7) They stole this idea from me first.

8) The client made me do it that way.

9) I guarantee this will pull in more customers than you've ever seen.

10) That may work on TV, but it won't on radio.

11) I was almost nominated for a Clio once.

12) This won't cost you anything; it'll save you money.

13) I rewrote this copy six times before we cut it.

14) I can't just edit that part out without changing the whole spot.

15) The dubs will definitely be at the station before the first one's scheduled to run.

16) I sent it to your Fax machine.

17) I never got it on my Fax machine.

18) We worked on this all night.

19) All my other clients are waiting because I wanted to do your spot first.

20) No problem.

21) We did it this way to save you money.

22) Everybody I've played this for thinks it's great.

23) It's hard to believe that you always write your own copy.

24) It won't sound like that on the air.

25) These studio monitors always exaggerate every little sound.

Radio Station Account Executives to Clients:

1) This is the best deal in town.

2) Any time on our station is prime time.

3) Everybody listens to our station.

4) Not just anybody listens to our station.

5) All the other stations in town copy us.

6) Our rate card's not negotiable.

7) This price is even lower than the national accounts pay.

8) You'll have exclusive sponsorship of this program.

9) Just because you don't listen to us doesn't mean your customers don't.

10) If you sign now I'll lock you into that price.

11) We don't trade.

12) I'd rather be in Sales than be a disc jockey.

13) Call me whenever you have a problem and I'll straighten it out.

14) We'll come up with a really good promotion to help you out.

15) I'm going to work with the copywriter personally to make sure your commercial comes out right.

Clients to Radio Producers:

1) I already faxed you the copy.

2) I thought we agreed that you would cut this spot on spec.

3) I stress quality in my store and in my ads.

4) I'll call you tomorrow.

5) My wife/partner/boss has to hear it first.

6) We're already getting great results with our current ads.

7) We tried that before and it didn't work.

8) I've had a lot of experience in broadcasting myself.

9) I never said that.

10) The station gives me better spots than this for free.

11) I just want you to make one change in the copy.

12) I wouldn't still be in business if I didn't know all about advertising.

13) People love the commercials I've been producing myself.

14) If I like the commercials you make for me, I'll give you lots of referrals.

15) The check's in the mail.

Please feel free to drop me a note c/o **db** to add to this list.

You can also leave me messages via the following: PC-LINK (id WA1YUA), The Source (BFW623), or on a variety of non-commercial telephone Bulletin Board Systems throughout North America. Try a National Echo, such as "Ham Radio Conference."

SON OF SHAMELESS SELF-PROMOTION

Thanks to everyone who's already ordered my audio cassette training program, *How To Produce Great Radio Commercials*, is becoming very popular. This lively four-cassette package is full of tips, hints, and examples of creative radio production. It costs only $99.95, including domestic shipping (U.S. finds only.) To get your copy, please send your check or money order to:

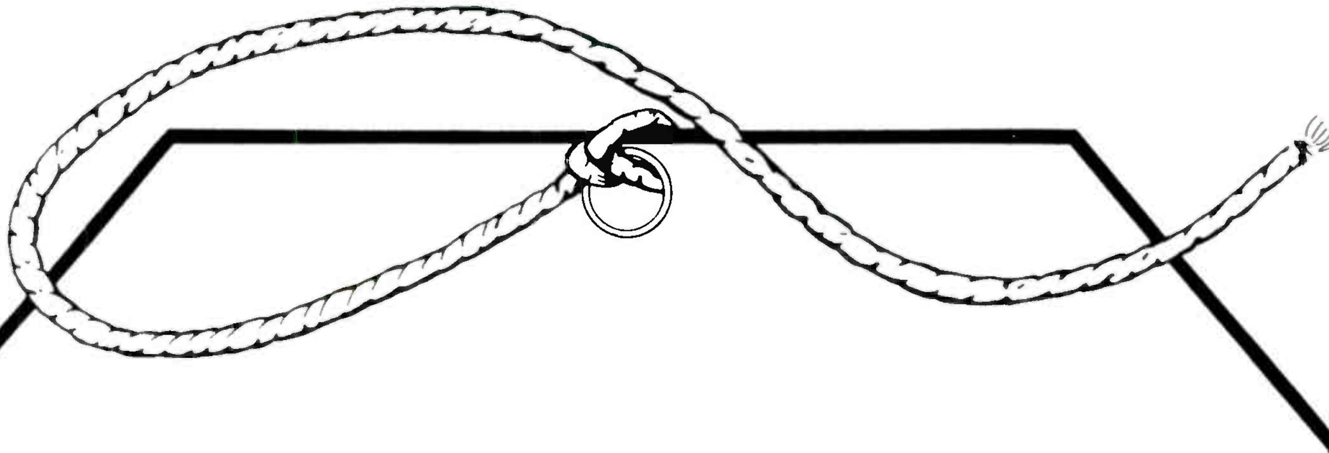
Porkpie Productions

P.O. Box 176

Colchester, CT 06415

Keep those letters and demo tapes coming! And if you're ever in southern New England, give me a shout on the Amateur Radio 2-Meter FM band. I usually monitor 147.09/.69.

Now, get out there and talk to some potential new clients! db



db

serving: recording, broadcast and sound contracting fields

Buyer's Guide—Microphones (including Wireless Mics), 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. In addition, there are paragraph-form Guides to Recording Tape and Tape Accessories. Manufacturers' addresses conclude the Guides

As usual, please be aware that we attempt to contact every manufacturer, but not all are cooperative or prompt enough for our necessary deadlines.

Model	Type	Pattern	Frequency Response ±dB	Impedance, Ω	Sensitivity, 1k	SPL at % distortion	Dimensions-in., L, D, W	Weight, oz	Finish	Connector	Price, $	Features
AKG ACOUSTICS, INC.												
C414 VLS	cond	multi	20-20k	180	38	134 0.5	5.6 1.8 1.4	11	black chrm	XLR	-	FET cond. with large diaphragm capsule,provides four selectable patterns with 3 bass-rolloff switch, phantom power.
C460/ CK61 VLS	cond combo	card	20-20k	120	43 0.05	134 0.8	6.8	4.9 black	satn	XLR	-	Switchable bass roll-off, attenuator — 4 positions.
C522	cond	stereo x-y	20-20k	300	40	128	8.5	10	satin black	5- pin	-	Hand-held or boom x-y stereo mic. Internal rechargeable battery or 8-52 V phantom power,ideal for ENG/SNG.
C535EB	cond	card	20-20k	200	43.1	130 1	7.2 1.8	11	satin black	XLR XLR	-	Hand-held studio quality cardiod mic, 4 position bass roll-off, output padding switch,8-52 V phantom power.
C562	cond	bound layer	20-20k	600	33	130 1	.375 6.3	2.1 2.1	satin black	XLR	-	Boundary layer mic mounted on flat round plate. Screw holes for mounting, phantom powering updated.
C426	multi	stereo	30-20k	200		133 0.5	9.25 1.65 1.3	1.04	satn black chrm	12 pin DIN	-	New version of C422, newer technology for improved response.
C1000S	elect cond	card	50-20k	200		137 0.5	8.7 1.3	4.7	dark grey	XLR	-	Hand-held electret condenser phantom power or 9 V.
D321	dyn	hypr card	40-20k	300	57	128 1	7.3 1.9	11.7	dark grey	XLR		Hyper-cardiod mic designed for tough vocal situations. New patented handling noise compensation system elastically suspends magnet to virtually eliminate all handling noise.
ALTEC LANSING CORPORATION												
D645	dyn	card	80-15k	150	56		6.6 1.4	6	beig matt		380.00	Offers an outstandingly smoooth frequency response with great naturalness and accuracy of inflection.
D658	dyn	card	80-13	155	61		6.6 2.16	8.9	grey		100.00	Moving coil dynamic, designed for the performer. Head design provides wide linear frequency response allowing greater gain-before-feedback.
C644	cond	card	50-18k	150	50		7.5 1.95	12.	black	A3M	220.00	Ideal for speech or singing reinforcement systems where there is a tendecy towards feedback.
C649	cond	card	40-18k	150	45	1.06	6.94	8.0	beig		390.00	Wide range,high input Z without distortion. A true cardiod pattern. Designed for voice.
D90P	dyn	omni page	50-15k	150	58		7.5 1.6	16	satn chrm		236.00	
D91P	dyn	omni close	180- 10k	150	60		3.6 2.5	4.5	char grey		164.00	
D646	dyn	super card	80-15k	150	57		7.0 1.9	8.0	slvr	A3M	410.00	
654A	dyn	card	50-15k	200	56		7.25 1.875	8.0	satn nckl	A3M	240.00	
AMS/CALREC												
CM 1050C	cond	card	30-20k 3	200		130 0.5	5.5 .87	4.0	black satn	XLR	215.00	Fixed capsule.
CM 1051C	cond	card	40-20k 3	200		130 0.5	5.5 .87	4.0	black satin	XLR	215.00	Fixed capsule with bass roll-off.
CM 2001C	cond	omni	20-20k 3	200		130 0.5	6.6 .87	4.0	black satin	XLR	295.00	Detachable capsule. Hand-held.
CM 2003C	cond	omni	20-20k 3	200		130 0.5	6.25 .87	4.2	black satin	XLR	280.00	Detachable capsule.
CM 2020C	cond	card	30-20k 3	200		130 0.5	6.25 .87	4.2	black satin	XLR	280.00	Detachable capsule.
CM 2051C	cond	card	30-20k 3	200		130 0.5	6.25 .87	4.2	black satin	XLR	280.00	Detachable capsule with bass roll-off.
CM 2056C	cond	card	40-20k 3	200		130 0.5	7.25 .87	4.5	black satin	XLR	330.00	Detachable capsule with bass roll-off and mesh windscreen.
AUDIO-TECHNICA —See our ad on Cover II												
AT4073	cond		30-20k	250	62	129 1.0	9.1 .81	4.2	black alum	XLR	750.00	Transformerless, high output, low noise. Very light weight. 12-48 V phantom power.
AT4031	cond	uni	30-20k	200	46	145 1.0	6.3 .81	4.9	black chrm	XLR	325.00	Pressure gradient capacitor, low self noise,high SPL capability. 9-52 V phantom.
ATM25	dyn	hyp	30-15k	600	57		4.65 1.5	13		XLR	238.00	Largediaphragm high SPL capability,good low-end response. Suited for instrument mic'ing.
AT4051	cond	card	20-20k	250	35	143 1	6.1 .81	4.2	black chrm bras	XLR	550.00	Transformerless. Interchangeable head capsule(omni/hyper). 48 V phantom power.
AT4049	cond	omni	20-20k	250	35	142 1	6.1 .81	4.4	black chrm brass	XLR	550.00	Transformerless. Interchangeable head capsule(cardiod/hyper). 48 V phantom power.

Model	Type	Pattern	Frequency Response ±dB	Impedance, Ω	Sensitivity, 1k	SPL at % distortion	Dimensions-in., L, D, W	Weight, oz	Finish	Connector	Price, $	Features
AT4053	cond	hyper	20-20k	250	35	143 1	6.1 .81	4.2	black chrm	XLR	1550.00	
AT4071	cond	shot-gun	30-20k	250	25	127 1	15.5 .81	5.8	black chrm bras	XLR	900.00	
BRUEL AND KJAER												
4011	cond	card	40-20k 2	180	40	110 0.5	6.75 .75	5.8	black chrm	XLR	1497.00	Transformerless,phantom powered.Switchable 0,20dB attenuator. Flat response both on and off-axis.
3529	cond	omni	10-20k 2	30	24	135 1.0	6.5 .63	5.3	black chrm	4pin XLR	5378.00	Matched stereo pair (4003/4006),phase matched,amplitude, true omni nose-cones.
3530	cond	omni	20-20k 2	30	36	135 1.0	6.5 .63	5.3	black chrm	XLR	4185.00	Similar to 3529
4003	cond	omni	10-20k 2	30	24	135 1.0	6.5 .63	5.3	black chrm	4pin XLR	1226.00	Dynamic range from 15 to 154dB SPL(A)typical. Uses 2812 power supply.
4006	cond	omni	20-20k 2	30	36	135 1.0	6.5 .63	5.3	black chrm	XLR	1226.00	Dynamic range from 15 to 143dB SPL(A)typical. Uses 2812 power supply.
4004	cond	omni	10-40k 2	30	38	148 1.0	6.5 .63	5.3	black chrm	4pi XLR	1226.00	Dynamic range from 24 to 168dB SPL(A)typical. Uses 2812 power supply.
4007	cond	omnl	20-40k 2	30	50	148 1.0	6.5 .63	5.3	black chrm	XLR	1226.00	Dynamic range from 24 to 155dB SPL(A)typical. Phantom powered.
COUNTRYMAN ASSOCIATES												
Isomax 2	cond	omnni hyper card	20-20k 3	600	57	150 1	.81 .43	2.8	matt black	XLR	293.09	Multi-purpose mini-mic,four patterns,wide dynamic range.
Isomax 3	cond	omni hyper fig 8	20-20k	600	57	150 1	.81 .43	4.0	matt black	XLR	324.09	Same as Isomax 2 except provided on variable-length goosneck.
Isomax 4	cond	card	70-18k 3	270	52	120 1	.81 .43	4.4	matt black	XLR	478.31	Podium/handheld,optimized for voice,variable-length goosneck, high-pass filter, electronic vibration isolation.
Isomax TVH	elec cond	hyper	70-18k 3	270	52	120 1.0	.81 .43	3.2	matt black	XLR	447.38	High gain before feedback.Rejects background noise and phase cancellation from multiple mics. Active vibration isolation.
EM-301	elec cond	omni	50-15k 3	600	52	130 3.0	.75 .34	3.0	matt black	XLR	384.19	Waterproof for mounting into conference tables,pulpits, baptistries. Internal isolation minimizes mechanical noise transmission.
Head mic	elec	card/	50-15	600	57	150	.31	3.4	black	XLR	338.87	Low profile headset mic. Equalized for warmth and clarity.
CT MARKETING —See our ad on page 35												
CXM	cond	cont-act	25-20k	600		155 .01	.75 .05	1.0	brwn PVC	XLR ¼"	266.00	Contact condenser mic for drums and stringed instruments.
CXS	cond	cont-act	25-20k	600		155 .01	.75 .05	1.0	brwn PVC	XLR ¼"	439.00	Contact condenser mic for stereo piano.
CX2	cond	cont-act	25-20k	600		155 .01	.75 .05	1.0	brwn PVC	XLR ¼"	329.00	Contact condenser mic for monaural piano.
DRM WIZ	cond	cont-act	25-20k	600		155 .01	.75 .05	1.0	brwn PVC	XLR ¼"	495.00	8-mic system which also incorporates mic mixer. Additional MIDI output.
CROWN INTERNATIONAL —See our ad on page 13												
PZM-30 R	elec cond	hemi	25-15k 6	240	65	150 3	5 6	6.5	black gold	XLR	349.00	Pressure Zone Microphone©. Smaller models available.
PZM-30 FS	elec cond	hemi	20-15k 3	240	67	150 3	6 5	6.5	slvr alum	XLR	349.00	Pressure Zone Microphone©. Smaller models available.
GLM-100	elec cond	omni	20-20k 3	240	71	150 3	.75 .31	1.0	black PVC	XLR	199.00	Miniature clip-on mic for voice and instruments. Model GLM-100F for wireless transmitters.
GLM-200	elec cond	hyper	60-20k 6	100	69	150 3	.75 .31	1.0	black PVC	XLR	229.00	Miniature clip-on mic for voice and instruments. Models GM-200/EQ and GM-200/EQR for wireless transmitters.
PCC-160	elec cond	half super	50-18k 6	150	53	120 3	6.7 3.2	11.5	blac steel	XLR	285.00	For stage-floor pickup of drama,musicals,opera,lecterns,news desks. Bass-tilt switch. Model PCC-200 is gated.
LM-200	elect cond	super	80-15k 6	100	68	150 3	16 1.1	10.2	black steel	scr trm	289.00	Lectern microphone with swivel mount for noise-free adjustment. Phantom or 12 V DC adaptor. Pop filter, low-cut.
CM-200	elec cond	card	80-15k 6	200	73	151 3	7.5 1.8	7.0	black alum steel	XLR	209.00	For hand-held stage vocals and instruments.Wood handles available. Model CM-100 is PZM omni hand-held mic.
CM-310	elec cond	card	60-17k 6	200	77	151 3	7.3 2.0	7.0	black alum steel	XLR	309.00	Differential cardiod has outstanding gain-before-feedback. Wood handles available.

Model	Type	Pattern	Frequency Response ±dB	Impedance, Ω	Sensitivity, 1k	SPL at % distortion	Dimensions-in., L, D, W	Weight, oz	Finish	Connector	Price, $	Features
FOSTEX CORPORATION OF AMERICA—See our ad on Cover III												
M85RP	ribbon	super hyper	500-12k	600	60	153 1	2.0 6.4	9.8	black alum	XLR	395.00	Noise-cancelling near-field mic. Effective in isolating snare from hi-hat or excluding ambient noise.
M77RP	ribbon	card	40-18k	200	56	147 2	1.8 6.8	12	black alum	XLR	460.00	Excellent mic for kick and snare. Has three position EQ switch.
M88RP	ribbon	bi	40-18k	600	58	55- 2	149 1.8 5.3	2.0	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.
M22RP	ribbon	M/S	40-18k	600	55	148 2	9.6 2.8	26	black alum	XLR	1095.00	Broadcast applications in stereo announcing/interviewing. Useful for ENG and location work. Applications on percussion instruments where two mics would cause phasing.
M20RP	ribbon	M/S	40-18k	600	54	148 2	9.6 2.8	25	black alum	XLR	695.00	Trimmed-down less expensive version of M22RP.
M11RP	ribbon	uni	40-18k	600	54	148 2	7.0 2.5 2.6	20	black zinc	XLR	595.00	Smoothest cardiod pattern in the RP series. Offers flat frequency response.
M51RP	ribbon	card	80-20k	250	58	149 2	2.0 6.7	10	black zinc	XLR	250.00	Studio quality microphone,rugged enough for road.
HM ELECTRONICS,INC—See our ad on page 3												
EM43-4	elec	omni	20-20k	2.2	65		.29 .78	1.5	black alum	TA4F	75.75	Built in by-pass capacitor to eliminiate RF interaction. High gain-before feedback and high SPL capability,
HM58	dyn	card	80-14k	600	75		6.6 2.0	10	grey	XLR	133.25	Designed for high quality professional applications.
RM77	elec cond	card	150-15k	600	74		7.5 2.0	11	grey	XLR	133.25	Built-in reverb with variable control. Mute switch.
NEUMANN(GOTHAM AUDIO)												
TLM- 170	cond	multi	40-18k	150		140 0.5	6.0 2.4	22	nickel matt	XLR	1750.00	Advanced studio microphone.Transformerless.
U89	cond	multi	40-18k	150		122 0.5	7.3 1.8	14	nickel matt	XLR	1700.00	Used for on-air broadcasting, narration,voice-over and film scoring.
U87A	cond	multi	40-16k	200		117 0.5	7.8 2.2	18	nickel matt	XLR	1875.00	Improved version of this studio standardmicrophone. 10db greater output than U87.
KM130	cond	omni	40-20k	50		140 0.5	3.6 0.8	2.8	black brass	XLR	600.00	Miniature condenser mic. Transformerless output.Dynamic range of 124dB. Capable of remotely powered capsule.
KM140	cond	card	40-20k	50		138 0.5	3.6 0.8	2.8	black brass	XLR	600.00	Similar to KM140, but with dynamic range of 122db.
KM145	cond	card	40-20k	50		138 0.5	3.6 0.8	2.8	black brass	XLR	600.00	Similar to KM140, but with dynamic range of 121db.Built-in low-frequency roll-off.
KM150	cond	hyper	40-20k	50		142 0.5	3.6 0.8	2.8	black brass	XLR	725.00	Similar to KM140, but with dynamic range of 124dB and hyper-cardiod capsule.
RSM- 1915	cond	M/S	40-20k	50		134 0.5	8.4 1.2	6.0	matt	XLR -5	2650.00	Transformerless M-S/X-Y stereo shotgun mic with active matrix.
PANASONIC/RAMSA												
WM-S1	cond	card	50-18k	600		148	0.5 1.3	0.5	black alum	XLR	199.00	Miniature condenser includes acoustically damped mounting system.Optimized for cymbals,hi-hats and strings.
WM-S2	cond	card	120-15k	250		138	0.5 1.3	0.5	black alum	XLR	160.00	Miniature condenser similar to WM-S1, but optimized for toms and brass.
WM-S5	cond	card	70-16k	600		158	0.5 1.3	0.5	black alum	XLR	270.00	Miniature condenser similar to WM-S1,but optimized for brass,snare and other high-SPL sources.
WM-S10	cond	card	120-15k	250		138	0.5 1.3	0.5	black alum	XLR	210.00	Headset mounted miniature condenser. Ideal for vocalists who play drums or keyboards. Also good for flutes and harmonicas.
PASO SOUND PRODUCTS												
M501	dyn	card	50-15k	250	20		6.2 1.7	15	grey	XLR	90.00	Unique dual shock mount to eliminate handling noise. Non-reflective finish.Includes 18ft. cable.
M601	dyn	card	50-15k	250	20		6.5 1.7	9	grey	XLR	104.00	Same features as M501.
M701	dyn	card	40-16k	250	20		6.5 2.0	11	grey	XLR	134.00	Same features as M501.
M800	dyn	card	40-18k	250	20		6.5 2.0	11	grey	XLR	160.00	Same features as M501.

Model	Type	Pattern	Frequency Response ±dB	Impedance, Ω	Sensitivity, 1k	SPL at % distortion	Dimensions-in., L, D, W	Weight, oz	Finish	Connector	Price, $	Features
PEAVEY ELECTRONICS												
PVM48	elec cond	card	20-20k	200	57		5.7 1.1	13	slate zinc	XLR	219.50	High output.Excellent feedback rejection. Ideal for podiums, choirs,vocal groups.
PVR1	elec cond	omni	20-20k	200	52		5.5 .81	7	black bras	XLR	199.50	Flat frequency response. Good for general studio mic, for choirs or systems calibration.
PV	dyn	card	50-12k	600	59		6.5 2.2	8	grey zinc	XLR	99.50	Inexpensive but well made,having metal housing,on/off and internal pop filter. Lo-Z capable.
PVM580 TM	dyn	hyp	40-15k	400	52		5.7 1.9	14	black zinc	XLR	219.50	Neo-dynamic microphone. Diaphragm laminated with titanium, magnetics are neodymium. Good for hand-held vocals.
PVM380 N	dyn	card	40-15k	400	52		5.7 1.9	14	black zinc	XLR	199.50	A neo-dynamic ball type vocal mic. Neodymiummagnetics provide high sensitivity, natural vocal reproduction.
PVM38	dyn	card	40-16k	300	56		5.7 1.9	9	slate zinc	XLR	199.50	Designed for hand-held applications. Low handling noise. Pop filter and windscreen included.
PVM45	dyn	hyper	40-16k	300	56		5.5 1.1		slate zinc	XLR	199.50	Highly directional. Ideal for drums, acoustic instruments
SENNHEISER ELECTRONIC CORPORATION												
MD431	dyn	super	40-16k	200	57	120 1.0	7.8 1.2	8.8	black alum	XLR	429.00	High gain-before-feedback, handles high SPL. Triple-layered steel mesh grill. Magnetic reed on/off switch.
MKE 4032	elec cond	super	70-20k	200	46 1.0	140 1.9	8.1	7.5	black alum	XLR	595.00	12-48 V phantom or AA battery operation. Built-in blast filter, shock mount. Rugged, handles high SPL.
MKH20	cond	omni	25-20k	150	32	142 0.5	6.0 1.0	3.6	black alum	XLR	925.00	High frequency boost switch, and 10dB pad. Ideal for concert. acoustic and M-S recording.
MKH40	cond	card	40-20k	150	32	142 0.5	6.0 1.0	3.6	black alum	XLR	925.00	Versatile mic with switchable 10dB pad and bass attenuation. Transparent response.
MD518	dyn	card	50-16k	200	58	120 1.0	7.0 1.2	6.5	black alum	XLR	219.00	Versatile hand-held microphone. Uses include vocals, rack toms and sax.
ME80	elec cond	shot	50-15k	130	46		12.3 0.8	12	slvr alum	XLR	239.00	On-camera microphone or handheld interview mic for ENG/EFP. Increased gain-before-feedback for podium or lecture use.
MKE-2	elec cond	omni	40-20k	200 1000	46	126	.43 1.0	0.1 .23	black flesh	XLR pig	254.00	Two impedance options, fleshtone color option. Small, ultra light for broadcast, church and theatre in wireless system.
MD421	dyn	card	30-17k	200	54	175 1.0	8.4 1.2	13.6	black	XLR	399.00	Versatile, durable. Handles high SPL. Five position roll-off provides equalization up to 1000Hz.
SONY PRO-AUDIO												
C-48	cond	multi	30-16k	150	39	128 1.0	2.2 9.1	20	satin nickel	XLR	1050.00	Selectable patterns,10 dB pad,lo-cut switch, 9 V battery or phantom power. Vibration-proof structure.
C-535 P	cond	card	30-16k	200	40	138 1.0	0.8 6.1	4.9	black alum	XLR	495.00	Slim-line design with 10 dB pad. Rejects SCR, TV and other electronic noise. Excellent transient response.
ECM- MS5	elec cond	stereo card	70-20k	150	40	130 1.0	1.9 8.4	7.6	alum	XLR -5	1250.00	Three capsule design for M-S recording. Built-in M-S matrix field-rugged construction. 12-48 V 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.
SHURE BROTHERS, INC.—See our ad on Cover IV												
SM7	dyn	card	40-16k	150	57		5.8 7.5 3.8	27	grey alum steel	XLR	550.00	Independently switchable bass roll-off and presence-boost switches. Internal air suspension. Heavy gauge storage case included.
SM81LC	cond	card	20-20k	150	40	146 1.0	8.3 0.9	8	chpgn steel	XLR	380.00	10 dB attenuator, 3-position bass roll-off. Omnidirection cartridge(R104) available. Pop filter, swivel adaptor Included.
SM84	cond	super	80-20k	150	46	129 1.0	1.0 0.4	1.6	black brass	XLR	300.00	Super-cardioid lavalier.Chest resonance dip filter gives natural response.Windscreen, multiple mounting options.
SM87	cond	super	50-18k	150	49	142 1.0	7.6 1.9	6.3	grey alum	XLR	329.00	Tailored vocal response reduces need for equalization. Accepts 11 to 52 V phantom power.
SM91	cond	hemi	20-20k	150	45	144 0.1	0.6 3.7 5.0	9.3	black cast steel	XLR	310.00	Low profile boundary effect mic. External pre-amp with12 dB per octave roll-off switch. Accepts battery ot phantom power. Omni capsule available in SM90.
SM94- LC	cond	card	40-16k	150	48	141 1.0	7.5 1.0	8.8	grey steel brass	XLR	250.00	Accepts phantom power up to 52 V or 1.5 V AA battery. Wide dynamic range ideal for sampling applications.
SM98	cond	card	40-20k	150	54	153 0.1	1.2 0.5	0.4	black brass	XLR	250.00	Full-range response in miniature unit. Optional A98SPM adaptor for super-cardioid pickup. Many optional mounting accessories.
SM99	cond	super	80-20k	150	48	130 1.0	1.2 14.7	5.8	black steel brass	XLR	240.00	Miniature gooseneck microphone with lownoise pre-amp built into base. RFI protection.Surface or stand mounting. 5-52 V phantom power. 102 dB dynamic range.
TELEX COMMUNICATIONS												
TE10	cond	card	30-20k	150 -200	75	140		7.4	matt black	XLR	183.00	Condenser with "bright" natural sound. Sudpended by flexible "fingers" which isolate it from mechanical vibration.
TD11	dyn	card	50-16k	100 -250	77			9.2	matt black	XLR	163.00	Die-cast case, reinforced steel mesh windscreen Multi-stage pop filters.
LM-100	cond	omni	20-20k	150	74		.75 0.4	1.0	matt black	XLR	220.00	Lapel mic system includes LM101 micwith 3-foot cable, and PS-10 power supply. Various mounting clips.
LM-300	cond	card	100-15k	150	82		0.7 0.3	1.0	matt black	XLR	220.00	Lapel mic system includes WLM60 mic with 3-foot cable, and PS-10 power supply. Various mounting clips.

Model	Type	Pattern	Frequency Response ±dB	Impedance, Ω	Sensitivity, 1k	SPL at % distortion	Dimensions–in., L, D, W	Weight, oz	Finish	Connector	Price, $	Features
YAMAHA MUSIC CORPORATION												
MH100	elec	card	10-10k	1.6k	70					¼"	49.00	Headset and microphone in one unit. Uses lightweight pads that are easy on ears even with extended use.
MZ101	dyn	card	40-17k	250	76		6.2 0.9		mett brown	XLR	135.00	Noted for it's clean mid-range and high-end. Poly-laminate diaphragm. Unique 3-point suspension. Gold-plate connectors.
MZ102 BE	dyn	card	40-18k	250	76		6.2 0.9		mett brown	XLR	190.00	Deep,lower mid-range quality. Beryllium diaphragm for tight reponse. Die-cast zinc body. Gold-plated connectors.
MZ103 BE	dyn	card	40-18k	250	76		6.1 0.9		mett grey	XLR	235.00	Wide range mic with resistance to off-axis sound. Beryllium diaphragm, 3-point suspension and gold plated connectors.
MZ104	dyn	card	30-17k	250	77		7.0 1.4		mett brown	XLR	145.00	Good instrument mic. Good bass response. Lowered sensitivity to avoid hig SPL overload.Gold-plated connectors.
MZ105 BE	dyn	card	40-18k	250	77		6.0 1.4		mett brown	XLR	200.00	Designed to avoid unwanted bass buildup with close mic'ing. Beryllium diaprhrgm. Gold-plated connectors.
MZ106S	dyn	card	40-18k	250	77		7.2 2.0		mett grey	XLR	140.00	Ideal for vocal use. On/off switch with switch lock for lock-On. Two-layer laminated polyester film diaphragm.
MZ205 BE	dyn	card	40-18k	250	77		4.3 1.3		mett grey	XLR	295.00	Vocal microphone with right-angle XLR connector. Beryllium diaphragm. 3-point suspension. Gold-plated connectors.

Wireless Microphone Systems

AKG ACOUSTICS

WM185 modular microphone system provides a flexible group of products configurable for diverse applications. Microphone heads (D330BT, D321, C535, CK410) can be combined with standard transmitter body T185N. Adaptor A85 can interface external mics or instruments. Transmitter/receivers are either true diversity or non-diversity and operate between 174 and 216 MHz.

Price:

$3,000.00 to $5,000.00

AUDIO TECHNICA—See our ad on Cover II

ATW1031 UniPak and ATW1032 comprise a diversity wireless mic system. Features include a rack-mounted receiver with adjustable output and squelch. Transmitters are crystal-controlled and are available as body pack with instrument or microphone input, or as a hand-held mic. Battery life is 10-hours continuous. Ten VHF frequencies are available.

Price:

$650 to $750 depending on mic element.

COUNTRYMAN ASSOCIATES

EMW series are tiny omnidirectional lavaliere microphones designed for wireless use on stage or in broadcast applications. Exhibiting a very low profile and low handling noise, the mics have three onboard EQ settings adjustable for tie-clip positioning or hiding under clothing.

Price:

On request

HM ELECTRONICS—See our ad on page 3

EM43 is an omnidirectional electret microphone designed to work in RF environments on wireless mic transmitters. It is a high impedance microphone (2.2K Ohms) having a 20-20 kHz frequency response. Two low impedance models (200 Ohms) are also featured: RM77—a cardioid electret, and HM58—a cardioid dynamic microphone.

Price:

$70.00 to $144.00

NADY SYSTEMS

Numerous wireless configurations are offered featuring proprietary system with optional mic heads from Shure, AKG, EV and Audio Technica. 1200GT (top-of-the-line model) is a true diversity system allowing up to twenty units to work on high-band frequencies, utilizing built-in companding noise reduction.

Price:

From $119.95 to $1,699.95

PASO SOUND PRODUCTS

MA25 mic/transmitter features adjustable gain, separate audio and RF on/off switches, low cut filter and integral pop filter, operating at 10 possible frequencies between 174 MHz and 200 MHz. R8 true diversity receiver automatically selects between outputs of two independent VHF receivers for cleanest signal.

Price:

On request.

SAMSON TECHNOLOGIES CORP.

BS-87 is a diversity 10-band digitally-synthesized VHF-selectable transmitter and rack-mountable receiver. Features include dbx noise reduction, Shure SM-87 mic, receiver auto-scanning to detect cleanest signal, balanced and unblanced outputs, channel selector and sensitivity switching.

Price:

$2,195.00

BS-MKE-2 is the same as above except with a belt-pack transmitter that is also frequency-selectable. It features a removable Sennheiser MKE-2 lavaliere mic.

Price:

$1,995.00

TD-757 diversity 10-band-available, has dbx noise-reduction, E-V N/DYM 757 hand-held microphone. The receiver has adjustable AF control, balanced and unbalanced output, and mute, sensitivity, and power switches.

Price:

$1,225.00

TD-831 is the same as the TD-757 except equipped with an Audio-Technica AT-831 and belt-pack transmitter.

Price:

$1,050.00

SENNHEISER ELECTRONIC CORPORATION

VHF1H features hand-held microphone transmitter SKM4031 and miniature receiver EK2012. Utilizing VHF carrier, unit is suited for ENG/EFP.

Price:

$2,940.00

VHF1B features body-pac transmitter SK2012, mini lavalier mic MKE2 and miniature receiver EK2012 utilizing VHF carrier. Suited for ENG/EFP.

Price:

$3,515.00

VHF2H features hand-held microphone transmitter SKM4031 and diversity receiver EM2003, utilizing VHF carrier.

Price:

$3,330.00

VHF2B features body-pac transmitter SK2012, mini lavalier microphone MKE2 and diversity receiver EM2003, utilizing VHF carrier.

Price:

$3,905.00

UHF2H is similar to VHF2H system, but utilizing UHF carrier.

Price:

$6,430.00

UHF2B is similar to VHF2B system, but utilizing UHF carrier.

Price:

$7,105.00

UHF2EH features hand-held microphone transmitter SKM4031TVH, and diversity receiver EM2003TVH, in a portable canvas bag with battery. Utilizes UHF carrier.

Price:

$6,070.00

UHF2EB features body-pac transmitter SK2012TVH, lavalier mic MKE2, and diversity receiver EM2003TVH in a portable canvas bag with battery. Utilizes UHF carrier.

Price:

$6,876.00

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

W1020S Wireless Body-Pack System consists of W10BT Body-Pack Transmitter, W20R Receiver and WL83 (omnidirectional) or WL84 (supercardioid) microphone. Receiver features 9-pole linear phase filters for high selectivity and accommodates both low and high-level inputs.

Price:

$1,200.00

W1025S Diversiphase(TM) Wireless Body Pack System is an advanced body pack system for professional use featuring the W10BT Transmitter, WI83 or WI84 lavalier microphones and W25DR Receiver. Receiver phase-locks two signals delivering 98dB (A-weighted) dynamic range.

Price:

$1,700.00

W1520 Hand-held Wireless System consists of one wireless microphone (W15HT/58 or W15HT/87) and one W20R receiver. Mic heads are interchangeable. High efficiency circuitry provides up to 12 hours peak performance with 9V alkaline battery.
Price:
$1,450.00 to $1,600.00 (depending on microphone)

W1525 Diversiphase Hand-held Wireless System consisting of one hand-held microphone (W15HT/58 or W15HT/87) and one W25DR receiver. Both microphones can handle 138 dB SPL (maximum) and feature a quartz-locked transmitter. Companding circuitry delivers clean sound even at low levels.
Price:
$1,950.00

TELEX COMMUNICATIONS, INC.

HT-400 is a two channel wireless microphone with integral transmitter and antenna. Models available with microphone heads: Telex TE-10 condenser, Shure SM-87 condenser or Shure SM-58 dynamic. Format is interchangeable.
Price:
$885.00 to $1,095.00 (depending on head)

HT-100 is a single channel wireless microphone with integral transmitter and antenna. Includes on/off switches for both audio and mic. Includes models with Telex TE-10 condenser, Telex TD-11 dynamic, Shure SM-58 dynamic and Shure SM-87 condenser heads.
Price:
$370.00 to $620.00 (depending on head)

VEGA (A MARK IV COMPANY)

R-42/T-88 Pro Plus hand-held system consists of the R-42 Pro- Plus true-dual-diversity receiver and the T-86 Pro Plus transmitter. The system features 108 dB (typical) signal-to- noise ratio and wide RF dynamic range. 16 poles of IF filtering give high adjacent channel rejection. Transmitter features an EV N/D757 element.
Price:
$4,213.00

R-33/T-86 Pro plus portable system consists of R-33 Pro Plus miniature portable receiver and the T-86 Pro Plus portable transmitter. The R-33 receiver is ideal for camera mounting.
Price:
$2,398.00

67B/77DII Pro Plus portable system consists of the 67B Pro Plus true-diversity receiver and the 77DII Pro Plus body pack transmitter. The 67B operates on four 9-volt alkaline batteries or external power from a 12-volt camera belt pack or other +10.5 to 18 volt d.c. source.
Price:
$2,548.00

Traveler portable system consists of the 66B portable receiver and either the T-37 Pro bodypack transmitter or the T-36 Pro hand-held transmitter (with Electro-Voice BK-1 Black Knight Element). Designed for on-location and portable use. System ranges up to 1200 feet.
Price:
$1,635.00 (1-B bodypack), $1,980.00 (1-HE handheld)

Pro 2HE hand-held diversity system consists of the R-32 Pro true-dual-diversity receiver and the T-36 Pro hand-held transmitter. System operates on any crystal-controlled frequency from 150 MHz to 216 Mhz, at a range of up to 1200 feet.
Price:
$2,110.00

Pro 1-B bodypack system consists of the R-31B Pro non- diversity receiver and the T-37 Pro bodypack transmitter. Transmitter accepts virtually all electret lavalier mics. Recessed control switches can be operated by feet.
Price:
$1,520.00

Ranger 2 true diversity systems include eight bodypack and hand-held configurations, including Ranger 1 systems for applications not requiring diversity. All feature wide dynamic range and operate on VHF high-band frequencies.
Price:
$1,025.00 to $1,425.00

Reporter portable systems are useful for both studio and on- location applications. Available either as a bodypack or a hand-held system, it features CVX audio processing for wide dynamic range.
Price:
$1,255.00

Magnetic Tape

AGFA CORPORATION

PEM469 is a studio mastering tape delivering high output and low noise characteristics and standard bias for compatibility with a variety of tapes and machines. Low print-through characteristics minimize pre and post echo, while excellent slitting provides consistent phase stability. Available in ¼-in., ½-in., 1-in. and 2-in. widths.

PEM468 is a studio mastering tape featuring extremely low print-through, high output and low noise for a wide dynamic range. Excellent slitting for consistent phase stability. Batch number and web position printed on back coating for permanent identification. Available in ¼-in., ½-in., 1-in. and 2-in. widths.

PEM291D is a digital mastering tape compatible with both PD and DASH formats. It features superior carrier-to-noise ratio, ensuring extremely low error rate, a consistent and reliable formulation, and superior winding characteristics. Available in ¼-in., ½-in. and 1-in. widths.

PEM526 is designed specifically for high-speed cassette duplication, providing excellent durability under the stress of bin-loop applications. Also features excellent high- frequency retention after numerous passes and extremely low print-through. Available in ¼-in., ½-in. and 1-in. widths.

R-DAT (PACKAGED/DUPLICATOR) is designed specifically for the rotary-head digital audio tape cassette format. Cassettes feature precision coating with pure metal pigments. Special back-coating ensures perfect mechanical performance and security for data stored. Available in R-60, R-90 and R-120 lengths.

PE649/949/1249 is a premium iron oxide bulk audio cassette tape featuring a high output, low noise using standard IEC Bias I. It also offers extended headroom in both low and high frequencies for critical music duplication.

PE619I/919I is a bulk audio cassette tape which features an improved binder system (relative to original 19 series formulation) for cleaner running and overall better handling. An extended high-end response iron oxide tape, it is bias compatible with industry's Standard I designation.

PE647/947 is a chromium dioxide, Bias II product for the most critical applications in either high-speed or real-time applications. Featuring excellent high-frequency response, low-noise and wide dynamic range, it can be used with either 120 μs equalization or 70 μs equalization.

AMPEX CORPORATION

456 Studio Mastering Tape is suitable for all demanding recording operations. Known for its reliable batch-to-batch consistency, accurate and clean slitting, this tape has become a studio standard. Available in ¼-in., ½-in., 1-in. and 2-in. widths. Coating is 0.55 mils, backcoat 0.05 mils and basefilm 1.42 mils.

478 Low Print Mastering Tape is specially designed for applications where print-through must be minimized. Additional characteristics include high M.O.L., low bias noise and low distortion, and batch consistency. Available in ¼-in. and ½-in. widths. Coating is 0.52 mils, backcoat 0.04 mils and basefilm 0.83.

467 Digital Open Reel tape is manufactured in "clean room" environment to assure blemish-free surface (minimizing necessity of error correction). Has excellent slitting characteristics plus end-to-end and reel-to-reel consistency, and works well with standard machine alignments. Available in ¼-in., ½-in. and 1-in. widths. Coating thickness is 0.20 mils, backcoat 0.04 mils, and basefilm 1.42 mils.

467 Digital U-matic Cassettes feature the same specialized oxide formulation as the open reel variety, but are manufactured and qualified to meet digital PCM criteria. Conductive backcoating reduces static build-up and provides for precise tape packing to minimize edge damage. Available in 30, 60, 75 or 80 minute lengths.

467 DAT Cassettes contain advanced metal particle formulation to ensure high output levels, minimum error corrections and exceptional durability for multiple pass performance. Features a unique professional label documentation system. Available in 45, 60, 90 and 120 minute lengths.

472 Studio Audio Cassettes are available in both normal bias (ferric) Type I and high bias (cobalt modified ferric) Type II formulations. Tape characteristics include totally flat frequency response and unsurpassed sensitivity for accuracy in reproduction. It is housed in a 5 screw shell with improved pad design for superior azimuth tracking. Various lengths are offered: 5, 10, 15, 30, 45, 60, and 90 minutes.

3M (SCOTCH)

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. 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. Best tape for applications where both music and speech are being recorded. Available in ¼-in., ½-in., 1-in. and 2-in. widths.

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 1/4-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. Available in both the standard album box or exclusive hanger/shipper box.

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.

XSM-IV audio cassettes are metal particle bias (IEC Type IV) for the ultimate quality music recording applications. Available in album boxes, 60 and 90 minute lengths.

Tape Accessories

POLYLINE CORPORATION —See our ad on page 2

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

TENTEL CORPORATION

T2-H20-ML Tentelometer Tape Tension Gauge. For use on all open reel tape recorders (¼-in. to 2-in.).
Price:
$325.00

T2-H12-2 Tentelometer Tape Tension Gauge. For use on PCM 3324 DASH recorders.
Price:
$725.00

T2-H7-AC Tentelometer Tape Tension Gauge. For use on all audio cartridge machines.
Price:
$345.00

WS-120 Field Calibration Weight Set. Used for verifying calibration, it provides improved accuracy at specific tensions.
Price:
$49.00

XEDIT CORPORATION

S-3D Editall Deluxe full size ¼-in. splicing block provides three cutting angles.
Price:
$50.00

S-2 Editall Compact ¼-in. splicing block provides two cutting angles.
Price:
$36.00

S-3OT Editall Otari replacement ¼-in. splicing block. Provides three cutting angles.
Price:
$50.00

S-3.5 Editall Deluxe full size ½-in. splicing block. Provides three cutting angles.
Price:
$65.00

SA-2 Editall Curved troth full size 2-in. splicing block. Provides three cutting angles.
Price:
$160.00

MD-25 fits all ¼-in. digital formats. It is an exact retrofit for Mitsubishi.
Price:
$145.00

EC-D1 fits all 1-in. digital formats such as Otari and Mitsubishi.
Price:
$300.00

Editabs are precision pre-formed die-cut editing tabs; available in nine models, Cx-1 through CX-9, covering audio cassette to one-inch tape sizes.
Price:
Depending on size and quantity

ADDRESSES

Agfa-Gevaert
275 North Street
Teterboro, NJ 07608

AKG Acoustics
77 Selleck St.
Stamford, CT 06902

Altec Lansing
10500 W. Reno
Oklahoma City, OK 73128

Ampex Corporation
401 Broadway MS 22-02
Redwood City, CA 94063-3888

AMS/Calrec
PO Box 31864
Seattle, WA 98103

Audio-Technica US, Inc.
1221 Commerce Dr.
Stow, OH 44224

Bruel and Kjaer Instruments
185 Forest Ave
Marlboro, MA 01752

Carvin Corporation
1155 Industrial Ave
Escondido, CA 92025

Countryman Associates
417 Stamford
Redwood City, CA 94063

Crown International
1718 W. Mishawaka Rd
Elkhart, IN 46517

C-T Audio Marketing
3050 SW 14 Pl., Suite 3
Boynton Beach, FL 33434

Electro-Voice
600 Cecil St
Buchanan, MI 49107

Fostex Corporation
15431 Blackburn Ave
Norwalk, CA 90650

Geltzer and Company
1180 Ave. of the Americas
New York, NY 10036

Gotham Audio Corporation
1790 Broadway
New York, NY 10019-1412

HM Electronics
6675 Mesa Ridge Rd
San Diego, CA 92121

Mitsubishi Pro Audio Group
225 Parkside Dr.
San Fernando, CA 91340

Paso Sound Products
14 First St
Pelham, NY 10803

Peavy Electronics (AMR)
711 A ST.
Meridian MS 39301

Polyline Corporation
1233 Rand Ave
Des Plaines, IL 60016

The Phelps Group (Ramsa)
11999 San Vicente Blvd, Suite 301
Los Angeles, CA 90049

Samson Products Corp
485-19 S. Broadway
Hicksville, NY 11801

Sennheiser
6 Vista Dr.
Old Lyme, Ct 06371

Shure Brothers
222 Hartrey
Evanston, IL 60204

Studer Revox
1425 Elm Hill Pike
Nashville, TN 37210

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

Telex Communications
9600 Aldrich Ave. S.
Minneapolis, MN 55420

Tentel
1506 Dell Ave
Campbell, CA 95008

3M Center
Bldg. 236-1B-06
St. Paul, MN 55144

Uher of America
7067 Vineland Ave
N. Hollywood, CA 91605

Vega
9900 Baldwin Pl.
El Monte, CA 91731-2204

Xedit Corporation
133 S. Terrace Ave
Mount Vernon, NY 10550

Yamaha-Pro Audio
P.O. Box 6600
Buena Park, CA 90622

BRUCE BARTLETT

How To Operate A Small Concert Sound System

• Suppose you've been asked to run a sound-reinforcement system for a musical group. Here are some tips on doing the job efficiently and with the best sound quality. We'll cover the necessary equipment, where to place it, setup and wiring, system equalization, mic'ing, and mixer operation.

EQUIPMENT

The following list shows the equipment you'll need:

- Mic stands and booms
- Microphones
- Microphone foam pop filters
- Direct boxes
- Mic cables
- Snake
- Mixing console
- Effects (usually reverb, delay, and compression)
- Cables from console to and from effects
- Cables from console to graphic equalizers
- Cables from graphic equalizers to power amps
- ⅓-octave graphic equalizers (2 or 3)
- Power amplifiers (2)
- Rack for signal processors
- Rack for power amplifiers
- House speaker systems (2 total, 1 per side)
- Monitor speakers (2 to 6)
- Speaker cables (heavy gauge, #12 recommended)
- Notebook, masking tape, felt-tip pen, gaffer's tape, flashlight, ground-lift adapters
- Carrying cases for equipment

If your speakers are bi-amped, you'll also need an active crossover, an extra power amp, and associated cables.

EQUIPMENT PLACEMENT

After carting in your equipment, set it up as shown in *Figure 1*. Put a complete speaker system on the left and right sides of the stage, up to the edge of the stage. Put wedge-shaped floor monitors in front of the people needing them. Use two to six across the front of the stage, or as side fills. The keyboardist and drummer might need separate monitors.

Place your amplifier rack on stage near the rear of the stage, off to one side. Put the amplifiers on stage, rather than at the mixer, so that the speaker cables can be shorter. Long cables waste power through resistive heating.

Position the snake box either at the front of the stage in the middle, to one side of the stage, or at the rear of the stage. To prevent hum pickup, try to keep the snake away from power outlets and power cables.

Locate the mixing console in the audience area, about ¾ back if your snake is long enough. If the audience will be dancing, put the console and the snake out of the way of the dancers. Place the signal-processor rack (graphic equalizers and effects) within easy reach.

Run the snake between the console and the stage. If the snake doesn't reach, move the snake box or the console. Try to run the snake where it won't be stepped on, and tape it down with gaffer's tape.

PRELIMINARY SETUP

Now that the major equipment is positioned, it's time to make preliminary adjustments to it. Neutralize or zero the console. That is, set all controls to "off", "flat" or "zero" so as to have no effect. Be sure that the master faders are down, and the master monitor (aux) send is down. This will prevent feedback and pops when you plug in the mics later. Turn up the gain trims for maximum sensitivity and remove any input pads. Or, if you already know the approximate settings needed, set the gain trims and pads accordingly.

Turn off the power amps and set their volume controls about halfway up. Set the graphic-equalizers' level controls to 0 or unity gain, and set all their controls flat.

POWER WIRING

Next you'll install wiring for AC power. Run a heavy extension cord from where you plugged in your mixer to the stage. Plug an outlet strip into the extension cord onstage. Into this outlet strip, plug the power cords for your power-amp rack and the electric musical instruments.

Also power your processor rack and the mixer off an outlet strip. Caution: First make sure that the system current (the sum of the equipment fuse ratings) does not exceed the breaker or fuse rating for that power circuit.

If you power the stage equipment from a stage outlet, and power the mixer from another outlet near the mixer, this is likely to cause ground loops and hum. It's usually better to power everything from the same circuit if possible.

AUDIO-SIGNAL WIRING

Figure 2 is a block diagram of the audio wiring for a non-bi-amped system; *Figure 3* is the same for a bi-amped system. The following connections refer to *Figure 2*:

Starting at the console,

1. Connect the console main output (left and right) to the graphic EQ inputs.

2. Connect the console monitor output (one of the aux outputs) to another graphic EQ input.

3. Connect the console aux send (effects send) to the effects input; connect the effects output to the console effects return.

4. Connect the house graphic equalizers' outputs to the inputs of the house stereo power amplifier. Some snakes have send-lines built in for this purpose.

5. Connect the outputs of the house stereo power amp to the house speakers (left and right). If your power amp is mono, connect the speaker systems on both sides to the amp in parallel. First be sure that the total load on each power- amp channel is not less than two ohms. Wiring two identical speakers in parallel halves the impedance; wiring them in series doubles the impedance.

6. Connect the output of the monitor graphic equalizer to the input of the monitor power amp.

7. Connect the output of the monitor power amp to a monitor speaker.

8. Interconnect the monitor speakers with speaker cable.

If your system is bi-amped, please refer to *Figure 3* and make the following connections. They refer only to the left channel of each component, but you will connect both channels.

If you want effects in the monitors, but your console does not permit this, connect the effects output to a spare console input and turn up its monitor send.

1. Repeat steps 2, 3, 6, 7 and 8 described above.

2. Connect the console main output to the graphic EQ input.

3. Connect the graphic EQ output to the active crossover input.

4. Set the crossover frequency to the upper frequency limit of the woofer, which should be about the same as the lower frequency limit of the midrange.

5. Connect the active crossover's high-frequency output to the input of the high-frequency house power amp.

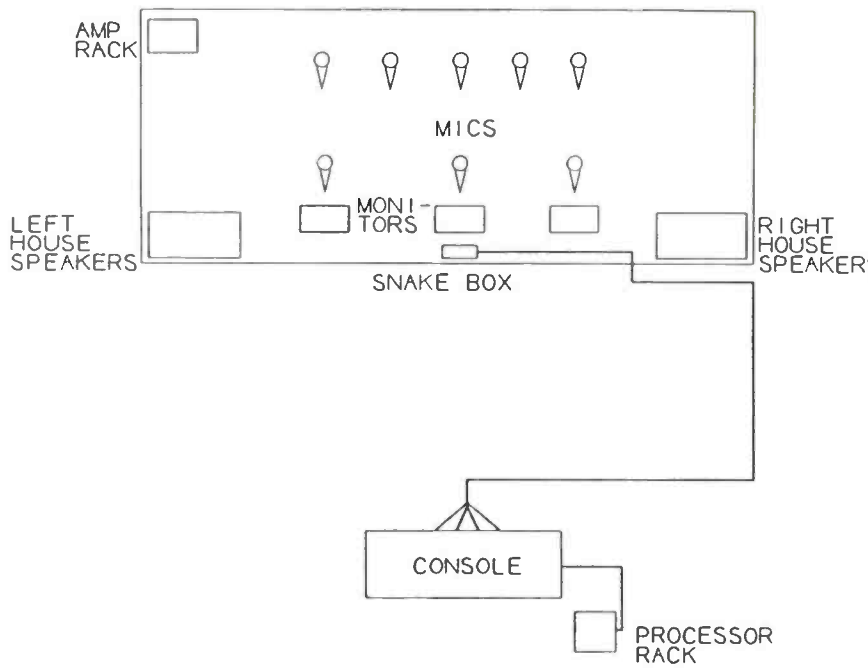


Figure 1. The equipment layout.

6. Connect the house power amp output to the left house midrange and tweeter.

7. Connect the active crossover's low-frequency output to the input of the low-frequency house power amp.

8. Connect the output of the low-frequency house power amp to the left house woofer.

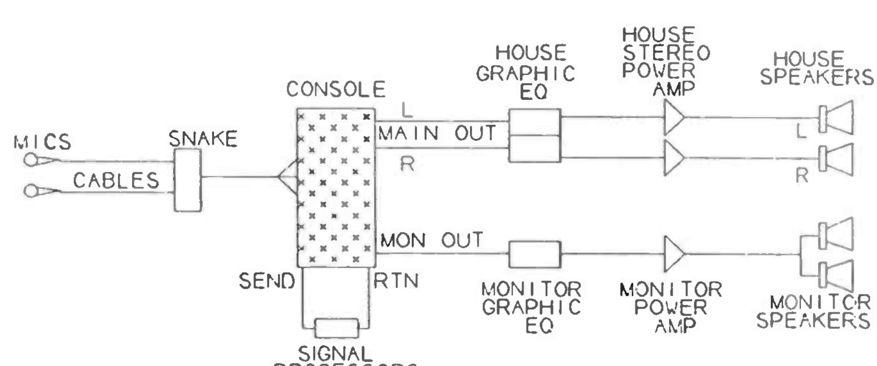
Repeat all the steps above for the right channel.

Next, plug the snake's XLR-type connectors into the console mic inputs, matching the numbers on the snake cables with those on the console.

If you want effects in the monitors, but your console does not permit this, connect the effects output to a spare console input and turn up its monitor send. That monitor send will act as an effects-return control for the monitors, and that input fader will act as an effects-return control for the house speakers.

If you want to compress the vocals (as a group), connect a compressor between the console access jacks for that group. Or, if you want to com-

Figure 2. A block diagram of a concert sound system that is not bi-amped.



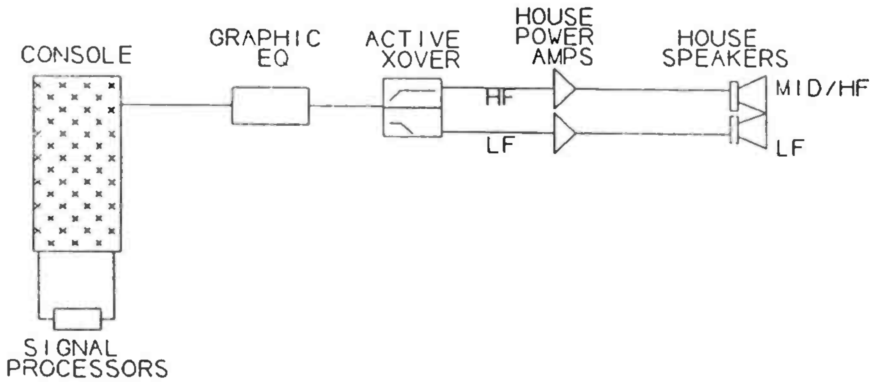


Figure 3. A bi-amped sound system. This is shown after the console, with only one channel shown.

press only the lead vocal, connect a compressor between the access jacks for that input module.

HUM CHECK

After first checking that the console master faders and master monitor-send knob are turned down, power up all the equipment. If you hear hum, you might want to float the console by plugging a 3-to-2 power adapter onto its power cord. This is not recommended because it creates a safety hazard if the mixer chassis becomes electrically hot.

For balanced microphone cables, the cable shield should be soldered at both ends. But for balanced line-level cables, the shield should be soldered only at the mixer end.

If the shield is connected at both ends, you might create a ground loop which causes hum. If necessary, you can lift the ground at the stage end of the cable by inserting a ground-lift adapter (*Figure 4*).

GRAPHIC EQUALIZATION

At this point, you might want to equalize the house system. Here's one way: Plug a set of high-quality headphones into the console and monitor its main output. Play a good cassette or CD through the console and house speakers. First be sure to set the noise-reduction and tape-type switches on your cassette deck appropriately, and clean the heads. Using the graphic equalizer, equalize the house speakers to sound like the headphones. Or just equalize the house speakers by ear to sound like a good home stereo system.

Note: If your speakers are bi-amped, first adjust the relative gains of the low- and high-frequency amplifiers for a good tonal balance, then adjust the graphic EQ.

Another way to equalize the sound system is to use pink noise and a real-time analyzer. Using a pink-noise generator, play pink noise through the console into the house speakers. Place the microphone from the real-time analyzer where your head will be behind the console, or in the middle of the audience area.

Note the display on the RTA screen. On your equalizer, pull down the frequencies that are bumped up on the RTA display, until the average display curve is relatively flat. If necessary, boost frequencies that are low in level (but try not to apply more than 6 dB boost). You might want to try several different mic positions and pick an average.

Once the RTA display is flat, roll off the high frequencies starting above 1000 Hz, down about 10 dB at 10 kHz. This is necessary because the microphone and RTA hear differently than our ears do.

On stage, put your mic stands approximately where they will be used, and insert your microphones in the mic stand adapters.

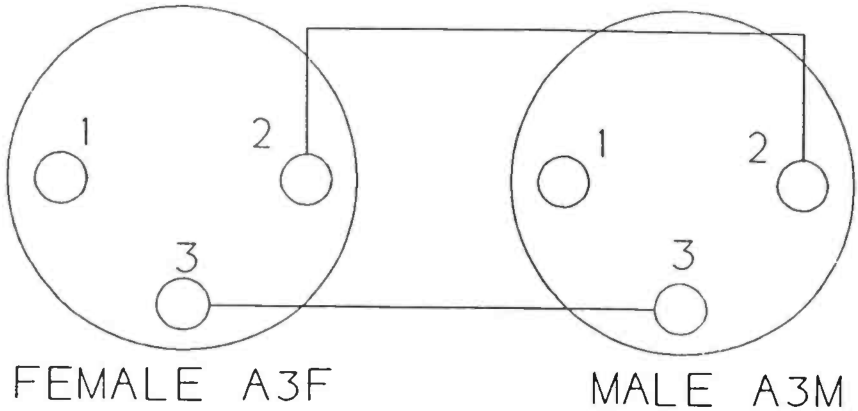
Turn off the pink noise and play a cassette or CD through the system. Fine-tune the equalization by ear so that the system sounds like a good home stereo system, or like the headphones plugged into the console.

Again, if your system is bi-amped, first adjust the gains of the low- and high-frequency amplifiers for flattest RTA response before adjusting the graphic EQ.

MIC'ING

Now you're ready to mic the stage. First, make a list of numbers matching the console input numbers. Next to each number, write the name of the instrument or vocal you want to plug into that input (as shown in *Figure 5*). Then, next to each instrument or vocal, write the model number of the microphone or direct box (DI) you

Figure 4. A ground-lift adapter for a line-level cable.



want to use. You will refer to this list as you connect mic cables to the snake box and to the mics.

On stage, put your mic stands approximately where they will be used, and insert your microphones in the mic stand adapters. Some people prefer to run mic cables first, then bring out the microphones last.

Place the input list by your snake box. Plug a mic cable into snake box connector 1. Run this cable out to the mic you wish to use on this input, and leave a coil of cable at the mic stand (so you can easily reposition the mic stand). This procedure also prevents a tangle of cables at the snake box, making it easier to change connections.

Plug the mic cable into the mic or DI. Using masking tape and a pen, label the DI box according to which instrument it will connect to. If you're using a mic-stand boom, run the cable down the boom and loop it around the boom near the mic stand.

Run cables for all the mics and DIs as you just did for input 1. Place each mic close to its instrument, usually at the point where the instrument is loudest. This maximizes gain-before-feedback.

Have a spare mic, cable, and stand off-stage ready to use in case a mic fails.

TALKBACK MIC

Near the console, place a mic with which to talk to the musicians on stage through the monitor speakers. Plug this talkback mic into an unused console input (behind the console). Turn up the monitor send (aux send) for that input about halfway, and turn up the monitor master send about halfway. Talk into the mic and verify that the musicians can hear you clearly. If not, turn up the monitor send or monitor master send. This way you can talk to the musicians without disturbing the audience.

MIC CHECK

Now that the mics are plugged in, you're ready to do a mic check. First, tape a designation strip of masking tape along the top of the console faders. Referring to your input list, write on the tape the name of the instrument or vocal that each fader affects. Also label the meters, master sends and returns.

Turn on phantom power for any condenser mics. Plug your headphones into the console and monitor cue or solo. Be sure the solo or cue gain is turned up. Set each mic to cue or solo in turn, and verify that you can hear it. If not, double-check the mic cables, phantom or battery powering, and mic switches; and verify that each mic goes to its proper input. If a DI box hums, flip its ground-lift switch.

Turn up the monitor master send about halfway. Have someone on stage talk into each mic in turn, and gradually turn up its monitor send (aux send) until the volume is adequate on stage. Mark the point on the monitor-send knob where feedback occurs and don't exceed this level during the concert.

If all the sound is lost, refer to your system block diagram and trace the signal through your system by checking the level indicators at each stage.

MONITOR EQUALIZATION

If you can't get enough gain before feedback with the monitors, you might want to notch out the major feedback frequencies. For each vocal mic, slowly turn up the monitor send until the system just starts to ring or feed back. Find the control on the graphic EQ that corresponds to the feedback frequency (by ear, by trial and error, or by RTA display). Reduce the level at that frequency until the feedback stops. Turn up the monitor send a little more until feedback occurs, and repeat the process for about five feedback frequencies. Do this for each vocal mic and for instrument mics that seem especially prone to feedback.

MIXER OPERATION

Now that the monitors are set, bring up the master faders to design center (the shaded portion of fader travel, about ¾ up or 10 to 15 dB from the top). Now have someone talk into a mic, and slowly bring up its fader. Provided that the mic is assigned to the main output channel(s), you should be able to hear the mic over the house speakers. Repeat this procedure for each microphone and DI box.

INPUT LIST

1	KICK	RE-20
2	SNARE	CM-200
3	RACK TOMS	421
4	FLOOR TOM	421
5	OH L	SM-81
6	OH R	SM-81
7	BASS	DI
8	KEY L	DI
9	KEY R	DI
10	GUITAR	SM-57
11	VOC L	CM-200
12	VOC C	ATM-41
13	VOC R	N/D 757
14	DRUM VOC	CM-310
15	CASSETTE L	
16	CASSETTE R	

Figure 5. A sample input list.

You might want to equalize each microphone so that it sounds the way you like it. You'll need to do more EQ when everyone plays at once.

When the band is ready to play, place all the input faders about 10 dB below design center as a starting point. When the band starts playing, if any mic sounds distorted or its LED clip light flashes, turn down the input gain trim or switch in the pad until the clip light just goes out.

Adjust the balance between faders to create a good mix. Also readjust the EQ for individual inputs and turn up the effects sends for signals that you want to enhance with effects (such as reverb or slap echo). If the band is playing in a highly reverberant room, you may need no effects — they'll just muddy the sound.

The console main output meters should peak around 0 VU when the sound system is as loud as you want to hear it. If the console meters peak above 0 VU for adequate loudness, gradually turn down the console level and turn up the power amps to compensate. If the console meters peak below, say, −10 VU, gradually turn up the console level and turn down the power amps to compensate. The monitor send and effects send should also peak around 0 VU.

If you hear hum or loss of signal, suspect the cables first. Solo each input through headphones to determine which one has the problem, and fix it (usually by replacing the cable). If all the sound is lost, refer to your system block diagram and trace the signal through your system by checking the level indicators at each stage. Note where the signal stops, and work on the problem there.

THE STRIKE

After the concert, strike or tear down the sound system. Pack the mics away first so that they won't be stolen or damaged. Wrap the mic cables in lasso fashion to prevent kinks as follows:

1. Hold the XLR-type connector in your left hand, with the holes or pins of the connector toward your wrist.

2. Grasp the cable overhand with your right hand about 3 feet from the connector, loop the cable clockwise, and grasp the top of the loop with your left hand.

3. Grasp the cable overhand with your right hand about 3 feet from the connector cable.

4. Turn your right hand clockwise upside down so that the dangling part of the cable hangs between your hands.

5. Put your hands together, and rotate your right hand counterclockwise to form a loop. Grasp the top of the loop with your left hand. The dangling part of the cable should hang between two loops.

6. Repeat steps 2 to 5, alternating between standard and inverted loops, until the cable is coiled.

7. Tie the cable with a twist tie or Velcro strips. The tie or strips should be permanently attached to the cable.

Here's another method of coiling cables suggested by Bob Heil. Wind a cable onto a cable drum (rotating spool).

Plug the next cable into it, and wind it onto the drum. Continue this process until all your cables are wound on the drum. You might want to wipe the cables clean with a damp cloth as you wind them. At your next gig, simply unwind the spool and disconnect cables as needed.

Label any equipment that failed and fix it as soon as possible.

By following these suggestions, you should be able to run a concert sound system efficiently and with clean sound. db

JOHN BARILLA

Sound at MSG—Like Standing in the Middle of a Hurricane

That's how engineer/producer Phil Antonucci describes doing live broadcast audio for the Madison Square Garden Network. Like standing in the middle of a hurricane.

MSG (a major purveyor of sports television) provides home-base broadcast operations for the New York Rangers, the New York Knicks, WWF wrestling, and a host of other sporting events. Here, the name of the game is non-stop action—grueling athletics liberally spiced with gratuitous, often violent entertainment.

This kind of cross-fertilization of careers is rather unusual in the television business.

THE AUDIO WARRIOR

Sometimes Antonucci can be found perambulating the frigid wilderness of the hockey rink, dodging the sweaty ice-age warriors and an occasional stray projectile. He does this armed only with a shotgun microphone—the staple weapon for audio engineers on safari. At other times he can be found in the relative safety of the mixing booth, making split-second decisions on simultaneous audio feeds for national television. Even here within the strategy room, the pressure is always on. Tyrannical technical directors chant commands with absolute authority. There are no second takes. To survive, an audio engineer must rely on ancient animal instinct until the final cue.

At this, one might conclude that engineering for MSG bears many similarities to its culinary counterpart: Too much MSG = one big headache! But for guys like Phil Antonucci, this is the stuff that, creatively speaking, really turns his propeller. You see, Phil Antonucci lives a double life. While one Phil Antonucci dutifully presides over the broadcast sound for these sporting events at night, the other Phil Antonucci wakes up in the morning, flips the switch in his modest electronic cottage and proceeds to write, perform, and record virtually all of the hard-driving music MSG sports fans expect to hear between segments of the action.

The idea here is to keep the viewer glued to his (or her) chair with music even more exciting than the action in the arena. Perish the thought that you might be tempted to make a sandwich and delay returning to the TV room. Even if you should tarry a just a bit too long at the refrigerator,

Figure 1. Phil Antononucci with his Tascam Syncaset 238.

Figure 2. Two Tascam Model 5A consoles are bridged together to form a 16 x 8.

Antonucci's music is designed to call you back to your position in a hurry.

This kind of cross-fertilization of careers is rather unusual in the television business. In general, there tends to be a rather strict division between so-called "above-the-line" personnel (such as writers, producers, etc.) and "below-the-line" personnel (such as engineers, technicians, etc.). Usually, success in one area totally precludes credibility in the other area, but Phil Antonucci seems to be a notable exception to that rule. For him, being on-line near the fracas and fray of the game provides the fodder from which he fashions his music. Being at center court every week, sensing the unbridled excitement of the crowds (something he admits occasionally verges on being "scary"): all of these things course through his veins, lodge in his memory and feed the creative muse of sports music. This dynamic balance seems to infuse his music with an immediacy and verisimilitude only possible for one who is constantly on the scene.

Reflecting on the diversity in his career, Antonucci notes: "The experience just came from doing it. Being thrust into a situation saying, 'Now you gotta deliver pal, let's go!'—I love that kind of challenge. The excitement of a hockey game or basketball game gets you up. When you're there, when you're standing on the court and the place is exploding...some nights it's like standing in the middle of a hurricane! It helps to see it. When you're actually out there with the players it really does have an effect. Instead of just imagining it, you're there."

The increased fees and new accounting procedures were perceived as a real drag by the television networks...

SPORTS MUSIC

So how the heck do you define "sports music," anyway? It's a relatively new word in the proliferating vocabulary of production specialties. Only a few years ago the field of sports music, per se, didn't really exist. Sure, they used music between and under sports telecasts, but most of it was stock stuff straight out of the cans or a snatch of some well-known rocker off the Top 40 charts.

According to Antonucci, until recently, networks were able to use any popular song they wished, provided they paid an all-purpose annual fee to the various performing rights organizations. But times change, and usage of both current and classic popular tunes skyrocketed in fast-paced television-like news and sports shows. BMI and ASCAP (the performing rights organizations) didn't figure on such heavy usage when they designed the original fee, so they got together and changed it to make it more profitable to their writers. Instead of the blanket fee, they instituted a pay- per-play type of system. The increased fees and new accounting procedures were perceived as a real drag by the television networks, and many of them decided it was time to commission some original

Figure 3. A Commodore 128 provides MIDI control.

music which they could use without restriction.

Enter PAS Music (Phil Antonucci Songs). Making them an offer they couldn't refuse, Antonucci (and the two partners he had at that time) designed a one-year package providing MSG on a season-by-season basis every musical item they could possibly desire, all cut to spec. What they required broke down to basically three categories:

a) Feature Pieces—Three or four minutes of intense music paying tribute to a particular team or some individual star's athletic prowess. These special segments are usually retrospective, highly motivational video/musical blowups of some heroic exploits from a previous game. The music is usually played constantly throughout the segment, sometimes under the narration, and sometimes blasting through to underscore a great play.

Figure 4. Phil Antonucci poses among the keyboards and console.

For Phil Antonucci, it's liberating not to have to spend half a day deriving sufficiently complex sounds.

b) Theme Songs—Shorter pieces (15 seconds on up to about a minute) which set the tone for a regular pregame segment (such as "Coaches Corner," which is a weekly prognosis from the teams' head honcho).

c) Bumpers—30-second pieces played between segments of the programming to make transitions smoother. These could be used at "time-outs" or between shows. Whenever there is a possibility of the dreaded "dead air" syndrome, usage of a bumper is indicated. This is doubly true in sports television. It must always be made to appear that something is happening—even when nothing is really going on.

In constructing his agreement with MSG, Antonucci wisely provided for the future of his company by retaining the copyrights to his compositions. Essentially, MSG is granted a license for exclusive use of the pieces over a specified period of time after which control of the songs returns to Antonucci. While the songs are constructed specifically for MSG, he will be able to build a library of his pieces and sell them to another market after the licensing period has elapsed.

How does one go about writing and producing sports music? First, Antonucci meets with the producer of the segment in question. The producer may show him clips of the sequence he wants scored—if they are available. Usually, they are not. So the producer throws around some images, ideas or even specific pieces of music which communicate a similar message. As a result of this evocation, Antonucci will construct a piece of music, and the video cuts will often be made to the music, rather than vice versa. It's usually the most efficient way of getting the final product. Sometimes however, he does receive a "hit list" with crucial timings. At the specified number of seconds into the piece, a transition of some sort (a cymbal crash, drum fill, tempo or key change, etc.) is called for.

It's all self-contained (multi-timbral mixing and effects are done internally).

Occasionally, the timings will need to be changed after the composition has been written. A producer will have everything (seemingly) laid out in his script or storyboard, but after assembling some of the footage and inserting the DVE (digital video effects), he may feel that for maximum impact the timings should be shifted. A few years ago this would have been a bit of a trauma for a composer. It would have required a lot of razor blade editing and a compromised musical product. Today, Antonucci simply reassembles the building blocks of the composition in his MIDI sequencer. Antonucci's creative methodology involves first visualizing the movement of the sport in question and combining that with the tone and pacing of the proposed video cuts. Holding this information in his mind, he then proceeds to find a rhythm which expresses the essential message. The key tool here is the drum sound. Antonucci becomes positively galvanized by hearing a particular sound, the sound suggests a rhythm, the rhythm suggests a bass line or a keyboard harmony and so forth.

His choice for drum sounds come from the selection in Korg's versatile M-1 synthesizer. Antonucci waxes effusive about that unit:

"The M-1 is the staple of this studio. It's an incredible machine with its combination sounds. The beauty of the M-1 is that it takes up just two channels on your console. It's all self-contained (multi-timbral mixing and effects are done internally)."

THOSE TWEAKING SOUNDS...

It is this kind of all-in-one "workstation concept" which has made the M-1 so incredibly popular. For Phil Antonucci, it's liberating not to have to spend half a day deriving sufficiently complex sounds. Like many busy producers, he does not enjoy tweaking sounds. For him, it's a diversion from the creative path. Instead, he wants to plug in, audition several sounds from the M-1 library, and be able to say, "Yeah, that's a great sound," and move on to the next track.

While some purists may balk at the fact that Antonucci does not care much for customizing sounds, from the practical point of view it's a position that cannot be knocked. Productivity in the studio soars when one is liberated from a task. Electronic cottage operators like Antonucci tend to be "one-man-shows," often writing, performing, producing and engineering simultaneously. It's a tremendously rewarding way to create music, but it has its limitations in terms of how much energy one can invest in a particular aspect of the project. Since signal processing is often included as part of the sound with units like the M-1, fewer pieces of outboard equipment are necessary to come up with a professional sound. This is both cost-effective and time-efficient. Of course there are compromises to be made along the path, but when working under an intense deadline, it's understandable why one would prefer a sound module with great out-of-the-box sounds, rather than one with deep programmability.

I listened to some of Antonucci's music. It was motivational music of the highest quality. Even within the context of sports music there seemed to be a diversity of styles reflecting the image of the particular sport. While music for the Rangers was bold and aggressive, music for the Virginia Slims Golf Tournament was cool and cerebral. All of it was produced with an intense, high-tech sound that seemed so much bigger than the appearance of his modest facility would lead one to believe.

All Antonucci's MIDI production is done on a Commodore 128 running Sonus software (*Figure 2*). It is not by any stretch of the imagination an upscale system, yet it does everything that Antonucci needs it to do. That, of course, is the bottom line. On producing with his system Antonucci comments:

A respected composer/producer and audio engineer, he should serve as an encouragement to electronic cottage owners throughout the world.

"My hardest thing is starting. I'll try to come up with a verse, and create the whole thing in one or two sequences. Then I come up with a release. The form in sports music is A-B. There is no chorus (per se) in sports music. It's just 'bed' music (under the narration) and a release, with an intro and ending. Maybe five or six parts maximum which can be edited to make several variations on the same theme. Now I can edit those parts anytime I want. For instance, I might make the end of the song out of a verse. I'll take the verse and copy it to a different memory location and I'll edit it all right here (in the computer) without ever having to play any more. That's the beauty of these things."

BEWARE OF TRAPS

"But one trap you can get caught in is making it sound too mechanical. So you have to concentrate as much as you can on keeping the flow of the music happening. Once you piece all your various sequences together, you want to go back and add things to each sequence so that when the sequences change it's actually flowing. That really brings things together!"

"What I strive for all the time is to make it sound like it's being played. What puts me over the edge (towards realism) is being able to add guitar." (Antonucci is a skilled guitar player holding degrees in music and performance).

"It's the live element—the wildcard—that's not electronically linked, but humanly played."

In order to integrate these wildcards with the MIDI sequencer, Antonucci locks his computer to a Tascam 238 Syncaset (8-track recorder) using standard FSK tape sync. Since he is writing theme music—not doing the audio layback to video tape—SMPTE time code is not required. Asked how he got started in the field of recording, Antonucci confessed that he gleaned most of his practical knowledge from the pages of **Modern Recording and Music Magazine.**

"Starting around 1977," says Antonucci, "when I bought a 4-track machine, I wanted to know how to use it. I found out a lot of information from that magazine. Believe it or not, I was religious with that magazine, and I taught myself just about everything. I learned it by doing it. By reading and then doing it. That's the best way."

Phil Antonucci has come a long way using the method of reading and doing. A respected composer/producer and audio engineer, he should serve as an encouragement to electronic cottage owners throughout the world. The message is this: You don't need every toy in the store to make effective music. Every time you tune in to a Rangers or Knicks game on national television, think about what Phil Antonucci does in his unassuming electronic cottage. It's not what you got. It's how you use it that counts! db

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People, Places... & Happenings

• **HM Electronics** has named **Eunice Davis** as Pro Audio General Sales Manager. Davis had been Regional Sales Manager since April, 1987. Prior to her work with HME, she operated her own rep firm for seven years, successfully handling several Pro Audio manufacturers. She will direct domestic and international sales. The company has also named **Randy Opela** as Marketing Product Manager after two years of service to the company. He will be responsible for new product development, product research and product introductions of the Pro Audio Line.

• **Jerome E. Werner** has been named President of **dbx**, Inc., replacing Michael Kelley who resigned at the end of February. Mr. Werner was Vice President and Chief Operating Officer at Klipsch & Associates for the past eight years before joining **Carillion Technology**, dbx's parent company.

• **Rupert Neve** has joined **Amek** to develop new and advanced ranges of equipment which will incorporate the qualities and feature for which he is renowned. He will also make some circuit enhancements.

• **Al Zang** has become manager of Professional Products for **Sennheiser Electronics Corporation**. Zang, a fifteen year veteran of the audio industry, will help Sennheiser continue to develop their overall direction of professional products.

• **Trident** has appointed two new sales managers—**Mike Abercrombie** in the US and **Kim Templeman Holmes** in Europe. Abercombrie worked for Neve, and prior to that, he handled sales for **Sony Digital Products** for six years. Holmes also worked for Neve. In addition to sales experience, he worked as a sound engineer.

Milton T. (Bill) Putman, founder of **Universal Recording** in Chicago, **United and Western Recording** in Hollywood, **Coast Recorders** in San Francisco, and **United Recording Industries** passed away on April 13, 1989. He was 69.

Bill was involved in research projects involving all aspects of electronics, acoustics, and Amateur Raio. He created the **UREI 800** series monitors, the **1176 limiter**, and the first low noise tube microphone premplifier. He pioneered half speed record mastering, the **"Double Feature"** record, and was involved in many of the top ten record releases in the forties, fifties, and sixties. For many years, he was the only engineer that Frank Sinatra would allow in the booth.

From the beginning, Bill shared his knowledge with everyone. He wrote numerous articles for trade magazines, conducted seminars and classes, and his advice was available to anyone who would ask. Bill was an instructor at the BYU Audio Recording Seminars in the 1970s, and is the recipient of their Harris Fine Arts Award. He was a Fellow of the AES, and was an officer of the Chicago Acoustical and Audio Group, he was a member of the Hollywood Sapphire Group and had "Grammy" nominations from NARAS.

Bill Putnam not only taught us how to record, but he taught us how to build our studios, and he designed and built the equipment to make them work.

We thank our friend Shelly Herman for the above.

• **Mark Johnson** has been appointed as Director, Technical Marketing at **Meyer Sound Laboratories** Inc. He's been with the company since 1987. Previously, he has worked at **Opryland USA** Inc. as audio engineer, sound designer and Technical Coordinator.

• **Neve** has been named the exclusive distributor of **Mitsubishi Electric professional digital audio products**. The Mitsubishi product line includes three 32-track digital tape recorders—the X-880, X-850, X-800, the 16 track X-400, and the four two-track recorders—the X-80, X-86, X-86C and X-86HS. The XE-2 digital audio editor is among the complimentary ancillary equipment also available.

• **Shinji Miyata**, President of the **Mitsubishi Pro Audio Group**, and **William Windsor**, President of **Quad Eight Electronics**, a new California corporation, have announced that Quad Eight Electronics, Inc., has purchased the assets, goodwill, engineering, and manufacturing facilities of the Quad Eight audio mixing console operation of the Mitsubishi Pro Audio Group.

Quad Eight Electronics, Inc., has also assumed all warranty and non-warranty service obligations of previously purchased consoles manufactured by Quad Eight/Westrex and sold to customers in North America and the Far East. Digital Entertainment Corporation UK will continue the service obligations of all consoles previously sold in the United Kingdom and Europe, and will be supported by Quad Eight Electronics, Inc..Quad Eight Electronics, Inc. will continue to manufacture the **Westar** console under the **Virtuoso** name.

• **Drew Daniels**, formerly applications engineer with **JBL**, has accepted a position with **Walt Disney Imagineering** as a senior member of the technical staff.

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