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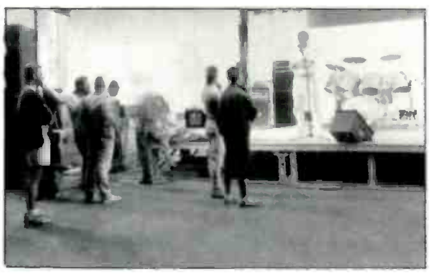
The sound contracting engineer



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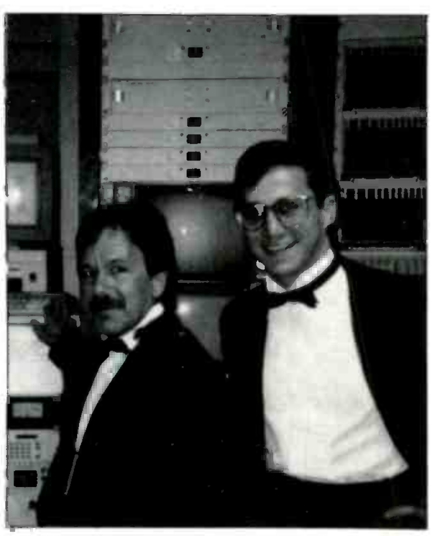
THE ELECTRONIC COTTAGE

The broadcast engineer



see page 56

About the Cover

● The National Symphony Orchestra in performance at the West Lawn in Washington, D.C. See Ed Learned's article on outdoor reinforcement for symphony orchestras, beginning on page 30.

Calendar

• **Synergetic Audio Concepts** (Syn-Aud-Con) of Norman, IN, has announced **Intelligibility Workshop II** to be conducted under the supervision of Dr. Larry Humes of Indiana University in Bloomington, IN, May 24-26, 1990. The workshop will deal with the measurement of speech intelligibility and its uses in planning sound reinforcement systems with acceptable intelligibility characteristics.

Intelligibility II will employ new technical tools to advance the understanding of the participants with respect to the proper measurement of speech intelligibility of a sound system. Special emphasis will be made to understand the role of the pinna and ear canal in intelligibility; therefore, hearing threshold as well as pinna response measurements will be made on each participant in the workshop. In-The-Ear recordings will be made in the pressure zone of the eardrum of several participants of the workshop. DAT recordings will be made of 5, 10, 15, 20 and 25 percent articulation loss of consonants.

Attendance at the Intelligibility Workshop will be limited. The workshop will be held in Bloomington, IN, approximately an hour south of the Indianapolis airport.

For further information, please contact: Synergetic Audio Concepts, R.R. #1, Box 267, Norman, IN 47264. Phone: (812) 995-8212. Fax: (812) 995-2110.

• **ASTM Committee Meeting on Environmental Acoustics** (E-33) to be held on April 30 to May 2 1990, at the Stouffer Harborplace Hotel, Baltimore, MD. **For more information, contact Stephen Mawn at (215) 299-5521.**

• **ASTM** is also sponsoring a two-day trainingseminar on Architectural Acosutics. It is designed for architects, designers, engineers, and it includes laboratory demonstrations and a thorough review of fifteen related ASTM standards.

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STEVEN C. RYPKA

The Las Vegas Hilton Showroom Audio System

Sound reinforcement systems are becoming increasingly sophisticated as demonstrated by the new installation in the Las Vegas Hilton's main showroom.

As the Hilton's head audio engineer for the past eight years, I have tried to improve the performance of the showroom audio system whenever possible. Many of these improvements were made possible with the help of audio technician Bud Wolfe, the other half of the showroom sound crew. We recently finished a major upgrade of the front of house system that uses some new developments in both hardware and software to achieve many interesting capabilities.

A LITTLE BACKGROUND

Although the showroom in the Las Vegas Hilton is the largest in Nevada, it is a fairly intimate venue. The room seats approximately 1,600 persons, with about 375 of those seats located in a balcony. The stage is large, with a 65-foot main curtain in the center with 30-foot side stages on each side, and is about 85 feet deep. The house sound booth, ideally located, is on the center line of the auditorium and just underneath the front edge of the balcony. A custom frame houses six 24-inch-tall racks of outboard gear and provides support for two Yamaha PM-3000 consoles.

The 32-channel console's auxiliary, group, and stereo bus outputs are normalled to the respective subinputs of the 40-channel console's buses. The master Mute and VCA controls can be operated independently or as one master section that controls both boards (from either console). This 72-input configuration allows great flexibility for large shows, co-headliners, and special events.

Effects devices include an Eventide H3000 Ultra-Harmonizer, a Lexicon PCM-70, two Yamaha REV-5's, and two Yamaha SPX-90 II's. These units are linked via MIDI to a Lexicon MRC midi remote controller, allowing quick program changes of all units simultaneously. There are two dbx 166 gated compressor/limiters and two dbx 900 racks equipped with eight 903 compressors, nine 904 noise gates, and a 902 de-esser.

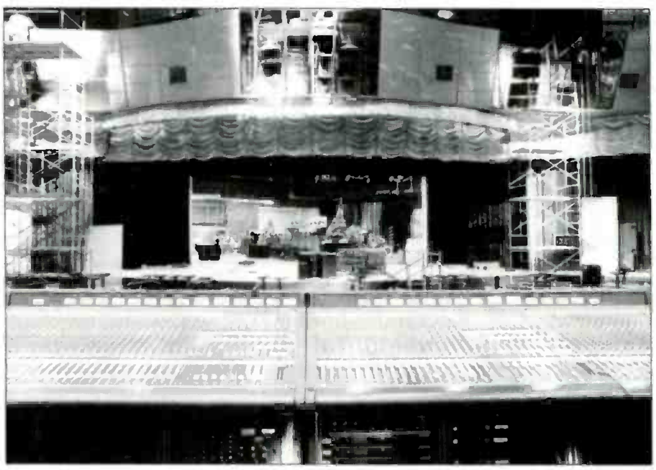


Figure 1. A view of the Las Vegas Hilton Showroom stage and speaker rigging from the double console position.

A twenty row by twenty-six position patchbay allows flexible equipment configurations and includes mic lines from both 50-pair snakes (custom built by A-1 Audio) as well as nearly 100 additional mic lines from other parts of the room. Tape machines include an Otari MX-5050-BQ-II 4-track reel-to- reel, a Broadcast Electronics Five-slot cart machine, and a Nakamichi cassette deck.

A third PM-3000 (40 channel) is used as a monitor board. The monitor system contains 12 Klark-Teknik DN-360 1/3 octave equalizers, 4 TDM 24CX-4 crossovers, and 14 Crest 4001 power amplifiers. An assortment of bi-amplified wedges include Meyer UM-1's and custom cabinets using JBL components. These cabinets were fabricated by A-1 Audio and include both single 15-inch and double 12-inch configurations, all with 2-inch compression drivers. The side-fill system consists of two Meyer MSL-3 cabinets, each mounted on top of a Meyer USW-1 subwoofer cabinet.

A wide selection of microphones includes models from AKG, Beyer-Dynamic, Countryman, Crown, Electro-Voice, Mylab, Sennheiser, Shure, and Yamaha. This configuration of the house soundbooth and monitor system has been functional for about two years.

A NEW PROJECT

The latest improvement to our system was a complete upgrade of the front-of-house components, including equalizers, delays, crossovers, amplifiers, and the weakest link in the chain, the speakers. More than a year of research included gathering information, studying new equipment, and discussions with other sound engineers that came to the Hilton. Some of our best information came from these outside engineers, who shared their thoughts and ideas on what would constitute the best possible system.

Figure 2. A closer view of the house sound booth showing the effects racks below the console and, at top left, the control computer.

The decision was made to do this project completely in-house, for the following reasons; we had the full support of Hilton management, we would be able to make the most of our budget by utilizing in-house labor, and the technical knowledge and ability of our crew was top-notch. We received the news of our budget approval in late July of 1989. From that point on we went into high gear. The design goals of the project were fairly basic: full-range frequency response, smoother coverage throughout the room, increased dynamic range (headroom), and maximum flexibility of the system.

SPEAKER SELECTION

The first decision concerned which type(s) of speakers to use. I decided to utilize a full-range, pre-designed, arrayable speaker cabinet system rather than trying to form coherent arrays from lots of separate components as was previously done. To help make the final decision on the main cabinets, I was able to arrange a "shootout" on the Hilton stage between four different manufacturer's speaker systems.

Although none of the systems tested performed poorly, we choose the Eastern Acoustic Works KF-850 system for its natural, clear sound, and smooth coupling (equipped with factory-standard RCF drivers for mids and lows, TAD 4001 drivers for the highs). Matching E.A.W. SB-850 double 18-inch subwoofer cabinets were chosen to extend the main system to the lower octaves. We decided to use Meyer Sound Laboratories 650-R2 double 18-inch subwoofers for our separate sub-bass effects system in the side walls.

Apogee Sound AE-2 loudspeakers were chosen to cover our under-balcony areas. These speakers have remarkable punch and clarity in a small, low-profile design. In addition, the smaller E.A.W. KF-300 speakers were specified for both rear wall effects and for overhead fill speakers directly over the front edge of the stage.

ELECTRONICS SELECTION

For a number of years I have been deeply involved with personal computers and programming and had been looking for ways to incorporate the power of a personal computer into the everyday operation of a large sound system. When I learned about the Crown IQ system, which uses a computer to control and monitor multiple amplifiers, I knew things would never be the same. The engineers at Crown have come up with a very valuable tool that points to the future of modern sound systems. Further research and A/B comparisons with other "industry standard" amplifiers confirmed the high quality of the Macro-Tech amps. The Crown products were chosen to power our new system.

T.C. Electronics 1128 ⅓ octave programmable equalizers were chosen for their excellent audio quality as well as their many other powerful and unique features. If our new system was to be flexible, I felt that the equalizers used for the front of house system had to be more than the standard manual ⅓ octave graphic. Storage and recall of preset curves would enable any engineer to have his exact EQ settings, and still have my standard "house" EQ curves available when needed. The 1128s also have a well-executed computer interface that makes them even easier to use.

We followed the manufacturer's recommendations concerning frequency dividing networks (or processors): E.A.W. MX-800 units for the E.A.W. KF-850/SB-850 cabinets, BSS FDS-360 units for the KF-300 speakers, and devices from Meyer and Apogee for each of their respective speaker systems. Klark-Teknik digital delays were selected for the two-zone under-balcony system and for the side and rear effects sends.

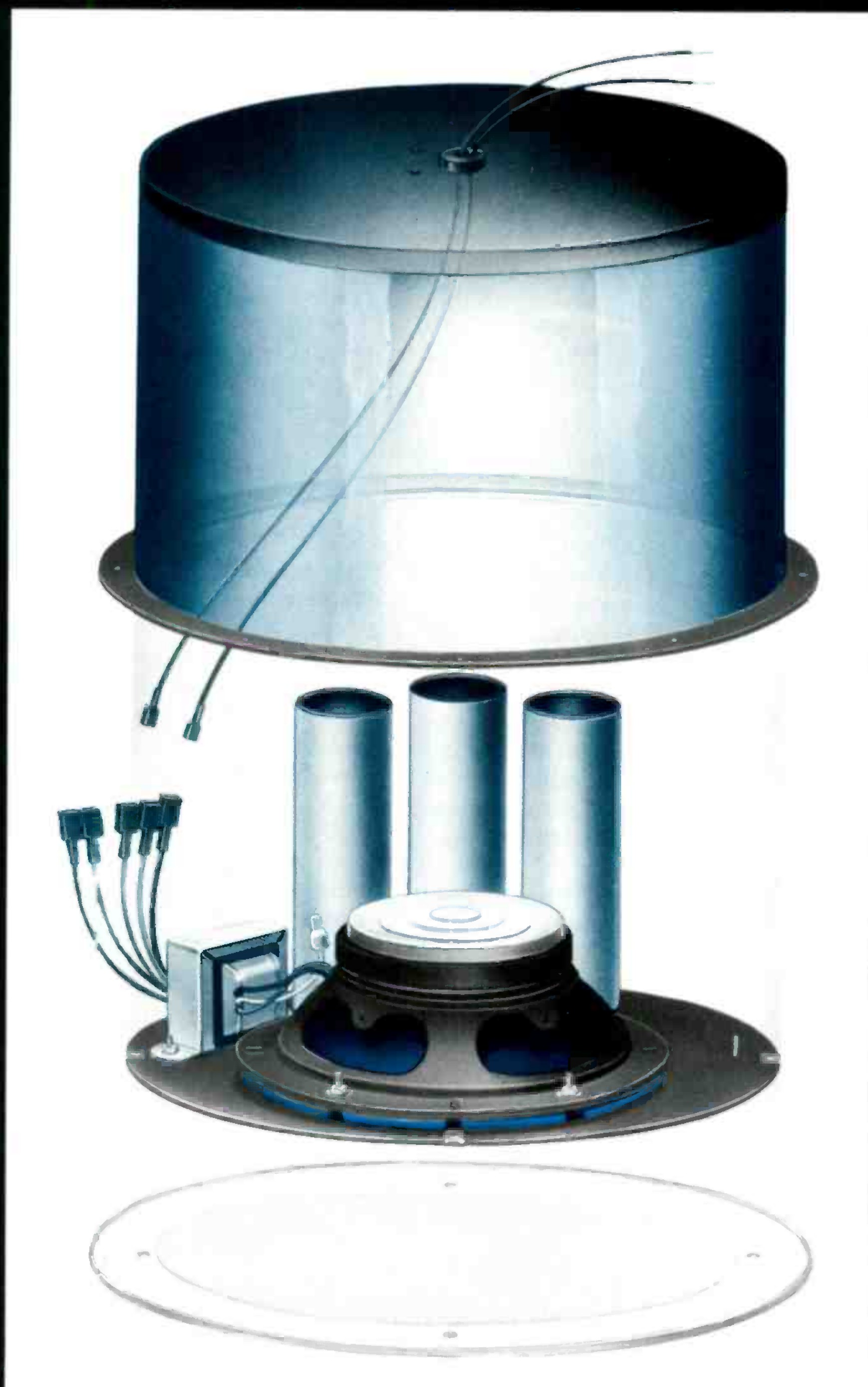
SYSTEM DESIGN

The previous sound system included three overhead clusters, left, center, and right. The components of that system were stacked on steel platforms that hung approximately twenty-four feet above the edge of the stage. In addition, there were speakers for side and rear effects and a 70-volt under-balcony system. Since the old clusters were already ideally located, we maintained the same basic positioning, improving the coverage, zoning, and of course, the components themselves. It was decided that each overhead cluster would contain four KF-850 cabinets, two for the balcony and two for the middle of the main floor. Each cluster would also contain two KF-300 cabinets aimed almost straight down as near-fill for the area just in front of the stage. The left and right clusters would also house four SB-850 subwoofer cabinets each.

An additional set of two KF-850 and two SB-850 stacks were specified for each side of the stage to be used as an optional stage-level fill system. This was done to satisfy those engineers who required a lower image source for their mix. The under-balcony coverage split the ten Apogee AE-2 speakers into two separate delay zones of six and four cabinets each. Two Meyer 650-R2 subwoofers and one KF-850 cabinet were designated for sub-bass and side wall effects. Four additional KF-300 cabinets were specified for the rear wall effects system.

Since maximum zone control was necessary to attain the smoothest coverage, we used one amp channel per driver throughout most of the system. The only exceptions to this are the two 18-inch drivers in each subwoofer cabinet which are driven by one amp channel, and the monaural under-balcony system where we use a total of four amp channels for the ten 16-ohm AE-2's. Crown MA-1200 amplifiers were assigned to power all high-frequency drivers with MA-2400's for the mids, lows,

Ceiling Dynamics



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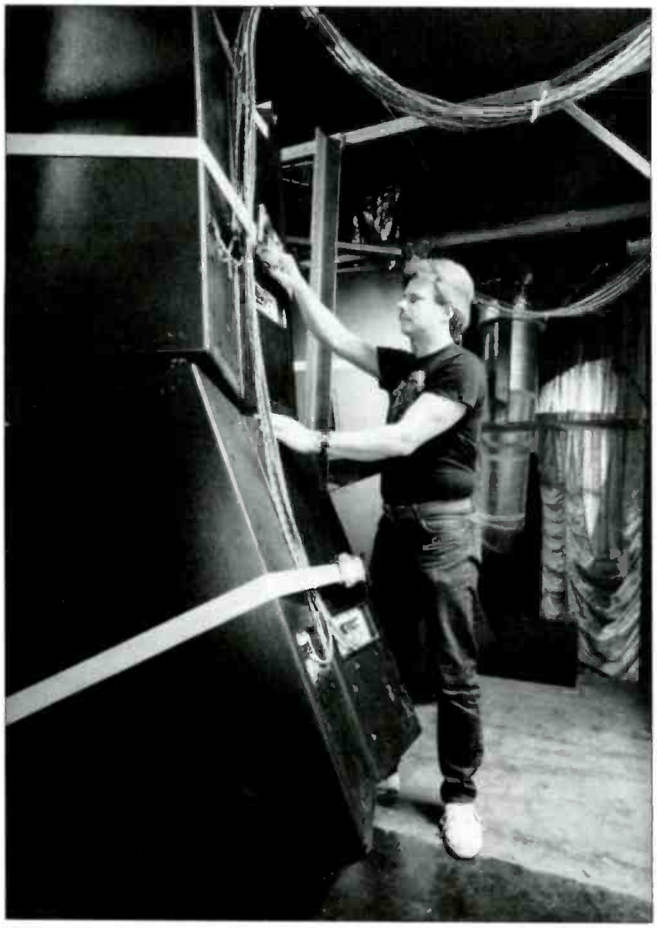


Figure 3. An example of one cluster of the KF-850s, this one at the house right. The author is seen making adustments to the rig.

and subs. We specified a total of eighteen E.A.W. KF-850's, twelve SB-850's, ten KF-300's, ten Apogee AE-2's, four Meyer 650-R2's, and thirteen Crown Macro-Tech MA-1200 and 33 MA-2400 amplifiers. A new, computer-grade, three phase, 112 KVA isolation transformer was ordered to provide over 300 amps of clean A.C. power to the system including a 200 amp disconnect, used to supply external A.C. power distribution systems.

With the new capabilities of the Crown IQ system, we decided to locate the amplifiers as close to the speakers as possible. The short cable runs would maximize power transfer and damping factor, giving us the best possible performance, while the IQ system would enable us to maintain total control over the remotely installed amps. Since most of the system would be over the stage, we needed a home for five racks totalling twenty-eight amplifiers. This led to the design and construction of two overhead amp rooms located directly above the left and right main clusters.

Along the way, a new product was brought to my attention that was very interesting. The device, called B.A.S.E. which stands for Bedini Audio Spatial Environment, was tested on the old sound system and proved to be a fabulous device that really worked as claimed. The unit enhances stereo signals, giving a greater sense of depth and texture to the mix. Since the Hilton Showroom lends itself well to stereo sound, and the B.A.S.E. units actually enhance separation for more of the audience, three units were incorporated into the system.

As this design was finalized, all aspects of the system were drawn on the computer using CAD software (see example). Multi-pair wiring connections, to be done on telephone punch blocks, were laid out on spreadsheets that contained a description of every wire's function, its color code, and number. Price information was also stored in a spreadsheet, enabling us to track our progress and optimize the system design while staying within the budget.

THE ACTUAL INSTALLATION

The entire project spanned a six-month period and consisted of many different phases, many of which overlapped considerably. We actually started the installation shortly after our budget approval, even though the design was far from complete. I used project management software to help schedule the different segments of the job. The actual hanging of the clusters was scheduled for a dark period in December, just prior to Christmas. Because we had the time, Mr. Wolfe and I were able to do most of the wiring work ourselves. Our intimate knowledge of the existing system enabled us to make good use of existing conduit runs. Thousands of feet of old speaker cable were removed from its conduit to make way for the A.C. and line-level runs to the new overhead amp rooms. An electrical contractor was hired to install the new A.C. isolation transformer and a 400 amp disconnect to feed it.

Using our spreadsheet layouts of the punch blocks, the multi-pair line-level connections were made. This process went very smoothly, having been triple-checked on the computer prior to installation. The A.C. power requirements of the amplifiers were calculated and each portion of the system was balanced between the three phases. Three separate 30 amp A.C. circuits were pulled to each amplifier rack.

Seven T.C. Electronics 1128 units were installed in the house booth for equalizing the three main clusters, under-balcony system, stereo effects system, and a patchable spare. Another addition was the Northgate Computer Systems 20 MHz 80386 computer equipped with 4 MB of RAM, a 67 MB hard disk drive, a high-resolution mouse, and a color VGA monitor. The remaining electronics, (EQ's, delays, and crossovers) were installed in a rack on stage left.

When the E.A.W. shipment arrived in Las Vegas at A-1 Audio (a major vendor for the system), the speakers were stored in their warehouse. A-1 graciously let us use their facility for pre-testing of the speakers. A TEF analyzer, brought in by Steve James and Jim Fox of Electro-Acoustics in Las Vegas, was used to confirm the response and proper polarity of all the drivers.

In December, the old system was taken down and the new one was hung. The rigging design uses a swivel-frame/trolley assembly attached to a structural I-beam. This allows each cluster (four cabinets per trolley assembly) to be pulled back over the steel floor and then swiveled

Figure 4. These racks of Crown Macro-Tech amps are one of two such located overhead, just above the house right cluster.

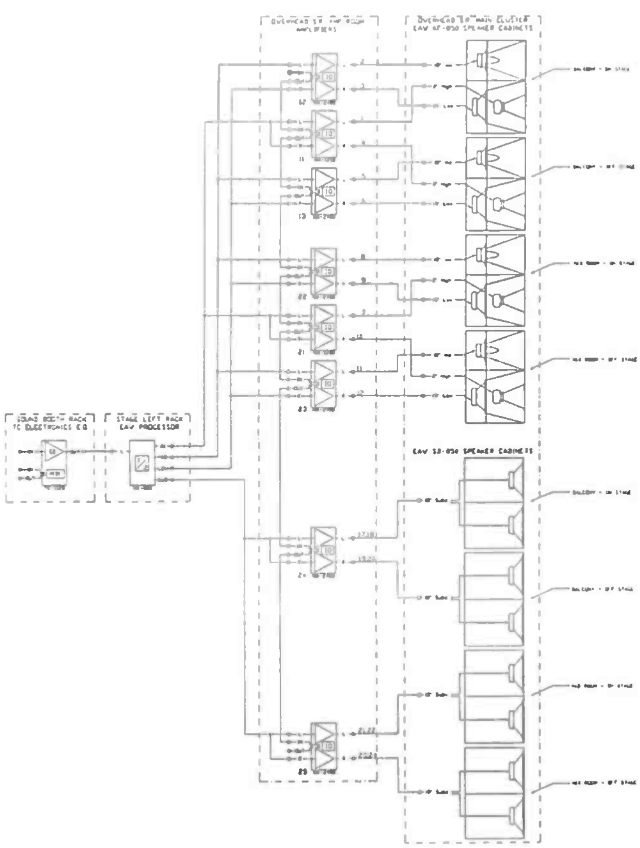


Figure 5. A computer drawing by the author of the house left main cluster.

ninety degrees for necessary maintenance. Before installing the IQ system software, we tuned the system for a maximum operating level using the amplifier inputs, adjusting the gains so that even without the computer control, the system would default to the manually set levels. Limiter thresholds in the various processors were set to allow maximum safe operating levels before limiting.

HOW WELL DOES IT WORK?

I feel that all of our goals for this project have been met and that the new audio system performs as expected. Very little equalization is necessary to get a smooth, full sound. Reaction of audiences, entertainers, and engineers has been extremely positive. The system is very quiet, yet will cleanly reproduce the peaks of a live show with ease. The real news, in my opinion, is the flexibility and control that the computer has brought to the system.

Now we can use the IQ system to load a preset from the disk to configure the amplifier settings for the entire room in a matter of seconds. If we have a full house, we run with zero attenuation; if the balcony is empty, we attenuate the amplifier channels driving that part of the room. We no longer have just one mode of operation for every show, and every show has the potential to sound just that much better. The repeatability of these settings is precise.

We can also monitor every amplifier's operation, from the sound booth, and know exactly how much headroom we have or if there is a problem with part of the system. I can mute every amplifier, feed pink noise to all outputs, then un-mute each amp channel momentarily in succession to verify the operation of every driver in the system in a matter of minutes.

Using the computer to control the T.C. Electronics equalizers is also a wonderful experience. The ability to save and load presets of a complex equalization curve is marvelous. For example, before changing a curve it can be copied it to a new address. Then it can be altered and compared to the original by switching back and forth. Other benefits include the ability of the computer to calculate and trace the additive effects of each band's setting; and the integral spectrum analyzer for each unit that will show you problem frequencies immediately. Having all of this capability on a color monitor that you can read easily under almost any lighting condition is fantastic.

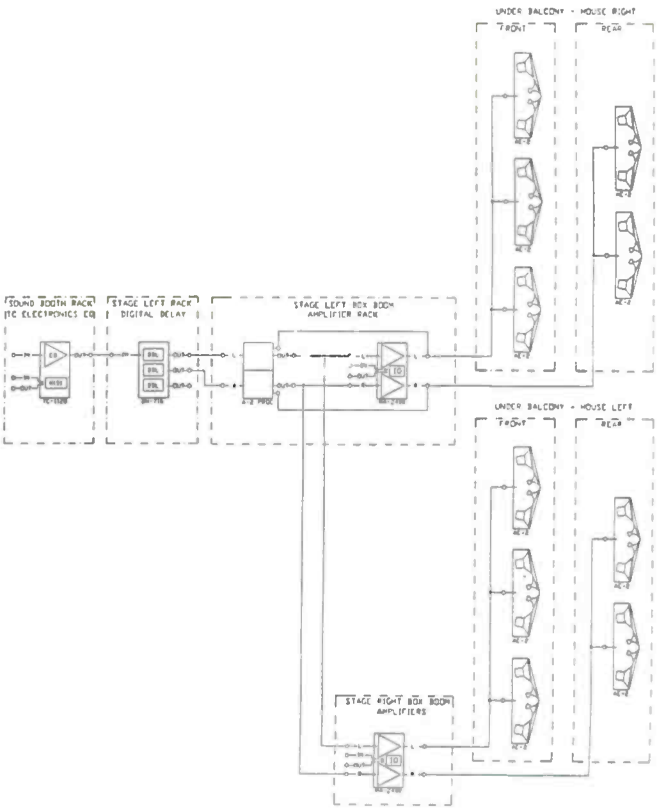


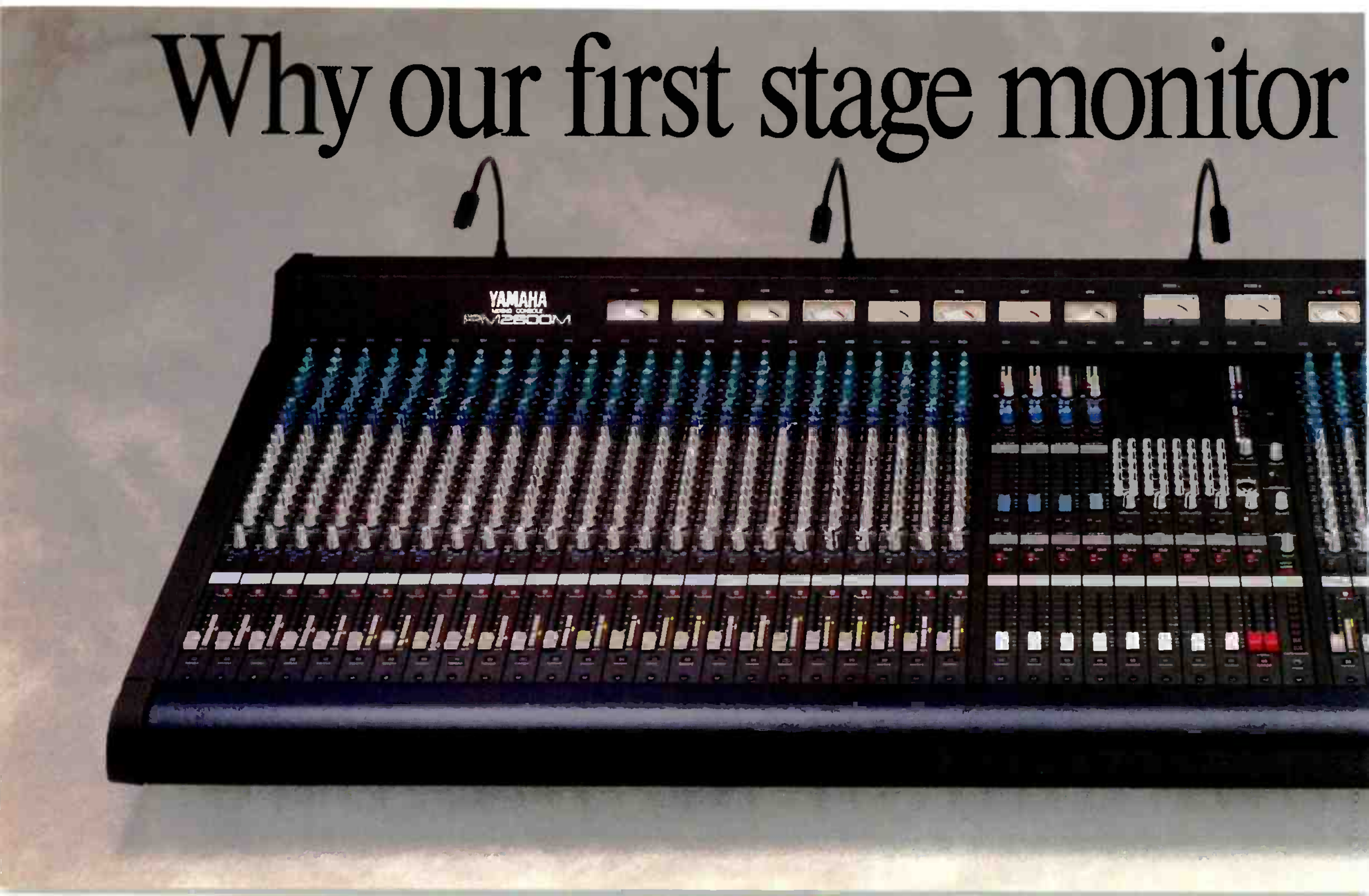
Figure 6. Another computer drawing this time showing the under- balcony fill system.

I have set up the computer with multi-tasking software called DESQview 386 from Quarterdeck Office Systems. This windowing control program allows me to run many programs at once on the 386 machine. One window can run the Crown IQ System software, while at the same time another window is running the T.C. Electronics program. Switching between programs is instantaneous and both windows can be displayed simultaneously if necessary.

An additional Sequencer/Editor/Librarian program controlling a MIDI interface card is now connected to our six effects devices. This allows saving of all user presets to disk for future retrieval, as well as performing complex program changes in real-time during a show. We all know that computers are not totally infallible, in fact they can be very susceptible to power surges and other problems. Because of this, we have incorporated this technology in such a way that if something should go wrong with the computer, the system will still be 100-percent operational.

CONCLUSION

The marriage of new audio equipment with the PC has increased the capabilities of our sound system tremendously. The audio quality of the



new system is excellent, with plenty of headroom for any live performance. All of this power, used creatively and with taste, can greatly enhance the enjoyment of our showroom customers, and make the engineer's job that much more enjoyable and fulfilling.

Las Vegas Hilton Showroom Sound System Specifications

HOUSE SOUND EQUIPMENT

House Sound Booth:

There are two Yamaha PM-3000 consoles (one 40-channel and one 32-channel) with all 72 channel inputs, plus 6 stereo aux returns available. These consoles can operate independently or linked together, sharing VCA and mute masters, aux and stereo bus outputs. The sound booth is centrally located in an ideal position.

Processing devices available include:

2 dbx 166 gate/compressor/limiters
6 dbx 903 compressor/limiters
9 dbx 904 noise gates
1 dbx 902 de-esser

Effects devices available include:

1 Eventide H3000 Ultra-Harmonizer
1 Lexicon PCM-70
2 Yamaha REV-5's
2 Yamaha SPX-90 II's

Tape Machines available include:

1 Otari MX-5050-BQ-II, ¼-inch, 4-track reel-to-reel
1 Broadcast Electronics 6 slot cart machine
1 Nakamichi auto-reverse cassette deck

Main PA System:

The PA system consists of three overhead clusters (left, center, and right). Each cluster consists of four Eastern Acoustic Works (E.A.W.) KF-850 (large, 3-way, tri-amped, full range) cabinets and two E.A.W. KF-300 (small, 3-way, bi-amped, full range) cabinets. The left and right clusters also have four E.A.W. SB-850 (double 18-inch) subwoofer cabinets each.

Optional, stage level, left and right stacks consist of two E.A.W. KF-850 and two E.A.W. SB-850 cabinets per side.

The under-balcony system is a monaural composite of the left/right stereo mix, in two delay zones, consisting of a total of ten Apogee AE-2 under-balcony speakers.

A separate aux bus feeds a sub-bass enhancement system consisting of four Meyer 650-R2 cabinets located in the side walls of the showroom.

Another aux bus feeds a surround-sound effects system consisting of two E.A.W. KF-850 cabinets (side walls) and four E.A.W. KF-300 cabinets (rear wall).

MONITOR EQUIPMENT

There is one 40-channel Yamaha PM-3000 console, providing nine discrete mixes and eight additional mixes via the Matrix outputs. There are twelve Klark-Teknik 1/3 octave equalizers and twelve bi-amp speaker outputs. There is one Yamaha SPX-90 II signal processor available.

Monitor speakers available include:

4 JBL bi-amped, 3-way, single 15-inch wedges (A-1 Audio design)

4 JBL bi-amped, 2-way, double 12-inch wedges (A-1 Audio design)

4 Meyer bi-amped, 2-way, UM-1 wedges

2 Meyer bi-amped, 2-way, MSL-3 side fills

2 Meyer subwoofers, double 15-inch (part of side fill system)

2 Yamaha 2115's, full range, 2-way wedges

2 Yamaha 4115's full range, 2-way, self-powered

Snakes:

There are two custom 50-pair snakes, each equipped with multiple sub-snakes off the main box. Each box has a hard-wired split with individual ground-lift switches on the "B" side of each pair. One snake normals into each house console. Tails are provided for patching into the monitor console(s).

Microphones:

There is a large assortment of microphones from: Beyer, Shure Bros., AKG, Sennheiser, Countryman, Yamaha, and Electro-Voice. There are eight passive direct boxes, all with Jensen transformers, and six active Countryman direct boxes.

A.C. Power:

There is a 112 KVA computer-grade isolation transformer providing power to the entire sound system. The transformer also feeds a 200 amp, 3-phase disconnect box located on stage left for connecting external A.C. power distribution systems.

Communications:

There is a multiple channel Clear-Com headset system throughout the showroom. Communication channels exist between all lighting personnel, the fly rail, the stage manager, etc. A private line exists for audio communications only.

Miscellaneous notes:

All house speakers are driven by Crown Macro-Tech power amplifiers (MA-1200 for highs, MA-2400 for mids and lows). The system is *extremely* powerful (over 40,000 watts), accurate, and has sufficient headroom for any type of music. Coverage is very uniform throughout the room. All main house feeds are via T.C. Electronics 1128 programmable equalizers. All KF-850 cabinets are equipped with TAD high-frequency compression drivers. All effects devices are controlled via the MIDI interface allowing maximum flexibility and quick program changes. All monitor speakers are driven by Crest power amplifiers. db

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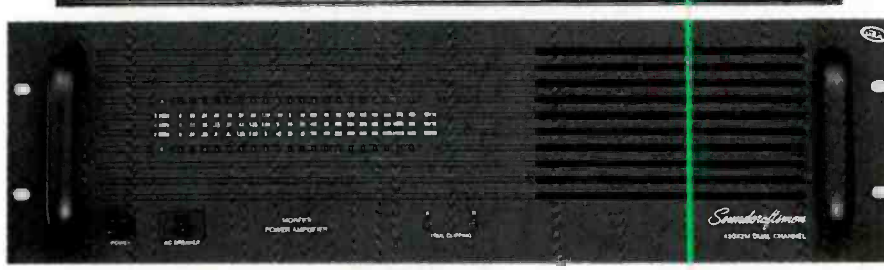

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The 1989 Newport Jazz Festival

This year's festival, held August 17th to the 20th, is the thirty-fifth in a series. It has been a rocky series, having been driven from conservative Newport, Rhode Island by large crowds in the late 60s, brought by the likes of Jethro Tull and Led Zeppelin. In 1971, large crowds coming for the rock, rather than the jazz concerts tore down fences surrounding the concert venue, bringing about the ban.

So, in 1972, it became the Newport Jazz Festival—New York and it was that until 1981. For those ten years in the Big Apple, the Festival was presented in Carnegie Hall, later Lincoln Center, but also Yankee Stadium, and even once, on the Staten Island Ferry.

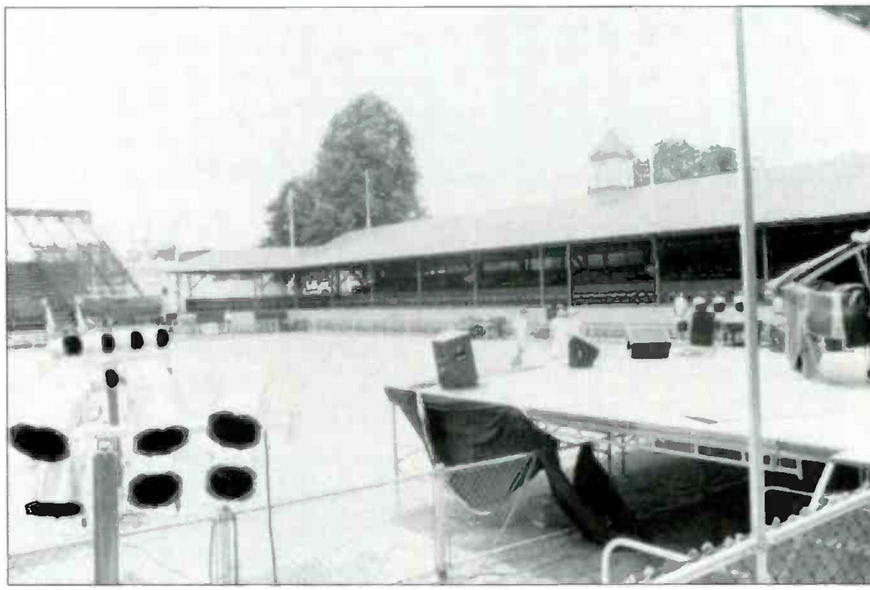

Figure 1. The tennis court set up as viewed from the stage.

The lure of Newport, however, remained. According to long-time Festival producer, George Wein, "The thought of coming back was very appealing, so we approached the State of Rhode Island and the city council and found that most people were receptive to our coming back." Starting in 1981, the Newport Jazz Festival was back in Newport, RI.

THE 1989 FESTIVAL

The Friday night concert was again held at the Newport Casino, the 19th century tennis club (now also the site of the Tennis Hall of Fame,) where the first concert was held. Audio and video equipment was being set up on center court. We had arrived in the early afternoon to meet with John Philips of Festival Productions, who with the sponsorship of JVC, were supervising the installations. We were also scheduled to meet with Gene Shively, of CMF. Gene would be making a digital tape of the event for JVC. A twenty-four track analog master also was being made—all this in conjunction with the video taping for the broadcast in November, 1989 that was on PBS.

A massive Unitel truck was parked just outside the center-court area,

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The Professional Electronic Cottage and Broadcast USA—a Synergetic Combination!
Winter NAMM and NAB show issue
GUIDE:
Speakers: performance & monitor

MAR/APR

Sound Reinforcement: theory, and application for various venues—NSCA show issue
GUIDE:
Power Amplifiers

MAY/JUNE

Broadcast, Recording & Sound Reinforcement in Houses of Worship
Summer NAMM issue
GUIDE:
Consoles & Mixers

JULY/AUG

Live Sound—producing it and,or recording it.
GUIDE:
Tape, tape recorders and accessories, Microphones

SEPT/OCT

Audio Post-Production—Television and Film
AES in L.A. Show issue
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Signal Processing Equipment, Part I

NOV/DEC

The Recording Studio—What's happening, what's ahead
GUIDE:
Signal Processing Equipment, Part II, Studio Accessories

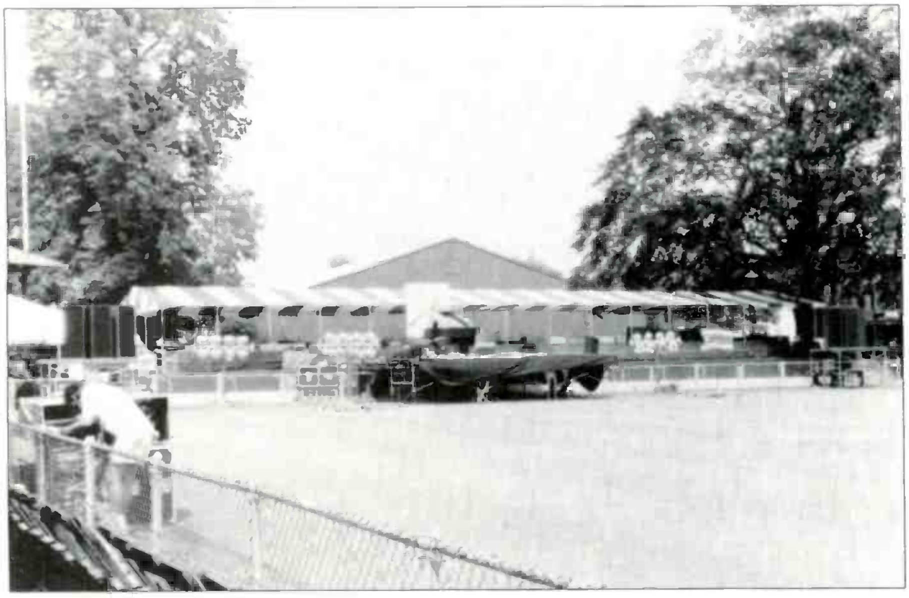


Figure 2. The mix point for the tennis court set up is seen at the left in this view toward the stage.

and cables were being strung to the stage area for the audio, video and lights. The truck also contained a fair-sized section just for audio—24 track Otari MTR-90 and big Auditronics 48/24-bus console. Unitel's Terry L. Kulchar, Senior Audio Engineer, but known as "Tweeter," explained the truck. "This is the video and audio mastering for the PBS television broadcasting, editing and audio sweetening to be done later. In addition to the Auditronics 750, we've got six Yamaha IM406 6/2 submixers. Monitors are Visionik-80 monitors with subwoofers powered by Crown D-150 amps. We have two dbx 160X, two UREI 1178s, and an Aphex 300 stereo compressor/limiter, a dbx 904 noisegate, and a Lexicon L-200 digital reverb as standard. We also have an Otari 5050 II ¼-in. and a Technics cassette deck for on the road instant demos."

LIVE SOUND

Joe Shalhoup did the mix from mid-court (see *Figure 2.*) He explained the setup for the concert. "I'm using a small console and two equipment racks to create a mono mix for the audience. I work for Capron of Boston and we are doing all the concerts. It's a separate setup, different from here, that will be used at the Saturday and Sunday concerts. But this one has the two big Meyers, fed by Crowns for the main sound. The bleachers in back, however, are being faced by five E-V units. This supplementary systems has 150 ms delay.

"The stage has six Capron-built E-V-component stage monitors along

Figure 3. The mix console position has a clear view of the stage.

with two Meyer side fills. We are using Shure, Sennheiser, and AKG mics on stage along with four more Sennheisers aimed at the sides and from the front for audience reaction. There are also Countryman boxes and C-Ducer contacts for the instruments. We're not using any wireless mics."

"I'll be giving the sixteen-channel feed to the Unitel Truck from here as well as handling the live sound.

We asked how he handled the all-to-frequent requests from artist people to take over the sound mixing.

"More often than not when that happens, and its not really that often, I'll explain how I know this equipment and can really do the best job for them. But if their soundman really seems to know the equipment and how to use it (I can determine that in a few minutes,) then I'm perfectly willing to sit back and let him do it."

The sound check by artist Mel Torme went well. We could not stay for the concerts.

EQUIPMENT LIST (STAGE SOUND)

Console:
Yamaha M1516

Microphones:
Shure, Sennheiser, AKG wired
Countryman direct boxes
C-Ducer contact mics

Figure 4. The mix console sits on one of two portable equalization, playback, and amplification racks.

EQ:

Klark-Technik 3rd-octave

Aphex Aural Exciter, Type C

Compressors and Limiters, Delays:

Brooke Siren stereo limiter

Lexicon Prime Time

Other equipment:

Furman power conditioners

Tascam MK II cassette deck

Teac CD player

db

JOHN BARILLA

THE ELECTRONIC COTTAGE

The Art Of Equalization: Part 1

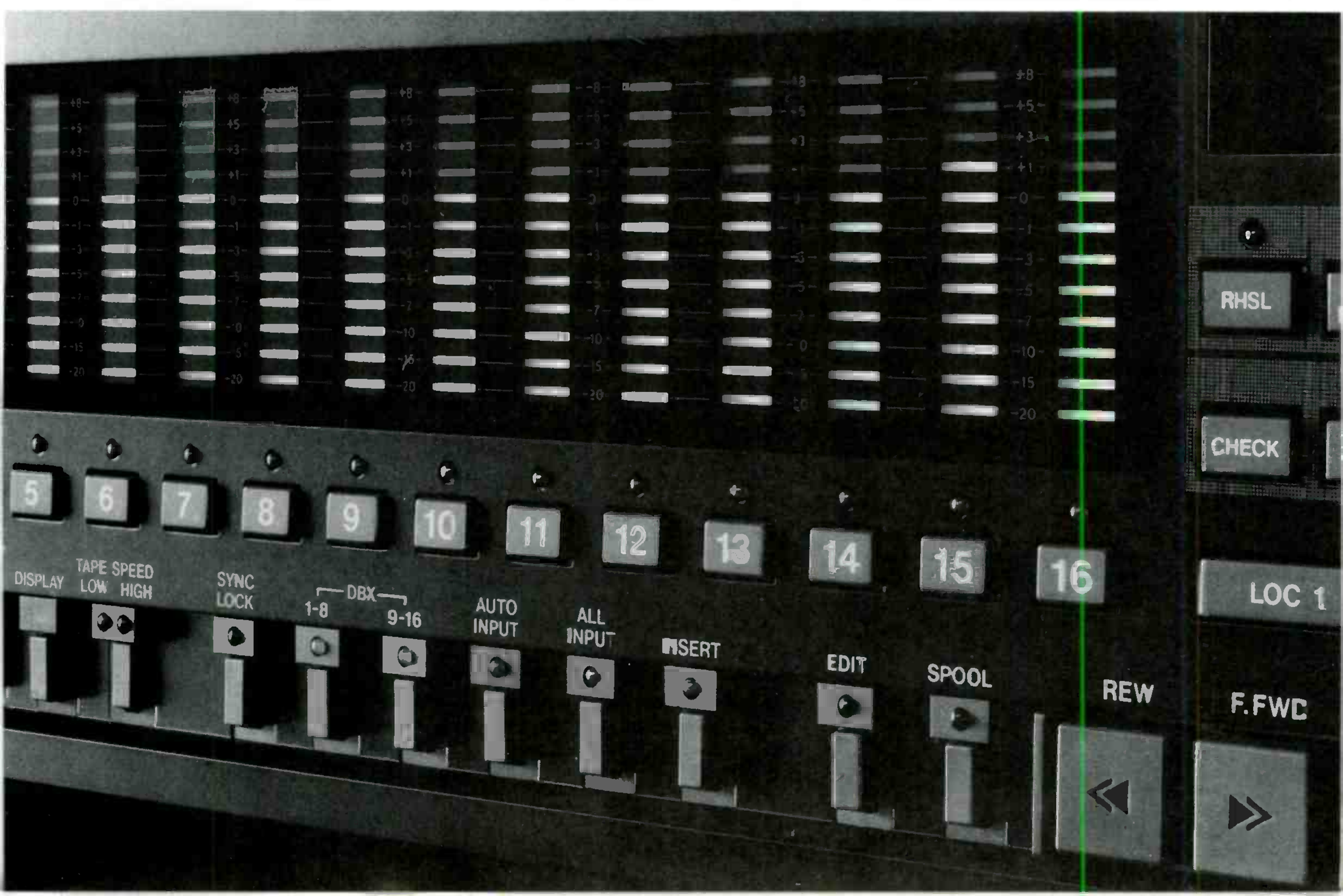
• The difference between an outstanding audio mix and a mediocre one often boils down to subtle differences in equalization. A just barely audible boost or cut at key frequencies can often turn an otherwise lifeless series of sounds into a brilliant sonic picture.

Likewise, a mix can quickly turn harsh and unlistenable if equalization has been excessively applied. How do we know *where* to start and as importantly, *when* to stop?

It has often been argued that a person either has "ears"—that is, acute sonic awareness—or doesn't; that while you can teach someone to operate a recording studio, you can't teach him aesthetics. But this elitism/defeatism attitude is patently false. While certain people are innately more talented in this area then others, every sound engineer should be able to master the art of equalization by assiduously applying the basic principles presented in this article. Herein lies the true art of equalization.

WHAT IS EQUALIZATION?

Does that sound like too elementary a question for you? Think about it for a moment. Today, equalization is often pragmatically defined: it is whatsoever you decide to do with those knobs on your mixing console which are called equalizers. This approach is pretty much underscored by our consistent use of the acronym "EQ", instead of the word "equalization." Of course, EQ is much easier to say, but it fails to remind us of what we should be doing when we



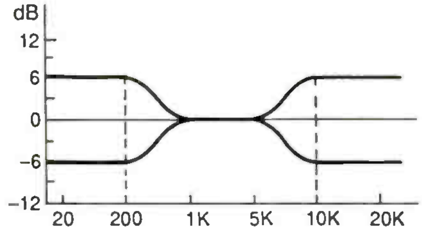


Figure 1. A representation of shelving equalization.

twiddle those knobs: equalizing, or *compensating for perceived deficiencies* in the sound—relative to the purpose of the sound in it's context.

Am I advocating a philosophy of equalization that merely strives to capture the sound and never to manipulate it? Certainly not!

While there are times when faithfulness to reality is the rule (for example, live classical recording), in most cases, equalizers can validly be used as a creative tool for sonic manipulation—a first-order signal processor of more power and utility than reverb and delay. If the context demands a snare drum with more snap than a live drum, or one that a drum sample is capable of delivering, we can respond by exaggerating a peak inherently in the drum, or perhaps, even creating one at some totally unnatural frequency.

This use of equalization—even if it results in a bizarre sound—can be valid because it compensates for the perceived deficiency of the source, relative to the context in which it will be placed.

The dangerous "thin ice" of EQ'ing however, is in arbitrariness and subjectivity. When one starts making rules like, "I always add highs, because highs make everything sound better," he is headed for an artless, perhaps even abrasive recording that may seem at first titillating, but will not hold up under repeated listenings. No, the art of equalization is nowhere to be found in subjective judgements or rigid rules. Like any art, mastery is to be found in truly understanding the tools and the medium in which you are working.

INEQUALITY AMONGST EQUALIZERS

All equalizers are not equal in the kind of performance they are suited to deliver. Most of this is a function of the design criteria: what the manufacturer has determined are the best trade-offs for its market. Practically speaking, equalizers are meant to do specific things, to the exclusion of other things. So boosting 5 kHz on one manufacturer's recording console may not have the identical sonic effect as the same degree of boost on another brand. We will address this issue again, but for now, let's leave aside the nuances of different manufacturers and review the operational similarities of the basic types of equalizers.

SHELVING EQUALIZERS

The antipodes of the audio spectrum—the low and high end—have the unique ability to add warmth or brilliance to a sound. This is achieved most subtly when a wide range of low or high frequencies are affected simultaneously, rather than specific frequencies. Shelving equalizers are designed to perform this task. *Figure 1* shows a graphic representation of this. It is quite apparent why this type of curve is called a "shelving" curve, for it appears to be lifting (or lowering) the entire low or high end as though it were on a platform or "shelf." This lift is applied uniformly to all frequencies below or above what is called the "turnover point": the place where the curve begins to flatten out. The turnover point is usually in the neighborhood of 10 k for a high shelf and about 200 Hz for a low shelf.

Shelving equalizers have many good features to commend them to the user, not the least of which is that they are "smooth" to the ear and very easy to apply or remove from a mix. They allow a sound to retain much of it's characteristic definition (which is most often a mid-range function), while gently stretching it at the ends. It is difficult to permanently screw-up a sound with a shelving equalizer. Unfortunately though, with the following two types of equalizers, this is not the case.

PEAKING EQUALIZERS

If you want to boost or cut a discrete area of the sound (rather than a global bass or treble effect), the peaking equalizer is the instrument of choice. In a sense, it is a more powerful tool than the shelving equalizer, because you can zero in on specific frequencies and boost or cut in that range. When we say *specific* frequencies; we are not saying that it

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Other "brightness enhancers" only boost existing high frequencies, pumping as much as an additional 12dB, which can distort the amp or even blow your speakers... in addition to sounding unnatural. In fact, you could probably achieve the same effect more flexibly and economically by using any equalizer.

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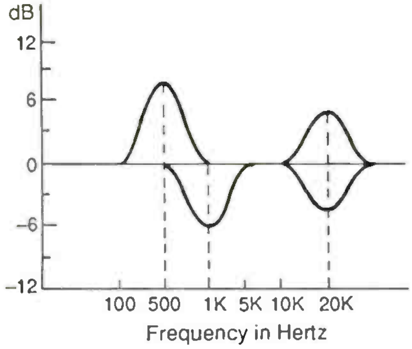

Figure 2. The familiar haystack-shaped curves associated with peaking equalizers.

affects only one frequency, but rather a comparatively narrow range of frequencies surrounding the *center frequency* we have chosen.

In *Figure 2*, we see the familiar haystack-shaped curves associated with peaking equalizers. (The width of the haystack is fixed, and becomes a characteristic of the particular manufacturer's sound.) This quality is usually described by a specification called "Q". "Q" is a measure of how far to either side of the center frequency the range of the equalizer extends, before diminishing to minimal effectiveness. "Q" is specified by a simple number, usually between 0.3 and 3.0: where the *smaller* the number, the *larger* the haystack.

Usually the "Q" (which is determined by the manufacturer) is moderate—not too wide, not too narrow. Nonetheless, it is rather specific: being extremely active at the center frequency, but virtually ineffective as little as an octave away. While the "Q" is a given factor in peaking equalizers, the user can generally choose from a continuous range of center frequencies, enabling one to boost or cut almost anywhere in the audible frequency range. This selectivity comes in real handy when it is necessary to modify highly sensitive mid-range frequencies.

PARAMETRIC EQUALIZERS

These are the big guns. And like most firearms, you should be extremely careful when you operate them or you could end up shooting some big holes in your sound. Essentially, they are like peaking equalizers with one added parameter: varia-

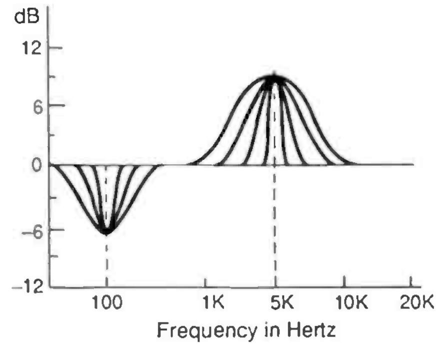

Figure 3. High "Q" permits manipulation of narrow frequency bands.

ble "Q". In a sense, it gives you a certain control over the design of the equalizer. It can be made very frequency specific (a high "Q"), which will allow you to boost or remove a *narrow* band of sound with almost surgical precision (See *Figure 3*). You can also add an audible "spike" to the frequency spectrum of any sound by boosting at the same high "Q". Additionally, you can select a rather broad band of sound using a low "Q" setting, and of course, there are lots of possibilities between the two extremes.

This is a power tool that can be of great creative value. It can help create excitement when a track has become bland. It can also be used remedially, when there is a particularly offensive frequency that needs to be attenuated without materially affecting the bulk of the sound. (A good example might be removing a 60 Hz hum.) But as wonderful as it is, a parametric can be dangerous if not used cautiously. You can end up with a very uneven, peaky, or holey sound—if you use it to excess.

There are a few other miscellaneous units which should be mentioned briefly, under the rubric of equalizers. Low pass filters do just what their name implies: they pass all frequencies *below* a certain predefined point (say 16 k), which steeply rolls-off all highs above that point. High pass filters do an analogous service to the low end: they pass all frequencies *above* a certain pre-defined point (say 45 Hz), which steeply rolls-off all lows below that point. If you stick a high pass and a low pass filter together and place their roll-off points closer together, you get a band-pass filter, which allows a particular band of audio to pass through while attenuating everything above it and below it. A device which tunes in an extremely narrow band of audio (much tighter than a parametric), and attenuates that band is called (not surprisingly) a notch-filter.

These miscellaneous tools of equalization (notch-filters) once comprised a major section in the classic electronic music studios during the 1960's and 70's, and are still found in pro-audio facilities, but aren't seen much around today's electronic cottages. Manufacturers have left them off of most synthesizers feeling that today's sounds are fascinating enough without sophisticated filtering. Likewise, they hardly ever appear on reasonably-priced recording consoles. This is lamentable, since they are valuable and powerful tools and do not add noise to the mix, since they are passive devices. Now that we have a handle on the basic tools of equalization, let's spend the rest of this study discussing a beneficial mindset for learning the art of equalization.

THE ART OF EQUALIZATION

Conceptual EQ. Let me state out front, that I am not against the "Neanderthal" method of equalization. You know, twist the knobs until it sounds good. I use that method sometimes myself, but only when I've exhausted all rational means, or I'm so tired that I can't think conceptually. But if I can think conceptually; I will, because it saves me an awful lot of time and energy. Not that I sit down with a calculator and try to figure it all out on paper. It's just that a conceptual understanding of the physics of sound, coupled with a little common sense, can put you immediately "in-the-ballpark" in determining what frequencies should be treated. From there it's usually a simple matter of fine tuning.

In a sense, it is a scientific approach I am speaking of. Here's how to approach conceptual equalization: Based on whatever your knowledge of sound is (which will undoubtedly grow over the years); you *construct a hypothesis*. For example: Let's say you want to put more "thok" into a kick drum. To do this you must accentuate a component of sound that is already present. The kick drum sound can be divided into at least two parts: the resonance from the drum shell, which propagates large bass waves, and the sound emanating from the slap of the beater against the skin. One sound is low—the other is high. The low sound you need not bother with. It's the high sound you need to manipulate. You try to identify in your mind how high the high sound might be. Is it high like a cymbal (10 k or more)? No, of course not. It's more like hands clapping or sticks beating against each other. Sticks—they are made out of wood. Wood might mean "boxiness" which is associated with low mid-range (400 to 800 Hz). But it's not like a wooden crate. It's more like two slats from that crate. A thinner piece of wood, therefore, a higher pitch. So it's probably an octave above the "boxiness of a crate" but well below the sizzle of a cymbal (maybe two octaves below).

If the cymbal is at 10 k, two octaves below would be half of 10, then half again which is 2.5 k, and an octave above 800 Hz is 1.6 k.

So the "thok" of the kick might possibly be found between 1.6 k and 2.5 k.

Even if your hypothesis is as simple-minded and vague as this one, it is at least a point of departure. You no longer need to be concerned with highs or lows. Your goal is to search the lower mid-range, starting at 1.6 k.

Now test your hypothesis. At first, you might be wrong as often as you are right. But so what? Make another hypothesis based on the new evidence and test that one. Each time you make a hypothesis and test it, learning occurs—no matter if the results are positive or negative.

With each experience the associations in your mind grow into a database, which allows you to formulate your hypotheses on the basis of more intelligent information. You will no longer be in the realm of, "hmm, let's see what this does." Instead, your inner voice will say, "hmm, let's see if this does what I think it will do." And that makes all the difference when it comes to learning the art of equalization.

Mastering the scientific approach to equalization: testing hypotheses, accumulating information from your experiments, etc., will lead, eventually, to a genuine artistic freedom where results are achieved on the basis of knowledge (rather than chance), and are consistently replicable (not hit and miss)—all of which is the essence of professionalism.

If you need proof that there is freedom in discipline, I can only offer this parable: A certain keyboard player was considered to be a creative monster. Perhaps a monster in chains would be a better description, for his mind took him on glorious musical fantasies, but he could never perform them properly. Why? Because he had never practiced basic scales and arpeggios—his fingers could not move fast enough to follow his thoughts.

One day though, he committed himself to leave aside his composing for a season, and practice scales and arpeggios daily. He did so for several months. When he went back to composing again, he found that he was looking at the keyboard as an entirely new instrument. At last, he was able to actualize his thoughts into a beautiful performance. The discipline of practice had liberated this creative monster in chains.

And so it is with anyone who would learn the craft of the recording studio—and most especially, the esoteric art of equalization. Discipline, study, and experimentation are the surest pathways to creative freedom.

In the next issue of **db**, we will continue our discourse on the art of equalization by examining the overtone series, formants, and a lot more practical techniques. Stay tuned!

RONALD BENNETT

Education in Sound Reinforcement

How many times have you been in a club, concert hall, or a live theater and been frustrated that the sound wasn't all it could be? In the days when a sound system was no more that a pair of A-7's, a powered mic mixer with rotary pots, a few RE-15's and an SM-53; live sound systems were simple to set up, and it was easy to achieve the maximum system performance, such as it was.

Today however, even small clubs and churches have complex mixers, high-powered amplification, and distributed multi-speaker systems. The sound engineer not only has to understand and command the gain structure of the system; they also need to be experts in lighting, drayage, rigging, power distribution, and have the understanding and skill to troubleshoot these complex and interactive systems. Clients now expect a sound company to come into a place like the Los Angeles Coliseum or the Mall in Washington, D.C. and provide intelligible sound reinforcement for both spoken word and music for an audience of 150,000 people: including stage lighting, public address, intercom, a live broadcast feed, and a monitor mix that satisfies everyone's demands—set up all in one day. Clearly, today's sound companies require engineers with in-depth training in a variety of disciplines: able to install, operate and troubleshoot systems with hardware that they will encounter in the field.

HOW DO YOU LEARN THE TRADE?

All to often, the only way sound engineers learn their trade is in the "School of Hard Knocks." This system has given us a few competent engineers, but all too often; the sound engineer knows only one method of operation, and has inherited someone else's bad habits in addition to their own. Some engineers have gotten the reputation of being difficult to work with or stubborn. This is all too often just a shield for their lack of technical understanding of today's complex systems. Sound companies have taken people with a bit of raw talent and an interest in audio (coupled with a little of that "on-the-road" gypsy spirit) and educated them through an apprenticeship program. In most cases, this is a slow and less than satisfactory way of training: yielding personnel that know only one type of system and hardware.

Figure 1. Full Sail stage with the instructor discussing monitor settings.

A SCHOOL FOR SOUND REINFORCEMENT

There has been a lot of coverage of the educational opportunities in audio in the trade press. There are a number of fine schools offering education in recording and broadcast audio, and one of the leading schools is Full Sail Recording in Orlando, Florida. They offer an intense program (up to 33 weeks—1,424 hours of class time plus evening workshops) in Recording Science for which they have received numerous awards. Full Sail has graduated over 10,000 students and has an industry placement record of 93-percent, which speaks extremely well for the program. Unique to the Full Sail program is their Sound Reinforcement and Concert Lighting course.

I first visited the main campus facility, a world-class facility designed by John Storyk and Full Sail founder and C.E.O., Jon Phelps. It is one of the first VARMS concept complexes to be commissioned (the VARMS concept is the interconnection of Video/Audio Recording and Mixing Suites). I won't dwell on the facility which serves as a commercial recording studio/video post-production company and as the campus for the school. Although both operations share the same building, each are operated separately, and the operation of the commercial unit keeps the Full Sail instructors in tune with the latest developments in audio and video production.

Figure 2. Michael MacDonald of Yamaha addresses the students.

THE STAFF

Full Sail has a staff of almost 100 people, the majority of which are involved in the school. Dana Roun, Director of Education/Sound Reinforcement at Full Sail Recording School was my host for the visit. He gave the background of the school and the "nickel-tour" of the main campus. Full Sail was founded in 1978, and the Sound Reinforcement Program (the focus of this visit) began in 1982. Mr. Roun, who has an extensive hands-on background in sound reinforcement (with the likes of Boston, The Pat Travers Band, and Bon Jovi) has dedicated himself full time to the Full Sail program. It was interesting to note that the students sought his advice not only for technical assistance, but for personal questions and guidance. Mr. Roun had started at Full Sail as a guest lecturer, a program which Full Sail actively pursues (the day I was there, John Meyer of Meyer Sound and Michael MacDonald of Yamaha were on-site talking with some of the sound reinforcement students).

We left the main campus and went across town to the temporary location of the sound reinforcement program, a 50,000 square-foot building which had been an open floor-plan sales office. With its high ceiling and hard roof and walls, Mr. MacDonald said, "This certainly is a real-world environment: if they can teach the students to get a good sound in there, they will be prepared to handle sound anywhere."

As with all of the course, this introduction is conducted in both the classroom and hands-on with the Full Sail concert sound system.

THE COURSE

The Sound Reinforcement and Concert Lighting course runs for four weeks and can be taken as a stand-alone course or as part of the full program. Mr. Roun stated, "we want to have an intense and concentrated program; we have lab format and classroom format. The lab format has six to eight students with one instructor. We begin by introducing a concept; then each student sets up a system and operates the system from a non-operational start, i.e., the board zero'ed out. In the third phase, the students are asked to leave and the instructor introduces flaws into the system and the student must troubleshoot the system and explain how they discovered the problems. This process is repeated for all of the learning steps and processes in the curriculum."

The Sound Reinforcement and Concert Lighting course is broken into nine segments which begin with Sound Reinforcement Technical Systems—where a study of the components that comprise a modern sound system are introduced. This includes an introduction to electronics, equalization, major system interfaces, signal processors, console signal flow, amplifiers, and loudspeakers. As with all of the course, this introduction is conducted in both the classroom and hands-on with the Full Sail concert sound system.

Concert Lighting Technical Systems covers the interfacing of lighting fixtures, dimmers, and lighting consoles. This section of the course features a study of lighting basics for stage, theater, and instruments, dimmers' spots, rigging, trusses and lifts, lighting design and production: up to and including actually "calling" the show.

Figure 3. The instuctor is "sabotaging" the mix console. Students must then discover the problem he introduced.

The Engineering segment covers the correct operation of the equipment for house engineering, monitor engineering and simulcast engineering. This is done through actual hands-on engineering experience and includes real-time troubleshooting.

TOURING SOUND EDUCATION

Time is spent with the students discussing the real consideration of the "touring life." This is a very human look at the people and jobs that make a national tour a success. The purpose of the touring crew is explained along with show production procedures and the ins and outs of "staying on top of it all." Tour structure and income potential are studied in detail with instructors who have been there.

Clearly, a graduate of this program is ready to handle any system...

Each student designs a large scale system, both lighting and sound, and submits this as a mock bid. An overview of the many different sound and lighting systems in use in today's touring companies is given. In addition to all of the technical skills discussed and taught, time is spent learning how to get a job in the industry. What professions are available and how to be psychologically prepared to get those jobs is presented by industry personnel. In addition, students learn the ins and outs of Remote Recording. They learn to interface the school's state-of-the-art remote unit with a live sound system and what the specific duties are for each member of the team.

FULL SAILS' EQUIPMENT

For the first years of the course, Full Sail rented systems from local sound companies and rented a venue for the lab portion of the course. The success of the program demanded a full time system and Mr. Roun assembled a cutting-edge system based around Yamaha consoles and Meyer loudspeakers.

The system components represent actual equipment that the students will encounter in the field. The system consists of a Yamaha PM3000 console for the house, a Yamaha PM2800 monitor mixing console, Meyer loudspeakers and amplifiers and Yamaha SPX1000 digital effects units, plus the normal complement of microphones. Full Sail also has a remote recording unit and a full concert lighting system, all of which are used in this study course.

Clearly, a graduate of this program is ready to handle any system, fixed or remote, and operate it to its fullest potential. I heard a student mixing during my visit, and especially considering the building, it sounded clean and open. A tribute to the system, but even more, a tribute to the program. db

Wonder Plus (1DIR)

Wonder Plus is a DOS shell. What's a DOS shell? It's a PC program that permits access to, and manipulation of, all the files and programs on a hard disk in a computer.

• When you have a 20 meg or more hard disk filled with assorted programs and perhaps thousands of files, you can use all the help you can get to access them with ease.

Working with PC or MSDOS directly through its command structure is no picnic with heavily-used hard disks. DOS shells help this access immensely and make easy access to all files and programs possible.

Here at the **db Magazine** office, where this sort of computer need abounds, we have been using Bourbaki's DOS shell for some time now.

Recently a completely revised and updated version of the 1DIR shell now called Wonder Plus has been issued. We've been using it for a while and it really permits access to programs and files far faster and easier than any DOS command could.

Wonder Plus uses a user-created menu system that permits you to structure the program to your specific needs. It can easily operate from the keyboard or with a mouse. Once trained to know and access the programs on your disk, you only need to hit a function key or use your mouse to go to the program name, hit the Enter key, and your program comes up on the screen. When you quit the program, Wonder Plus reappears on your screen (in color if you have that capability) ready for its next assignment.

While it is on the screen, you can scroll up or down to see files or subdirectories. You can switch to any other drive to see what is there and act upon those files and programs. You can display hidden files, and act on the attributes of those files to make them unhidden, even erased, if you wish.

DOS shells use memory, of course, while they are active. But Wonder Plus can be set to use conventional memory or EMS memory. More important, it can be set to remove itself completely when you invoke a program, only to reset itself when you quit the program. So, while it is using memory to be on the screen, when we activated Microsoft Word to type this manuscript, Wonder Plus was removed from memory leaving all that was available before for Word's (or any other program's) use.

One of the best reasons for using Wonder Plus is the ease in which you can move from one program to another. It literally only takes a few mouse clicks.

It's a full-featured DOS shell in that it also permits creation or erasure of subdirectories and or files, formatting of floppies, copying of disks and files from any drive or directory to any drive or directory; in short, anything that DOS permits is done with ease and a few mouse clicks or function-key combinations.

Do you have files or directories that are restricted to others using the same system? Activate the Security Module and password-protect them.

We can't stress strongly enough the fact that the menu structure can be completely custom configured to suit any requirement. The manual and reference card that come with the program are excellent, so even if you are not excessively DOS wise, you will have little trouble creating a personalized menu structure that works best for your needs.

It's a must-have program for PC users, and at $95 can almost be called a bargain. db

For more information circle 67 on the reader service card.

ED LEARNED

Outdoor Sound Reinforcement for Symphony Orchestra

Classical music performances comprise at least half of the total live music performances in the United States every year. The artists range from soloists to small ensembles to full symphony orchestras. Quite often, these performances are given in concert halls designed and built for the performance of acoustic music. The best possible marriage between the architect's manipulation of acoustics and the artist's mastery of balance results in performances pleasing to the ear, rendered without amplification of any kind.

The past 25 years has seen an explosion of innovation and technology in the sound reinforcement field, embraced fully by popular music forms. The dependence of today's pop musicians on sound reinforcement systems, coupled with the dramatic increase in the size of concert venues, makes sound reinforcement an integral part of any popular musical performance. Classical music, however, remained predominantly acoustic, although various pop flirtations with classical music forms (e.g. Emerson, Lake and Palmer's ill-fated orchestra tour of the 70's) foreshadowed a possible marriage between classical music and state-of-the-art sound reinforcement.

NEW THINKING FOR OUTDOORS

The growing appetite of the American public for outdoor classical performances necessitated new thinking regarding the role of sound reinforcement systems in classical music. For the first time, orchestras were faced with an environment hostile to natural enrichment and projection of their sound. These performances often took place on a temporary stage in a park or other open area, without the luxury of even a bandshell for projection. Crowd size expanded logarithmically at these shows; for the far reaches of the audience to even partially hear the performance, sound systems became necessary.

Due to familiarity, mic technique for these events mirrored classical recording technique: minimal mic'ing, featuring the use of omni-pattern mics, coincident pairs, and distant area pickup were the rules of the day. As venue and crowd size continued to grow, the need for better control of the sound soon dictated a new approach. RCI Sound Systems, the major mid-Atlantic symphony contractor is one of several U.S.-based companies making important innovations in this *brave new world* of symphonic sound reinforcement. RCI currently services both the Baltimore and National Symphony Orchestras.

THE INTERVIEW

I spoke with RCI's most experienced symphony engineer, Craig Jensen, about his company's approach to the tricky problems posed by these large outdoor events.

db: RCI's Baltimore Symphony Orchestra concerts over the past few years have been held at Oregon Ridge. Give us a "lay-of-the-land."

CJ: Oregon Ridge is located in Hunt Valley, Maryland. It's a defunct ski slope, a fairly small ski slope, with the stage at the bottom of the hill. So it forms a fairly large natural amphitheater that could seat 50,000+.

db: The stage is placed so the orchestra "plays" to the hill?

CJ: That's right. From the stage, looking at the hill, the incline of the first 100 yards or so is rather moderate. Then it gets quite steep as the actual ski slope kicks in. The slope is bracketed on both sides by thick trees that help contain the sound somewhat, but can also cause a slight echo problem.

db: Has the echo ever posed a problem for you or the orchestra?

CJ: Not really. It's only slightly noticeable from the stage, maybe a little more so at the mix point, which is about 100-125 feet out from the stage. When it's really obvious is during loud passages: if you have a piece that ends on a loud note, you can hear the slapback. That usually gets a few chuckles out of the audience.

db: How big is the stage area, and what does it look like?

CJ: The stage is an 80-foot by 40-foot wood slab, built on a small natural mound that elevates it above the near audience area. The stage is surrounded by wooden walls, located at the sides and rear of the slab, so that the downstage side is the only open one. There are wooden baffles overhead, suspended about 20-feet up from the floor (*see Figure 1*). These are hung from the roof, which is steel frame structure covered with a heavy duty blue-and-white tarp. That gives us a dependable waterproof covering for the stage.

db: Was the purpose of the walls and overhead baffles to provide a shell for natural projection of the sound into the audience?

CJ: In a small sense that might have been part of the reasoning, but the main thing was really monitoring. When the orchestra is in their concert hall environment, the far corners of each section can hear each other. The basses can hear the first violins, and vice versa, so they can play together. That's the most important thing for an orchestra stage,

Figure 1. A detail of the Oregon Ridge Orchestra Shell.

and it's one of the reasons why concert halls are as live as they are.

db: And outside, which is essentially anechoic, you don't have any liveness.

CJ: Right. That creates the need for a shell like the one at Oregon Ridge. We also do the National Symphony Orchestra several times a year on the West Lawn of the Capitol in Washington, D.C. That stage is essentially open, it's covered only by a large tent. You can walk up there when the orchestra is playing and notice the difference right away. It's a far less desirable situation for the musicians, but since it's only done three times a year; they basically wing it.

Figure 2. One of the typical speaker stacks used at Oregon Ridge.

db: You mentioned that you've done jazz and pop shows at the Oregon Ridge site using the shell. How did that work out?

CJ: It was too live; very reminiscent of the problems you have when you do pop shows inside a concert hall. You can't get the good tight mix that you'd like. The jazz show was great—it was low—keyed enough to work. But don't do a rock band there! *(both laugh)*

db: What do you use for a symphony speaker system?

CJ: On each side, I use 4 RCI 490 cabinets, sometimes called "super boxes," and 2 RCI 240 cabinets, sometimes called "long throws." We erect a 50-foot scaffold tower, a 6-foot by 4-foot section, I believe, on each side of the stage and fly the cabinets inside of these. The four super boxes are hung together in a column, with two long throws up on top *(see Figure 2)*.

db: Where were these towers placed with respect to the stage; were you in front of, or even with it?

CJ: The stage is built on uneven land, so we weren't as flexible as we'd have liked on the scaffold placement. We were able to buy a bit of room downstage, so the speakers were a good five-to-six feet in front of the downstage lip. No mics were even close to the downstage edge, so we had a good start on gain-before-feedback *(see Figure 3)*.

OREGON RIDGE AUDIO THROWS

db: You've got a very long area to cover at Oregon Ridge. How do you insure adequate level at the rear without toasting the people in front?

CJ: One advantage I have is the slope. You can walk back from the stage as far as 200 yards and still be in the audience area, yet the line-of-sight distance is less. With the system flown, you still have speakers in your face up on the hill. The long throws do an excellent job of covering the upper slope area; in fact, I usually run these on a separate matrix output so I can push them more than the lower boxes. This fills out the farthest audience area, yet the cabinets are up high enough that I don't overpower the nearfield.

db: That's a great idea!

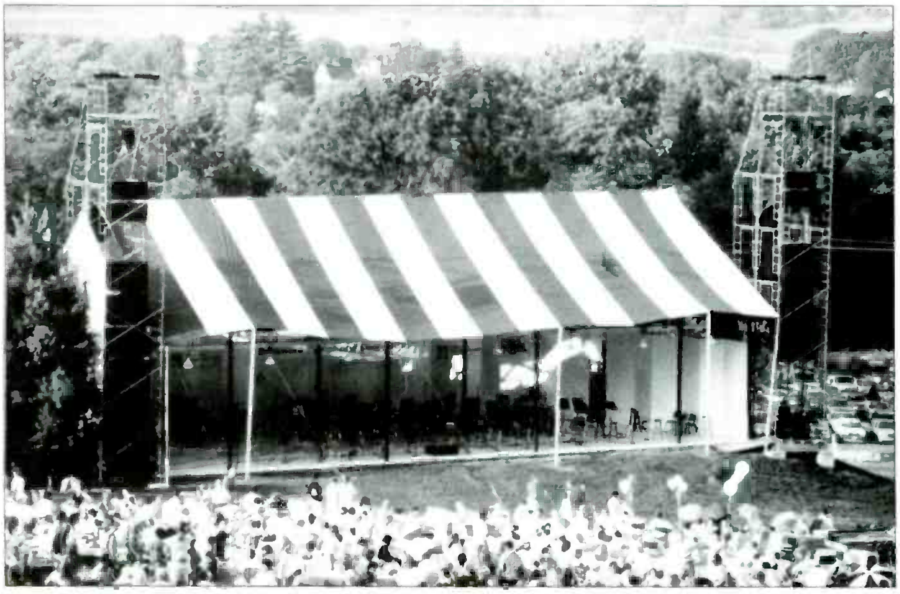

Figure 3. Overall view of the Shell and stacks, Oregon Ridge.

CJ: I'll often use a three-tiered system. I run the bottom pair and top pair of super boxes on separated matrix outputs as well.

As the speakers go higher, I can run them hotter, all of which helps my distance coverage. I should also mention that the first twenty feet of immediate downstage lawn area is roped off, so no one can get any closer than that to the stage or either stack.

db: Do you run the system in stereo?

CJ: I never run stereo; instead of left-right on my graphics, I run up-down. The lower set of cabinets require a bit more EQ than the upper ones due to stage proximity. The upper cabinets are forty feet in the air; I really don't have to worry much about feedback problems there. Plus, the audience is really spread out, so stereo would be detrimental to most people. Better control up-down means I can better cover the rear areas, more beneficial to the guy two-hundred yards away who just paid $40.00 and brought the whole family out for a concert/picnic.

db: Perhaps delay stacks could be used to fill in rear areas. How do you feel about them?

CJ: We use them, and in most cases I would favor them. However, it becomes a bottom-line issue: with more labor, time, and gear involved it becomes a trade-off between what the orchestra *really* needs to cover the audience and cost. At the West Lawn location for the NSO we use them, and they're very effective. The site is fairly flat, and the audience for those concerts can be upwards of 150,000, so sheer size necessitates delays.

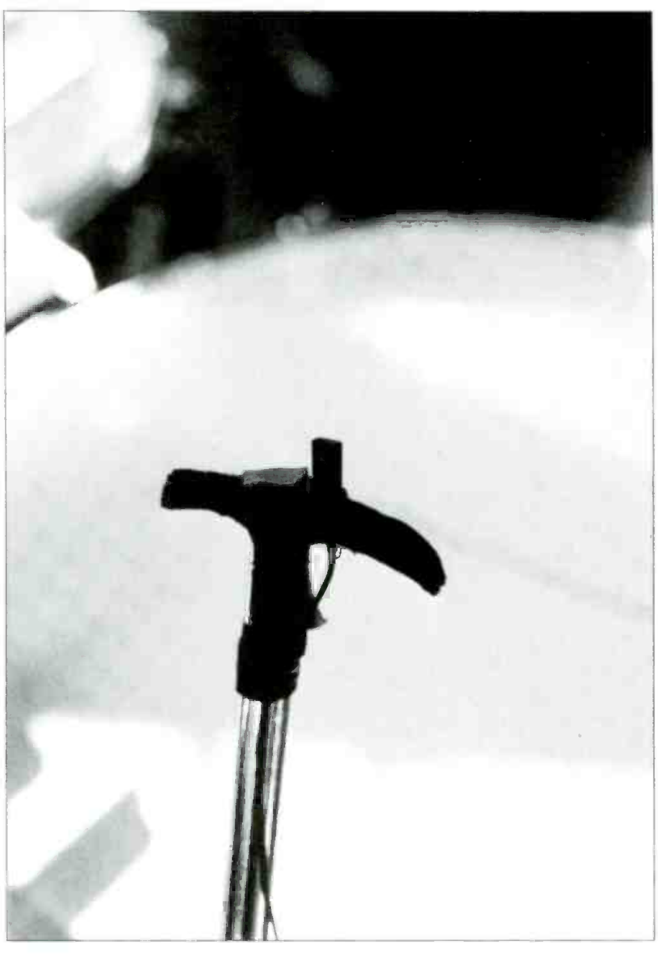

Figure 4. An omni Iso-Max is shown positioned over typani.

TOTAL COVERAGE

db: What do you figure your total area of coverage at Oregon Ridge is?

CJ: It's probably in the neighborhood of 150,000 square yards, which is a huge area. And we can have huge crowds. Our coverage is helped somewhat by the containment provided by the trees on either side and the hill itself. Delay stacks were deemed superfluous due to cost and these physical advantages.

db: You're dealing with acoustic music, dependent for years on the acoustic environment of concert halls for projection towards the audience. Now you find yourself outside, where natural acoustics can no longer handle the coverage requirements, let alone balance and tonal considerations. What do you feel are the most important elements in reinforcing this music?

CJ: Outdoors, with 10,000+ people, the subtleties of what the orchestra is trying to do will not come across like it will in the concert hall. You aren't going to hear the second oboe playing a harmony with the first oboe, in a quiet passage, from the back of the audience area. It's necessary to exaggerate these types of things level-wise so the audience can hear them.

Figure 5. Microphones over the string section.

db: Traditional mic technique for classical reinforcement dictates the use of omni-directional patterns or coincident pairs for area pickup. What's your view on this?

CJ: To get the necessary gain to cover subtleties two-hundred yards away—it's just not going to happen with distant mic'ing. You need better control, and all the gain you can get. We went to using as many as forty mics on a symphony orchestra. We found that by following scores and getting cues from people who knew the particular pieces well; we could bring up any section of the orchestra very easily and obviously if need be.

db: The idea of placing one's music in the hands of a third party was an idea that terrified most musicians at first, I suppose. Nowadays, many pop groups routinely have an engineer recreate their sound for them—it's no big deal. But it is very new ground for classical music: your multi-mic technique places more responsibility for orchestral balance on the engineer than ever before. Do you work with the conductor in balancing the orchestra to their desired ends?

GETTING STRINGS LOUD ENOUGH

CJ: Before I talk about conductor-engineer trust, let me mention our most basic problem outdoors: getting the strings loud enough. The strings are probably the one instrument most people relate to a symphony orchestra. They are the bulk of the orchestra; they play most of the moving melody lines, yet are the quietest instruments. Just about everything else projects better outdoors. Some people in pop music start with the drums: they feel if they get the drums sounding good, they have a basis for a good mix going. In orchestra mixing, it's getting those strings loud enough; trying to get the SPL needed without letting them get too shrilly, working to maintain a nice warm sound.

db: Some purists might argue that a simple coincident pair over the strings could do a better job of preserving inter-section tonality and balance.

CJ: I still don't think you can beat the gain we get from a mic on every two to four players. On cellos, we might go to one mic on very two players. The concertmaster might get his (or her) own mic for solo pieces. That's another distinct advantage to multi-mic'ing: we can spotlight the concertmaster for a solo passage, something area mic'ing wouldn't allow.

db: Creating voicing through instrument level changes is a given for many pop engineers. Is this something you feel is important for classical multi-mic mixing technique?

CJ: I feel the symphony orchestra does a better job than any other idiom of music in controlling dynamics—playing in sections with each other. That's part of the conductor's job, to control those sectional balances. The orchestra is still mixing itself: I'm not moving faders a lot during performances. And just because I have all these mics doesn't mean I have to use all of them. I'm there to improve projection, not balance. Sometimes symphony personnel might come up and tell me they feel the strings aren't quite loud enough during a certain passage. I can highlight a certain section more easily than I could with coincident pairs scattered about. And multi-mic'ing is far superior for pieces that have lots of solos.

db: Can you give us an example of a heavy solo piece?

CJ: Ravel's *Bolero* is a perfect example: a very popular piece, which was played twice last year. It's a very building piece, with cascading solos during the entire piece. For that, I had a cue sheet of every solo coming up, so I could give it just that little bit extra and get it heard. It goes back to the idea of exaggerating for an outdoor show.

db: Did the conductor ask you to do that?

CJ: I have no problem taking direction. But it's funny: for the most part, they've left me alone. I'm letting the symphony mix itself; I just exaggerate the parts so people can hear. I think the conductors, trained as they are, can hear the amplification I'm doing and approve of the job. If I hear anything, it's usually from the symphony administrative people, like the artistic director, who know symphony sound very well—they gave me the cue sheet. They're happy because response to the outdoor shows has been overwhelmingly positive.

db: Do you have a background in classical music?

CJ: I was a music major in college, and played trombone in various high school and college symphony orchestras. I've studied classical music a lot, and enjoy listening to it in my spare time.

MULIPLE MICS

db: Aside from the artistic considerations, there are technical considerations to using multiple microphones. The potential for feedback increases as the number of open mics increase, as does the problem of unwanted instrumental bleed. How do you counter those problems?

CJ: I could have a single coincident pair over the strings, and during an ensemble fortissimo passage, the brass is going to be louder in mics back in the brass, which I thought would easily be enough, and I really missed out on some of the trumpet harmonies. I should have had two mics just *for* the trumpets! It's another point about area mic'ing: I got a decent blend, but I just couldn't get the SPL I needed to pull those trumpet harmonies out. If it's a Fourth of July show, you've got 50,000 people, and *Bugler's Holiday* is on the program—well, people two-hundred yards away have to hear that part.

db: How loud are your clients asking you to run the system: is the PA basically coasting, or are you getting into real level?

CJ: I'm not getting into real level, and that goes back to the feedback question you mentioned. A lot of people think that forty mics puts you into a feedback nightmare as opposed to a smaller number of coincident pairs or whatever. I don't push the system that much; the PA is there primarily to pinpoint sections better. I don't run any particular mics that loud; the ones I push the most are the string mics. String level is always my major concern. I have another thing to consider with respect to string sound: no string player ever brings their best instrument outside in 95-degree weather. Instead of the $10,000 violin, they bring out the $1,000 violin that they used in high school or college.

db: Considering summer weather in the Washington area, especially the humidity, I can't blame them.

CJ: The top instruments don't cost more because they're older, necessarily, but because they sound better. The cheaper ones just aren't as nice; they're not as rich, and can't give a tone like the good instrument. They can start to sound screechy when I get them loud enough; often I find I have to cut the 1.6-2 kHz area quite a bit to compensate.

SPL OUTPUT

db: How many dB-SPL are you pushing at the mix point?

CJ: The dynamics of some of these pieces are as wide as you could imagine: there could be a flute solo for ten seconds followed by total ensemble playing, fortissimo. At the loudest, we're pushing 100 dB-SPL at the mix. On the other hand, you've got soft string phrases that you can't get loud enough. To the people sitting two-hundred yards back, it's virtual silence unless you have an extremely quiet audience. For the most part, you've got an ambient noise level of 55 dB-SPL from wine bottles opening, children playing and crying, and adults chatting. It's a festival atmosphere, not a serious classical crowd. Fortunately, the programs are pops-oriented, so I don't have a lot of the pieces that are all over the place dynamically. We probably average between 85 and 90 dB-SPL.

Figure 6. A close up view of the 'cello mic'ing.

db: Your normal console for these jobs is a Yamaha PM-3000 40-channel desk. Do you use any outboard effects, such as digital reverb?

REVERB

CJ: I *definitely* reverb the orchestra, especially the strings. Reverb is part of the sound of the orchestra in a concert hall, and people expect to hear it. There isn't any reverb outdoors, of course, so we recreate it digitally. It makes all the difference in the world—I couldn't live without it.

db: What sort of reverb units do you use, and how do you program them?

CJ: I've been using the small hall program on a REV-7, which might surprise a lot of people. It seems to work better than the large hall program, which can sound a little unrealistic outdoors. I use this on eighty-percent of the string mics, which gives me a nice overall reverb thanks to bleed. I'll use a second reverb, like an SPX-90, to spot reverb those color solo passages here and there. I know these pieces; if I don't know them, I can get and read a score. I wait until that spot comes up, and then add just a touch of reverb. It gives that nice, concert hall sound.

db: Do you return the reverb flat, or do you EQ the return with a certain acoustic in mind?

CJ: The reverb can create a feedback problem in the low mids and lows. I'll dial some of that out: an 80-100 Hz high pass filter on the low end, maybe take out a bit of 315 Hz and go from there. I don't take out too much of this stuff, because then it can start to sound thin. If I bring the reverb back on an input channel, I'll EQ it there. If I have to use an effects return I'll use the reverb's on-board EQ, if it has one, with the EQ on the return to achieve the desired effect.

db: I've used RCI's system for pop shows, and as I recall you have incorporated system limiting in your

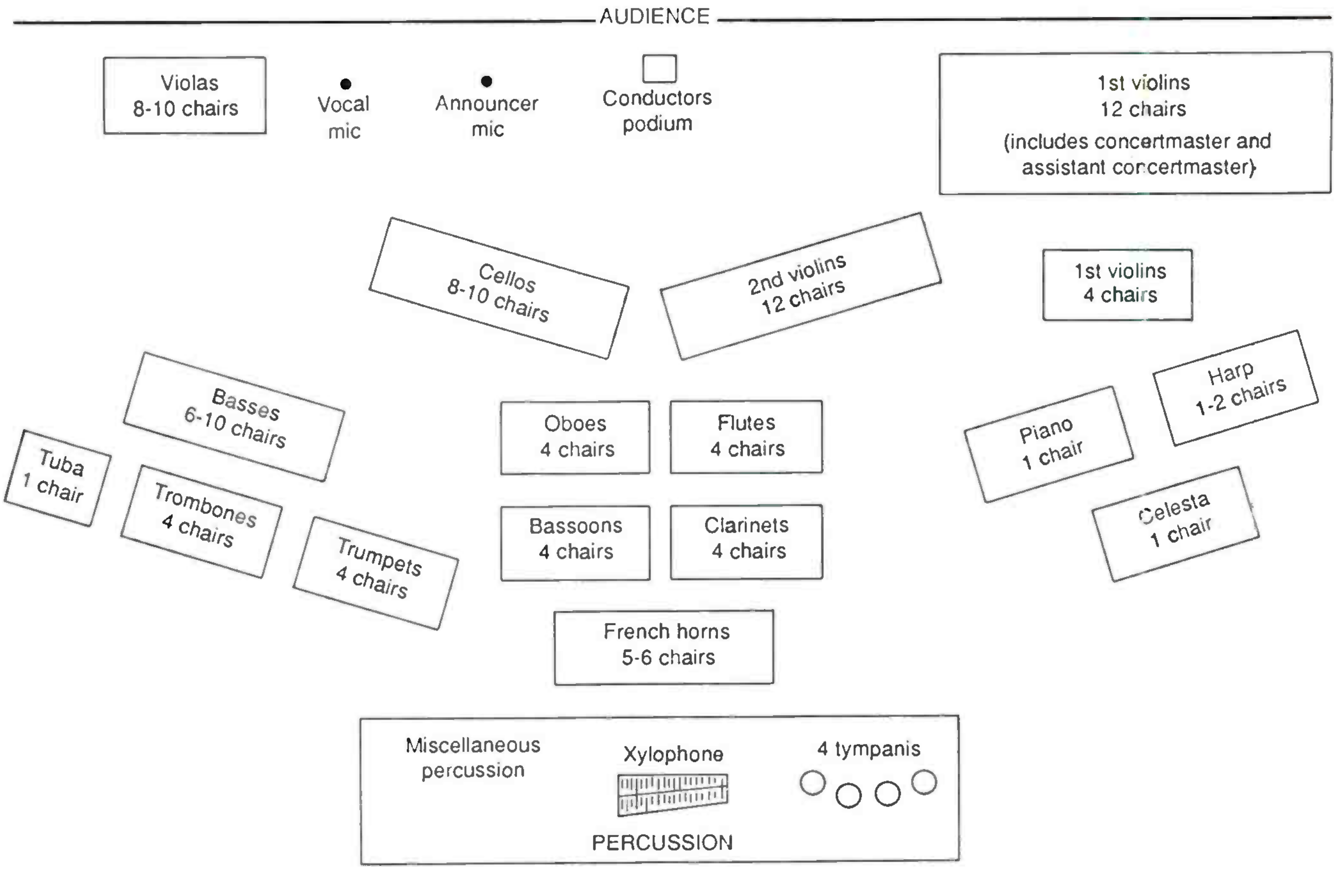

Diagram 1. The Baltimore Symphony Orchestra stage layout.

Strings:	
5 First Violins:	1 AKG 460, 4 Sennheiser ME-40's
3 Second Violins:	3 Sennheiser ME-40's
3 Violas:	3 AKG 451's
3 Cellos:	3 AKG 451's
3 Double Basses:	3 Sennheiser 421's
Woodwinds:	
2 Oboes:	2 AKG 451's
2 Flutes:	2 AKG 460's
2 Bassoons:	2 AKG 451's
2 Clarinets:	2 Electro-Voice ND-408's
Brass:	
2 French Horns:	2 Crown PCC-160's *or* 2 Sennheiser 409's
1 Trumpet:	Electro-Voice ND-408's Y-d
1 Trombone:	2 Sennheiser 421's Y-d
1 Tuba:	1 Electro-Voice RE-20
Percussion:	
2 Timpani:	2 Countryman Isomax
1 Xylophone:	1 Countryman Isomax
1 Misc. Perc.:	1 Countryman Isomax
Miscellaneous:	
2 Grand Pianos:	2 AKG 460's *or* 1 AKG 460, 1 C-Tape
1 Harp:	1 Countryman Isomax
1 Celesta:	1 AKG 451
1 Vocal:	1 Shure SM-87's
1 Announce:	1 Shure SM-57's

Diagram 2. Instruments and respective microphones used for the Baltimore Symphony Orchestra performances.

those mics than the strings—and we're talking about a brass section that is upstage center or left quite a distance away. There's really nothing you can do about that. Having several mics closer to the strings definitely gives me better control than I might get with distant mics.

db: That situation requires you to be sensitive enough to use that bleed to your advantage, and to know when to turn mics off.

CJ: Well, with forty mics on the orchestra, you can bet that when the brass is blaring away in the big part of the piece I'm not using much of the brass mics! Sometimes I wish they'd play softer, but they're used to a concert hall and no mics: they know it's supposed to be loud.

db: Have you ever considered cutting back on the brass mics?

CJ: There have been times, where I only had a 32-channel console instead of the 40 I'm used to, that I didn't fully mic the brass to save channels. I figured I'd have no problem picking them up, and I ended up kicking myself. There was a piece we were doing, *Bugler's Holiday* I think it was, where there is this important ensemble trumpet section. I put two

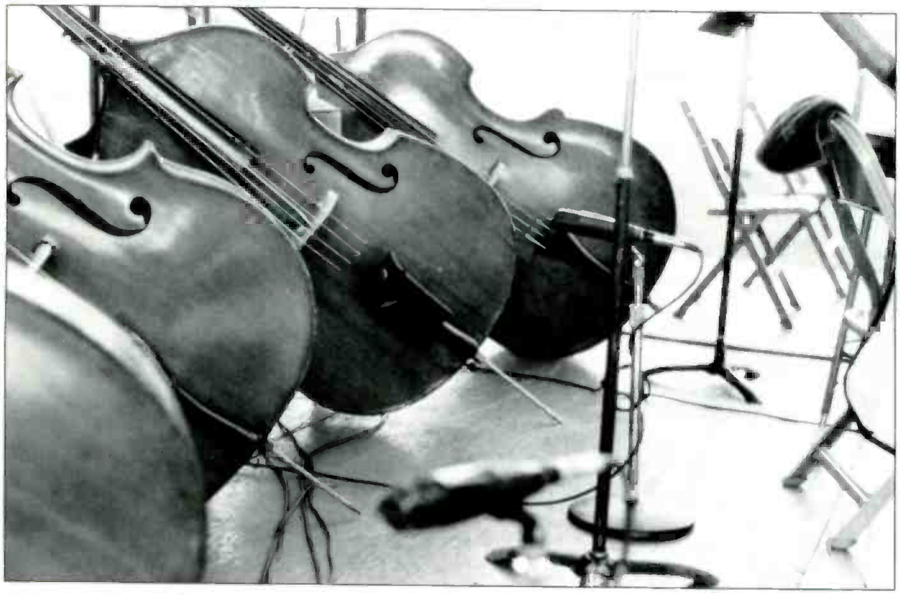

Figure 7. Double bass instrument mic'ing.

drive rack. Is that something you even need for a job like this?

CJ: I suppose some God-awful thing could happen that might send a dangerous spike through the system, like a mic falling off a stand. But at the level I run things, damaging the system is a very minute possibility. I take the purist attitude and patch around the limiters. I really don't need another device in the audio chain; I keep it as simple as possible.

MULTI-MIC TECHNIQUE FOR SYMPHONY ORCHESTRA

I asked Craig to give us a real basic mic'ing for orchestra, as the specific demands of particular pieces will dictate changes in the actual numbers and locations of microphones. The corresponding photographs are a combination of both the Oregon Ridge and West Lawn sites (*see also diagrams of the BSO's stage layout and channel assignment and mic list*).

db: What design of mic do you prefer?

CJ: I use as many condensers as I can get, especially on strings. They have the clean, full sound I'm looking for, and they also give me the best available gain at the mic. While response does vary between brands, I find they're flatter than dynamics for the most part.

db: Since you're outside, you must use a lot of windscreens.

CJ: I've got them all over the place, and definitely on every condenser. Most of these tend to be downstage on the strings, where any wind will get picked up. The ones on the woodwinds are more upstage, and are protected by the shell somewhat, but if I have enough windscreens I use them back there too.

WIND NOISE

db: What sort of console EQ or filtering do you use to minimize wind noise?

CJ: Any of the sub-harmonic stuff is not going to help in an outdoor situation. I prefer to keep the mics flat: I don't use the roll-off switches on mics like the Sennheiser 421, ME-40, or the AKG 451 and 460. If I've got a board that has a versatile, flexible EQ with adjustable high-pass filter, I'd much rather do it at the board. On strings, I'd use a 40 Hz cut. I'd use that on most mics in the orchestra except double basses and tubas. To be honest, I could probably do it there too, and no one would really notice. Fortunately, I don't often have a bad wind problem. The shell and top offer some protection, but concert timing helps even more. In the summertime, at least on the East coast, the winds tend to die down at sunset, which is around the time we start.

USING OMNIS

db: I've been involved with outdoor symphony performances in my home town of Detroit. Because I used a lot fewer mics than you, we employed a mixture of cardioid and omni patterns. You have a real forty mics up—do you use omnis at all?

CJ: I am using a few omnis, but in one case, the pattern type is really secondary. I'm using a Countryman Isomax inside of a harp. It gives a gorgeous flat sound—I don't really have to do much of anything with channel EQ. That's really why I'm using that mic there, not because it's an omni. I also tape them to the tip of a straight mic stand or boom and lean it into a percussion table, xylophone, or timpani (*see Figure 4*). That also gives a very nice sound, and because there is so much acoustic SPL hitting these mics, I don't have to turn them up as much and risk picking up any nearby bleed.

db: Where are these instruments located?

CJ: They are far, far upstage, where I get great gain-before-feedback. The harp mic is inside, where I get great isolation, and the timpani are so far upstage there really isn't anything nearby to bleed into them. It's always the downstage mics that are more susceptible to bleed. I actually do benefit from the omni pattern on the timpani, because I can use one for every two drums and get real even pickup between them.

db: What do you do if you have more than one harp?

CJ: Harps have a very distinct sound. Sometimes an oboe can sound very similar to a clarinet, a viola similar to a violin, but there is nothing at all that can sound like a harp in an orchestra. It's what I would call an accent instrument. When it's played, it needs to be heard: it needs to stick out a little bit. I could area mic two harps, but I'd never get it loud enough to do that. With the mic inside, you get *plenty* of harp—certainly you'd never even get near a feedback problem. So I'd use one Isomax inside each harp.

MICS ON THE FIRST VIOLINS

db: The first violin section is the orchestra's largest. You mentioned a concertmaster special earlier; that is one of the first violins. How exactly do you handle this section?

CJ: The concertmaster and assistant concertmaster are the two premier violinists in the orchestra. I use one mic, split between them, for that reason, and also because if there are spot violin solos during a piece, they will play them. I use an AKG 460 cardioid condenser, and try to get it in as close as I can, with some compromise for bow clearance. I place the stand on their upstage side, with the mic angled away from the upstage side and towards the audience. That helps keep most of the bleed, which is coming from upstage, off-axis.

db: What about the rest of the first violin section?

CJ: I use Sennheiser ME-40's, which are cardioid condensers. They don't quite have the same quality of high-end that the 460's do, and I like that, because I have a problem with too much high-end on the violins. These are two-to-three feet away, placed and angled the same as the concertmaster's special. I try and do that with every string mic I can. Every group of four players gets one mic (*see Figure 5*).

THE SECOND VIOLINS

db: Now, we have the second violins. Do you use the same type of mic here as well?

CJ: I do use the ME-40 here: because it's the same instrument, I like to use the same type of mic. The parts are not the same, however. Second violinists aren't seconds because they aren't as good players; seconds play a completely different harmony than firsts, and it's equally important that they be heard. The mics are placed and slanted similar to the other violin mics, and every group of four gets one mic. I do cheat on the first group of four, as the first two seconds are the best players; I favor them slightly.

OTHER STRINGS

db: There are a smaller number of violas; what do you do there?

CJ: Usually, there are ten chairs, so I go 2-4-4 with the three viola mics. The group of two includes the principal, so I have solos covered. I use AKG 451 cardioid condensers here: since the viola is a lower registered instrument than the violin, I find I don't have as many high-end problems with them.

db: What about the cellos and double basses?

CJ: On the cellos, I use 451's. I originally started with Sennheiser 421's, which are dynamic cardioids, but I really fell in love with the cleaner, livelier sound I got from the condensers (*see Figure 6*). For the double basses, I do use 421's (*see Figure 7*). I'll always use the first mic of each section on the first two players, favoring the principal. The other mics basically split pairs of players, but that can depend on how many we have that day. I used to clamp the mics on their music stands, but I've evolved to floor stands with short booms. I try to get them as close as I can without getting in the way, around two-to-three feet from the instruments.

Figure 8. Flute mics are shown.

THE WOODWINDS

db: We now come to the woodwind section, which has a diversity of instruments, including sectional diversity. Let's start with the flutes.

CJ: The standard setup is three flutes and one piccolo. The others double on piccolo when the need arises—in *Stars and Stripes Forever*, they all play piccolo. Piccolo is an instrument that doesn't require a mic too close in order to pick up fine! The two mics split the four players; throughout the woodwind section, I favor the first chair, who is the principal and the best player on that instrument. I find I can get the ensemble sound when I need it—the unison parts are not compromised. Yet, any solo spots will be played by the principal, so I still need to be able to highlight that. Occasionally, I might have a bass flute. If I can't give that a separate mic, I will at least cheat a mic towards it. I mic these from overhead, like the violins, using a pair of 460's or 451's (*see Figure 8*).

db: What's the oboe situation?

CJ: It's sort of similar to flutes. You have four players, with three oboes and one English horn. The English horn is similar to an oboe, a little longer, with a little ball shape on the end. I use a pair of 451's, splitting the pairs and favoring the principal and English horn. Placement is similar to the basses. I use a floor stand with short boom, coming up from the floor about three inches away. The mic is aimed at the center of the instrument, halfway up the tube. On these instruments, sound comes out the holes in the body as well as the bell. By mic'ing halfway up, I get a better balance between high and low notes. I use the same technique for clarinets and bassoons too, by the way.

db: What surprises do the clarinets hold?

CJ: There are four players that can play three different instruments. Typically, it's three B-flat clarinets and a bass clarinet. Some-

Figure 9. The Baltimore Symphony Orchestra in performance at Oregon Ridge.

times you have an E-flat clarinet player, or several B-flat clarinet players double on E-flat clarinet. The E-flat is a higher register, shorter instrument that goes substantially higher. It can sound sharper, more brilliant than the B-flat, and it cuts really well. I usually don't worry about a separate E-flat mic unless there is an extended solo. We use two EV 408's, which are their top-of-the-line, swivel-mounted, neodymium, cardioid dynamics. I'm very pleased with these mics on clarinet; it's the closest thing to a condenser you can find in a dynamic. I split pairs of players, favoring the principal and bass clarinet, which is a lower registered instrument. If I know about a bass clarinet solo, I would definitely give it a dedicated mic and change somewhere else.

db: Finally, we have the bassoons.

CJ: Again, we have four players, with three bassoons and one contrabassoon. I use a pair of 451's, splitting the pairs. I favor the principal and really favor the contrabassoon. The contra is an extremely low register instrument, and doesn't get very loud.

THE BRASS

db: We now enter the brass section. You've listed two mics for six French horns. How do you manage that?

CJ: What I really like to do is use two Crown PCC-160's, the cardioid PZM mics, and place them on foam, on the floor, behind the section. Strangely enough, I've had arguments with players telling me to mic them from the front. I'm convinced that the majority of the quality sound comes from the bell, as it does on all brass instruments. It's not like a woodwind, where there are all these holes for sound to come out. And as the instrument is played, the bell points to the rear, away from the audience. My real problem is space: the French horns are usually backed right up against the percussion section. The PCC-160's are useless unless you can buy yourself some room in the back. I like to back the mics off a bit to get a better blend, but that's always a logistical problem with the French horns. If I've got to go close, I use a pair of Sennheiser 409's, which are dynamic cardioids. Again, I favor the principal, split the second pair, and forget the third pair.

db: For trumpets, your layout shows a pair of mics y-d. We'll add the caveat that you don't have the channels, as your layouts change from concert to concert. What do you use?

CJ: I've been using EV 408's, like on the clarinets. But I could really use SM-58's if I had to. There's no question that the trumpets are the loudest thing on stage, so I only use these mics for sections that have to be heard, like *Bugler's Holiday* or marches. I use two mics, splitting each pair of trumpets and favoring the principal. Usually, the trumpets are too loud with their mics off.

db: How about the rest of the brass?

CJ: For trombones, I use a pair of Sennheiser 421's y-d, splitting the two pairs and favoring the principal. My tuba mic is an EV RE-20, a large-diaphragm, dynamic mic that's great for low-end instruments. I suspend it over the bell, about a foot-or-two away. The RE-20 is a heavy mic, so we use the large Atlas studio boom stand with the twenty-five pound triangular base.

PERCUSSION

db: You touched on percussion earlier. The two timpani mics are Isomax elements, taped to a stand and positioned one per pair of drums. You also have two extra Isomax for extraneous percussion. What gives there?

CJ: It depends on the program. For *The Star Spangled Banner*, I like to have a snare drum mic. All it's for is the opening of the piece, but it's so crucial that you have to have it. I might never use it again, but I need it for that opening. Many Gershwin pieces have *heavy* xylophone parts, so I need a mic for it. With percussion, you're talking about instruments that don't have sound coming out of a bell, hole, or whatever. You can't just point a mic in one direction and know you've got it. That's where I like the omni pattern of the Isomax.

db: You've got a better margin of error.

CJ: Exactly. On a xylophone, which is about five-feet long, I can put one in the middle and get an even balance between notes. Most extraneous percussion I can get with one omni, because the parts are played loudly. That's the beauty of orchestras: they take care of themselves. The conductor tells them what has to stand out in a certain passage, and it does stand out.

PIANO

db: You have two piano channels: listed are a C-tape and a regular mic. Is this your standard method?

CJ: My favorite method is to use a pair of AKG 460 condensers in a near-coincident pair. I place these in the middle where the strings cross, making sure I don't put them too close to the hammers. If the lid has to be closed, I just tape them to the lid; otherwise, I point them inside with boom stands. If I don't have a soundcheck, I'm more comfortable with a

C-tape and a 460. I know I can get that loud enough on the fly.

db: The only instrument left is the celesta.

CJ: I think of the celesta as a very percussive piano. It looks something like a small upright piano, although there's no acoustic soundboard. It's more like hammers hitting plates. I use an AKG 451, on a floor stand with a short boom, and point the mic at the back center of the instrument.

VOCAL MICS

db: Finally, you have two vocal mics, one of which is a dedicated announce. What's your preferred choice?

CJ: That depends on the vocalist, but I'm fond of the Shure SM-87, a cardioid, condenser vocal mic. Of course, depending on the program, we might have to use several vocal mics. I had to do that with Carol Channing: we needed four vocal mics, all at different locations on stage. I used SM-58's for that one, because it was a handheld situation. I use an SM-57, with a windscreen, for my announce mic because it looks neater.

db: Is the announce positioned near the podium?

CJ: It depends on the conductor: some just swing a boom over to the podium and talk. Others want to step down and move; for them, I put it on a straight stand a few feet away from the podium (*see Figure 9*).

db: You mentioned that you might occasionally use D.I. boxes. What sort of stuff might that be?

CJ: Most of our outdoor concerts tend to be pops programs; anyone who has gone to a symphony more than twice knows what that is. You are going to hear Gershwin, some marches, some movie themes, that sort of thing. With these types of programs, you are going to find trap sets, electric basses, sometimes electric keyboard or synthesizer of some kind. Occasionally, you even have electric guitar, although when that happens I mic the amp, just like you would in rock-and-roll. The electric bass and electric keys I would take D.I., though. One show, we used a synthesizer program for digital cannons on the *1812 Overture*, although they blew that off this year. I don't know why; it's an effective way of doing it if you've got the right sampling.

FOURTH OF JULY AND IT'S IMPLICATIONS

db: You do three concerts with the National Symphony Orchestra, all at the West Lawn of the Capitol building. I understand that the Fourth of July concert there demands a larger commitment than the Memorial and Labor Day concerts.

CJ: That's putting it mildly! The minimum audience size is 100,000 and it could go as high as 300,000. The scope of the event is just mind boggling. This year, we even did a national TV simulcast over the PBS network, through our local affiliate WETA.

db: With that many people, you must add the delay stacks you mentioned earlier.

CJ: We add the delay stacks *and* a backfill as well. For this one day, people sit behind the orchestra, on the Mall. We put up additional scaffolding and hang four additional 490 cabinets on each side of the rearstage area, pointing toward the Washington Monument. The main area delay stacks are three 390 cabinets: these are much smaller three-way cabinets, typically used for smaller PA jobs or as part of a monitor system.

db: What sort of spacing did you use between the stage and the delay stacks?

CJ: It's about two-hundred feet, and since the audience tends to fan out, we put them out wider than the main stacks. We still used the long throws, though, to help push the sound out there. Each delay stack had it's own AC generator, and audio was fed from the house mix location via a wireless link, so we didn't have to run any wires through the crowd.

MIC CHANGES FOR BROADCAST

db: Did your mic'ing change due to the simulcast requirements?

CJ: It certainly did. At the suggestion of the TV engineer, we put a Countryman on every violin. We velcro'ed an Isomax to the bridge of each individual instrument.

db: Now that's the ultimate in multi-channel symphony mic'ing. How many channels did that turn out to be?

CJ: It was pushing ninety. We didn't actually take all those mics; we were fed two different string submixes from the recording truck, and did our own thing on the rest of the orchestra. The mics we shared directly were split via a transformered mult box. If you want to talk about gain-before-feedback, that was the ultimate.

Since the Countryman is a very flat mic, you get just what the player gives, which is nice. No problem getting the strings loud enough here! But we did start to get into all sorts of extraneous noise: when the instrument was set down, when someone accidentally hit their bow against a string. Plus, a lot of players found the idea of a mic attached to their instrument a bit disconcerting—little aside comments to each other during the pieces were picked up.

db: The whole idea of it has profound implications with respect to balance.

CJ: I usually don't have the luxury of a mic on every player, and I'm not sure I'd want to. I think of our multi-mic technique as close area mic'ing, rather than close mic'ing. By mic'ing groups of four, I have a mic within five-feet of any player. I still get a natural, blended sound—something I think would be hard to get with a mic on every player. Yet, I still get the gain I need, something I wouldn't get with distant omnis or coincident pairs. I also like the idea of a little air between the mic and a stringed instrument.

db: Even in the studio, I don't think true close-mic'ing has ever been tried, although I could be wrong. Given the time constraints live, it would be difficult to do. The conductor would have to come out and mix, just to get the voicing correct.

CJ: I think that if the conductors knew how much power engineers had over the sound, they'd be *very* concerned. That certainly isn't the case with four coincidental pairs, but with multiple mic'ing an engineer can drastically change the overall sound of an orchestra, for better or for worse. I've given the option of mic'ing every stringed instrument to the BSO, and they rejected the idea, believing that it would be very difficult to get that nice blend.

db: The day may come when conductors sit at the console and actually mix the performance like pop engineers, voicing the sections as they see fit. You could emphasize strong players, de-emphasize weak ones, change tonal characteristics—the possibilities are mind boggling.

CJ: The technical possibilities are interesting: I think we could achieve a new standard of quality in capturing the sound of a symphony orchestra. But at some point, the human element becomes a factor and performance begins to suffer. I really don't think you can improve on the orchestra balancing itself. I think our method of close area mic'ing is the best compromise between natural orchestral balance and the control needed for projection outdoors.

RCI SPEAKER SYSTEMS

Author's Note:

I spoke with RCI Sound Systems' design engineer and managing partner, Rick Shepard, about the company's unique enclosures. His comments on the evolution of their designs reveal the many concerns involved in the design of a sound system, especially when it must "fly":

The R490 was inspired by the demands of a long-time client, whose annual show at the Capitol Centre in Landover, Maryland grew large enough that our system of stacked components was no longer going to be acceptable—we had to come up with a "flying" PA system within a year. Around this time (summer 1981), there really weren't any commercially available "flying" PA systems except for Meyer; so we decided to build our own.

After talking with many sound engineers, and some of the designers at JBL; we concluded that a horn loaded full-range box was our best choice. We felt the horn loading would give us the ability to optimize the output of each device in the cabinet, while also offering excellent pattern control.

A full-range "all-in-one" cabinet would be easier to set up than components, with less chance for configuration errors.

Along with cabinet design, we also had to develop hanging hardware to support the system. About the only thing everyone in the industry could agree on was the use of chain-motor hoists. We looked at what the large national companies were doing, and realized that it was too restrictive for a smaller company like ours. We wanted the most flexibility (more coverage angles) we could get from a single frame; we couldn't put up a "wall of sound," nor would that have been appropriate for some of our clients. Once we settled on that, the design phase of the hanging hardware paralleled the development of the cabinet. Our first prototypes were ready for trial in December, 1981; we used them for small concerts at local venues to shake the "bugs" out of them. A few minor modifications were made before we considered the design a success. With cabinet dimensions finalized, we could now complete the hanging frame.

NO MARGIN FOR ERROR

There can be no errors in hanging a PA; to sleep at night, we took our ideas to a company that specialized in material handling, since our project was a form of that. After consultation with their engineers and designers, we selected a steel design instead of aluminum: for its longevity and resistance to abuse. A system of staggered points was incorporated into this frame, permitting a rake of 22 and-a-half degrees, 30 degrees, 45 degrees, and 60 degrees horizontal on the cabinets themselves, if desired. The individual cabinet hangers were attached to their respective enclosures only after assembly outside the cabinet was completed. The hangers carry the weight of anything hanging below it, not the cabinet; we didn't want to trust the cabinet to carry more than its own weight—a problem I'd noticed in some other designs. Nylon webbing with Aeroquip fittings was used to suspend the cabinets from the frames and from each other. We got both our straps and the "D" rings for our hangers from a company specializing in truck body building; coincidentally, they also supply Claire Brothers. We sent all our hardware to a laboratory for destructive testing, and found that the weakest link was the "D" ring, which broke at 8,300 pounds (the typical load for this fitting is 750 pounds). Everything else tested to well over 10,000 pounds, so we felt very confident about the safety margin.

RCI cabinets are designed so that they will couple acoustically when "flown" or stacked—a big problem with some of today's designs is that they don't use the laws of physics. The R490 is a 4-way enclosure with a 90-degree horizontal by a 40-degree vertical dispersion. The low-frequency section uses an 18-inch JBL 2240H woofer in a quasi "W" design horn with a 32 Hz flare. Two 12-inch JBL 2202H woofers are located to either side of the low frequency section: these supply midrange in separate 120 Hz exponential horns. Upper midrange is supplied via a Community PC494 horn in conjunction with a JBL 2245J 2-inch compression driver. This is mounted in the center of the low frequency section. The tweeter is a Yamaha 4281B: mounted above the high-mid horn. The cabinet is run 3-way active, with an on-board passive crossover between the compression driver and tweeter; points are 250 Hz, 1.2 kHz, and 6 kHz. We're now using BSS 360 crossovers with 18 dB/octave slopes, and have time and phase aligned the components using a TEF analyzer.

The entire project was completed on time, though not by much! We've been very pleased with the system, and continue to amaze crews and sound engineers with how flexible it is.

A few years ago, we found that we needed a longer throw, where distributed systems or delays were not practical or available. We developed the R240 as a long-throw adjunct to the R490. The R240 was designed primarily for outdoor use, but has proved to be very effective indoors at large halls, or at theaters with deep balconies.

This cabinet has a 40-degree by 40-degree dispersion, and is a mid/high box only. Two 12-inch JBL 2202 woofers are used in the same exponential horn design; highs are handled by two PC442 horn/2425 driver combinations. The same crossover frequencies are used, and the cabinet is configured to couple acoustically with the R490. db

The Creation of a Console

Herewith, case histories of the initial design and subsequent installation of a new console in several venues.

Fort Lauderdale's not a bad place to be, with the exception of the occasional tropical storm or hurricane. Design engineer Tom Graefe certainly thinks so; even though a good part of his time is spent winging off to audio conclaves to address the growing number of people who are discovering the merits of the Sony MXP-3000: a console that bears his mark.

DESIGN CONSIDERATIONS

In 1985, using the MCI JH600 console as a starting point, Graefe undertook the task of coming up with a new console that was quieter, more flexible, and easier to manufacture. As solid as the archetypal MCI was, changes in the way people worked had resulted in numerous modifications. Here was a chance to incorporate the best of those after-market design concepts with the latest high-technology circuitry.

Technological advances included: new hybrid discrete transistor circuits in the front end, quieter op amps and power supplies, a complete reworking of the grounding systems, thermal coupling of components to a heat dissipating conductive substrate in the circuit boards, and 0.01-percent tolerance components. Large crystal oxygen-free copper cable replaced foil-shielded cable for higher conductivity and to keep noise down. Attenuation to where the cables ran within the console also helped to lower the noise figures. Newer ACN's, VCA's, and critical peak detectors topped off the operational refinements.

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Graefe knew that EQ would be the real sticking point. With so many different types of EQ on the market, the only logical way to proceed was to continue the modular approach. After determining what circuits were most often used, five different types of EQ were implemented: a four-band sweep frequency parametric with switchable Q, a four-band sweep frequency parametric with fixed Q, a ten-band graphic, a "Pultec-sounding" LCR (inductive/capacitive/resistive), and a four-band state-variable EQ.

GROUNDING AND CONNECTIONS

"Good grounding is Rule Number 1," notes Graefe. "You have to separate grounds for certain reasons. We run our LED's from a constant current source. But even that little amount of transient difference when they're turned on, if dumped into the analog or audio ground, causes a slight click or pop, at 70 to 80 dB down. So we have that all dumped into a peripheral ground which comes together with other grounds at one point only." Special considerations were given to incorporating new digital control technology with analog audio circuits. "Digital and analog grounds are separate. Otherwise, you dump all that digital stuff into the console and it starts screaming at you."

Integration of the automation-system controls into the console layout presented additional challenges. "There is an automation system in there on an eight-bit bus that runs over ninety inches in the console, and getting that quiet takes a lot of attention to detail. The size of the grounds, the positioning of the grounds, where the grounds are run on the board itself, are equally as critical as the whole system."

Graefe found that soldering crimped connectors reduced noise by 6 dB. When combined with the new grounding system and quieter chips, noise figures at the ACN bus were reduced to better than 90 dB. Depending on the specific routing and modules used past that point, the noise figures fall in the middle to upper 80 dB range. Although efforts to improve the noise figures led Graefe to design considerations that pushed the price of the console above originally projected figures, those efforts paid off. The console soon became so quiet that Sony had to buy new test gear to measure the little noise that remained.

ERGONOMICS

Feedback from MXP-3000 users helped Sony improve the overall physical layout of the console. Graefe explains, "We removed all the buttons, knobs and assignment switches that are normally cluttering up the fader area—the ones that you accidentally hit at 4 a.m. when *you're* fading out, and went with an infrared remote control with a display on the fader."

Graefe takes obvious pride in how much attention was paid to little details. "Using hot-stamped knob markings instead of engraved ones meant that the markings won't become filled with dirt, making them difficult to read. The somewhat boxy meter housings were purposely made that way for safe placement of near-field monitors."

Figure 1. Students at work at the console at the Peabody Conservatory of Music's Recording Arts and Sciences Department.

PATCH BAY CONSIDERATIONS

Nowhere is a console more susceptible to external noise than at the patch bay. Ground loops, aberrant equipment interfaces, and poor conductivity often conspire to make fiction out of noise figures. Graefe explains the special considerations that had to be made to preserve the noise figures he worked so hard to get.

"We went to our own moldings there so that our jack rows are totally isolated from one another. All the chassis grounds are brought back separately to a central ground point. There's no current carried in the chassis anywhere in the patch bay so you have no induced hum or noise. We use metal frame jacks, not the plastic PC mount ones. We're using insulation displacement-type shielded ribbon cable with gold connectors that go to the patch bay."

PHYSICAL AND THERMAL CONSIDERATIONS

Another important but subtle design consideration of the MXP-3000 is its physical displacement in the control room. According to Graefe, if you're spending a lot of money on a control room with flat response to 20 Hz, accurate reproduction of that low end will be destroyed by a console the size of an airplane wing that goes from floor to eye level. Because the MXP-3000 takes up less space, and because of its overall shape, its effect on low-end response in the room is minimal.

Highly efficient heat dissipation, important for long life and stable circuit operation, is achieved without cooling fans. Even so, most of the MXP-3000's working surfaces (including the top panel) are cool to the touch. The thermally conductive substrate in the circuit boards and design considerations which maximize convection currents help keep the console cool. The lack of cooling fans has other advantages. No fans means no physical or electrical fan noise, and fewer moving parts.

Graefe believes the approach to quality taken with the MXP-3000 will make it a solid part of the Sony line for a long time. "I fully expect that twenty years from now the thing will be still sitting around working, with the kind of parts we're using. It's state-of-the-art technology as available today. In terms of analog technology, I don't think anybody's going to come along with a radical new concept."

SONY AT PEABODY

For Alan Kefauver, who is the coordinator of the Recording Arts and Sciences department at the Peabody Conservatory of Music in Baltimore, MD, the decision to go with a Sony MXP-3000 was based on several factors. Even though he visited the Sony factory in Florida to take a first-hand look at the console, the fact that WPS (Washington Professional Systems) had one operating in their showroom was a big plus. The offer to come to the WPS showroom for some hands-on work allowed Kefauver to satisfy his concerns about how well the console would perform.

"I had a module go up once, with a session that afternoon. They drove the module up—you can't ask for much better service than that."

CUSTOMER SUPPORT

Having the console "in-house" also allowed WPS to come to his rescue on several occasions; Kefauver explains, "during the installation, for example, the old mic lines from the studio to the concert hall were found to ring so much that they burned out the mic preamps. Even though it was not Sony's fault, WPS gave me new preamps." According to Greg Lukens, head of sales at WPS, "The MXP-3000 has transformer-less mic preamps with a 200 kHz bandwidth.

The 500-foot, foil-shielded mic cables were oscillating at 200 kHz. We solved the problem by installing Sony's optional transformer mic preamps to support the unusually long mic lines." Keeping the customer happy is one of the many things WPS does well, even at the last minute. Kefauver remembers, "I had a module go up once, with a session that afternoon. They drove the module up—you can't ask for much better service than that."

Following Sony's guidance on grounding for the MXP-3000, the console was connected to the existing Peabody ground field which consists of nine copper stakes interlaced and buried in the ground. According to Kefauver, "When we built the studio; we put in a dedicated ground field. We're not even on the house ground or power company ground. We have our own grounds for safety ground, shield ground, A.C. ground, everything. We run separate neutrals on all of the A.C., and use isolated ground receptacles which run to our own ground field."

The longer Kefauver uses the console; the more he is convinced that it was the right choice.

LOW-LEVEL METERING

Kefauver was the first in the U.S. to order the MXP-3000 with VF (vacuum fluorescent) metering. "For classical music recording, I need metering that goes down to –40 dB to –60 dB. A string quartet playing Mozart gets damn quiet. Standard VU meters which only go down to –20 dB won't do the trick for me."

Other specified options to the 36-input mainframe include: the hard-disc automation, a number of each of the different EQ's, a mix of input modules with and without transformers, an extended patch bay, and extra headphone amps. His experience with mic preamps in a wide range of consoles, including Mitsubishi, SSL, and Neve, still led Kefauver back to Sony.

"When I work in New York; I find that most engineers like to track on Neve and mix on SSL. The Sony MXP-3000 transformer-less mic preamps are incredibly clean and quiet—transparent." The MXP-3000's broadcast mode, which allows the output of the mic preamp to feed both the channel path and monitor path independently, also appeals to Kefauver because it allows him to do a multi-track backup while doing a live stereo mix.

WORKING DOUBLE-TIME

Another striking example of the console's versatility is that Peabody has used the console to record two concerts at the same time: a feat which says all you need to know about its cross-talk performance. Kefauver feels the console is well-suited for training students in the Recording Arts and Sciences program, "it has proven to be extremely user-friendly for freshmen to seniors."

The longer Kefauver uses the console; the more he is convinced that it was the right choice. "Every morning at 8 o'clock it's on and running just as it was the night before when whomever left at midnight...it always works. The students and the faculty here all comment on how nice it sounds—how musical. The bottom line's real simple. It works; it sounds good and it's got dealer support. It's hard to say that about a lot of equipment in this industry."

SONY AND THE ARMY

Graig Greco, who oversees recording for the U.S. Army Band in Washington D.C., had no budget restraints to keep him from going after a more expensive console. Regardless of that, his choice was the Sony MXP-3000.

Greco also appreciates WPS' customer support. In his case, getting and maintaining good support is not easy.

Greco's specs called for thirty-six inputs, eight wild faders, vacuum florescent meters, six of each type of EQ, single mic preamps and transformerless inputs. The decision to run without transformers came after testing a wide selection of their own microphones. Like Kefauver, Greco had worked with Neve and SSL consoles. Like Kefauver, it was the sonic qualities that attracted him to the MXP-3000.

Using Dolby SR and dbx, much of the music he records is complex big band and orchestral material. After hearing how transparent and natural the console was, Greco says WPS didn't have to do a lot of selling. He remembers some of the things that caused him to go Sony: "Greg Lukens had some brass arrangements on tape for a demo. He knows I record a lot of brass. The accuracy of the valve sounds really impressed me. Big band arrangements and trumpet arrangements can create some very strange artifacts when combined. You need to be able to hear what's going on. I feel comfortable with the MXP-3000. The people I work with are purists—they need a certain sonic quality."

Greco also appreciates WPS' customer support. In his case, getting and maintaining good support is not easy. "We're a government office. 'Red Tape' means problems. WPS worked out ways to deal with these problems." Throughout its two years of operation, Greco says the console has been very stable. Based on his experience with Sony reliability, the studio also bought two 5000 series Sony reel-to-reel tape machines.

IN THE PRIVATE SECTOR

At Cubic Studios near Baltimore, Mark Davison runs a 36-input MXP-3000 with full disc-based automation and four wild faders. EQ's include: two parametrics, two graphics, two of Sony's "Pultec-like" LCR EQ's, and thirty standard ones.

Like an increasing number of musicians who record their own projects; he had become tired of lugging tapes from the studio in his home to larger studios for mixdown. His recording projects had reached a stage where his existing console was no longer adequate. With the disc-based automation, for example, Davison can now store several mixes of the same piece. He can then choose parts of each mix and combine them for a finished master.

Davison's ten-year history of dealings with Washington Music (the MI side of WPS) led him to Greg Lukens. Already familiar with Sony quality and proven track record, the fact that WPS had an MXP-3000 on the floor was a major consideration. Other key factors were: automation, EQ, and price. Though he had worked with SSL and Neve (and had also looked into Amek, Trident, Soundcraft, and Soundtracs), customer support concerns added to his decision to go Sony. "They've always been there when I've needed them," says Davison, "and that's what counts." db

RONALD BENNETT

Audio in the Theater

The Broadway-bound show Annie 2 represents the new wave of thinking in production in modern theater. It is a major production, but it closed with poor reviews while still in Washington D.C. after only a few performances. Statements made by the producers at the time (January 1990) said revisions to the book and songs would be made and a new production would be mounted. There were no negative reviews regarding the sound heard in Washington's Kennedy Center.

Abe Jacob created the sound design. Audio technology for live theater has been slow to advance when compared to the advances in recording and broadcast technique. But now, large scale console systems, intelligent equalization, MIDI, and digital processing technology were incorporated by Mr. Jacob—one of the theater's true innovators. Mr. Jacob pioneered and introduced the concept of sound design to the Broadway Theater. His many shows include: *Hair*, *Jesus Christ Superstar*, *Pippin*, *Chicago*, *A Chorus Line*, *The Act*, *The Rocky Horror Picture Show*, *Beatlemania*, *Dancin'*, *Evita*, *Woman of the Year*, *Big Deal*, and *Black and Blue*.

He was responsible for the concert sound for Jimi Hendrix, the Mamas and Papas, Peter, Paul and Mary, and many others. Mr. Jacob is also presently the Executive Secretary for the newly organized Theatrical Sound Designers Association.

THE SHOW'S SOUND SYSTEM

The system he developed for *Annie 2* uses Yamaha consoles for audio control, A Crown PIP system for amplification, and Meyer loudspeakers throughout. The Meyer UM1's are in a cluster of six over the house location and are utilized primarily for vocal reinforcement. The Meyer UPA's are on each side of the presidium for reinforcement of the orchestra. The two systems do not intermix—only vocals appear in the center cluster and only music appears in the side cabinets.

The main mix console is a Yamaha PM3000-40. It accepts the fifteen Sennheiser RF inputs and the rhythm section of the orchestra, including all the drums and piano. The five-foot mics also appear on the console.

A Yamaha PM1800-16 is used for the orchestra sub-mix, primarily strings and woodwinds. A Yamaha DMP7 Digital Mixing Processor is used for effects mixer and receives input from cart machines and DAT machines, and with some specific presets, mixes those as effects, changes EQ and routes them to the PM3000 console. The DMP7 is able to store and recall all of its equalization, level and signal processing settings on cue: a truly powerful device for live performance. Additional signal processing includes Yamaha DEQ7 digital equalizers and Yamaha SPX1000 digital effects processors. Meyer CP10 equalizers are used for both program EQ's and insert EQ.

Mr. Jacob's design philosophy aims at creating a system of "maximum control with maximum invisibility."

He also said, "to reinforce, not amplify the system, the system should be transparent." The system for *Annie 2* is large, versatile, robust, sophisticated, and yet simple to operate. Duncan Edwards, Mr. Jacob's assistant and the production sound engineer for *Annie 2* mixes the show.

Figure 1. Abe Jacob at the console.

He said, "the setup is mixing console and reinforcement system, it is all designed to not break the illusion of the show."

Special effects, both via cart machines and DAT are used throughout the show, but as Mr. Jacob said, "We really only use special effects to create moods."

For example, he uses a Yamaha DEQ7 to create a 1930's radio effect. A Lexicon 480L and the Yamaha REV5 are used together to create a Yankee Stadium sound.

For microphones, they use a Sennheiser MKH70 foot microphone for foot mic'ing, a MKH40 studio microphone is used for the orchestra, with a MKE2 with K3U capsules used for strings, and the radio frequency (RF) mic is the Sennheiser SK2012TVL.

HOT RF

RF mics in the Washington. D.C. area can present some real problems. With all of the government and security communications, the area around the Kennedy Center is one of the hot test RF areas in the United States. A technician was brought in to do RF spectrum analysis and, based on the testing, the system was set to 500 to 600 MHz RF in the UHF band because it was found that this band had the least amount of RF interference. The choice of the 500-600 MHz provides them with some degree of RF protection.

Duncan Edwards is the Production Sound Engineer for *Annie 2*, and Daryl Bornstein is the Sound Design Associate. Mr. Edwards said, "The stage managers are just beginning to learn to deal with RF microphones. They are now designating time for people to come in to be dressed with RF microphones."

Mr. Jacob noted, "In the past, they had put RF microphones on actors at the start of the show, and during breaks and intermission, they would move the belt packs and microphones to different actors to get their lines. The new system puts the microphone on the performers' heads and allows them to change only the RF transmitters if needed. I prefer to leave a speaking performer's RF mic on, barring a special situation, to avoid mix-ups and problems associated with equipment changes—this cuts down on the number of mistakes that can be made."

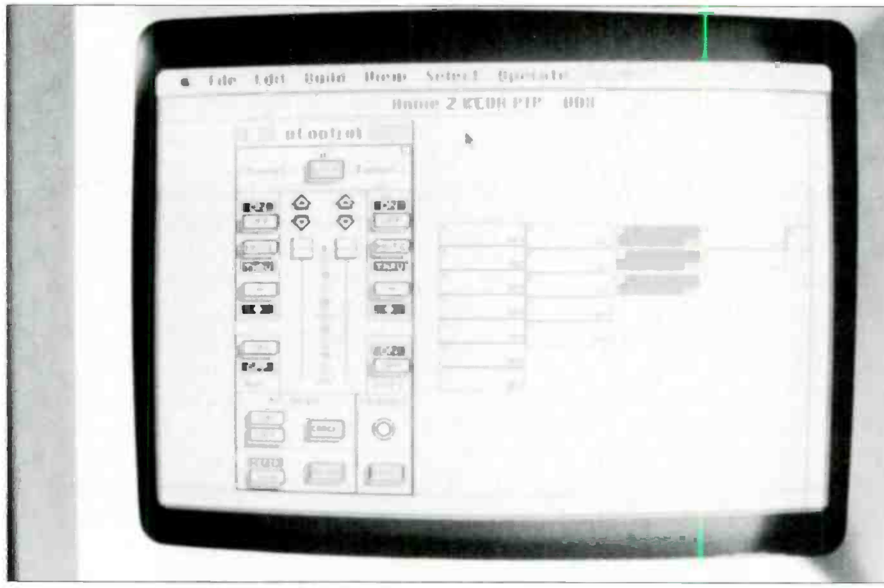

Figure 2. The PIP screen on the Macintosh screen.

STAGE FOLDBACK

In the stage foldback system, they are using Meyer UPM loudspeakers with sub-woofers. They are trying to put only the orchestra in the foldbacks, but often have requests from the performers to include the dialogue RF microphones in the foldback system.

Through a combination of methods, they have been able to show the performers that it is not the best way to work: it is a real educational effort.

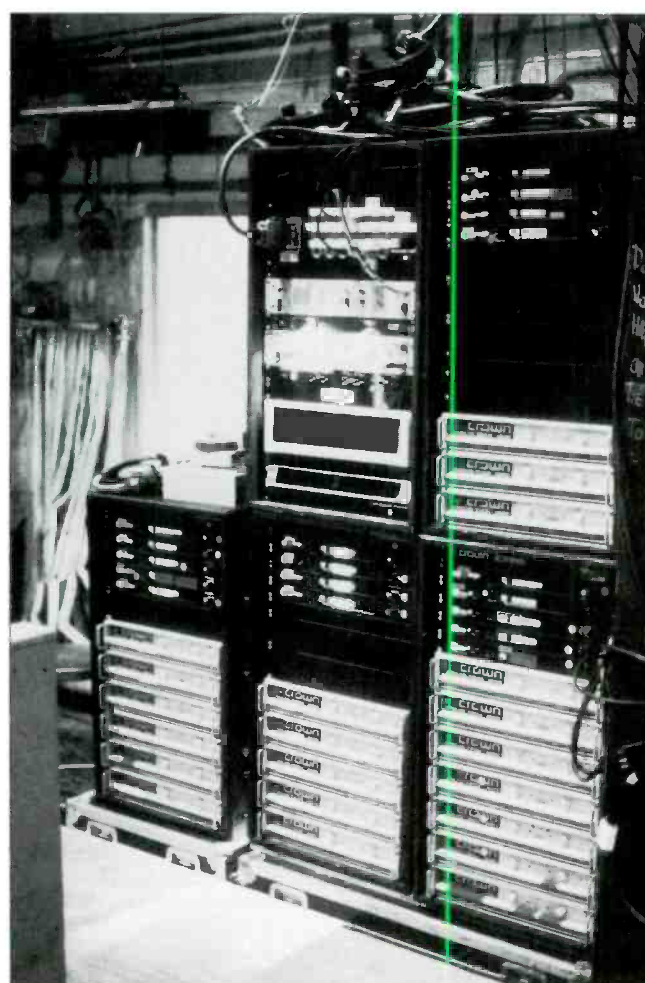

Figure 3. Amplifier racks for the Kennedy Center production.

A very clever low-cost tool used on *Annie 2* is a video camcorder which has been installed to allow the mixer, via a monitor at the mix location, to monitor the RF system. The camcorder also records a picture of the RF system and an audio feed during a performance to allow the mixer to trace down any report of a microphone problem and determine if it is an audio or RF problem. This video recording also serves as a system history of each performance and helps in planning any system changes if they are in order.

The Kennedy Center Opera house has a 2,000 seat capacity. Mr. Jacob designed a PL system with a Vega eight-channel wireless PL and four channels of walkie-talkie. A special part of the performance required a sound effect to come from a prop on the stage. Mr. Jacob designed a unique RF wireless effect unit that uses a small battery-powered guitar amplifier and a battery-powered RF receiver buried into the prop to actually have sound effects coming from within the performance area.

SYSTEM CONTROL

The system output is through a Crown PIP system with MA 1200 amplifiers. This allows the operator to interrogate the amplifier and adjust levels and have interactive control of the system. The Meyer SIM system utilization lets them not only equalize the system, but actually time-align the system for maximum intelligibility and frequency response for both music and speech. "The time delay in the main system is time-delayed to the principle source on stage, wherever that might be. In the future, we hope to be able to program the system to follow the source as it moves around. We will be able, during the show, to change the central time-delay location," said Mr. Jacob.

The SIM system set-up for *Annie 2* is being performed by Kyle Takemori of Meyer, and using the IQ PIP system and MA 1200's from Crown; for any single seat in the house, he can control everything in the theater: he can turn-on/turn-off channels, set up time-delays, adjust levels, equalize the system, balance the system—basically run the system from one location. Mr. Jacob commented on the SIM system, "Once I have the SIM equalization set-up for the principal room in which the play is to be performed, there is a direct correlation in subsequent locations. If the theater is the same size as the room that the show was EQ'ed in, I feel that the system can go from room-to-room; the EQ's and the time-delay do not appreciably change."

Documentation and system history are now of major importance in large audio systems, not just for the basic design, but for the day-to-day management of the system. The entire system is documented in organized three-ring binders for hardcopy, and all system information is stored in the Macintosh computer which also controls the PIP system.

Once again, Mr. Jacob has creatively utilized the latest audio technology and thought, and has developed a robust high quality reinforcement system which will work well in any venue. The sound system is transparent to the audience and truly maintains the illusion of the live performance. db

MARTIN GLASBAND

#107 TEK TEXT

Soundly Engineered A.C.: The Answer To Audio Noise

It's truly surprising how little is understood among electricians and studio personnel about the A.C. power systems that feed sophisticated audio electronics in recording studios. Why is it that technicians resort to ground lifters whenever unwanted noise appears in a studio sound system? Does anyone realize how dangerous that is? Why do they attempt to quiet a studio by driving ground rods which often fail? Why does there seem to be a widely held belief that A.C. power and grounding are inherently "dirty" things? The answers to these questions are quite simple. There are simply no adequate standards in existence at this time. As a result, A.C. power and grounding are lightly regarded and remain very misunderstood.

Recently, at an electrical seminar, it was stated by the instructor that only now are the unique requirements of high-tech installations filtering down to us (electricians). But that was about all he had to say. Aside from isolation transformers and isolated grounding receptacles, not much else is being disseminated. And the noise goes on. So, in the spirit of putting the myths to rest once and for all, it has been undertaken in this article to provide for a comprehensive and clear understanding of A.C. power and grounding techniques for recording studios which will provide for both safe and noise-free operation. There are two areas that will be addressed:

1) A.C. Supply (source of most noise within the studio)

2) Grounding (transmission of noise within the studio and from outside sources)

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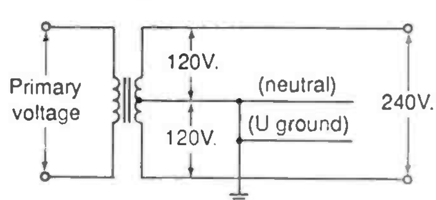
A.C. SUPPLY

To properly appreciate what is being presented here, it is first necessary to accept the idea that studio electronics begin at the A.C. isolation transformer outputs, not at the amplifier or console power supply as is commonly believed to be the case. There are a number of A.C. systems which can furnish a 120-volt supply. *Figure 1* is an example of the most common type of 120-volt A.C. system in the U.S. This system has a dual voltage output (120/240 volts). The center tap on the outputs is grounded and is commonly called the neutral. Even if only one side of the outputs is utilized for studio operation, this system is not truly compatible with audio electronics because it is *unbalanced*. In other words, the wires in a 120-volt circuit have unequal potential to earth or reference ground. White wires (neutrals) in the wiring system are generally an indication that unbalanced power is being fed to the studio electronics.

Figure 1. The most common U.S. 120-volt system.

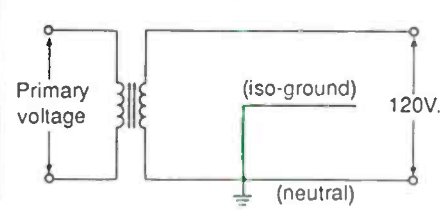


UNBALANCED POWER

Both 120/240-volt delta and 120/208-volt three-phase systems employ a neutral circuit wire when 120 volts is used and, likewise, supply unbalanced power. All systems with neutral conductors supply unbalanced power because neutral conductors are grounded and have no potential. *Figure 2* is an example of how most studios with a 120-volt isolation transformer are commonly supplied. As is evident, the same situation exists here as with the other systems. The grounded (neutral) side of the system unbalances the supply.

Before we discuss balanced A.C. (equi-potential line voltage), let's examine an RF filter on the line side of a power supply and look at the problem unbalanced power creates. As demonstrated in *Figure 3*, the capacitors (which act like resistors in an A.C. circuit) pass current directly into the equipment chassis. There is

Figure 2. The most common studio isolation system.



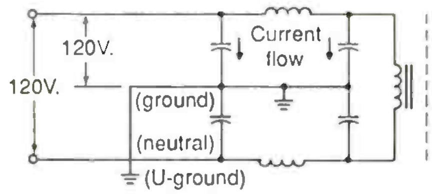


Figure 3. Capacitors act as resistors in A.C. circuits.

a cumulative effect of all RF filters in a studio. The more equipment is used, the more noise gets dumped into the grounding system through the "U"-ground terminals on the receptacles. Though exotic grounding methods have been devised to route the noise away from the system, the matter of eliminating RF at its source has not been addressed. Basically, the more pieces of equipment that are used, the higher the noise floor of the system. Unbalanced A.C. is probably the single greatest *source* of noise in a recording studio.

BALANCED A.C.

There is only one way to deal with this source of noise—balanced A.C. Just what exactly is a balanced A.C. supply? In the electrical industry, this is referred to as a 120-volt split phase (or two-phase) system. Both line conductors have equi-potential (about 60 volts to ground and 180 degrees out of phase) voltage. Now, let's look at *Figure 4*: an RF filter operating with a balanced power input. Note that current flows through the capacitors on *both* sides of the RF filter. However, in this case, no current gets dumped into the chassis because both sides of the filter, operating at the same voltage of 180 degrees out of phase, cancel the potential at the common chassis ground. It's easy to see why balanced A.C. is the only effective way of eliminating A.C. noise at its source.

So, let's now look at *Figure 5*: some 120-volt A.C. systems which are balanced. You probably recognize this transformer as the same one used in *Figure 2*. Most studios with a 120-volt isolated power system have one of these. The difference here in this application is that the ground has been lifted from the secondary—this changes the whole system. Formerly, it was operating in the 120-volt single-phase mode, now it will run as a two-phase system. The main problem with this, however, is that there is no longer a large enough available fault current on either side

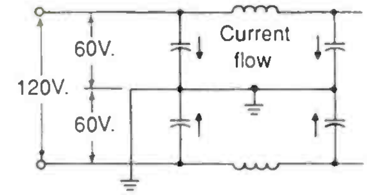


Figure 4. This is an RF filter with a balanced power input.

of the output to cause a circuit breaker to trip. In the event of a short to ground, the entire system will simply revert back to its former single-phase 120-volt mode of operation. The secondary side of the system will only reference to ground.

To a studio engineer, it would probably be somewhat discomforting to think that all of his equipment was draining somewhere out there in the middle of a ground circuit. Probably worse though, is what an electrical inspector would say if he noticed that the studio was operating with an ungrounded system. None of this is intended to discredit the viability of the system. No doubt that the equipment will run a lot cleaner. It's ironic though, when one considers that commercial use of electricity has been around for a comparatively short period of time. Way back when the industry was in its infancy, this was the only type of system commercially available. Now, once again, we are resorting to its use to solve our high-tech problems. Today, ungrounded systems are practically unheard of. Two-phase ungrounded systems are even more scarce. The problem may be in convincing the local authorities that it is safe and viable.

THE BEST SOLUTION

Probably the best solution to all of the above is to change all of the circuit breakers to two pole GFCIs (ground fault circuit interrupters). This should also entail the replacement of all the white neutral conductors with some other color. In the event that some circuit suddenly, for whatever reason, shorted to ground, the GFCI would immediately trip and the integrity of the system would be maintained. This would also rate a solid "10" on a safety scale. Because of the effectiveness of this type of protection, the local electrical inspector will probably OK the system. But in the event that he doesn't, you would be well-advised to change transformers. A local code variance

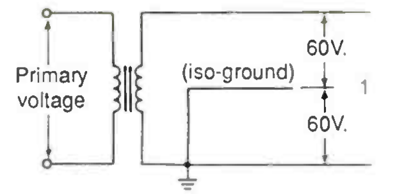


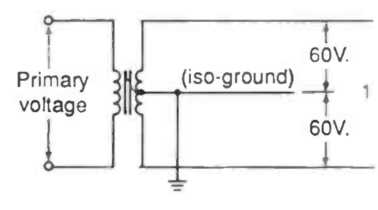
Figure 5. Most studios with a 120-volt isolated power system have one of these.

is possible: citing the use of ungrounded isolated systems in hospital operating rooms (ref. N.E.C. 250-5b ex. 4 & 517-104).

There is one other minor hitch in operating a 120-volt isolation transformer in this way. Commonly, the potential between each side of the system and the ground reference is not perfectly balanced. A difference of 5 to 10 volts is common. This, however, is only a minor imbalance compared to a single-phase system and should probably be ignored. There will still be a drastic reduction in the noise floor. It may be a good idea though, to use a dual voltage monitoring device such as a pair of analog, or analog to digital voltmeters between each A.C. line and ground. This can be a made-up unit and plugged into any receptacle at a convenient location. It is possible, though, to avoid all of these problems.

Let's now look at *Figure 6*: one more type of 120-volt A.C. system. This is the optimum type of the two-phase system for studios. It is referred to as a 120-volt two-phase grounded system. The transformer is called a 240 to 120/60-volt isolation transformer. The primary voltage doesn't necessarily have to be 240 volts. 240 was only used as a specification example for the primary side. 120, 208, 277, and 480 volts are also very common voltages. Care must be taken when ordering the transformer in that the primary voltage is what is available. Also, a center tap on the secondary side is a crucial requirement. 120/60 indi-

Figure 6. This is the optimum type of the two-phase system for studios.



cates a center tap secondary. Be extra sure when ordering your transformer because of the importance of a center tap. The reason for this will soon be apparent.

This transformer should not be confused with a 60/120 output transformer. Both provide 120 volts, but only a 120/60 can be grounded while still maintaining a two-phase supply. Only the 120/60 system has by design, two outputs 180 degrees out of phase, with a common center tap. When the center tap is grounded, equal coil output windings on both sides, rather than a typically uneven magnetic flux dispersion, determine the voltage at each output. This guarantees a near-perfect balance. Furthermore, referencing the center tap to ground enables the system to generate enough fault current potential to trip a circuit breaker. This is a very important safety consideration. Costly GFCI breakers are not needed. The grounded secondary will also please the inspector.

Two pole breakers must be used to feed every circuit in all 120-volt two-phase systems. But as far as overload protection is concerned, conventional two-pole magnetic trip circuit breakers may not provide adequate protection. At lower than usual operating voltages, they may not trip under overload as designed (240-volt circuit voltage). To remedy this, it is recommended that HACR-type breakers be used. HACR breakers provide a secondary tripping mechanism (thermal overload) which relies solely on current. Many standard breakers are already rated as HACR-type breakers, so they shouldn't be hard to find. False alarms are not a problem.

NEW ISOLATION TRANFORMERS

In case you find yourself purchasing a new isolation transformer, there are some other important specifications to be aware of. First of all, there is the matter of the electrostatic shield. It's not commonly known that an isolation transformer's protection against unwanted eddy currents can vary considerably. Use of a transformer with an electrostatic shield will greatly reduce unclean A.C. as a noise source. The specification that applies is the transformer's common mode rejection. In most cases, 80 dB is adequate, but higher noise rejection is available if needed. With the advent of newer and quieter digital equipment, 100 or even 120 dB may be necessary to maintain a sufficiently low noise floor.

The other specification one should consider is transformer temperature rise. Temperature rise is simply an indication of how hot the transformer will run under maximum load conditions. 150 degrees is standard in most cases. But if you are considering growth, or perhaps at times pushing the transformer beyond its rated output, an 80 degree temperature rise will allow the transformer to be overloaded by as much as 35 degrees.

This lack of clear direction pertaining to isolated grounding within a split-phase power system is probably the main reason why studio electrical systems have remained in the Dark Ages up to now.

This could save a lot of money down the road, if your plans call for it. A lower temperature rise could also save money in air conditioning expenses. To maintain clean A.C., it is recommended that several 1:1 A.C. line transformers be kept on hand for use with unbalanced equipment, or otherwise transformerless power supply gear. Without them, noise dumped into the A.C. or ground system is very likely. But despite all of our best efforts, some noise remains in the studio. Not all noise is studio A.C. related. We must look elsewhere to find answers.

GROUNDING

For a variety of reasons, current gets dumped into the grounding system, both from within the studio and from outside sources. The tendency is that high gain and high impedance equipment, being very sensitive to EMI, will detect this noise and pass it along into the rest of the audio system. Depending on the severity of the noise, other equipment as well, can similarly be affected. The task, therefore, is to *suppress* noise transmission without compromising the A.C. ground. Ground lifters effectively block noise transmission, but they do so at the expense of safe operation. By not maintaining short circuit potential in metallic equipment chassis, "live" chassis conditions are possible and circuit breakers are rendered ineffective. The solution is not to create a completely independent audio ground—this would be solving one problem and inadvertently creating another.

It's somewhat comical, if not tragic, to think of all the ground rods stuck in the earth, abandoned, for they failed to perform as intended. Your local electrical inspector would frown on their use anyway, as a separate or supplemental studio ground, because more than one ground source is prohibited. Besides, another ground path would create a loop which could result in even more noise. Within the National Electrical Code (N.E.C.) there are a multitude of regulations surrounding (but relatively few directly addressing) our objective, which is to provide a safe and noise-free grounding system. To add to this confusion, there is no reference at all to 120-volt two-phase grounding, and only one small paragraph which refers to an isolated grounding system. This lack of clear direction pertaining to isolated grounding within a split-phase power system is probably the main reason why studio electrical systems have remained in the Dark Ages up to now. Unfortunately for recording studios, the code is more concerned with safety than with "clean" operation.

WORKING YOUR WAY THROUGH CODE

So here we will make an attempt to beat a path through the various code sections which do apply, and formulate a method of grounding which takes into account both the applicable code safety requirements, and the "clean" requirements of studio electronics. First of all, what is isolated grounding? Isolated grounding is a method of grounding that reduces electrical noise by employing a discreet grounding conductor which does not connect to the conduit system. It utilizes isolated ground receptacles on which the grounding terminal is not mechanically connected to the mounting means of the receptacle. According to the code, the isolated ground circuit wires are to terminate at an equipment grounding bus in the main iso-panel.

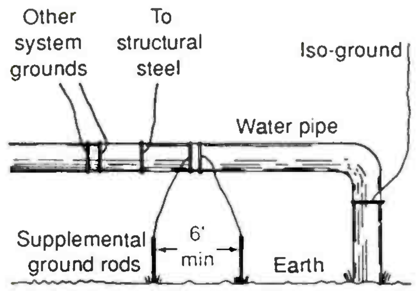


Figure 7.This is an example of what is termed building-noise-reduction grounding.

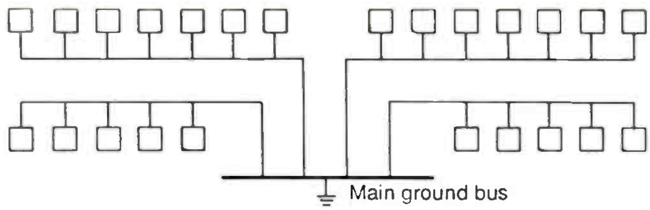


Figure 8. This method is often referred to as daisy-chain grounding.

For our purposes, that too must be isolated, because the panel box is connected to the conduit system.

The next logical question is, where do we go from here? Where do we find a "clean" ground source that is legally permissible? First let's look at the grounding electrodes that are allowed. In the N.E.C., Article 250-54 specifies that only one electrode (system) may be used for all services and equipment in a building. Additionally, in Article 250-26c, effectively grounded cold water pipes and structural steel are listed as the primary types of electrodes to be used for a main grounding connection. This greatly reduces our options. Cold water pipes and structural steel are electrically-bonded together, thus they are part of the same electrode system. This doesn't preclude the use of *supplemental* ground rods, provided that they are integrated into the building's grounding electrode system.

In Article 250-26c, it also states, "The grounding electrode shall be as near as practicable to, and preferably in the same area, as the grounding conductor connection to the system." In most systems, the grounding connection is made at the neutral bus. In a 120-volt two-phase system, there is no neutral bus.

Unlike a single-phase system, none of the power conductors are grounded. However, the system's equipment *is* required to be grounded. Therefore, it is concluded that the only possible point of main grounding connection is the equipment ground bus (main iso-ground bus). So according to the code, it is intended that you connect the iso-ground bus directly to the closest water pipe or structural steel member (bypassing the iso-transformer).

SPECIAL CONDITIONS

When using a 120/60 center tap system, the center tap is regarded as the equivalent to a neutral. The center tap, therefore, is to be grounded as described above. Additionally, the main iso-panel *box* is to be connected to the secondary center tap. The iso-ground *bus* needn't be connected to the center tap, as will be explained shortly.]

In many cases, the nearest cold water pipe or structural steel member is adequate. But keep in mind that in the code, the words "effectively grounded," (primary electrode), and "as near as practicable," are used. Herein lies a matter for interpretation and judgement. Just how "effectively grounded" (clean in an audio sense) *is* this primary electrode? And, just how close is "practicable" (again, clean in an audio sense)? Recently, in a six-story building in Los Angeles, some technician did a spectrum analysis on the structural steel. The results were a veritable rainbow of frequencies present. So, structural steel was ruled out as a primary grounding electrode. Additionally, the cold water pipe in the area (5th floor) already had numerous transformer secondaries grounded to it. Anyone who has ever lifted a live center tap from a cold water pipe has seen the sparks fly. So, how could this water pipe be an effectively grounded electrode, especially when the iso-ground connection is made in close proximity? Quite simply, it is not. Grounding electrode systems are allowed to be as high as 5 ohms above true earth ground.

If an isolated ground circuit in a studio is permitted to circumvent noise in the grounded conduit system, how then could a cold water pipe, with other systems already grounded to it, be any better? There is possibly only one way: *Figure 7* is an example of what is termed building-noise-reduction grounding. The objective here is to be as close as possible to the earth, making any improvements to the grounding electrode system between the iso-ground contact and the other system's ground contacts. This method of primary grounding effectively suppresses noise transmission from sources outside of the studio.

If necessary, a plumber can be called in to install an underground water pipe to a remote faucet for a more isolated ground contact point. Or when there is very dry or sandy soil, a deeply buried chemical ground system may be used where supplemental ground rods are indicated. These improvements to the building's main grounding electrode system are in line with code requirements. Though there may be other primary grounding methods, it is doubtful that there is much room for improvement. When this method is not used, studios are more subject to outside sources of interference. Some noise can be difficult to trace and very expensive to eliminate. Proper primary grounding can be a very valuable first line of defense.

BROADCAST RF

If broadcast RF is a problem, primary grounding is not the answer—shielding is. The only reliable solution to broadcast RF interference is to wrap the control room with copper mesh. Commonly, this method is used in radio stations, where the broadcast tower is in close proximity to the broadcast booth. When installing a room shield, care must be taken to insure continuity throughout the entire screen. Proper grounding of the shield is, of course, also mandatory. Alternative materials which can be used include lead and mu-metal. But these are quite expensive and, in most situations, aren't a significant enough improvement to warrant the extra expense. If outside noise sources other than broadcast RF are suspected, a test can be made by lifting the main ground conductor. If the noise ceases, the source of the interference is certainly outside of the studio. But

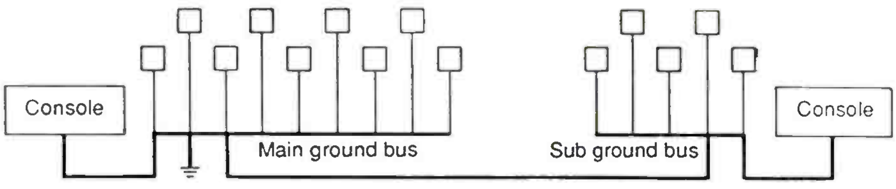


Figure 9. An example of a star-grounding network system.

if the noise continues, the problem is probably within the studio grounding distribution. So, let's now look at some grounding distribution methods and explore the ways of suppressing noise transmission from sources within the studio itself.

Figure 8 is an example of how an isolated ground system is commonly wired. This method is often referred to as daisy-chain grounding. Iso-ground conductors are looped in series from receptacle to receptacle (as chassis are looped from chassis to chassis). What you have here is a typical problem studio with a multitude of non-linear ground planes. Various potentials exist between chassis. 12-gauge wire, commonly used as a ground conductor, is more resistive than one might think. 50 feet contains approximately .1 ohms resistance.

The problem is generally one where the resistance between chassis is less than the resistance between each chassis and the reference ground. There is no common linear ground plane. EMI transmission along a common ground circuit often creates problems. One massive ground wire run throughout the system to every "U"-ground would be ideal, but the cost would be prohibitive.

There is, however, a cheaper alternative: *Figure 9* is an example of what is called a star-grounding network. Though star-grounding has been around for a few years, its implementation has been rather slow. In conjunction with a properly designed, isolated main grounding system, it is a very efficient way of suppressing noise transmission, and its use is highly recommended. For the sake of emphasis, and for those who have not yet heard of star-grounding, let's review it. In essence, each iso-receptacle has its own *discrete ground conductor* routed to a common ground bus. Where there is more than one breaker panel, modifications can be made to avoid extremely long ground runs. Additional ground buses appropriately called "star clusters" should be installed in each panel. These additional ground buses are individually fed by a large ground wire originating at the main iso-ground bus. The objective is to maintain a relatively linear ground reference plane. Each chassis is individually referenced to the same reference ground plane.

If current somehow gets dumped into the system, the potential is uniformly raised in every chassis from a single reference ground point. Additionally, the resistance between any two chassis, in every case, is greater than the resistance between each chassis and the system ground reference. Therefore, noise transmission is greatly reduced.

Even a very noisy chassis tends to become invisible to the rest of the system. To achieve a more uniform potential in equipment chassis, the length of the ground wires should be considered. Use of larger conductors compensate for longer runs. There is a table of trade size conductor properties in Chapter 8, Table 9 of the N.E.C. for sizing ground wires.

Properly balanced A.C. supply eliminates the source of most noise, and properly designed grounding will effectively block any residual noise transmission.

Perhaps the most difficult place to install a star-ground system is in equipment racks. Metallic chassis bonding and plug strips defeat a star-ground system. There are, however, ways of coping with this situation. Racks made of wood, or other non-conductive material used in conjunction with field-made plug strips (such as 2100 series Wiremold), maintain the integrity of the star-grounding. At the very least, digital and high impedance equipment should be mechanically isolated and discretely grounded. In some cases, where switching power supplies are used in digital gear, high-frequency hash may be dumped into the grounding system. This can pose a unique problem in a studio grounding network. The reason for this is that high-frequency noise requires a very large ground conductor to be effectively dissipated. This can be especially true when digital and non-digital gear are located in close proximity to the ground bus. Rather than go to the expense of installing huge ground wires, an alternative method is suggested.

A cheaper solution is to install a separated grounding system for digital equipment. This would, of course, be called a double-star system. This alternate system, needing only to meet minimum code requirements as digital equipment, is far more "forgiving" than analog equipment.

A separate ground bus is installed, with its own main ground conductor, to the nearest available primary-grounding electrode. The iso-ground circuit wire can be switched to the alternate ground bus whenever needed: this will effectively isolate digital equipment noise from the rest of the audio system.

CONCLUSION

Running an audio system without proper equipment grounding will work, but it is very unsafe. A.C. circuits need a ground reference because circuit breakers are rendered ineffective without a low-resistance ground fault path.

By not maintaining this short circuit potential, extremely hazardous conditions can develop. As far as an electrician is concerned, there is no other reason for a ground wire. To electronics personnel, however, a ground represents a shield against unwanted noise. Grounding, as with electricians, is used as a reference.

But these are two distinctly different applications. It takes careful integration of both to insure that a studio grounding system both protects electrically and shields electronically. A.C. noise problems can easily be avoided.

Properly balanced A.C. supply eliminates the source of most noise, and properly designed grounding will effectively block any residual noise transmission. There is no need for compromise when careful A.C. engineering is employed. db

BRENT HARSHBARGER

Audio for the Church

Equalization in Church Sound

• One of the most widely used devices in audio is the equalizer. An equalizer can be an extremely useful tool if used correctly. Equalization is one of the most involved ingredients in the field of audio, and if used wrong, the worst component in audio. To begin this topic, we will be looking at the various types of equalizers, followed by some applications on how to use them properly.

To clearly understand equalization, we have to understand a little about what sound is. Sound is made up of vibrations. For example, if you pluck a guitar string, it will move back and forth. Each time it moves in the opposite direction it pushes air molecules; and we determine its frequency by how many times it goes back and forth in a complete cycle in one second. The audio spectrum starts at 20 cycles per second (or Hertz) and goes to 20,000 Hertz (20 kHz), although only young people can hear in this range because our hearing deteriorates with age (*Figure 1*).

Octaves, *Fundamentals*, and *Harmonics*. These are terms that describe the relationship between different frequencies. Musical sounds are made up of fundamentals, which are usually the strongest frequency. For example, if you hit a concert "A" on the piano, the fundamental frequency is 440 Hz. An octave above that is two times the fundamental or 880 Hz, and an octave below would be 220 Hz. Again, we will use the 440 Hz to explain harmonics. Harmonics is what gives us timbre in music.

Therefore, if the fundamental frequency (also called the first harmonic) is 440 Hz; the second harmonic is 880 Hz, or twice the fundamental; and the third harmonic is three times the fundamental, which would be 1320 Hz. Knowing about octaves, fundamentals, and harmonics will help you understand the equalizers and what they control.

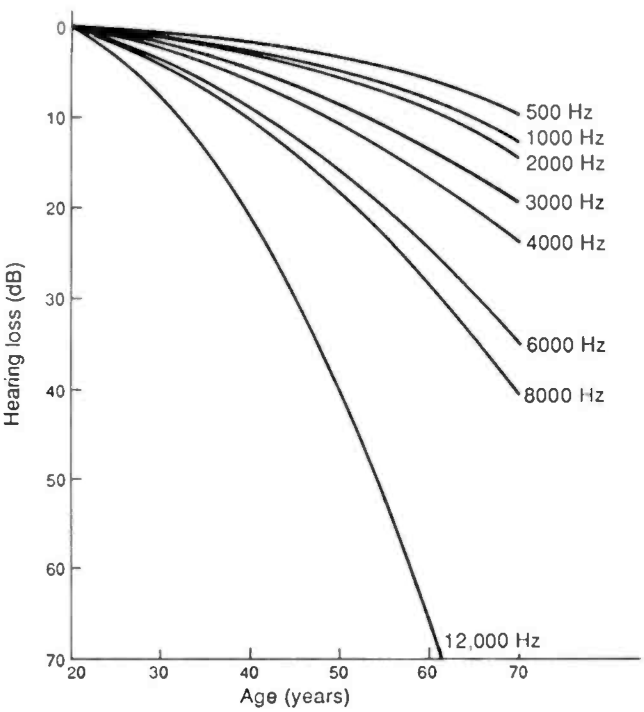


Figure 1. Hearing loss tends to deteriorate with age, although environment also plays a significant part.

EQUALIZERS ARE EVERYWHERE

Everyone has used an equalizer at some point in time. The equalizer (in one of its simplest forms) is the bass and treble controls on a consumer stereo. By turning the tone controls in either direction, we increase or decrease the tone. This type of equalizer works by boosting or cutting above or below a specified frequency. *Figure 2* displays an example of how this works. The frequency that we are going to choose is 100 Hz. Use the treble and bass pots. If we were to take the bass pot and turn it all the way to the right; we would increase or boost all the frequencies from 100 Hz and below to a +12 dB level. Now, if we were to turn the bass pot all the way to the left; we would decrease or cut all the frequencies below 100 Hz. It would work the same way for the treble pot but at the other end of the frequency spectrum. Therefore, if someone came up on Sunday and said that you have an 80 Hz shelving bass control; you would know exactly what you are controlling with that pot.

Most low-to-medium priced mixing consoles have a three-band equalizer on each channel. This consists of high (treble) and low (bass) frequency shelving type, with a third pot for a mid-frequency. The mid-frequency is what is called a peak/dip or boost/cut, which works on a single frequency. *Figure 3* displays how this works. The single frequency in this illustration is 800 Hz, which is called the center frequency. Even though we are only increasing or decreasing a single frequency centered at 800 Hz, the adjacent frequencies will also be affected. How many of these

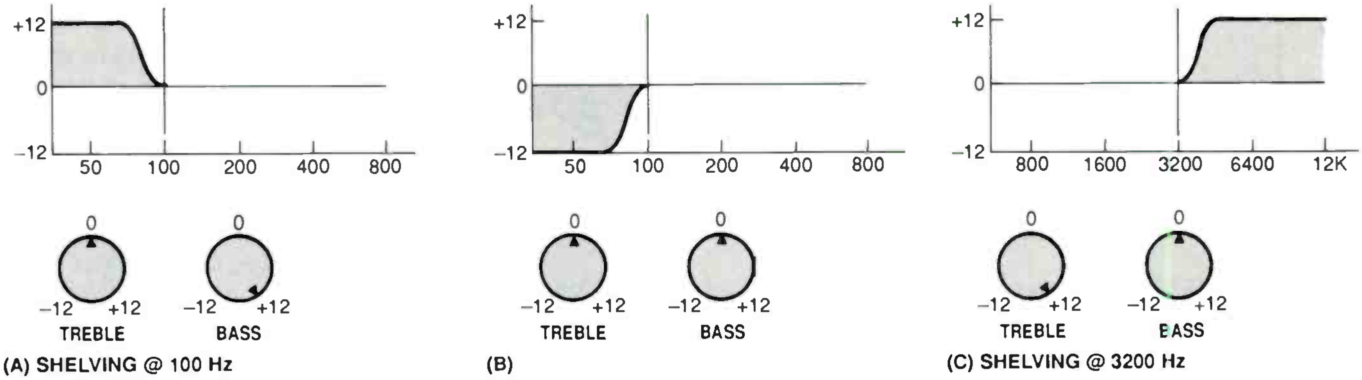


Figure 2. Basic tone controls. At (A) bass boost, at (B) bass cut, and at (C) treble boost.

adjacent frequencies is determined by what is referred to as "Q". If the Q is referred to as a Hi-Q, then it affects very few frequencies adjacent to the center frequencies; and if it has a Low-Q, then it affects several frequencies (*Figure 4*).

SWEEPABLE EQ

So far, we have been talking about several "fixed" equalizers, which simply means that the frequencies are set and cannot be changed. More and more mixing consoles are coming out on the market with variable center frequencies (what is commonly called sweepable or sweep eq). *Figure 5* shows that we can now move the boost/cut center frequency to a more desired location. For example, sometimes an acoustic guitar has a tone that when strummed hangs on longer than any other. If it sounds undesirable, we can remedy this by boosting the sweepable frequency, then turning the sweep pot until that frequency jumps out, and then turning the boost/cut pot to cut that frequency until it sounds natural (and in place with the rest). Also, be aware that poor mic'ing can also cause this problem; therefore, check your mic'ing first, before you work it out with EQ.

Now, onto everyone's favorite play toy—the graphic equalizer. The two most popular graphic equalizers are the octave and the third octave. The ISO (International Standard Organization) has a standard on the center frequencies for the graphic equalizer, which most manufacturers follow. The ISO center frequencies for an octave EQ are:

16, 31.5, 63, 125, 250, 500, 1,000, 2,000, 4,000, 8,000, and 16,000 Hz.

Although most manufacturers follow this standard; you will still only find 10 sliders instead of the 11 frequencies indicated because most speakers cannot reproduce this low, and if they could, we couldn't hear it. The third octave equalizer has three bands per octave which are:

16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1,000, 1,250, 1,600, 2,000, 2,500, 3,150, 4,000, 5,000, 6,300, 8,000, 10,000, 12,500, and 16,000 Hz.

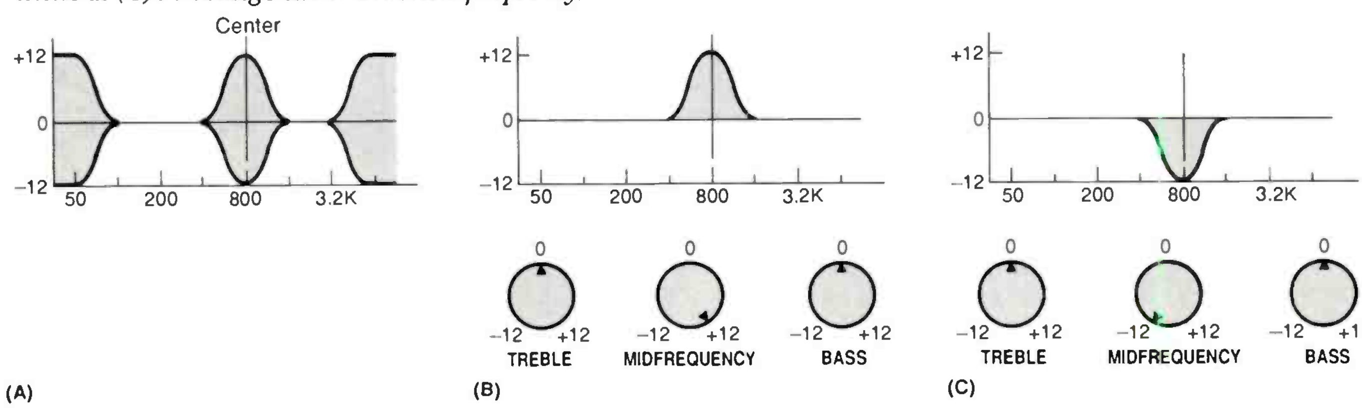
Again, notice that most third octave equalizers do not have the frequencies below 31.5 Hz. Graphic equalizers have fixed frequencies and fixed "Q's" and they work with the boost/cut characteristic displayed in *Figure 6*.

TYPICAL CHURCH EQ

In a typical church audio system, we usually have two equalizers in the audio path; the first being the EQ in the mixing console's channel, while the second is the house or monitor EQ. Both equalizers serve different purposes. The equalizer in the mixing console is used to give the microphone the flattest (or smoothest) frequency response. The graphic equalizer is used to give the speakers (along with the rest of the house) the flattest frequency response. Therefore, any tonal adjustments should be made with the equalizer on the mixing console, while the graphic or house equalizer should never be touched once it is set.

Before going on any further, I would like to use the story I used last month as an illustration, but taking it a step further. For those of you tuning in for the first time; or for those of you who missed last issue's segment, I'll briefly go over it. The story was about a church choir having problems with hearing: "We just can't hear!," the choir said, although the SPL was 6-9 dB above what the house was getting. So we turned off the monitors until the service started and brought the level up until the music pastor said it was

Figure 3. A three-way equalization control. At (A) maximum range, at (B) midrange boost centered at 800 Hz, while at (C) midrange cut at the same frequency.



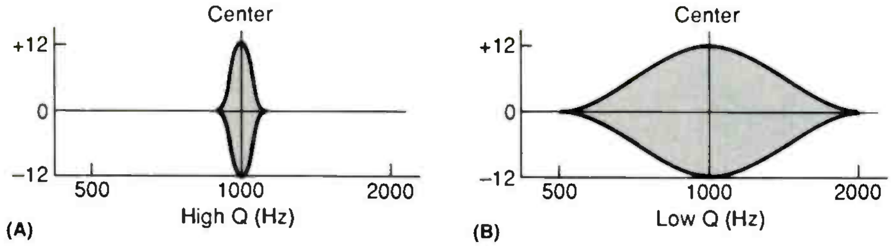


Figure 4. At (A) Hi-Q offers a narrow frequency band, while at (B) low-Q spreads it out.

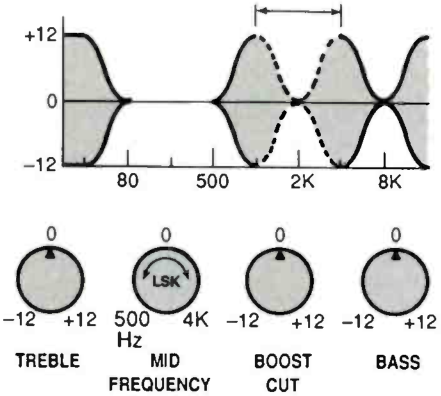


Figure 5. Sweepable control permits you to move the center frequencies around.

OK. Well, when the pastor said *stop*, the level was lower than we normally would have set it. Problem solved. Or was it? Yes and no. Yes, we cured the symptom, but we didn't solve the problem. What was the problem? The problem was the location of the choir monitors. The choir was sitting in an acoustic shadow; therefore, they were not getting enough direct sound from the speakers—making it unintelligible. It wasn't that the level was too low, because when we went from nothing to something; the choir then had a sense of direction of where the sound was coming from.

My point of reference to equalization is to make sure that you are working on the cure, instead of a symptom. Make sure your mic placement is right. Make sure you don't have too many mic's open at one time. And you better make sure your speaker placement is proven correct; or you will make things even worse by equalizing.

SUITING EQ TO THE NEED

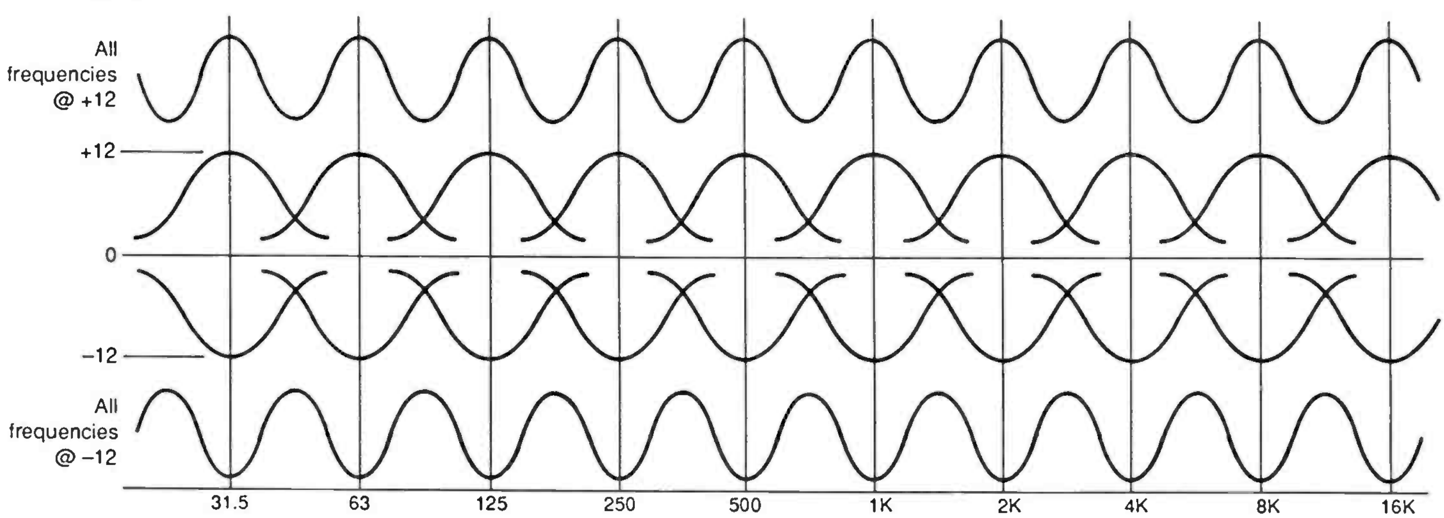
Sound systems are used for different purposes, so equalization curves should suit the need. For example, my friends who go to a Church of Christ service don't believe that musical instruments should be used in worship, so they would require a different curve than let's say; an Assemblies of God service, who usually have an aggressive music program in comparison. The curve at the Church of Christ would look like *Figure 7a*, while the Assemblies of God Church would use the curve in *Figure 7b*. Some of you are probably saying, "I thought you wanted a flat frequency response." Well, that is true for the most part. In a system designed for music reproduction; the sound would come across too bright for the average listener, and would also cause a sibilence problem in a system designed for speech. Therefore, you would equalize flat with a preferred listening curve which is similar to *Figure 7c*. I should also add at this point that you do not want equalization to compensate for hearing loss because this varies widely with each individual.

EQ EQUIPMENT

To equalize a sound system you will need at least, a pink noise generator, real time analyzer, and a sound pressure level meter. If you don't have these tools, then you have no business trying to equalize a sound system; and you should leave it to a qualified sound contractor, electro-acoustic consultant or technician.

Before turning on your pink noise to start tuning, we have to have all the sound system devices set, including our crossover. First, place the RTA's microphone in front of the speaker array in the seating area. Next, turn on the pink noise, and starting with the low frequencies,

Figure 6. Graphic equalizers have fixed frequencies and fixed "Q" and they work with the boost/cut characteristics shown.



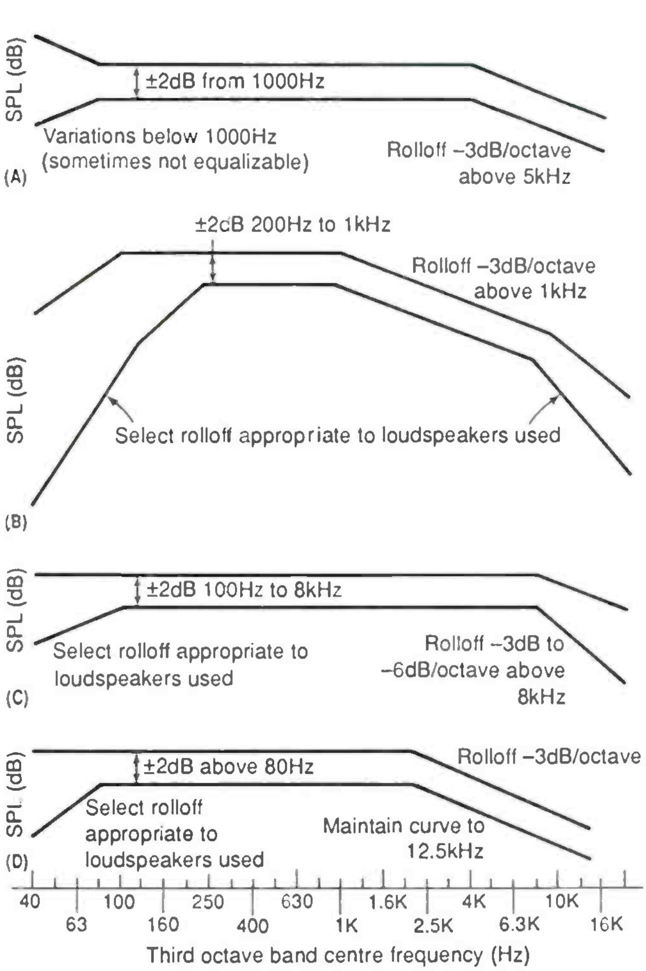


Figure 7. Different equalization can be set for different conditions. Note that controls are not necessarily set "flat."

work your way up the spectrum and flatten the response. Now, add the preferred listening curve that works best for your application.

Finally, turn off your pink noise and turn on a microphone that you would normally use in a service (usually the pulpit works best here), and slowly bring up the gain until you get feedback. Cut the frequency giving you problems about –2 dB, and also cut the adjacent frequencies about –1 dB. Repeat this until no one frequency stands out. When you reach the point that you hear multiple feedback frequencies; you will start to work against yourself if you continue to try to notch feedback. It's time to plug in the CD player and take a listen and make your final decision on your tuning job. Sometimes you will have to make minor adjustments on the overall curve to make it just right.

I hope this overview has given you a better understanding of how and why we use equalization in church audio applications. In our next segment, we are going to be looking at multi-track recording, along with an inside view of a church taking advantage of multi-track technology. db

RANDY HOFFNER

Broadcast Audio

New Zealand: Home Of The World's Newest Television Network

• G'Day, Mate! I've recently returned from New Zealand, where I had the privilege of helping to launch TV3, New Zealand's first private television service. Although New Zealand has had privately-owned radio stations for several years, this marks the first competition ever for the government-owned television services. It was a unique experience to spend five weeks among the "Kiwis," as New Zealanders call themselves. I enjoyed getting to know the people, the country, and its television broadcasting structure, which differs greatly from our own.

GEOGRAPHY AND BACKGROUND

New Zealand is a country that consists primarily of two islands, which although narrow, span a total top-to-bottom length of about 1,000 miles. In land area, the country is about the size of England. Cook Strait, the twelve-mile-wide waterway which divides the two islands, creates a wind tunnel between them that has earned the capital city of Wellington, located at the bottom of the North Island, the title of "The Windiest Capital City in the World."

New Zealand is located about 1,200 miles southeast of Australia, its closest neighbor. The northern section of the North Island enjoys an almost sub-tropical climate. This is the site of Auckland, the country's largest city, comprising nearly a third of its population. The southern tip of the South Island, at about 47 degrees southern latitude, is the second-closest land mass to Antarctica, after the lower tip of South America. Although the South Island's climate is colder than that of the North Island, the fact that these are geographically isolated islands gives even the South Island a much more moderate climate than might be expected by its latitude.

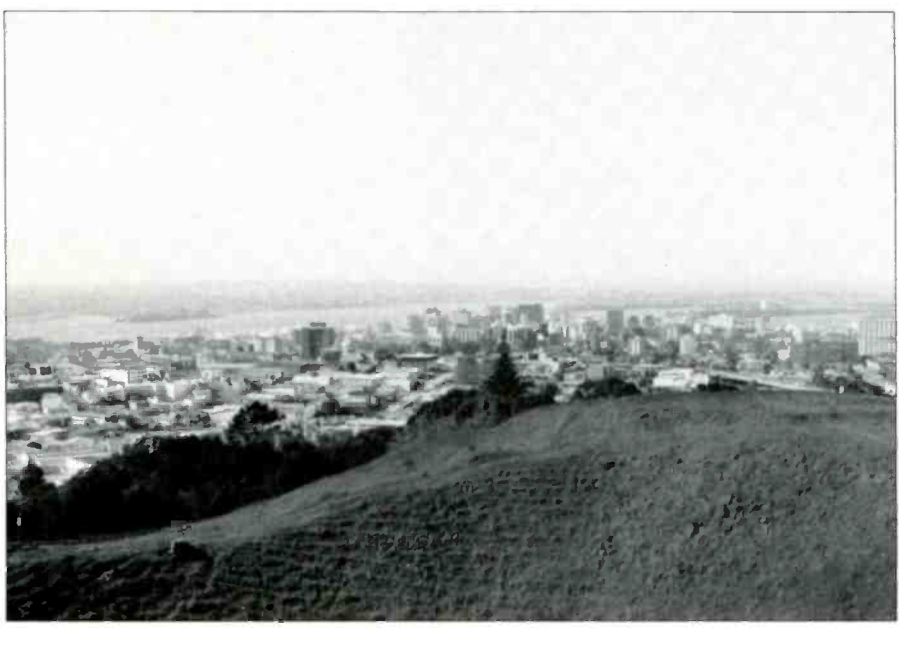
New Zealand is certainly one of the most beautiful countries in the world. It is located on a volcanic ridge, and extinct and dormant volcanoes abound. The most prominent landmark in the Auckland harbor is the island of Rangitoto, a dormant volcano that last erupted about 300 years ago. Volcanic activity results in hot springs at Rotorua, in the central North Island, that is a favorite attraction for both locals and tourists. The plant life in the north includes palm trees, giant tree ferns, and the ancient giant kaurie trees. Auckland's climate, its coastal location, and its large complement of beautiful and uncrowded beaches have nurtured an outdoor lifestyle with emphasis on such activities as "barbies" and boating. The South Island is home to the Southern Alps, where some of the world's best skiing is enjoyed, and a large section of the west coast of the South Island is covered with a rain forest.

The grassy, mountainous terrain supports a vast livestock farming industry, with sheep and dairy cattle being raised in abundance. New Zealand boasts that it is home to 70 million sheep, but I have it on good authority that sheep under one-year-of-age are not included in that count and that in fact, at certain times of the year, there are over 100 million sheep in this country of 3.5 million people. It is no surprise then that agriculture, and in particular sheep and sheep products, are New Zealand's principal exports.

The indigenous population of New Zealand is the Maori people. The Maoris are a Polynesian people that came to New Zealand in canoes about 400 years ago. English colonists arrived in the nineteenth century.

A treaty was signed in 1840, between the Maori people and the English explorers, that permitted peaceful colonization to take place. 1990 is a year of celebration to commemorate the 150th year of that treaty. An amiable pattern of

Figure 1. Downtown Auckland, with an extinct volcano crater in the foreground. Ridges visible on the hillside are artifacts from terrace farming by Maoris in past centuries.



cooperative living and intermarriage between New Zealanders of English descent and the Maoris has been the norm since; with recent generations embracing a strong sentiment for the preservation of the Maori heritage and culture.

TVNZ

From 1962, which marked the beginning of television broadcasting in New Zealand, until November 26, 1989, the Kiwis had only the two networks of the state-owned television service to watch. This service was a part of the Broadcasting Corporation of New Zealand (BCNZ). A few years ago, BCNZ was reorganized, and two government-owned corporations, Television New Zealand (TVNZ), and Broadcast Communications Limited (BCL) were created.

These are expected to function as businesses competing in the commercial marketplace. Concurrently, the groundwork was laid to open television broadcasting to private entities. TVNZ consists of the two television networks TV1 and TV2, which now sell commercials to support themselves. BCL owns and operates the transmission facilities used by TVNZ, and contracts to perform services for both TVNZ and outside entities. Several TV3 transmission facilities share antennas with TVNZ, and BCL has constructed and maintains some of TV3's transmission facilities as well.

TV3

On the evening of November 26, 1989, the long-awaited competitor to TVNZ emerged when TV3 went on-air. TV3's debut was preceded by a continuous one-and-a-half-week transmission of their test pattern, accompanied by promotional activities with local radio broadcasters and newspapers, to increase awareness of TV3's presence (although it is difficult to imagine that anyone in the country was unaware that TV3 was coming). This test pattern broadcast enabled the local populace to tune in TV3 on their sets. New Zealand television receivers are similar to those used in Europe, being tailored for a situation where there have traditionally been only a few television services offered, as opposed to our all-channel, cable-ready sets in the United States which cater for our multiplicity of outlets. Here in the U.S., if we wish to view a local TV station on channel 4; we simply switch the channel selector to "4",and there it is. Older New Zealand receivers have five or six program selection buttons—each of which must be preset to the desired television channel. The electronic-tuning counterpart of these button-type sets has a number of "programs" which may be preset to any broadcasting channel (frequency allocation) that the viewer wishes. Like TV1 and TV2, the channel or frequency allocation used by TV3 varies depending on the geographical area, but that channel would not appear on a "program" or a button on a viewer's set, because there was no reason to have tuned it in before. So the viewer, or a serviceman, must preset a "program" to the appropriate channel in order to view the new service.

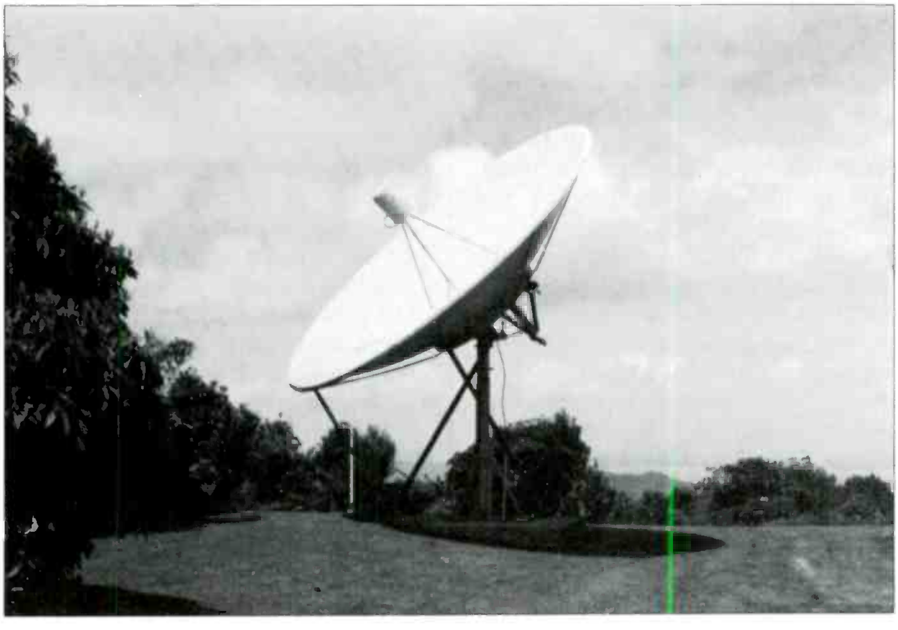
TRANSMISSION

The structure of television broadcasting is completely different in New Zealand from what we know of in the United States. TV3 is a network that will cover the entire country when it is completed, and at its inception serves about 75 percent of the population. The network originates a single program at its Auckland studios, and this program is linked throughout the North and South Islands to its transmitting stations, all owned and operated by TV3. United States television broadcasting networks are very complex, and each actually consists of a number of networks, but New Zealand's TV3 is actually somewhat like a television station with seven transmitting sites instead of one. Even time-shifting by the network is unnecessary, because although New Zealand is about a thousand miles long, it is so slender that it falls completely within one time zone.

TRANSMITTER SITES

On launch day, the TV3 network consisted of seven transmitter sites, scattered around the North and South Islands to cover the major population centers. There will be additional transmitting sites added over the next two years, beginning to come on-line in January, 1990. The high-power transmitting stations will be augmented with a number of low-power repeaters to fill in coverage gaps in the mountainous terrain. The network headquarters is in Auckland, the country's largest city, and of course, there is a transmitter site serving Auckland. There are also transmitter sites in the Hamilton/Cambridge area, the Western Bay of Plenty, Palmerston North, and Wellington on the North Island: all the transmitters are co-sited with TVNZ transmission facilities. The program audio and video signals are carried on a microwave studio-transmitter link from the TV3 studio to the Auckland transmitter, and via a digital linking network

Figure 2. The satellite earth station atop Waiatarua, site of the transmitting tower that serves Auckland.



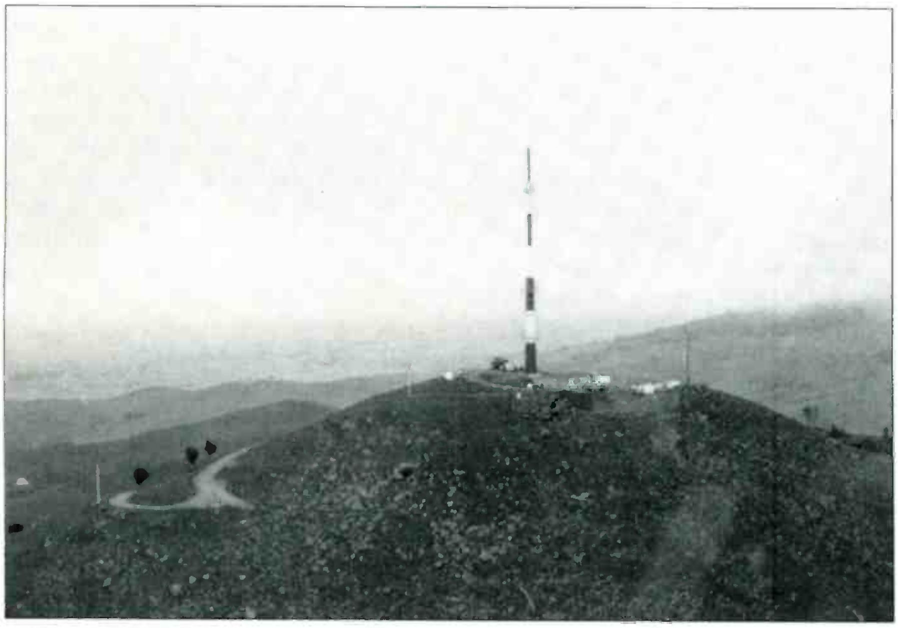


Figure 3. TV3's new tower on Ruru Hill, serving the Hamilton/Cambridge area of the North Island.

supplied by New Zealand Telecom to the other transmitter sites. The transmitters are all controlled from the Auckland studios using a single PC to control and meter all seven sites simultaneously via a two-way data stream on a network consisting of STL subcarriers, UHF radio links, and a Telecom digital circuit. The multi-site control/metering system is expandable, and is expected to ultimately control twenty sites.

CHANNELS

Until recently, there were only ten VHF television channel allocations in New Zealand, channel 1 through channel 10 (yes, there really is a channel 1). These VHF channels are each 7 MHz wide conforming to Standard "B", and PAL video is broadcast on them. Channel 1 is quite low in frequency, extending from 44 to 51 MHz, and is used only for translator service in a few areas. The frequencies occupied by the New Zealand VHF channels differs from those occupied by corresponding channels in the United States, which is not surprising considering that each channel is 7 MHz wide as opposed to our 6 MHz channels. Although the frequency blocks occupied by many "B" Standard VHF stations in Europe are largely the same as those in New Zealand, the channel numbers assigned to those blocks are not the same; so while a European television receiver will work in New Zealand, the channel numbers will be different. As an example, in Auckland, TV3 is on channel 7, which ranges from 195 to 202 MHz. That spectrum is occupied by portions of channel 10 and channel 11 in the U.S., and in Europe corresponds to channel 8. Channel 11 was recently added to the New Zealand VHF TV spectrum. The spectrum allocated to channel 11 was formerly used by the New Zealand military. There is a UHF television spectrum allocation in New Zealand, although as of yet, there is no UHF service.

STEREO TV

Television stereo is available in New Zealand, using the BBC-developed NICAM 728 digital stereo system. The NICAM (Near Instantaneous Compression And Multiplexing) 728 (kilobits per second data rate) system permits the transmission of two channels of audio, either stereo or two completely different programs; on a second, low-amplitude aural carrier located just above the monaural FM carrier.

This system has a 32 kHz sample rate providing 15 kHz audio bandwidth, and initial 14-bit resolution compressed to 11 bits. It is interesting to note that this is the only television stereo transmission system in use worldwide that does not rely on some form of sum-and-difference matrixing. The monaural signal is transmitted completely separately from the discrete left and right channel stereo signal.

This system would thus afford the opportunity, if desired, to transmit separate mono and stereo mixes, although to the best of my knowledge, that is not done in practice. The usual technique is to send discrete two-channel stereo to the transmitter sites, at which point a sum of left and right is generated to modulate the monaural carrier.

Atypically, the Kiwis did not follow the lead of their closest neighbor, Australia, on TV stereo, as the Australians use the German analog stereo transmission system, in which the sum of left and right channels appears on the main aural carrier and the left channel only modulates a second FM carrier.

This is a type of matrixed stereo system, albeit a far different one from our own. New Zealanders cannot buy a stereo television receiver on a visit to Australia with the expectation that it will receive stereo back home.

FACILITIES

TV3 headquarters features large studios and state-of-the-art production, editing, and distribution facilities. On-air playback is from one-inch and D-2 composite digital video tape and commercial and news segments are aired from half-inch tape cassettes using a robotic cassette handling system. Additionally, TV3 has a studio in the capital city of Wellington. There are also "itinerant contribution" facilities for news feeds at several locations around the country. These are unattended phone booth-like facilities where a reporter may plug in to a Telecom link and feed news pieces back to Auckland. Like cable television, satellite news-gathering has not yet arrived in New Zealand. TV3 does, however, have access to world satellite news services, including NBC Skycom and Visnews.

CONCLUSION

The long-awaited launch of TV3 is now history, and the world's newest television network is providing a brand new television service to the people of New Zealand. It was an exciting project to be involved in, and I hope you have found reading about New Zealand and its television broadcasting system interesting. db

LEN FELDMAN

Lab Report

Shure Model 1200 Powermixer Amplifier

The Model 1200 is shown atop two optional Model 3200 speakers.

GENERAL INFORMATION

• The Model 1200 Powermixer, by Shure Brothers Incorporated, is one component of that company's Audiomaster System. Other elements of the system include the Model 3200 Loudspeaker, the Model A 1200MX Expansion Module and the Model A 1200C Portable Case. The Model 1200 Powermixer tested for this report is rack-mountable or portable. It is a high-power, 6-input mixer amplifier, expandable to 8 or 10 inputs. Inputs accept both high and low impedance microphones as well as amplified instruments or other high level sources. Two of the six available inputs can be used for high (line level) inputs or for microphone inputs, while the remaining four inputs are intended only for microphone input. High and low impedance inputs may be used simultaneously. Individual input attenuation controls, with overload indicators, are continuously adjustable to prevent input overload. The unit has built-in reverberation

Figure 1. Frequency response of the Shure Audiomaster 1200.

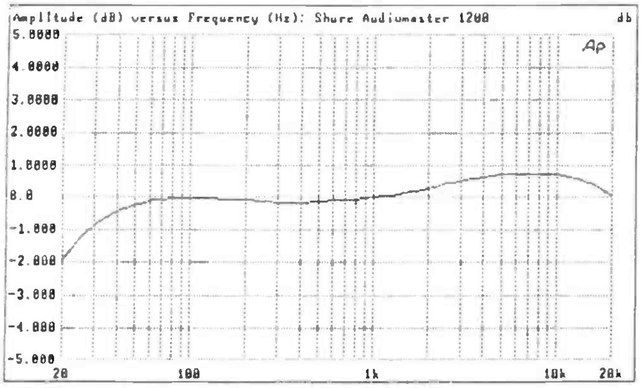
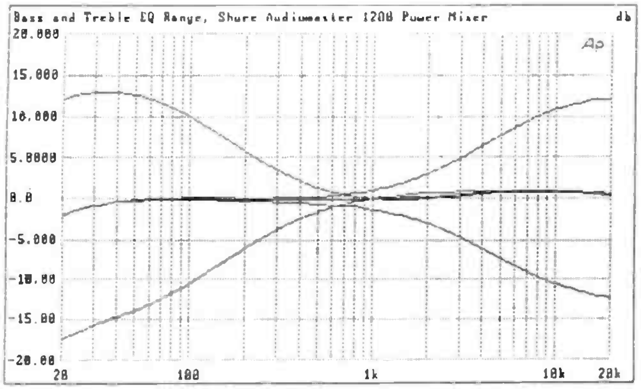


Figure 2. The range of bass and treble controls versus frequency.



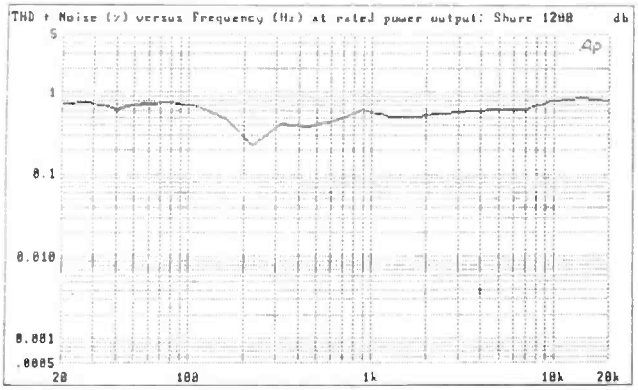
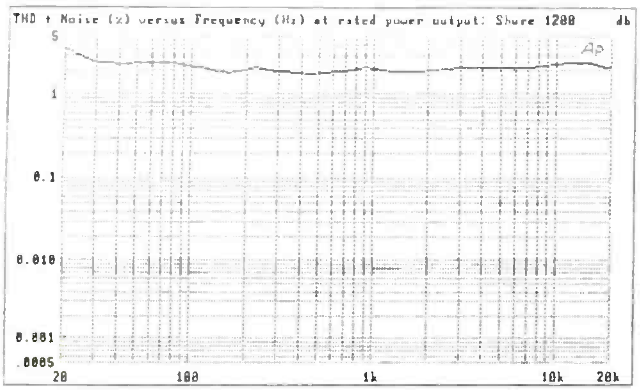


Figure 3(A). Harmonic distorion plus noise versus frequency at rated output of 120 W/channel at 8 ohms.

with individual channel and overall level adjustments. Other effects devices or an equalizer can be easily connected via a post-master control program loop. A limiter in the power amp section prevents overload distortion over a wide input signal range.

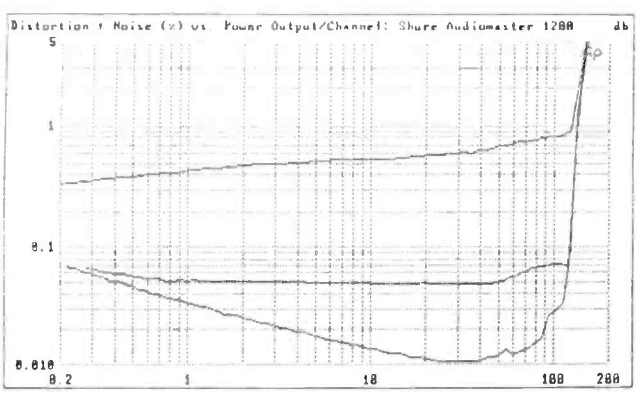
Pre-volume monitor "send" controls and a pre-master volume Tape Output provide extra flexibility in performance or recording applications. The control panel (about

Figure 3(B). Harmonic distorion plus noise versus frequency at rated output of 200 W/channel at 4 ohms.



which more detail in a moment) is sensibly engineered with color-coded knobs that prevent errors during adjustment. LED indicators give instant visual notice of overload and operating status. Shure tells us that the unit was designed for churches, schools, clubs, civic and business organizations and small performing groups.

Figure 4(A). Harmonic distortion plus noise versus power output per channel (8-ohm loads). Best curve is 1 kHz; next best is 20 kHz; poorest is for 20 Hz.



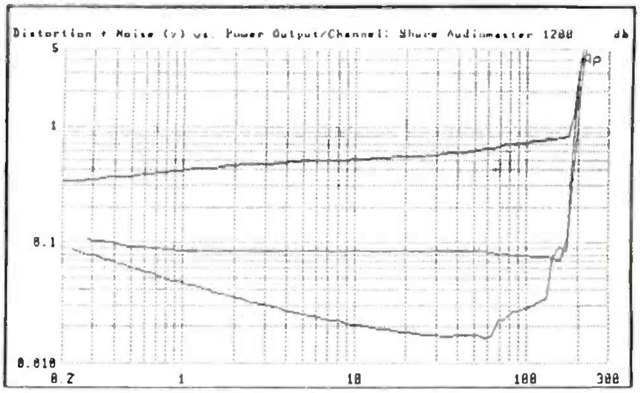


Figure 4(B). Harmonic distortion plus noise versus power output per channel (4-ohm loads). Best curve is 1 kHz; next best is 20 kHz; poorest is for 20 Hz.

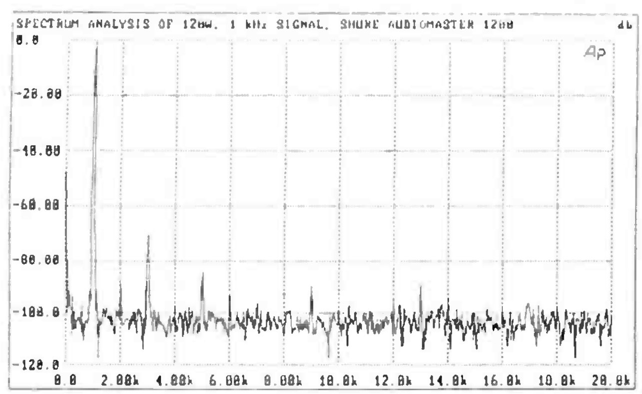
The unit is fairly lightweight and compact, as such products go, measuring only 7-inches in height and weighing only about 27 pounds.

CONTROL LAYOUT

All six input modules, positioned vertically along the left section of the Model 1200, have identical, colored control knobs. Starting at the top, a "Monitor" knob controls channel monitor "Send" independently of channel volume control. The user would adjust this knob when using a monitor system or for auxiliary or tape recording mix. Next, a reverberation control determines the amount of reverberation added to each channel. Concentrically mounted bass and treble equalization knobs come next. Below the EQ knobs is an input overload indicator. The input attenuator below that indicator, has an attenuation range from 0 dB (no attenuation) to –30 dB, with calibration marks at 0, –6, –12, –24 and –30 dB. Finally, the channel volume control allows individual setting of channel input for the desired signal mix. When a channel is unused, this volume control is set fully counterclockwise.

Over at the right side of the panel is a master control module. Its uppermost control is a monitor master volume knob. Below that is a "Reverb Return" knob that controls the reverb mix level to program mix. Dual concentrically mounted bass and treble controls alter tonal response of the monitor mix signal. A power amplifier limiter in/out switch, when in the "in" position, prevents overload distortion at the speaker and headphone outputs by not allowing the power amplifier to be over-

Figure 5. Spectrum analysis of 120 W. A 1 kHz signal using an 8-ohm load.



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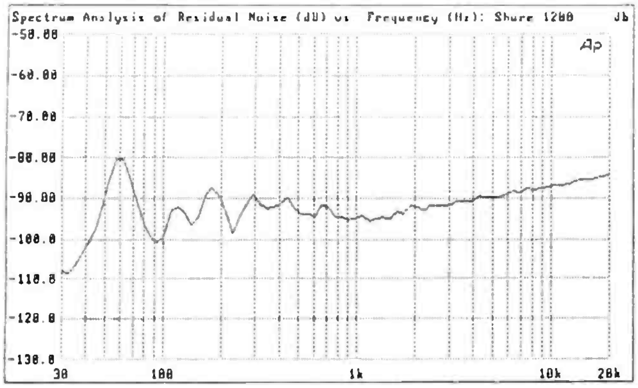


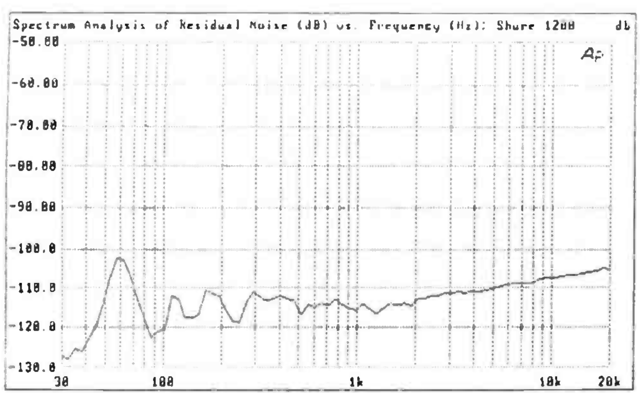
Figure 6(A). Spectrum analysis of residual noise, 500 mV input, 1 watt output reference level.

driven. A master volume control adjusts overall level of the mixed signal at the Program Mix output, Speaker Output, and Headphones Output. Further down along this master module are a headphone volume control and a standard ¼-inch headphone jack that accepts 4- to 4000-ohm mono or stereo headphones.

The main power switch is located at the lower right hand corner of the front panel. To its left is a green LED that lights up when power is turned on. A yellow LED above it lights up whenever signal is present at the amplifier output. A red LED mounted still higher on the panel lights just before clipping occurs or, if the limiter is turned on, at the onset of limiting. Another red LED flashes to warn of overload or unsafe operating conditions. It lights continuously if the power amp shuts down because of overheating, or because of an abnormal speaker load, such as a short circuit.

The rear panel of the Model 1200 is equipped with low-impedance XLR balanced input connectors as well as ¼-inch high-impedance unbalanced phone jacks for each of its six input channels. Above the phone jacks associated with Channels 1 and 2 are Mic/Aux switches that permit these channels to be used either for microphones or for high level inputs. Blanks in the rear panel accept Shure's Model A1200MX Input expansion modules, each containing two channels, so you can add a total of four more inputs. The rear panel also houses a Monitor Output jack, a Tape Output Jack to Speaker Output jacks, Program Mix Output and Power Amplifier Input jacks (these two comprise the external device loop facility) and a Reverb Defeat jack.

Figure 6(B). Spectrum analysis of residual noise, typical control settings at midpoint, referred to 120 watts output.



All of these jacks, including the speaker output jacks are ¼-inch phone plug type jacks. We found it a bit unusual that Shure chose to use phone jacks of this type for accessing the amplifier's speaker outputs, but realized that if you were to purchase the Shure Model 3200 loudspeakers that are an integral part of the Shure Audiomaster system, these too would be fitted with such jacks, and connection would be made using an accessory cable (A50SC) fitted with phone plugs at each end. Two such cables were supplied by Shure for our tests, as was a cable fitted with XLR connectors at each end. An unswitched AC convenience outlet and a phantom power on/off switch completes the rear panel layout. The phantom power switch, if turned on, will supply 24 volts of DC to condenser microphones connected to any of the inputs.

LABORATORY MEASUREMENTS

Figure 1 shows the frequency response of the amplifier, when fed with a 50 mV (high level) signal via one of the balanced inputs. Published specification were easily met, since Shure quotes response as extending from 40 Hz to 20 kHz, +1, –3 dB and our test sample clearly did better than that, with bass response off by only –2.0 dB at 20 Hz.

Figure 2 shows the maximum boost and cut range of the EQ controls for each channel. Results were almost exactly the ±10 dB at 100 Hz and 10 kHz claimed by Shure in their published specifications.

Figure 3A is a plot of total harmonic distortion plus noise versus frequency, with the amplifier driving an 8-ohm load and with power output regulated to maintain a constant 120 watts output at all frequencies shown. At 1 kHz, THD plus noise measured 0.58 percent, while at no frequency did THD plus noise exceed the 1 percent rating assigned by Shure for a 1 kHz test signal at 120 watts, into an 8 ohm load. The same test was repeated using a 4-ohm load. Results, shown in *Figure 3B*, were not quite as good. While Shure claims a THD with this load of 1 percent for 200 watts output, we measured closer to 2 percent at that power output level.

Figures 4A and 4B show how THD plus noise varied as a function of power output using an 8-ohm load (*Figure 4A*) and a 4-ohm load (*Figure 4B*). Notice that highest distortion was obtained for a test frequency of 20 Hz in each case while, as expected, lowest distortion occurred using the mid-frequency test signal of 1 kHz.

We wanted to separate actual THD from residual noise for at least a 1 kHz test signal. To do this, we employed the spectrum analysis capabilities of our test equipment, with which we analyzed the spectral content of a 1 kHz test signal at an output level of 120 watts, into an 8-ohm load. The chief component of distortion seen in the graph of *Figure 5* is, as expected, a third harmonic of the fundamental 1 kHz test signal. This harmonic distortion component was some 70 dB below the reference output level. That corresponds to a 3rd harmonic distortion percentage of around 0.03 percent.

Shure quotes signal-to-noise ratio referred to rated output as –80 dB "with typical control settings." We're not quite sure just where those control settings were when they made their measurements, but for every kind of S/N measurement we made, results were *actually* much better than that. For example, using the amplifier measurement standard developed by the old IHF and

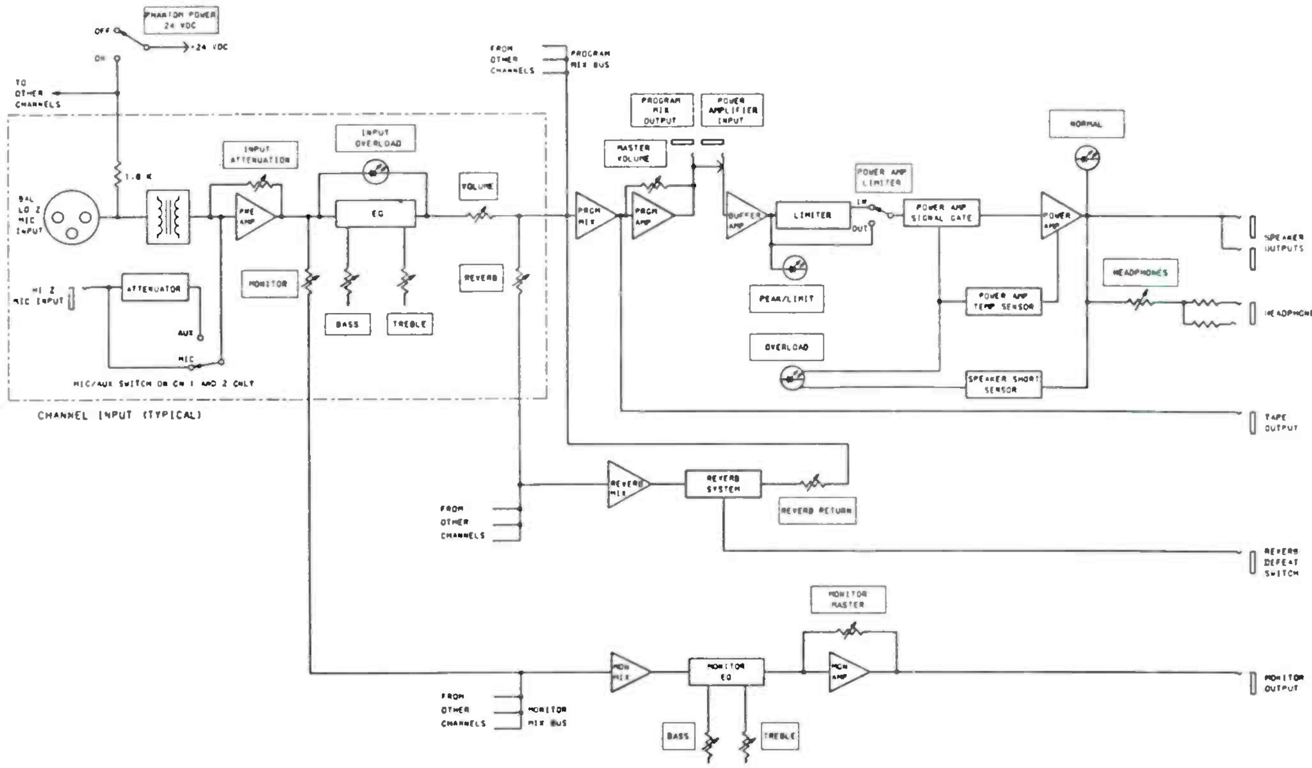


Figure 7. A block diagram of the Shure Audiomaster 1200.

now sanctioned by the EIA, we measured S/N of 98.35 dB referred to an input level of 500 mV, with input attenuator and volume controls adjusted to deliver 1 watt output. We also analyzed the spectral content of this noise, using the same reference input and output levels, with results shown in *Figure 6A*. The chief "noise" contribution was actually power supply hum, with a noise peak clearly visible at 60 Hz and smaller peaks noticeable at 120 Hz and 180 Hz.

This analysis was repeated with controls set to their mid-points and with dB readings referenced to 120 watts of output (*Figure 6B*). The fundamental shape of the curve remained the same (with the same power-supply related peaks visible), but of course, the dB readings of S/N at the various frequencies were now much higher because of the higher output reference level.

Finally, we measured SMPTE-IM distortion at the equivalent of 120 watts (for 8-ohm load conditions) and 200 watts (for 4-ohm load conditions). The test signal consisted of 60 Hz and 7000 Hz sine wave signals combined in a 4:1 amplitude ratio. The two readings obtained were 0.116 percent and 0.558 percent respectively.

CONCLUSIONS

To fully appreciate the versatility of this amplifier/mixer, you should not only spend some time using it, but should also study the signal flow-path as illustrated in the block diagram of the Model 1200: found in the owner's manual and reproduced here as *Figure 7*. Shure Brothers are past masters at this sort of equipment, having been in the business of sound reinforcement products for as long as we can remember. It occurred to me as I was using the equipment that a compact all-in-one mixer amplifier such as this could easily replace some multi-component systems I have encountered recently—systems that take up many times the space of the Model 1200 and don't offer nearly as many features or as much flexibility.

As an experiment, I set up a system consisting of a couple of high level inputs (a mono AM radio feed and a TV sound feed) to the first two channel inputs (having set them for high-level input) and connected a couple of microphones to channels 3 and 4. In almost no time flat I had a good balance for all of these inputs and, after a bit of experimentation, the small amount of reverb that I added to the mic channels made my "amateur" singing group sound a lot better than they had a right to sound. A bit of bass boost on one of the mic channels improved the overall sound quality of my "hobbyist" announcer's voice. While all of these experiments with the Shure Model 1200 Powermixer were of an unofficial nature, let me assure you that the equipment itself is very professional in every sense of that word. Even the colors of the knobs made sense and those colors will be especially helpful if you have to work as a mixer in conditions of subdued lighting where even the clear nomenclature found on the front panel of this mixer/amplifier may not be readable. All in all, Shure Brothers Incorporated have created the kind of product that I would have expected from them—nothing short of a superb one. db

Editor's note: Shure AUDIOMASTER systems, of which this unit is a part, include speaker systems, microphones, cables, etc. For example, one of the matching speakers, Model 3200, are $470.00 each.For complete information on these systems circle 75 on this issue's Reader Service Card.

VITAL STATISTICS

SPECIFICATION	MFR'S CLAIM	db MEASURED
Frequency Response ±3 dB,	40 Hz to 20 kHz	+0.8,–0.5 dB
Power Output, 1 kHz, 1% THD	200 W @ 4 ohms	185 W @ 4 ohms
	120 W @ 8 ohms	125 W @ 8 ohms
Low & Hi Freq EQ	±10 dB @ 100 & 10 kHz	Confirmed
Input Sensitivity (full output)		
Balance Lo Z	0.5 mV	0.48 mV
Hi Z	6.3 mV	6.0 mV
Aux	50 mV	Confirmed
PA Input	0 dBV (1 V)	Confirmed
Voltage Gain @ 1 kHz		
Lo Z (Speaker)*	95 dB	96 dB
Hi Z (Speaker)*	73 dB	Confirmed
AUX (Speaker)*	55 dB	Confirmed
PA (Speaker)*	29 dB	Confirmed
Output Clipping Level		
Speaker	28 V	28.2 V
Monitor	7.9 V	8.0 V
Program Mix	7.9 V	8.0 V
Phones	6.3 V	6.5 V
Tape	7.9 V	8.0 V
Impedance (Inputs)		
Lo Z Mic	1 kohm	Use with 75 to 600 ohms
Hi Z Mic	130 kohm	Use with 100 k or less
Aux	50 kohms	Use with 10k or less
PA	50 kohms	Use with 10k or less
Impedance (Outputs)		
Monitor	2.4 kohms	Use with 2 k or more
Program Mix	2.4 kohms	Use with 2 k or more
Tape	2.4 kohms	Use with 2 k or more
Phones	430 ohms	Use with 4 ohms or more
Speaker	—	Use 4 ohms or more
Signal-to-Noise Ratio	80 dB	98.3 dB
Power Requirements	120 VAC, 50/60 Hz, 420 Watts	Confirmed
Dimensions (HxWxD, in.)	7-½ X 19 X 13-½	Confirmed
Weight	27 lbs.	Confirmed
Price: $ 1,200.00		

*Program mix 30 dB less gain than speaker out, monitor out is 24 dB less than speaker out, phones out is 13 dB less than speaker out and tape out is 44 dB less than speaker out.

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Amplifiers

Introduction to the Charts

We've tried to make the charts of amplifiers as self-explanatory as possible, with slanting headlines on each column that explain what we wanted to show you.

These charts represent entirely what each of the respective manufacturers have sent us in response to our (sometime repeated) requests. You will also see that there are numbers of blank sections within the charts. If they don't have a specification available, we can't list it. But note that many do not have anything under the Features column. This column is where we have invited each manufacturer to state, in as few words as possible, what is special about the product. You can safely assume, then, that when this column is blank, it is because the manufacturers told us nothing.

Note also that we ask for amplifier continuous power not only at the traditional 8 and 4 ohm resistive loads, but also at 2 ohms. As you know, when you parallel speakers, the load is halved. Accordingly, in the real worlds of studio monitors and headphone lines, and the even more real world of performance and stadium systems, effective loads back to an amplifier can well be 2 or 3 ohms. Since modern solid-state amplifiers can handle such loads successfully, we ask each manufacturer for this specification. Note that not all give it. It's, therefore, safe to assume that if it is missing, the amplifier may not be reliable at low loads.

Distortion at normal and full power ratings is also specified. While many amplifiers today can boast of almost vanishing distortion, remember that if you will be pushing an amplifier hard up against its rated power and beyond, distortion will then be rising rapidly. No audio product is really made to be abused, and amplifiers are no exception.

One group of important specifications deals with dimensions and weights. Amplifiers, particularly high-power ones, are not lightweights. A few racks can have weights adding up rapidly.

Finally, the price. What we have asked each manufacturer for is the suggested retail price. Different retail dealers establish their own.

On to the charts...

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
ALTEC LANSING															
9442A	100	150			10-50k -3	.10	.10	.10	.10	10-50k -3	.89	5.25 19 14.75	32	$770.00	Choice of XLR, ¼-in. and terminal inputs, accesory octal socket, peak and protection LEDs.
1270C	2	220	400		20-20k 0.5	.05	.05	.3	.1	7-33k -3	.89	5.25 19 15	51.5	$1850.00	Choice of XLR, ¼-in. and terminal inputs, peak and protection LEDs, two-speed fan cooling.
1407A	1	75	75		20-20k -3	.01	.01	.2	.1	10-30k -3	.78	5.25 19 12.5	24.2	$580.00	Includes XLR (male and female), phono and terminal inputs, direct and X-former output, protection.
1415A	1	150	150		20-20k -3	.01	.01	.2	.1	10-30k -3	.78	5.25 19 12.5	30.8	$698.00	XLR (male and female), phono and terminal inputs, driver protection relay, convection cooling.
2200A	var				20-15k 0.5			.5	.25	10-30k -3	.61	7 19 17.4	70	$5100.00	Eight 75W power amp modules that can be configured for numerous output combinations up to 600W.
2204A	4	75 (4ch)	150 (2ch)	300 (1ch)	20-15k 0.5			.5	.25	10-30k -3	.78	5.25 19 16	31.5	$1750.00	Four 8 ohm/75W power outputs that can be paralleled and/or bridged, X-former isolated inputs.
9444A	2	200	300		10-70k -3	.05	.05	.2	.1	10-85k -3	.81	5.25 19 12.75	39	$958.00	Electronically balanced XLR or terminal inputs, octal socket, peak and protection LEDs.
ARX SYSTEMS —See our ad on page 2															
SS1200VC	2	400	600		10-20k			.03	.05	10-20k 0.25	1.5	3.5 12 19	28	$1349.00	Conventional power supply for drive output stages.
SS600VC	2	250	325		10-20k			.03	.05	10-20k 0.25	1.5	3.5 12 19	22	$999.00	As Above.
ASHLY AUDIO, INC.															
FET-2000C	2	300	500	675	20-20k	.004	.01	.004	.01	20 20k	1.7	5.25 19 16	60	$999.99	Barrier strip inputs,MOSFET,UL listed. FT2000M same with peak reading meters—$1059.99.
FET1500C	2	200	300	360	20-20k	.004	.01	.004	.01	20 20k	1.7	3.5 19 16	42	$799.99	Barrier strip inputs,MOSFET,UL listed. FT1500M same with peak reading meters,XLRs—$859.99.
FET1000C	2	120	190	225	20-20k	.004	.01	.004	.01	20-20k	1.7	3.5 19 16	37	$659.99	Barrier strip inputs,MOSFET,UL listed. FT1000M same with peak reading meters,XLRs—$699.99.
BGW SYSTEMS, INC.															
7500T	2	200	250		20-20k		.05			3-100k	1.22	5.25 17.5 10.5	36		Modular construction barrier-strip term.,plug-in crossover.
SPA-1	2	250	400	600	20-20k				7.05			5.25 19 13.7	41		Signal processing subwoofer amp parametric EQ.
SPA-3	2	250	400		20-20k				7.05			5.25 19 13.7	43		Signal processing amplifier, active,balanced inputs.
750FG	2	280	450		20-20k		.01		.06	20-20k	1.5	7 17.3 12.9	55		Low feedback design, available in a studio version.
GTA	2	350	600	900	20-20k		.03			20-20k	1.48	7 17.5 14.5	72		Grand touring amp, solid state DC speaker protection.
GTB	2	275	400	800	20-20k		.03		.1	20-20k	1.48	5.25 19 13	50		Accepts 2 BGW crossover cards for bi- or tri-amping.
8500T	2	300	450	850	20-20k		.05			20-20k	1.6	5.25 19 13.8	50		Cost-effective version of GTB designed for fixed installation.
BRYSTON (BRYSTON VERMONT, LTD)															
2-B	2	50	100		20-50k	.01	.01	.01	.01	20-50k	0.75	1.75 19 10	18	$725.00	Modular design, gold plated contacts, bridgeable to mono.
3-B	2	100	200		20-50k	.01	.01	.01	.01	20-50k	1	5.25 19 9	35	$1175.00	Modular design, gold plated contacts, bridgeable to mono.
4-B	2	250	400		20-50k	.01	.01	.01	.01	20 50k	1	5.25 19 10.5	41	$1775.00	Modular design, gold plated contacts, bridgeable to mono.
6-B	2	500	800	500	20-50k	.01	.01	.01	.01	20-50k	0.75	5.25 19 13.5	50	$1895.00	Modular design, gold plated contacts, bridgeable to mono.
270	2	50	100		20-50k	.01	.01	.01	.01	20 50k	0.75	3.65 19 10	20	$500.00	Modular design, gold plated contacts, bridgeable to mono, 70V unit.
370	2	100			20-50k	.01	.01	.01	.01	20-50k	0.75	5.25 19 9	35	$695.00	Modular design, gold plated contacts, bridgeable to mono, 70V unit.
2-B	2	250			20-50k	.01	.01	.01	.01	20-50k	0.75	5.25 19 13.5	50	$1050.00	Modular design, gold plated contacts, bridgeable to mono, 70V unit.

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
CARVER CORPORATION															
M120	2	45	60	90	20- 20k		.1	.5	.5	20- 20k 0.5	1.5	1.75 19 13.5	13	$499.00	XLR-barrier strip inputs, headphone jack,clip indicators.
M300	2	110	150	180	20- 20k		.1	.5	.5	20- 20k 0.5	1.5	1.75 19 13.5	13	$499.00	XLR-barrier strip inputs, headphone jack,clip indicators.
M600	2	200	300		20- 20k		.1	.5	.5	20- 20k	1.5	3.5 19	13	$499.00	XLR-barrier strip inputs, headphone jack,led power indicators.
M900	2	350	450		20- 20k		.1	.5	.5	20- 20k	1.5	1.75 19 11.56	13	$649.00	XLR-barrier strip inputs, headphone jack,led power indicators.
M1200	2	450	600		20- 20k		.1	.5	.5	20- 20k 0.1	1.5	3.5 19 10.83	21	$1249.00	XLR-barrier strip inputs, headphone jack,led power indicators,cooling fan.
M1250	2	350	450		20- 20k		.1	.5	.5	20- 20k 0.1	1.5	3.5 19 10.75	11	$1450.00	XLR-barrier strip inputs, headphone jack,led power indicators,cooling fan.
CARVIN CORPORATION															
FET 301	2	100	200	300	5- 80k			.005	.1	20- 20k .05	1	5.25 19 12	26	$379.00	MOSFET technology, speaker protion, XLR AND ¼-in. connectors.
FET 400	2	100	200		5- 80k			.006	.1	20- 20k	1	5.25 19	29	499.00	MOSFET technology, speaker protion,bridgeable,clip LEDs .0510
FET 900	2	200	300	450	5- 80k			.006	.1	20- 20k .05	1	5.25 19 10	35	$669.00	Same features as the FET 400, yet has 900 watts total output.
CREST AUDIO															
4801	2	400	575	750	20- 20k	.015	.015	.025	.025	20- 20k	1.1	3.5 19 15	48	$1589.00	Front panel lights,active signal clip limit,load relays.
6001	2	525	700	880	20- 20k	.015	.015	.025	.025	20- 20k	1.2	3.5 19 15	51.5	$2189.00	As Above.
CC151	2	200	325	650	20- 20k	.015	.015	.025	.025	20- 20k	.187	5.25 19 14	46	$1189.00	As above, can feed 70V.
CC301	2	360	540		20- 20k	.015	.015	.025	.025	20- 20k	1.2	5.25 19 13	50.5	$1989.00	As above, also available as 70V vesion for $1389.00.
7001	2	560	810	850	20- 20k	.01	.01	.05	.05	20- 20k	1.4	3.5 19 15	49.5	$2589.00	Front panel lights, active signal clip limit,load relays.
8001	2	750	1225	1400	20- 20k	.015	.015	.05	.05	20- 20k	1.75	5.5 19 13	80	$3189.00	As above,can feed 70V.
FA-901	2	280	375	500	20- 20k	.015	.015	.025	.025	20- 20k	.775	3.5 19 13	33	$879.00	Front panel lights,signal clip limit dc-thermal protection:clip limit,load relays,can feed 70V line.
FA-2401	2	350	600	770	20- 20k	.015	.015	.025	.025	20- 20k	.775	3.5 19 13	55	$1769.00	As FA901 above.
CROWN INTERNATIONAL —See our ad on page 3															
Macro Tech 600	2	235	325	340	35k	.05	.05	.001	.05	20- 20k 0.1	.77	3.5 19 18	39.6	$1295.00	Chassis ground-lift switch,discrete power supplies,mono bridging, two input sensitivity settings.
Macro Tech 2400	2	525	800	1200	20- 20k	.05	.05	.001	.05	20- 20k 0.1	.77	3.5 19 16	67.8	$1995.00	As above.
Macro Tech 1200	2	320	465	600	20- 20k	.05	.05	.001	.05	20- 20k 0.1	.77	3.5 19 16	44	$1295.00	As above.
Micro Tech 600	2	235	340	410	20- 20k	.05	.05	.05	.05	20- 20k 0.1	.77	3.5 19 16	37	$995.00	As above.
PB2	2	320	400		20- 20k	.05	.05	.05	.05	20- 20k 0.1	.77	3.5 19 16	32	$1045.00	Full protection circuitry,bridge and parallel mono,forced air cooling,dual input sensitivity.
PS-200	2	100	170		DC- 35k	.01	.001	.05		DC- 20k 0.1		1.3 19 10.2	35	$819.00	Ultra low distortion,chassis ground lift jumper,passive cooling,proof-of-performance indicators.
PS-400	2	320	400		DC- 35k	.01	.01	.001	.05	0- 20k 0.1	1.76	7 19 10.2	54	$1259.00	As PS200 above.
CT-1600	2	540	875		20- 20k	.05	.05	.05	.05	20- 20k 0.1	.77	7 19 16	61	$1880.00	For sound contractors,8-ohm or 70V constant voltage lines,many Micro Tech features.
ELECTRO-VOICE, INC.															
AP2600	2	200	300		7- 85k		.03		.05		.77	5.25 19 12.75	39	$999.00	Available with precision stepped attenuators as model AP2600SA.
AP2300	2	100	190		7- 85k		.03		.05		.77	5.25 19 12.75	39	$641.65	Same as above.

Model	Number of Channels	Cont.power/channel at 8 ohms,all channels driven	Cont.power/channel at 4 ohms,all channels driven	Cont.power/channel at 2 ohms,all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
HILL AUDIO —See our ad on page 20															
LC400	2	120	200		20- 20k				.01	20- 20k -.5	1	20 19 14.25	17	$1249.00	XLR,binding post, ¼-in.inputs, 5-way central logic protection.
LC800	2	250	400		20- 20k				.01	20- 20k -.5	1	24 19 14.25	30	$1399.00	As above
LC1200	2	350	600		20- 20k				.01	20- 20k -.5	1	24 19 14.25	35	$1749.00	As Above
ML200	1	120	200		20- 20k				.01	20- 20k -.5	1	24 14.25 19 24	28	$1249.00	As Above, 100V line out,alsoML400 model,250W,33 lbs, $1399.00.
DX1000	2	375	600		20- 20k				.02	20- 20k -.5	1.55	3 19 12	38	$2199.00	XLR,¼-in. phone inputs,binding post outputs, LED output indicators.
DX1000A	2	500	800		20- 20k				.02	20- 20k -.5	1.55	3 19 12	40	$2499.00	XLR,¼-in. phone inputs,binding post outputs, LED output indicators.
DX2000	2	375	600		20- 20k				.02	20- 20k -.5	1.55	3 19 18	75	$2799.00	XLR,¼-in. phone inputs,binding post outputs, LED output indicators.
DX3000	2	375	600		20- 20k				.02	20- 20k -.5	1.55	3 19 18	80	$3099.00	XLR,¼-in. phone inputs,binding post outputs, LED output indicators.
INDUSTRIAL RESEARCH PRODUCTS, INC.															
DH4020	2	100	140		20- 20k			.02	.1	20- 20k -1	1	1.75 19 14	13.5	$1042.00	100 kHz switching power supply, MOSFET high freq. output.
INDUSTRIAL STRENGTH INDUSTRIES															
PA700	2	230	350		10- 30k			.1	.2	20 35k -1	1.23	5.25 19 13.38	34	$699.00	Built-in electronic X-over, balanced XLR inputs, mono bridging, protection, automatic fan cooling.
INNOVATIVE ELECTRONIC DESIGNS, INC.															
6208	1	200			20- 20k			.5	.5	20- 20k	.9	6.1 2.1 12.7	2.4	$783.00	Class D (switching mode) design card type, greater than 80% efficiency at full output.
6270	1				20- 20k			.5	.5	20- 20k	.9	6.1 2.1 12.7	2.38	$783.00	200 watts into 25 ohms or 70.7v, all specs at 25 ohm load. Both amps have balanced outputs.
JBL PROFESSIONAL															
6210	1	40			20- 20k -1	.1	.1	.1	.1	20- 20k		8 8.5 2.75	6.5	$310.00	Converts 4400 series or any other 8-ohm monitor into self-contained power system.
6211	1	40			20- 20k -1	.1	.1	.1	.1	20- 20k		8 8.5 2.75	6.5	$345.00	Same as above and also includes active balanced inputs, but with mic/line selector switch.
6215	2	35	45		20- 20k -1	.1	.1	1	1	20- 20k	1.1	1.75 19 9	10.5	$685.00	Active balanced bridging input circuitry, rear-panel switch for bridge/dual mono or stereo.
6230	2	75	150		20- 20k -1	.1	.1	1	1	20- 20k	1.1	5.25 19 11	26.3	$730.00	Same features as above.
6260	2	150	300		20- 20k -1	.1	.1	1	1	20- 20k	1.1	7 19 11	44.5	$1045.00	Same features as above.
6290	2	300	600		20- 20k -1	.1	.1	1	1	20- 20k	1.1	7 19 14	63	$1570.00	Same as above yet also includes fan for cooling.
PASO SOUND PRODUCTS,INC.															
P4061										30- 18k	.25	19 12.5 5.25	20	$630.00	Rack mount, 600 ohm input,overload protection, telephone paging output.
P4121										30- 18k 3	.25	19 12.5 5.25	34	$714.00	Rack mount, 600 ohm input,overload protection, telephone paging output.
P4201										30- 18k 3	.25	19 12.5 5.25	38	$970.00	Rack mount, 600 ohm input,overload protection, telephone paging output.
T5121										30- 20k 3	.1	19 12.5 6	23	$620.00	Rack mount, 600 ohm input,overload protection, telephone paging output.
PEAVEY ELECTRONICS															
CS-1200	2	350	600		20- 40k		.01		.03	5- 60k	1.4	7 19 17.75	70	$1299.99	Plug-in electronic X-over capability, transient-free turn on/off, thermal protection.
CS-1000	2	300	500		20- 40k		.01		.03	5- 60k	1.4	5.25 19 14.13	50	$999.00	Plug-in electronic X-over capability, transient-free turn on/off, thermal protection.
CS-800	2	240	400		20- 20k		.04		.03	5- 40k	1.4	7 19 13.5	54	$799.99	Plug-in electronic X-over capability, DDT compression, fan-cooled, thermal protection.
DECA 528	2	210	250		20- 20k	0.0	0.0		.15	10- 20k	1.0	1.75 19 14	12	$749.99	Digital energy comversion amp, DDT compression, fan-cooled, MOSFET design.

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
DECA 1200	2	325	600		20-20k	0.0	0.0		.06	10-20k	1.3	3.88 19 18	37	$1399.99	Digital energy conversion amp, DDT compression, fan-cooled, MOSFET design.
DECA 724	2	225	350		20-20k	0.0	0.0		.1	10-20k	1.0	3.88 19 18	37	$999.99	Digital energy conversion amp, DDT compression, fan-cooled, MOSFET design.
M-7000	2	200	350		20-20k		.01		.03	10-30k	1.4	5.25 19 16.5	47	$749.99	Fan-cooled, DDT compression, electronic X-over capability, transient-free turn on/off.

QSC AUDIO PRODUCTS, INC.

Model	Number of Channels	8 ohms	4 ohms	2 ohms	Power Bandwidth	IM at 1 watt	IM at full power	THD at 1 watt	THD at full power	Frequency Response	Sensitivity	Dimensions	Weight	Price	Features
1100	2	50	65	90	20-40k		.01	.01	.01	20-20k –1	.83	1.75 19 7	12	$568.00	Gain control, headphone jacks.
1200	2	50	70	90	20-40k		.01	.01	.01	20-20k –1	.83	1.75 19 7	12	$598.00	Rear gain controls, XLR and ¼-in. jacks, octal sockets.
1400	2	200	300		20-40k		.01	.01	.01	20-20k –1	1	5.25 19 9.5	34	$798.00	As 1200 above.
1700	2	325	500		20-40k		.01	.01	.01	20-20k –1	1	7 19 10.8	57	$1248.00	As 1200 above
MX1500	2	330	500		20-40k		.01	.01	.01	20-20k –1	1	3.5 19 17.9	47	$1098.00	
MX2000	2	375	625		20-40k		.01	.01	.01	20-20k –1	1	5.25 19 15.9	75	$1498.00	
3500	2	300	450		20-40k		.01	.01	.01	20-20k –1	1	3.5 19 15.9	50	$1488.00	Front removeable channel modules, detented gain controls, dual mono construction.
3800	2	375	600		20-40k		.01	.01	.01	20-20k –1	1	5.25 19 15.9	75	$1958.00	As 3500 above.

RAMSA (PANASONIC)

Model	Number of Channels	8 ohms	4 ohms	2 ohms	Power Bandwidth	IM at 1 watt	IM at full power	THD at 1 watt	THD at full power	Frequency Response	Sensitivity	Dimensions	Weight	Price	Features
WP9440	2	350			10-60k	.06	.06	.06	.06	20-20k –.5	1.23	5.25 19 18.88	75	$2190.00	Intelligent VI limiting, soft overload characteristics.
WP9220	2	200	300		10-85k	.06	.06	.06	.06	20-20k –.5	1.23	5.25 18.88 15.06	38.6	$1090.00	Has ability to drive high phase-angle, loads with ease.
WP9110	2	100	150		10-85k	.06	.06	.06	.06	20-20k –.5	1.23	3.5 18.88 15.06	28.6	$840.00	Detented input attenuators with removable knobs.
WP9055	2	50			10-85k	.05	.05	.05	.05	20-20k –.5	1.23	1.75 18.88 13.13	19	$590.00	Signal, peak and protect LEDs, XLR and phone inputs.

RANE CORPORATION

Model	Number of Channels	8 ohms	4 ohms	2 ohms	Power Bandwidth	IM at 1 watt	IM at full power	THD at 1 watt	THD at full power	Frequency Response	Sensitivity	Dimensions	Weight	Price	Features
MA6	6	100	150		20-20k	.2			.2	5-80k	.775	5.25 19 11	44	$1349.00	Built-in limiters, auto bridging, 2-speed fan.

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

Model	Number of Channels	8 ohms	4 ohms	2 ohms	Power Bandwidth	IM at 1 watt	IM at full power	THD at 1 watt	THD at full power	Frequency Response	Sensitivity	Dimensions	Weight	Price	Features
210	1	6	10		100-15k			1	3	100-15k	40mV	2.75 9.5 5.63	2.13	$125.00	Balanced mic input, unbalanced line in-put, ext. 12V power.

SOUNDCRAFTSMEN—See our ad on page 13

Model	Number of Channels	8 ohms	4 ohms	2 ohms	Power Bandwidth	IM at 1 watt	IM at full power	THD at 1 watt	THD at full power	Frequency Response	Sensitivity	Dimensions	Weight	Price	Features
300X4	2,3 or 4	600 205	900 300	450	20-20k	.05	.05	.008	.05	20-20k	1.0	5.25 19 14	60	$1299.00	Multi-channel MOSFET, 2, 3 or 4 channel mode Indicators, front panel-mounted circuit breakers.
PM860	2	210	315	450	20-20k	.05	.05	.008	.05	20-20k	1.2	5 8.5 14	20	$599.00	Has high current design to allow stability with 2 ohm loads.
450X2	2	210	315	450	20-20k	.05	.05	.008	.05	20-20k	1.2	5.25 19 11.75	28	$849.00	High current MOSFET amp with balanced or unbalanced inputs.
900X2	2	375	675	900	20-20k	.05	.05	.008	.05	20-20k	1.22	5.25 19 16.5	59	$1599.00	Same as above.
RA7501	2	275	420	320	20-20k	.05	.05	.05	.05	20-20k	1.21	7 19 15	47	$949.00	Class H signal tracking design for maximum efficiency.

SPECTRA SONICS

Model	Number of Channels	8 ohms	4 ohms	2 ohms	Power Bandwidth	IM at 1 watt	IM at full power	THD at 1 watt	THD at full power	Frequency Response	Sensitivity	Dimensions	Weight	Price	Features
701	1-inf	33	58	86	20-20k	.05	.075	.025	.025	0-20k	+5 dBv	2.5 10 1.88	.88	$108.00	A modular amp suited for bi, tri, multi-way, used in noise masking, broadcast, recording.
701BP	1-Inf	122	172	200	20-20k	.05	.075	.025	.025	0-20k	+5 dBv	5 10 1.88	1.76	$216.00	Is two model 701s bridged together with the same qualifications.
712B	2	30	50	80	20-20k	.05	.075	.025	.025	0-20k	0 dBv	5.5 19 14.5	22	$595.00	A stereo rack-mount, self contained power amplifier.
712	2	100	100	100	20-20k	.05	.075	.025	.025	0-20k	0 dBv	5.5 19 14.5	24	$760.00	A stereo rack-mount, self contained power amplifier.

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
STUDER REVOX AMERICA, INC. —See our ad on page 2															
Revox B242	2	200	230	400	20- 20k		.01		.01	20- 20k -3	1.55	7.7 6 14.2	37.5	$3000.00	MOSFET drive and special bipolar power transistors, two power transformers, mono bridgeable.
SUNN															
SPL7350	2	225	375		10- 50k	.05	.05	.1	.1	10- 50k -1	1.23	5.25 19 15	40	$899.99	Delayed on/off, protection, soft clipping.
SPL7250	2	160	250		10- 50k	.005	.005	.1	.1	20- 20k -1	1.23	3.5 19 14	30	$699.00	As above.
SYMETRIX															
A-220	2	20	20		20- 40k				.01	20- 20k -1	.5	1.75 19 8	9	$349.00	Phone and XLR input, front panel headphone jack.
TOA ELECTRONICS															
P-300D	2	300	480		10- 20k		.05	.02	.01	20- 20k	1.23	8.75 19 18.75	77	$2218.00	Separate power supplies, stereo, mono switching, calibrated attenuator.
P-150D	2	150	220		10- 20k		.05	.02	.001	20- 20k	1.23	5.25 19 18.75	44	$1238.00	Magnetic breaker front panel switch, HPF switch, calibrated attenuator, high current headroom.
P-75M	2	75	110		10- 20k		.05	.02	.001	20- 20k	1.23	3.5 19 18.75	29	$938.00	As above.
P-300M	1	300	480		10- 20k		.05	.02	.001	20- 20k	1.23	8.75 19 18.75	62	$1458.00	70/25V transformer, front panel breaker switch. Also available: P-150M, 150W, $1118.00.
YORKVILLE SOUND, INC.															
AP3000	2	475	750	1200	20- 20k		.003	.04		20- 20k	1.4	3.5 19 15.75	45	$1599.00	MOSFET drivers, also available AP1200 with 625W/C@2 ohms—$1199.00; AP500 with 250W /C@2 ohms—$799.00. All are bridgeable.
YAMAHA CORPORATION —See our ad on page 10															
PD2700	2	350	500		10- 50k 1		.03		.05	10- 50k	1.23	5.25 18.89 17.25			Bridgeable mono, forced air cooling.
P2250C	2	170	250		10- 50k 1		.01		.05	10- 50k	1.23	5.25 18.88 16.56	41.8	$895.00	XLR and ¼-in. inputs, barrier strip output, forced air cooling.
P2350	2	175	250		10- 50k 1		.03		.05	10- 50k	1.23	5.25 18.87 11.25			XLR and ¼-in. inputs, barrier strip and ¼-in. outputs, forced air cooling.
P2150C	2	100	150		10- 50k		.01		.007	10- 50k 1	1.23	5.25 18.88 16.63	37.4	$695.00	XLR and ¼-in. inputs, binding post and 1/4\in. output jacks, forced air cooling.
P2075C	2	50	75		10- 50k		.05		.003	10- 50k -1	1.23	3.88 18.88 14.38	19.8	$495.00	XLR and ¼-in. jacks, binding post and ¼-in. output jacks, compact and light-weight.
PC1602	2	160	240				.01		.015	10- 50k 1	1.23	5.5 18.88 17	47.8	$995.00	Comprehensive protection circuitry, XLR in and through connectors, 480 watts in mono.
PC2602	2	260	400				.007		.015	10- 50k 1	1.23	7.25 18.88 17	57	$1295.00	Protection circuitry, XLR in and through connectors, model PC2602M has 26-segment backlit LCD meters

ADDRESSES

Altec Lansing Corporation
10500 West Reno Avenue
Oklahoma City, OK 73126-0105

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

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

BGW Systems
13130 S. Yukon Avenue
Hawthorne, CA 90250

Bryston Vermont, Ltd.
979 Franklin Lane
Maple Glen, PA 19002

Carver Corporation
20121 48th Avenue West
Lynnwood WA 98046

Carvin Corporation
1155 Industrial Avenue
Escondido, CA 92025

Crest Audio, Inc.
150 Florence Avenue
Hawthorne, NJ 07506

Crown International
1718 Mishawaka Road West
Elkhart, IN 46517-1000

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

Hill Audio
5002B N. Royal Atlanta Dr.
Tucker, GA 30084

Industrial Research Products
411 Busse Road
Elk Grove Village, IL 60007

Industrial Strength Industries
15431 Blackburn Avenue
Norwalk, CA 90650

Innovative Electronic Designs
9701 Taylorsville Road
Louisville, KY 40224

JBL Professional
8500 Balboa Blvd.
Northridge, CA 91329

Paso Sound Products, Inc.
14 First Street
Pelham, NY 10803-1401

Peavey Electronics
711 A Street
Meridian, MS 39301

QSC Audio Products
1926 Placentia Avenue
Costa Mesa, CA 92627

Ramsa (Panasonic)
6550 Katella Avenue
Cypress, CA 90630

Rane Corporation
10802 47th Avenue W
Everett, WA 98204-3400

Soundcraftsmen
2200 South Ritchey
Santa Ana, CA 92705

Spectra Sonics
3750 Airport Road
Ogden, UT 84405

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

Sunn
1130 Columbia Street
Brea, CA 92621

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

TOA Electronics, Inc.
601 Gateway Blvd.
S. San Francisco, CA 94080

Yamaha Corp. of America
P.O. Box 6600
Buena Park, CA 90622-6022

Yorkville Sound, Inc.
4600 Witmer Industrial Estate, Unit #1
Niagara Falls, NY 14305

DAN BUXBAUM

1990 Winter NAMM Round-Up

This year's winter NAMM show has certainly been an impressive one. A wide variety of products were displayed and introduced, and even the odd fire-breathing monster made an appearance (the giant, green plastic type, courtesy of Ibanez). The following is a compendium of various manufacturers with products designed for the professional audio environment. This is a non-comprehensive summary, and my sincerest apologies to anyone left out due to space, and my own, limitations. Also, manufacturers who did not display new products have been omitted. Now, with all the technicalities out of the way, let's begin our alphabetical listing:

APHEX SYSTEMS

Aphex introduced their new processor, the EXPRESSOR compressor/limiter. This single channel unit employs standard features such as: adjustable input, threshold, attack, release, output, and ratio, as well as hard or soft knee compression, link, and slave. The EXPRESSOR's special features include adjustable High Frequency Expansion to counteract the dulling effect of high compression ratios, and the Spectral Phase Refractor (SPR) which was first introduced on the Aural Exciter Type III. SPR corrects the bass delay anomaly associated with the recording and reproduction processes, to restore clarity and openness, and significantly increases the apparent bass energy level without adding any amplitude equalization or "bass boost."

APPLIED RESEARCH & TECHNOLOGY

ART unveiled a whole new family of impressive signal processors and delay systems. Leading the way is the SGE MACH II processor which offers twelve *simultaneous* effects and 20 kHz bandwidth. The MACH II has over **seventy** different effects including: exciter, EQ, compressor, limiter, noise gate, expander, sampler, envelope filter, pitch transposer, line EQ, stereo panner, stereo chorus and flange, twelve distortions, twenty-one delay types (two full seconds), and twenty-four different reverb algorithms. It also features real-time MIDI and two-hundred memories.

The DR-X is a studio digital reverberator/dynamics processor/pitch transposer/sampler which offers 160K bytes of audio ram, bandwidth to 20 kHz, sampling, ten simultaneous audio functions, an exciter, EQ, compressor, etc. It also features twenty-one different delays (two full seconds), twenty-four reverb algorithms, two-hundred memories, and comprehensive real-time MIDI control.

The MultiVerb III signal processor is four-hundred-percent more powerful than its predecessors, and features a new series of algorithms with four-times the previous resolution. The MultiVerb III has over fifty-three effects (up to four simultaneously) including: sampling, twenty-one delay types, stereo panning, pitch transposing, twenty-four reverbs, **two-hundred** memory locations, and full programmability and performance MIDI compatibility. The MultiVerb LT offers the sounds and performance of the MultiVerb III, but at a very competitive price ($299.00). The LT has the simplicity of one-touch control commanding the **one-hundred-and-ninety-two** studio multi-effect combinations. This unit is also MIDI addressable.

ART's new delay systems, Delay System V and Delay System VII, feature new circuit designs with full 16 bit ADA converters and offer 20 kHz bandwidth. Both units feature rotary type pots or encoders allowing the user to manually "tweak" delay, width, chorus, and flange settings. Because of the extreme range of control, chorus settings deeper and richer than most digital units allow, can be achieved. The Delay System V is a non-programmable delay/super choruser that features controls for: range, bypass, feedback invert, and infinite repeat along with rotary pots for delay, speed, depth, and width. The Delay System VII uses the same audio current, but is fully programmable. The mix is also programmable and all parameters are visible on L.E.D. readout. The Delay System VII is also a sampler allowing the user to capture samples, truncate them and vary their pitch, and may be triggered manually or automatically.

AUDIO-TECHNICA U.S., INC.

Audio-Technica has also added to their popular line of mics. Added to the versatile AT871 UniPlate microphone is a new, smaller version (AT851A Micro UniPlate), a phantom-powered design (AT871R), and a new omnidirectional model (AT841A OmniPlate). The size and versatility of these mics enable them to be used effectively in a wide range of applications, especially in situations that demand surface mounting and minimum visibility. Audio-Technica also introduced their PRO22 microphone which has been designed for close-up applications, and their new 600 Series line of performance headphones.

COMMUNITY LIGHT AND SOUND INC.

Community Professional Sound Systems previewed two exciting new sound reproduction systems designed exclusively for the demands of the cinema environment. The TheatreStar III is the first three-way loudspeaker design of its type built exclusively for behind-the-screen cinema applications. The TheatreStar has several unique features including the use of a Community M200 loudspeaker for midrange frequencies between 400 Hz and 3500 Hz for extremely clear dialog intelligibility. An exclusive trapezoidal-pattern high frequency horn provides uniform sound distribution throughout the viewing audience. The TheatreStar III utilizes its very precise Wavefront Coherent design which allows the full frequency spectrum to arrive at the listener's ears at exactly the same moment in time, providing remarkable definition and clarity in its sound reproduction.

The SurroundStar II is a compact, unobtrusive two-way loudspeaker system designed to provide uniform, realistic sound quality from the surround channels of a cinema sound system. The SurroundStar II features a powerful eight-inch low frequency/midrange speaker crossed over at 2000 Hz into a unique trapezoidal-pattern high frequency horn. The SurroundStar II is only 6-and-1/4-inches deep, and is wrapped in black acoustically-transparent cloth, blending easily and almost invisibly into most theater designs.

The RS880, VBS415, and 880EQ are the latest developments in Community's RS Series of Wavefront Coherent loudspeaker systems, and are designed to operate as a complete system. The 880EQ is the controller: it provides a crossover between the RS880 full-range system and the VBS415 subwoofer. The RS880 is a three-way trapezoidal, arrayable loudspeaker system designed for professional applications. The RS880 and the VBS415 enclosures are fitted with handles, and optionally, with three or six "D"-ring flying points.

The new M4 CoAx bracket system enables any of Community's PC400 Series high-frequency horns to be coaxially mounted directly in the center of a matching M4/PC/500 Series horn, thereby improving polar response and adjacent horn performance. The M4 CoAx bracket system is available bolted into a PC1500 horn for new installations, or as a bracket and hardware package for retrofitting existing installations of current modal PC1500 horns.

DOD ELECTRONICS

DOD introduced their new series of mixers, the 820 and 1220 stereo mixers. These mixers are compact, yet maintain studio-quality sound reproduction. Features of these mixers include: high and low impedance inputs, 15 dB cut/boost EQ'ing, table or rackmount configurations, phantom power, RCA-type stereo tape outputs, auxiliary inputs, individual monitor sends, effects send and mono/stereo returns, 60 mm dust-shielded faders, and rubberized knobs. The 820 series features eight stereo channels, while the 1220 series has twelve stereo channels. There are six different models of these mixers, and the various configurations include the 820 and 1220 with line-only applications, and the 820 XL and 1220 XL with high/low impedance in table-top formats, to the 820 RM and 1220 RM with full-balanced and un-balanced input capability in rackmount configurations. The prices of these mixers are quite affordable, and range from $399.95 to $699.95.

ELECTRO-VOICE

Electro-Voice introduced their new N/DYM Series II line of dynamic mics. EV is employing a new vibration-isolation system, *DynaDamp*, on these mics to reduce handling noise, cable-transmission noise, and clicks and thumps. The soft but firm feel, and black matte finish of their *Warm-Grip* sleeves make these new mics distinctive. The trademarked *DynaDamp* and *Warm-Grip* (better absorption and increased protection) contribute to an improved N/DYM series of dynamic mics.

We also got a chance to hear EV's new MT-4 Manifold Technology concert system firsthand, at a concert featuring Giant and Extreme, two rock bands. The MT-4 system was used for the mains, and the sound at the show was both clear and "punchy." The new MTS-1 stage speaker system, derived from the MT-4 Manifold Technology concert system was also introduced. This unit is a compact, manifolded, two-way, biamped speaker system and is recommended for use in situations where space is limited and efficiency is a major factor.

FOSTEX CORPORATION OF AMERICA

Fostex introduced their new Model 280 Multitracker, a four-track recorder with impressive programmable memory functions. The three programmable memory cue points are the zero reference, memory 1 and memory 2; and each one can be set, confirmed, and changed easily. This unit can automatically locate the "zero" mark and the memory 1 and 2 cue points—a real time-saver and headache-preventer. The Model 280 can also automatically return to the memory 1 cue point upon reaching the memory 2 cue point. These automatic operations really allow the user to concentrate on being creative—the main objective, anyway.

The Models 454 and 812 (four-bus and eight-bus, respectively) are Fostex's new recording mixers. The Model 454 is affordable (under $1,000.00), and some of its many features include in-line monitoring, dual parametric equalizers for low end and midrange, and output solo. The two effects sends are: mono, post EQ and fader, and stereo, with a three-position selector—pre EQ and fader, post EQ and fader, or tape (playback). The Model 812 features dual parametric equalizers for the low end (60 Hz to 1 kHz) and midrange (400 Hz to 6 kHz). The high end has a shelving-type equalizer fixed at 10 k/Hz. This unit also has two mono effects sends and a stereo aux send. The stereo aux send bus has both pan and gain controls and is switchable from the post-fader signal to the tape playback signal. An important function of this bus is its ability to monitor a "wet" input signal while recording "dry." This is an outstanding multi-track mixer that is also fully MIDI-compatible.

SENNHEISER ELECTRONIC CORPORATION

Sennheiser introduced a variety of new microphones, new stereo headphones, and even a new mic stand. The

MKH 50-P48 is a new member of the symmetrical transducer, tranformerless MKH Series, designed to meet the demands of digital recording. This condenser mic exhibits a highly frequency-independent supercardiod polar pattern for attenuating sound pick-up from the sides and rear of the mic. The MKH 50-P48's low noise floor, low distortion and high sensitivity combines for a dull dynamic range.

The MKH 70-P48 RF condenser super-cardiod/lobe is a new, long shotgun mic. The mic is designed for long distance sound pick-up, and is lightweight, yet very reliable as the RF condenser principle is virtually immune to humidity and moisture. A new symmetrical pull-pull transducer design significantly reduces both intermodulation and harmonic distortion products. The new MD518 dynamic mic is designed for sound reinforcement of high sound pressure signals including vocal, guitar, and percussion mic'ing. A smooth cardioid pickup pattern insures maximum rejection of unwanted sounds from the rear of the MD518 and allows for use in close proximity to stage monitors and sidefill cabinets. The MD518 is highly insensitive to the strong magnetic effects of loudspeakers; therefore, it can be utilized with confidence near large stage speakers.

Two new headphones introduced by Sennheiser are the HD 450 Studio Stereo Headphone and the HD 25 Studio Monitoring Headphone. The HD 450 is a supra-aural Open-Aire model that is a durable, acoustically accurate, high impedance headphone. The HD 450 Studio is designed to hold up to the rigors of studio or field use, and comes with a canvas carrying case. The HD 25 utilizes dynamic drivers in a closed supraural design to offer a lightweight and comfortable professional headphone. Also, if you wish to cue with single muff monitoring, the HD 25 allows one driver to rotate off the ear and onto the user's temple.

Additionally, Sennheiser's new mic stand (the SEMS series) has some new features that gear it towards the studio or for the touring life. The boom joint does not contain any support hardware that might break in shipping. Reinforced knobs are used that can withstand the rigors of touring. Also, plastic friction washers are used to insure that the boom will always hold its position.

SHURE BROTHERS INC.

Shure Brothers introduced a new addition to their illustrious L Series line of wireless microphone products—the L2 Handheld Transmitter. The L2 comes in three different versions that are all compatible: the Model L2/58 features the famous SM58 dynamic microphone element, the Model L2/96 incorporates the condenser element used in the high-performance SM96 vocal condenser mic, and the L2/Beta 58 featuring the acclaimed Beta 58 element. The mic transmitter heads are interchangeable, so any of these three elements can be used with the same L2 transmitter. These various heads can be changed in seconds—making the L2 Handheld Transmitter extremely practicable and useful in many different situations.

SONY CORPORATION OF AMERICA

Sony Professional Audio introduced new additions to their already prestigious wireless microphone lines, both in UHF and VHF versions. The WRT-67 Dynamic UHF wireless mic, using a uni-directional dynamic capsule, offers warm response on the low end and the "punch" required for vocals. This wireless mic operates on batteries (a single "AA" battery provides for over three hours of continuous operation) or optional A.C. power supply.

The newest development in Sony's UHF wireless microphone technology is the WRR/WRT-28 System. The WRT-28 (the transmitter) operates on a carrier frequency from 947 to 952 MHz, with RF output power of 30 mW, and frequency stability of ±0.005-percent. The WRR-28 (wireless mic) features a balanced microphone level output and a water-resistant case. This unit is a marvel in that Sony has incorporated such a high-quality transmitter into such a compact and lightweight system. The performance is high-end quality, and the smallness of the unit allows for easy concealment—ideal for live performances.

The VHF 400 Series wireless mic system incorporates miniaturized frequency synthesis components which eliminate the need for a larger equipment inventory to obtain reliable wireless operation. An amazing feature of this system is that it is capable of operating on any of 48 separate frequencies—the system's receivers are equipped to handle all 168 channels in the assigned band. The major components of the VHF 400 System are: the WRT-410, a one-piece unit combining the transmitter and a hand-held dynamic mic, and the WRT-420, a bodypack transmitter supplied with an ECM-77B lavalier-type mic.

Additional products shown were the MU-R201 stereo digital reverb (shown for the first time), and the TCD-D10 PRO portable DAT recorder coupled with the ECM-MS5 M-S stereo mic. The MU-R201 employs 16-bit linear quantization and a 26 kHz sampling frequency. This stereo digital reverb has ten basic algorithms including: reverb, gated reverb, delay, reversed reverb, auto pan, and various combinations of functions which help the user to save time and control costs. An extremely useful feature is that this unit's algorithms are defined by 26 parameters which can edit a preset sound output—this enables you to tailor the preset sound to your own specific tastes and needs. This makes the MU-R201 ideal for recording studios, post-production suites, live performers, broadcasters, and sound reinforcement agencies. The MU-R201 also includes a MIDI interface for effective control of any parameter, memory number, or effect level in a MIDI-based system.

The TCD-D10 PRO portable DAT recorder is designed for both studio and field use. Among its many features are: S/P DIF and AES/EBU-type digital input/output ports, 2x oversampled digital A/D and D/A converters, and a rechargeable battery—providing up to 1.5 hours of continuous operation on a single charge. The ECM-MS5 stereo mic is also well-suited for indoor and outdoor applications. Weighing in at less than eight ounces, and measuring just 8.75 inches in length, the ECM-MS5 is extremely comfortable to use, yet its matched uni-directional electret condenser capsules assure a phase-coherent image, while a low-cut switch enables users to roll-off low frequencies to achieve optimum recording by reducing unwanted wind noise or vibrations. This mic also comes supplied with cable and urethane windscreen.

SOUNDCRAFTSMEN

Soundcraftsmen introduced two new professional power amplifiers, the Model DJ600 and the Model DJ900. Both units were designed specifically for the Disc Jockey and Club and Commercial Installation Market, where high power as well as superb sound reproduction is required. The DJ600 Power Amplifier combines all of the best features of Mosfet amplification and multi-impedance, high current design to provide great sound. The new DJ600 *multi-impedance* amplifier is designed specifically to provide not only the high current necessary to drive low impedances, but also to provide unmatched musicality in driving loudspeakers of *any* impedance, from as low as 2 ohms to as high as 32 ohms.

The DJ900 Power Mosfet Amplifier combines the best features of Class A, Class B, and Soundcraftsmen's proprietary and unique Phase Control Regulation circuitry to also provide great sound. The new ultra-high-current design of the DJ900 allows this amplifier to perform effortlessly under the most demanding conditions—the DJ900 can handle impedance as low as 2 ohms! The DJ900 Power Amplifier's Mosfet output stages offers the utmost audio clarity and distortion-free sound reproduction.

TASCAM

Tascam, the professional division of TEAC, unveiled an impressive array of new items ranging from DAT Recorders and mixdown decks, to high-speed, four-track recorders and thirty-two-channel consoles. The Porta Two HS is a high-speed version of Tascam's popular Porta Two four-track machine. This new recorder utilizes 3 3⁄4 in./sec. tape speed which provides users with a greater sonic quality to their cassettes.

Especially impressive are the two new MIDI-compatible mixer/recorders: the 644 MidiStudio (four-track version) and the 688 MidiStudio (eight-track version). Both these units provide excellent versatility and sound in all small studio recording and mixing applications, while also incorporating built-in MIDI-to-tape synchronizers which allow MIDI tracks to chase-lock to each unit's internal multi-track recorder. The flexibility and usefulness of these products, all in a self-contained, lightweight unit, is quite impressive. Tascam also introduced a balanced version of their quality unbalanced 112 model mixdown deck, the four-track, two-channel 112B.

The new DA-30 DAT recorder is an affordable ($1,899.00), stereo deck which features 64-times oversampling and Delta-Sigma modulation in the analog-to-digital converters, while the digital-to-analog converters feature 18-bit technology with eight-times oversampling. Combined, this results in the achievement of a S/N ratio in excess of 94 dB. Finally, the M-3500/24 and M-3500/32 (twenty-four and thirty-two tracks, respectively) are new mixing consoles. Both units are performance-oriented consoles, with a wide range of features, and affordable prices. The M-3500/24 is priced at $7,499.00, while the M-3500/32 is priced at $8,499.00.

YAMAHA CORPORATION OF AMERICA

Although Yamaha specializes in its wide, quality assortment of musical instrument lines at the NAMM show (including guitars, basses, drums, guitar and bass amps, etc.), various pro audio, non-instrument products are also displayed. One such unit generating a lot of press, and certainly deservedly so, is the DMR8X—a self-contained mini studio. This amazing unit has built-in features including: an eight-track DAT recorder, a twenty-four track mixer, a SMPTE synchronizer, and various signal processors including EQ, compression, reverb, limiting, delay, etc. The DMR8X additionally has its own automated mixing system, also built-in.

The Yamaha DMR8X has the ability to take your analog recording signals, convert them to digital, and do everything other eight-tracks can do, only in a digital format. The DMR8X processes, mixes, and records, all digitally, in one package! This product accepts all the major digital and audio formats and also generates MIDI Time Code. Be prepared, however, that this amazing product does have a list price of $35,000. I wanted to end this 1990 Winter NAMM round-up with a grand finale, and the DMR8X certainly provides the fireworks in that department. db

New Products

DYNAMIC VOCAL MIC

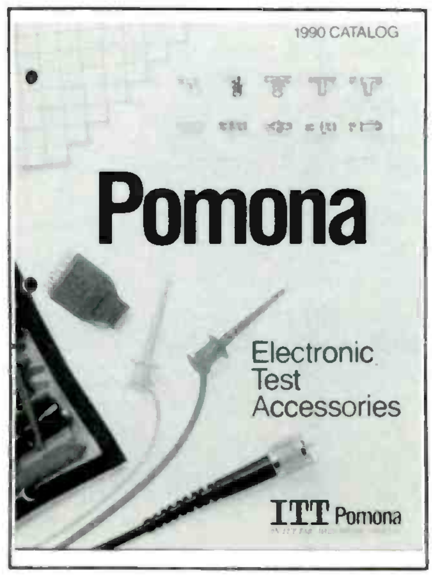
• The new N/DYM N/D857 microphone is designed specifically for concert vocal applications, where high gain-before-feedback and extended response are critical. The addition of a unique acoustical path corrector provides increased sensitivity and an extremely uniform polar pattern throughout the vocal frequency range. This gives the N/D857 a tight, supercardiod polar pattern with superb off-axis rejection and increased gain-before-feedback, allowing high monitor levels. The N/D857's switchable bass roll-off employs a second-order high-pass filter with an 80-Hz corner frequency, for a 12-dB-per-octave roll-off that further eliminates handling noise without compromising vocal sonic quality. The roll-off also provides increased flexibility with various stage situations, including the high bass output of large subwoofer arrays, low-frequency standing waves and vocal styles. Because it is designed to handle high SPL and has a tight polar pattern, the N/D857 is also suited for mic'ing amplified instruments.

Mfr.—Electro-Voice, Inc.
Price: $450.00
Circle 60 on Reader Service Card

CATALOG

• This is a totally new 138-page catalog of Electronic Test Accessories. Ten major product categories are presented via an easy-to-use index, also including the company's most popular selection of jumpers and cables, boxes, plugs and jacks, connectors, adapters, single-point test clips, and static control devices. Designed to provide a complete guide and reference to Pomona's growing family of IC Test Clips and Adapters, and Cable Assemblies and Adapters, helpful selection guides are provided for the user.

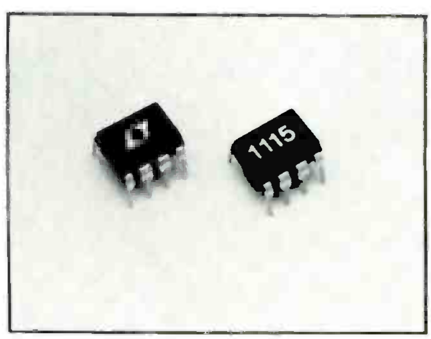
Mfr.—ITT Pomona Electronics
Price: Free
Circle 61 on Reader Service Card



AUDIO OP AMP

• The LT1115 is a new audio operational amplifier with 0.9 nanovolts per root Hertz noise at 1 kHz and less than 120 nanovolts (rms) noise over the DC to 20 kHz audio spectrum. The minimum slew rate of the LT1115 is 10 volts per microsecond and its minimum gain-bandwidth product is 40 MHz. Minimum voltage gain is 2 million. Applications for the LT1115 include high-end preamplifiers, CD audio players, digital audio tape recorders and players, hydrophones, infrared detectors, and low noise frequency synthesizers, where extremely low noise and distortion are also necessary. The LT1115 is available in 8-pin plastic dual-in-line packages and 16-pin small outline packages, and is available from stock.

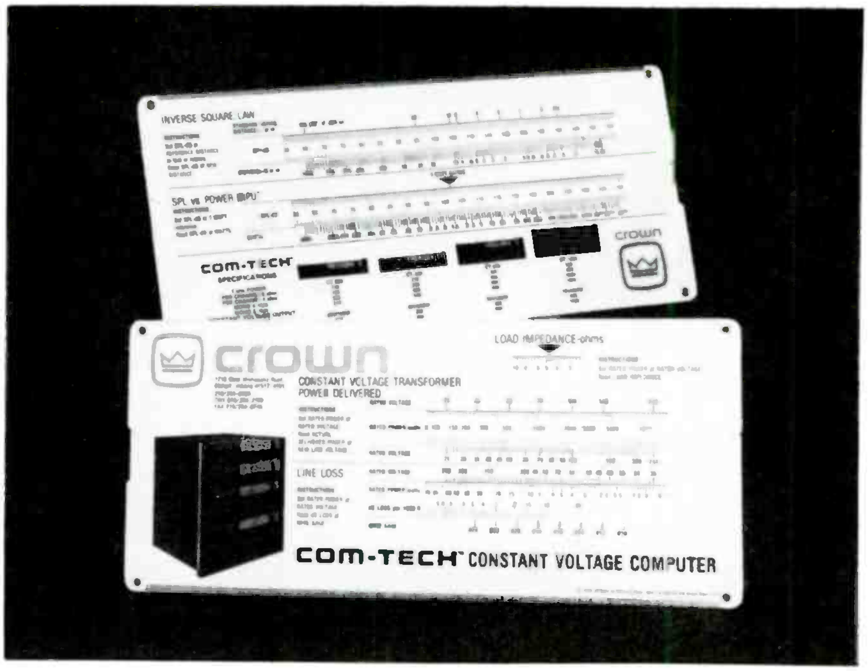
Mfr.—Linear Technology Corporation
Price: $2.95 (100-up pricing in the 8-pin package)



Circle 62 on Reader Service Card

VOLTAGE COMPUTER AND MIC SLIDE RULE

• Crown's new Constant Voltage Computer can be used to calculate the correct transformer taps to use when constant voltage lines other than 70V are desired. In addition, it also quickly calculates line loss, the effects of inverse square law, and the relationship between power input and SPL. This device is valuable because it not only tells you which transformer taps to use to achieve a desired voltage, but reveals the load impedance to the amplifier as well. It does all this merely by aligning the number of watts desired on its "Rated Power" scale with the number of volts needed on its "Rated Voltage" scale. Most importantly, the voltage computer takes the guesswork out of system engineering. With its use, the exact number of amplifiers can be determined for any job, and line losses can be figured precisely. Complete with its own easy-to-follow instructions, the Constant Voltage Computer is the perfect companion for all Com-Tech amplifiers, which have the flexibility to drive constant voltage lines at 25, 35, 50, 100, or 140V.



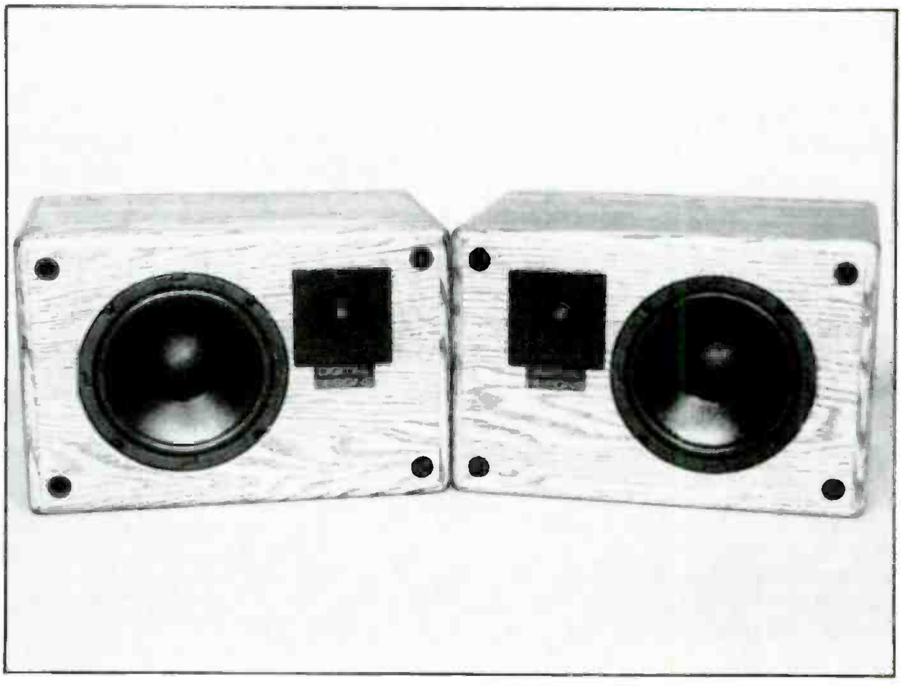
Crown's Microphone Sensitivity Slide Rule helps you to convert easily from one microphone-sensitivity specification to another. These include open-circuit sensitivity, power sensitivity, and EIA sensitivity. It also lets you calculate the mic output voltage for a given input sound pressure level. In addition, the slide rule helps you determine a microphone's impedance if you know its power sensitivity. The jacket provides useful application examples and information on microphone directional characteristics.

Mfr.—Crown International, Inc.
Price: $5.00 (for each item)
(covers postage and handling)
Circle 63 on Reader Service Card

STUDIO MONITORS

• Model LS161 studio monitors are mirror-imaged, two-way, acoustic-suspension driver systems, engineered to be compatible with today's recording technology. By mirror-imaging, these monitors eliminate the response dips which can occur in the upper harmonic structure of the human voice range. Other design features include: point source vocalization, minimum phase shift, fast transients due to low moving mass, time corrected driver placement, a self-resetting, thermo-controlled fuse on the tweeter, connections are gold-plated, solid steel, 5-way binding posts, and the cabinets are available in natural oak or black oak lacquer.

Mfr.—Digital Designs
Price: $349.00/pair
Circle 64 on Reader Service Card



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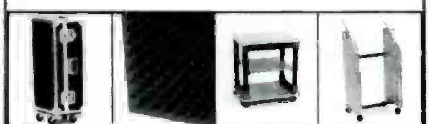

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People, Places... & Happenings

• As a non-profit organization which has traditionally attracted wide volunteer support from the professional audio community, an organizing committee of the Special Olympics is once again seeking help in providing sound reinforcement for the many sporting and entertainment events held during the upcoming **International Summer Special Olympics** slated for July 19-27, 1991 in Minneapolis, Minnesota.

Promising to be the largest Special Olympics showing to date, the summer 1991 games will attract over 6,000 special athletes and 2,500 coaches from 90 countries worldwide. Preparations are also being made for the arrival each day of over 50,000 spectators, who will witness athletic endeavors including swimming, diving, field and track, basketball, bowling, canoeing, cycling, tennis, softball, power lifting, and team handball.

Those interested in donating time, equipment, or cash contributions to this event should contact audio organizers **Beryl Moore, Joe Wisler,** or **Gil Nichols** at **Crown International**. Please call Monday through Friday during regular office hours at (219) 294-8000, or write to them in care of Crown at P.O. Box 1000, Elkhart, IN 46515.

• **Underwriters Laboratories Inc.** (UL) and **JMI Institute** have signed a joint Memorandum of Understanding that will provide quality assurance assistance to Far East and U.S. corporations interested in gaining access to the European Community (EC).

Under the new agreement, UL and JMI may accept each other's evaluation of manufacturers' quality systems for registration to the International Organization for Standardization's (ISO's) 9000 Series Standards. The EC has adopted the ISO 9000 Standards and identified them as European Norm (EN) 29000 Series Standards. These standards are expected to become the governing documents throughout Europe for evaluating manufacturers' quality systems.

The ISO 9000 Standards provide models for various quality assurance system levels, as well as general guidelines for establishing and maintaining a quality management system. The UL/JMI joint memorandum is intended to provide a mechanism for either organization (UL or JMI) to evaluate a manufacturer's quality assurance system and to register the system with UL, JMI or both UL and JMI to ISO 9000 and EN 29000 Standards.

UL is known nationally and internationally for its work in product testing, safety certification, quality assurance and standardization. JMI is known for its work in evaluating electronic products and building materials for the Japanese government.

• **Telarc International** announced the opening of a European office in Paris, France. Heading the operations there will be **Gerard Schoumann**, as Director of Sales and Marketing in Europe. Mr. Schoumann, who graduated from the Sorbonne in Paris with a degree in Law, has an extensive background in the recording industry, beginning with **Polygram**, where he was the Label Manager for **Deutsche Gramophone** in France. In 1986, he became the Marketing Manager of Classical Repertoire for **CBS Masterworks** in Europe, and in 1988, he was named Director of the Classical Department for that label in France.

• **Altec Lansing Corporation** recently hosted two sound clinics held in Europe. These clinics drew record attendance in Frankfurt, West Germany, and Zagreb, Yugoslavia. More than 200 Altec distributors and their customers, as well as consultants, electrical engineers and contractors, attended each of the two-day clinics hosted by **Altec Lansing-Europe**, a division of **Mark IV Vertriebs, GmbH.**

The international forum—a *United Nations of Sound*—saw delegates representing companies from Austria, Bulgaria, Denmark, England, Finland, Germany, Greece, Holland, Italy, Norway, Rumania, the Soviet Union, Switzerland, Turkey, the United States, and Yugoslavia. The program at the clinics was translated from English into German, Russian, and Serbo-Croatian.

Programs, seminars, demonstrations and presentations ranged from architectural acoustics to sound system design, and from understanding sound products to computers in acoustical engineering and electronic design. Altec Lansing President **Dave Merrey** was quoted as saying, "to my knowledge, the clinic in Frankfurt was the first full-scale clinic Altec ever conducted in Western Europe, and the one in Zagreb was the first sound clinic *any* major pro sound company has ever held in the eastern part of Europe. We are excited over the enthusiastic response and proud that we can add yet another chapter to our history of pioneering the achievement of better sound throughout the world."

• **Neve Electronics International** announces the appointment of **Hazel Simpson** as Director of Sales. Ms. Simpson will be responsible for sales both in the UK and via Neve's world-wide network of distributors. In related news, Neve announces the appointment of **Joe Naccarato** to the position of General Manager, **Rupert Neve Canada, Inc**. Prior to joining Neve, Mr. Naccarato was Sales Manager for **Studer Canada**. As part of this new appointment, Mr. Naccarato will be responsible for all Neve and **Mitsubishi Digital Pro Audio** sales in Canada.

Also, **Neve North America** has named **Charles Conte** to the position of Public Relations Administrator for the North American market. Mr. Conte comes to Neve from **Studer Revox America** in Nashville, where he has been Public Relations Manager since October 1987. As PR Administrator, a newly created position within Neve, Mr. Conte will be responsible for writing all Neve and **Mitsubishi Digital Pro Audio** press releases, and will work directly with the trade press to promote both product lines.

• **DOD Electronics** announces the hiring of **Richard Bos** as Marketing Manager. In his new position, Mr. Bos will assume responsibility for corporate and product marketing strategies, including new and existing product development, market research and analysis, advertising, promotions, P/R and trade shows. Mr. Bos comes to DOD with an extensive and successful background in marketing and advertising both at the corporate level and in ad agencies. He has 25 years experience as a lead guitarist for various local scene rock and roll bands and has played warm up for a number of top bill rock groups. His business and gig experience includes the U.S., Europe, East Germany, Asia, and South America. He has resided and worked in a number of foreign countries.

• **The Hollywood Association of Recording Professionals** (H.A.R.P.), the professional trade association for commercial recording studios, has elected **Terry Williams**, co-owner of **Lion Share Studios**, as the organization's 1990 President. Since its formation early last year, H.A.R.P. has brought together some of L.A.'s most respected studio owners to address traditional business issues such as group health insurance, employee benefits, and equipment purchasing, as well as trends and concerns exclusive to the recording industry. H.A.R.P. is currently engaged in an aggressive membership drive, which will be launched by direct mail to local studio owners in the coming weeks.

• **The National Association of Broadcasters' Engineering Conference Committee** has selected **Hilmer I. Swanson**, a senior staff scientist at **Harris Corporation's Broadcast Division**, Quincy, IL, to receive NAB's Engineering Achievement Award. Swanson is responsible for much of the technology used in AM transmitters today. Work by Swanson has dramatically lowered the power requirements for transmitters, saving AM radio stations an estimated $50 million in extra power costs over the years. At Harris, Swanson made significant contributions in the development of pulse modulation techniques and more recently, digital modulation for AM radio. Many of Swanson's patents represent the same standards and benchmarks used by transmitter designers worldwide.

• **AMS Industries plc** (UK) announced plans to relocate their wholly owned subsidiary, **AMS Industries Inc.**, to Northern California under the control of **Jim Stern**, President and CEO, in a move to further expand their U.S. operations. **Nigel Branwell**, who has run the operation from Seattle for the past two years, remains in Seattle and will become the representative for AMS products in the Northwest area. Also, AMS products will be represented in the states of Tennessee, North and South Carolina, Georgia, Florida, Mississippi, Alabama, Louisiana, and Texas by **Interface Audio** under the direction of **Ridge Nye**, President.

• **FSR, Inc**. has just purchased a 30,000 square-foot manufacturing facility in West Paterson, New Jersey. The company, which manufactures audio and video control equipment as well as custom metal fabrication, will move into their new home in March. Their new address is 244 Bergen Boulevard, West Paterson, NJ, 07424.

• **Applied Research & Technology, Inc.** (A.R.T.) has moved to new facilities. The new factory is approximately double the size of their previous factory. The new building was designed to utilize the latest in production and assembly automation, yet integrate all other business functions. In addition, A.R.T. continues to add to its manufacturing technology base with four new pieces of equipment designed to facilitate the entire printed circuit board assembly process, from component insertion to final test.

• **International Music Company** announced the appointment of **Woody Moran** as the new Director for all **AKAI Professional and Digital** products in the United States. Mr. Moran will be responsible for coordination and direction of all AKAI activities in the U.S.

• **Studer Revox America, Inc**. (SRA) has relocated its Western Regional Sales office to larger premises in the San Fernando Valley. The 3,200 square feet of available space at the new office complex has been divided into a new showroom and demo area, enlarged office space, plus a fully equipped service center. The new West Coast address for SRA is 16102 Hart Street, Van Nuys, CA, 91406, the phone and fax numbers remain unchanged.

• **WaveFrame Corporation** announces its planned merger with **Cybermation, Inc**. of Long Beach, CA, maker of the CyberSound editorial system. Concurrent with the merger, WaveFrame announces the raising of $3.5 million in capital to fund growth. WaveFrame will be headed by **Charles Grindstaff** as President and Chief Executive Officer, **John Melanson** as Chief Technical Officer, and **Steve Krampf** as Senior Vice President of Sales and Marketing.

In related news, **Martin Audio Video Corporation** has been selected by **WaveFrame Corporation** to represent its product line in the New York, tri-state and metropolitan area, including WaveFrame's AudioFrame Digital Sound Production System, which is used by musicians, the recording industry and jingle companies, as well as in the production and audio enhancement of film and video.

DIGITAL PERFECTION

THE MULTIVERB II

A.R.T. proudly introduces a stunning new version of one of the finest sounding effects processors ever made. The Multiverb II offers four studio quality effects at once, pitch transposing, Performance Midi® and new reverb algorithms that border perfection—with noise and distortion figures dramatically lower than comparable units! The Multiverb II offers more power and performance than anything in its class.

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- Real-Time Performance Midi®
- Up to four top quality effects at once
- Pitch transposing - 2 full octaves
- The finest sounding algorithms available
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- 200 memory locations • Full parameter variability
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THE MULTIVERB EXT

Sampled Perfection! This new super-effector does everything that the Multiverb II does with delay time of two full seconds and studio quality sampling! You can create shimmering layered sounds and sample on the fly using manual or automatic triggering. Besides using four effects at once, you can control everything in real time with Performance Midi®. The EXT provides the amazing reverbs and musical accuracy that A.R.T. is famous for, with the power to create all you can imagine.

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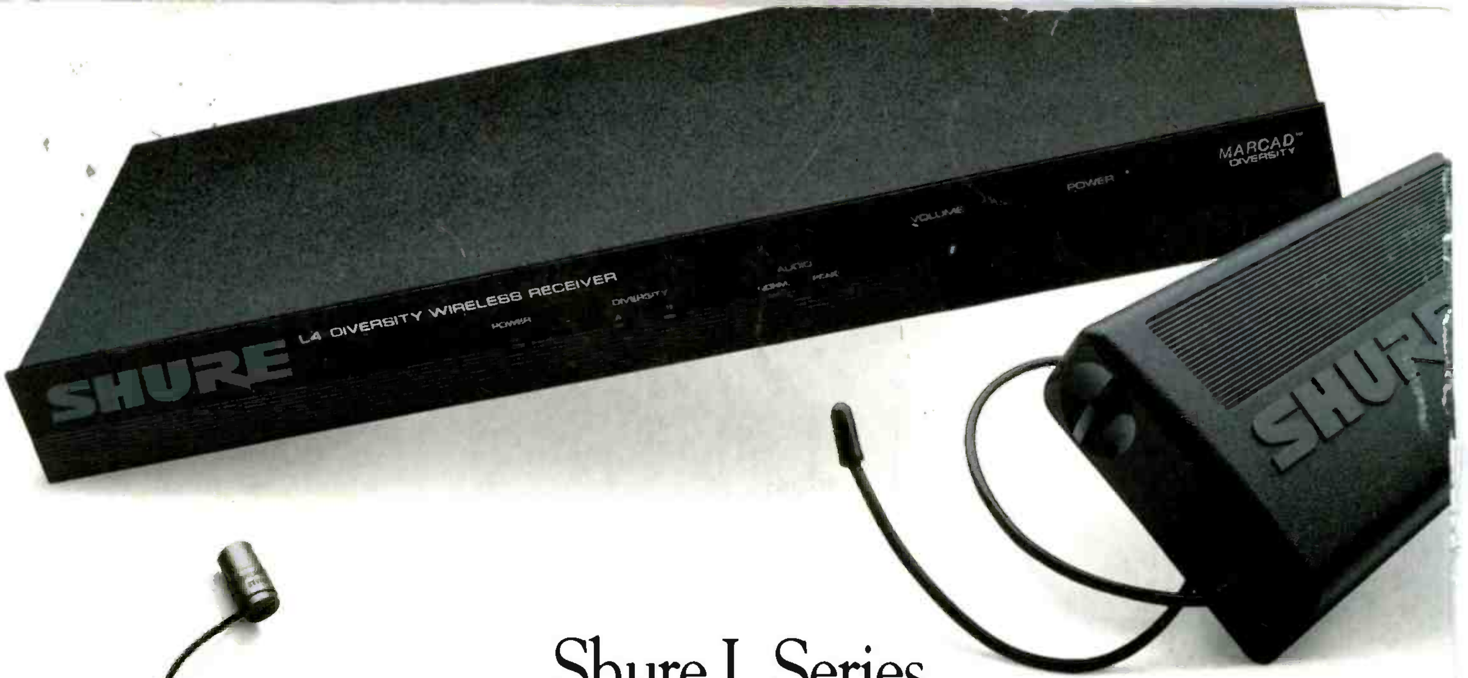

Shure L Series brings reliability to affordable wireless.

Why take chances with anything else?

If you're providing wireless microphone systems to churches, schools, or other value-conscious users, you need reliable equipment you can sell at an affordable price—and make a profit doing it.

That's what the new L Series from Shure is all about. The L Series sets a new standard of value in its price range, offering features, performance and reliability other "economy" systems can't match.

We didn't forget the details.

Designed and built by Shure in the U.S.A., L Series systems include many of the features that set professional-quality wireless systems apart from the "toys." L Series receivers are sturdy, metal-cased, and rack-mountable. Antennas are detachable and may be placed in remote locations, providing excellent performance in situations where many other wireless systems have trouble.

Our L1 Body-Pack Transmitter has features like a separate audio mute switch and a universal 4-pin "Tiny QG" connector that accepts a variety of microphone and musical instrument sources. And L Series lavalier systems come with the 839W, a reliable Shure condenser microphone designed for clear, natural vocal pickup.

Performance meets economy.

Even though L Series components are economically priced, they incorporate sophisticated RF technology. The L4 Diversity Receiver utilizes "intelligent" MARCAD™ circuitry to monitor signals from its two independent RF sections, blending them in the optimum proportion—not merely switching them. The result is reliable, uninterrupted audio with no clicks, no pops. And all L Series systems feature Shure "Mirror Image" companding, plus high-gain, low-noise MOSFETs, a high-fidelity quadrature detector, and a 3-pole Chebyshev audio low-pass filter. It all adds up to outstanding audio quality with exceptional freedom from noise and distortion.

Why risk callbacks with anything else?

Other systems may not meet expectations. But you can recommend a Shure L Series system with confidence. So why risk callbacks—and your reputation—with anything else?

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