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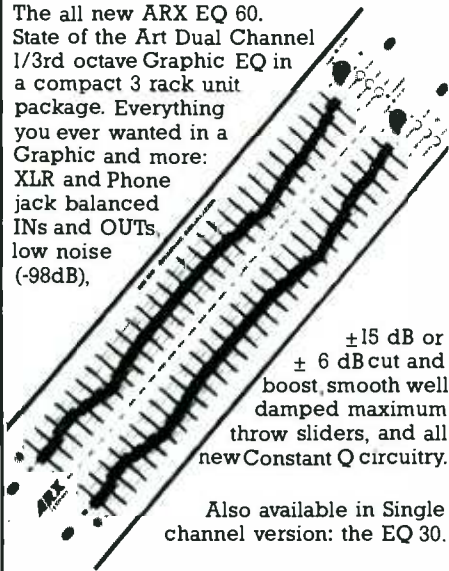
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### ABOUT THE COVER

● Advision's main room. Apparently out of the mainstream of England's London music scene, Advision has been able to build a world-renown studio and mobile operation from their location in Brighton. Proof, that if you do it right, people will beat a path to your door! See Randy Savicky's article beginning on page 10.

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## CALENDAR

● Among the many highlights of the technology program at the **National Association of Broadcaster's Radio 1991 Convention** is a day-long seminar on digital radio. The convention is scheduled for **Sept. 11-14 at San Francisco's Moscone Convention Center.**

**The 4th Digital Radio Station Seminar** on Sept. 13 will offer presentations on equipment and software for digital audio storage and distribution. Discussions during the afternoon will focus on Digital Audio Broadcasting with presentations by each of the DAB system proponents. An additional materials fee is charged for this seminar. A panel with representatives of each DAB system will close the day's agenda.

For more information on this and other seminars during the convention, please call **NAB Science & Technology** at (202) 429-5346. For registration information, please call (800) 342-2460.

● **Roland J. Zavada**, longtime standards advocate, will deliver the Keynote Address for the **133rd SMPTE Technical Conference and Equipment Exhibit** which will be held from **Oct. 26 to Oct. 29** at the Los Angeles Convention Center. Zavada's speech, tentatively titled "Managing the Moving Image," will cover the Society's 75-year history from an engineering point of view. He will present the Keynote Address during the opening session on Saturday morning. Other featured speakers of the conference are **Gregory Peck**, who will speak at the Honors and Awards Luncheon on **Oct. 26**, and **Burton (Bud) Stone**, of Deluxe Laboratories, who will speak at the Fellows Luncheon on **Oct. 27**.

● **The Sony Professional Audio Training Group** will offer a seminar on the **CD Mastering Format Oct. 28** and an applications seminar on the **Sony APR-24/APR-**



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5000 Nov. 7-8 in New York. The CD Mastering Format Seminar will cover the theory and data structure of the CD Mastering recording format. A/D conversion and quantization and encoding/decoding schemes unique to the Sony CD Mastering System will be discussed in detail. In addition to the recorded signal format, digital audio interface formats are explained. The fee for this seminar is \$200.00. All of the APR-series professional analog audio tape machines, including the APR-24, APR-5002 and the APR-5003V, will be featured at the APR Series Applications Engineering Seminar, which costs \$300.00. Operational aspects will be explained and demonstrated, and all stages of training will be reinforced with practical "hands-on" sessions.

For additional course information or registration, please contact the Sony Professional Audio Training Group's technical training office at (305) 491-0825, extension 186.

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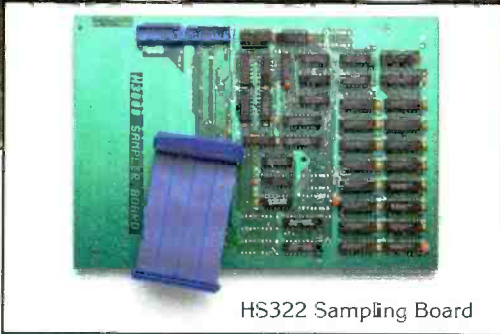
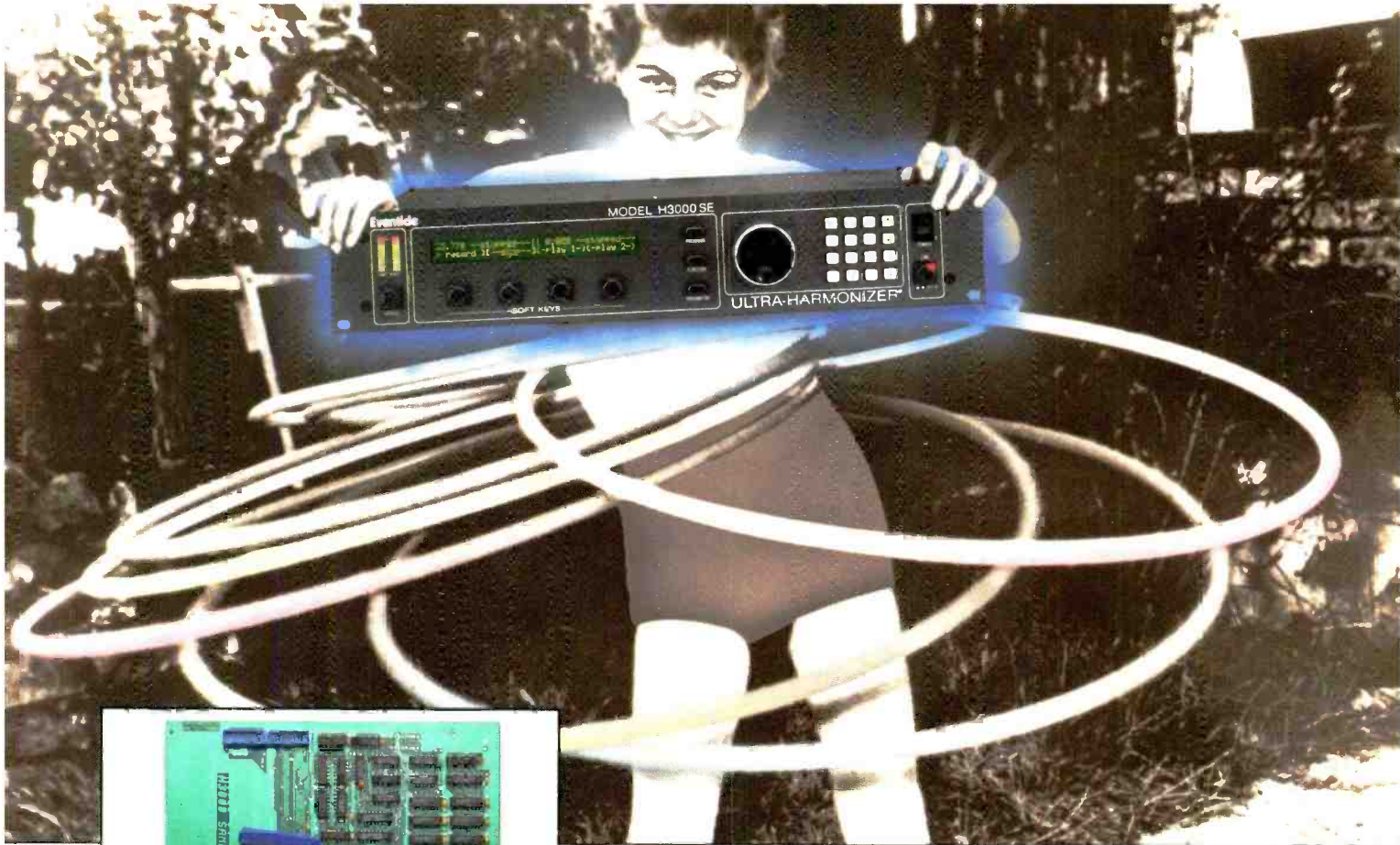
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See the ELAR Booksource Insert

4 db September/October 1991

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HS322 Sampling Board

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The HS322 board records two stereo or mono sounds, with the flawless audio quality that's

made the H3000 a studio favorite. You also get fast, flexible, precise control. Start recording and playback manually, via MIDI or audio trigger. Edit Start, Stop and Loop points using the famous Eventide Knob to "rock" audio, or enter locations on the keypad.

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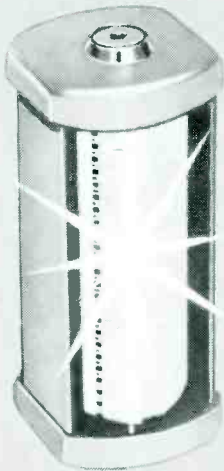
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**LETTERS**

The Editor

One of the more difficult tasks assigned the less-experienced and equipment-endowed engineer is the recording or reproducing of the human voice—singing or speaking. Toward that end we have been reading John Barilla's excellent series in each issue. His **Designing Vocals** series has so far not only hit the mark but given us some insight into what we can do with our existing mics and sound shaping electronics.

We've managed to develop a decent living out of the local talent pool that rotate through our clubs. Of course, much of the work is recording live in those clubs with a minimum of mics and onto a Fostex R8 with a Tascam 12/8 console in a portable case. Often, we are also asked to augment the house sound. Toward that end the wagon also has several speaker systems, and a well-travelled Crown DC-300A.

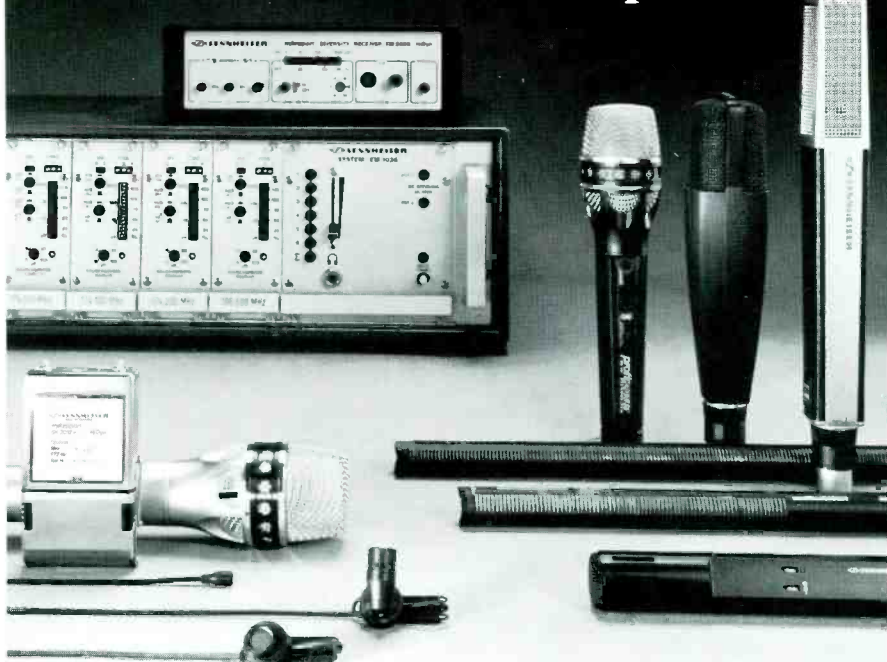
The bottom line is how much we have learned from **db Magazine** articles, and particularly Mr. Barilla's **Designing Vocals**. Will there be more?

Jerry E. Cohn  
LIVE Music, Inc  
Anaheim, CA

*From the Editor:*

*Jerry, we though you'd never get to the last question, but you did. John's column is in this issue and covers more on **Designing Vocals**. What's more, he has told us that he has more to say. It will be in the November/December issue.*

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These are true 3-way designs—unheard of in this size class, but a fundamental principle in all EAW full-range systems. An advanced midbass horn and ultra-rigid carbon-fiber cone driver cover the entire midband, producing over 130 dB SPL with lower distortion than comparable two-way systems. The custom-designed woofer uses a flat wire wound voice coil and massive, optimally aligned magnet structure to achieve exceptional efficiency and surprisingly impressive bass.

True 3-way design also makes the KF300 Series a genuine Virtual Array™ system. Horizontal coverage is uniform from 400 Hz all the way to 18 kHz, and precisely matched to the enclosure angles of two KF300i's arrayed with our new SB330 subwoofer. The result is consistent, predictable coverage without comb filtering, lobing or hot spots.

The KF300 Series includes a variety of hardware configurations and powering options, such as the horizontal AS300 enclosure, designed for distributed systems. Each version delivers coherent output, controlled directivity and wide full power bandwidth in concert sound fill coverage, theatrical systems, dance clubs, corporate theater and other demanding applications.



The MX300i CCEP™ (Closely Coupled Electronic Processing™) unit provides overload protection, LF excursion control, phase compensation and idealized crossover characteristics for the KF300i.

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# Editorial

Some of you are seeing db Magazine for the first time, having picked up your copy at the AES Convention. Hopefully, you like what you are reading and will join our growing family of professional audio subscribers. Whether you have entered this industry for only a few days or for many years, we have something for you. Welcome.

We do not do record reviews, leaving that chore to those consumer music magazines that are out there. So, why am I now writing one such review? That's because, while the recording in question does contain music, it is its technical engineering qualities that make it important to note.

The recording in question is from Delos International, Inc. and it is called Engineer's Choice. The engineer is John Eargle who has had an over-25-year career as a recording or chief engineer for RCA and Mercury Records, and is now Director of Recording for Delos.

The CD album contains twenty-two cuts of excerpts from Delos releases, each picked by Eargle to illustrate a microphone and/or orchestral pickup technique.

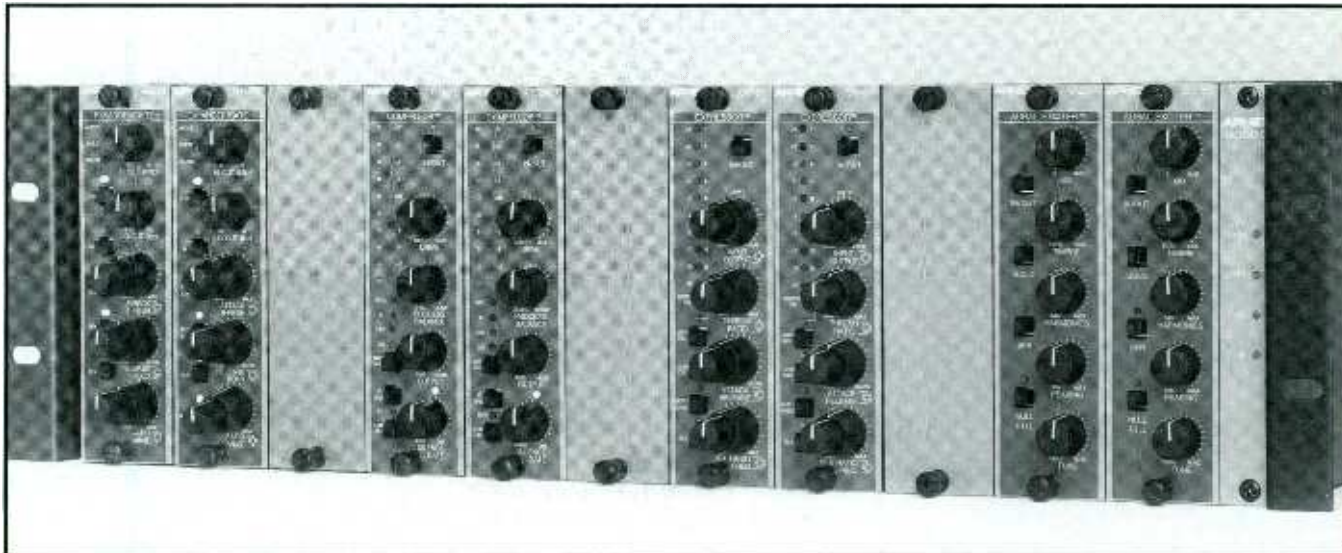
All the selections are of classical works, but range from oversized orchestra to solo piano, vocal, or chamber selections. Each is an example of recording/microphone placement at their very best.

Here's a quote from the extensive liner notes that Eargle wrote: "...I have selected 22 examples covering musical styles from classical to modern and ensembles ranging from solo instruments to orchestra. I will explain in detail how and why each microphone setup was used. There are nine examples which feature the Seattle Symphony Orchestra directed by Gerard Schwartz. This should come as no surprise, inasmuch as that ensemble records more CDs today than any other U.S. orchestra! The Seattle Opera House has become a laboratory, so to speak, and has given me the opportunity to experiment and refine orchestral recording techniques."

The CD is a textbook of classical recording technique and therefore belongs in the tool chest of every present or future soundman. I strongly recommend it. L.Z.

**Meet db Magazine at the AES Convention at Booth #1722**

# Space Saver...



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- ❖ Expander/Gate - simply the world's finest gate, no one ever met our \$10,000 challenge to find a better one!
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# Advision: The U.K.'s Top Mobile Recording Facility

*More Than The U.K.'s Top Mobile Recording Facility—Advision: A Powerful Three-Way Combination—Remote Truck, Recording Studio & Post Facility.*

**J**UST LIKE THE UNITED STATES, THE United Kingdom recording studio marketplace has undergone some profound fundamental changes in the past year. Continuing pressures on studio rates, the growing number of project or private studios, high interest rates and new business tax rates in parts of the country have made it even harder for U.K. studios to compete. Some studios have been forced to leave the heart of the U.K. recording industry—London's West End and SoHo—because of mammoth rent increases and a new business tax rate. Faced with these pressures, a number of studios have changed hands or gone under, and even such London recording institutions as Air Studios have made the decision to relocate to new facilities.

In this era of turmoil and increased specialization in the U.K. studiomarket, Advision Ltd. remains one of the leaders. Today, led by Director Doug

Hopkins, Advision is comprised of several individual facilities, offering a highly effective and symbiotic combination—Europe's top mobile recording facility, a newly opened state-of-the-art recording studio, and a top-flight post-production facility.

Having recently relocated from the West End of London to Brighton, the seaside resort on England's south coast, Advision has always charted a unique course through-

the United Kingdom to offer eight-track recording facilities.

Until its recent move, Advision was located for 21 years in the heart of London's West End where it had grown into an expansive, multi-faceted facility—three recording studios offering both analog and digital 48-track recording, a digital editing suite for compact disc production, a programming suite, video/film post-production facilities, and the Advision Mobile.

The Advision Mobile continues to be widely used across Europe for live concert recording and television sound work. It has been used to record many major music festivals, including, most recently, the Nelson Mandela Freedom Concert at Wembley, Knebworth '90, The Prince's Trust, and Paul McCartney's latest "Unplugged" album.

Other notable live events handled by the Advision Mobile include the Commonwealth Games

and the Donnington Monsters Of Rock '90 Festival in addition to work for Anglia Television, Granada Television, London Weekend Television, Tyne Tees Television and Yorkshire Television.

A short list of music clients includes Bon Jovi, Bruce Springsteen, Eric Clapton, Prince, Placido Domingo and U2.



*Figure 1. The Advision Mobile is used throughout Europe for live concert work, television sound work, and many music festivals.*

out its nearly forty year history.

Advision was formed in 1954 as a film production company and sound studio. By the early 1960s, it had dropped all film work and was specializing in sound work only. In fact, Advision was the first studio in the United Kingdom to use magnetic recording tape and, in 1967, it became the first studio in the

*Randolph P. Savicky is president of RPS Communications, Centerport, NY, a public relations and marketing company specializing in the professional audio, video and broadcast industries.*



# In the right hands they're incredible tools.

We'd like to introduce you to the Crown PZM®-30F and 30R\*, two extremely versatile and indispensable microphones for the recording studio.

Like other professional tools, it takes some experience and skill to discover everything they're capable of doing. But, when used properly, they do what no other microphones *can* do.

Unlike conventional mics, the pressure-zone design of the 30 series uses a miniature condenser mic capsule to receive direct and reflected sound simultaneously. Direct sound from the source and reflected sound from a wall, floor or other boundary combine in-phase to produce a wide, smooth frequency response free from phase interference. The result is increased sensitivity, superior clarity

and "reach" with little or no off-axis coloration. All you receive is clear, natural sound.

Because of their unique PZM design, the 30 series excels when used to record sound near hard reflective surfaces such as pianos or instruments surrounded by reflective baffles. They're also an excellent choice for recording room ambience or, when used in pairs, for recording near-coincident or spaced-pair stereo.

With some experimentation, you'll find the 30R and 30F open a whole range of possibilities and solutions not available with any other microphone. We think you'll find them an important part of your recording "toolkit."

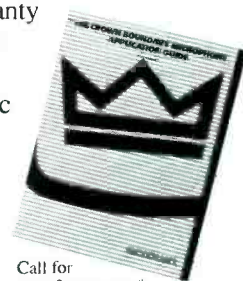
Like all Crown professional mics, the 30 series carries a full three year

unconditional warranty against malfunction with a lifetime warranty on the acoustic system.\*\*

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\*The PZM-30F features a flat frequency response, the 30R features a rising response. Also available with a smaller profile as PZM-6F and 6R

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"We feel that the Advision Mobile is one of the best equipped and most versatile mobile production facilities operating in Europe," said Ulltan Henry, Advision's technical engineer. "There are lots of inputs for live gigs—which is one of those many areas where you can never have enough—and the truck is outfitted with an extensive collection of top-flight, tried and true studio gear.

"We are seeing a continued and even stronger demand for the best audio quality possible at live events," Henry continued. "Concert audiences are more demanding for studio quality-like sound, and the home viewer and listener is even more demanding. They know how good their home stereo system can sound, so when they're watching an event like the Mandela concert on television, they expect sound quality that will rival their home systems."

Fully air-conditioned and featuring closed circuit TV monitoring, the Advision Mobile is built around a 62-input Helios console with full equalization, four auxiliary sends, split 24-track monitoring with aux sends, mix-minus, solos, cuts, and VU and PPM metering. There are in-line monitoring facilities for 48-track recording, and it is possible to record from the monitor independent of solos or cuts.

Sony PCM-3324 24-track digital tape recorders are available for 24-track or 48-track digital recording, and there are two Otari MX80 24-track analog tape machines and two Studer stereo tape machines. On hand are 24 channels of Dolby A noise reduction, with Dolby SR (Spectral Recording) available as well.

The Advision Mobile also includes an extensive selection of microphones and a well-rounded collection of outboard gear.

"Much of the work is brought back from the Advision Mobile to the studio for mixing," Henry noted. "We feel that we have the unique ability to know how to capture the live sound correctly so that any potential problems are eliminated while we are recording a concert or event. Our house engineer, Pete Craigie, is available with the mobile to oversee the project from recording to mixing. We can't try to blame any

audio problems on another remote facility!

"We also feel that this ability to continue working on a project once the live event is over is particularly attractive to our clients," Henry added. "In this way, we are able to insure the audio integrity of the work we produce throughout the whole project."

## THE STUDIO

Advision's new studio facility, which opened in October 1990, is housed in a converted church close to Brighton's many shops and leisure facilities. A fully residential recording studio, only one hour from London, Advision offers one of Europe's finest recording rooms, technical expertise, the latest in digital technology, and first class accommodations and catering.

"We wanted to open a residential facility," Henry said, "but we didn't want the typical English residential facility, a mansion in the country with horses. Instead, we wanted something unique and different.

"We found that in Brighton," he said. "Brighton offers the sun and the sea, as well as clubs, pubs and nightlife. It's the perfect place to work during the day and then go out at night."

The studio itself, which is naturally lit by high arched windows, is approximately 36 feet by 33 feet by 30 feet. There is a dead/soft booth that is approximately 10 feet by 10

feet, a live/hard booth that is approximately 10 feet by 15 feet, and an isolation area that is approximately 10 feet by 10 feet. A floating riser for drum kits, etc., measures approximately 12 feet by 15 feet. "The church's natural acoustics give the studio a wonderful sound quality that would take a great deal of exacting planning and careful construction to recreate in any other space," Henry said. The control room is a spacious 15 feet by 27 feet with an annex for meetings, coffee, and such. Both are also lit with natural daylight.

The decision to scale down the studio's scope from its London size also resulted in an even better recording facility with a wealth of top studio gear, Henry explained.

"With three studios, you can be sure we had a lot of equipment," he said. "What we decided to do was keep all the best equipment and consolidate it into one superb studio. This also allowed us to expand our range of outboard gear and our selection of mics, offering a much wider and deeper inventory than any multi-room facility could possibly dedicate to one room."

## EQUIPMENT

The equipment includes a Solid State Logic SL 6048 E 48-input console with G Series Studio Computer and Total Recall, two Sony PCM-3324 24-track digital tape machines, Otari MTR-90 24-track analog tape machine, Otari MX80

*Figure 2. The studio is housed in a converted church in Brighton and is naturally lit by high arched windows.*





Figure 3. The decision to scale down the studio's size resulted in an even better facility. Here, the new SSL is being moved in.

24-track analog tape machines, and Studer half-in. and quarter-in. tape machines with center-track time code.

Noise reduction includes a Dolby XP24 rack and four Dolby 361 with A type noise reduction, and digital mastering is offered via Audio + Design ProDAT, Sony PCM-701 and Sony C9 Betamax.

"For music-to-picture work, we use TimeLine's Lynx Synchronization system," Henry said. "The Lynx modules are easily configured for various machine setups and have proven quick and reliable to use.

"Let me add, however, that we also offer the friendliest atmosphere in the industry," he continued. "Great equipment is only part of the secret of a studio's success. If you don't have the right people and the right attitude, you won't have a successful studio."

Some of Advision's studio clients include David Bowie, Dire Straits, Elton John, George Michael, Paul McCartney, Peter Gabriel, Pink Floyd and ZZ Top.

"We're also getting a lot of new clients," Henry noted, "including a lot more dance music than we attracted when we were back in Lon-

Figure 4. Upon completion, the control room is 15 feet by 27 feet with an annex for meetings. Both are lit with natural daylight.



don. We're also looking to increase our share of album projects. In fact, Jason Bonham is coming in soon for six weeks to work on his new LP."

To house clients of this caliber, Advision offers a bright and sunny rooftop apartment. The sitting/dining room has a view of the sea through a rose window and glass doors that open onto two sun terraces. The apartment is completely self-contained with a fully-fitted kitchen and three double bedrooms, including one with its own private bath.

The basement of the studios includes a main recreation and lounge area, kitchen and dining room. The basement also includes another two large double bedrooms, including one with its own private bath.

## POST

Advision's third facility is a post-production suite based around a Lexicon Opus tapeless recorder/editor/digital mixer. Henry explained that the Opus system was originally installed in the Advision Mobile approximately one year ago.

"Many broadcast companies were very enthusiastic about the idea of a system such as this being available for hire on an as and when needed basis," Henry said. "However, despite the enthusiasm, most British video work is still based in London's West End where parking is a problem.

"Our Opus has now been installed within a video facility in this area known as Wiseman," he continued. "The system is heavily booked, with many clients being converted to the benefits of hard disk editing."

Several television series for Channel 4 are now post-produced on the Opus system, including "The Crystal Maze" and "Star Test," Henry noted.

As Advision enters into its fifth decade in the studio business, it seems perfectly situated and streamlined to continue as one of the leading facilities in the years ahead.

## THE EQUIPMENT

### Advision Mobile—Equipment

62-input Helios console

Sony PCM-3324 24-track digital tape recorders

Otari MX80 24-track analog tape machines

Studer stereo tape machines

Dolby-A noise reduction (24 channels)

**Monitoring:** Altec/Crown, Yamaha NS10, Auratones

Multi-standard time-code generators and readers

Color closed circuit video monitoring

250 meters of 24 mic and communications cable

4 x 24 way custom-built splitter boxes

Comprehensive range of cables, stands, direct boxes, etc.

**Mics:** AKG, Beyer, Crown, Electro-Voice, Neumann, Shure and Sony.

**Outboard equipment:**

UREI 1176 limiters/compressors

Drawmer compressor

Drawmer dual gates

Audio + Design Compex limiter

Kepex Gain Brains

Yamaha SPX90

Ursa Major 8X32 reverb

Lexicon PCM60 reverb

Ursa Major Stargate reverb

Dynamite gates

Scamprack with parametric EQ and gates

UREI graphic EQs

**The Studio—Equipment**

Solid State Logic SL 6048 E 48-input console with G Series Studio Computer and Total Recall

Sony PCM-3324 24-track digital tape machines

Otari MTR-90 24-track analog tape machine

Otari MX80 24-track analog tape machines

Studer half-inch or quarter-inch tape machines with center-track time code

Denon DRM24HX cassette decks

Audio + Design ProDAT

Sony PCM-701

Sony C9 Betamax

Sony 5850 PAL U-matic

TimeLine Lynx Synchronization system

Dolby XP24 rack

Dolby 361 with A type noise reduction

**Monitors:** Dynaudio Jade 2, 3 k watt monitors, Yamaha NS10 Auratones

**Outboard equipment:**

AMS RMX16 reverbs

Quantec QRS

Lexicon PCM60

Yamaha Rev7

Yamaha SPX90

Lexicon 480L

AMS DDL

Korg SDD3000 digital delay

Lexicon Super Prime Time

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# Creating a Church Recording Studio

*In 1970, sermons at the First Church of the Nazarene in Bethany, OK, were recorded primarily for the pastor's benefit so he could analyze the outline, delivery and continuity of his sermons. The church did not have a permanent tape recorder, so an AKAI 770X was carried up and down two flights of stairs every Sunday for four years.*

**W**ITHIN THE NEXT FIVE YEARS, there became a demand for cassette copies of the sermons as well as copies of special services and big musical services since music is of major importance to the church's worship. As a result, Sunday services were recorded and duplicated on a small scale.

In 1974, there was a major overhaul of the sound system. The church later invested in a high-speed duplicator to produce more tapes in a shorter amount of time. Some congregants were serving as missionaries in foreign countries and requested tapes of the services.

In 1978, the church decided to move the recording operation to another location within the church. A room had originally been set aside for members to bring their tape recorders to record any service. After

gathering ideas and putting them together along with a few pieces of equipment, we had a recording room.

## INITIAL INSTALLATION

We installed two organ mic lines and two audience mic lines and terminated them at the studio. There were also some great times with those Shure M68 mixers and an E-V/Tapco 12-channel mixer. Around 1982, we took a giant leap forward and purchased a Tascam M15 mixer! It came with only eight input modules, and we knew we would add more. With our tape ministry doing as well as it was, we went on a "module-a-month" plan, filling out the board to 24 channels to mix and 8 outputs. Cabinets were added to the studio, and the table top was cut down to insert the

mixer and have its controls at working desk height (see *Figure 1*).

At this point, we decided that in order to hear all the sound that we should be hearing, we should have some excellent studio monitoring speakers. Some old floor monitor boxes were worked over, and some 15 in. 2-way speakers were installed. A Tascam 80-8 8-channel recorder with a dbx encode/decoder was the church's next big purchase. After that, there was a need to keep the recorders from being over-recorded and distorting the tapes, so a UREI 1176n 2-channel limiter was added to the sound system. It keeps levels up to the desired setting without distorting the recordings. The church next decided to purchase a reverb unit to aid in more quality recordings. Purchasing a unit turned out to be almost the hardest decision we faced because of the vast number of units available. We selected the Yamaha REV-7, which is used almost exclusively for all music in our mix. The Aphex Aural Exciter also adds a nice touch to the vocal solos and groups and their place in the mix (see *Figure 2*).

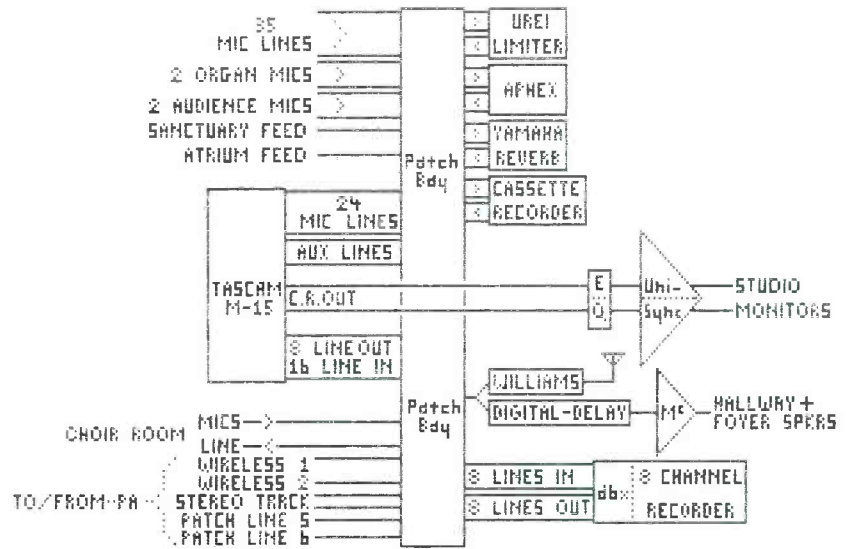
Currently, the studio consists of a 24-channel mix board, 12-channel side mixer, digital reverb, 2-channel limiter, audio processor, auto-reverse stereo cassette deck and an 8-track/8-channel recorder. The studio monitoring system consists of a 2-channel equalizer, 2-channel power amplifier, and two studio quality monitor speakers (see *Figure 3*).

This setup is unique in several aspects. The first is mixing all audio activity as if it were a live album. Next is the fact that there are two mix locations, one for house sound

*Figure 1. The view of the studio. The sanctuary is off to the left through the window.*



Figure 2. The control-room wiring-patching system.



STUDIO PATCH BAY LAYOUT

FLOOR MICS	ORGAN MICS
CHOIR MICS	AUD MICS
TASCAM M-15 MIC INPUTS	CHOIR ROOM
EU-TAPCO MIC INPUTS	
BALANCED / UNBALANCED TRANSFORMER BANK	
LINE OUTPUTS	
LINE OUTPUTS	
LINE INPUTS	
LINE INPUTS	



and one for recording. The third aspect is that in most cases, one microphone feeds two mix board inputs without Y transformers. Fourth is the installation of patch lines between the two boards and assignment of wireless mics and trax feed. The fifth aspect is the installation of audience mics and organ mics which are only available to the recording board.

The mix board was chosen because of its flexibility; since it is a studio board, not a PA board, it has specifications which match most studio boards. There are eight main outputs which are normalled to the 8-track inputs. One of the two stereo outputs is normalled to the auto reverse cassette, one mono output is normalled to the reverb input, and the other is patchable to feed the hard-of-hearing transmitter and the foyer speaker system. As a result of the board's flexibility, it is possible to route all of the above signals to their desired destination in one take. If the output is used to feed television or radio, then the

mix is basically the same as a recorded mix.

An 8-channel/8-track recorder was basically chosen because of its adaptability to the board and its price at the time. It records eight channels at one time, which puts it in the multi-track category. The 8-track has all of the features its big brothers have, except punch-out. One big feature is the built-in dbx encode/decode adapter, while another nice addition is the remote controller which sits on the board.

The auto reverse cassette deck was selected for its automatic features. The unit records in either direction, a plus, since there is no time in the middle of a sermon to turn a tape over. This particular deck also has dbx encode/decode built in. The remote control unit is also a very handy addition.

#### OTHER EQUIPMENT

The digital reverb is a fairly new addition to the studio and was selected because of its versatility. It is a stereo unit with mono capabilities

and there are thirty preset and sixty user-modified settings. The front panel is the user end of a small computer. The wired remote control has become an often used item, even with limited control.

A 2-channel limiter was the next item on the never-ending list of "wants" for the studio. The limiter, which was chosen for its accuracy and reliability, has an extremely low noise figure—a major concern in the studio. It is a very useful tool, especially for controlling an input with dynamics to spare.

The newest unit in the studio is one that re-creates and enhances harmonics. It keeps a solo voice above an orchestra and choir.

The studio monitoring system is made up of a 2-channel equalizer, stereo power amplifier and 2-way speakers. The system is equalized for the flattest possible sound, considering the room acoustics.

The control room is set up on the live-end-dead-end (LEDE) arrangement. A thick carpet covers the rear wall to eliminate bounce,



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and a window in front of the mixer isolates the studio sound from the sanctuary sound.

An extensive patch bay system rounds out the studio's versatility. All needed inputs and outputs are grouped as to mic and line signal levels, and some are normalled. In the middle of the bay separating mic levels and line levels is a special panel with impedance matching transformers plus switching to access the patch lines to the PA board on the sanctuary floor (see *Figure 3*). When the PA operator is asked to play a soundtrack, it can be sent through one or two patch lines if a stereo track is used. The normalled outputs through the patch bay are as follows: the 8-track outputs are normalled to Aux A, Aux B, and mixdown inputs to the board; the 8-track inputs are normalled to their respective group master outputs, the cassette recorder inputs are normalled to the Mon B output; and the cassette outputs are normalled to Tape 1 at the control room selector. The audio processor is also inserted into group masters seven and eight.

### TO Y OR NOT TO Y

There was a big debate as to the addition of mic Y transformers due to a phantom power problem, the loading down of the output of the mic, and the expense of thirty-five transformers. After weighing all the facts and a trial hookup, a decision was made simply to Y the mic jack to both mixers. A test was made for loss of signal and degradation of quality to the two mixer inputs. To our surprise, there was very little audible distortion and only about one decibel of signal loss. After several years of dual use and several different types of mics, we felt we made the right decision in not using a Y transformer.

An early decision was made to re-create as much of a live feel as possible, since the philosophy of the studio is to re-create the feel of being tenth-row center. Every service is a live album and therefore requires a lot of attention. There are two boundary-type mics, hung from the ceiling near the first row, to audibly separate the orchestra from the audience. We also added two hanging mics above the organ pipes.

*Figure 3. A close-up of the studio equipment rack and Tascam 80-8-recorder.*



### EQUIPMENT LIST

Tascam 15 24 x 8 mixer  
 E-V/Tapco C12 Catalina 12 x 4 mixer  
 Tascam 80-8 recorder/reproducer with DX-8  
 Tascam 3340S 4-channel recorder/reproducer  
 Sony TC-755 2-channel recorder  
 Revox A77 half-track recorder/reproducer  
 Tascam V95RX cassette recorder  
 Uni-Sync 50 2-channel power amplifier  
 Yamaha REV 7 digital reverb  
 UREI 1176n dual limiter  
 Aphex 103 Aural Exciter

### BASIC OPERATION

The studio is centered around the Tascam M-15 mixing console. Aux A controls the output to the Williams transmitter and the foyer/nursery amplifier. Aux B sets the input level to the reverb. The green knobs, Auxes 3 and 4, set the input to Aux B mix which is set to about the 12 o'clock position. The reverb return is first routed through echo receive 7 and 8 controls and assigned switches. To rough mix the 8-track,

we select Tape A, 1 through 8, and adjust Mon A, 1 through 8. Mon A on Echo Rcv (Receive) is selected for reverb. To set up for a mixdown, input select on modules 1 through 8 is located and set to the unmarked, square box position. Next, the trim is set to 11 o'clock and Mon B is selected at the Control Room module and set to the desired listening level. Since the cassette recorder input is normalled to Mon B output, a mix can be performed.

### NORMALLED CONNECTIONS

Cassette play to Control Room—  
 Line 1 in  
 Monitor B output to Cassette line in  
 Aux B send to Reverb Left/mono in  
 Reverb Left to Echo Receive 7  
 Reverb Right to Echo Receive 8  
 Aphex Left to Bus Access 7  
 Aphex Right to Bus Access 8  
 Aux 1 through 8 out to 8-track inputs 1 through 8 respectively  
 8-track out to Tape A, 1 through 8, respectively  
 Patch line 1 (wireless 1) to M-15 line in 11



Figure 4. The Sanctuary. This view is from the sound-reinforcement console position

Patch line 2 (wireless 2) to M-15 line in 12

Patch line 3 (track R) to M-15 line in 13

Patch line 4 (track L) to M-15 line in 14

## PATCH LINE DO'S AND DON'TS

The patch is set up in a waterfall arrangement; the top two rows are outputs and the bottom rows are inputs of like devices. In other words, mic outputs connect to mic inputs and line outputs patch to line inputs. The mics are all balanced (three wire connections), and the line in/outs are unbalanced (shield and signal connections). Do not patch between mic and line *in or out*. There are signal level differences (1,000 times) and phantom voltage (+40 volts DC) on the mic lines. Also, do not patch one end of a patch cable to a mic outlet and touch the other end to your mouth. If a patch cord does not appear to work, exchange it with another until it does work, then mark the one that doesn't work. When finished with the project you are working on, disconnect all patch cords and return all controls to the setting you found them (or as near as you can).

## TYPICAL ORCHESTRAL SETUP

We use Sony ECM-44 clip mics for first and second violins. The players, at our instructions, clip them below the bridge pointed towards

the played strings. Due to the proximity of the strings, we have to pull off the extreme high end EQ. We use Sony ECM-30 clip mics on violas and cellos with the same EQ considerations. Sony EMC-33s are used on flutes in a peeking-over-the-music-stand boom arrangement. A Neumann KM-84 on a short stand is placed at the music stand to cover the clarinets. On trombones, a middle height stand is used under the music stand with a Neumann KM-84 mic. Rather than a tall boom stand on tuba, we use a Sony ECM-30 clip mic on the bell of the upright Yamaha.

Sometimes the french horns are mic'd with a Neumann KM-84 on a short stand to the right of the player. The trumpets are the biggest problem in that their play levels change many dB during the same song. We have tried many ways to solve this dilemma and have settled on two Neumann mics at music stand height. We also use two Neumann mics on the grand piano—one on the high end and one on the low strings. A Kurzweil electronic piano is used direct through Shure A-95 impedance-matching transformers. When we have guitar-bass, we take a DI out from it. We have had a harp from time to time, and have found a way of mic'ing it that works best for us—we use a Sony ECM-30 clip mic and wrap it one time around the main post, about 18 in. from the floor, and point the mic toward the sounding

board. We clip the mic to its own lead wire and then route the wire in front and then to a mic outlet. This close-mic'ing technique is employed for a number of reasons.

First, we have to satisfy both PA and recording requirements. Second, it is a live recording situation and we have to make the best of it. If we place a string-group mic in the air, our mix will suffer because there are stronger sounds than strings. We have used a brass-group mic with fairly good results, except that if it is in the air for height, it tends to be obtrusive to worship. Third, the technique with minor EQ adjustments comes close to group mic'ing, except we have to pay a little more attention to the mix. When we do a mixdown of the music and want to get the most from it, we first have to plan our musical strategy; we plan and mix each section as if it were an overdub session. With proper mic'ing techniques and equalization, that planning pays off.

Occasionally we hear a brass player cleaning out his spit valve or a string player plucking his strings to check his tuning, or an occasional wrong note. Ahh, such are the things to look forward to in a live situation. Then we have to decide (in the mixdown) whether to pull that part out, or bring that group in and redub, or try again later. For the choir area, we use eight to twelve KM-84s depending on the occasion. These mics are placed mainly to satisfy PA rather than recording.

## SUMMARY

The art of doing a live recording has been superseded by multi-tracking and studio sessions. The tapes produced by our churches should represent our best, and with proper planning, they can be a reality. We have been pleasantly surprised by a request from a local radio station for our music for their Sunday programming. When we did a television program, we did our own audio mix because we felt it put our best foot forward. Our television producers were free to do their camera magic and did not worry about the audio. It's a lot of work, but very rewarding when we get comments like, "Get a tape...it sounds better than live." db

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**E**LEMENTS OF EVERY MILITARY hardware, from tanks and guns to a "fly-by" by every type of helicopter and aircraft used in the Gulf theater of operations, were included. To enhance the public's enjoyment of

this extraordinary event, an extensive sound reinforcement system was required. The challenge presented by an event of this magnitude was substantial; I was pleased that RCI Sound Systems, audio contractor for the parade and

"thank-you" picnic, subcontracted my engineering services for the event.

Sound reinforcement needs for the celebration were twofold: an extensive speech-only reinforcement system was required to cover the

Figure 1. The Ellipse stage and audience area.

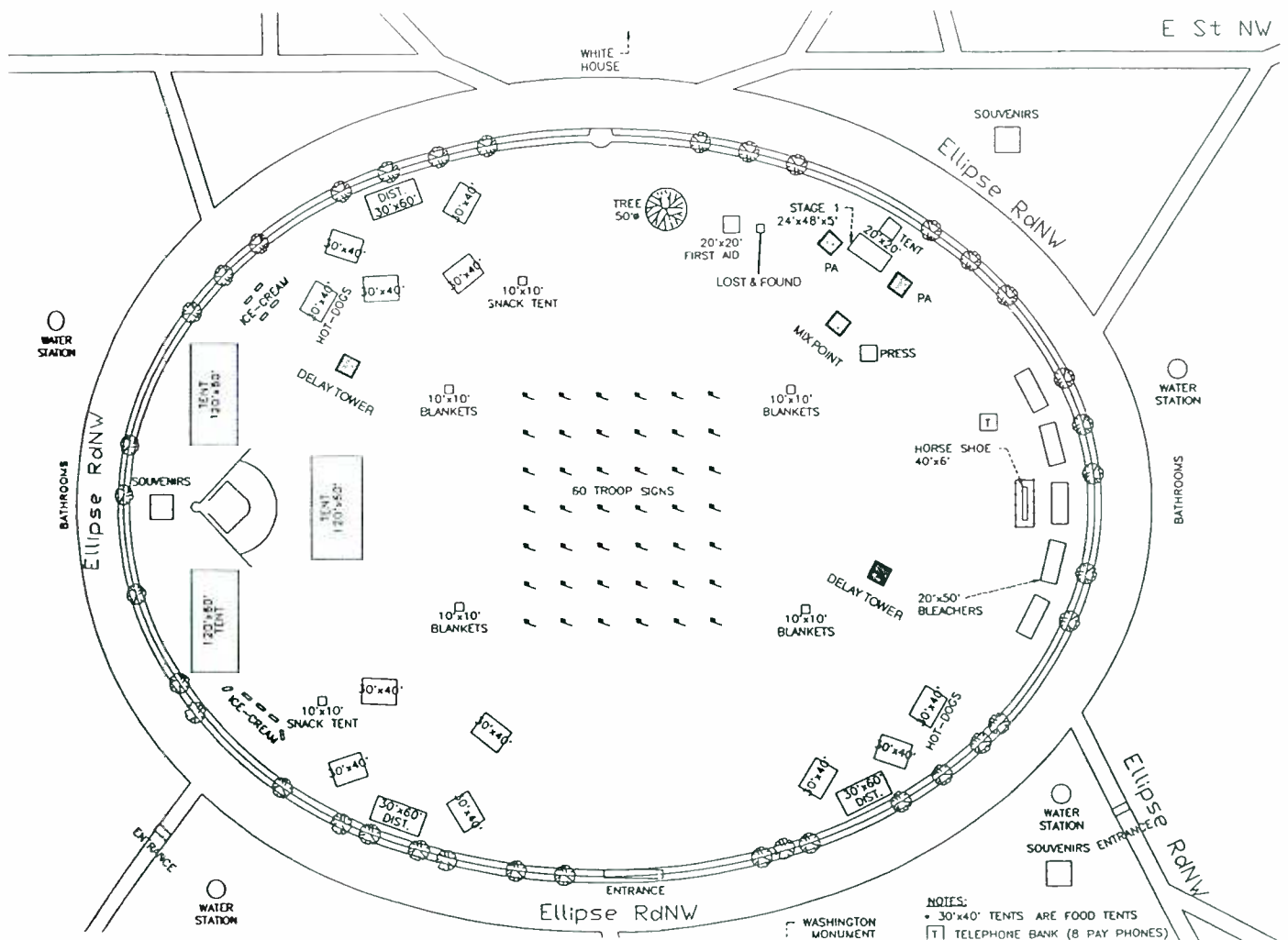




Figure 2. RCI Sound Systems engineers Lorne Greene (L) and Craig Jensen (R) adjust the picnic grounds system. Delay systems and wireless transmitters are visible in the background.

entire parade route, and the picnic grounds needed a large concert sound system, capable of handling the varied audio needs of different military bands and covering a crowd of 25,000 military personnel and their families.

Rick Shepard, RCISS managing partner, planned to use the company's standard concert rig (see *db Magazine*, March/April 1990), augmented by large delay towers, to cover the Ellipse picnic ground's stage and audience area (see *Figure 1*). The delay stacks each had their own AC generator (60 amps,

single phase 120 volts), and were fed from the central mix point via a Yamaha DD1-3 into dedicated wireless links; there would be no cable runs to the distant stacks, insuring both a clean "look" and safety. A Yamaha PM-3000 40-channel console, with a full complement of signal processing equipment, controlled the house system (matrix outputs fed mains left and right, delay towers and a press mult) and also provided four monitor mixes for stage foldback (see *Figures 2 and 3*). RCISS used some of today's most modern equipment

Figure 3. A view of picnic grounds stage with the main house stacks flown from scaffolding bays on either side.



to insure quality sound reinforcement over a large area. The parade system posed a whole different set of challenges; it proved that in our haste to modernize, we shouldn't neglect the value of our older gear.

## WHAT WAS NEEDED

Hargrove, Inc., of Washington D.C., the overall event coordinator, envisioned sound reinforcement as an essential element of the parade. To an average viewer, the parade would look like troops marching down Constitution Avenue with odd military hardware accompanying them. Narration would greatly enhance the experience for all concerned; not only could an announcer identify the unit and/or equipment rolling by, he could also document their roles and subsequent successes in the Gulf operations. These announcements would both inform and honor, which was the whole idea behind the parade in the first place.

## ZONES

The reality of this sound-for-narration idea was sobering: cover two miles of parade route with even speech reinforcement, paying particular attention to the President's reviewing stand and VIP seating areas. With such a long route, it was obvious that a single narrator wouldn't work; the route had to be split up into "zones," so each area could get a description of what was in view at that particular moment. That meant separate systems for each zone. Whatever equipment was used had to generate enough level to compete with both military brass bands and tank engines; sound had to be clearly audible on both sides of the street, with coverage for an estimated audience of one million. Lastly, we had to be careful that the sound from one zone didn't spill into another, resulting in a confusing cacophony of narration. The system had to be tamperproof and weatherproof in the middle of a major city with no security to watch it at night. No cable runs could touch the ground in non-secured areas.

The only place to put a parade paging system on a city street, for both coverage and security reasons, is in the air. RCISS decided to use street lamp poles as the means





Figure 4. A close-up of the Unistrut fastening system for the D.C. lamp posts.

to elevate speakers to a height of 14-16 feet. Shepard had three immediate concerns: "permits, permits and permits. We had to get permission from the DC government to actually hang our equipment from the lamp posts and string the wire between them. We started pursuing these permits a

good two weeks before work was scheduled to begin. I've learned that any delay with city government can be fatal to your schedule. Any streets crossed by cables required a *minimum* clearance height of twenty feet; crossing Constitution Avenue required an extra five feet of clearance for parade



Figure 5. A four-horn cluster located on the corner of 7th and Constitution at the beginning of Zone #1.

floats and military hardware. We figured we'd need about a week to get the system up and working, so we also had downtown parking to consider." Nicely understated.

Most of the work on the parade route took place between May 31 and June 7; the normal business of Washington, D.C. did not stop while we were out on the street stringing wire and hanging equipment, so the job became a creative game of dodging cars and searching for parking spots. Constitution Avenue is a major artery during rush hour, with parking restricted from 4 p.m. to 6:30 p.m., and rush hour stops for no one. Even *with* permits, motorists and police took a dim view of our rolling scaffolds and scissor lift blocking precious traffic lanes. Despite these obstacles, the work was completed on time.

## THE SPEAKERS

The speaker system used for the majority of the parade route was comprised of outdoor paging horns. For the most part, these were Electro-Voice 848A re-entrant fiberglass horns with either 1828 or 1829 drivers. Each horn incorporated a 70 volt transformer, tapped for either 15 or 30 watts, depending on the driver. Not exactly cutting-edge technology, but perfect for this application. "These paging horns are the most efficient means of conveying speech to a large area," said Shepard. "Their pattern is easy to control, and they have a relatively full bandwidth for speech. In areas where I could only put horns on one side of the street, I knew they had the 'throw' to reach the opposite side with sufficient spl. They are weather-resistant, reducing the likelihood of problems from inclement weather. The only exposed surface is the horn itself; that and the fiberglass construction offer some protection from vandalism, an important consideration when you have to leave them up on a city street overnight with no security."

The fastening system was based on two pieces of B-109 Unistrut (1 1/4 in. x 1 1/4 in.), bolted across the street lamp fixture with 1/2 in. x 9 in. galvanized bolts (see Figure 4). At selected street corners, additional Unistrut was attached to form a vertical boom; this provided



bicle were two TOA SM-60 speakers (each containing two 4 in. speakers) that had their own volume controls, so the occupants

could adjust their own levels accordingly.

Each zone had its own complete set of mixing, electronics and amplifiers. Most mixers were six channel JBL, Shure or Yamaha rack-



*Figure 8. Musicasters in the VIP bleacher area.*



*Figure 9. Workmen putting the final touches on the Presidential reviewing stand. Note the bulletproof glass enclosure for President Bush.*



*Figure 10. A JBL Control 5 suspended from the reviewing stand's roof.*

mount mixers. There was a press mult in each zone; most of these had twenty-four outputs. Each location had a  $\frac{1}{3}$  octave graphic (Yamaha or UREI) for overall system EQ. Crown MT-1200 and BGW 750 amplifiers, strapped in mono, provided system power.

Most zones used a single amp to power all speakers, with the exception of zone 3, which had the largest variations in equipment. The two mix points in zone 3 (see *Figure 11*) had numerous amps and graphics, with independent sends: paging horns south, paging horns north, Musicasters south, Musicasters north, reviewing stand, President's cubicle. Zone 6 was another area of special concern. Due to the proximity of the Vietnam Veterans Memorial, spl was restricted. A small Virginia-based sound company, Onyx, was contracted by RCISS to put four E-V Delta-Max cabinets on poles in this area, firing directly into the Bacon Drive bleachers from the same side of the street. This pinpoint, high-fidelity reinforcement helped contain the sound to the bleacher area, preserving the reverent ambience at the Vietnam Veterans Memorial.

Each narrator/announcer was positioned on a twelve-foot high platform, adjacent to the press stand in each zone. We took great care during setup to avoid focusing any horns or speakers in the direction of the announcer's platform, hoping to preserve the best gain-before-feedback we could.

Mic'ing the narrators posed another challenge—we hoped to avoid hand-held or stand-mounted mics that could cause inconsistency in level. For instance, if the narrator turned to view what was coming next, the mic would not move along with him. The distance between mouth and mic would change, so we would experience a level drop. We solved this problem by giving each narrator a Beyer DT-108 single-muff headset. The headset-mounted mic stayed positioned tight to the mouth; if the narrator turned, so did the mic. A special push-to-talk switch, located on the table in front of the narrator, was incorporated into the mic line, so the narrator could control when the mic was on or off; we didn't have to worry about open mics. The ear-

*Figure 11. RCISS managing partner Rick Shepard operates Mix Point 3S during system testing on June 7th.*



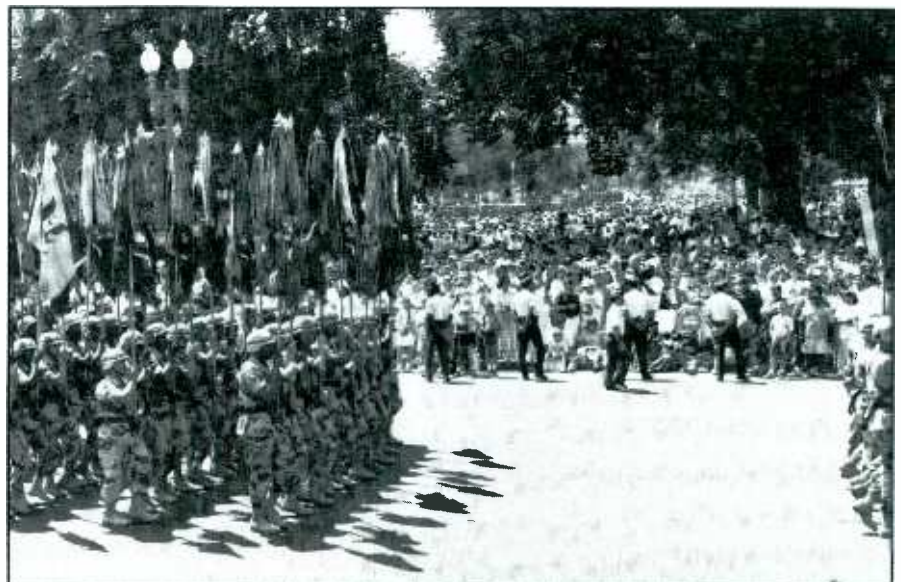
piece was fed from a foldback send on the mixer; the narrator monitored his voice through the headset with excellent gain-before-feedback, so he wasn't confused by the sound of the PA bouncing off buildings, down streets, etc. A Shure SM-58 hard-wired hand-held mic

was positioned on each table as a backup.

### THE BIG DAY

June 8 proved to be a long day. We arrived on site at 5:45 a.m., fifteen minutes before all the streets were officially closed. Our generators

*Figure 12. Just a part of the huge crowd that saluted our troops during the parade.*



had been placed the day before, and were now surrounded by security fencing. Most zones had a 40 amp, 120V generator; zones 3N and 3S had 60 amp generators. Each electronics/amps package was dropped off at its respective location, quickly attached to the pre-run, pre-tested wires, and checked. Our two days of pre-testing paid off—everything worked when it had to.

The security sweep of the President's reviewing area took two-and-a-half hours, at which point a huge crowd had already gathered. The parade came off without incident; a crowd estimated at 750,000 gave the troops a "Welcome Home" they'll never forget (see *Figure 12*).

We did have to deal with one last-minute add-on: the Air Force decided to run their own set of wires to each zone. These were fed from a central announce location, so an Air Force announcer could narrate the

elaborate aircraft fly-by to all zones simultaneously. These wires were hastily run during the evening of June 7, and were just completed when we arrived at the site.

Nevertheless, we dutifully checked these lines, incorporating them smoothly into our equipment. Unfortunately, someone forgot to tell the Air Force about the need for extra clearance when crossing Constitution Avenue with wires.

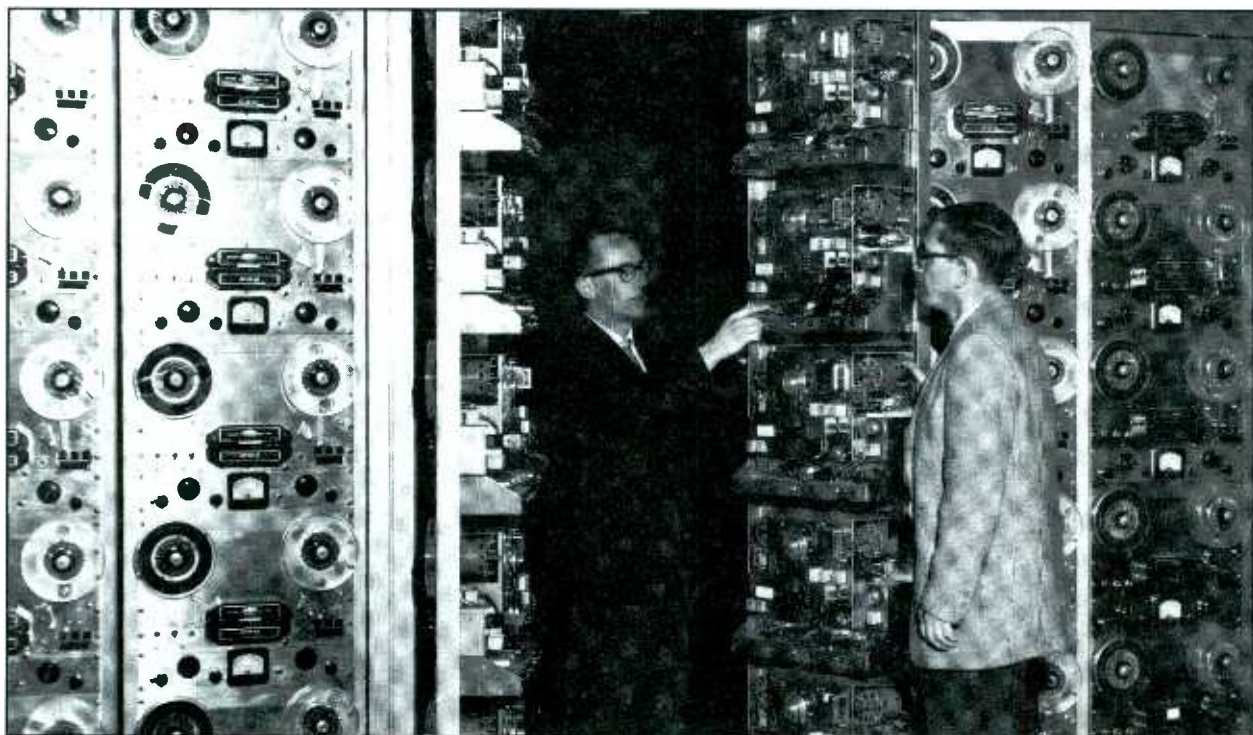
The last float in the parade, representing a "thank you" from the troops to all those Americans who had sent letters to "Any Servicemen," measured twenty-two-and-a-half feet in height. As it rolled down the street, it neatly severed the Air Force feed wires, so when the fly-by occurred, we had no sound source! Such is the penalty for overlooking small details; I'm sure glad it didn't happen to us.

As the parade ended, the picnic began. The troops and their families were bussed to the Ellipse picnic grounds for free food and entertainment by the Army Blues (a twenty-piece big band), the Air Force's High Flight (a Manhattan Transfer-ish vocal group) and the Navy's Country Current (a country music group).

While the picnic sound crew was hard at work, the parade crews packed up all their electronics and returned them to RCISS offices in Rockville, MD by mid-afternoon.

Everyone enjoyed a truly spectacular fireworks show later that night. After a well-deserved day off Sunday, we were back at it on Monday, taking everything down, a job which took two days. It also made me appreciate just how long 14,000 feet of wire is! db

## Historical PERSPECTIVES



*The Earlham College Monster Language Lab was used by students in 1962-1963 at Earlham College in Indiana. The one-of-a-kind Language Lab, designed by Crown Engineer Wayne Blakesley (right), was used to record and play back for students in foreign language classes. Max Scholfield, the head of engineering at Crown during the early 1960s, is pictured at left.*

# Small Group Setup for Live Recording

MALCOLM CHISHOLM

*Life in studios is getting awfully complicated these days. New this, digital that, advanced and integrated the other, all very complex, all very glitzy, and most about half-understood.*

**I**N FACT, IF NOT FOR BUDGET CONSTRAINTS, the modern engineer wouldn't get any work done at all; he'd be too busy reading books on the new equipment. It would be nice if somewhere in the midst of all the complexities there was something simple and dumb that just worked.

There is.

Before getting to all those wonderful toys, with their hundreds of knobs and infinite capabilities, there's the matter of putting out microphones and picking up sound from the instruments. That's the raw material. If it sounds good, you can do tricks with it and maybe make it sound very good. If it sounds lousy, you can do tricks with it and make it sound lousy. The only way to make a silk purse from a sow's ear is to start with a silk sow.

Short of going out and playing everything himself, the engineer can't upgrade the sow. A bad band will be bad under any conditions, and a great recording only reveals the horrible details of just how bad it really is.

On the other hand, the engineer can damn near kill the sow. It's possible to condemn a good band to bad performance with an inappropriate setup, and it happens, because musicians are peculiarly vulnerable to environment.

A musician who spends his entire life practicing by himself can learn to play very well. By himself. Musicians who play in groups learn to do that by practicing in groups, as often as not on stage during public performances.

The performance environment is where musicians develop their ensemble skills, and it's the one in which they are accustomed to playing at their best. That environment can be fairly described as too many players cramped into too little space at one end of a large, very

noisy room. Since that's where your clients learn to play at professional levels, a good case could be made for duplicating it in a studio. Make 'em feel at home. It won't work. For one thing, putting almost anything close to a wall screws up isolation, and for another, audiences are pretty much out unless you're faking a remote. Lots of fun, by the way, but only when it's intentional.

Just because you can't have it all doesn't mean you can't have some of it, though, and the basic bandstand setup works very well in a studio. This brings us to the fundamental purpose of setting up for a session, which is not, repeat *not*, to make the engineer's job easy, nor to correct for rotten studio acoustics, nor even to arrange things for pretty pictures.

## THE FUNDAMENTAL PURPOSE OF SETUP IS TO MAKE THE PLAYERS HAPPY AND COMFORTABLE

Happy and comfortable in this context defines a situation in which musicians can play as well as possible, and know it. Given that, they'll likely perform to the max. They'll sound terrific, and with a little luck so will you. If not, they probably won't, which means you can't, no matter what you do in the control room. A band that goes into a studio knowing it's good but comes out sounding like a bunch of amateurs is not a great ad for the business. Worse, you can't defend yourself.

It's a fact that nobody can tell the difference between rotten playing and a rotten mix. Nobody includes the mixer, who has repeatedly got egg on face by blaming the band for miserable sound before getting up off his dead butt and listening in the studio.

Since we all know that the engineer is automatically to blame for everything that goes wrong in a studio, and musikers are not famous for admitting to crummy playing, it's in your interest to arrange things so musicians can play to the best of their abilities.

Besides, it's your job.

Engineers are hired to get good sound. If you do, you're a hero. If you don't you're a bum, and nobody much cares how you did it either way.

It's commonly said that eighty percent of a mixer's work is the setup, and it's true, but setup is not just a matter of getting appropriate isolation or using the best possible mics.

Great sound comes from well-recorded great performances, and the studio setup is critical to a group's ability to perform, as opposed to merely playing some notes in the same place at the same time.

A last argument for good setups is that they are competitive. A mixer who consistently makes poor setups will turn out consistently shabby sound from both first rate and not so great bands, and one who uses competent setups will drive that turkey into the ground like a tent peg.

And so, finally, on to the what, why, and how.

In the beginning, there were drums.

## DRUM MIC'ING

There still are, and we start with them. Mic'ing drums is its own subject, but since that's been covered in a previous article (see **db Magazine**, November/December 1990), we'll pass on except to put the drums in the middle of the room with a low three-sided absorptive screen around the back and sides of the kit. Proper screen height is just over three feet.

The screen is not there to kill the drum sound in the room. Nothing but a closed room will do that, and a booth puts the drummer out of contact with the rest of the musicians. There's always headphones, but if they were any good, we wouldn't spend multiple thousands of dollars building acoustically-correct control rooms for speakers. Besides, when's the last time you saw people wearing headphones on stage?

Headphones are occasionally useful, but they're nowhere as good as speakers, and they don't even compare with live sound for cueing. Avoid them. If you need cueing leave the studio speakers on (which works better than you might think), or do what the experts do: use stage monitors.

Since a decent setup mostly eliminates the need for artificial cueing anyway, back to the drums.

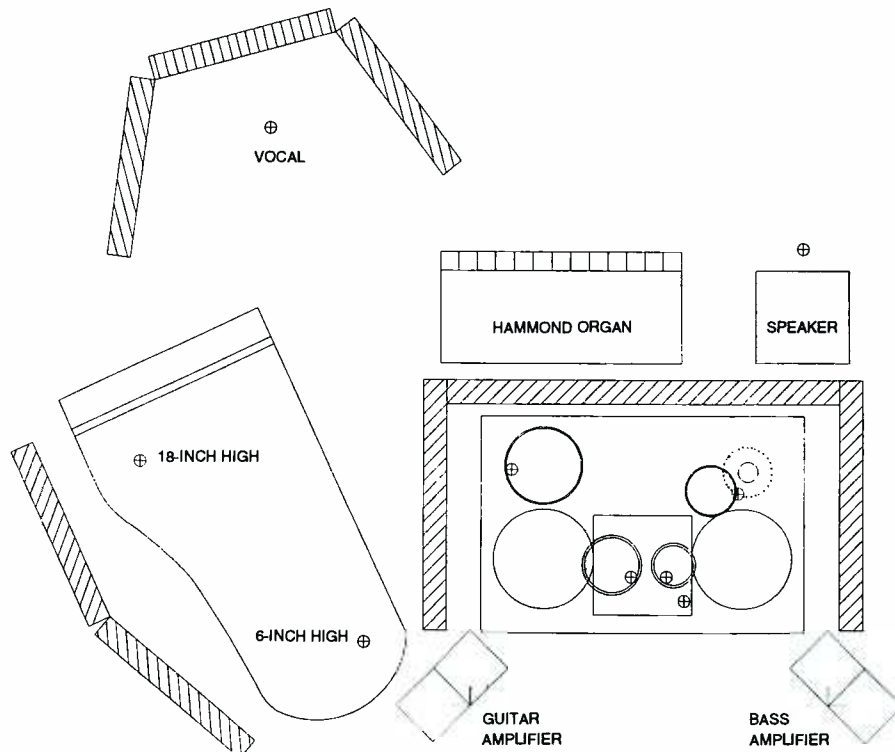
The drum screen is used partly to give the drum mics something absorptive to work into and partly to prevent the drummer's hearing three sets of reflections from the kit. With the screen in place there's only one, from the control room wall and glass. He (she?) can play the other way, but the screen makes things a little more comfortable. Drummers like it.

With the drums centered and screened, the piano goes beside them with the hinge side toward the kit. A couple of ribbon mics work best on piano, but if you don't have them, use what you've got. Don't use more than two mics; you'll never get them in phase. Positions are shown in plan on *Figure 1*. The high end mic is usually 18 inches off the strings, and the low end mic is at about six.

If you don't have ribbons, or the studio acoustics are a problem, a six by eight foot screen will get you respectable isolation on the piano.

## ELECTRIC/ELECTRONIC INSTRUMENTS

Bass and guitar are usually electric, in which case the amps can be placed at the front edges of the drum screen with their sides toward the kit. Put the amps up on chairs or the equivalent. The musicians can hear them better that way, and it improves isolation quite a lot.



*Figure 1. Absorbive flats and mics in a typical small setup.*

Electric instruments are typically picked up with direct boxes, but if mics are needed, the speakers can be close mic'd at their edges, or the amps can be put in three-sided 4 x 4 absorber screens similar to the drum unit to get some isolation with the mics a few feet off.

Given acoustic instruments, the 4 X 4 absorbers afford surprisingly good isolation with the instruments in about the same positions while preserving a tight setup. Don't forget to give the bass a resonator. A four foot square of wooden flooring (even plywood will do) on 2 X 4s will amplify and improve the bass sound a great deal. Bases and celli are made to work on wood floors, and without the floor, you're working with half the instrument.

Mics are dealer's choice here, but if you've got a condenser on a bass, look out for very low frequency energy. You may find the bass is badly out of balance on a small speaker

because of excessive signal below 60 Hz. Some 700 Hz boost will bring it up on the little speakers, and a 60 Hz cutoff will bring it down on the big ones. The cutoff helps a lot on poorly damped electric basses, by the way, and they're getting real common of late.

Drums are directional. Except for cymbals, there is surprisingly little sound back of a kit, and even that can be killed by making the back of the drum screen higher than the sides. Coming up to five feet will do it. Looks funny, but it works well.

The quiet area back of the drum kit makes vocal placement obvious. Close to the piano eliminates some cueing problems and generally works out nicely.

When a Hammond organ is on the menu, it can be placed in the otherwise useless spot just back of the drum screen. That keeps the keyboards close, and both can cue off the vocalist's live sound.

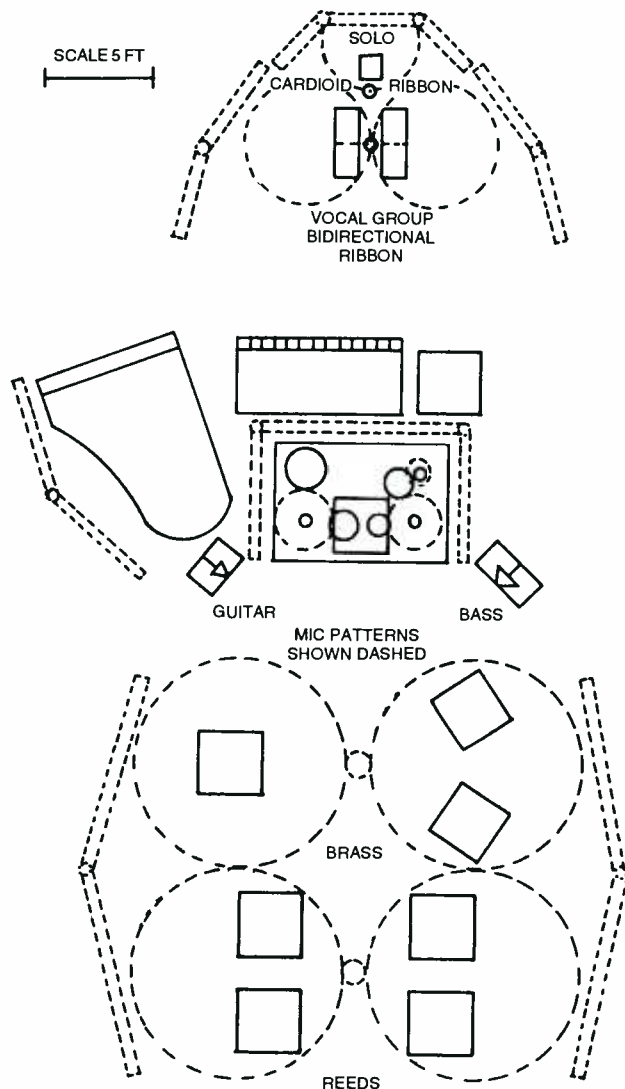


Figure 2. Typical mic pickup patterns are shown in this layout.

than most, which makes stringing a short section out in a line a really bad idea. The obvious solution is to seat them facing each other at ninety degrees to the drum kit, and the logical pickup is bi-directional mics in the middle. You can run into ego problems here as a good many section players tend to think of themselves as soloists in concert rather than members of a group. Actually musicians are both, but they sometimes feel very strongly about being individually mic'd. If that happens, use individual mics with the same seating pattern. It's not optimal, but when clients insist on telling you how to do your job, do it their way and let 'em find out why you like yours.

The overall result of all the above is a very tight setup in which no instrument is playing into another's mic, the players can hear each other without cue systems, and in which they can pass a note without standing up. It's about as close as you can get to the normal performance environment. The communications are excellent, the feeling of "groupness" is strong, and when the band hits a tutti, you can hear the room modes WHOMP! in response. That's the sound of power. Very satisfying stuff.

As to performance, you know you've got a session when the musicians remark that they're having a good time, and the setup system detailed here is designed to produce that feeling. Additionally, it's dead reliable. It works every time without anything much in the way of adjustments after the session starts.

Best of all, it's probably somewhat familiar. A great many live session engineers set up along these general lines as we all face the same problems and so find similar solutions.

To put it another way, this setup is neither a revelation nor a revolution. It is, however, a well-integrated system that has been worked out and used over a course of nearly thirty years of live session work without ever having needed to be revised on session.

There are a bunch of ideas here. If you see one you like, try it. It works for me, and it might work for you. ☐

A Hammond needs no screening, but a vocal does. Three-piece six-footers at the usual four-wide do very well. Four inch casters bring the final height up to about six-feet-five, and with the vocal mic pointed down as in film practice, that's enough. Down mic'ing vocals from eye level has a bunch of other advantages anyway, so nothing's lost.

Vocal backups are best worked as close to the lead as possible, especially for gospel and the like. A very useful trick for that situation uses either a bi-directional ribbon or two cardioid whatevers at right angles to the lead mic, and about three feet away. With a ribbon the lead-background isolation is absolute, and it's only two steps from one mic to the other. A ribbon mic is excellent

for gospel, where you have very tight arrangements and interchanging lead vocalists. Depending as always on studio acoustics, extra screens may prove useful on either side of the background mic(s).

### HORNS

Horn bands are a special problem, mostly because when brass instruments play loud they tend to go sharp, but the reeds go flat. That doesn't create a problem for big bands as the sections are separated enough so they don't mix, but with a only a few instruments you get studio intermodulation distortion, and it sounds terrible.

That's not to say horn bands can't play in tune. They can and do, but they need to hear each other better



# AUDIO FOR THE CHURCH

## Why Is Church Audio So Poor?

● Many years ago, I set out on a mission to analyze why church audio as a whole sounded poor, and how to improve its overall quality. My findings were not all positive; many of the problems were or are not going to be easily solved, but they can be improved by education.

I have found that poor sound quality is mainly due to a lack of education, either by the operator or the person who installed the sound system. Church sound systems are complicated because they require two different approaches in design—one for speech and one for music. If a system is designed for speech, the music minister may be unhappy and want to replace it with a system designed for music. However, a music system may make speech difficult to understand or cause extra feedback problems.

Audio is not as mysterious as people may lead you to believe. People who act as if audio is a mysterious art form usually don't understand the principles themselves. In Don & Carolyn Davis' book *Sound System Engineering*, they describe the differences between art and science in working with sound systems in stating, "The science is in the step-by-step, logical procedure that one follows when faced with a completely unknown problem. The art is in recognizing the cause of the problem when it is first discovered. This comes with experience. When art fails, science takes over." Audio is a science that deals with acoustics, electronics and electro-acoustics (the transformation from acoustic to electrical and back). A popular belief is that if you know electronics, you can be a great audio engineer. Knowledge of electronics is a great start, but you will have problems with making microphones work in the same room with a speaker, or making speech intelli-

gible in a very reverberant room, if you are not familiar with audio.

To improve the performance of church sound systems, you have to increase your knowledge of acoustics, electronics and electro-acoustics. There are several committees that set standards and provide training for the sound and communication industry, such as the Audio Engineering Society (AES), the Institute of Electrical and Electronic Engineers (IEEE), and the National Sound and Communications Association (NSCA), but these standards are more or less unwritten (a true standard is one specified by a group or committee that governs the industry). However, churches have specialized needs as well, and need a group to set up and provide training and standards for technical ministries. Such a group would provide training and certification for different levels of knowledge and experience. This group as a whole would improve relations between manufacturers and contractors, thereby meeting the needs of church technical people.

For this segment, I have discussed testing and certification with many people including manufacturers, church members, contractors and audio consultants. From these conversations came the following test. This test is for you to see how many questions you can answer correctly, and to see which areas of your knowledge need improvement. The other reason for the test is to show you a sample exam that would be given for certification.

There should be multiple levels of certification, such as Operator, Senior Operator, Technician, Senior Technician and Master Technician. Part of the certification would require a specified time period of experience, preferably under someone with certification one level higher than the level trying to be

obtained. To get certification of Operator, for example, you would have to complete course material, have so many hours of hands-on operation under a Senior Operator, and complete a test before certification is awarded.

The following would be task-required for each level of certification.

### Operator

- Equipment set-up and tear down
- Working knowledge of mixing console and processing equipment
- Mixing procedures
- A determined amount of operating experience
- Satisfactory test score

### Senior Operator

- A determined amount of operating experience
- Working knowledge of all audio systems
- Basic electronics, i.e. DC/AC
- Semiconductors
- Basic understanding of electro-acoustics
- Satisfactory test score

### Technician

- Complete all requirements for Operator and Senior Operator
- Detailed knowledge of audio systems and their interconnection
- Working knowledge of electro-acoustics
- Basic understanding of acoustics
- Good trouble-shooting skills
- Working knowledge of basic audio measurement (RTA)
- Satisfactory test score

### Senior Technician

- Complete all of the above
- A determined time of experience as a technician
- Detailed knowledge of electro-acoustics
- Advanced level of system and component trouble-shooting
- Working understanding of acoustics
- Experience using RTA

Working knowledge of RT60 and advance measurement

Satisfactory test score

## Master Technician

Complete all of the above

Detailed knowledge of acoustics, electro-acoustics and electronics with an understanding of psychoacoustics

Be able to design and write specifications of a working sound system

Detailed knowledge of the five parameters of sound system design

Experience in advanced audio measurement

Experience with an Audio CAD Program

Satisfactory test score

## THE TEST

Please take the following test, ponder the concepts, and write in care of **db Magazine**—we would like to hear your comments.

1. What is the purpose of an equalizer?
2. What is the procedure for setting up the mixer's gain structure?
3. Every time you double the amount of microphones that are turned on, you lose/gain \_\_\_\_\_ decibels before it feeds back.
4. Every time you double the power, for example, when you go from 100 watts to 200 watts, you have a level change of \_\_\_\_\_ dB.
5. How many decibels does it take to perceive double the volume?
6. If you have a mic line and a quarter inch line from a keyboard, which will have a stronger level?
7. What device would you use to match a line, such as from a keyboard, to a mic level?
8. What is the difference between a balanced line and an unbalanced line?
9. What is the ratio for the proper spacing between mics?
10. When wiring a mic, the hot pin is always on pin number?
11. The shield is on what pin of an XLR?
12. An echo send or Effect auxiliary is usually pre or post fader?
13. A monitor send or Monitor aux is usually pre- or post-fader?
14. On a mixing console, the high and the low controls are what type of EQ filter?
15. On a mixing console, the mids use what type of EQ filter?
16. An Omni-type mic has a 180 degree pick up pattern. (T or F)
17. A cardioid mic pattern is a good mic to use in a high decibel environment because \_\_\_\_\_.

18. Which type of mic has more gain before feedback?

19. If I have a level of 100 dB with one mic on, what will my level be with eight mics on?

20. What is the international standard on the use of polarity on XLR connectors?

21. What is a good way to locate the primary source of hum in a system?

22. When installing a reverb unit in a church sound system, should it be placed at the input of the main power amp or in a loop of the mixer?

23. What is constant Q in equalizers, and what is its advantage?

24. Where does hum come from?

25. What is the result when two channels are wired in opposite polarity?

26. Define RT60.

27. What is a preferred listening curve?

28. What's the difference between a speaker's sensitivity and its efficiency?

29. Which produces less handling noise—an omni or cardioid mic, and why?

30. What is RFI, and what is one way to prevent it?

31. What is input overload, and what device is used to prevent it?

32. Do constant-voltage amplifiers put out a constant voltage, and under what circumstances can a direct-coupled amplifier drive a 70 volt line?

33. What is critical distance and how many values for it can exist in a room with a multiple-horn speaker cluster?

34. What is the standard rule-of-thumb for providing adequate damping to a woofer as it relates to cabling?

35. What advantage does bi-amplification have over passive crossover networks?

36. What is the primary reason for using High Q devices in a speaker cluster?

37. When people in the first third of the audience from the stage complain that they can't understand what is being said, this is usually caused by what?

Select *one* of the following:

- Echo
- Equalization
- Speaker dispersion
- Crowd noise

38. In a fan-tailed hall, the reverberation tends to congregate in what part of the hall?

- The front
- The back
- The middle
- Evenly distributed

39. In a "shoe box" hall with raised seating, the reverberation tends to congregate in what part of the hall?

- The front
- The back
- The middle
- Evenly distributed

40. The condition of scattered sound is known as?

41. The condition that describes the bouncing of sound is known as?

42. The scattering of sound is known as?

43. The attenuation of sound (in the acoustic environment) is known as?

44. The bending of sound is known as?

45. The bending of sound by density variations in the conducting is known as?

46. Room modes are known as?

47. Carpet on the floor of a hall will \_\_\_\_\_ the RT60 of the \_\_\_\_\_ but not for the \_\_\_\_\_.

Choose from:

- Increase
- Reduce
- Highs
- Lows

48. Define Echo.

49. If a small room (1000 cu. ft.) has a RT60 of one second, and a large room (10,000 cu. ft.) has the same floor, wall and ceiling materials, the RT60 will be about?

Choose from:

- 1 second
- 10 seconds
- 5 seconds
- 3 seconds

50. The proper way to set up a mix is to turn your gain all the way up with the channel slider on 0 dB (or at unity gain), and then slide your masters up just before you get feedback. (T or F)

51. Phase and polarity mean the same thing. (T or F)

52. When using a compressor, usually an ideal compression ratio is 2:1 for vocal work. (T or F)

53. What is a compander?

54. A noise gate is used in what situation?

55. What are the five parameters for a good sound system design?

56. What is noise reduction, and how does it work?

(A very general answer is fine.)

57. Should you use noise reduction with desolver tones or timing codes?

58. What does the abbreviation MIDI mean?


59. What is MIDI?

I would like to thank some of the industry's leaders for taking time out of their busy schedules to contribute to the compilation of this test:

**John A. Murray**, Pro Sound manager of TOA Electronics

**Art Noxon**, president of Acoustic Sciences Corporation

**Monty Ross**, engineer, Rane Corporation

**Larry King** of Klepper, Marshall, King 

# Using Drums in the Church

*Psalm 150:4,6—Praise Him with the drums...Praise Him with the cymbals, yes, loud resounding cymbals.*

## Church Drums

**T**HE EMERGING POPULARITY OF gospel and contemporary Christian music has not only brought change to the entertainment and music industries, but to the church as well. This change has introduced new rhythmic and upbeat music into the church, thus emphasizing the use of drums.

The fear, though, of many music directors and pastors is that the addition of drums will not only raise the overall volume of the music, but overpower the other instruments.

This fear can easily be dispelled by using the proper techniques of tuning, baffling and mic'ing drums. By observing the following guidelines, the drums can become a dy-

namic addition to the church service.

### FIRST THINGS FIRST

Imagine using a \$1,000 mic on a \$100 guitar, or a cheap set of strings on a Stradivarius violin. The results would obviously be disappointing. What might not be obvious, though, is that poor sounding drums will not be helped by expensive mics or an elaborate sound system. The equation still holds true—garbage in, garbage out. So the place to begin is to properly tune the drums. The best person for the job is a drummer, but if he or she is inexperienced in tuning, you would want to follow these basic guidelines.

Each drum should be tuned to a specific pitch, similar to a timpani. Personal taste will dictate what pitches will be used. Begin with the tom-toms. If there are three, pitch them low, medium and high, and if just two, low and high. With drum key in hand, begin by tightening the first lug so that the drum head reaches the desired pitch. Continue diagonally across the head and tighten the second lug so that the head matches the pitch of the first. Proceed in the same manner until all eight lugs are tightened and the head is completely tuned. If a bottom head is being used on the same drum, tune it similarly only at a slightly lower pitch. The kick and snare drum should be tuned in the same manner. It is common practice to remove or cut a hole in the front head of the kick drum and insert a pillow or rug inside (see *Figure 1*). This is done to reduce the resonance of the drum and produce a deep punchy sound. It also facilitates an opening for a mic.

If after tuning the drum still sounds flat or dead, it is possible the head may need changing. Check for dents. The more there are, the less the head will resonate. Replace with either a plastic head or an oil-filled head. Another method of tuning that I have used with much success is match tuning. Match tuning is using a sampled drum from a drum machine as a standard and tuning the real drums to it. There are many inexpensive drum machines on the market (under \$300) that produce excellent sampled drum sound with incredible authenticity. Two

*Figure 1. The kick drum pillow and mic in position.*



people, each equipped with a wireless headset, are needed to tune the drums in this way. First, mic the real drums and connect the drum machine to your sound system. Have one person in the audience monitoring the sound while the other person strikes the real drum followed by the corresponding drum sample. Simply tune the drum by matching it to the electronic sampled drum. Mic placement will play a major role in the final sound and can only be arrived at through trial and error. Let your ears be the final judge.

After the drums are tuned, the next things to check for are any rattles, vibrations and ringing. This may sound trivial, but remember that a small squeak won't sound small after it's amplified through a sound system. Play each drum individually and check for any ringing from other drums. Inevitably, the snare drum will cause you the most frustrations. When any drum in the set is hit loud enough, especially the bass drum, the snare will vibrate. To minimize this, simply tighten the snare. If it persists, a small piece of duct tape placed at the end of the snare wire should dampen the rattle.

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### **Probably no other instrument has the combination of mic'ing possibilities and techniques as do the drums.**

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When using duct tape on any of the drum heads or cymbals, remember that too much tape will eliminate overtones, causing the drums to sound dead and/or the cymbals to lose their brightness. Gauze pads or tissue paper can also be used to reduce vibration and ringing. Simply fold a piece of tissue paper to the size of a matchbook. Move this pad to the location on the drum head where the unwanted noise is occurring, then secure it with duct tape. The ringing in cymbals can be reduced with masking or duct tape applied on the underside of the cymbal in radial strips.

### **TO BAFFLE OR NOT TO BAFFLE?**

Now that the drum set is tuned up and free of rattles, rings and vibrations, the next question is whether you need to decrease the overall loudness of the drum set by using baffles. The answer lies in the acoustical properties of the church. A church whose interior is constructed of mainly hard, nonporous surfaces (especially in the area where the drums are located) will require baffling. Baffles (or gobos) are walls or partitions constructed of soft, porous material which absorb sound. In the church setting described above, the drum set should be enclosed by baffles (see *Figure 2*). To maintain the drummer's vision, glass or plexi-glass continues from the top of the baffle to the top of the highest cymbal.

If the interior of your church is filled with padded pews, carpeted floors and other porous materials, baffling may not be necessary. Most churches, however, fall into a middle category, a combination of both hard reflective surfaces and soft absorbent surfaces. One such church is the Christian Victory Center in Hempstead, New York. The church meets in a theater where the playing area is a mixture of hard and soft surfaces. A spacious stage made of wood is surrounded by tall brick walls and rows of thick, heavy curtains. The combined surfaces

make necessary only a partial baffling of the drums (see *Figure 3*). The baffle is 3½-feet high and made of plexi-glass and masonite; both highly reflective surfaces. The direct sound of the drums gets reflected back by these materials into the highly absorbent curtains which are directly behind and above the drums. The shorter baffle allows for a more 'live' feel between the drummer and the rest of the musicians. Previously, the drums were totally enclosed. The side and rear baffles were made of highly absorbent material. The front wall was plexi-glass and the ceiling was masonite.

The advantages were extremely low sound levels on stage and total control over the volume through the sound system. This type of baffling produced a true studio drum sound. However, the disadvantages were considerable. First, the drummer was acoustically isolated from the rest of the band. A regular floor monitor or hot spot was prohibitive because it would feed back into the mics, requiring a special headphone system. Ventilation was also a problem. Fan noise would inevitably be picked up by the five mics inside the booth. Finally, this partitioned box was not aesthetically pleasing to the eye. The present baffling set up, therefore, was constructed to utilize the best of both worlds. Remember, the

*Figure 2. In a church setting, baffle-enclose the drums.*



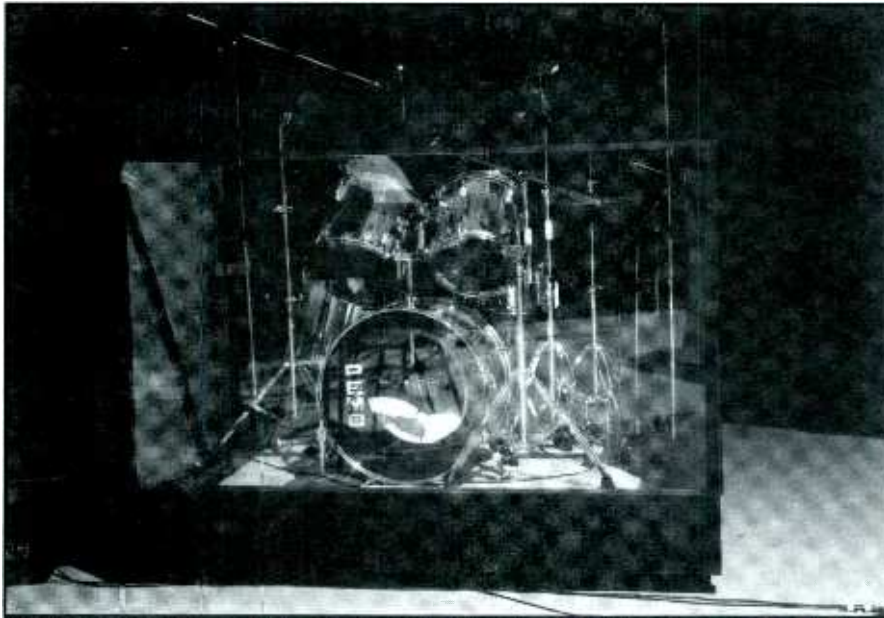


Figure 3. A partial plexi-glass baffle.

main question when designing your own baffling system is "What are the acoustical properties of my church?"

### MIC TECHNIQUE

Probably no other instrument has the combination of mic'ing possibilities and techniques as do the drums. Each technique is based on either the style of music, acoustical environment or desired sound. To achieve a contemporary drum sound however, the most common way of mic'ing is the close mic'ing technique. Starting with the snare drum, most audio engineers prefer the sound of either a moving coil or capacitor mic for snare. The Shure SM57 or Sennheiser 421 both reproduce the sound of the snare very well. Place the mic at a thirty to forty degree angle, horizontal to the drum head and about one inch from the rim and one inch above the head. Make minor moves from that point to get the sound you like. If you have the luxury of using two mics, place one under the snare in a similar fashion as above. This mic will pick up more of the snare's buzz.

There are two common methods for mic'ing the high-hat. The first is mic'ing it separately using a condenser mic. A good choice is the AKG 451. Place the mic approximately three inches above the high-

hat thirty degrees down from horizontal, pointing away from the snare and toward the drummer's left elbow.

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**If your church doesn't own much equipment, or your mixing console has limited inputs, I would suggest using just three mics—one for the bass drum and two overhead.**

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The second method of mic'ing the high-hat is using one mic for both it and the snare drum. The mic of choice would then either be a Shure SM57 or 58. Place the mic between the snare and high-hat and adjust according to the desired sound. Be careful not to place the mic near the edge of the high-hat; the two cymbals coming together produce a rush of air which will be picked up by the mic.

The kick or bass drum is the foundation of the drum set and along with the bass guitar, supports the rest of the musical elements. It is therefore important to mic it properly. As stated earlier in the section

on tuning, the front head should be removed along with enough dampening material to reduce any unwanted resonance. The mic can either be placed on top of a pillow resting against the head at a ninety degree angle or on a mic stand pointing directly at the head. Either of these methods will produce a fuller sound with maximum bass response. If the mic is pointed slightly away from the head toward the side of the drum, more of its overtones will be emphasized. Since the bass drum produces high levels of sound pressure, a moving coil or dynamic mic should be used. Some good choices are the Shure SM57, the Neumann U67 and the Sennheiser 421.

Tom-toms can be mic'd one of two ways, like the snare from above or from underneath. Place the mic at a thirty degree angle and about two to three inches from the drum head. Mic'ing it any closer than two inches will give the drum a nasal quality. When mic'ing from underneath, remove the bottom head. This mic'ing technique will produce a flatter and more percussive sound. My favorite mic for toms is the Shure SM58. Some other good choices are the Sennheiser 421 or the AKG 414.

Finally, to cover the cymbals, two overhead mics are used. Place them about 15 in. above the cymbals and three to four feet apart. Any high quality condenser mic should be used because it gives the cymbals an open, airy sound. My choice is either the Shure SM81 or the AKG C 1000S.

If your church doesn't own much equipment, or your mixing console has limited inputs, I would suggest using just three mics—one for the bass drum and two overhead. The only adjustment would be to lower the overhead mics in order to pick up the toms, snare and high-hat.

In closing, remember that the presence of a drum set in some churches is still taboo. Even if your drummer played with feathers, it's still a psychological barrier to some congregants.

My suggestion would be to purchase a decent quality drum machine and start using it in services. After the faithful become used to hearing it, the transition to the real thing won't be as traumatic. ▣

# 3-D Audio: Wave of the Future?

*Many people connected with the audio industry could not help but hear the recent clamor about the new three-dimensional audio systems on the market.*

**T**hese systems claim that while using only two speakers, they can locate sound far out beyond the edges of the speakers and even place sounds behind the listener's head in true 3-D sound. While these claims are extremely easy to make, delivering on them is not so easily accomplished. This article provides a simplified explanation of how the brain perceives sounds in three dimensions, and then takes a quick look at products from several manufacturers which claim to provide 3-D or enhanced experience in sound. Please note that the information provided here is gleaned from the results of many long hours of research done by various manufacturers, universities and private audio research labs over long periods of time and some of this research has included very complex mathematical models of sound and hearing done on large mainframe computers. A tip of the hat to researchers at Hughes Aircraft, Archer Communications, Pete Meyers Productions, Roland Corporation, Bedini Audio, and the many others who continue to contribute to this fascinating and ever-expanding body of knowledge.

## THREE AURAL CLUES

The human brain is remarkable in its ability to perceive sonic directionality. There are three basic elements in sound perceived by the ears and decoded by the brain to in-

terpret exactly where a sound is coming from. These three elements are: a) differences in amplitude, b) difference in phase or arrival time and c) differences in certain frequencies induced by the shape of the outer ear. It is the combination of these three elements which allow humans to pinpoint the location of a sound in space with uncanny accuracy.

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**The record buying public's fascination with the idea of 3-D sound was evidenced by its embracing the ill-fated and short-lived concept of quad.**

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The first clue, difference in amplitude, is caused by the fact that the head creates a sonic shadow to sound waves. Consequently, a sound coming from more than several degrees off axis from the nose will be perceived as louder by the ear on the side of the arriving sound. This loudness difference will help tell the brain the general direction of the sound, whether off to the right, left, or directly in front or back. It should be noted here that until the advent of these "3-D" processes in recording and mixdown, this was the primary method used to create a stereo spectrum in a conventional stereo recording via the pan control on the recording console during mixdown.

Make the sound louder in the left speaker, and the brain will perceive the sound as coming from that general direction. However, the lack of extremely clear imaging and directionality in conventional stereo demonstrates that this method alone is not sufficient to provide the necessary information to the brain for accurate localization. It is this very deficiency in standard stereo which has led to the discovery and exploitation of the other localization clues.

The second clue, difference in phase and arrival time, is caused by the fact that a sound arriving off axis will arrive at one ear first and then arrive at the other ear several milliseconds later, thus creating an out of phase condition with the sound the first ear hears due to the difference in arrival time. The brain takes this phase information and decodes it, adds it to any amplitude difference present, and is able to localize sound quite well with just these two clues. An analogy might be an artist drawing a 3-D object on two-dimensional paper using perspective and shading. A good rendering can provide a great deal of depth and detail, but is still not a complete 3-D picture. Difference in phase is used in a majority of 3-D audio systems and does manage to provide fairly accurate localization clues, but to get the whole picture, there is a third aural clue which completes the aural image

and transforms this 2-D "sketch" into full 3-D sound.

The third aural clue, change in frequency, has to do with the actual shape of the pinna or outer ear. Depending on from which direction a sound comes, the shape of the outer ear actually attenuates certain frequencies, mostly in the upper register, while allowing other frequencies to pass unchanged. When combined with the other two localization clues, it is this slight aural shaping of the harmonic spectrum which provides the brain with the third and final clue as to the location of a sound in space. This creates for the listener a sense of depth and space and directionality so accurate, that people with closed eyes are able to point directly at the point of origin of a sound with great accuracy.

#### QUADRAPHONIC ROOTS

The record buying public's fascination with the idea of 3-D sound was evidenced by its embracing the ill-fated and short-lived concept of quad. Four-channel stereo's promises of "being there" and "true 3-D sound" in theory were wonderful, but their implementation proved impractical, expensive and the listener's enjoyment of the material was extremely dependent on his location within the sound field.

Undaunted by the catastrophic failure of quad, several companies began research programs to study just how the ears and brain processed sound and how this might be recreated by two speakers. Although albums and soundtracks using a two speaker 3-D process are just now entering the market, the research on the subject began back in the early eighties. Pioneers like Pete Meyers, John Bedini and even Hughes Aircraft foresaw a time when 3-D audio from two speakers would become a reality. They studied the process, learned how it works and began to devise ways to implement the process in recorded sound.

#### RECREATION OF THE 3-D AURAL CLUES

It is well and good to understand how the ears and brain work together to provide localization in real life situations. It is, however, another matter entirely to some-

how recreate and/or process sounds captured by a recording medium, EQed, mixed, sliced, diced and chopped, played back through two speakers, and yet still provide the listener with enough of the aural clues to locate the recorded sounds in 3-D space.

As it has turned out, the leaders in the field went down two different pathways. One path led to the development of methods to expand the stereo image beyond the edges of the speakers and provide a much tighter stereo image with regard to placement of individual instruments within the stereo spectrum. The other pathway led to the creation of a true, 3-D audio experience which would allow the mixing engineer to place sounds far to the listener's right and left as well as above, below and even behind the listener's head. The companies manufacturing these devices are split neatly into two categories based upon whether they try to "expand" the sound or to "three-dimensionalize" it.

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**One would hope that the advent of 3-D audio does not turn out to be just another "emperor's new clothes" phenomenon, but will turn out to be a truly viable technology which will enhance the listener's enjoyment of the music.**

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The first category is the "sound widening" group. These companies market devices which enhance the width of the stereo field and sharpen stereo imaging. Representative of this category are products such as B.A.S.E. from Gamma Electronics and Sound Retrieval System from Hughes Aircraft. These manufacturers do not claim behind the head 3-D sound, but only to restore a greatly enhanced sense of ambience, space and imaging to any musical source, live or prerecorded. These devices may be used as playback devices or during tracking to capture the effect directly to a multi-track master.

The second category is the "3-D" group which claims to provide sound above, below and even behind the listener. Examples of this group are Roland Sound Space from Roland Corporation and the much hyped Q-Sound from Archer Communications. While the Roland product seems to be slowly gaining favor from various studios and producers, Q-Sound burst onto the scene with a roar. Beginning with a super-hyped Super Bowl commercial in 1990 (which was reportedly *underwhelming*), to the recent release of albums by Sting and Madonna using the Q-Sound process, Archer Communications has by far gained the most public attention.

#### IS SOMETHING ROTTEN IN DENMARK?

In fulfilling its commitment to keep its readers informed, **db Magazine** will in future articles report on the progress of this new technology, now in its infancy. It must be noted, however, that not all the attention and press these 3-D products have received has been positive. There are complaints about the so-called "sweet spot," which is a very small location between the speakers where the 3-D effect may be heard, concern about mixes sounding overly bright, and the British Broadcasting Company reportedly is considering banning all 3-D encoded material from broadcast because of problems with mono compatibility. Some of these complaints are troublingly reminiscent of statements detractors made about quad during its brief rise and fall. One would hope that the advent of 3-D audio does not turn out to be just another "emperor's new clothes" phenomenon, but will turn out to be a truly viable technology which will enhance the listener's enjoyment of the music. After all, isn't that the reason for doing all this in the first place? Compact discs and audiophile LPs sound pretty good just as they are and while it would be nice to hear the cellos behind the right ear and the lead guitar behind the left ear, consumers can hope that manufacturers will never lose sight of that old and time-honored saying, "If it ain't broke, don't fix it!"

# A Theater Sound System

*The MUNY (Municipal Theatre Association of St. Louis, MO), a twelve-thousand-seat outdoor amphitheater built in 1917, provides its patrons with grand theatrical productions in a natural outdoor setting.*

**S**INCE ITS ESTABLISHMENT AS A not-for-profit organization in 1919, The MUNY has been producing large-scale Broadway style shows, operettas and ballets in this outdoor musical theater which is the largest in the country (see *Figure 1*). Over forty-four million people have enjoyed these incredible productions. The stage itself includes a forty-eight foot revolving platform which allows elaborate scene changes in a little over one minute!

Equally impressive is the sound system for this summer's series of shows. *The Riverfront Times* said about the first show of the season (*It's Delightful, It's Delovely, It's Cole Porter*), "Kaye Ballard is a delightful storyteller when Porter supplies the words. And you could understand those words, thanks to

*Figure 1. The main entrance to the MUNY amphitheater.*



the best sound I've heard at the MUNY."

To put together the incredible sound system for this year, The MUNY called upon the services of Otts Munderloh, a Broadway sound designer whose experience

includes *Dreamgirls*, *Jerome Robbins' Broadway*, *Sophisticated Ladies*, *Little Shop of Horrors* and many other productions. "Four Apogee 3X3-IIs are flown thirty-five feet high on a cable that runs the width of the facility over the stage apron in a distributed system approach to cover two-thirds of the seating area," says Munderloh (see *Figure 2*).

*Figure 2. The speaker system is Apogee 3X3-IIs flown from cable above the audience.*



## THE SPEAKERS

"There is a single 3X3-II mounted on the pylons that supports the cable for additional coverage at the extreme sides of the seating area. We're also using three AE-5s on either side of the stage on the proscenium arch to cover the first third of the seating area in addition to the six AE-2s used for front audience fill (see *Figure 3*). There is a single AE-3 at the edge of the seating area on either side for extra fill

*Barry Luz is Manager of Marketing and Technical Training at Apogee Sound in Petaluma, California.*



mounted on the pylons that support the steel cable where the 3X3-IIs are flown. The sound quality of the system is exceptional," he said.

## A NEW SHOW EVERY WEEK

The schedule for the actors and production staff is grueling. This season there is one show each week with performances every day and seven shows in all. A show will open on Monday evening while the next show is being rehearsed throughout the week. Saturday mornings, from midnight to 6 a.m., run-throughs are done for the next week's show to focus lights. At 6 a.m. everyone goes home for a short rest with the rehearsal on Sunday at 3 p.m. to hear the cast sing with the orchestra in a rehearsal space. Then, at 7 p.m. Sunday, the last performance of the previous show is held. The sound crew comes in Monday at 10 a.m. and final set changes for the next show are completed. The dress rehearsal of the new show runs from 1 p.m. to 6 p.m. with the cast and orchestra on the main stage. The new show opens 8 p.m. Monday.

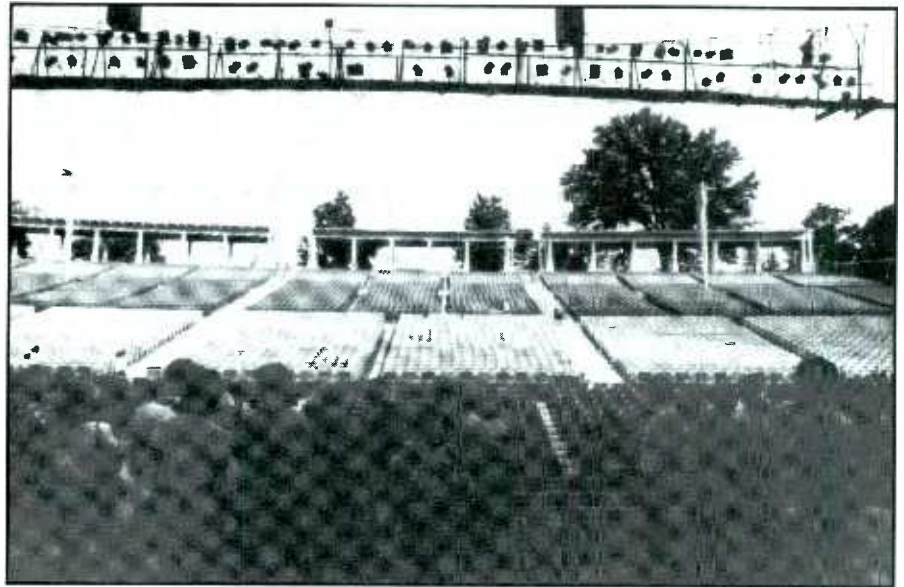
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**There are several mixing consoles in use, including a Yamaha PM-3000 with forty inputs from which Munderloh mixes the show and a Yamaha PM-2000 submixed into the PM-3000 for more orchestra inputs...**

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"Most of this rehearsal time is spent finding out that the wireless mic is hidden under the costume or the transmitter antennae is bent instead of the 'luxury' of working on sound cues, equalization and sound levels," said Munderloh.

"In the old days, when you had five-foot mics, a six hour rehearsal was enough to write down where the actors were positioned on the



*Figure 3. This view is from the stage and shows the expansive outdoor amphitheater.*

stage and turn on the appropriate microphone. Now, in the nineties, audiences expect every person in the cast to be heard as if you were standing next to them. With twenty wireless units, the rehearsal time constraints are very difficult. This year we're using the Sennheiser wireless UHF units with the computer-controlled remote system. It's been working very well for us and because it's an outdoor venue, anything that isn't on a microphone can't be heard. I'm also using pit singers on mics. We're also using a large orchestra by Broadway

standards, approximately thirty-two people."

## CONSOLES

There are several mixing consoles in use, including a Yamaha PM-3000 with forty inputs from which Munderloh mixes the show, and a Yamaha PM-2000 submixed into the PM-3000 for more orchestra inputs (see *Figure 4*). A Soundcraft 200B is also used to submix six reed instruments with its output being sent to the PM-3000. Some of the signal processing equipment includes a Lexicon

*Figure 4. Apogee AE-2s mounted along the front edge of the orchestra pit for front fill.*

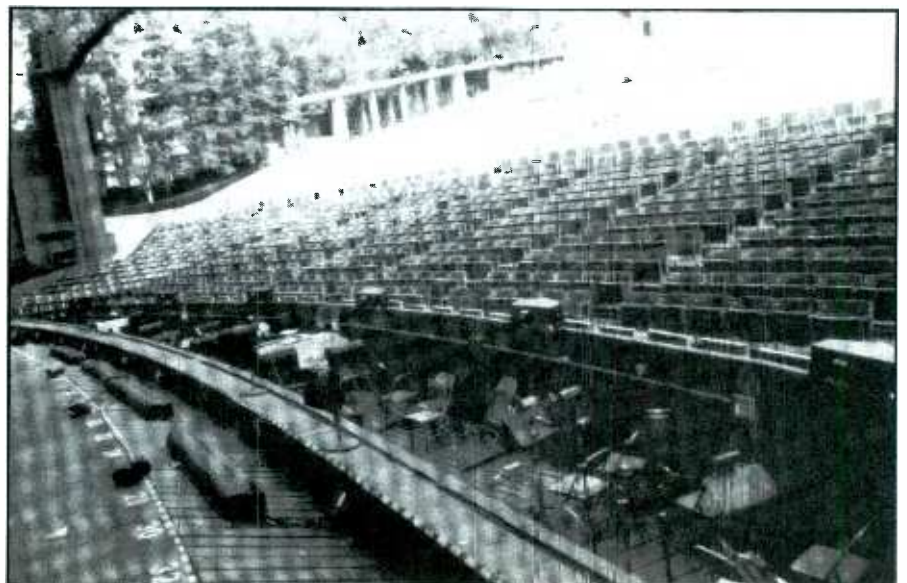




Figure 5. The mix position and its view of The MUNY stage.

PCM-70 delay, and a SPX-1000 digital reverb and several dbx 900 mainframes with compressors/limiters for all wireless units on insert points of the console.

## COMPUTER EQUALIZATION

Another integral part of the sound system is the use of Apogee Sound's proprietary equalization technique CORREQT, (Computer Optimized Room Resonance Equalization Technique). This technique involves the use of very narrow-band frequency analysis and equalization as Ken DeLoria, president of Apogee Sound and developer of the CORREQT technique, explains. "Many people will tune a sound system using very sophisticated test instruments without ever really listening to how live speech and music sounds through the system. The goal of the sound system is to convey whatever signal source it is given in the clearest, cleanest, most accurate fashion. To do that consistently, the CORREQT engineer has to work with the venue's acoustics. His task is to

minimize the detrimental acoustical anomalies of the facility. Listening to how speech and music sounds through the system, in addition to extensive instrumentation-based analysis, are extremely important and form the basis of our technique," DeLoria said.

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**All of this technology  
and effort is put forth  
to achieve one goal: to  
provide to the  
patrons of The MUNY  
the best show  
possible.**

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One of the advanced features of CORREQT is the ability to use music and speech, as well as test signals, for the source which stimulates the system. This allows *real-time* equalization during the show with the audience present.

After DeLoria's initial equalization and setup of the system, a second EQ specialist, Alexander Yuill-Thorton II (Thorny), was brought in to attend the first live show and further refine the system's equalization. This served to accurately "fit" the EQ curves to the venue's acoustical characteristics with a live audience in place.

All of this technology and effort is put forth to achieve one goal: to provide to the patrons of The MUNY the best show possible. If the reviews from the *St. Louis Post Dispatch* are any indication, the following reviewer's statement holds true:

"The MUNY's new sound system is well-nigh perfect. I spent the second act prowling the vast seating area and I heard every word without a trace of crackles, buzzes or static (see Figure 5). The system has excellent balance and both the orchestra and the performers sounded splendid, from my regular seats down front all the way to the top, only a couple of rows from the free seats. I checked the center and both sides, practically from top to bottom, and never missed a thing."

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# RECORDING TECHNIQUES

## How To Use EQ

● Even if you can record a performance accurately, you might not like how it sounds. This is where the equalization (EQ) process comes in. EQ lets you improve on reality: it makes an instrument sound warmer or less harsh, removes noises and adds punch.

EQ adjusts the bass, treble and midrange of a sound by turning up or down certain frequency ranges. To do this, it operates on the spectrum of the sound source—its fundamental and harmonic frequencies. The spectrum helps give the instrument its distinctive tone quality or timbre.

If some of these frequencies change in level, the tone quality changes. An equalizer raises or lowers the level of a particular range of frequencies (a frequency band), and so controls the tone quality. That is, it alters the frequency response. For example, a boost (a level increase) in the range centered at 10 kHz makes percussion sound bright and crisp. A cut at the same frequency dulls the sound.

### TYPES OF EQ

Let's review the types of equalizers from simple to complex. The most basic type is a *bass and treble control* (often labeled LF EQ and HF EQ). Its effect on frequency response is shown in *Figure 1*. Typically, this type of EQ provides up to 15 dB of boost or cut at 100 Hz (for the low-frequency EQ knob) and at 10 kHz (for the high-frequency EQ knob).

You have more control over tone quality with a *multiple-frequency equalizer*: you can boost or cut several frequency bands (see *Figure 2*).

*Sweepable EQ* is even more flexible; the exact frequency range needing adjustment can be "tuned in." (see *Figure 3*) Sweepable EQ is

often incorrectly called "parametric," which also allows control of bandwidth. You won't find a true parametric equalizer in home-studio equipment, however.

The *parametric equalizer* allows continuous adjustment of frequency, boost or cut, and bandwidth—the range of frequencies affected. *Figure 4* shows how a parametric equalizer varies the bandwidth of the boosted portion of the spectrum.

A graphic equalizer (see *Figure 5*) is usually external to the mixing console. This type has a row of slide potentiometers dividing the audible spectrum into 5 to 31 bands.

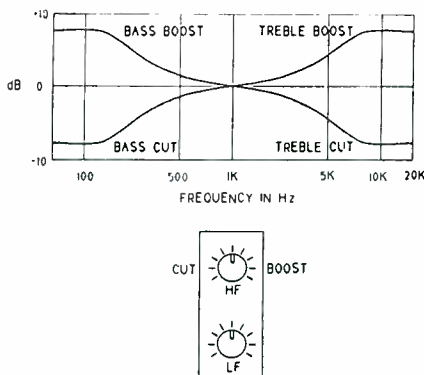
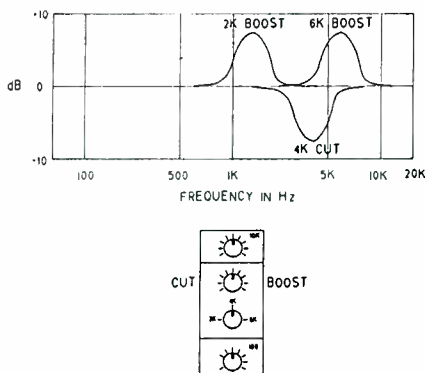


Figure 1. Bass and treble EQ.

Figure 2. Multiple frequency EQ.



When the controls are adjusted, their positions graphically indicate the resulting frequency response. Usually, a graphic equalizer is used for monitor-speaker EQ.

So far we've classified equalizers according to the frequency bands they control, but they can also be classified by the shape of their frequency response. A *peaking equalizer* (see *Figure 6*) creates a response in the shape of a hill or peak when set for a boost. With a *shelving equalizer*, the shape of the frequency response resembles a shelf, as in *Figure 7*.

A *filter* causes a roll-off at the frequency extremes. It sharply rejects (attenuates) frequencies above or below a certain frequency. *Figure 8* shows three types of filters: low-pass, highpass and bandpass.

For example, a 10 kHz lowpass filter (high-cut filter) removes frequencies above 10 kHz. Its response is down 3 dB at 10 kHz and more above that. This reduces hiss-type noise without affecting tone quality as much as a gradual treble roll-off would. A 100 Hz highpass filter (low-cut filter) attenuates frequencies below 100 Hz. Its response is down 3 dB at 100 Hz and more below that. This removes low-pitched noises such as room rumble, microphone handling noise and mic breath pops. Finally, a 1 kHz bandpass filter attenuates frequencies above and below a frequency band centered at 1 kHz.

The crossover filter in most monitor speakers consists of lowpass, highpass and bandpass filters that send the lows to the woofer, mids to the midrange, and highs to the tweeter.

A filter is named for the steepness of its roll-off: 6 dB per octave (first-order), 12 dB/octave (second-order),

der), 18 dB/octave (3rd order) and so on.

## HOW YOU USE EQ

If your mixer has bass and treble controls, their frequencies are pre-set (usually at 100 Hz and 10 kHz). Set the EQ knob at 0 to have no effect ("flat" setting). Turn it clockwise for a boost, counterclockwise for a cut. If your mixer has multiple-frequency EQ or sweepable EQ, one knob sets the frequency range while another sets the amount of boost or cut.

Table 1 shows the fundamentals and harmonics of musical instruments and voices. For any particular instrument, turn up the lower end of the fundamentals for warmth and fullness. Turn down the fundamentals if the tone is too bassy or tubby. Turn up the harmonics for presence and definition; turn down the harmonics if the tone is too harsh or sizzly. Avoid excessive boost because it can distort the signal. Try cutting the lows instead of boosting the highs.

Here are some suggested frequencies to tweak for specific instruments. If you want the effects described below, apply boost. If you don't, apply cut.

**Bass:** Full and deep at 60 Hz, growl at 600 Hz, presence at 2.5 kHz, string noise at 3 kHz and up.

**Electric guitar:** Thumpy at 60 Hz, full at 100 Hz, honky at 600 Hz, presence at 2-3 kHz, sizzly and raspy above 6 kHz.

**Drums:** Full at 100 Hz, wooly at 250-600 Hz, trashy at 1-3 kHz, attack at 5 kHz, sizzly and crisp at 10 kHz.

**Kick drum:** Full and powerful below 60 Hz, papery at 300-800 Hz (cut at 400-600 Hz for better tone), click or attack at 2-6 kHz.

**Sax:** Warm at 500 Hz, harsh at 3 kHz, key noise above 10 kHz.

**Acoustic guitar:** Full or thumpy at 80 Hz, presence at 5 kHz, pick noise above 10 kHz.

**Voice:** Full at 100-150 Hz (males), full at 200-250 Hz (females), honky or nasal at 500 Hz-1 kHz, presence at 5 kHz, sibilance ("s" sounds) above 6 kHz.

## SETTING EQ BY EAR

You can set an equalizer by ear as well as by knowing the frequency ranges of an instrument. One way

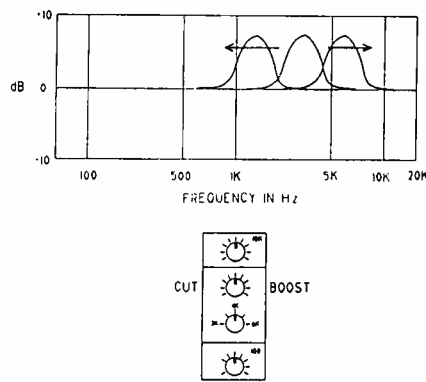


Figure 3. Sweepable EQ.

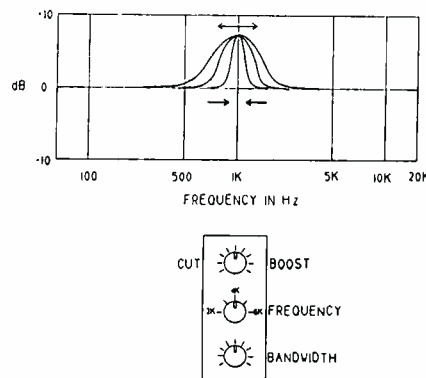


Figure 4. Parametric EQ.

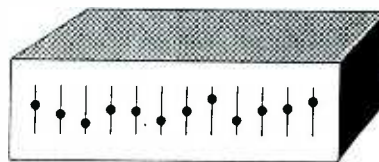


Figure 5. A typical graphic equalizer.

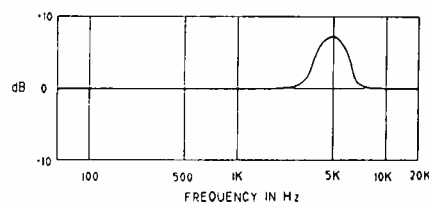
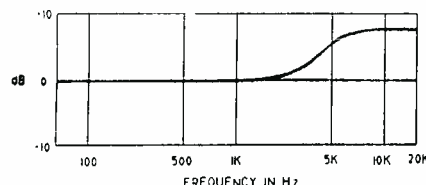


Figure 6. A peaking EQ.

Figure 7. Shelving EQ.



is to tune the equalizer to the approximate frequency range you need to work on (you'll soon know where by experience). Then apply full boost or cut so the effect is easily audible. Finally, fine-tune the frequency and amount of boost or cut until the tonal balance is the way you like it.

For example, if a close-mic'd vocal sounds unnaturally bassy, reach for the low-frequency EQ (say, 100 Hz) and turn it down, adjusting the cut for the desired tonal balance.

If you hear a coloration in the tone quality of an instrument, set a sweepable equalizer for extreme boost. Then sweep the frequencies until you find the frequency range matching the coloration. Cut that range by the amount that sounds right. For example, a piano mic'd with the lid closed might have a tubby coloration—say, excessive output around 300 Hz. Set your low-frequency EQ for boost, and vary the center frequency until the tubbiness is exaggerated. Then cut at that frequency until the piano sounds natural.

## WHEN TO USE EQ

Before using EQ, try to get the desired tone quality by changing the mic or its placement. This gives a more natural effect than EQ.

Should you apply EQ during recording or mixdown? If you mix more than one instrument to the same track, you can't EQ them independently during mixdown unless their frequency ranges are far apart. Suppose a recorded track contains lead guitar and vocals. If you add a midrange boost to the guitar, you'll also hear it on the vocals. The only way around this is to EQ the lead guitar independently when it's recorded.

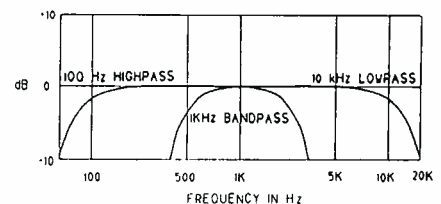


Figure 8. Bandpass EQ.

If a track contains bass and cymbals, you can EQ the low end of the bass without affecting the cymbals because the bass produces mostly low frequencies, while the cymbals produce mostly high frequencies.

If you assign each instrument to its own track, the usual practice is to record flat (without EQ) and then equalize the track during mixdown. I record with EQ, even when multi-tracking. Why? The monitor mixer in my board has no EQ, so when I play back the multi-track recording through the monitor mixer, it doesn't sound right unless the tracks are already equalized. Later, when I start to mix down, the tracks already sound good and need little tweaking.

If you're using a bass cut or treble boost, you can get a better signal-to-noise ratio by applying this EQ during recording, rather than during mixdown. But if the EQ used is a treble cut, applying it during mixdown will reduce tape hiss.

### USES OF EQ

Here are some applications where EQ comes in handy.

- **Improving tone quality.** This is the main use of EQ. It can make an instrument sound better tonally. For example, you might use a high-frequency roll-off on a singer to reduce sibilance, or on a direct-recorded electric guitar to take the "edge" off the sound. As another example, boosting 100 Hz on a floor

tom gives a fuller sound, or cutting around 250 Hz on a bass guitar aids clarity. The frequency response and placement of each mic affects tone quality as well.

- **Special production effects.** Extreme EQ reduces fidelity, but it can also make interesting sound effects. Sharply rolling off the lows and highs on a voice, for instance, gives it a "telephone" sound. A 1 kHz bandpass filter does the same thing. An extreme boost at 5 kHz can accent the impact of a snare drum.

- **Reducing noise and leakage.** You can reduce unwanted low-frequency sounds—bass leakage, air-conditioner rumble, mic stand thumps—by turning down low frequencies below the range of the instrument you're recording. For example, a fiddle's lowest frequency is about 200 Hz, so you'd set the equalizer's frequency range to 40 or 60 Hz and apply cut. This roll-off won't change the fiddle's tone quality, because the roll-off is below the range of frequencies that the fiddle produces.

Similarly, a kick drum has little or no output above 5 kHz, so you can filter out highs above 5 kHz on the kick drum to reduce cymbal

leakage. If this filtering is done during mixdown, it will also reduce tape hiss.

Filtering out frequencies below 100 Hz on most instruments reduces air-conditioning rumble and muddy bass.

- **Remixing mono tracks.** EQ can actually change the mix between instruments within a single track. To illustrate this, suppose you have an old mono jazz recording of bass, drums and sax. Let's say you want to remove everything but the sax solo, and overdub new bass and drum tracks with a contemporary, bright sound.

Here's how. Filter out the lows and highs on the original recording to remove bass and cymbals. You're left with the midrange, which is mainly sax. Copy this sax recording to one track on your multi-track recorder. Now overdub bass and drums on other tracks in sync with the sax solo.

This trick was used in the motion picture *Bird* to produce a high-fidelity sound track from Charlie Parker's original recordings. A contemporary studio bass player and drummer played along with Parker's original sax solos, which were gleaned from Parker's records by filtering out the original bass and drums.

- **Compensating for the Fletcher-Munson effect.** As discovered by Fletcher and Munson, the ear is less sensitive to bass and treble at low volumes than at high volumes. So, when you record a very loud instrument and play it back at a lower level, it might lack bass and treble. To restore these, you may need to boost the lows (around 100 Hz) and the highs (around 4 kHz) when recording loud rock groups. The louder the group, the more boost is needed. As an alternative, use cardioid mics with proximity effect (for bass boost) and a presence peak (for treble boost).

- **Making a pleasing blend.** When several instruments are heard together, they sometimes "crowd" or overlap each other in the frequency spectrum. That is, it may be difficult to distinguish the instruments by tonal differences. By equalizing various instruments at different frequencies, you can make their timbres distinct, which results in a more pleasing blend.

TABLE 1. Frequency ranges of musical instruments and voices.

INSTRUMENT	FUNDAMENTALS	HARMONICS
	Hz	Approx. range
Flute	261-2349	3-8 kHz
Oboe	261-1568	2-12 kHz
Clarinet	165-1568	2-10 kHz
Bassoon	62-587	1-7 kHz
Trumpet	165-988	1-7.5 kHz
French horn	87-880	1-6 kHz
Trombone	73-587	1-7.5 kHz
Tuba	49-587	1-4 kHz
Snare drum	100-200	1-20 kHz
Kick drum	30-147	1-6 kHz
Cymbals	300-587	1-15 kHz
Violin	196-3136	4-15 kHz
Viola	131-1175	2-8.5 kHz
Cello	65-698	1-6.5 kHz
Acou. Bass	41-294	1-5 kHz
Elec. Bass	41-300	1-7 kHz
Acou. Guit.	82-988	1-15 kHz
Elec. Guit.	82-1319	1-3.5 kHz (through amp)
Elec. Guit.	82-1319	1-15 kHz (direct)
Piano	28-4186	5-8 kHz
Soprano	247-1175	2-12 kHz
Alto	175-698	2-12 kHz
Tenor	131-494	1-12 kHz
Bass singer	87-392	1-12 kHz

This procedure also evens out the contribution of each frequency band to the total spectrum, yielding a mix that is tonally well-balanced.

If you have two instruments that sound alike, such as lead guitar and rhythm guitar, you can make them more distinct by equalizing them differently. You might make the lead guitar edgy by boosting 2-3 kHz, and make the rhythm guitar mellow by cutting 2-3 kHz.

The same philosophy applies to bass guitar and kick drum. Since they occupy about the same low-frequency range, they tend to mask or cover each other. To make them distinct, either fatten the bass and thin out the kick a little, or vice versa.

• **Compensating for response deficiencies.** The mics, tape recorder, monitor speakers and the mixing board itself may not have a flat frequency response. EQ can partly compensate for these deficiencies. If a mic has a gradual high-frequency roll-off, for exam-

ple, a high-frequency boost on the console may help restore a flat response. On the other hand, if a mic "dies" above a certain frequency, no amount of boost can help it. Some directional mics have proximity effect—a bass boost when used up close. A bass roll-off on the console can compensate for this boost.

Many purists shun the use of EQ, complaining of excessive phase shift or ringing caused by the equalizer. Instead, they use carefully placed, high quality mics to achieve a natural tonal balance without EQ. The resulting sound is said to be less strained and more natural.

• **Compensating for mic placement.** Often, you must place a mic very close to an instrument to reject background sounds and leakage. Unfortunately, a close-placed mic emphasizes the part of the instrument the mic is near; the tone quality picked up may not be the same as that of the instrument as a whole. EQ can partly compensate for this effect.

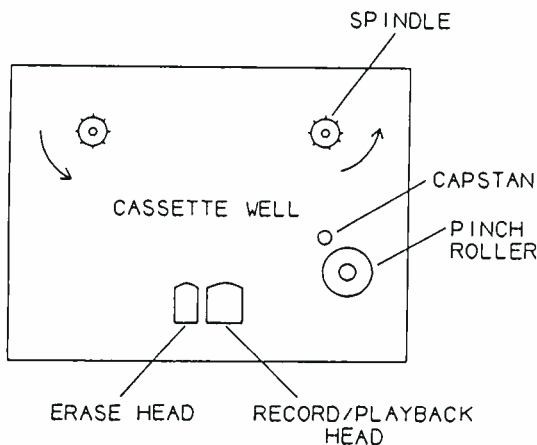
For example, an acoustic guitar picked up with a mic next to the sound hole sounds bassy because the sound hole radiates strong low frequencies, but a complementary low-frequency roll-off on your mixer can restore the natural tonal balance. This use of EQ can save the day by fixing poorly-recorded tracks in live concert recordings. During a concert, the stage monitors might be blaring into your recording/P.A. mics, so you're forced to mic close in order to reject monitor leakage and feedback. This close placement, or the monitor leakage itself, can give the recording an unnatural tone quality. In this case, EQ is the only way to get usable tracks.

You may be fortunate enough to use optimally-placed mics in a great acoustic environment. In that case, EQ is not wanted nor needed, but your recording will sound better with EQ than without it. □

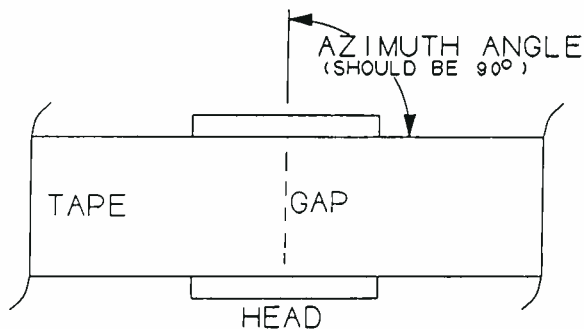
## ERRATA

*In the May/June issue we omitted two illustrations from the Bartletts' column. They are here included. A photo of a cassette multi-track recorder was also omitted and is not reproduced herein.*

*Figure 1. These components are found in the cassette well.*



*Figure 2. Tape head azimuth angle.*



## Designing Vocals: Part III

● Parts I and II of this series on designing vocals have focused on the important foundational principles for capturing an inspired performance on tape—and doing it with all the necessary technical savvy. These are extremely important steps in the process, for a poorly recorded track or a spotty performance will undoubtedly come back to haunt you in the mix. In the case of flawed tracks, it's amazing what a creative and technically competent engineer can do to redeem the song, but it's really very wise to make sure your vocal tracks are solid, both technically and performance-wise before going into a mix. It's like the parable of the house built on solid rock versus the house built on sand: which one do you suppose could survive a storm with its torrential rains and blustering winds?

Mixing can be compared to a storm in that it starts out in utter chaos—all the tracks are separate, competing entities, needing to be placed in proper relationship to each other. Establishing order from chaos is a rigorous task whose outcome can be frustrating or satisfying depending on certain factors. For example, the more well-defined the individual tracks are, the easier it will be to find their proper niche. With vocals, it is best to have dynamics somewhat restricted during the recording process, as it can be rather nightmarish to try placing a vocal in the mix when it is unpredictably loud or soft. When applied during a mix, a compressor or limiter might have to work so hard to compensate for this that the track would end up sounding flat and lifeless. So the importance of starting a mix with good tracks should not be underestimated.

Still, the mix itself is probably the most critical stage in the recording process; it is (proverbially speaking) “where the rubber meets the

road,” for it is at this stage that all relationships between instruments, vocals and the sonic environment are permanently defined. The product of this mixing session is what ultimately reaches the ears of the listener, so an engineer needs to exercise the greatest degree of discernment or else even well-recorded tracks will fail to shine. Good tracks, particularly vocal tracks, can easily get bogged down in a bad mix. If they are dull or harsh, too far under or over the music, or are buried beneath a wave of excessive effects, your brilliant engineering and production will all be in vain. It is therefore extremely important that we carry a high level of technical and aesthetic awareness through the final stages of production. With this in mind, let's take a look at some of the critical factors in mixing vocals.

### CONTROLLING TRANSIENT DYNAMICS

Vocal tracks will (in most cases) have been subjected to some form of dynamics control—limiting, compression or both—while they were being recorded, but at mix time it is frequently necessary to further control the transient dynamics. (When I use the term transient dynamics, I am referring to the short-term peaks that make a track difficult to record and also to place in the mix.) Smooth sounding vocals are definitely a mark of professional-sounding recording. High-end studios have very sophisticated (and expensive) devices which apply both compression and limiting, and assure that the output of the track remains at a steady level—irrespective of the input. If you, like most of us, cannot afford the big-bucks devices, you can still achieve a sophisticated vocal sound by recompressing and/or limiting the track on mixdown.

Whether you'll utilize compression or limiting depends on lots of factors. You'll need to experiment in both modes to see which works best. The safest route in most cases though, is to limit the vocal on recording, so as to preserve the nuances of the performance without overloading the tape; limiting (with the threshold properly set) just lops off the large dynamic peaks in the performance without really affecting the moderate and lower level signals. Then later, upon mixdown, if you really want a much tighter uni-dynamical sound, you can apply some heavy compression to iron out all the wrinkles, or some lighter compression to just tighten it up a bit.

There are many differences between types of compressors/limiters. Some respond to peaks, and some (RMS sensitive devices) respond to an average energy profile, rather than every little blip that passes by. Each type has its strengths and weaknesses and each vocal performance is a unique event, so while you can easily develop some rules of thumb, don't allow yourself to lapse into formula. It's usually best to (as quickly as possible) try several different settings of limiting or compression and see which renders the desired effect. Controlling short-term dynamics is one of the first things that should be done before you attempt to place the vocal track within the mix. When this has been accomplished, you can confidently move on to adjusting EQ and creating a vocal ambience.

### SOME TIPS ON EQUALIZATION

If there is but one axiom you should remember with regard to equalization of vocals, it is this: “Don't overdo it!” Many people equate an excellent vocal sound with a certain shimmering bril-

liance in the high end. This, I am sure, started out as an attempt to duplicate the frequency response of expensive studio microphones—which can accurately capture the high-end harmonics in a voice. But as time rolled on, people became entranced by the magic of high frequencies, no longer to compensate for deficiencies, but for the sheer thrill of coming up with the hottest possible sound. With this quest, of course, came problems—lots of them. When excessively laden with highs, a vocal can quickly become shrill and robbed of its power, and words containing strong *s* sounds (sibilants) become altered into *sh* sounds.

### DE-ESSING

Some of this can be treated after the fact by running the vocal through a de-esser (a fast limiter which is maximally sensitive to a frequency responsible for the sibilance). Although the de-esser can momentarily suppress peaks in the fundamental range of sibilance and fool the ear into thinking it's not there, this device cannot totally undo the mutation of the sound. Some vocalists are more prone to "messy esses" than others because of the anatomy of their mouth, but in most cases, otherwise normal vocalists are made to sound like hissing dragons due to the excesses of an engineer who EQ'd the voice at some dangerous frequency. Often, the problem of sibilance is not picked up until it is too late—when the master is being prepared for duplication. So if there is a moral in this message, it is this: unless you have a real good reason to do so and you're absolutely sure that it won't cause excessive sibilance, avoid positive EQ in the range of 6 k to 9 k. Negative EQ might frequently be used to diminish sibilance, but positive EQ should probably be avoided like the plague!

The next logical question is, how do we get sparkle into the vocal and do it safely? If your mixing console has a high frequency shelf (usually affecting every frequency above 10 k), a small boost here (2-4 dB) will usually do quite nicely. Other consoles have peaking equalizers (affecting a select narrow band) in the high frequency range. In this case, a couple of options are open to you:

a modest boost in the 10 k range will give you a strong dose of very powerful highs, but be conservative here because 10 k is close to 9 k; it is just outside the range of sibilance and if the cue (bandwidth) of the equalizer is wide enough to overlap, you could possibly be augmenting the sibilance, so monitor carefully. Another option preferred by many engineers is to give a liberal dose of positive EQ in a portion of the range most distant from the sibilance, say 15 k, 16 k or higher. The effect here is to boost the weaker high harmonics of the voice. Since the ear is not so sensitive to frequencies up that high, 6 dB or more can be added without any deleterious results.

While we're on the subject of EQ, it's worth noting a few more key frequencies that can really help you place a vocal in the mix. For example, 5 k is usually considered some sort of a magic number when trying to get vocals to cut through in a mix. It is a powerful upper midrange frequency that can really alter the apparent "presence" of a track. In other words, adding 5 k will cause the track to proceed (move forward towards the listener), and subtracting 5 k will cause the track to recede (move away from the listener). This is, of course, only an illusion; it is a psychoacoustical phenomenon based on the fact that human ears are maximally sensitive in the region of 5 kHz. Any track that is exciting the eardrum at that frequency will seem like it's closer. Excesses in this area are to be avoided lest the voice becomes brassy and hornlike.

### OTHER AREAS

Other areas of great power can also be found. There are narrow frequency ranges called "formants" which differ for male and female vocalists, and also between individuals. They are tied in to the anatomy of the throat and head, and resonate at a characteristic fixed frequency irrespective of the pitch of the note being sung. (For a more detailed discussion on formants and EQ, see my article, *The Art Of Equalization: Part 2* in the May/June 1990 edition of **db Magazine**.) You really have to fool around with a good sweepable equalizer to home in on a formant,

but once you hit it, you will find that small amounts of boost in this area will increase the apparent power of the track appreciably. Statistically, there are considered to be two formant areas worthy of your investigation: the low formant, which is roughly around 500 Hz for men and 1 k for women, and the high formant, which is centered around 2.8 k for men and around 3.2 k for women. If you use these figures as starting points and carefully sweep the area below and above, you can usually find some sort of a "hot" area which will prove useful in shaping your vocal sound.

One quick word on background vocals. Getting background vocals to blend with a lead vocal is sometimes a little difficult when both lead and background vocals are full bandwidth tracks. In other words, when both leads and backgrounds have been EQ'd to sound magnificent—with sizzling highs, a tight, powerful midrange and a warm low end—it is possible that their timbres will compete with each other, thereby diminishing the overall clarity of the mix. In that case, something has got to give, and in most cases, it ought to be the background vocals. Usually, by shaping the backgrounds in some sort of complimentary way, things will fit a lot more easily. One of my favorite techniques is radically rolling off both the highs and lows on the backgrounds, leaving the midrange intact. In this way, the backgrounds retain much of their power and fullness, but draw less attention to themselves because they are not full range tracks. Under alternate circumstances, the exact opposite technique can also work quite well: rolling off lots of midrange power, but leaving the high end sizzle and low end woof. The point is to differentiate the lead and the background vocal, and this is usually accomplished by weakening the backgrounds so that they don't dominate such a large region in the frequency spectrum.

One major factor remains to be explored: creating a vocal ambience with echo, early reflections, reverb, chorus, doubling, delay and so on. We will do so in depth when **Designing Vocals** continues in the next issue of **db Magazine**. ☐



# NEW PRODUCTS

## AUTOMATION CONSOLE

- This console has over ten thousand different console configurations, and can be called up with all parameters stored to hard disk and instantly recalled in less than 30 mS. It features MIDI, SMPTE and EBU interfaces along with remote control of external tape machines. There are three input connectors per channel that select or recall forty-eight out of one hundred and forty-four sources at any time. Each output channel, either group or matrix, is fitted with two outputs giving a maximum of ninety-six output points and up to forty-eight outputs simultaneously. A 32-way matrix of inputs/outputs along with sixteen output buses are available so the console can be configured for Front of House, Monitor or Recording. There are also sixteen stereo auxiliary returns with eight VCA control-



led DCA groups.  
Prices: Memory Frame 32  
8/16/s, \$145,000; Memory  
Frame 48 48/16/s, \$340,000

Manufacturer: Infoscene  
Technologie Inc.

Circle 60 on Reader Service Card

## DIGITAL AUTOMATED CONSOLE/8-TRACK RECORDER

- The DMR8 combines an 8-track digital recorder, digital mixer, locator and mixing automation, allowing production facilities, artists, producers and engineers to accomplish all phases of digital audio production in one desktop-sized unit. The DMR8 has 20-bit sound quality comparable to or better than that of compact discs. It is a totally integrated system, permitting the user to accomplish the entire process in the digital domain from recording to mixdown without having to think about repatching, digital format conversions, and the multiple A/D,



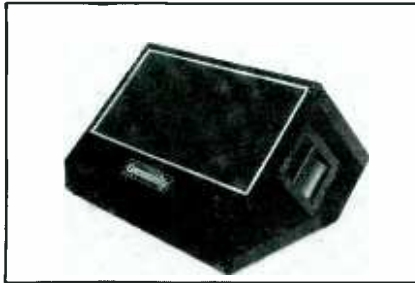
D/A conversions. Of equal importance are the DMR8's automation features. Console setups and real time mixes are stored to either a 64K Ram card or on the tape itself. The DMR8 can digitally memorize, then instantly recall all static control settings—including panning, EQ, track assignments, effects and fader levels. Real time automation of all parameters is also available. Motorized faders indicate channel level changes relative to time code.

*Manufacturer: Yamaha Corporation of America*  
*Price: base, \$34,000.00*

*Circle 61 on Reader Service Card*

### WIDE-RANGE SPEAKER

• The RM220 loudspeaker is operable from 100 Hz to 18 kHz and posts power handling figures of 200 watts pink noise/500 watts program, while maximum output is greater than 127 dB at 1 meter. Drivers incorporated in the loudspeaker's 3-way proprietary Wavefront Coherent design start at the



low end with twin 8 in. ferro-fluid cooled dual-spider woofers. Proceeding up the frequency spectrum to the midrange, each loudspeaker packs a single M-200 compression driver, which employs a one-piece Mylar diaphragm and features a 2 in. exit throat. High frequencies are directed to another compression driver which is equipped with a titanium diaphragm and a 1 in. exit area. Crossover points are at 800 Hz and 3 kHz. A 220 System Controller, a proprietary dedicated system, is

home for the IntelliSense circuitry which provides continuous monitoring and dynamic equalization at impending overload levels and independent compression.

*Manufacturer: Community Light & Sound, Inc.*

*Prices: the RM220 is \$1,400.00 and the 220 System Controller is \$749.00*

*Circle 62 on Reader Service Card*

### NEW CONDENSER

• The CM-230 tridundant condenser microphone's capsules work in conjunction with an interface equipped with three transformer-isolated outputs. Designed for use with either a stand or a standard gooseneck (a permanently mounted female 5/8 in.-27 threaded Atlas flange is provided to accommodate the latter), the CM-230 comes with two shielded cables with 4-pin XLR-type connectors. The typical frequency response of the CM-230

ranges from 80 Hz to 15,000 Hz. Its impedance is 240 ohms (balanced), with 1000 ohms being recommended for the minimum load impedance. Other specifications include a signal-to-noise ratio of 66 dB at 94 dB SPL, a maximum SPL of 120 dB SPL (which produces three percent THD), and a power sensitivity of -56 dB re 1 mW/Pa. Accessories included are a foam windscreen, mic swivel mount, two mic cables and a carrying pouch.

*Manufacturer: Crown International*  
*Price: \$795.00*



*Circle 63 on Reader Service Card*

### MINI MIC

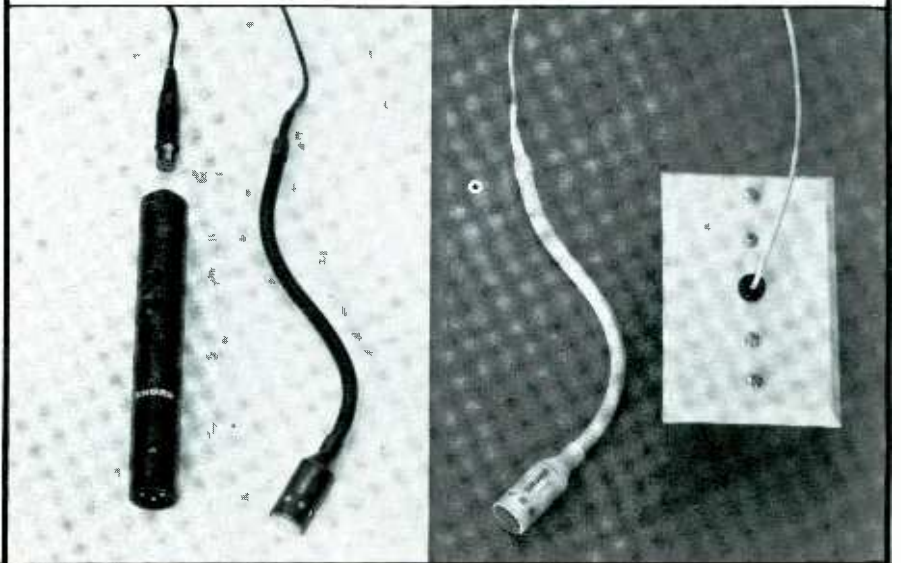
• The SM102 miniature hanging condenser mic has a uniform cardioid polar pattern, low noise, high sensitivity, and smooth, natural frequency response from 40 to 20,000 Hz. Its miniature mic head is permanently mounted to a 6 in. gooseneck, allowing the capsule to be aimed and angled when the assembly is suspended from overhead. A 30-foot attached cable exits from the opposite end of the gooseneck and connects to an included preampli-

fier. Two types of preamps are available: a wall plate version and an inline version. Both contain a low-cut switch, and the wall-plate version will fit in a standard electrical wall box.

*Manufacturer: Shure Brothers, Inc.*

*Price: from \$215.00 to \$260.00, depending upon finish color and preamplifier style*

*Circle 64 on Reader Service Card*



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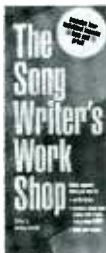
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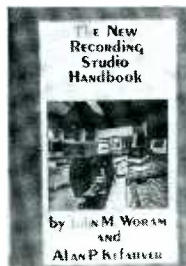
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## LIVE SOUND! by David Scheirman

This excellent video is targeted at first-time users and musicians new to the field of sound reinforcement. However, the video contains insider tips and sophisticated approaches to using the equipment.

The video covers:

- Equipment selection
- Loudspeaker placement and setup
- Mic selection and placement
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- Mixer Position
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- Running the Mixer

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



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

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## CONCERT SOUND AND LIGHTING SYSTEMS by John Vasey

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

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## HANDBOOK FOR SOUND ENGINEERS: THE NEW AUDIO CYCLOPEDIA by Glen Ballou



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

1,400pp. Hardcover

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## BROADCAST SOUND TECHNOLOGY by Michael Talbot-Smith



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

224pp. Hardcover

**\$42.95 #7-991**

## DIGITAL AUDIO OPERATIONS by Francis Rumsey



Leaving the higher levels of theory to other digital audio texts, this handbook emphasizes principles for the studio and those aspects of digital audio appropriate for day-to-day sound engineering operations. It describes the sampling process, error correction, editing systems and different recording options. This book is

written to help producers and engineers in the studio get the best possible results from the high quality standard equipment in use today.

256pp. Hardcover

**\$39.95 #6-991**

## STEREO MICROPHONE TECHNIQUES

by Bruce Bartlett



This book is extremely timely for sound engineers and video or audio producers. Also, as Digital Audio Tape (DAT) production becomes less costly to use in the field, all electronic media will be trying to achieve the highest level sound production possible.

This book tells how to position the correct microphones in the proper locations in order to record optimal quality stereo sound. The many illustrations and clear organization easily explain the theory behind stereo mic'ing methods, and describe specific techniques, including comparative evaluations. In addition, it offers suggestions on session procedures and stereo troubleshooting as well as recent developments in binaural and transaural stereo and stereo boundary arrays.

192pp. Paper

**\$24.95 #3-991**

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



This book is written to assist in the design, purchase, and operation of a sound system. It provides the basic information on sound systems that is most needed by ministers, members of Boards of Trustees and worship and music committees, interested members of congregations, and even employees of musical instrument dealers that sell sound systems.

To be of greatest value to all, it is written to be both nondenominational and "non-brand-name."

183pp. Paper

**\$24.95 #1-991**

**HANDBOOK OF RECORDING ENGINEERING** by John Eargle



This handbook covers the entire range of sound recording and emphasizes the technology of the field as well as the aesthetic aspects of actual recording. It features a sequence of six chapters covering the basic tools of the trade, from the acoustical recording environment through the reproducing environment. Eargle details the actual production decisions

which are made in classical and popular recording. Also covered are the physical and operational principles of tape, disc, and digital recording, as well as the economic and physical aspects of the low-cost studio.

405pp. Hardcover  
**\$66.95 #11-991**

**THE RECORDING SERIES BASIC MULTI-TRACK RECORDING TIPS** with Rick Shaw

Using a "hands on" approach, this video gives you an overview of the basics in setting up a personal studio. Among items covered are: hooking up instruments, connecting equipment,

setting levels, working with the console, doing a recording, overdubs and mixdown. This is a very important video for all persons planning on setting up a studio.

Approximately one hour long  
**\$34.95 #18-791**

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

by Jim Mandell



Here is a comprehensive survey on the state of recording studio business and management in the nineties that includes startup and equipment cost comparisons from low budget to world class operations; equipment purchasing strategies; rate-setting factors; actual examples of pre-session contracts; how different studios

handle billing, credit applications, payment guarantees, conflicts and collections; how to write publicity releases that will get into print; what to avoid in advertising;

336pp. Paper  
**\$29.95 #10-991**

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# db BUYER'S GUIDE

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

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

### CROSSOVERS

#### ALTEC LANSING CORPORATION

Model 1631A is a two-way electronic crossover using plug-in modules to select crossover frequency and configure specific equalization to provide flat power response for various horn/driver combinations. The high-pass output has a level control and the low-pass output has a delay adjustment of 0 to 25 ms.

Dimensions: 1.75 in. X 19 in. X 4.875 in.

Weight: 4.74 lbs.

Price: \$660.00

The 1632A Electronic Dividing Network is a dual channel two-way or single channel three-way active crossover, 24 dB/octave, selectable from 50 Hz to 10 kHz; elect. balanced in/out with xfmr in/out optional; 30/60 Hz HP inputs, hard limiters on all 4 outputs; sub-modules to customize response.

Dimensions: 1.75 in. X 19 in. X 9.75 in.

Weight: 8 lbs.

Price: \$1,150.00

The 15594A Low Pass Crossover/Equalizer Module is a plug-in module for the 9400 series power amplifier; has 18 dB/octave roll-off pre-programmed at 125 Hz, 500 Hz, 800 Hz, 1250 Hz; customer programmable for other frequencies; programmable 12 dB HP roll-off with pre-sets at 16 Hz or 32 Hz.

Dimensions: 1.6 in. X 2 in.

Weight: 1.6 oz.

Price: \$96.00

The 15595A High Pass Crossover/Equalizer Module is a plug-in module for the 9400 series power amplifier; 18 dB/octave roll-off pre-programmed at 125, 315, 500, 800, 1250 Hz; customer programmable for other frequencies; sub-modules available to customize frequency response to horn/driver.

Dimensions: 1.6 in. X 2 in.

Weight: 1.6 oz.

Price: \$96.00

## ASHLY AUDIO INC.

XR22E is stereo two-way 12 dB/octave.

Price: \$379.99

XR77E is stereo three-way, mono four/five way 12 dB/octave.

XR-4000B is a 24 dB/octave, XLR-equipped stereo four-way crossover.

Price: \$769.00

## ARX SYSTEMS —See our ad on page 2

The EC-1 is a low noise Linkwitz-Riley 24 dB Phase correct electronic crossover. It is ideally suited for all studio installation and live sound applications.

Dimensions: 1.75 in. X 19 in. X 6 in.

Weight: 2.5 kg

Price: \$299.00

## BRYSTON/Brystonvermont Ltd.

The 10PBX is a 2-way stereo 3-way mono crossover with 12 switchable turnover points, 3 switchable slopes of 6, 12 +18 dB per octave; balanced input/output; high frequency gain or cut control and mute switches; S/N ratio of -90 dB; distortion of 0.005 percent; 20 k ohm input impedance and output impedance of 100 ohms.

Dimensions: 1.75 in. X 19 in. X 10 in.

Weight: 12 lbs.

Price: \$1,295.00

The 10PBX LR is a 2-way stereo, 3-way mono Linkwitz-Riley slopes with fixed cross-over points; high frequency gain or cut control; S/N ratio of -90 dB; distortion of 0.05 percent; 20 k ohm input impedance and output impedance of 100 ohms.

Dimensions: 1.75 in. X 19 in. X 10 in.

Weight: 12 lbs.

Price: \$1,350.00

## DOD ELECTRONICS CORPORATION

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

Dimensions: 1.75 in. X 19 in. X 6.5 in.

Weight: 3 lbs.

Price: \$475.00

## ELECTRO-VOICE, INC. —See our ad on page 3

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

Dimensions: 1.73 in. X 19 in. X 7.28 in.

Price: \$820.00

## FURMAN SOUND, INC.

The Model TX-324 stereo 2-way/mono 3-way crossover features 24 dB/octave rolloff slopes. Field Select allows optimizing filters for long-throw (Butterworth) or near field (Cauer); hard limiters on each output with adjustable threshold provide speaker protection; includes on/off transient muting; ground lift switch; in/out level controls; limit threshold indicators. Optional balanced configuration.

Dimensions: 1.75 in. X 19 in. X 8 in.

Weight: 7 lbs.

Price: \$419.00

Model TX-424 stereo 3-way/mono 4 or 5-way crossover has features similar to the Model TX-324.

Dimensions: 3.5 in. X 19 in. X 8 in.

Weight: 9 lbs.

Price: \$549.00

Model TX-524 stereo 4-way crossover has features similar to the TX-324.

Dimensions: 3.5 in. X 19 in. X 8 in.

Weight: 9 lbs.

Price: \$679.00

The Model TX-3A is a 12 dB/octave tunable crossover that may be used for either stereo 2-way or mono 3-way applications. Includes calibrated input/output level controls, power indicator and ground lift switch. Optional balanced configuration.

Dimensions: 1.75 in. X 19 in. X 8 in.

Weight: 7 lbs.

Price: \$319.00

## LT SOUND

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

Price: \$295.00



## PANASONIC PRO AUDIO SYSTEMS

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

Dimensions: 1.75 in. X 19 in. X 7.875 in.

Weight: 6 lbs.

Price: \$275.00

## PEAVEY ELECTRONICS CORPORATION —See our ad on Cover II

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

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

Weight: 8 lbs.

Price: \$399.99

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

Price: \$799.99

## RANE CORPORATION

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

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

Prices: \$389.00 and \$499.00

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

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

Weight: 4 lbs.

Price: \$339.00

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

Price: \$449.00

## SYMETRIX

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

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

Weight: 10 lbs.

Price: \$1,095.00

## WHITE INSTRUMENTS

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

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

Weight: 9 lbs.

Price: \$3,400.00

## DELAYS

### EVENTIDE, INC. —See our ad on page 5

The PD860 precision delay features stereo; 20 kHz frequency response; delay adjustable from 15 milliseconds to 5.24438 seconds; adjustable in microsecond increments.

Dimensions: 1.75 in. X 19 in. X 12.5 in.

Weight: 8 lbs.

Price: \$2,695.00

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

Dimensions: 1.75 in. X 19 in. X 12.5 in.

Weight: 15 lbs.

Price: \$5,495.00

The BD955 broadcast delay features mono; 15 kHz frequency response; variable delay; dump and catch-up functions.

Dimensions: 1.75 in. X 19 in. X 12.5 in.

Weight: 10 lbs.

Prices: \$3,360.00 (3.2 seconds maximum) or \$4,300.00 (6.4 seconds maximum)

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

Dimensions: 1.75 in. X 19 in. X 9.4 in.

Weight: 5.5 lbs.

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

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

Dimensions: 1.75 in. X 19 in. X 9.4 in.

Weight: 5.6 lbs.

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

### **KLARK-TEKNIK ELECTRONICS, INC.**

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

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

Weight: 5.5 lbs.

Price: \$1,625.00

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

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

Weight: 5.5 lbs.

Price: \$3,500.00

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

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

Weight: 5.5 lbs.

Price: \$3,900.00

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

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

Weight: 5.5 lbs.

Price: \$3,900.00

### **LEXICON, INC.**

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

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

Weight: 12 lbs.

Price: \$1,050.00

### **PANASONIC PRO AUDIO SYSTEMS**

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

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

Weight: 19.5 lbs.

Price: \$4,500.00

### **ROLAND PRO AUDIO/VIDEO GROUP**

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

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

Weight: 11 lbs.

Price: \$1,095.00

### **SOUND CONCEPTS, INC.**

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

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

Price: \$975.00

### **WHITE INSTRUMENTS**

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

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

Weight: 9 lbs.

Price: \$3,400.00

# EQUALIZERS

## ALESIS STUDIO ELECTRONICS

The MEQ-230 Precision Equalizer has dual 30 band,  $\frac{1}{3}$  octave EQ in single 19 in. rack space; interface provided by means of  $\frac{1}{4}$  -in. and RCA jacks; center frequencies range from 25 Hz to 20 kHz and are set to ANSI/ISO standards; each band provides 12 dB cut/boost; in/out switch.

Dimensions: 1.75 in. X 19 in. X 4 in.

Weight: 2.5 lbs.

Price: \$249.00

## ALTEC LANSING CORPORATION

The 8558B Programmable Microaudio Equalizer offers eight memories; only one rack space; no front panel controls; 28  $\frac{1}{3}$  octave filters with 12 dB of cut/boost; fixed HP/LP filters; elect. balanced in/out; xfmr in/out optional barrier strip only.

Dimensions: 1.75 in. X 19 in. X 7 in.

Weight: 5.9 lbs.

Price: \$1,320.00

The 1750A Cut-Only  $\frac{1}{3}$  Octave Mono Equalizer has 28 constant-Q filters from 31.5 Hz to 16 kHz; 15 dB of attenuation per filter; 20 dB of broadband gain; variable HP/LP filters; elect. balanced in/out with optional xfmr, XLR and barrier strip.

Dimensions: 3.5 in. X 19 in. X 9.75 in.

Weight: 10.7 lbs.

Price: \$1,200.00

The 1753A Boost-Cut  $\frac{1}{3}$  Octave Mono Equalizer has 28 constant-Q filters from 31.5 Hz to 16 kHz; 12 dB cut/boost per filter; 20 dB broadband gain; variable HP/LP filters; elect. balanced in/out with optional xfmr, XLR and barrier strip.

Dimensions: 3.5 in. X 19 in. X 9.75 in.

Weight: 10.7 lbs.

Price: \$1,200.00

## APPLIED RESEARCH AND TECHNOLOGY —See our ad on Cover III

The HD 31, Model 350 is an active balanced  $\frac{1}{3}$  octave 31 band equalizer featuring constant Q filters; 60mm sliders; switchable 15 and 7.5 dB level scale; switchable subsonic and ultrasonic filters; hard bypass at no power and S/N of 115 dB.

Dimensions: 3 in. X 19 in. X 6.25 in.

Weight: 8 lbs.

Price: \$425.00

The HD 15, Model 340 is an active balanced  $\frac{2}{3}$  octave 15 band equalizer with constant Q filters; 60mm sliders; optional XLR connections; switchable subsonic and ultrasonic filters; hard bypass at no power and S/N of 115 dB.

Dimensions: 3 in. X 19 in. X 6.25 in.

Weight: 8 lbs.

Price: \$425.00

## ARX SYSTEMS —See our ad on page 2

The EQ 30 and 60 are ultra low noise constant Q  $\frac{1}{3}$  octave graphic equalizers featuring balanced XLR and jack inputs/outputs and switchable  $\pm 6$  dB or 15 dB of cut/boost.

Dimensions: The EQ 30 is 3.5 in. X 19 in. X 10 in.

The EQ 60 is 5.25 in. X 19 in. X 10 in.

Weight: The EQ 30 weighs 3 kg

The EQ 60 weighs 4.5 kg

Prices: The EQ 30 is \$899.00

The EQ 60 is \$1,349.00

The Multi Q is a six channel/band fully variable parametric EQ. Featuring ARX Auto Patch, the Multi Q allows the user to select any number of channels without the need for patch cables.

Dimensions: 1.75 in. X 19 in. X 6 in.

Weight: 2.5 kg

Price: \$688.00

## ASHLY AUDIO INC.

GQ-215 is a stereo 15-band  $\frac{2}{3}$  octave graphic equalizer.

Price: \$549.99

GQ-131 is mono with 31 bands of  $\frac{1}{3}$  graphic eq. There is a stereo version, model GQ-231.

Price: GQ-131-\$599.99, GQ-231-\$1,099.99.

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

The 905 three-band parametric equalizer features instant before/after comparisons available by switch bypass; symmetrical peak/dip; and switchable notch mode on each band.

Dimensions: 5.25 in. X 1.5 in. X 9.5 in.

Weight: 0.75 lbs.

Price: \$499.00

The 1531X graphic equalizer has selectable 15 band stereo ( $\frac{2}{3}$  octave) or 31 band mono ( $\frac{1}{3}$  octave) equalizer on ISO centers; constant-Q and symmetrical peak/dip curves with selectable 7.5 or 15 boost or cut; and switchable HP filtering at 20 Hz, 60 Hz or 120 Hz.

Price: \$419.00

## DOD ELECTRONICS CORPORATION

The DigiTech MEQ 28 Mono 28-band MIDI Programmable Graphic EQ is a two space, rack-mount mono graphic EQ that is fully MIDI controllable and programmable with 99 user-definable programs. It features 28 bands of 12 dB cut/boost equalization.

Dimensions: 3.5 in. X 19 in. X 8.5 in.

Weight: 7 lbs.

Price: \$569.95

The DOD 830 Stereo 15 band per channel,  $\frac{2}{3}$  Octave Graphic EQ is a two rack space EQ featuring 20 Hz to 20 kHz equalization; 12 dB cut/boost; low cut filter; 90 dB S/N; THD 0.006 percent and 5 percent frequency tolerance.

Dimensions: 3.5 in. X 19 in. X 8.5 in.

Weight: 7 lbs.

Price: \$319.95

## ELECTRO-VOICE, INC. —See our ad on page 3

Model 2710  $\frac{1}{3}$  octave graphic EQ features 27-band,  $\frac{1}{3}$ -octave equalizer; constant range variable-Q filters; minimal interference between adjacent filters; user-selectable high- and low-pass filters; built-in pink-noise generator for noise masking; system equalization and other applications.

Dimensions: 3.5 in. X 19 in. X 10.25 in.

Weight: 11.5 lbs.

Price: \$1,130.00

## FURMAN SOUND, INC.

Model GQ-31 is a 31-band single rack space graphic equalizer. Design results in extremely low noise, even with large amounts of boost or cut. Features include  $\pm 12$  dB of equalization; gain control; LED indicators for overload; EQ in, and power, as well as Loc Cut button and ground lift switch. Optional balanced configuration.

Dimensions: 1.75 in. X 19 in. X 8 in.

Weight: 6 lbs.

Price: \$369.00

The Model GQ-15 stereo graphic equalizer is the same as model GQ-31, except it has two channels, each with 15 bands spaced at  $\frac{2}{3}$  octave intervals. Single rack unit height.

Dimensions: 1.75 in. X 19 in. X 8 in.

Weight: 6 lbs.

Price: \$379.00

The Model GQ-62 stereo 31-band graphic equalizer is the same as model GQ-31, except it has two complete 31-band channels in one double-height rack chassis.

Dimensions: 3.5 in. X 19 in. X 8 in.

Weight: 10 lbs.

Price: \$699.00

The Model PQ-4 parametric equalizer has constant-Q equalization curves; peak/shelf switches on top and bottom bands; extra wide range of bandwidth and EQ adjustment. Includes input level control; EQ in button, as well as overload; EQ status and power indicators; high and low level inputs/outputs; and footswitch jack, allowing use as a preamp. Balanced configuration is optional.

Dimensions: 1.75 in. X 19 in. X 8 in.

Weight: 6 lbs.

Price: \$379.00

## KLARK-TEKNIK ELECTRONICS, INC.

The DN410 is a dual (5) band/Single (10) band parametric equalizer with 100 percent frequency overlap on all bands; +15/-25 dB boost/cut;  $\frac{1}{12}$  to 2 octave bandwidth; separate variable high/low pass filters (each channel); separate EQ in/out switch on all bands plus overall noise less than -94 dBm; 20 Hz-20 kHz, unweighted.

Dimensions: 3.5 in. X 19 in. X 9.25 in.

Weight: 10 lbs.

Price: \$1,195.00

The DN405 is the same as above, but with single (5) band only.

Dimensions: 1.75 in. X 19 in. X 9.25 in.

Weight: 7.7 lbs.

Price: \$775.00

The DN360 is a dual channel 30 band  $\frac{1}{3}$  octave graphic equalizer with switchable 12 dB/6 dB scale on faders; switchable high pass filters; electronically balanced inputs, unbalanced outputs; transformer balancing optional; noise less than 90 dBm; 20 Hz-20 kHz unweighted.

Dimensions: 5.25 in. X 19 in. X 8 in.

Weight: 10 lbs.

Price: \$1,795.00

The DN300 is a single channel 30 band  $\frac{1}{3}$  octave equalizer with continuously variable high and low pass filters; switchable 12 dB/6 dB fader scale; noise less than 90 dBm; 20 Hz-20 kHz unweighted; electronically balanced input; unbalanced output; transformer balancing optional.

Dimensions: 3.5 in. X 19 in. X 8 in.

Weight: 7.7 lbs.

Price: \$1,150.00

The DN301 is a single channel 30 band  $\frac{1}{3}$  octave Cut only graphic equalizer with continuously variable high and low pass filters; switchable 12 dB/6 dB fader scale; electronically balanced input; unbalanced output; transformer balancing optional; noise less than 94 dBm; 20 Hz-20 kHz unweighted.

Dimensions: 3.5 in. X 19 in. X 8 in.

Weight: 7.7 lbs.

Price: \$1,150.00

The DN332 is a dual 16 band  $\frac{2}{3}$  octave graphic equalizer with +12 dB boost/cut; switchable high pass filters; electronically balanced inputs; unbalanced outputs; transformer balancing optional; noise less than -90 dB; 20 Hz-20 kHz unweighted.

Dimensions: 3.5 in. X 19 in. X 8 in.

Weight: 7.7 lbs.

Price: \$1,095.00

### LT SOUND

The PEQ is a dual-channel, 4-band parametric equalizer with selectable peak/dip or shelving response on upper or lower bands, overall hard-wire bypass and individual bypass on middle 2 bands. Bandwidth variable from 0.15 to 2 octaves.

Price: \$595.00

The PEQ-1 is a single-channel version of the PEQ-2. Utilizes a single-rack space.

Price: \$349.00

### ORBAN, A DIVISION OF AKG ACOUSTICS, INC.

The Model 642B dual channel/stereo is a fully parametric equalizer with 4 bands per channel, switchable to 8 channels mono; each band with separate bypass, Q, frequency and fine tuning control; high pass and low pass filters per channel; minimum 40 dB notch per channel.

Dimensions: 3.5 in. X 19 in. X 11.25 in.

Price: \$1,200.00

The models 672A/674A mono/stereo 8-band graphic parametric equalizers have long throw faders controlling boost and cut for each band; high pass and low pass filters with separate outputs for use as 2-way crossover.

Dimensions: 3.5 in. X 19 in. X 5.25 in.

Prices: \$725.00 for the 672A

\$1,525.00 for the 674A

Model 764A features programmable, digitally-controlled parametric equalizer version of the 642B; controls up to 99 channels of masters and slaves; stores up to 99 presets; has four bands, dual channel, with high and low pass filters; and programmable input attenuator.

Dimensions: 3.5 in. X 19 in. X 9.625 in.

Price: starting at \$1,900.00, depending on configuration

### OXMOOR CORPORATION

The DEQ-1 High Resolution programmable  $\frac{1}{3}$  octave equalizer has 29  $\frac{1}{3}$  octave filters adjustable in  $\frac{1}{2}$  octave spacing; 8 presets with security; balanced inputs/outputs; PA-422.

Dimensions: 1.72 in. X 19 in. X 13.5 in.

Weight: 13 lbs.

Price: \$1,060.00

The DEQ-II High Resolution Programmable  $\frac{1}{3}$  octave Equalizer has 29  $\frac{1}{3}$  octave filters adjustable in  $\frac{1}{2}$  dB steps; high/low pass filters selectable on  $\frac{1}{6}$  octave spacing; large LCD display and front panel controls make programming simple; 8 presets with security; balanced inputs/outputs; PA-422.

Dimensions: 3 in. X 19 in. X 13.5 in.

Weight: 15 lbs.

Price: \$1,400.00

### PEAVEY ELECTRONICS CORPORATION —See our ad on Cover II

The AEQ 2800 is an automatic equalizer with up to 12 complete EQ memories; automatic EQ curve fit; 28-band EQ on 3rd octave centers; user friendly;  $\pm 12$  dB in 1 dB steps; 40 X 2 character liquid crystal display; 128 complete EQ program memories.

Price: \$499.99

The PME 4 is a 4-band parametric equalizer with control over 11 octaves via state-variable filters; four bands with calibrated adjustment; 18 dB boost/cut;  $\frac{1}{2}$  to 2 full octave range.

Dimensions: 1.75 in. X 19 in. X 8 in.

Weight: 6 lbs.

Price: \$229.99

The Autograph is programmable with automatic EQs with up to 128 user-selectable program memories; complete with real-time analysis EQ capability;  $\pm 12$  dB in 1 dB steps;  $\pm 6$  dB in 0.5 dB steps; 8 settings; MIDI-controllable sliders; rack mountable.

Dimensions: 1.75 in. X 19 in. X 8 in.

Weight: 7 lbs.

Price: \$549.99

The PME 8 is a stereo version of the PME 4.

Dimensions: 3.5 in. X 19 in. X 8 in.

Weight: 10 lbs.

Price: \$349.99

The PME 4000 is a parametric control over 11 octaves; top and bottom bands switchable (peak to shelving); +4 balanced in and out; 4 parametric bands.

Price: \$349.99

The EQ 215 has two  $\frac{2}{3}$  octave graphic equalizers;  $\pm 6$  or  $\pm 12$  dB ranges; level control; EQ bypass; +24 dBv input and output capability.

Price: \$399.99

The EQ 31 has 31 bands of graphic EQ; 150 centers;  $\pm 6$  or  $\pm 12$  dB ranges; level control; low and high cut filters; +24 dBv input and output.

Price: \$379.99

## RANE CORPORATION

The ME 30 and ME 15 MicroGraphic Equalizers feature constant-Q  $\frac{1}{3}$  and stereo  $\frac{2}{3}$  octave performance in single rack space packaging; with switchable  $\pm 6/12$  dB boost/cut; input level; hard-wire bypass and 20 mm center-detent sliders.

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

Weight: 5 lbs.

Prices: \$359.00 and \$369.00

The GE 27 and GE 14 Graphic Equalizers feature constant-Q  $\frac{1}{3}$  and  $\frac{2}{3}$  octave performance in two rack space packaging; with 45 mm center-detent sliders; level control; hard wire bypass; low noise and low distortion circuitry.

Dimensions: 3.5 in. X 19 in. X 8.5 in.

Weight: 9 lbs.

Prices: \$499.00 for GE 27 and \$529.00 for GE 14

The SP 15 Studio Parametric Equalizer/Notch Filter provides 5 bands, each with 4-octave sweep; bandwidth from 1.5 to 0.03 octave; +12/-15 dB boost/cut; individual bypass; overall bypass and gain control; and fully balanced input/output. Noise and distortion specifications exceed 16-bit digital performance.

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

Weight: 5 lbs.

Price: \$599.00

The FPE 13 Flex Series Parametric Equalizer features three fully parametric full-range bands in a single channel half-rack format. Vertically or horizontally mountable, the unit provides fully balanced three-pin and  $\frac{1}{4}$  in. input/output; exclusive I/O patch point; overall gain and bypass; and bandwidth range from 0.03 to 2 octaves and 10 Hz-20 kHz frequency range for each band.

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

Weight: 4 lbs.

Price: \$289.00

The FME 15 Flex Series MicroGraphic Equalizer is a single channel  $\frac{2}{3}$ -octave Interpolating Constant-Q graphic equalizer with dual boost/cut range switch; input/output level controls; exclusive Patch I/O jack; and fully balanced three-pin, terminal strip and  $\frac{1}{4}$  in. input and output connectors.

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

Weight: 4 lbs.

Price: \$289.00

The MPE SERIES Programmable Equalizers feature the MPE 28  $\frac{1}{3}$  octave and MPE 14 Dual  $\frac{2}{3}$  octave equalizers with 128 memory locations plus a software package that enables curve weighting (adding 2 curves together); real time program changes; remote control; copying; data-dumping, full MIDI mapping and other functions.

Dimensions: 1.75 in. X 19 in. X 8.5 in.

Weight: 6 lbs.

Prices: \$749.00 for the MPE 28 and \$799.00 for the MPE 14.

## SABINE MUSICAL MANUFACTURING COMPANY, INC.

The FBX Feedback Exterminator is a microprocessor-controlled, parametric, filtering device which automatically seeks out and eliminates feedback in sound systems and continuously updates the filters as necessary.

Dimensions: Single space rack mount

Price: \$550.00

## SOUNDCRAFTSMEN

The PRO-EQ 22 C-MOS 0.1 dB Differential/Comparator Octave Equalizer is a two-channel device with 10 octave-wide bands of adjustment for each channel featuring C-MOS Digital Switching; Differential/Comparator 0.1 dB True Unity Gain controls; LED True Unity Gain indicators; EQ defeat totally bypasses equalizer; Pre/post EQ processor loops.

Dimensions: 3.5 in. X 19 in. X 11 in.

Weight: 15 lbs.

Price: \$349.00

The PRO-EQ 44 is a C-MOS 0.1 dB Differential/Comparator Third Octave featuring C-MOS digital switching; two independent channels of EQ;  $\frac{1}{3}$  octave 40 Hz/1 kHz; alternate  $\frac{1}{3}$  octave 1 kHz/16 kHz; exclusive differential/comparator unity-gain circuits; balancing LEDs for instant adjustment to unity gain; pre-post EQ loops and EQ defeat switch.

Dimensions: 3.5 in. X 19 in. X 11 in.

Weight: 15 lbs.

Price: \$549.00

## SUMMIT AUDIO, INC.

The EQP-200A is a dual program equalizer utilizing tube gain make-up stages with 990, balanced output. All units are hand-crafted and burned in for ten days or more.

Dimensions: 3.5 in. X 19 in. X 10.5 in.

Weight: 19 lbs.

Price: \$2,100.00

The EQF-100 Full Range Vacuum Tube Equalizer is a full-range, single channel, four band equalizer with Hi/Lo pass filter section; musically selected center frequencies; with bands one and four peaking of shelving selectable; 990, balanced output.

Dimensions: 3.5 in. X 19 in. X 10.5 in.

Weight: 21 lbs.

Price: \$2,200.00

## SYMETRIX

The SX201 parametric EQ has three overlapping bands; +15 dB boost; -30 dB cut; 0.05 octave to 3.3 octaves bandwidth; 119 dB S/N ratio; 20 Hz to 20 kHz response (+0, -1 dB).

Dimensions: 1.75 in. X 8 in. X 5.5 in.

Weight: 5 lbs.

Price: \$259.00

## WHITE INSTRUMENTS

The Model 4700/4700-2 is a digitally-controlled  $\frac{1}{3}$  octave equalizer; has one or two channel; controllable from the front panel with password protection or software control via RS-232 or PA-422 interface with Pilot 447 software provided.

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

Weight: 9 lbs.

Prices: \$875.00 mono/\$1,375.00 dual

Model 4710 is a digitally-controlled  $\frac{1}{6}$  octave 55 band equalizer in one rack space. Controllable from the front panel with password protection; has 10 memory locations and 10 separate preset locations in non-volatile storage.

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

Weight: 9 lbs.

Price: \$1,550.00

The Model 4650/4660 is a 60 mm slider controlled  $\frac{1}{3}$  octave filters 31.5 Hz-16 kHz;  $\pm 12$  dB, 10 dB gain; variable high/low pass on 4660; XLR and  $\frac{1}{4}$  jack connectors; input/output transformer available (4622).

Dimensions: 3.5 in. X 19 in. X 5 in.

Weight: 7 lbs.

Prices: 4650 is \$699.00/4660 is \$750.00

Model 4675 is a 60mm slider controlled stereo  $\frac{2}{3}$  octave; filters 40 Hz-16 kHz  $\pm 12$  dB range, 10 dB gain; variable high pass, fixed low pass; XLR connections; servo-balanced differential input/output circuit.

Dimensions: 3.5 in. X 19 in. X 5 in.

Weight: 7 lbs.

Price: \$795.00

Model 4400 has L-C active  $\frac{1}{3}$  octave filters 31.5 Hz-16 kHz;  $\pm 10$  dB range; variable high/low pass; 3 outputs and crossover socket for optional bi-amp/tri-amp operation; input/output transformers available; noise -90 dBu worst case.

Dimensions: 3.5 in. X 19 in. X 8 in.

Weight: 15 lbs.

Price: \$1,050.00

The Model 4500 is R-C active  $\frac{1}{3}$  octave filters 31.5 Hz-16 kHz  $\pm 10$  dB range; variable high/low pass; 3 outputs and crossover socket for optional bi-amp/tri-amp operation; input/output transformers available; noise -80 dBu worst case.

Dimensions: 3.5 in. X 19 in. X 5 in.

Weight: 7 lbs.

Price: \$790.00

Model 4100A has L-C active stereo octave band 31.5 Hz-16 kHz  $\pm 10$  dB range; variable high pass; fixed low pass; bi-amp available; input/output isolation transformer available; noise -92 dBu worst case; L.A. approved.

Dimensions: 3.5 in. X 19 in. X 5 in.

Weight: 11 lbs.

Price: \$975.00

The DSP 5000 has 12 bands of parametric equalization, digital crossover and delay all in one rack space; 19 bit, user configurable single channel in, 4 out; remote control capability via PA-422, MIDI or contact closures.

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

Weight: 9 lbs.

Price: \$3,400.00

## MULTI-EFFECTS PROCESSORS

### APPLIED RESEARCH AND TECHNOLOGY —See our ad on Cover III

The SGX-2000, Model 500, is for guitar. Tri-channel programmable tube and solid state preamp with stereo digital effects; full 20 kHz bandwidth; 24 bit processing; seven band equalizer.

Dimensions: 3 in. X 19 in. X 9 in.

Weight: 15 lbs.

Price: \$829.00

The SGX NightBass, Model 490, is for bass guitar. Tri-channel programmable tube and solid state preamp with stereo digital effects; full 20 kHz bandwidth; 24 bit processing; seven band equalizer and selectable crossover.

Dimensions: 3 in. X 19 in. X 9 in.

Weight: 15 lbs.

Price: \$839.00

The Power Plant, Model 410, is a dual channel guitar preamp. Channels are switchable between clean and dirty with their own separate EQ effects loop; separate guitar, line and power amp and headphone outputs.

Dimensions: 1.5 in. X 19 in. X 10 in.

Weight: 11 lbs.

Price: \$329.00

## DOD ELECTRONICS CORPORATION

The DigiTech DSP 256XL Digital Effects Processor features 21 different studio-quality effects, up to 4 simultaneously, and is built tough enough for road use.

Dimensions: 1.75 in. X 19 in. X 8.5 in.

Weight: 5.5 lbs.

Price: \$439.95

The DigiTech DSP 16 Effects Processor contains 128 MIDI changeable programs utilizing 16 different reverb and delay effects; a 3-band EQ provides tailoring of the sound.

Dimensions: 1.75 in. X 19 in. X 8.5 in.

Weight: 4.5 lbs.

Price: \$299.95

## LT SOUND

The ECC is a digital delay system with microplate reverb. Delay and reverb may be used simultaneously or independently; delay range is from 1 ms to 1 s; effects include doubling, chorus, flange, plate reverb with delay, acoustic chamber and tremolo.

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

Price: \$995.00

## PEAVEY ELECTRONICS CORPORATION —See our ad on Cover II

The Ultraverb II is a digital multi-effects processor with 15 kHz bandwidth; 256 internal programs; 128 user editable effects; real time MIDI control; each preset transfer step reversible up to last keystroke; full MIDI access.

Dimensions: 1.75 in. X 19 in. X 6.5 in.

Weight: 6 lbs.

Price: \$349.99

All effects of the AddVerb II, except reverbs and specials, may be modified and stored at any of 199 program presets; full MIDI control capability; 50 reverb presets; 40 programmable delay/echo and modulated presets; 10 combinations; presets may be mapped to any of 128 MIDI program numbers.

Dimensions: 1.75 in. X 19 in. X 8 in.

Weight: 6 lbs.

Price: \$319.00

The Multifex contains four 16-bit, digital, multi-effects modules in one 19 in. rack mount package. Each module delivers user-adjustable echo; pre-delay; early reflections; room size; tonal color; reverb time; left and right stereo channel delay; left and right stereo echo feedback; chorus rate; depth; delay time; feedback and multi-effects algorithm facilities.

Dimensions: 1.75 in. X 19 in. X 9.125 in.

Weight: 7 lbs.

Price: \$1,099.99

ProFex is a programmable MIDI controlled multi-effects preamp featuring digital stereo multi-effects processor; switchable for line level input of instrument level; independent effect blocks can be combined in series or parallel in any order to form multi-effect chains; each effect block has independent mix and level control; programmable noise gate in all programs; 128 presets mapped to 128 programs for front panel, MIDI or footswitch access.

Price: \$799.99

The DSR 1000 is a MIDI capable digital stereo reverb/multi-effect processor; six powerful multi-effect algorithms; all effects re-mappable; full suite of echo/chorus/reverb facilities; 16-bit processing.

Price: \$349.00

The QFX is a 4-channel digital multi-effects processor with full MIDI implementation; 16-bit processing; stereo/mono; 1 I.U. 19 in. rack package; re-mappable effects positioning; up to 2.75 seconds of digital delay available.

Price: \$1,099.00

# REVERBS

## ALESIS STUDIO ELECTRONICS

The Quadraverb Plus features 1.5 seconds of delay memory for sampling; independently adjustable multi-tap delays; programmable panning; new ring modulator and resonator configuration along with the 20 K bandwidth reverb; delay; chorus; flanging; parametric EQ; leslie simulator; and comprehensive onboard digital effects mixing system of the original Quadraverb.

Price: \$499.00

The Microverb III, a 16-bit stereo digital reverb and effects processor, has 256 preset programs: 112 reverbs; 32 gated/reverse reverbs; 80 delays; and 32 multi-tap and effects programs. The 19 in. rack mountable unit features 15 kHz bandwidth and two bands of EQ (100 Hz and 4 kHz) for fine tuning of programs.

Price: \$249.00

The Midiverb III is a digital stereo multi-effects unit capable of generating four effects at a time: delay; reverb; and chorus or flange. Features 200 memory locations, with 100 reserved for factory presets. Real-time MIDI control.

Price: \$399.00

## APPLIED RESEARCH AND TECHNOLOGY —See our ad on Cover III

The Multiverb ALPHA, Model 470, is a 24 bit full 20 kHz digital signal processor capable of combining seven effects at once. Has programmable seven band EQ; reverb; two octave of Pitch Transposing; 20 delay types including sampling.

Dimensions: 1.5 in. X 19 in. X 9.25 in.

Weight: 11 lbs.

Price: \$499.00



The Multiverb LT, Model 420, is a studio digital effects signal processor with instant access to 192 pre-programmed presets of up to three effects at once. Effects include reverb; delay; chorus; flanging; gated and reverse reverb and panning.

Dimensions: 1.5 in. X 19 in. X 9.25 in.

Weight: 10 lbs.

Price: \$299.00

#### **EVENTIDE, INC.** —See our ad on page 5

The H3000KS Kitchen Sink Ultra-Harmonizer has all SE and B features plus Vai Presets and HS322 Internal Sampler Board (23.71 seconds mono/11.35 seconds stereo sampling, pitch shifting and time compression/expansion, more).

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

Weight: 13 lbs.

Price: \$4,590.00

The H3000SE Studio Enhanced Ultra-Harmonizer has 19 algorithms, including vocoder; dense room; multishift; band delay; string modeller; phaser, stutter and patch factory; 200 presets; function generator (programmable parameter modulation); soft functions (user-definable Soft Keys).

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

Weight: 13 lbs.

Price: \$2,995.00

The H3000B Broadcast/Post Ultra-Harmonizer has 14 algorithms, including TimeSqueeze (stereo time compression/expansion with machine control); stutter and patch factory (white noise generator, filters, pitch shifters, delay lines and more); 80 presets; function generator, soft functions.

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

Weight: 13 lbs.

Price: \$2,995.00

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

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

Weight: 13 lbs.

Price: \$2,495.00

#### **KLARK-TEKNIK ELECTRONICS, INC.**

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

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

Weight: 16.5 lbs.

Price: \$2,865.00

#### **LEXICON, INC.**

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

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

Weight: 18.9 lbs.

Price: \$4,795.00

#### **LT SOUND**

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

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

Weight: 7 lbs.

Price: \$695.00

#### **PEAVEY ELECTRONICS CORPORATION** —See our ad on Cover II

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

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

Weight: 5 lbs.

Price: \$249.99

#### **ROLAND PRO AUDIO/VIDEO GROUP**

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

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

Weight: 22 lbs.

Price: \$3,995.00

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

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

Weight: 2 lbs., 10 oz.

Price: \$850.00

## ADDRESSES

**Alesis Studio Electronics**  
3630 Holdrege Avenue  
Los Angeles, CA 90016

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

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

**Applied Research and  
Technology**  
215 Tremont Street  
Rochester, NY 14608

**ARX Systems**  
28271 Bond Way  
Silverado, CA 92676

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

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

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

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

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

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

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

**Lexicon, Inc.**  
10 Beaver Street  
Waltham, MA 02154

**LT Sound**  
7900 LT Parkway  
Lithonia, GA 30058

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

**Oxmoor Corporation**  
2111 Parkway Office Circle  
Birmingham, AL 35244

**Panasonic Pro Audio  
Systems**  
6500 Katella Avenue  
Cypress, CA 90630

**Peavey Electronics  
Corporation**  
711 A Street  
Meridian, MS 39301

**Rane Corporation**  
10802 47th Avenue West  
Everett, WA 98204

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

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

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

**Soundcraftsmen**  
2200 South Ritchey  
Santa Ana, CA 92705

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

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

**White Instruments**  
1514 Ed Bluestein Boulevard  
Austin, TX 78721

# HOT LINE.....

## A Broadcast Audio Question and Answer to Randy Hoffner

*Dear Mr. Hoffner:*

First of all, I would like to thank you for the enlightening articles you have in **db Magazine** under the **Broadcast Audio** column, especially "Multichannel Sound Around the World" which ran in the November/December 1989 issue. Since I am at a radio and television broadcasting station here in Singapore, the articles are particularly relevant to me and my colleagues.

However, here is a query that I would be much obliged if you could shed some light on. We had viewers comment that our station is not "punchier" or "brighter" sounding, and that it has a narrower stereo spread than our neighboring stations. In house, we have two schools of thought to tackle this problem. One group says that including an exciter and/or compressor like the Aphex Aural exciter and the Compellor between the studio and transmitter link is the solution. The other group says that there should be no processors in the above link (at the most, a limiter to prevent an overload), but individual sources/programs (sound balancers or operators in the various areas) should do their own "brightening," compatible level control and imaging. This means a flat unenhanced broadcast chain that is compatible for various types of pro-

grams. What is the conventional or modern broadcast practice?

Another question related to this problem developed. Since we adopted the NICAM 728 stereo system, we have had a mixture of both mono and stereo programs. However, a marked loudness difference between stereo and mono programs was apparent for those viewers having mono sets (non NICAM receivers) and was more than accountable due to "center channel buildup" when stereo was summed into mono and in the reverse, a general drop in loudness when stereo was heard on a mono set. It is very noticeable when commercials of both varieties have to be aired alternately in between breaks.

I hope you can share some views and provide some guidance in solving these problems (which I hope have been presented clearly).

*Jibby Jacob  
Singapore*

### **The Reply From Randy Hoffner:**

*Dear Mr. Jacob:*

Thank you for the kind words about my **db Magazine** articles. Of course, compliments are always appreciated, but it is nice to hear

that my writing is found useful by those in the business. Let me now address your two questions. Please remember that a lot of what is said below falls into the category of personal opinion.

Your first question raises a number of issues, some of which are perennial topics of discussion among radio and television audio engineers. The issue of "punch" or "brightness" in the on-air sound is distinct from stereo spread or separation. It can be dissected into two components. You have raised the question of which type of broadcast chain is preferable: a "flat," unenhanced chain or one that incorporates processing. In the abstract, of course, that depends upon the philosophy of the individual broadcaster. In practice, there is advantage to including processing in the broadcast chain. A big advantage in my opinion is that the broadcast station is much better able to maintain a consistent on-air sound. If all sweetening and processing is done by the individual balancers or operators, every program will have a different "sound," depending on the taste of the individual sound balancer. In addition, of course, it will be much more difficult to assure consistent audio levels from program to program, and from commercial to commercial. It is also

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true that you would be hard-pressed to find a television or radio broadcast station in the United States that does not do some processing in the broadcast chain.

You have not said as much, but if your studio-to-transmitter link is a microwave system, or any system using FM subcarriers to carry the audio signals, it requires at the very least a protection limiter, one that is sensitive to the pre-emphasis curve the system employs, to prevent overdeviation of the FM transmission channels and consequent high frequency distortion. Even if you are not using FM subcarriers in your studio-to-transmitter links, and even though you are transmitting stereo digitally with the NICAM system, your transmitted monophonic signal (the one that most of your viewers are hearing) is itself carried on an FM transmission channel.

While United States television aural transmitters and studio-to-transmitter links use a 75 microsecond pre-emphasis network, producing a boost of 16dB at 15 kHz, your equipment may use 50 microsecond pre-emphasis, but this only reduces the severity of the problems that pre-emphasis can generate and does not eliminate them. Without a frequency-sensitive limiter in front of FM subcarrier stages to protect them from overdeviation, the overall audio level to their inputs must be reduced so dramatically that the resultant signal-to-noise ratio will be compromised.

The use of a compressor in conjunction with the broadcast limiter will serve two functions: it will increase the "punch" of your on-air sound and raise the average audio level over the inherent noise floor of your studio-transmitter link and your monophonic aural transmitter. The choice of which compressor to use is, of course, a matter of taste. You have mentioned the Compellor, which is one of a number of high quality devices available on the market.

If you wish to add brightness to your on-air sound, you would be better advised to use one of the very good split-band audio processors available and increase the high-end balance by judicious adjust-

ment of its control, keeping in mind what effect pre-emphasis will have on the high-frequency content of your audio program material. To put it in a nutshell, I recommend the use of a high-quality broadcast compressor/limiter combination, set up for your pre-emphasis curve, to be placed ahead of the studio-to-transmitter link in your broadcast audio chain.

There should be nothing in your broadcast audio chain that would compromise the stereo separation, which leaves stereo "spread" a function of the audio balancer. Some United States television and FM radio stereo encoders, because they operate in the sum-and-difference domain, have circuitry that increases stereo spread by manipulating the amplitude of the stereo difference signal, but NICAM does not work this way, so that option is not readily available to you.

In response to your second question, the issue of the relative loudness difference between a mono signal and the mono sum of a stereo signal arose when we were contemplating the conversion of the NBC Television Network to stereo operation. This within itself is not really a big problem, as I hope to show. There may, however, be other problems with other causes that generate the phenomena you are describing.

Let us assume that we have two audio paths, which we will call *A* and *B*. If *A* and *B* have identical signals (from the same source) at equal level of 0 dB on them, these two signals are perfectly correlated. This would correspond to a split mono signal. A very simple example is a sine wave from a single oscillator connected to both paths. If these two signals are summed, the resultant signal has an amplitude twice that of either of its components. Stated differently, if *A* and *B* each have an in-phase (correlated) 0 dB signal on them, the sum of *A* and *B* is a signal of +6 dB. If *A* and *B* have totally uncorrelated signals corresponding to maximum stereo separation of amplitude 0 dB on them, their sum will be +3 dB. For typical stereo television program material, the degree of correlation between channels varies from complete correlation to to-

tal lack of correlation. The worst-case difference between two-channel mono and separated stereo is 3 dB, and the typical difference is in fact around 1-2 dB.

When we converted the NBC Network to stereo, the transition was made this way. On a particular day, we started feeding two-channel audio, either stereo or two-channel mono, on two audio subcarriers of our satellite system. By the time that day arrived, all of our mono affiliates (at that time, over 150 stations) had to have put in place a summing network to combine the two audio signals, or else they would be airing only one channel of any stereo program. We had no significant problems with mono versus stereo levels, and that has maintained from 1985 until today.

You have not mentioned how your mono audio signal gets fed to its transmitter. At least some NICAM encoders furnish a mono sum audio output for feeding the mono aural transmitter. If this is the way our mono transmitter is fed, and if the level of that signal into your aural transmitter is set correctly, no substantial difference should be noted between stereo and two-channel mono. Feeding the mono transmitter in this way can complicate things, however, because of the differing limiter requirements of NICAM and FM audio transmission. Other ways of feeding the mono transmitter are possible, and some could cause level problems if not properly set up. For instance, if there is some scheme whereby a switch is made between single-channel mono and stereo sum, the relative levels of those two sources could be disparate. It is also possible that if *extreme* separation is present in stereo segments, this could result in noticeable loudness differences between mono and stereo material.

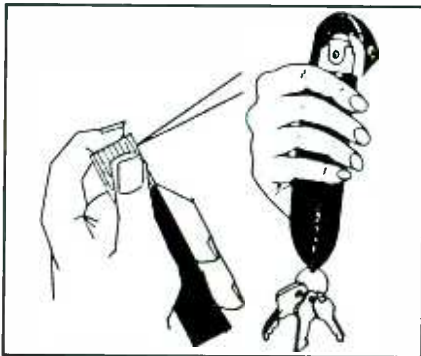
I hope that my views on the questions you have asked will give you some ideas on how to go about solving your audio problems. I would be happy to continue this dialogue if you find it useful. I wish you good fortune in your work, and again, I thank you for your kind words.

*Randy Hoffner*  
NBC-New York

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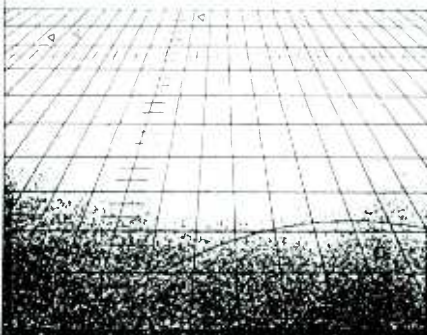
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# PEOPLE, PLACES & HAPPENINGS

- Construction is complete for the **University of Utah's Dolores Dore' Eccles Broadcast Center**. Facility tours of the complex will include visits to the 3,500 sq. ft. television studio for public television station KUED which encompasses two floors, the Audio Post Production Suite, the open plan VTR/Rack Room, control rooms for radio and television, production rooms and suites for radio, television and future Student Broadcast activities, and administrative spaces located on both levels. Unique to the second floor is a viewing window located in the corridor and the full-length window located in the multi-media room both providing unobstructed views into the KUED studio.

- **Sennheiser Electronic Corporation** was awarded the "ART-IST Stage Design Prize" at the recent MUSIKMESSE Fair in Frankfurt, Germany, for their SEMS Microphone stand. Special emphasis was placed on the ergonomic and operation features of the mic when it was designed.

- After being destroyed by fire in July, 1990, the **Music Annex's Studio C** is back in operation. Studio C features a new 56-input Soundcraft 3200 console, with Discmix II automation, a Studer A827 24-track recorder, Otari MX15 2-track, UREI 813C time align monitors and a full complement of outboard gear, including a Lexicon 480L and an Eventide HD3000 SE. The Music Annex's recording studios are located in Menlo Park, CA...Also adding equipment to its studios is **Ron Rose Productions** in Southfield, MI. AudioFile II Plus, which is the

latest generation of hard disc-based digital audio recording and editing equipment, has been installed in Ron Rose Productions' Miami Vice studio. The equipment's internal processing unit is a transputer—the same space-age technology used in F-15 fighter planes—which processes incredible amounts of data in split seconds. Ron Rose Productions is the first company in the metro Detroit area to have this new, updated version...The first multi-console order for Korea has been delivered to **Seoul Broadcasting Systems** in South Korea. Eight Harrison PRO-790s, which range from twelve to twenty-four input channels, have been installed in SBS's new radio facilities in Seoul...Nutmeg Recording in New York City has installed Solid State Logic's ScreenSound digital audio-for-video editing/mixing system for work on its post production projects. The four-studio facility features two 24-track video interlock rooms, a third MIDI room and a fourth room designed around the ScreenSound system...A Versadyne 1500 Series high-speed tape duplication system has been delivered to **Precision Sound Corporation** of Burnaby, British Columbia, Canada. The new system will be used primarily for music duplication. Other Precision Sound purchases include an Otari MTR-12 mastering deck, Dolby encoders, Versadyne SR-150 slave reader, Versadyne PT-250 production totalizer, and an assortment of test equipment.

- A new AutoCAD-compatible acoustic design program entitled **CART (Computerized Acoustic Ray Tracing)** has been developed

by John Storyk in a collaborative effort with Walters-Storyk in-house CAD consultant **Marcolm Young**. CART is an automated process which calculates and graphically displays acoustic ray behavior—specifically, how sound rays of varying frequencies bounce, reflect and re-radiate against the interior surfaces of any given space. The CART program, in conjunction with the TEF System-20 Sound Lab audio analyzer, has contributed to design programs for Storyk projects including Studio 9 at Howard Schwartz Recording in New York City, and new facilities for JSM Music and Sound Shop.

- **Gene Nyland**, vice president of Operations at **Ampex Recording Media Corporation's** Opelika, AL, manufacturing facility, retired Aug. 30 with twenty-seven years of service to Ampex...**Ronald Renschel** has been appointed marketing manager, **Professional Audio Products, Sony Business and Professional Group**...As part of the reorganization of its research and development staff, **Renkus-Heinz Inc.** has appointed **Frank E. Ostrander** as chief engineer...**John Bolstetter** has been named a vice president of **Mark IV Audio, Inc.**, and **Al Watson** has recently been named vice president of engineering at **Electro-Voice, Inc.**, one of seven companies in the Mark IV Audio Group...**Charles Meyer** has been promoted to vice president of engineering at **NVision, Inc.**

He will assume full responsibility for all research and development for NVision's line of digital audio/video distribution and transmission equipment.

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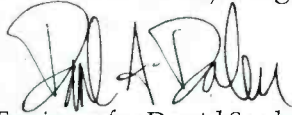
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