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db Magazine

serving: recording, broadcast and sound contracting fields

LAB REPORT: **Crown's Macro Reference Power Amplifier**

SOUND REINFORCEMENT IN THE PARK

Modernizing the Elgin Theatre

db Looks At State-of-the-Art Church Audio

The UniPoint line of miniature microphones that revolutionized church sound pickup…as well as everything from TV game shows to concert recordings…is now better than ever.

New Circuits, New Condenser Capsules

While others try to play catch-up, we've been busy staying well ahead. UniPoint now sounds even better with new circuitry to provide greater headroom. Improved 2nd generation elements offer smooth, extended response, while making polar response even more uniform and predictable.

Better in Every Detail

Further responding to your needs, we've given you more choice of microphone lengths. The electronics module is now more compact and can be powered by a standard 1.5V battery. We've also improved its electrostatic and RF shielding.

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UniPoints are now even harder for audiences to see. Finishes have been refined to reduce glare, and an all-white option is now available for the AT853 (the "choir mike"). An optional electronics package now installs behind a standard electrical plate for simple plug-in operation.

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With the introduction of A-T Wireless systems, you can create a "wireless podium" using the ideal UniPoint microphone for each specific application. And now there are three UniPlate™ boundary microphones available. UniPoint has never been more flexible.

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May/June 1991 Vol. 25, No. 3

db MAGAZINE

serving: recording, broadcast and sound contracting fields

THE SOUND CONTRACTING ENGINEER

AUDIO FOR THE CHURCH

THE RECORDING ENGINEER

THE ELECTRONIC COTTAGE

THE BROADCAST ENGINEER

see page 6

ABOUT THE COVER

• The main sanctuary of the Calvary Church in Charlotte, NC. While it features the world's 13th largest playable pipe organ, it is also home to a sophisticated sound system powered by Crown amplifiers and with a JBL main speaker cluster and satellites. *Photo by Ken Beebe.*

CALENDAR

• "The Practice of Digital Audio," a one-day seminar covering practical implications of digital technology, will be held on May 11 at the University of Southern California's School of Cinema-Television. An introductory session will cover the basic principles of digital audio, including the analog-to-digital conversion process, storage media, data compression, interface formats and editing systems.

Other sessions will focus on levels and monitoring, switching, distribution and interfacing, and time code and sampling rates. There will also be open forum panel discussions.

For more information, please contact Don McCroskey, SMPTE, Hollywood Section, at (818) 846-0918.

• Synergetic Audio Concepts will host three-day audio engineering seminars at the Farm in Southern Indiana on May 16-18, July 25-27, Sept. 19-21 and Oct. 17-19.

For more information, please call (812) 995-8212 or FAX (812) 995-2110.

• The University of Iowa will offer its annual Seminar in Audio Recording June 10-21. Lecture-demonstration topics will include stereophonic and ambisonic microphone techniques, mic technology, digital processing, digital editing, and the preparation of clients' tapes for CD and stereo LP mastering. **Jerry Bruck, Robert Ludwig and Lowell Cross** will be the principal instructors.

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Please contact Lowell Cross at (319) 335-1664 or FAX (319) 335-2777, for more information.

• Courses in recording engineering and sound design are being offered by the UCLA Extension for their spring and summer quarters. *Sound Design for Film and Television* will take place from 10 a.m.-3 p.m. on June 15 at Serafine Incorporated in Venice, CA. **Frank Serafine**, whose credits include Poltergeist, Tron, The Hunt for Red October, and others, is the instructor. Fee is $125.

An introduction to the principles and procedures of recording engineering for the non-technician is the focus of *Introduction to Audio Engineering*, which will take place from 7-10 p.m., June 24-July 22. Fee is $295.

Techniques of Multi-track Mixing for Music Mastering will be offered from 7-11 p.m., June 28-Sept. 11. Students can participate in mixing pre-recorded 24-track master tapes in a variety of musical styles at the Digital Sound Recording Studios. Fee is $950.

Please contact the UCLA Extension at (213) 825-9064 for more information.

• As part of Duquesne University's nationwide Summer Music Institutes, their annual guitar workshop will take place from June 17-21 in St. Paul, MN, and from Aug. 5-9 in Pittsburgh, PA. Class offerings will feature studies in various styles, maintenance, recording techniques and music technology. There will also be live concert and video presentations.

Please contact the Duquesne University School of Music at (412) 434-6080 for registration forms and additional information. db

GREG DETOGNE

A World-Class Church Audio System

Built by the Moller Organ Company, the Calvary Grand Organ took four years to build and was presented to its owners wearing a price tag of $2.5 million. Exactly 11,499 pipes were used in its assembly which range from 40 feet tall to the size of a soda straw, and if that doesn't impress you, just lay an eyeball on the handcrafted Honduran mahogany console.

Bearing a resemblance to the cockpit of a 747, the organ houses five keyboards and a pedal board, 306 draw knobs and approximately 200 other controls. Additionally outfitted with sophisticated electronic controls provided by Solid State Logic, as well as future MIDI capabilities, this grand instrument is used by organist Dan Miller to play everything from fifth century hymns to Southern gospel and contemporary music.

Presently the cornerstone of the musical ministry at the Calvary Church in Charlotte, NC, it almost goes without saying that the Calvary Grand Organ is also the main focal point of the church's 6,000 seat sanctuary, which provides the instrument with 2.5 million cubic feet of air space containing excellent acoustics and a reverberation time of just under three seconds.

THE SOUND SYSTEM

By itself, the pipe organ is a hard act to follow, especially when it comes to constructing a house sound system that has to be aesthetically transparent, yet capable of serving sound reinforcement

Figure 1. Housed in Atlas/Soundolier racks, 26 Crown PS-400 and 15 PS-200 amps send power to the main sanctuary.

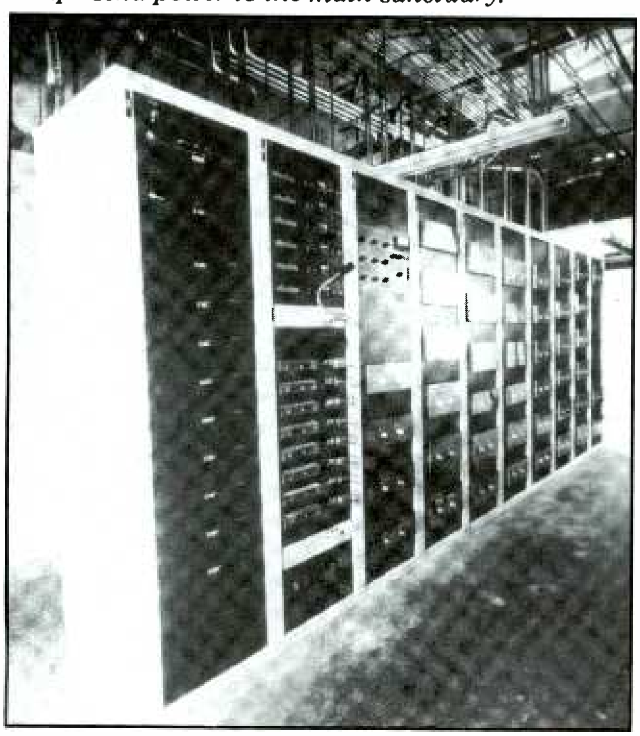
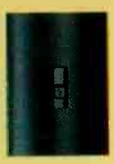


Figure 2. The house engineer has an excellent view of the entire sanctuary including the main cluster and its satellites.

We've added a whole new twist to our LM microphones.

Crown introduces a new low-profile addition to our series of lectern microphones, the LM-300. Like our other lectern microphones, the LM-300 is aesthetically pleasing and unobtrusive. It's built for years of trouble-free use. It installs quickly and easily. And, like our other LM mics, it gives you beautiful, natural sound with superior gain-before-feedback. But there's a twist. Literally.



A low-frequency roll-off switch reduces low-frequency rumble.

The new LM-300 adds a noiseless gooseneck section for more flexible microphone positioning. Now, you can place the microphone exactly where you want it without worrying where it might wander off to.

A noiseless gooseneck improves the positioning capabilities of the LM-300

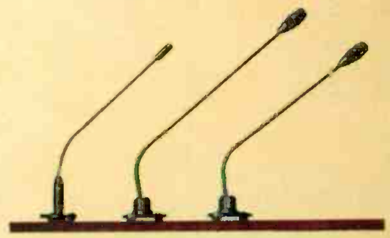
But we didn't stop there.

Along with the new gooseneck, we've provided additional mounting options: a standard 3-pin XLR panel connector, or the accessory *lockable* shock mount for reduced mechanical noise and theft prevention. We've also improved the already excellent characteristics of the microphone element. You'll enjoy smooth, wide-range frequency response, better control of ambience and higher output. A new low-frequency roll-off switch helps reduce low-frequency rumble and "P-popping". Couple this with superior RF supression and 12-48V phantom powering and you'll find the LM series perfect for all your lectern micing problems.

Optional lockable shock mount for the LM-300 reduces mechanical noise.

Like all Crown microphones, the LM series carries a full three-year unconditional warranty against malfunction with a lifetime warranty on the acoustic system.*

For more information on our complete line of microphones, see your Crown dealer or call toll-free: 1-800-535-6289.



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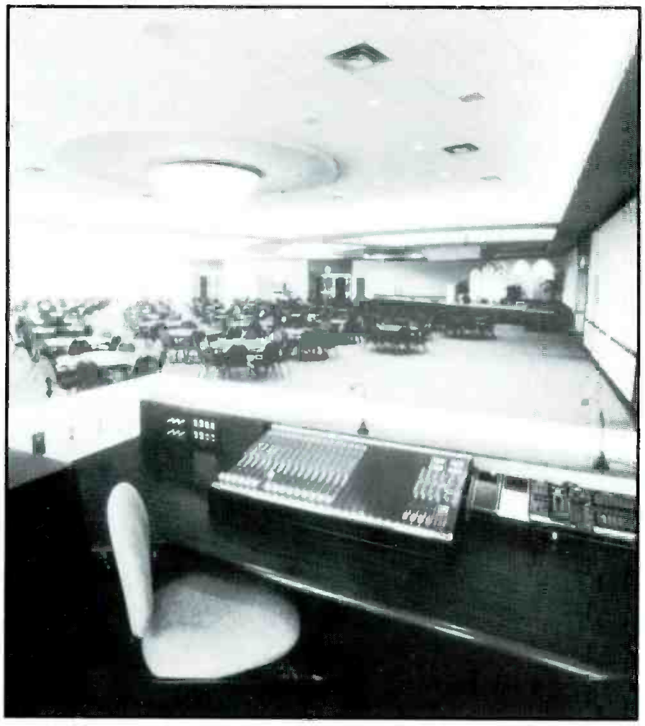


Figure 3. The Crown room has a distributed sound system with front-to-back delay.

Figure 4. Additional loudspeaker arrays are ceiling-mounted in the sanctuary to provide balcony area reinforcement.

needs ranging from speech to full-scale concerts.

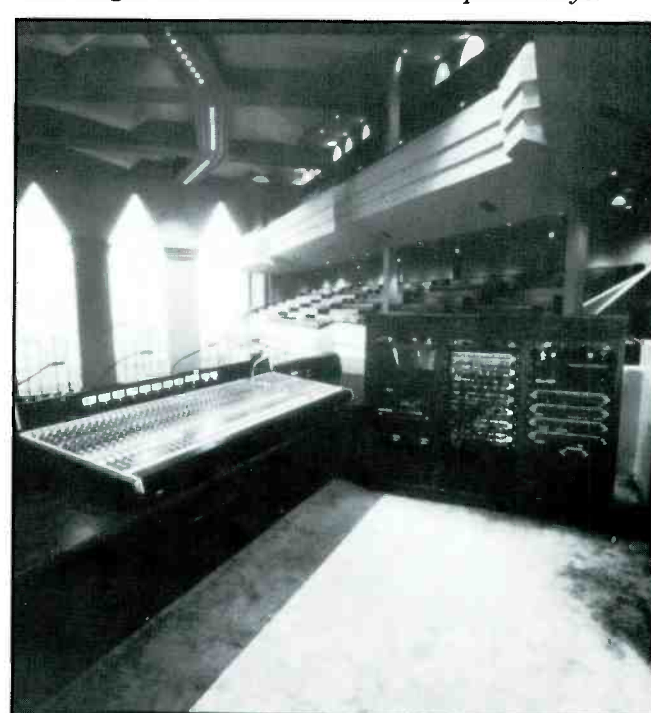
Answering the call to meet this audio challenge was Charlotte's Reliable Music, a sales, contracting and installation firm owned by partners Melvin Cohen and David Bustle.

After accepting the formidable task, Reliable Music came onto the scene during a phase in the church's construction where the general contractors had been changed, and nothing outside of placing a phone call to a consulting group had been done about the sound.

"One look at what was being asked of the sound system, and I realized that it was well beyond the scope and capabilities of most con-

Figure 5. High atop the sanctuary Calvary's recording studio has complete 24-track capabilities.

Figure 6. Another view of the house mix position showing a custom-built cabinet and patch bays.



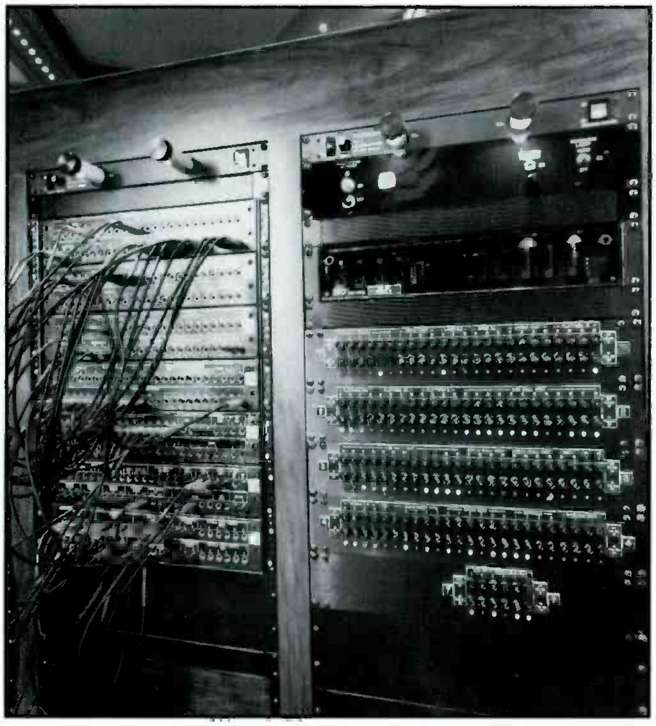


Figure 7. A close-up of the patch bays of Figure 6.

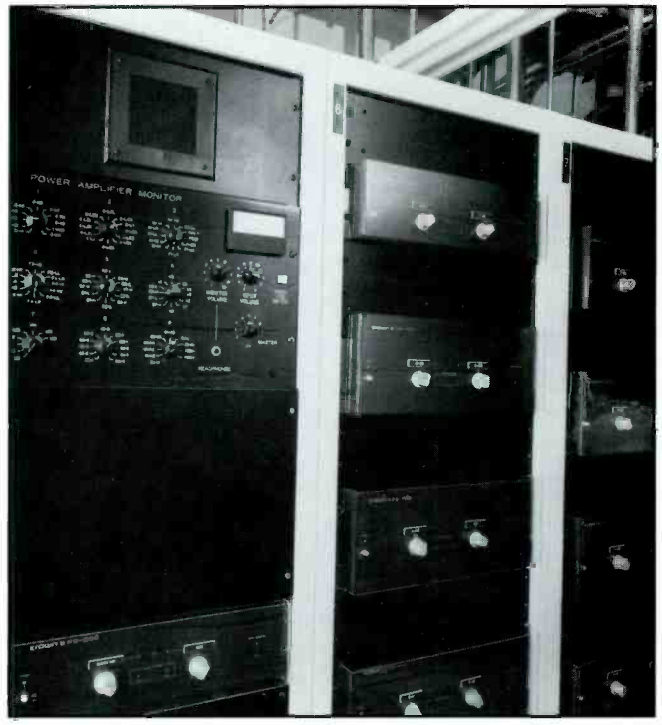


Figure 8. This close-up of the amplifier racks reveals the amplifiers and the monitoring panel.

tractors' design abilities," recalls Don Kendrick, Reliable Music's systems designer. "I strongly suggested that more effort be placed in the direction of finding a good team of consultants."

FOLLOWING THE GOOD ADVICE

Calvary's administrators heeded Kendrick's advice by contracting the services of Lenexa, Kansas-based Coffeen, Fricke, and Associates. Before Bob Coffeen and his designers George Damon and Bob Ledo mapped their plans, Kendrick, Bustle, Calvary Engineer Barry Wood, and Calvary Music Minister David German took stock of their design considerations.

"As we saw it, the system would have to serve three basic areas," Kendrick said. "These were the main sanctuary, a small chapel and a mid-sized room called the Crown Room. Overall, we also felt that the nature of activities in the sanctuary warranted on-site recording capabilities of high professional quality. The recording studio we envisioned would receive almost constant use, most of which would revolve around the recording of the multitude of concerts, symphonic presentations and special holiday events that are held there," he said.

MANY INPUTS NEEDED

Kendrick and Wood additionally felt numerous inputs would be required in the stage area, enough in fact, that four different stage mixes could be obtained. At the house mix position, they envisioned a console which could be utilized in conjunction with a smaller console which would provide additional inputs if the need arose.

On top of these recording needs, tie lines between the three distinct areas of coverage were specified, along with a sanctuary sound system utilizing a series of clusters. Concerns for amplification and electronics centered around simplicity, performance and reliability.

Against this backdrop, Coffeen/Fricke set about in a mindful manner to specify the system in exacting detail. No expense was spared in their quest to deliver an end product which would carry out the stated goals and deliver a system for all three acoustical environments within a budget framework of around $1 million (the project finally came in at a cost of $800,000, with very few changes being made to the original design).

The collection of photographs included, which were taken recently by Charlotte photographer Ken Beebe with the able assistance of Don Kendrick, illustrate the elaborate planning and craftsmanship that went into sound at the Calvary Church. Featuring Crown amplification and JBL loudspeakers, the sanctuary's sound system, like those found in the other two areas, is run at high voltage because of the long distances of the loudspeaker lines. Tie lines connect each individual area, and the use of custom and specially-fabricated components is evident throughout.

Taken as a whole, the Calvary Church represents an execution of sound design principles which bring form and function together, along with elegance and grace. db

JOHN LAWRENCE

Layback Editing

In the March/April issue, we offered a major article on the Vidfilm operation. This is by way of a follow up to that article.

FIRST THING WAITING FOR ME AS I walk into work is an urgent message to call Mike over at Mega-Huge Films' transfer department. Mike is panicked. His night crew has been transferring M & Es from their mag masters onto 4-tracks-with-time-code for me to lay back onto just-completed telecine masters here at Vidfilm.

Apparently, someone forgot to feed video reference to the time code generator and has been making dubs with film running at video speed, but time code printing at 60 Hz. I will now have perfectly-locked time code addresses reeling along like nobody's business, while the picture and the sound drift inexorably, relentlessly, apart.

Bummer. Either I'm going to have to tap a funky rhythm on the slew key of my synchronizer controller all day, making a new edit every three or four minutes, or Mike is going to have to put nine man-hours in the dumper and start over.

Right?

Wrong.

I don't care. I don't have to. I have an Adams-Smith 2600 A/V.

Once upon a time there was *time code*. Pretty simple, really. Just 30 frames every second, timed to 60 Hz, end of discussion. Then people decided they preferred Hoss, Little Joe and Ben in living color. Uh-oh. Enter drop-frame time code. Only 29.97 of those little guys make it out in a second, and just to complicate things, we get 59.94 Hz as a timing reference. And, of course, during all this time those nutty Europeans had to have their own damn standard, so any pal o' 50 Hz is a pain in the ass o' mine. Film rates? Don't get me started.

All these different rates mean that if you want to record audio from one standard onto any other standard, you must jump through various and sundry flaming hoops to stay in sync. For instance, PAL distribution copies are created by dubbing from an NTSC 1-in. master through a converter with Composite English on CH. 1. The audio department will lay back a Music/Efx track on the second channel of the newly created PAL 1-in. submaster so it can be dubbed into foreign languages. Just to do this simple job requires resolving the NTSC-coded, 24-track, sweetened master while transferring the desired audio to a second audio tape that gets striped with PAL time code. Then, and only then, can the show be laid back to the PAL dubs. If you're doing the same jog from mag, or tape created from mag (night shift at Mega-Huge), you'd better hope to have a properly-referenced dub. If not, go directly to the gearbox, do not pass go, start restriping. If you figure roughly double the running time to set up and make the show, you will need two hours of man and machines to complete the layback on a half-hour show. Plus, you've lost a valuable generation, since the thing will be re-recorded several more times before it is ready for broadcast in its new language.

Even more problematic are PAL Telecine masters. These nefarious telecine guys...they don't care, they just run all of their film elements 1 frame per second faster to make 25 frames per second. By the time the poor audio dog gets it, he has to rig up bizarre combinations of gearboxes, generators, flywheel defibulators, etc., to make "slow" 25-frame time code to print on his laydown transfers so the thing will end up running fast enough to stay in sync once it's locked to the master. Rube Goldberg joins SMPTE. Ugh.

The Adams-Smith 2600 A/V I use does some remarkable things that spare me this mind-bending waste of time and machines:

1. As my old dancing partner John Sachetti used to say, "it'll lock the secretaries to the water coolers." First of all, you don't even have to tell this system what kind of codes you want to combine. You just press PLAY (fourth button from the left), and the A/V tells you what code you've got on your tape. Once it knows the type of code it is seeing, the computer can count and synchronize ("Cross-Lock") 30 frames on one source, 25 on another, 24 on another, etc. So for NTSC to PAL conversions, I put the PAL submaster up, reference the system to 50 Hz, put up the 24-track with its 30-frame code, mark my sync points, mark my Record In and start recording. Simple.

2. PAL Telecine masters: I know the A/V can count different frame rates, but in this case the telecine has run the original material 4+ percent faster to make it go at 25 frames. No problem.

If I know the exact amount of vari-speed required, I can enter that value into the source machine's "Super-Sync" register, and from then on, all my marks will take that vari-speed into account. I just find "sync point," which auto-calculates the correct offset, mark Record In, Record Out as normal, and the computer does all the number crunching to keep the machines synchronized throughout. Once that vari-speed value is dialed in, the operator never needs to know.

Of course this is perfect for material that has been transferred at the wrong rate, as in the case of Mike at Mega-Huge. I just dial in + or – 1 percent to compensate for the difference between 60 Hz and 59.94, and away I go.

What about when I don't have any information (most of the time) about what has happened to a show? What do I do with a feature that has been compressed for television, or when somebody has screwed up a transfer and doesn't know or doesn't bother to tell me? What about opening credit music that has been stripped off wild, dubbed into Spanish, and then laid back wild onto another multi-track, so that now any reference or original program sync is shot to hell? This is where the A/V really shines:

I start out with my source material machine in "super-Sync" mode, no values entered, nothing marked (0.000 percent vari-speed). Let's say I find the first EFX door slam on the existing Comp track of my master. I Mark In that spot on the master, and I mark the same spot on the slave machine carrying the M&E I'm laying back. Now I run down to the end of the reel and find another door slam, piece of dialogue, music hit, etc., and mark that spot on my source as a "Sync" point. As I mark that same spot on the master, the A/V instantly calculates the amount of vari-speed it will take to get those two points on the slave perfectly synchronized with the same two points on the master. No matter what weird speed anomaly has occurred in the two time codes, as long as it has been a consistent anomaly, I'm guaranteed perfect sync. No number crunching, no calculator, no muss, no fuss. And it works just the same using any combination of time codes: PAL with NTSC, with 24 FPS, anything.

Oh, that's not all...It gets easier. I don't even have to capture those points on the fly, finger poised over the capture key, waiting, waiting, waiting, *Ooooh*!! I missed it!!

Or worse, trying to park machines on audio hits and then capture offsets. No, no. I sample the audio from the master and from my source into the waveform graphic displays built into the A/V, nudge the cursor across the audio display, and place marks on those things that look like the Chrysler building (door slam) on each display and, Voila!! I've got my vari-speed calculation. Accurate to a tenth of a frame....

....Wait a minute!! What am I saying?? I was just kidding. There's really no such thing as a 2600 A/V. How silly.

Actually, I have a Ph.D. in thermo-nuclear synchronizer and time compression physics and I do it all with a Cray that I have access to at the Defense Dept. You can't do it. Don't try.

Forget I mentioned it. db

John Lawrence is Audio Operations Manager, Vidfilm Services, Inc.Glendale, CA.

WALLY DUGUID

A Church Audio Checklist

ONE OF THE BIGGEST CHALLENGES for a local church sound committee is knowing which questions to ask when working with a system designer. Often, options for functional and technical features are not suggested and at times, the architect and consultant do not even approach the users for their input on the design.

The committee unwittingly approves and purchases a system that in some way is not completely suited to their particular needs.

In our contracting business, a simple, but fairly comprehensive checklist is provided to prospective clients to expose them to the field of available tools and techniques useful in quality sound reinforcement. Some explanations of basic concepts are included. The end result is, hopefully, educated consumers confident of their ability to make accurate choices and who are pleased with the final product.

There are a number of considerations to address when planning a new sound reinforcement system, just as there are when constructing a new building. Four main areas which determine the system's design are:

(1) The environment. What is the physical layout and, more importantly, the acoustic character of the room?

(2) The type of reinforcement desired. Do you want a subtle assist to the natural acoustic performance, or a system where the program's primary link to the audience is through the sound system?

(3) The type of control and functional flexibility desired. Does this system function with or without an operator, or should it do both? What kind of things would you like to be able to do with this system?

(4) Cost. What is the budget allowance and is it realistic in terms of the first three areas?

Within those areas are, of course, many sub-considerations that will be addressed when the design process is underway. The design of the acoustic interface, involving the selection of speaker components and their orientation with respect to the audience seating, is the most important aspect of the system design. The success or failure of the system's performance hinges on this critical factor.

The designer must consider the type of musical and spoken material to be reinforced, the acoustic properties of the room, the requirements of the listeners, as well as other considerations such as operational complexity and budget.

While the acoustic interface and the electronic design are best handled by an experienced sound system designer, the functional aspects of the system have to be chosen by the users. A partial list of considerations follows. Some of these questions do not have simple answers and may require guidance from an experienced designer. This list should help you in thinking about some options that are available and possibly help identify the direction you would like to go in terms of function. Some of these questions relate only to an existing facility, and some to new construction. You decide.

1. TYPE OF REINFORCEMENT PHILOSOPHY

Will this system be invisible and provide subtle enhancement to a predominantly acoustic program or does every instrument, voice, or group of voices need amplification, or should there be a blend of these two approaches? Is there another facility you know of which has the type of system performance you are looking for?

2. TYPE OF PROGRAM

Will speech, classical music, contemporary music, pre-recorded playback, or video reinforcement be the predominant material in use?

3. PEOPLE

Who uses this system besides the principal preacher/teacher? Musicians, class or seminar teachers, dramatic or touring groups all have special requirements that could be addressed by this system. Have they been included in discussions regarding function?

4. WHY

Do you really need a sound system? Or do you need a new one? Are the existing problems with the old system easily corrected by a professional who can fix a bad design with little capital outlay?

5. DESIGN PHILOSOPHY

What decisions are being made for you by the architect or consultant that you should have some input on? Can you meet with them to discuss all of your needs and how to implement them?

6. ACOUSTICS

Was a plan check and acoustic model executed (or measured) prior to the design? What are the particular acoustic properties of the room, are there problems relating to coverage and intelligibility, and are they being addressed at the source (acoustic treatment) or symptomatically (with electronics)? Is dissatisfaction with the old system an electronic or acoustic problem?

7. NUMBER OF INPUTS

How many microphones, keyboards, tape players and instruments do you want to control? What about future expansion?

8. MICS

Who uses mics and where are they used? (Pulpit, lectern, lavalier, wireless, choir [how many?], piano, organ, soloists, baptistry, direct boxes for electronic instruments and congregational response are several examples.)

9. NUMBER OF MIXES

How many different speaker systems do you want separate control over? Sanctuary, choir monitors, pulpit monitor, soloist monitors, instrumentalist monitors, radio/television feeds, tape recordings?

10. MAIN HOUSE SPEAKERS

A central cluster covers the congregation and sounds best in most sanctuaries. How do we make it visually attractive? If a cluster will not fit in a particular room, what are the alternatives and their drawbacks?

11. DELAY SPEAKERS

Are there areas the main house speaker system will not reach? (Under the balcony, transepts, and so on.)

12. NON-HOUSE AND OVERFLOW SPEAKERS

Where do you want speakers besides the sanctuary? (Nursery, cry room, foyer, bride's room, hallways, offices and fellowship hall, for example.)

13. PERMANENT MONITOR SPEAKERS

Do you want monitor speakers in the choir, pulpit, pastoral seating, or hidden monitors in the stage or platform?

14. PORTABLE MONITOR SPEAKERS

These can be used for soloists, choral groups, or special programs. Do you want to finish these in black, wood grain or carpet to match the decor? What about headphones for musicians?

15. EFFECTS, SIGNAL PROCESSING

Do you want to use reverberation or other electronic effects devices (limiters, companders, gates, de-essers, processors)? How about operator training for these?

16. MIX LOCATIONS

The best position for the sound system operator is in a seat where he or she hears what the person in an average seat hears.

Where is the optimum place that would allow practical operation of the system without being too conspicuous from a visual perspective?

17. MIC LOCATIONS

Where would you like to connect mics around the altar/platform area, or have a special location for a bell choir or instrumental performers? What about film or video sound inputs? Is there any need for extra long mic cables?

18. MONITOR LOCATIONS

Where would you like to connect portable monitor speakers? Where should permanent monitors (for example, choir) be located and how should they be trimmed out? Can they be made invisible?

19. RACK LOCATION

Is there a convenient closet or utility area in the vicinity of the altar for placement of the power equipment rack? Is it near convenient electrical power and can it be ventilated?

20. MEDIA MIX LOCATION

Where should the radio or TV audio mixer be placed? What about access and visual monitoring for the operator? Where is the physical interface point for TELCO or microwave connections?

21. RECORD AUDIO

On what type of machines (cassette, reel to reel, R-DAT) do you want to record messages, meetings, or weddings? Do you have a need for a multi-track machine?

22. RECORD VIDEO

Do you want provision for video recordings? Where do you want connections for consumer VCRs?

23. PLAYBACK

Is there a need for playback of prerecorded tapes? Do you need to connect film or video projectors to the sound system?

24. INTERCOM

Is there a need for communication facilities between parties at various locations like pastoral seating, instrumentalists, sound mixing area, usher station, choir room and so on?

25. HEARING ASSISTIVE

Are there members of your congregation who would benefit from the use of a system for the hard of hearing? Should it be an infrared or radio frequency system? What style of earpieces would the users like to have? Should the users be allowed to purchase their own receivers for convenience?

26. OPERATORS

Is this a system that should have an operator during each public meeting, should it be a set-it-and-leave-it system, or is there a practical way to make it function for both?

27. DEFAULT CONTROL

A default control brings the system up to operation with a few basic mics to pre-set levels. This feature allows the use of a complicated system when an operator is not available, as in the case of a weekday funeral or seminar. Is this necessary or is it possible to have the operators leave some controls at nominal positions for this purpose?

28. PATCHING, TIE LINES

Do your operators have need of and the ability for the inclusion of a patch bay?

29. SECURITY, ACCESS

Should access to the operation of the system be limited by locking cabinets? Should it be tamper-resistant or vandal-proof?

30. CABINETRY

What style of cabinet should be incorporated into public areas? Do you want the consultant or design firm to submit a design or build from yours?

Should the architect be involved in this design? Does the fire marshall have any interest in access and egress that this cabinet may interrupt? Where can you see samples of cabinets done for other installations?

31. POWER

Do you have a particular choice of a local electrician to supply the electrical power circuits for the system? Does local ordinance require a permit to provide the power for this sound system? Some touring groups carry sound and lighting that requires a connection to 100 or 200 Ampere single or three-phase power.

Would it be practical to have the electrician install a disconnect panel convenient to the stage/altar for this purpose? Is there a potential need for power line conditioning or surge protection? Is there access for an adequate ground source?

32. REMOTE CONTROLS AND INDICATORS

Are there any special custom controls or features you might want on this system, such as remote indicators for wireless mic transmitters, remote projection screen controls, provision for MIDI (Musical Instrument Digital Interface) connections (or other digital interface-RS-232, 422, etc.), or SMPTE time code in order to automate sound or lighting systems?

33. BUDGET

Is there a proposed budget set for this project? Is there consideration for the cost of a comprehensive technical power supply?

34. TIMING

When should this installation be completed? Is there a construction or remodeling project that it should be coordinated with, or a special or seasonal program that needs a new system?

35. PERMITS

Are there building or other permits required by local ordinance for the wiring or structural aspects of this system?

36. WIRELESS MICS

Would one or more wireless mics be useful in this application? Would it be practical to interface a receiver to multiple sound systems (for example, sanctuary and fellowship hall) in order to use the same mic in either room if desired?

37. ACCESS

Is there adequate access to areas needed by the contractor in which to pull cable, mount connector plates, hang speakers and reach cluster components in the future for service?

38. LIGHTING

Is there adequate lighting for the operators? db

BRENT HARSHBARGER

AUDIO FOR THE CHURCH

Figure 1. Sound and lighting is important to church productions. This is at the Columbus Convention Center in Columbus, Ohio.

• Music in worship has changed in the past decade due to a rise in contemporary Christian music. The change has passed the typical quartet to include all musical styles. With this growth and diversification, the typical column speaker and mixer amplifier are being replaced by sound systems with full range, including high powered sub-woofer systems and satellite clusters with digital delays. Churches are bringing in musical groups that have stage shows with riders equal to secular groups with a comparable following. These riders include 32 and 40 channel sound consoles with enough power to handle 90 to 100 decibels at the mixing position, and lighting rigs with computer controllers. Many churches have incorporated these type of concerts into their music programs. In order to reduce the cost of bringing in such concerts, churches are upgrading their sound and lighting systems to meet the requirements of these events.

Churches use sound and lighting systems to make an impact with their message. Easter presentations are becoming well-thought-out and executed productions, relying on current technology to add realism that has been viewed on Broadway or the silver screen.

To meet the needs of church programs, concert halls are converting their venues into houses of worship. Some churches' Easter and Christmas programs, for example, have outgrown their walls due to production and attendance size. Likewise, when concert halls are not being rented by the local church, the local or regional concert promoter is booking the hall for Christian rock or rap concerts, also known as "music with a message."

THE XENIA CHRISTIAN CENTER OF XENIA, OH

This church is known locally for its concerts. In the past, many visiting artists were traditional southern gospel groups which usually do not require the same type of systems required by today's mainstream Christian music. Xenia Christian Center found their sound system inadequate to handle new programs, and was forced to rent a hall. The church eventually called upon MS Entertainment of Dayton, OH, a sound and lighting rental and sales company that works with medium-to-large concert facilities and has also worked with many national acts. Xenia Christian Center found itself regularly renting sound equipment and lights from MS Entertainment, so the church decided to purchase a sound system that would meet the needs of regular worship and provide the extended performance requirements of special service programs.

Allen Day, Xenia Christian Center's technical director, and John Youker, owner of MS Entertainment, devised a plan to install a sound system that would be acceptable to the riders of national recording artists; would be flexible and have high intelligibility for regular service; and would be affordable. These requirements were not ranked in priority.

THE EQUIPMENT

The head end of the designed sound system was a Soundcraft 32 channel Delta 200 console. The processing equipment included an Electro-Voice 2710 for the house EQ, an E-V XEQ-3 time-alignable crossover that fits into a Crest 8001 for the sub-woofers, a QSC MX1500 for the mids and an E-V 7300 for the highs.

Youker custom-built the speakers to match the church's interior. The speakers were E-V Delta Max clones using E-V components. The sub-woofers, also E-V components, were built into the platform, and used four 18 in. drivers to drive the low end needed for concerts and special music presentations. Outboard processing gear selected was a quad gate from Furman, dbx 160s and a dbx 166. Reverb effects and digital multi-effects processors were selected from Applied Research and Technology and Alesis. A Telex "hard of hearing" system was also installed for regular services. The congregation's response was overwhelming!

Once the system was designed, the church decided to have a volunteer from the congregation help install it to save on cost. I strongly suggest not using volunteer church labor to install a sound system; in my experience, it almost always ends up costing much, much more than is saved, as there is more to installing a professional sound system than there is to installing a home stereo system.

Another example of integrating church programs with hi-tech sound equipment is Stage Tech Inc., of Columbus, OH, who "converts" concert halls into high-powered churches. When I visited Stage Tech, the stage was being set for Christian recording artist Whiteheart and Steven Curtis Chapman and Band. Stage Tech, who has been associated with Whiteheart's world tours for the past five years, was in control of sound and lights.

Figure 2. One of the speaker racks on stage at the Columbus Convention Center.

The tours use double lighting trust, a 96 channel microprocessor lighting console, 40 channel house, monitor boards and a house speaker that could be found at the Columbus Convention Center at any secular rock concert.

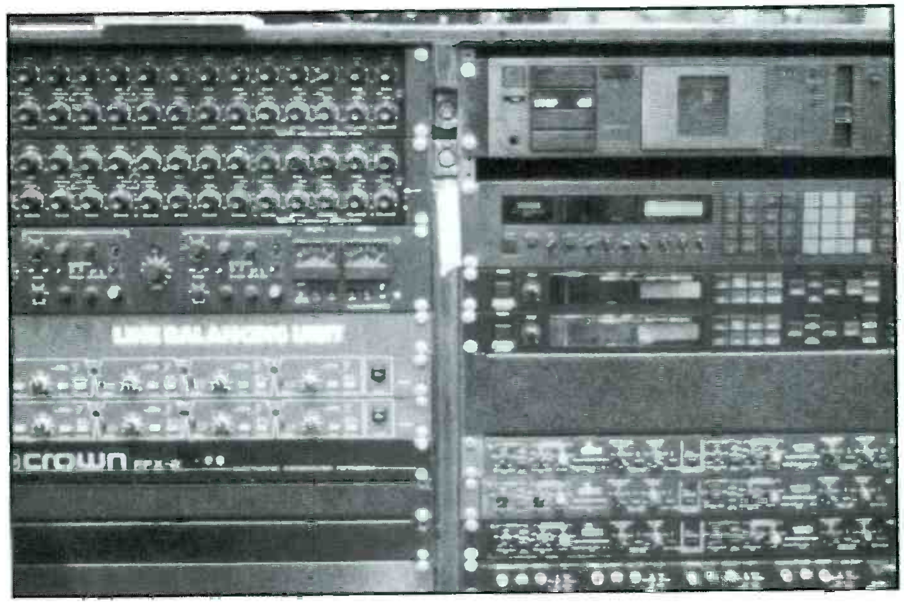


Figure 3. The Stage Tech equipment rack.

Bill Mayerchak, a Stage Tech sound engineer, took control of the house mix for Whiteheart's concert that evening. The system was headed up by a 40 channel Yamaha PM3000 that drove White 4400s and Crown crossovers. Crown Macro Tech amplifiers drove the custom-designed house speakers that used a JBL sub-woofer, E-V mids, horns and JBL high frequency drivers. (Stage Tech is looking to make an expansion in their house speaker arsenal and is testing concert boxes from Eastern Acoustic Works and E-V, as well as their own new design.)

IN CONCLUSION

The next Whiteheart tour will be a show to see! There will be the technological improvements in the house speakers, and Stage Tech is experimenting with putting the upcoming tour in quadraphonic sound.

"The latest Whiteheart album has such phenomenal layering of both synthesizers and vocals, that quad is the only thing that can do the live sound full justice," said Dave Mead, owner of Stage Tech.

With the possibility of Christian musical groups using quadraphonic systems on their tours, and churches bringing in large scale music and drama programs, the need for sound systems that was once limited to larger secular arenas (pun intended) is becoming commonplace. This brings up having knowledgeable people to operate these systems, which will be addressed in detail next issue. db

LAUREL CASH-JONES and FRED JONES

The Elgin Theatre—Modernizing Without Destroying the Past

The Elgin Theatre in Toronto, a stunning architectural masterpiece and one of Canada's most prized possessions, has had many famous performances over its seventy-eight year history. Celebrities such as George Burns and Gracie Allen, Milton Berle and Edgar Bergen have graced its elegant stage.

WITH THIS IN MIND, THE ONTARIO Heritage Foundation in 1989 chose to refurbish the aging Elgin Theatre and give it a face-lift. While they were at it, the Foundation decided to see if the acoustics could be improved using modern technology. Once acoustical contractors were chosen, they were under a strict edict that, in no uncertain terms, all the acoustical work, when finished, must not be visible to the eye and not vary or destroy the architectural grandeur of the Elgin. No small task.

Before getting to the technical information, perhaps some history of the theater would be in order. When the Elgin was dedicated on Dec. 15, 1913, it was christened as the *Loew's Yonge Street Theatre*. The city was so proud of the new theater that a star-studded event was held to celebrate the occasion. A private train brought owner Marcus Loew and his guests, Irving Berlin, vaudeville stars Weber and Fields, and the architect of the theater, Thomas Lamb, to this event.

Figure 1.The seventy-eight-year-old Elgin Theatre in Toronto, Canada was restored to its 1913 splendor, yet updated to 1990 sound.

However, the Yonge Street Theatre was planned only as a part of the theater complex which would be the Canadian flagship of the Loew's vaudeville theater chain. In 1914, another theater was added to the complex seven flights above the existing Yonge Street Theatre. What was unique about this at the time was that it was built as a rooftop theater. It was named The Winter Garden Theatre and became the first such "stacked" auditorium in Canada, and the country's only roofgarden theater.

After its use as a vaudeville house, the Yonge Street Theatre became a movie house and was renamed the Elgin Theatre after the famous Canadian, Lord Elgin. The Winter Garden Theatre retains its name to this day and will be referred to in this article, but all acoustical references will be to the Elgin.

Let us begin by giving a physical description of the Elgin Theatre. The 130 foot lobby features terrazzo flooring, faux marbled columns, gold leaf trim and friezes of reclining cherubs and other such gilded plaster details. As far as the architecture relates to the acoustics, the ceiling is relatively low, since the Winter Garden Theatre is located directly above. The Elgin, which has a seating capacity of 1,500 people, has a large balcony that covers approximately two-thirds of the orchestra seats.

The Elgin was known acoustically, prior to the remodeling, for its relatively short reverb time, giving

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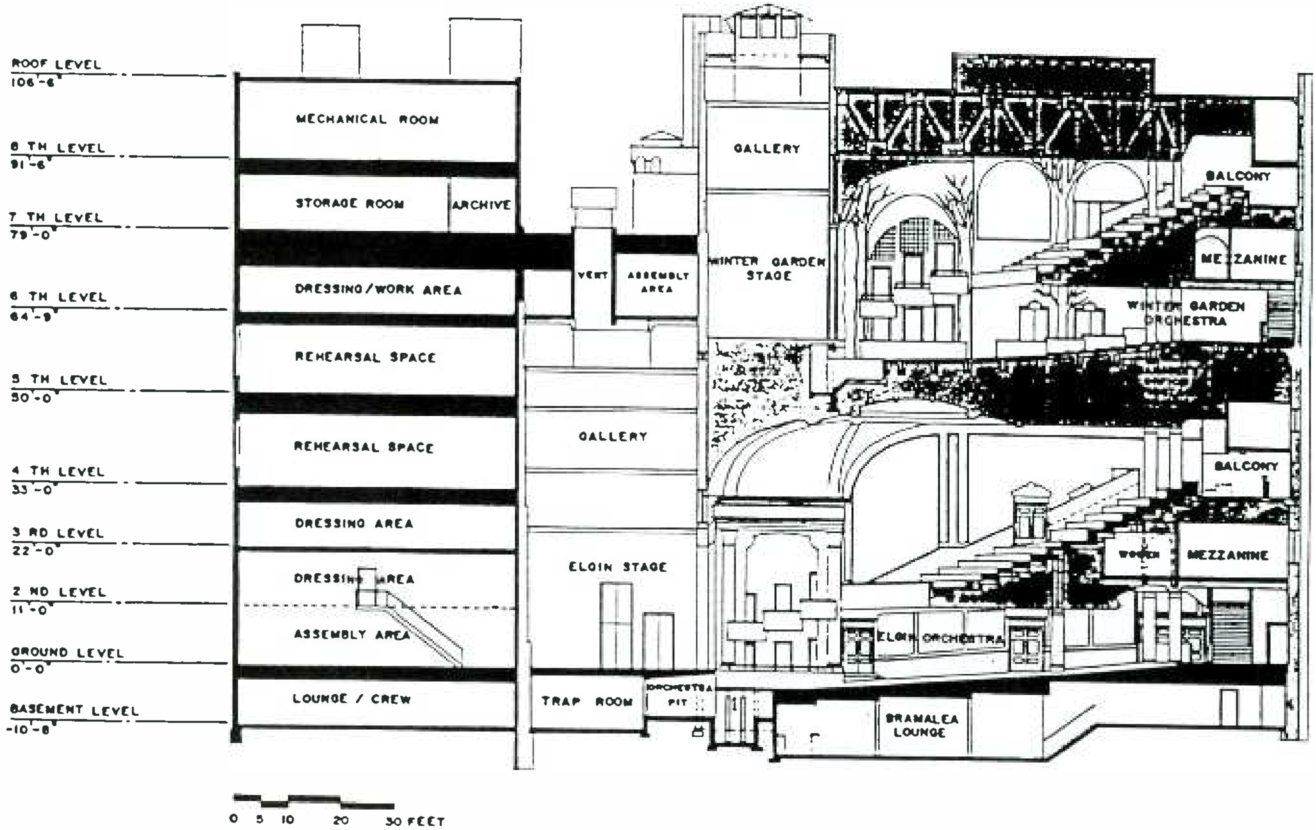


Figure 2. A side view of the Elgin Theatre and Winter Garden Theatre complex. The Winter Garden was added one year after the Elgin opened in 1913.

vocalists a high degree of intelligibility in the audience. This was probably due to the low ceiling height; the ceiling height below the balcony in many places is below eleven feet. The volume of a performance is lower, and reverb is basically non-existent. Reverb time above the balcony is about one second at mid frequencies with the hall empty, rising to about two and one half seconds at 100 Hz.

If we may digress for a moment, perhaps it would be helpful to discuss some level measurement techniques. As most acousticians will tell you, reverberation *level* should be considered as important as reverberation *time* in the measurement of the acoustics of a space, or in this case, the hall. Most audio professionals describe this value, wich is stated in dBs, as "G" or "G *level.*" Measurement of this reverberation *time* and *level* would then give you the number of dB that the hall itself acoustically produces in an increase in the level of an instrument or voice, compared to that of the original level of the instrument or voice measured at a distance of ten meters in an outdoor environment. To give you some idea, a "world class hall" is described as having a "G" value of between 2 dB and 5 dB.

THE TEAM IS ASSEMBLED

Mandel Sprachman, the architect of the restoration project, was responsible for bringing Robert Tanner in to improve the acoustics. Neil Muncy was brought in to design the electrical and do the installation work. Muncy then contacted Dr. David Griesinger of Lexicon, Inc. to help plan the audio system and to design and supply the necessary software.

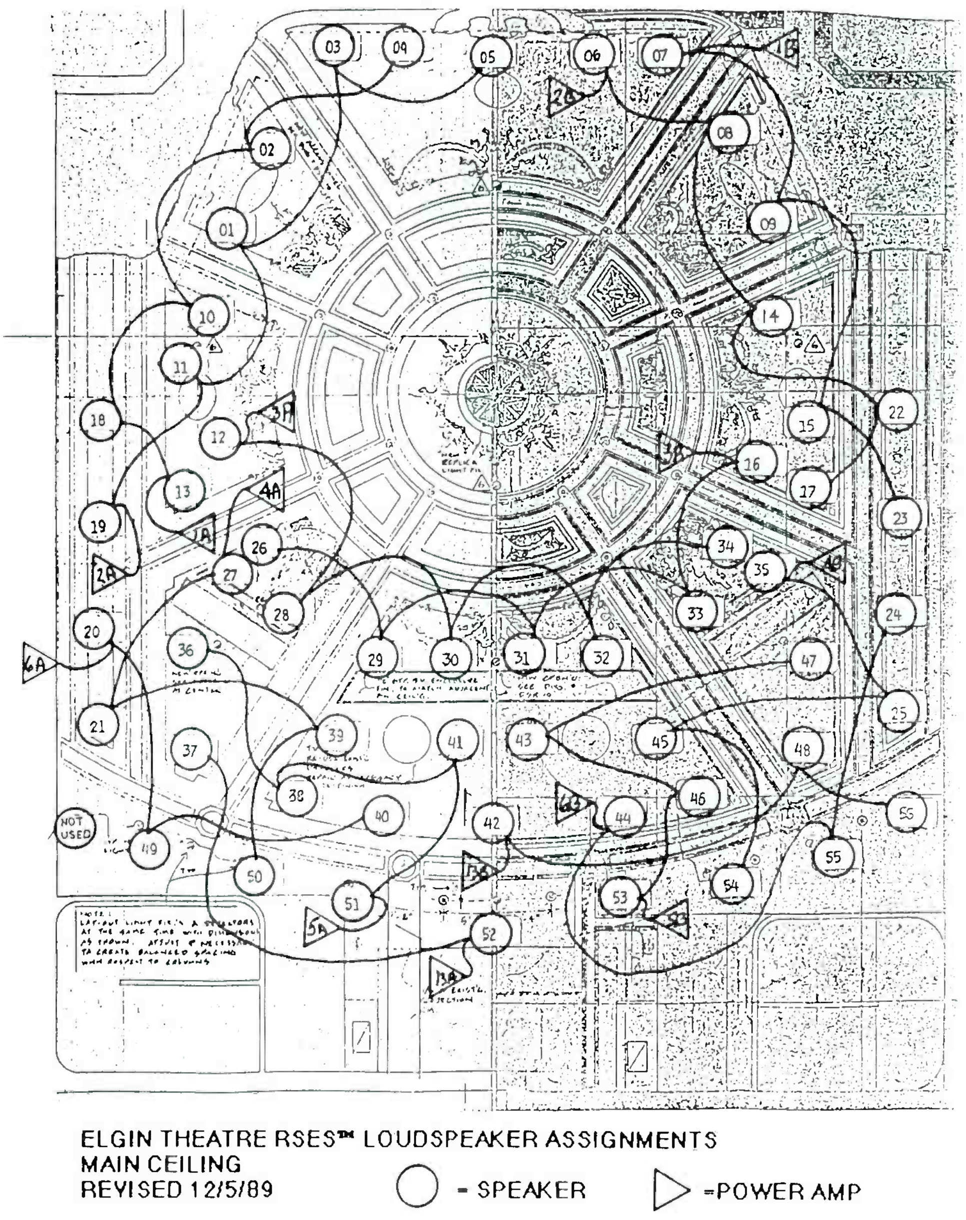
Griesinger settled upon the following method of acoustic measurement for the Elgin installation: A dodecahedral speaker to supply the impulse, and a 10ms multitone burst. The hall sound was analyzed with an octave band analyzer, many pulses were RMS averaged, and the resulting curves were plotted. The plots were then manipulated to yield standard acoustic measures such as "G," Early Decay Time (EDT), RT20, RT60, C80, and so on. "These curves told us a great deal about the hall," Griesinger said. "The most interesting part of the sound is in the first few hundred milliseconds. You can learn about the acoustic echo problems within the building itself, and about how the added energy from the system will mesh with the acoustics of the hall."

Upon experimentation in the lab and in the Elgin, there seemed to be an optimal shape to the added reverberation. It should build up over a period of 20 to 40 milliseconds, and remain at a high level for about 50 milliseconds before beginning to decay exponentially. This type of curve was found to blend optimally with the many loudspeakers and direct sound in the hall.

According to Griesinger, "Robert Tanner calculated that the 'G' in the Elgin was too low—the hall itself was basically too dead for ideal musical acoustics. Our system had to increase both the reverberation time and the reverberation level. It had to be audible not only when the music stops, it also had to be audible *while the music is running*. We must increase the 'G' at least 2 dB at some frequencies. To do this, the system must contribute *significant*

Figure 3. An under-balcony view at the Elgin with the sound-system installation complete.

Figure 4.The main ceiling loudspeaker tiling in the Elgin.



level and sound power to the room—at least as much sound power as the original instrument on the stage. Plus we have to do it without using close microphone techniques or active mixing," he added.

THE WORK BEGINS

Due to the requirement that the audio system be invisible when finished, a conventional sound reinforcement system could not be used. As a result, it was decided to use a multi-channel system with uniform coverage to be provided by a large number of speakers that would be hidden in the plaster ceilings above and below the balcony.

Originally, the acoustical designers thought they would need to place speakers in the walls as well as the ceilings, but it was discovered early on that the outside walls were made of the original masonry, and it was impossible to hide speakers in them.

The number of speaker arrays and mics was determined by the digital reverb devices necessary to accomplish the task of filling the hall. Two Lexicon 480L Digital Reverberation Processors were chosen for the job. "The decision to use the two 480Ls was based on the fact that they run sixteen reverberators, and gave us eight output channels. This made it an ideal choice for this project," Griesinger said.

In this system, the 480Ls supply all of the time delays. As a result of the large amount of output channels, no other delay processors or reverbs were necessary. The two 480Ls are remote controlled by a single Lexicon MRC MIDI Controller.

With the help of the National Research Council of Canada, the Paradigm 3sc bookshelf speaker was selected for the ceiling mounting. It was determined that these speakers were very low in coloration and were small enough to be recessed. All of the speaker cut-outs were covered by metal grills and painted to match the ceiling, thus making it almost impossible to see.

Muncy and his co-workers installed one hundred and fifteen of these speakers as follows: fifty three below the balcony, fifty six above, and six in the columns at the side. These were then wired into

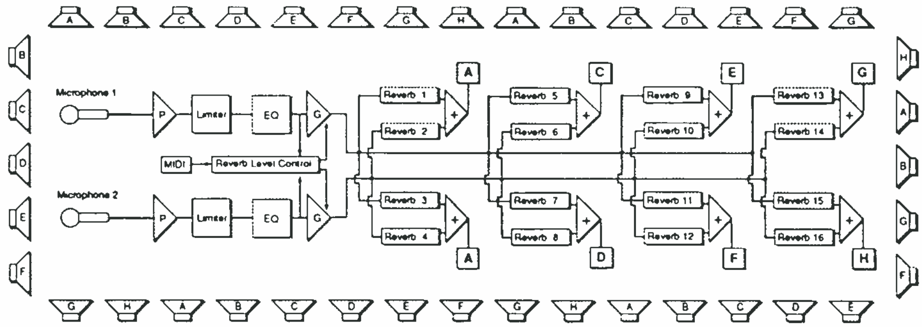


Figure 5. Lexicon's LARES Software System for the 480L.

eight banks in an interleaved formation known as a "tiling."

The eight outputs of the 480Ls were directed to the banks of loudspeakers, which were placed so that no two adjacent speakers were driven from the same output. The lack of coherence between them increases the apparent diffusion and reduces coloration. "Fortunately, the dense ceiling arrays formed image sources well beyond the wall, just as had been calculated," said Griesinger.

The mics used in the Elgin are two Bruel & Kjaer cardioids on short booms that have been attached to the balcony rail, about 45 feet from the stage. Directional mics were chosen to take advantage of the fact that the sound sources would be coming from one end of the room. This placement is due to the invisibility factor. If the mics were mounted any closer, they would interfere with the follow spot operation. All involved agree, however, that an increase in the quality of the system could be had if the mics were closer to the stage.

Muncy's equipment list for the installation, in addition to the 480Ls and MRC, included two one-third octave equalizers to control feedback and compensate for the loudspeaker frequency response, two mic pre-amps, two limiters for the Bruel & Kjaer mics, two racks of eight Ramsa stereo power amplifiers, and a non-interruptible power supply for the digital equipment to prevent system confusion from power glitches that may come from any other equipment in the complex.

NEW DEVELOPMENTS

The installation called for special computer software and led to the development of what is now being called LARES. "The advantages of the LARES system is its ability to greatly increase the gain before feedback of a mic/loudspeaker system, while maintaining very low coloration," Griesinger said. "In the Elgin installation, LARES increases the gain before feedback by at least 18 dB over a conventional reinforcement system."

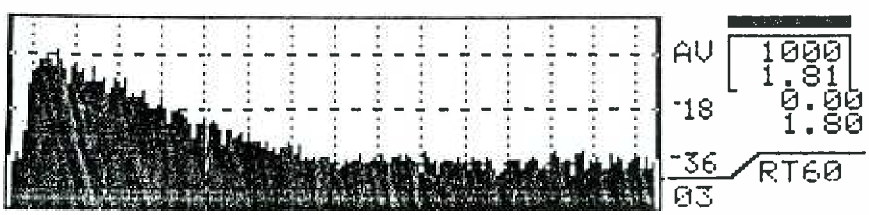
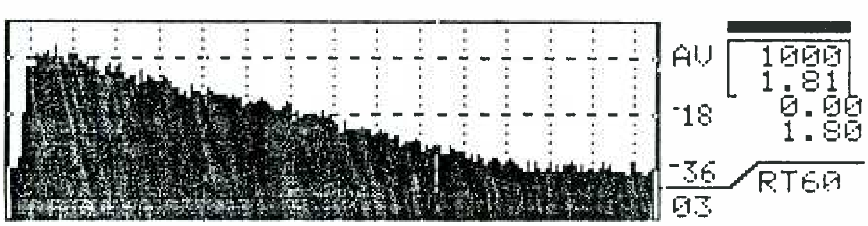


*Figure 6. Under the balcony with the system **off**. Taken at seat 429.*

*Figure 7. Same measurement, same seat, system **on**.*



This stability is said to be due to a very special digital reverberation algorithm, consisting of a large number of delay taps, all of which continuously change their delay values independently and randomly. This random changing is so fast that within one or two seconds, every delay has migrated to a new value, thus there is no correlation between the impulse response of any reverberator and itself after about one second.

The amount of gain that such a system will produce depends upon the square root of the number of channels used, which is a product of the number of speaker banks and the number of mics.

Using random reverberators has many advantages over a fixed delay or reverberator in that they can increase the gain before feedback by 6 dB, and since the timbre they produce changes more rapidly than the ear can detect, it provides a system with extremely low coloration. Since the LARES system uses a large number of reverberators and distributes their outputs through many interleaved loudspeaker banks, LARES depends upon having a multi-channel distribution speaker system, such as the one described above.

The amount of gain that such a system will produce depends upon the square root of the number of channels used, which is a product of the number of speaker banks and the number of mics. There are two mics, eight speaker banks and sixteen reverberators in the Elgin. Therefore, the resulting sixteen channels gives an additional 12 dB increase in gain before feedback.

"We have recently installed some refinements to the software," Griesinger stated. "We achieved a lowering of the coloration to inaudible levels and an increase in the gain before feedback to 20 dB over a conventional system."

Among the new refinements in the LARES software is the ability to detect when the sound level from the source is lower than the acoustic output of the reverberators—for example, when the source has stopped and the sound level in the room is decaying. Under these conditions, the gain is lowered about 6 dB. Full gain is restored as soon as the sound has decayed, or a new source from the stage can be heard. The system can also tell the difference between speech and music sources, and automatically adjusts the system.

The Lexicon 480L processors are adjusted with the MRC MIDI Controller to match the reverb time and reverb level needed by each type of performance, such as opera, spoken word, and so on.

Upon experimenting with the system, it was found that the optimum reverb level for speech is about 6 dB lower than that of symphonic music. Opera required intermediate values, with dialogue being closer to speech, A Cappella singing required about two dB more, and accompanied singing required about two dB more than that.

Reviews of the first musical show praised the theater's acoustics. In fact, the reviewer said it was his first chance to hear an unamplified chamber ensemble in the recently renovated Elgin Theatre.

"The improvements in gain before feedback achieved by LARES need not be confined to reverberation enhancement, as it was used in the Elgin," Griesinger said. "The system can be adjusted to be useful for conventional sound reinforcement systems, as long as it uses a multichannel distributed speaker system."

RESULTS

Results of the acoustic treatment show a dramatic improvement under the balcony. The array of speakers now produces equal distribution of sonic energy in all directions, and seems to surround the listener. You can stand directly under any of the speakers mounted in the balcony and not hear them specifically, even though the speakers are less than five feet above your head. (What this means is that no audience member will jump to his feet and shout "Haas Effect!")

As you can see from these measurements of the "G," Griesinger, Muncy and Tanner have succeeded in greatly improving the acoustics. Another important measure of the successes of (LARES) the system is the increase of the "G" in the lateral energy. This was primarily a problem under the balcony, where the most expensive seats are located. Due to the high absorption of the audience in this area, there was very little lateral energy.

The LARES system in the Elgin is running better than expected. There is at least a 2 dB increase in "G" throughout the hall, and a dramatic increase in lateral "G" below the balcony, with no audible electronic coloration.

When the system is turned on, the increase in spaciousness is dramatic. Lateral "G" values rise 4 dB or more under the balcony. The ceiling seems to disappear and you feel surrounded by the hall above. In the orchestra seats in front of the balcony, and in the dress circle above, the change in sound with the system is less dramatic, but still quite audible while the music is running. Level and spaciousness rise and the increased reverb time is wonderful.

"When you start out to design a system of this type, equalization and the shape of the reverberant decay must be carefully matched to the hall," says Griesinger. "You must take great care to rely on both measurement techniques and your own ears. I have found that the most popular techniques for acoustic measurement, such as MLS (MLSSA) and TEF (Crown-Techron), do not work with the LARES system!" The reason for this, he says, is that both systems use time integration to increase the signal to noise ratio for their measurements.

Since LARES is constantly changing with time, Griesinger believes those techniques simply cannot see it. To properly measure a LARES system, it is necessary to use a more old fashioned-type system that uses an acoustic impulse to excite the room and analyze the decay.

THE REVIEWS ARE IN

Reviews of the first musical show praised the theater's acoustics. In fact, the reviewer said it was his first chance to hear an unamplified chamber ensemble in the recently renovated Elgin Theatre. As you may have guessed, the Ontario Historical Foundation is very pleased with the acoustical results, as well as the visual beauty of the restoration in this fine architectural landmark. db

LEN FELDMAN

DAB—The Next Generation of Radio Broadcasting

THE AUDIO INDUSTRY IS IN GENERAL agreement that our current broadcast systems are not up to the task of delivering the full sound quality of our new digital audio program sources such as CDs and, more recently, DATs.

FM, with its high frequency response limited to 15 kHz and its dynamic range seldom greater than about 70 dB or so, falls short of being able to transmit the full audio range of CDs, let alone the more than 90 dB of dynamic range typically available from digital program sources. And if FM falls short, AM is even more woefully inadequate. So much for what the industry has no argument about. When it comes to implementing a new broadcast service, variously known as DAB (for digital audio broadcasting) or DAR (digital audio radio—not to be confused with the daughters of the American revolution), there is anything but consensus.

At the February joint meeting of the New York Section of The Audio Engineering Society and the Society of Broadcast Engineers, this lack of consensus was in evidence during a presentation in which several proponents of different DAB systems voiced their opinions concerning what form of DAB should become the standard for the United States. The presenters included **Ron Strother**, of Strother Communications, **John Leonard**, of Kintel Technologies, **Peter Dolan** and **Dr. R. Bruno**, representing CD Radio, Inc., **Emil Torick**, a consultant to the Eureka Consortium, **Michael Rau** of the National Association of Broadcasters and Paul Donahue and **Tony Massiello,** representing Project Acorn.

One of the issues raised by Strother was a key one: should provision be made for satellite delivery of DAB, for terrestrial (local, over the air) delivery or both? Strother maintained that, whether broadcasters like it or not, some form of satellite DAB delivery system will have to coexist with a terrestrial form of delivery. He proposed that the satellite system might be regional in nature, rather than national, and that it might therefore provide service to areas that otherwise lack diverse radio service now, such as in the Rocky Mountain states, etc. He noted that such service might be analogous to present-day "clear channel" stations that cover hundreds of miles. What raised some eyebrows in the audience was Strother's conclusion that, despite the desire on the part of the industry to develop a system that would offer "parity" between AM, FM and DAB broadcasting (and even compatibility between FM and DAB), ultimately, the FM and AM broadcast systems will disappear in the years ahead.

THE KEY TO ALL SYSTEMS—DIGITAL COMPRESSION

Leonard presented what he termed a compatible system for DAB. That is, the system can exist at the same frequency spectrum as current FM service. Furthermore, it exists side by side with current FM and does not interfere with it. Leonard was quick to point out, however, that his system could not address the issue of compatibility when it comes to AM broadcasting. The Kintel proposal, he maintained, would not even require new licensing. The system involves a technique called "power multiplexing."

What Leonard did not address, however, was the method of digital data compression which he admitted would have to take place, recognizing the fact that a current FM channel is only 200 kHz wide. In fact, since FCC rules permit each FM band to extend out to 120 kHz (providing the extremes of that range are 25 dB down in power), the effective bandwidth per channel might be thought of as 240 kHz. The digital carrier that is added to the existing analog carrier represents only about one percent of total power that would need to be taken away from the analog carrier. (–25 dB corresponds to less than one percent.) Accordingly, there would be an insignificant loss of coverage as far as the existing analog audience coverage is concerned. Kintel's studies reveal, however, that even with one percent of power assigned to the second carrier, coverage of the digital service would be as great as that of the analog service.

While Leonard's presentation certainly seemed attractive in terms of cost and compatibility, his failure to address the nature of the digital compression techniques used in the system left some unanswered questions. How severe would the compression have to be in order to be handled in the narrow bandwidth of an existing FM channel? How much sacrifice in fidelity would such severe compression entail?

SOME FAVOR EXCLUSIVE SATELLITE DELIVERY OF DAB

The next speakers were Dolan and Dr. Bruno. They favored and stressed the advantages of satellite delivery, such as high areas of coverage and high quality. After reviewing the hardware needed for proper coverage, Bruno described the waveform that would be used as being a combination of frequency diversity and time diversity. It also uses a receiver that channels adaptive equalization. Terrestrial repeaters would be used to, among other things, take care of filling in areas where there are signal nulls and multipath or other forms of fading. The receiver would also feature space diversity (use two antennas, spaced apart by at least the distance of the wavelength of the transmission).

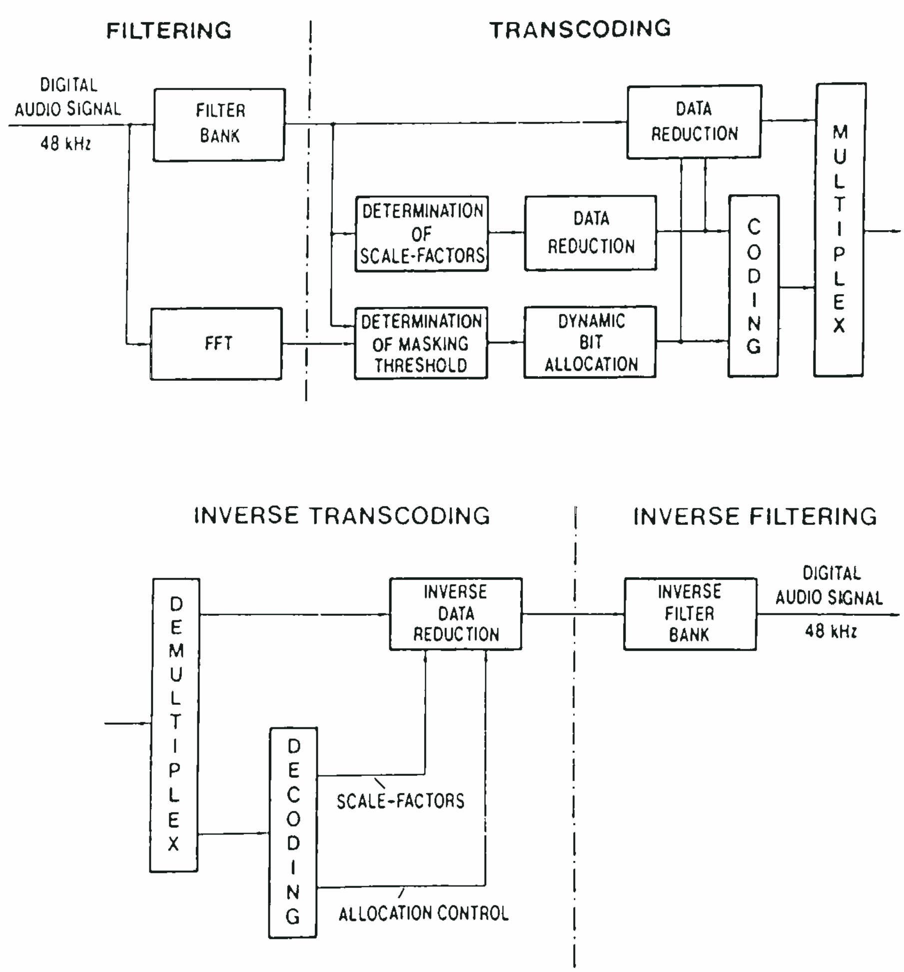
Bruno suggested that while compression would have to be used as a practical matter of accommodating a sufficient number of channels, such compression should not be so severe as to require transmission rates of anything less than about 256 kbit data rate per second. Adaptive equalization, referred to earlier, simply means the constructive addition of multiple sources of the signal. In the system, each channel switches its transmission frequency every two milliseconds, in a sort of "musical chairs" arrangement, where all programs continually "hop" in frequency. This is what is meant by "frequency diversity." The receiver is able to "follow" a given channel as this "hopping" takes place. The system is expected to operate in the so-called "L-band" of frequencies, which is located at 1500 MHz.

CD Radio envisions their system as being practical for the mobile environment. They claim that even in automobiles (presumably equipped with small, practical sized "dishes"), listeners would be able to travel from one end of the country to the other while listening to the same channel. They foresee coexistence of satellite DAB and current analog services, since a national DAB service would not lend itself to local advertising while certain localized programming, such as local news, local talk shows, weather reports and the like would, in turn, not lend themselves to the national satellite DAB broadcasting concept.

DAB AND THE EUREKA 147 PROJECT

Torick, the well-known engineer (formerly of CBS Technology Center and, more recently associated

Figure 1. A block diagram of the MUSICAM encoder (above) and decoder (below).



with Broadcast Technology Partners, the company that had been promoting the FMX system for improved FM transmission and reception), spoke next. Both he and Rau, the senior vice president, Science and Technology with the National Association of Broadcasters, (NAB) were there, essentially to support the system proposed by the Eureka 147, a consortium of more than a dozen European partners, including major electronics companies of European countries and several centers of technical education. This system brings together two concepts in sound broadcasting. The first of these is high quality bit-rate reduced digital sound and data.

The idea is to remove data that is subjectively redundant and inaudible. To put it simply, the transmitted signal contains only those parts of the original signal that can be heard by the listener. To understand this approach, you need to realize that the threshold of human hearing is not fixed, but can vary depending upon the material being broadcast. For example, if a single tone of high amplitude is transmitted at a given instant, other tones that are close to it in frequency, but of lower amplitude, will be completely "masked" by the predominant tone. Interestingly, this is much the same data encoding approach used in the recently announced Digital Compact Cassette technology (DCC) that Philips announced at the last Winter Consumer Electronics Show. In both cases, the elimination of non-essential digital data results in a data compression ratio of about four-to-one. In the case of the Eureka 147 project, this compression or encoding system has been given the name MUSICAM, which stands for Masking-pattern Universal Subband Integrated Coding And Multiplexing. As a result, transmission rate for the Eureka 147 system can be as low as between 96Kbits/second to a maximum of 128 kbits/second, averaging about 112 kbits/second.

COMBATTING MULTIPATH PROBLEMS

In addition to the spectrum conserving encoding scheme developed for terrestrial or satellite DAB (Torick maintains that the Eureka 147 system is equally applicable to both forms of broadcasting), the developers of the system have also come up with a modulation scheme resistant to echoes or multipath signal reflections—recognized as one of the chief problems associated with current FM radio broadcasting. Multipath problems are, of course, particularly annoying in moving vehicles where changing reception patterns occur constantly. The Eureka 147 system for combatting these problems not only offers high spectrum efficiency, but, according to its proponents, virtually unimpaired reception in moving vehicles or in a portable environment. Twelve to sixteen programs can be accommodated in a total bandwidth of about 4 MHz. That makes the system some three to four times more spectrally efficient than current FM radio.

The technique used to accomplish this is called Coded Orthogonal Frequency Division Multiplexing (COFDM). With COFDM, the digital data is transmitted on a large number of radio frequency carriers, each of which has a relatively low bit-rate. This technique is combined with a special decoding technique known as Viterbi decoding. The result, said Torick, is an extremely rugged transmission channel. When multipath or echoes occur in a moving vehicle they do not completely degrade all the carriers at the same time and it is this fact that combats the multipath problems that are so familiar to those of us who have listened to FM and stereo FM while driving in a car.

According to Torick, both the transmitters and the receivers of the proposed system rely upon full digital signal processing, and circuitry can ultimately be implemented in very large scale integrated circuitry (VLSI). Such implementation should result in relatively low-cost and easy-to-use radios or tuners. Several variations of the system have been, or are about to be tested in Europe and in Germany, in particular. For example, one test will involve a total bandwidth of only 3.5 MHz and will provide for 10 stereo channels. In another implementation goal, 1.1 MHz of bandwidth will be used to accommodate some five stereo programs.

A FREE DIGITAL DATA CHANNEL COMES WITH THE EUREKA 147 SYSTEM

In addition to providing a new high quality broadcast service, the techniques associated with this system permit the transmission of an additional data channel. This channel could be used, for example, to transmit information about the sound program (title of the song being played, name of the vocalist or performers, etc.), for text display on future receivers, or even to transmit still pictures for video display.

NAB EMBRACES DAB AND, PRELIMINARILY ENDORSES EUREKA 147

Rau explained why the NAB, after much soul-searching, decided to embrace the concept of DAB. Essentially, he maintained that it would be impossible and non-productive to fight against DAB in order to preserve the status quo of American broadcasting. As for the Eureka 147 system, the NAB's endorsement of that system was described as an initial endorsement, pending a technical review of all the proposed systems. Rau says this system offers possibilities for improvement as well as a variety of tradeoffs that could be used to customize the system. Eureka 147 has undergone more tests than any of the other proposed systems, says Rau, and has proven to be a viable system thus far. Additional tests will be conducted.

Project Acorn's Donahue and Massiello were up next. Donahue is vice president of Engineering for Gannett Broadcasting while Massiello is director of Technical Operations for CBS Radio. Just when the audience was settling into what sounded like a consensus of sorts (or at least a partial consensus), along came these two to criticize just about every aspect of the Eureka 147 Project. Using patent counsel advice as the reason why they could not detail the exact technical details of their system, they nevertheless claimed that they had developed a system for terrestrial broadcasting that would provide CD quality sound and eliminate multipath problems, all without degrading the quality of existing FM broadcast services. The system is said to be power efficient and can provide five programs in 1 MHz of spectrum space. Gannett has enlisted the efforts of additional firms in creating the Acorn System, such as SRI International and Corporate Computer Systems, Inc.

While Massiello did not detail the modes of digital compression techniques used in their system, what little he did disclose was that it takes into account psychoacoustic principles such as those proposed by the Eureka 147 project. Contrary to the Eureka 147 approach, however, Acorn strives to deliver a DAB signal that will fit within 200 kHz of bandwidth and will be compatible with present FM and AM broadcasting systems! They rejected the "L" band because of its limited range of coverage. Acorn Systems has developed a system that will allow each FM broadcaster to transmit an analog signal and DAB signal over their currently assigned frequency. They even claimed that no new licensing will be required. Acorn concluded that while AM broadcasters would probably have to shift to alternate spectrum space, the transition to DAB would be less costly using the ACORN system than it would be using the Eureka 147 or any of the other systems proposed thus far. Of course, this presentation, being devoid of many specific technical details, sounded almost too good to be true, but Donahue and Massiello assured their audience that they were not dealing in "smoke and mirrors."

What clearly came through in the meeting, which lasted well over two hours, is that it is a bit premature to zero in on a particular DAB system as a national standard. Many on-the-air tests must still be conducted and evaluated. Hopefully, the FCC, which in the recent past has been reluctant to endorse any single system for a new service (as, for example, in the case of stereo AM, where multiple systems have impeded the acceptance of this broadcast enhancement), will act more definitively in the case of DAB. In the event that the FCC fails to do so, it is to be hoped that the industry itself will have the good sense (and patience) to zero in on a single system which it can then recommend to the FCC, as was the case with our present standard for stereo TV sound. In any case, DAB is an important broadcasting option that is much needed, and that should stimulate a host of new professional broadcast as well as consumer products. db

RANDY FUCHS

Breaking All the Rules at The Walt Tucker Arena

It started off as an experiment in a garage. Now twenty years later and thousands of hours wiser, a 5,000 square foot room has evolved, giving birth to an experimental recording facility affectionately known as "The Walt Tucker Arena."

SCOTT PAGE (SAXOPHONIST FOR Pink Floyd, Supertramp, Toto and others) started off in 1971 with a small TEAC reel-to-reel and a Kasino solid-state PA head. Like most musicians, he set up the studio in his parents garage. "Imagine eight guys playing live trying to capture that special magic. It sounded like eight guys playing in a box. It sounded horrible because it was a box!" Page said. So he experimented. He deadened the room entirely. He put tweeters in the ceiling, speakers in the wall, subwoofers in the floor, and was able to design the size of the room by using delays and routing the system through different parts of the room. That was the beginning of EARMAX, the unique audio recording and mixing system that is today used in The Arena.

Figure 1.The Arena.

Many engineers who know better don't believe the quality of sound that comes from this facility. It is not a sonically-designed room. The room is 100 feet wide by 50 feet deep with skylights, vaulted ceilings (varying height from 12 feet to 25 feet), plenty of sunlight and no feeling of isolation. One whole side of The Arena is windows—just glass walls and brick. Sometimes during a session the windows are left wide open. Microphones are left on the ledge to capture additional ambience. "It doesn't make any difference," Page says, "because by the time you design the sound of the room environment that you're in, it's way over the noise levels of anything coming in through the outside."

To change the acoustical characteristics of the room, thick, soft 12 foot high matte black heavyweight theatrical curtains hang from a track that can be used to encircle the studio. The EARMAX concept has been developed to capture the essence of live performance in a recorded environment. EARMAX employs the best techniques from live sound and puts them in a studio environment and still allows for the excitement generated by the musicians. By allowing technology to work with musicians, instead of against them, it allows for a better performance. "I like recording a group all together. In the fifties a group of musicians just stood around and faced each other and they recorded great songs! I'm doing that with nineties technology!" Page said.

Each musician has his own 10 foot x 10 foot *instrument station* set on a circular tiered riser. This permits uninhibited visual eye contact from musician to musician. The musicians love it and they all get on the same "wavelength." The Arena becomes an emotionally inspiring room, thereby increasing the odds of producing a more magical product.

All instruments (except for drums) are plugged into the station and the signal is then routed to an isolated area outside of The Arena environment. There are no amplifiers or instrument speakers on the stations. This is how everything is isolated. The signal might go directly into one of the Soundcraft consoles, or to an isolated speaker with multiple mics in front of it, or both. Ultimately, the sound is then rerouted through the QSC Series

Three amplifiers to the instrument station and then to a pair of JBL Control 5 monitors as well as a pair of Wolcott Omnispheres (a 360 degree x 120 degree sound source). This ring of speakers encircles the musicians, and becomes the audio walls of this patent-applied-for EARMAX concept. Although the sound is taken from a different location, the engineer independently controls all the levels at each *instrument station*.

Not using headphones prevents band members from feeling isolated from each other, allows musicians to hear their instruments the way they're used to, and prevents ear fatigue. The Arena's goal is to provide the most comfortable playing environment anywhere.

During the re-mix, individual instruments can be put through the EARMAX system. By making the room sonically smaller or larger, the engineer can enhance the sound of a single instrument. "Imagine the sound of a kick drum going out into the room and reshaping it and then putting it back into the mix!" said engineer Paul Ray.

With live mics there is definite leakage. "Leakage can be your friend," Page says. "Controlling it is the key factor here. It's important the way the instruments mix in the air. Also, the idea of it is to be able to get the spirit of a live recording but still be able to control the recording, so you can get a studio-style record."

No gates are used during the live recording. All gating is used during mixdowns. "Mic'ing techniques are more important. I'd rather move things around till they sound good," Page says. "I like pointing them in different directions and trying combinations of mics for different kinds of sounds. For overhead you know what I love? It's the new Audio-Technica AT825 stereo mic. It's unbelievable!"

In addition to standard signal processing, The Arena has an old Paul Revera pedalboard system. An array of great classic guitar pedal effects are mounted and run through a Rocktron Hush single-ended noise reduction unit. Page enjoys using the pedalboard system with the Lexicon 300 and Digitech DSP256+ for added richness.

The heart of the system are the Soundcraft consoles located in the middle of The Arena. A Soundcraft 8000-40 x 12 monitor console is used as a matrix for routing the EARMAX system. The 16 foot x 18 foot control room features the new Soundcraft 3200-36 x 32. The control room can swap functions with the main room and easily transforms into a vocal booth. A Soundcraft 8000-40 x 12 *house* console is used for the monitor mix and a Soundcraft 200B-24 x 4 x 2 is used for the drum rack.

If additional inputs are needed, a Soundcraft Delta-24 x 4 x 2 is used, and a Soundcraft 6000-32 x 16 x 2 with patch bay is used for the MIDI room.

Figure 2. Scott Page at the Arena console.

A large screen JBL projection television system which projects a picture to a screen in the center of the room is used for all the audio for video projects in conjunction with "The Truck," an off-line remote video editing facility. "When we did the audio layback for the Bon Jovi video *Access All Areas*, we took 120 hours of footage and had to cut it down to 90 minutes," Page said. "It was shot all over the world in a lot different formats with different crews and because of all that, it was a potential nightmare. When you listen to the finished product though, it sounds and looks great." The audio was transferred to 3/4-in. SP Beta and that was then locked up to the picture. That was transferred over to the multi-track and the mix was built up from that point on. That was then mixed down to the Sony D2 Digital video decks also using Dolby SR noise reduction.

The greatest thrill for the Walt Tucker staff was the first time the completed EARMAX system was turned on. It was for a project for a European artist named *Ricky Gianco*. The band for that project consisted of Jeff Porcaro on drums, Steve Lukather and James Burton on guitar, Bob Glaub on bass, and a bunch of Tower of Power guys as well as Page on horns. The Walt Tucker staff knew it would work in the remix mode and in theory, they assumed it would work in the recording mode.

The system did what it was supposed to do. It worked great. It inspired all the players and today, the Walt Tucker staff moves on with what they like to call "that great pioneering spirit." If it was wasn't for people being open minded and caring about what things sound like, none of this could have worked," says Page.

"Some great companies like QSC, JBL, Soundcraft, Lexicon, DOD/Digitech, Wolcott, Audio-Technica, Rocktron and SWR have that pioneering spirit that makes it happen." db

SIDEBAR

The Walt Tucker Group is a horizontally-based company that caters to all facets of the music industry. Named after Page's two idols who were "pioneering visionaries," Preston Tucker and Walt Disney, this facility holds many different companies. The Arena is a 24 + 16 track recording facility featuring EARMAX. Walt Tucker Merchandising is a promotional merchandising division, and Walt Tucker Designs is a merchandising fulfillment house. Some clients include Yamaha, Rico Reeds, UMI (Conn, Benge, King), Artley, QSC, JBL, Lexicon, DOD, numerous post houses and music stores. Some independent businesses located in the facility that are tied together under joint venture agreement include Music Bank of America publishing, D-Squared Graphic Design, Audient Marketing Public Relations and "The Truck" remote video editing facility.

ED LEARNED

Live Sound Reinforcement: Asia-Pacific 1990, Part III

International touring by major American pop-rock acts has become an accepted, even necessary, practice in today's music scene. Popular music and musicians from the United States are admired and widely copied everywhere: I've encountered Michael Jackson's and Madonna's music and videos in all sixty-five of the countries I've toured in the last eight years. It's only natural that the production values embraced by these artists would be copied, too. MTV watchers enjoyed evidence of the internationalization of pop-rock extravaganzas this past February, 1991: the *Rock In Rio II* concert, televised from Rio de Janeiro, Brazil, showed that massive stage, light and sound production packages are no longer exclusive to North America, Europe and Japan. The popularity and financial clout of major artists, coupled with the proliferation of modern staging techniques around the world, insures that regions other than the "big three" will get to see more major artists, with their mega-productions, over the next ten years.

Touring at the "major-league" level is exciting; the most sophisticated production packages available are custom-designed for each artist. The Jacksons, Madonnas, McCartneys, and Pink Floyds of this world can take a "spare no expense" view of their production: staging and presentation must make a unique statement *and* entertain.

When these groups tour overseas, they expect uniformity in production. Any equipment contracted locally must meet the same stringent standards as what is used at home; staging and equipment unique to a particular tour must travel with that tour wherever it goes. Needless to say, transportation and labor expenses become horrendous; only upper-echelon pop groups can afford to tour internationally with their extravagant production. If local sound and lighting equipment doesn't measure up, costs continue to rise as more gear from home is required. This is one reason why you don't see an artist like Michael Jackson tour a country like Indonesia; although they'd love to have him, and have the population to support his concerts, they don't have enough top-quality equipment to service a tour on that scale. Is this situation changing?

While mega-tours by individuals and groups are impressive, we must remember that they comprise less than one tenth of one percent of the total live music performances given in any year. Massive production is not required for the presentation and enjoyment of music; it represents a single school of thought. Thousands of musical groups, representing different genres of music, give performances of value every day with a more modest level of production. In today's music scene, dependable sound equipment is imperative for *any* job; the difference between a theater concert and an arena-style production should be one of quantity, not quality. Thanks to the influence of Western music productions, the importance of quality sound is apparent to everyone; it has inspired rapid improvement in the quality of systems overseas.

When I first visited Jakarta, Indonesia, in 1983, my PA system was four Altec 15 in. woofers and two Altec multicell horns, powered with SAE 200 and BGW 750 amps, per side (see **db Magazine**, March/-April 1985). When I visited again in 1987, I had four 15 in. woofers, four 12 in. woofers, two 900 radial horns with Altec drivers, and four piezo tweeters, powered with BGW amplifiers, per side (see **db Magazine**, January/February 1989).

On my most recent trip, I didn't contract any local PA systems. The Charlie Byrd Trio sound requirements allowed me to use in-house PA systems at all Jakarta venues (see Indonesia—next issue). I did, however, meet several local sound engineers at our concerts, who told me about a new sound company in town that had sixteen Turbosound TMS-3 cabinets and eight subwoofers, powered with QSC amps. While this might not be enough for Michael Jackson, it's certainly enough to cover the sound requirements of most groups currently coming to Jakarta. In the past year, several outdoor concerts featuring Indonesian groups were presented using this system, with excellent results. There is keen interest in approximating Western arena and stadium productions, so the day is not far off when they will have the 60+ cabinets needed for a top-end production.

My 1990 tour revealed that there are now many sound companies in the region using "brand-name" house speaker systems familiar to many engineers here in the United States. Meyer, Apogee, Turbosound, or Electro-Voice DeltaMax systems, powered by Amcron (Crown), Crest, or QSC amplifiers, were available in most major markets, occasionally in smaller ones. Monitor systems had not quite caught up; one company in Singapore offered Meyer monitors, but most seemed to use stock wedges by E-V, Peavey, JBL or Yamaha. Consoles were predominantly Yamaha: most large companies use PM-3000s, although there was the occasional Soundcraft available. Ramsa consoles, while very popular in the United States, are just beginning to make inroads in the Asian market. I'd feel very comfortable contracting complete systems from local sound companies for a tour of *major* Southeastern Asian cities. Of course, my tours often wander away from major Southeastern Asian cities; until the technology fans out further from town, I will continue to carry my own stage equipment, a monitor system and essential front-of-house control equipment.

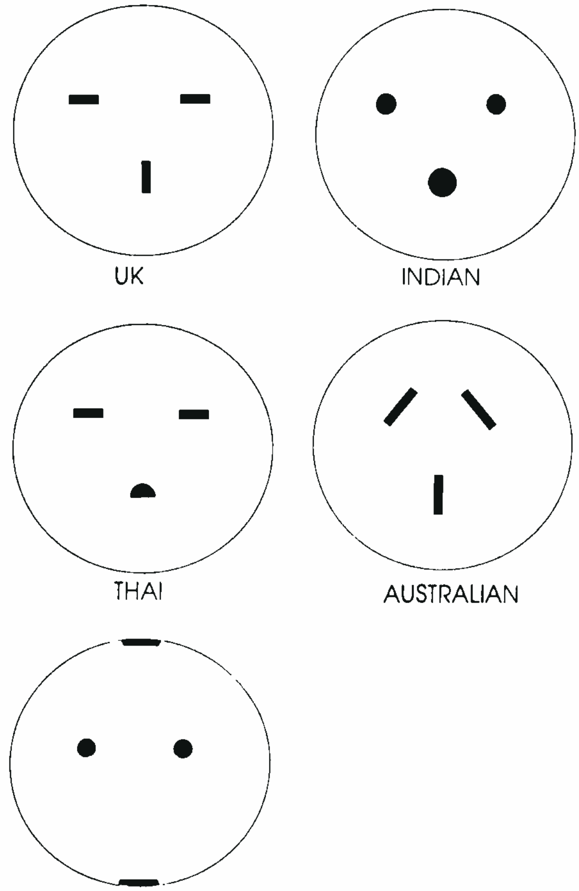


Figure 1: I can't recall any trip where I've encountered more different types of electrical outlets than this one. These illustrations of the different outlet-types will (hopefully) clear up any confusion.

SINGAPORE

Changi Airport is perennially rated among the top five airports in the world by most travel magazines. Upon arrival, it's easy to see why: customs formalities are quick and easy, baggage claim is spacious and well-organized, and it's got to be the cleanest airport anywhere. Departure, however, presents a different picture: Singapore is a major transit point in Southeastern Asia, so quite often there are large crowds in the departure area. Expect a hassle unless you are lucky enough to leave around mid-afternoon, when the airport is not quite as busy. Your equipment will be *thoroughly* searched on departure, so be sure to allow enough time for that. As of this writing, the new international terminal should be operational, which will reduce departure crowds substantially.

Charlie Byrd played two concerts, on consecutive evenings, at the World Trade Center auditorium. Located on the upper floor of the Center, the auditorium seated around 1,000 in a gently raked single seating area. Designed for corporate meetings and speeches, reverb time was only 1.5 seconds, and seemed very free of overt coloration.

Figure 2. Wayne Toups and Zydecajun performing at DBS Auditorium in Singapore.

I obtained AC power from a UK-type receptacle (see *Figure 1*) located on the backstage wall stage left; this delivered 225 volts of very stable power with a functional ground. The PA system was provided by R & R Associates. Engineer Bob Seah used six E-V Delta-Max (with 15 in. speakers) cabinets, powered by two Amcron MT-1200 amps, per side. We encountered a nasty buzz when we interfaced equipment. R & R's PA system was powered from a different electrical outlet (offstage right), which we discovered was ungrounded. A quick switch to a different receptacle solved our problem. The PA system sounded wonderful in this room and the band was very happy with their sound here. Our first concert was not well-attended, but apparently the "word" got out about the group; our second show was almost sold out, with many local jazz musicians in attendance. Charlie entertained a steady stream of guitar players in the dressing room during intermission, and we all enjoyed the company of our new friends at jam sessions around town that night.

Wayne Toups & Zydecajun performed a single concert at the DBS Auditorium, a small facility seating about 500 people. Located on the second floor of a bank building, all of our gear had to be brought up in a passenger elevator. Designed primarily for business meetings, the small dimensions of the room, coupled with thick carpeting and plush seats, limited reverb time to under one second, ideal for an amplified group like Toups & Zydecajun. The stage was also on the small side; we were able to extend it through the use of carpeted risers. I again contracted R & R Associates for sound, and we used the same sized Delta-Max system. Power came from a wall outlet just off stage left: 230 volts was supplied by this UK-style receptacle, which did have a functional ground.

This concert was a revelation: Singapore audiences are noted for their reserve, but you might have thought we were in Louisiana from the way the audience carried on; the crowd was on their feet for the entire second half of the show (see *Figure 2*)! The band played a short cameo performance the following evening at the Singapore Hard Rock Cafe. R & R had a system in house here as well, with Delta-Max house cabinets, E-V stock wedges, and Amcron power amps, controlled with a Soundcraft 400 console.

Figure 3. AUA Auditorium in Bangkok taken from the balcony. The PA is the black areas on the wall, on either side of the procenium. Center Director Pendleton Agnew is to the extreme left.

THAILAND—BANGKOK

Don Muang International Airport is only 15 miles from downtown Bangkok, but the city's infamous traffic jams insure that the journey will take *at least one hour*, if not more. If you have to depart during rush hour, allow two hours to be safe. Rainy season (June to September) can cause even more delays due to street flooding. Plan accordingly. The domestic and cargo terminals are immediately adjacent to the international terminal, which is handy if you have to make an immediate domestic connection (as we did with Charlie Byrd). Customs clearance is hard to figure: this time I arrived in late evening and was waved through with a perfunctory glance at my paperwork; on previous visits, when the airport was busy, equipment was thoroughly inspected on the way in.

Both Charlie Byrd and Wayne Toups & Zydecajun played the AUA Bi-national Center Auditorium. I'd played this particular facility before, with Shannon Jackson in 1983. Depending on seating configuration, this second-floor hall could seat up to 600 people, including a balcony area (see *Figure 3*).

There was a brand new house PA system; logistic and budgetary considerations dictated I use it for both groups, although I could (and did) configure it differently for each. The system was comprised of two JBL double-15 in. cabinets and two radial horns per side, mounted high in the wall on either side of the proscenium opening. This was powered by two JBL 6260 power amps (see *Figure 4*). I used one amp, powering one speaker and horn on each side, for Charlie Byrd; the entire system was used for Toups & Zydecajun. The acoustics were decent: reverb time was about 1.5 seconds, helped by the acoustic-tiled ceiling and carpeted floor. The hall was wider than it was deep, so echo off the back wall was reduced to manageable levels. I was pleased to note that the electrical system had been completely redone; we tied my transformer input tails directly to the new drop off stage right. This provided 225 volts and a good equipment ground, without the fluctuations in voltage peculiar to Bangkok. When I finally powered up, however, the polarity was reversed at my transformer output. The problem was not in my connection or the transformer: I discovered that the hot and neutral bars were incorrectly labeled at the drop, a problem that had gone undetected until now. I simply reversed my tails to achieve the proper polarity.

Mixing Charlie's trio here was a snap: the intimate nature of the room lent itself well to our acoustic approach. A local reviewer commented that the "balance was so sensitive you could hear the squeak of guitar strings." For Toups & Zydecajun, it wasn't so easy: the group easily overpowered the room with stage sound, so I tried to balance the vocals, accordion, and some drums to the stage level of the other instruments. Toups' concert was taped for public television, so I also had to manufacture a mix for the band. Using the stereo master of my Yamaha console, I fed the PA system with the left output, the audio for TV with the right. By using pan pots, I could choose to send bass, guitar and keys to the TV truck while keeping them out of the PA mix. It took some experimentation to get the TV mix right; we'd play a song, then run out to the video truck, parked outside, to monitor playback. After three songs, and several trips up and down the stairs, we were satisfied with the balance. The show was sold out, and we had another 200 people standing outside the hall, enjoying the concert on video monitors fed by the TV truck. db

(Continued next issue.)

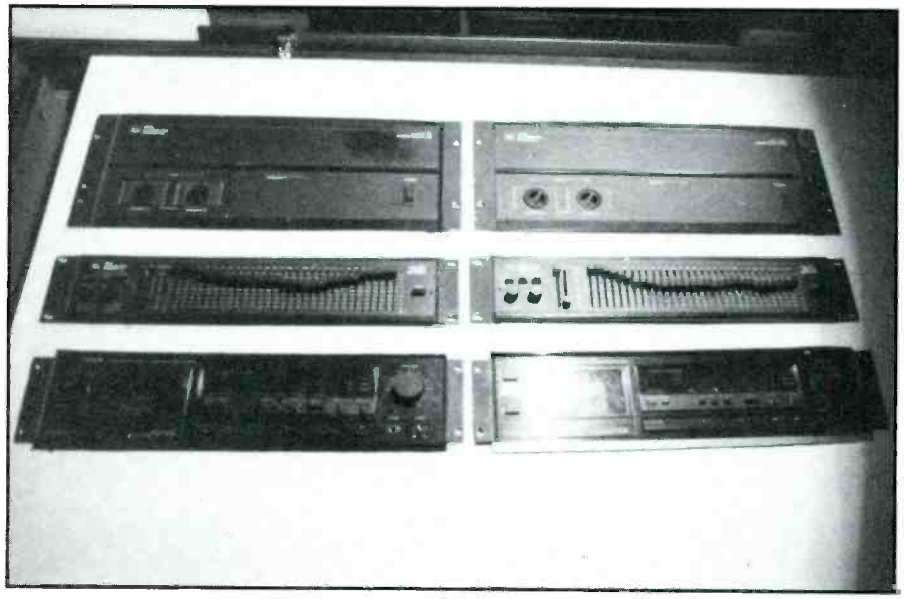


Figure 4. The inside audio control room at AUA, showing system power amps, EQs, and tape decks.

BRUCE BARTLETT AND JENNY BARTLETT

RECORDING TECHNIQUES

How To Care For Your 4-Track

• Your 4-track cassette recorder-mixer is an amazing piece of technology. It's a tool for making great recordings in a portable, economical package, but unless you apply some preventive maintenance, you won't get the full performance your machine is capable of. Plus, a little care goes a long way in preventing breakdowns.

Let's go over some simple procedures to keep your 4-track in good shape. First, identify the parts you'll be working with.

The cassette deck has three main parts: the heads, the transport and the electronics.

1. The heads are the shiny, or black metal blocks that contact the tape. Acting like electromagnets, they write your sound on tape and read it off the tape.

2. The transport pulls the tape past the heads, and it also rewinds and fast forwards the cassette.

3. The electronics inside the cassette deck amplify and equalize the signals going to and from the heads. That is, the electronics make the incoming signal stronger and adjust the highs and lows to compensate for certain losses during recording and playback. We won't be working hands-on with the electronics, but it helps to know about them.

Most 4-tracks include two heads placed left-to-right as follows: erase and record/playback (*see Figure 1*). The erase head produces an ultrasonic, oscillating magnetic field. As the tape goes past the erase head, the tape is exposed to a weaker and weaker magnetic field. This orients the tape's magnetic particles randomly and erases any signal on tape. In record mode, the record/playback head translates the incoming electrical signal into a magnetic field. As the tape goes past the head, this magnetic field aligns the tape particles in a pattern that corresponds to the audio signal. This pattern is permanently stored on tape.

The pattern has a magnetic field. During playback, the tape passes the record/playback head, which picks up this magnetic field, and converts it into a corresponding electrical signal. This signal is amplified and plays through your speakers or headphones.

The job of the transport is to move the tape past the heads. During recording and playback, the tape is pressed between a rotating metal post (called the capstan) and a rotating rubber wheel (called the pinch roller (see *Figure 1*). These pull the tape along. Metal guides alongside the heads keep the tape moving in the correct path. Inside the transport are rubber belts, wheels and gears that transfer motion from the motor to the capstan and cassette hubs.

A special circuit in the electronics generates an ultrasonic signal that drives the erase head. The ultrasonic signal, called bias, is also mixed with the audio fed to the record head. Bias must be added to the audio signal during recording to reduce distortion. The amount of bias, which is adjustable, greatly affects the sound of your recordings. For example, the playback level, tone quality, distortion and drop-outs (temporary signal loss) are all affected by the bias setting. In most recorder-mixers, the bias setting is internal. In that case, you won't be adjusting the setting, but you need to know when it might be out of whack so you can have it serviced.

Now that we know the parts to work on, let's get started.

CLEAN THE TAPE PATH

Over time, dust and oxide shed from the tape build up on your deck's heads. This layer of deposits separates the tape from the heads, causing high-frequency loss (a dull, muffled sound) and drop-outs (quick level drops). In addition, build-up of oxide on the tape guides, capstan and pinch roller can cause wobbly pitch, so it's very important to clean the entire tape path frequently, or before every recording session.

Use the cleaning fluid recommended in your recorder-mixer manual. Denatured alcohol (available at hardware stores or drugstores) and a densely-packed cotton swab are often used. Don't use rubbing alcohol because it can leave a film on the head. Note: Some manufacturers recommend using rubber cleaner on rubber parts — rather than alcohol — to prevent swelling or cracking. Allow the cleaning fluid to dry before putting in a cassette.

DEMAGNETIZE THE RECORD/PLAY HEAD

The record/play head can build up a magnetic field which can partly erase high frequencies and add tape hiss. You can get rid of this magnetism with a tape-head demagnetizer or degausser, which is available at Radio Shack and other music stores.

Essentially an electromagnet with a probe tip, the demagnetizer produces a rapidly vibrating magnetic field. Touch the probe tip to the head to magnetize it, and then slowly pull the tip away so the magnetism tapers off until none is left. It's a good idea to demagnetize the head after every eight hours of use. The technique of using a demagnetizer is critical:

1. Cover the probe tip with electrical tape, if necessary, to avoid scratching the heads.
2. Turn off your recorder-mixer.
3. With the demagnetizer at least one foot from your recorder-mixer, plug it in.
4. Bring the demagnetizer slowly to the head.
5. After touching the head with the probe tip, remove the demagnetizer *slowly* to at least one foot away and unplug it.

In this way, you're inducing a magnetic field, then gradually reducing it to zero. If you touch the demagnetizer to the head and quickly remove it, you'll magnetize the head even worse than when you started!

ALIGN THE RECORD/PLAY HEAD

For the best high-frequency response (brilliance and clarity), the head must be correctly aligned with respect to the tape. Note the gap in the record/playback head. This is the break in the electromagnet, and it must be exactly at a right angle to the tape edge. This adjustment is called azimuth alignment, shown in *figure 2*. You can align the head with the aid of either a standard alignment tape or a commercially-recorded cassette.

Dust off your recorder-mixer when necessary, or use a dust cover. Avoid spilling drinks into it by keeping them away from your work area.

Before starting, clean and demagnetize the tape heads. Look for a small spring-loaded screw next to the record/playback head. This screw affects the azimuth: the left-right head angle relative to the tape edge. If you have an alignment tape, put it in your cassette deck and play the 15 kHz tone. Adjust the azimuth screw for maximum signal level as shown on your meters. If you're lacking an alignment tape, put on a good commercially-recorded cassette that has plenty of cymbals or hi hat. Adjust the azimuth screw to the point where the cymbals sound the most crisp and clear. You'll find a peak in the clarity at a certain screw rotation, and a duller sound on either side of that.

HAVE THE ELECTRONICS CALIBRATED

Before a recorder-mixer is shipped to your dealer, its electronics are factory calibrated (adjusted for optimum performance from a certain brand of tape). If your machine is calibrated correctly, the playback signal should sound just like the input signal (except for some added tape hiss, and perhaps a little loss of clarity or high end). Occasionally these circuit adjustments drift, so the circuit may need to be re-calibrated. It's a complicated procedure best left to a service technician. In fact, some home and semi-pro recorders are not designed for easy calibration. The internal parts to be adjusted may not be easily accessible.

Bias setting is a critical part of calibration. Too high a setting reduces the level recorded on tape and rolls off high frequencies, giving a muffled sound. Too low a setting also reduces the level on tape, results in distortion and drop-outs, and makes the sound too trebly. If you hear these problems, suspect an improper bias setting. Most recorder-mixers have the bias adjustment inside the chassis to prevent tampering, but a few units include a bias knob on the front panel. Adjust it according to the instructions in your user manual.

Normally, it's sufficient to use the brand of tape for which the machine was adjusted, and leave the calibration alone. Gentle treatment will help your machine stay calibrated.

AVOID HEAT, COLD, DUST, AND KNOCKS

Keep your recorder-mixer near room temperature, because too much heat shortens the life of its transistors. Don't leave your 4-track in a hot car or car trunk. Likewise, if your 4-track sits for a long time in a cold car, during winter, the lubricants get stiff and batteries lose voltage. Accordingly, keep it in a warm house, and carry it out to your car just before taking it somewhere. Dropping or knocking your 4-track may cause the heads or other parts to go out of alignment, so treat it gently. When carrying your 4-track in a car, put it on a car seat to act as a shock absorber. Prevent damage by transporting the machine in a padded carrying case or factory packaging.

Dust off your recorder-mixer when necessary, or use a dust cover. Avoid spilling drinks into it by keeping them away from your work area. Don't allow smoking near the machine because smoke particles can get into the controls and make them noisy, or the smoke can accumulate on the tape, causing dropouts or dull sound.

DON'T LEAVE THE PINCH ROLLER ENGAGED

If you leave your 4-track in pause or play mode and turn off the power, the pinch roller will remain pressed against the capstan. (Some models automatically disengage the roller when power is shut off—a welcome feature.) Eventually, the capstan will press a groove into the roller. When the roller rotates, its deformed shape will make the pitch wobbly, so make sure to hit "stop" before turning the machine off.

TAPE TIPS

Your cassettes need special care, too. Keep them out of your car trunk and out of sunlight, because heat can erase a signal on tape or increase print-through. This is the transfer of a recorded signal from one layer of tape to the next, causing an echo or pre-echo. Excessive heat can even melt the plastic cassette housing. Ideal storage conditions are sixty to seventy-five degrees Fahrenheit, thirty-five to fifty percent relative humidity. Keep cassettes in their boxes to prevent dust build-up. Also, prevent tape erasure by keeping cassettes away from magnetic fields caused by degaussers, speakers, headphones and telephones. Let your tapes go through the X-ray machine in airports rather than carrying them through the metal detector; the X-ray machine does not erase tapes, but the metal detector can.

TROUBLESHOOTING

Listed below are some common 4-track problems, causes and solutions. If you need parts, phone the manufacturer of your 4-track to order them.

If you hear a scratchy sound when adjusting the mixer controls, they are dirty or worn inside. You might try spraying some contact cleaner in the potentiometers.

Wobbly pitch may be caused by a flawed cassette, a dirty capstan or pinch roller, a worn-out pinch roller or belt, or a defective servo circuit which controls the motor speed. First, try another cassette. Clean the tape path. To prevent roller wear, hit "stop" before turning off power. If the problem persists, replace the worn-out pinch roller and belt. If the problem still persists, have the servo electronics and bearings serviced.

Tape hiss is common in inexpensive machines, but it may become excessive. First, make sure you're recording at normal levels, you've turned on noise reduction during recording and playback, and you've turned down the gain trim (input attenuation) just enough to prevent input-overload distortion, but no more. If you do all of these procedures correctly, but still hear excessive hiss, try some other solutions. Try different brands of chrome or metal tape. If that doesn't help, your bias setting may be off, or your internal record-level setting may be too low. If this is true, the playback level on the meters will be several dB lower than the input level. Have the electronics calibrated.

Dull, muffled sound might be caused by dirty or misaligned heads, or electronics going out of calibration. Clean and align the heads. Make sure your high-frequency EQ knobs are not turned down. If these suggestions don't work, you need to have the electronics calibrated.

A "phasey" sound where the cymbals fade in and out indicates that the head is misaligned or that the tape is not traveling straight across the head. You could try another cassette, adjust azimuth alignment, or the tape tension could be off. If so, have the unit serviced.

You might notice a distorted, over-trebly sound on playback, even though you recorded at normal levels with EQ knobs set to flat (straight up). This problem is likely due to the bias level being too low or the internal equalization being maladjusted. Have bias and equalization checked.

If you hear a scratchy sound when adjusting the mixer controls, they are dirty or worn inside. You might try spraying some contact cleaner in the potentiometers. These are the flat can-shaped components under each knob. Sliding faders may need cleaning, too.

By taking these steps and administering some tender loving care, your 4-track will keep on making the same high-quality recordings it made when it was brand new. db

JOHN BARILLA

Hot Tips: Designing Vocals: Part I

• It is amazing how excellent musical tracks sound today, even if they were recorded in the humblest of electronic cottages. Very often, the tracks sound like they were recorded in a state-of-the art facility, and fact is, they were. If you are doing electronic production using drum samples and high-tech 1990s synthesizers, you are effectively accessing the studio in which those samples were recorded and the expensive electronics through which the signal was processed, not to mention the expertise of the engineers who captured and packaged the sound for you. A drummer and a team of engineers might labor all day long to get a digital recording of the quintessential snare smack, and you simply push a key and there it is, perfect performance every time. It is not hard to get great sounding tracks these days; that aspect of electronic recording is almost idiot-proof.

There is one aspect of recording, however, that is definitely not idiot-proof—recording vocals. It is lamentable how many otherwise decent productions are mired by inadequate vocal tracks. It is here that the art and science of engineering comes to the foreground. Unlike synthesizers and drum machines, you cannot push a button and get excellent vocal tracks. It takes both technique and musical sensitivity to elicit and capture a good vocal performance—and it is not easy to do. There is a long chain of events between the very thought of a sound originating in the psyche of a singer to the final placement of the recorded vocal in the mix. You are dealing with the intersection of tangible factors (such as amplitudes and frequency of waveforms) with intangible factors (such as the emotional complexion of the singer at a given point in time). The difference between an excellent vocal track and a horrible one can often boil down to knowing when to try another mic or offer the vocalist a cup of tea with honey and lemon. The point is this: there are a tremendous number of factors involved in producing vocal tracks, and the more factors you are aware of, the greater control you can exercise. Good vocal tracks are constructed thoughtfully from ground floor to rooftop. That is why this article is entitled "Designing Vocals."

TWO SIDES TO EVERY VOCAL

It is helpful to keep in mind that all the complexities of vocal design fit under one of two banners: the technical side and the talent side. Ideally, both sides should be equally considered, but all too often the system is out of balance. Two exaggerated examples: the inspired but impetuous musician-type who records a vocal without sufficiently testing levels. He ends up with a great, fresh sounding performance, but it is noisy or distorted. The opposite approach is the technoid engineer-type who is ultra careful and clinical in recording.

Unfortunately, he may be so busy trying to craft the sound that he is oblivious to a peak vocal performance while it is happening. The vocalist, now blown out and frustrated, is often unable to recoup, although she continues to sing. The result is a clinically perfect, but emotionally dead vocal track. Both extremes are to be consciously avoided, and the best way to do this is to go through a mental checklist of both technical and talent factors which need to be considered when designing vocals. Some technical factors include mic selection and placement, accoutrements (pop filters, acoustical baffles) and signal processing (pre-amps, compressors/limiters, etc.). Some talent factors include vocal quality, vocal problems (intonation, range), the emotional environment (mood) and performance strategy. Let's give each of these factors a closer look.

MIC SELECTION

The technical side of designing vocals begins with choice of mic. While the pundits might argue about which of the world-class mics is the superior vocal mic, most experienced engineers will agree that different mics work best for different voice qualities. Sometimes a relatively cheap mic will render a more appropriate sound than an expensive one. Some professional singers, once they stumble upon a unique studio sound, carry around their favorite mic from session to session. Often it is the mic's unique limitations that match it so perfectly with a certain voice.

For example, classic old-fashioned ribbon mics have a characteristic low-end resonance and high-end roll-off which lend them a reputation of having a "warm" sound. But a warm sounding mic is probably not the best choice for a warm sounding vocalist—the result might be rather dull sounding; but for a brilliant vocalist whose timbre sits right on the border of shrillness, the ribbon mic will be an excellent compliment, smoothing out lots of the brittle peaks associated with high voices.

Hitting on the most appropriate mic can save you lots of time later on in terms of experimental EQ and signal processing. Getting it right from the beginning is always the best way to go. If you cannot afford a selection of quality mics, get the one or two best mics you can afford and learn how to use them. You may have to make certain compensations for their deficiencies in frequency response, but you still can get a very acceptable sound from a moderately priced vocal mic. To save time, write down typical equalization settings for different voices so you do not have to experiment every time you record a vocal.

MIC PLACEMENT

Once you have decided which mic to use, focus on the appropriate placement of the mic for that particular vocalist. All kinds of questions should be asked here. For example: does the vocalist have a strong, dynamic voice or a softer, more evenly textured voice? (The dynamic vocalist should probably be placed further away from the mic in order to take advantage of the acoustical compression that occurs when a sound wave is forced through air. The softer vocalist does not need the acoustical compression, but instead needs to be placed closer to the mic to increase the signal-to-noise ratio, and also so he can "work" the mic if he should need to compensate for some lack of dynamics.)

Other questions also need to be considered when determining mic distance. For example, how much of the room (reflected sounds) do you want coming into the mic? If the answer is, as little as possible, then the vocalist should be placed closer to the mic. Occasionally the room sound may be flattering, in which case distance is the choice.

MIC ANGLE

The angle of the mic—relative to the stream of air coming forth from the vocalist's mouth—is also a factor. The vocalist does not always need to be on-axis with the pickup pattern of the mic; the only requirement is that he consistently be in the same relative position. By adjusting the mic angle, a certain amount of natural equalization can be achieved. One of the most useful positions is to have the mic raised to approximately eye level and pointing downward towards the vocalist's nose at about a 45 degree angle. In this way, the real brunt of the air pressure evades a direct hit on the mic capsule, avoiding explosive pops. Other positions may be beneficial, so you must experiment here.

Once you have locked into a sound, it is important to be able to retain it for the entire session—through punch-ins, through coffee breaks, and so on. The best way to achieve that is to note the position of the vocalist's feet relative to the mic. A strip of tape on the floor will help the vocalist's return to the same position later on.

ACCOUTREMENTS

This is a catch-all phrase referring to all the little gizmos—like pop filters or windscreens or shock mounts—that help keep unwanted impulses away from the mic. It can also refer to little necessities like music stands which can sometimes wreak havoc with the sound, or things like sound absorbent baffles or hard reflective panels which can selectively be moved to alter room acoustics. Accoutrements are all the physical things relative to the mic that have an effect on the sound. These may seem like little things, but they have a way of factoring into the equation.

For example, note the music stand mentioned earlier. When used by vocalists, music stands have a nasty way of reflecting out-of-phase signals back into the mic, making the vocals sound a little tunnel-like. The heavy duty sheet metal construction on some music stands may subtly ring at resonant frequencies, adding another problem to the sound. What are you going to do? If someone needs a stand, you must give it to him, but you have to locate it in such a way that it does not deteriorate the sound.

On the positive side, an accoutrement is usually a helpful item—like a pop filter. Simply a hoop with some nylon hosiery stretched over it (see **db Magazine's** July/August 1989 issue for a DIY version of this item), the pop filter (when attached several inches to a foot in front of a mic) effectively diffuses hard blasts of air—called "plosives"—which would normally wreak havoc with the mic. It is a great way to alleviate "pops" and also provide a way for the vocalist to mark his position (for example, one handspan in front of the filter). On the other hand, pop filters cannot be used in every case. If, for example, you are going for a very intimate "in-your-face" sort of sound, the pop filter will not help. For that kind of revealing sound, the singer must be on top of the mic, and a pop filter is not very effective when brought in this close. Yet, believe it or not, a pencil taped right across the middle of the mic's grille screen will effectively reduce much of the popping associated with close-up singing.

Shock-mounts (a bunch of glorified rubber bands suspending the mic) can be very helpful if your client likes to dance while he sings, because sometimes vibration from the floor can travel up the mic stand and modulate the input signal, causing distortion.

Sometimes studios have moveable baffles or even walls that have been covered with sound absorbent material, and sometimes the sound absorbers can be selectively moved out of the way revealing a hard, reflective surface. The ratio of hard to soft surfaces (reflection versus absorption) is a characteristic that can be used creatively to color the vocal sound in subtle, but quite audible ways. Portable baffles (gobos) can also be moved in or out from the vocal area, helping to redefine the acoustical space. The point of this section is simply that you need to be aware of the small factors that can add up, either enhancing the sound or detracting from it.

SIGNAL PROCESSING

Thus far, the focus has been on what occurs at the mic or in the environment around it, but what goes on at the other end of the mic cable is of no less importance. The object of the game is to get the lowest noise signal recorded on tape at the highest possible level (short of distortion), and this is true for digital as well as analog recording.

THE CHAIN OF COMMAND

In the chain of command from mic to tape, probably the single most important element is the mic preamp (usually found in the front-end of recording consoles). While mics have a certain amount of inherent noise and vary in their output levels, the most significant amount of gain is taken at the console mic preamp (perhaps 40 or 50 dB). The preamp then, must be an extremely clean unit; it must not contribute any significant noise to the signal. Since the signal-to-noise ratio of the entire chain is established here, it is important to find out whether the pre-amps normally installed on your board are adequate for the job.

On some lower priced consoles (which happen to be perfectly clean with line-level signals), the mic inputs can be a bit noisy. (That is how the price is kept so reasonable; the manufacturer skimps somewhere, and he does it on the mic pre-amps.) If that is the case with your board, get yourself a stand-alone mic pre-amp that has respectable noise specs. You only need to get one good pre-amp just for vocals (you do not need sixteen or however many inputs your board has). Even with a good mic pre-amp, you still need to learn how to operate it properly to absolutely maximize the signal-to-noise ratio—so be prepared to experiment; the results will definitely be worth the effort.

Finally, after going through the pre-amp and perhaps a channel on your board, the vocal signal must be recorded on tape. Here, too, we want to maximize the signal-to-noise ratio, or at least (especially in the case of digital) not crash the headroom of the recorder into distortion. Here compression and limiting are of some practical value. While classical purists would positively gag at this thought, I personally would never even think of doing a vocal session without applying some limiting or compression to the signal. For any pop music application, restricting the dynamics is always beneficial. While this can certainly be done during the mix, there is great merit in applying at least a modicum of limiting or compression while recording the track.

The question is, when do you use limiting and when do you use compression? The answer usually lies in understanding the overall impact of the production itself. For example, if the musical tracks are constant, driving and rather unidynamic—such as dance music or intense rock 'n roll, then your vocal must also be unidynamic if it is to be heard through relentless density of music. Here, compression (set with moderate ratios, but very low threshold) is the choice. This will smash down loud passages while effectively raising the relative level of soft passages. The dynamic range will be rather narrow, facilitating a hot signal on tape (while recording) and later upon mixing, making it easier to place the vocal within the intensity of the music.

On the other hand, limiting is usually preferable when the song itself contains lots of dynamics. A good example would be a sensitive, romantic ballad which utilizes the technique of "additive production" (for example, starting off with only a piano and ending up with an orchestra). In situations like this, where low level dynamics need to remain intact, it is good to simply put a ceiling on the output by applying limiting (high ratio and high threshold) to the signal. This prevents overstepping the headroom of the tape by trapping any ridiculous peaks, while still leaving the lesser dynamics in tact.

All the above-mentioned technical considerations are important factors in getting professional sounding vocal tracks, but almost any recordist can attain excellence in these things with proper equipment and proper training. Where most recordists fall short is in eliciting a great performance from the talent. Even the cleanest recording will not be esteemed if the performance is lifeless, and while a vocalist may be an excellent singer, it may take some interpersonal diplomacy to have that person "open up" in front of the mic. It is this important area that will be discussed in Part II of "Designing Vocals." db

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THE ELECTRONIC COTTAGE

MARK MAULUCCI

Drummers And Bands In The MIDI Studio

• Hello! In last issue's column about the guitarist in the MIDI studio, I discussed the integration of the "organic"- minded guitarist into the more hi-tech environment usually associated with the keyboardist. In this issue, I will take this concept one step further: the presence of drummers and whole bands in the MIDI studio.

When MIDI exploded in the mid-Eighties, one of its most attractive attributes was that one person could do it all. The individual could compose, arrange and perform a piece in its entirety without the aid of others. I must admit this appealed to me greatly. I'd been a songwriter for many years and had long struggled with obstacles in just trying to record demos. It was liberating to be able to program the drums, bass and keys without having to rely on others with busy schedules and strong opinions. I felt very self-sufficient and inspired. I could do it myself at my convenience, developing my engineering chops along the way. I had long dreamed of having a home studio and lo and behold, it became a reality.

For about five years, I wrote and recorded many of my songs. I constantly upgraded my gear since there was always some new, must-have toy coming out. Remember when there were two NAMM shows a year? One could barely keep pace with the rampant technology.

After a while, an interesting thing happened. The technological advancements slowed down a bit and so did my desire to do everything myself; the novelty was wearing thin. Being a roots-oriented guitarist, I found myself going back to the basics. Although I regularly performed with my band and had recorded a single at a professional multi-track studio, the bulk of my recording had been a solitary experience for five years. I decided it was time to invite others in to contribute to my music; I was tired of programming all the drum and bass parts. Think of all the lame drum patterns we've heard for the past six or seven years! Granted, sometimes we non-drummers can be very effective with a machine. I've always been of the "whatever the song needs" school, but guess what? Drummers do it better.

Every band I've ever played with was only as good as the drummer. He or she doesn't have to possess the technique of a Buddy Rich—they just have to swing. I mean, look at Charlie Watts! The guy just pushes the Rolling Stones. Sometimes, being a hair behind the beat is what makes the difference.

To be sure, some manufacturers have become hip to this. Roland has released the R-8 and R-5 human rhythm composers. They afford the programmer access to parameters such as nuance—where the stick strikes the drum or cymbal. This offers amazing variation in tone. "Feel" parameter changes can be programmed according to a set definition of values using the "Groove" function, or in a random fashion using the "Random Factor." The degree of randomness is user defined and allows one to create realistic rhythms such as a 16 beat hi-hat pattern with "human" accents and inflections. These settings can be assigned to a "Feel Patch," making it possible to produce incredibly varied performances from a single pattern!

The pad is the "brain" and can store 64 custom setups with a RAM card slot for 64 more. I use two Roland PD-21 pads and a DW-5000 bass drum trigger pedal.

Over the years, many drummers have been resentful towards or intimidated by drum machines. Ironically, they often make the best programmers. The drummer who realizes this is a well-equipped and versatile player. A producer friend of mine who is also a fine drummer brings both his sticks and drum machine to sessions. He's prepared for any situation. Even with the "Human Feel" function on my drum machine, I still have to do it myself unless I bring in a drummer. Enter the greatest invention since sliced bread: the pad controller. Just as the guitar synthesizer allows me access to the world of MIDI, so does the pad controller allow the drummer and percussionist. Even drummers who don't program can't help but be thrilled by being able to trigger any number of sounds in real time.

A CHANGED MAN

The pad controller changed my life; I live in a quiet residential neighborhood. I'm not really set up to mic drums in my MIDI studio; that requires a lot of space, a small fortune in mics and most importantly, great mic'ing techniques. Engineers spend years learning about mic placement and the tuning of drums—prerequisites to getting a great drum sound.

With the pad controller, one merely strikes a pad to trigger the sound source. All the painstaking work of getting optimum sounds in lab conditions has already been done for us. There is a wealth of 16 bit sounds on the market in drum machines, samplers, synthesizers and various expansion cards and disks. Getting a great drum sound becomes so simple and easy—no muss, no fuss.

I have a Roland PAD-80 Octapad in my studio. It has eight pads and six external inputs for a total of fourteen possible triggers. Each one can be assigned to any MIDI channel and note number. The velocity and curve and MIDI panning are all adjustable. It's possible to layer up to three voices simultaneously as well as mix or switch according to velocity.

The pad is the "brain" and can store 64 custom setups with a RAM card slot for 64 more. I use two Roland PD-21 pads and a DW-5000 bass drum trigger pedal. I find this to be a fairly universal set for most drummers. They have a real kick pedal, snare pad and hi-hat pad which can be layered for open and closed sounds. I then assign the eight pads on the Octapad to various tom-toms and cymbals.

A friend of mine uses the KAT pad controller in his studio to much the same effect. The point is, drummers can come in and play in a MIDI or direct recording situation. While it's possible for the drummer to play the parts right into the sequencer or drum machine, I find myself encouraging real time playing like in the "old days." The drummer, bass player, rhythm guitarist and/or keyboardist can all lay down tracks together. I produced a sax player friend's session this way. It has a groove that is very difficult to simulate via computers.

I recommend MIDI studio people explore pad controllers. It can open up your studio to a whole different clientele. db

LARRY ZIDE

Sound Reinforcement in The Park

LAST FALL, I TOOK A PLEASANT AUTO trip from **db Magazine's** Long Island base by ferry across Long Island Sound to Jaffe Acoustics' home base in lower Connecticut. There, in a handsome suite of offices, I found David Robb, senior consultant to the firm, waiting for me. What follows is an edited transcript of our conversations about the new sound system that was unveiled during the summer of '90 in New York City's Central Park.

db: How did Jaffe Acoustics get started in sound in the park?

DR: I guess that was about twenty years ago or so. In addition to the six columns above the music shell Jaffe also did a primitive delay. There were one pair of Bozak columns out just 100 feet or so, that were delayed. And of course, at that time there were no digital delays, so there was a Phillips tape loop delay to hit that delay line. And that worked for five or six years. And then at that point, they built the Interim system which is the one that was in use in the park up until last year. I think of that system more as a rock 'n roll kind of system what with large towers that hydraulically lifted up on either side of the stage and were thirty feet or so tall. It also had a number of components virtually all horn-loaded that kind of brute forced the sound from either side of the stage out to the areas of the park. They did not use any delay at that point.

db: Let's go right smack up to the present system and that day this past summer when we were all in the park for a New York Philharmonic concert that was threatened constantly by the sky.

DR: Yup, rain right up until 8 o'clock show time. Well, now we get the new system. The concept had been in discussion since 1980. There were some real design document stuff drawn up. There were some sketchy things that were drawn up I think as early '84, and we actually generated the initial equipment specification around '86 for this project. I was not at Jaffe Acoustics at that point in time. Our project manager was Chuck MacGregor who got a lot of the initial ideas all organized and passed on to the City and City's representatives. There were a number of different demonstrations done of this particular type of system and how it might work.

db: Well, the whole concept of the new system that is unique is the concept of it being wireless.

DR: That's a part of it. I think that's a necessary part of it because really, what we wanted to do, which is where the uniqueness comes in, is to provide both a distributive sound system so we no longer had to brute force all the sound from the stage out to the 700 foot mark, but that we could provide speakers that did not have that horn sound; that was one thing that everybody, critics and audience alike, really objected to with the past system. They wanted to sort of return to the kind of line array idea with Bozak, but they realized that it just couldn't work out at 700 feet. So the distributed system was really, I think, the idea, the concept for this, with the addition of the ambient speaker broadcasting stuff as well, and because the fact that it was going to be a distributed system, the Parks Department itself actually forced upon us the concept of wireless. I would personally just as soon have run wires for all these things. Logistically it would have been a bit more difficult, but it would have meant a whole lot less equipment.

I'd much rather trust to even a generator providing AC power than I would a whole bunch of batteries, but the Parks Department would not allow cabling to be either strung or laid on the ground, and they would not, even if they knew it was going to happen year after year, allow any permanent in-ground things to be done. So we only had one choice left and that was to broadcast.

db: So the design itself, the entire concept was done in-house, but then to execute it you had to go outside.

DR: Yeah, As we're only a consulting firm, we wanted, we needed, a high quality contractor. The manufacturer of the boxes was Eastern Acoustic Works, and Maryland Sound Industries was the contractor who actually made the boxes to then pass on to the City.

db: So then maybe right now what you ought to do is describe the system that now has been, I understand, accepted and will be used by The City of New York Parks Department.

DR: Okay. We started to talk about needing to find the person to build this for us. When we as consultants put together a specification, we like to make suggestions on who the client might use to implement all of our designs. And for this particular project, it was a bit different. We didn't go to the standard sound contractor type of person. We took three top people that were within several hundred miles of New York City.

db: You didn't see any point to going to West Coast sound people?

DR: No, I'll tell you one of the criteria that the City placed upon the project was that the selected contractor they would have preferred would be a New York City contractor just for paperwork's sake, but the City insisted upon and during the course of the project an office and a representative or somebody that they could talk to locally if they needed something done immediately. Maryland Sound ended up with two people that worked almost, it seemed like almost full-time, just doing paperwork with New York City. Very complex.

db: So Maryland Sound in effect became the actual contractor that did the project and submitted it to the city.

DR: Yes, that's right. They were selected during the bidding process. We had an initial round of bids and the bid did go out to several other people that requested it, but no one else chose to bid. It was obvious that it was a fairly complex project. So in the end, the only people that bid were the three that we felt could properly handle it.

db: We know, of course, that Maryland Sound did, in, fact build the system. Had you worked with Maryland Sound before?

DR: I've never worked with Maryland Sound as a contractor, but I was very familiar with them as a touring sound reinforcement company and you get to know what their technical levels of expertise are besides seeing what their touring equipment is. I've also visited their facilities. I guess I have not visited the amplifier manufacturer Audio Analysts, but I had visited Maryland Sound before this project went out to bid.

db: Interesting. Anyway, you were explaining how...

DR: The bid process—we went through a couple of rounds there as the initial prices came in and then there were about a half dozen options that were also to be bid upon. There was going to be a certain amount of time to do some experimentation between ourselves and the contractor before we actually decided what we really wanted. But it got late enough in the game that we really couldn't afford to do that anymore, so we ended up selecting a company.

Figure 1. The New York Philharmonic plays in Central Park. This photo was taken in 1965, the shell is not the current one.

We had three different options for speakers when it first went out to bid, and during the bid process, the New England Audio Resource box that was still being considered, which is basically that same Bozak-style thing, was deleted. And the contract was awarded with two remaining boxes, one of them being the one that ended up in it and the other one was another custom arrangement of a line array of JBL components that was custom-made by Maryland Sound. Part of the contract was to build one of these boxes so we could listen to it and then we would decide which way to go. That's how the whole bidding process was put together.

db: Alright, but Jaffe Acoustics was the one who was in effect selecting the contract. Is that correct?

DR: I'd have to say the City made the selection, but we had a heavy amount of input on it. Typically, in a construction job the client hires a general contractor that's going to oversee all these different things. Well, the City's representative, Peter Wexler, is not a contractor—he was just an overseer, an intermediary between all the different people involved and the City. He had the good sense to realize that he was not skilled in audio terminology; no one knew enough about the technology better than we did. In addition to our design of the project, we were also hired separately to then help oversee the construction of the project as if we were the general contractor for the sound contractor. So as a part of that process, I evaluated all of the contract bids. I helped chop dollars here and there, and ultimately, I made my recommendation that the City go with Maryland Sound based on the information that came in, and the city pretty much rubber stamped that. So yeah, we did it, but it was the City's decision.

Within the stage area itself, we've done a great deal of internal audio wiring so that there aren't a million mic cables running all over the floor, or speaker cables or intercom or whatever.

db: Then you were going to come to actually describing what the system is and how it came about. We want to graphically show that of course, but describe it as well.

DR: The system consists of, I guess, six general areas. There is a lot of wiring that, in addition to the sound system, we're talking about a whole performance package here that was all designed and built at the same time and is meant to interface with each other. As a matter of fact, parts of it can't work without other parts, and that includes a giant orchestra shell, a staging area

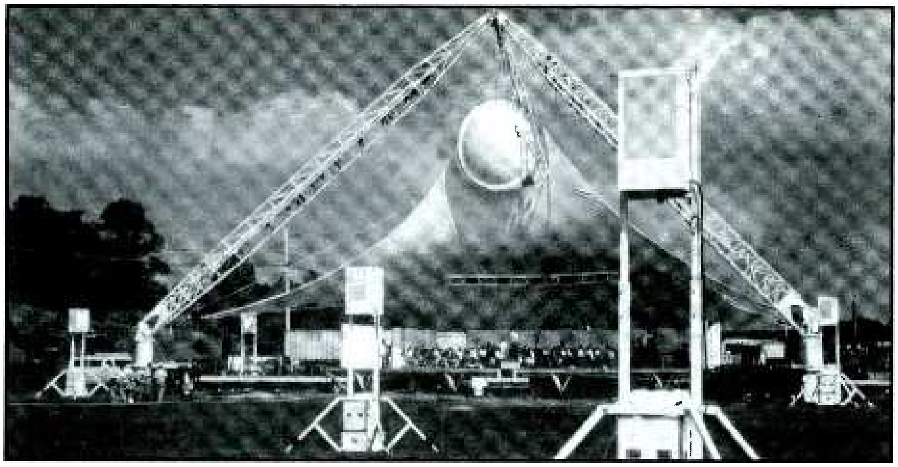


Figure 2. The wireless stands under test. The current stands are slightly different.

and roof and lighting system and projection system, along with the sound system.

Within the stage area itself, we've done a great deal of internal audio wiring so that there aren't a million mic cables running all over the floor, or speaker cables or intercom or whatever. Many of the portable stage sections have wiring built into them which are all externally interfaced to the sound system. That's one part. The next part would be the control booth area, located out about 200 feet. There's a mixing position and there are three booths that are attached to each other out from the stage. One of them is for audio, one's for lighting and one of them is for projection and the stage manager.

db: Now that is in fact hard-wired back to the show, the stage area.

DR: It's a typical snake. We have fifty-two channels of audio running to and from the stage and there is power also run out to that whole area.

db: Well obviously the console needs power and in fact many of your microphones need power.

DR: This is true. The reason that we were allowed to run cabling to that position is that there are police-barricaded emergency routes that are the same at every one of the locations in which this facility will be used. One of those routes runs from the center of the stage straight out for 200 feet. There's a 20 foot wide barricaded-aisle there that is there for police and emergency vehicles. In that aisle is where we run all of those cables out to the house mix position. At that point, that aisle then becomes a cross aisle and goes left and right from there and doesn't continue straight out into the audience. We could not keep going on out as you'll further see, and connect to other speakers that are on out there, but that was something that was already existing in the current setup and we took advantage of that to run our cables in that area.

db: Are the cables just laid on the surface?

DR: Yes. Going down one side of that aisle are the sensitive audio cables and going down the other side of that aisle are the data lines for the lighting and the power. So we've got a 15 foot separation or something like that between those singles until they get out to the console area. So that's the control area out there, then we have the speaker systems themselves. The systems are identical. There are twenty-four total. They are a formidable little package of stuff, and collapse down to a height of approximately 9½-10 feet and they're about, let's see, three foot square of a footprint in their collapsed position. That's how they're transported on a truck. When they are erected out in a field, the footprint becomes an eight foot square, and the top of the unit is just about 15 feet tall. What this is is a mechanical metal frame, mechanical structure of folding legs and a winch-activated lift mechanism that helps to erect this thing, and it consists electronically and acoustically of three subsections inside this metal frame.

The first one at the base of the tower is a case which holds the power for the tower. There are six 12-volt batteries in this case; they're deep-discharge marine-style batteries approximately one hundred amp hour per battery, which gives us a total of six hundred amp hours worth of power coming out of there. There are fuses in the base of this thing as the contractor found out during construction. Six hundred amp hours of power is a heck of a lot of power and they had a technician working in the battery compartment who welded his wedding ring to it. Anyway, there are some large fuses that are knotted down in the battery box, and they have a very high current of well-protected weather-proofed connectors that comes out of that compartment and goes on into the next compartment above that. This is where all of the electronics are. That's a standard electronics rack with rackrail in front and back and has removable front and rear covers.

We go into a delay, and from there we go into electronic crossover, from there we go into power amplifiers and then speakers. But there are subchains of things that happen here.

In addition to those covers, there are also several little (we're just calling them blobs) control compartments that stick out off the side of this thing that have access by keys so that once the whole case is sealed up, the only access to it is through a small door which only the technicians have keys to. Since the stuff will be sitting pretty much out in the audience much of the time, the electronics package consists of a receiver that receives a wireless signal. From there we go into a crossover which divides our signal up to a delay system. There are multiple outputs from the delay. We go into a delay, and from there we go into electronic crossover, from there we go into power amplifiers and then speakers. But there are subchains of things that happen here.

db: Let's talk about the actual components being used, whose amplifiers, whose crossovers, why, etc.

DR: Okay. Before I forget it, because it doesn't show up prominently in the block diagram, also in that electronics rack there, we supplied all of the devices that are in there that were chosen to be able to run off of the 12 VDC. Only the delay and the crossover require 120 volts. There is a converter in there that provides 120 VAC for that. And also in there, taking up a very large chunk of space, is a battery charger that will accept 120 volts and charge the battery pack back down through the cable. At the same time, it will also provide direct 120 volt AC operation of the whole system by working backwards through the battery charger so if any, heaven forbid, if any of these systems were to become so discharged and we did not have enough time to recharge them before the next performance or something, then they all have an external AC outlet that would allow us to direct power right out of the wall.

db: While they are performing would they also be trickle-charged at the same time?

DR: Yes. That is correct. A couple of the other safety things that we show there are listening tests we did with the amps to determine at which point during the discharge of the batteries that the quality of the amplified signal became poor enough that we would rather have the speaker turn off then to listen to some kind of distortion. And we chose a level at which there is a protection circuit in there that once the battery discharged at a certain point instead of continuing the discharge to put out a distorted sound, it just shuts the tower off. So that's a safety thing. Another safety is we have two channels we can broadcast on only one at a time, but each tower can receive one of two different frequencies being broadcast to it. We could potentially set up part of the tower to receive on one, and part to receive on the other.

db: But stereo could be done.

DR: It could be done. We do have transmitters from the mix position that can transmit on both frequencies simultaneously. As a matter of fact, that's how we do checks during the day at the location that we choose; whichever is the better quality signal at that point so during the day we can broadcast on both channels and we can go to a tower and listen as it receives on either channel and choose which one we think's the best. At that point, all the towers would be switched over to that frequency. Each tower has a pair of antennas, as a diversity system.

db: Can it be assumed that this is a fairly closed transmission system that would intrinsically be fairly immune to outside RF interference?

DR: You might assume that. New York City is probably second only to Washington, D.C. in saturation of the air waves. We checked out all the different possible ways of doing wireless here. We've had demonstrations by various manufacturers, we've talked about microwave, we've talked about everything for this, and so the one thing I thought was going to be the closest to being really foolproof were actually stymied by the FCC. We applied for permission to use the bands that are assigned to public broadcasting because all these concerts are free, it's classical music, and we thought they would give us permission to use those frequencies. They kind of led us on for a while, but then they shut us down. So what happens is we were kind of thrust at the mercy of where everybody else is, where all the Broadway plays with twenty-six wireless mics are, and everything else is, and we just had to figure out where was the cleanest area and where were we least likely to get the interference.

We do know what the frequencies are, we know what to watch out for, and obviously we, if wireless mics were to be used in this stage we would find out that we could choose wireless mics that would work with the system.

To do that, we selected our wireless manufacturer, Sennheiser, partially because they seem to have the best product going. We looked very strongly at Sony as well, but we ended up with Sennheiser and, I think, due in no small part to the fact that they're so nearby right here in Connecticut and are very familiar with the New York City area. They have repeatedly gone down at the request of various clients to do frequency searches of what goes on in New York City in certain various areas and to help select the frequencies that would be best used. And so, we relied almost exclusively on their knowledge of all that to help us select the frequencies in which we were going to broadcast.

After Sennheiser made their selection, we got samples of the two transmitter receiver units that we were choosing; one for each frequency. Maryland Sound, the contractor, sent out a person along with a representative from the City, and they went to every one of the twenty-four sites at which these broadcasts would be done to investigate and do testing to make sure that these things would in fact work there. There was no problem at ninety-five percent of the sites. The only problems we had were when we were really on fringe areas where the client had asked us if we would check out if it is it feasible for us to play in New Brunswick, NJ or up in Bridgeport, CT or anywhere. Those were areas where we started to get some interference because there were other well, there were other things happening.

db: How about wireless mics on stage?

DR: There's a lot of research that was done into RF for this job. We do know what the frequencies are, we know what to watch out for, and obviously we, if wireless mics were to be used in this stage we would find out that we could choose wireless mics that would work with the system. There are many, many, many frequencies out there that are available. The wireless technology is very good nowadays so we could be compatible. If we could get two dozen mics at the same time on a Broadway stage, you know there's a lot of frequencies out there. But it still can be made to work. db

(To be concluded in the next issue.)

TEK TEXT

DAN ROGERS

Speaker Placement III: Calculating Loss and Uniformity of Coverage

SOME THEORY

• This third program in our series is quite a bit different than the previous two. The first major difference is that you must enter more information than before. This also makes the actual calculations simpler.

The second major difference is that the program does not use a –12 dB coverage angle, but refers to a "first listener complementary coverage angle." I mentioned last time that there is a fixed relationship between the –6 dB coverage angle and the corresponding ideal -12 dB coverage angle. If the speaker location is not correct for the -6 dB angle, the actual first listener will not be on the -12 dB axis. The dB loss values displayed for the first listener must have the loss for the calculated angle added.

For example, if the first listener loss is –26 dB+AXIS, and the first listener complementary coverage angle is 130 degrees, then the speaker's polar response chart must be checked for the off axis loss at 65 degrees off axis. If that should be 10 dB (for example), then the total loss equals –26 dB and –10 dB for a total of –36 dB.

If the speaker has a real wide –6 dB coverage angle and/or a bad location, then the entire audience may be inside the –6 dB angle. This may seem innocent at first since the coverage appears to be ±3 dB. However, the loss due to distance may create horrible variations between the first and last listener. Also, consider where the remaining off axis sound is going. The second major problem is that the loss from the speaker to the audience may be the same as the loss from the speaker to the microphone. This has a tendency to reduce the useable gain before feedback to near zero. Problem number three is that the back wall and the rest of the room will become a reverberation chamber. Such a set-up is somewhere between a distributed overhead system and a central overhead cluster. Not a good application of theory, to say the least. The program will give a message if you break the rules.

Figure 1. The sample screen.

```
                                                               08-03-1990
P.A. SPEAKER ANGLE LOSS CALCULATIONS                           07:32:41
=========================================================================
H SPKR TO LAST LIST:  120 ft     0 in   V FLOOR TO AV LIST :   4 ft    6 in
H LAST LIST TO WALL:   10 ft     0 in   V FLR TO CENT SPKR :  20 ft    0 in
H 1ST TO LAST LISTN:  100 ft     0 in   ON AXIS SPKR TO REF:   4 ft    0 in
                                        SPKR -6dB COVR ANGL:  20.00 degrees
-------------------------------------------------------------------------
            VERTICAL     HORIZONTAL    THROW         THROW       VERTICAL
            HEIGHT       DISTANCE      DISTANCE      dB LOSS     TO THROW
SPEAKER TO: Feet   In.   Feet   In.    Feet   In.    decibels    degrees
- - - - - - - - - - - - - - - - - - - - - - - - - - - - - - - - - - - - -
LAST LISTENER :  15   6    120    0     121    0     -29.61        82.64
-6dB LISTENER :  15   6     49    7      51   11     -28.27        72.64
FIRST LISTENER:  15   6     20    0      25    4     -16.02+AXIS   52.22

-6dB BACK WALL:  26   0    130    0     130    2     -36.25        92.64
1ST LIST BK WL:  75   4    130    0     141    3     -30.96+AXIS  113.06
- - - - - - - - - - - - - - - - - - - - - - - - - - - - - - - - - - - - -
-6dB COVERAGE ANGLE   :  20.00 deg. SPEAKER TILT FROM VERT:   7.36 deg.
1ST LST COMP COVG ANGL:  60.83 deg. AVG LIST TO CENT OF SPKR:   15 ft.   6 in.
=========================================================================
/CR/ TO START AGAIN: ~
```

THE PROGRAM

As before, the program operates basically the same as the other two. Lines 100 through 9999 are identical and may be copied directly from the other programs. Be sure to read carefully when you run the program. The entry fields are very similar, and you may type in the wrong information if you forget which program you are running. *Figure 1* is a sample screen to test your program with.

STAY TUNED

The last program will calculate direct and reflection paths from the speaker to the microphone. I will also discuss compensating for sloped floors.

Again, your comments are appreciated. Please contact me through **db Magazine**.

The Basic Program

```
10 REM SPEAKER ANGLE LOSS
CALCULATIONS FOR P.A. SYSTEM
20 REM "PALOSS.BAS",A
30 REM V1.1
40 REM 08-03-90
50 REM DCR
100 REM ***** INITIALIZE
110 REM **** SYSTEM FUNCTIONS
120 ON ERROR GOTO 8000
130 CLEAR
140 KEY OFF
800 REM **** SET SYSTEM
VARIABLES
810 P3$=STRING$(80," ")
820 P4$=STRING$(80,"-")
830 P5$=STRING$(80,"=")
840 P6$=CHR$(254)
900 REM **** SET PROGRAM
VARIABLES
910 GOSUB 10000
1000 REM ***** DISPLAY SCREEN
1010 REM **** INITIALIZE
1020 CLS
1030 COLOR C0,C1,C
1040 FOR Y=1 TO 25
1050 LOCATE Y,1:PRINT P3$;
1060 NEXT Y
1100 REM **** HEADING
1110 REM *** FRAME
1120 LOCATE 3,1:PRINT P5$;
1130 LOCATE 22,1:PRINT P5$;
1140 IF G0>0 THEN LOCATE
G1,G0:PRINT LEFT$(P4$,G2);
1150 IF G3>0 THEN LOCATE
G4,G3:PRINT LEFT$(P4$,G5);
1200 REM *** SYSTEM FUNCTIONS
1210 GOSUB 6000
1220 REM *** TITLE
1230 COLOR C0,C1,C
1240 LOCATE 2,1:PRINT G$;
1300 REM **** LINES
1310 REM *** INITIALIZE
1320 RESTORE
1330 J1=0
1340 ON ERROR GOTO 1900
1350 REM *** GET DATA
1360 READ
F$,F0$,F1$,F2$,F,F0,F1,F2,F3,F4,F3$
1370 J1=J1+1
1400 REM *** SET NUMBER
1410 P1$=""
1420 IF F0$="N" THEN GOTO 1500
1430 P1$=STR$(J1)
1440 FOR J3=1 TO LEN(P1$):IF
LEFT$(P1$,1)=" " THEN LET
P1$=RIGHT$(P1$,2):NEXT J3
1450 IF F0$="0" THEN GOTO 1490
1460 IF LEN(P1$)=1 THEN LET
P1$="0"+P1$
1470 IF F0$="2" THEN IF LEN(P1$)=2
THEN LET P1$="0"+P1$
1490 P1$=P1$+ ". "
1500 REM *** DISPLAY
1510 LOCATE F0,F:PRINT P1$+F$;
1600 REM *** REPEAT 1610 GOTO
1350
1900 REM **** END OF DISPLAY
1910 RESUME
1920 1920 ON ERROR GOTO 8000
2000 REM ***** INPUT DATA
2010 REM **** INITIALIZE
2020 RESTORE
2050 REM **** START LOOP
2060 FOR J=1 TO J1
2070 REM **** GET PARAMETERS
2080 READ
F$,F0$,F1$,F2$,F,F0,F1,F2,F3,F4,F3$
2100 REM **** PROMPTS
2110 COLOR C0,C1,C
2120 LOCATE 23,1:PRINT P3$;
2130 LOCATE 24,1:PRINT P3$;
2140 COLOR C2,C3,C
2150 LOCATE 23,1:PRINT F3$;
2200 REM **** GET INPUT
2210 GOSUB 7000
2300 REM **** VALIDATE
2310 IF LEN(D$)><1 THEN GOTO
2350
2320 IF J=1 THEN IF
INSTR("QqEeXxTt",D$)>0 THEN
GOTO 9000
2330 IF ASC(D$)=27 THEN GOTO
8100
2340 IF D$="!" THEN GOTO 6100
2345 IF D$="*" THEN GOTO 100
2350 FLAG$=""
2360 GOSUB 20000
2370 IF FLAG$="REENTER" THEN
GOTO 2100
2380 IF FLAG$="START OVER" THEN
GOTO 100
2390 IF FLAG$="ERROR" THEN
GOTO 8000
2400 REM **** REDISPLAY
2410 COLOR C10,C11,C
2420 LOCATE F2,F1:PRINT D$;
2430 COLOR C0,C1,C
2440 PRINT
LEFT$(P3$,F4-LEN(D$)+1);
2500 REM **** SLOT DATA
2510 GOSUB 30000
2600 REM **** END OF LOOP
2610 NEXT J
2700 REM ***** CALCULATIONS
2710 GOSUB 40000
2800 REM ***** DISPLAY RESULTS
2810 COLOR C12,C13,C
2820 GOSUB 50000
3000 REM ***** END OF SCREEN
3010 REM **** PROMPT
3020 F$="/CR/ TO START AGAIN:"
3030 LET F0$="0":F2$="&"
3040
F=1:F0=23:F1=22:F2=23:F3=0:F4=1

3050 COLOR C0,C1,C
3060 LOCATE 23,1:PRINT P3$;
3070 LOCATE 24,1:PRINT P3$;
3080 COLOR C2,C3,C
3090 LOCATE F0,F:PRINT F$;
3100 GOSUB 7000
3110 GOTO 1000

6000 REM ***** DATE & TIME
SUBROUTINE
6010 COLOR C0,C1,C
6020 LOCATE 1,70:PRINT DATE$;
6030 LOCATE 2,70:PRINT TIME$;
6040 LET PREVT$=TIME$
6050 RETURN
6100 REM ***** BACK-UP ONE
FIELD ROUTINE
6110 REM *** CLEAR CURRENT
FIELD
6120 COLOR C0,C1,C
6130 LOCATE F2,F1:PRINT
LEFT$(P3$,F4);
6140 IF J=1 THEN GOTO 100

6200 REM *** RESET FIELD
6210 RESTORE
6220 J2=J-1
6230 FOR J3=1 TO J2
6240 READ
F$,F0$,F1$,F2$,F,F0,F1,F2,F3,F4,F3$
6250 NEXT J3
6260 J=J3-1
6270 GOTO 2100
7000 REM ***** STANDARD
KEYBOARD INPUT SUBROUTINE
7010 REM **** MASK
7020 IF F2$="&" THEN LET F2$=P6$
7030 IF LEN(F2$)>1 THEN LET
P$=F2$:GOTO 7060
7040 IF F2$="" THEN LET
P$="":GOTO 7060
7050 LET P$=STRING$(F4,F2$)
7060 P$=P$+" "
7070 COLOR C4,C5,C
7080 LOCATE F2,F1:PRINT P$;
7090 IF BELL1$="Y" THEN PRINT
CHR$(7);
7095 REM —- SET BELL PARAMS &
GOSUB
7100 REM **** CLEAR INPUT
VARIABLE
7110 D$=""
```

```
7200 REM **** CHECK FOR FIELD FULL
7210 IF LEN(D$)><F4 THEN GOTO 7300
7220 COLOR C2,C3,C
7230 LOCATE 24,1:PRINT "THIS FIELD IS FULL. /CR/ OR BACKSPACE.";
7240 IF BELL2$="Y" THEN PRINT CHR$(7);
7245 REM --- SET BELL PARAMS & GOSUB
7300 REM **** INPUT
7310 LOCATE F2,F1
7320 GOSUB 7900
7330 D1$=INKEY$
7340 IF TIME$><PREVT$ THEN GOSUB 6000
7350 IF D1$="" THEN GOTO 7330
7360 GOSUB 7900
7400 REM **** /CR/ CHECK
7410 IF ASC(D1$)<>13 THEN GOTO 7600
7420 IF F3=0 THEN GOTO 7800
7430 IF LEN(D$)>=F3 THEN GOTO 7800
7440 GOTO 7200
7600 REM **** BACKSPACE
7610 IF ASC(D1$)<>8 THEN GOTO 7700
7620 COLOR C0,C1,C
7630 IF LEN(D$)=F4 THEN LOCATE 24,1:PRINT P3$;
7640 IF LEN(D$)=0 THEN GOTO 7200
7650 COLOR C4,C5,C
7655 REM — NEXT LINE, F2$ WON'T WORK WITH LONG MASK, NEED MASK VARIABLE
7660 LOCATE F2,F1+LEN(D$)-1:PRINT F2$;
7670 D$=LEFT$(D$,LEN(D$)-1)
7680 LOCATE F2,F1+LEN(D$)-1
7690 GOTO 7200
7700 REM **** ADD CHR TO STR & DISPLAY
7710 IF LEN(D$)=F4 THEN GOTO 7200
7720 COLOR C8,C9,C
7730 LOCATE F2,F1+LEN(D$):PRINT D1$;
7740 D$=D$+D1$
7750 REM **** LENGTH CHECK
7760 IF LEN(D$)<F4+1 THEN GOTO 7200
7800 REM **** RETURN
7810 COLOR C10,C11,C
7820 LOCATE F2,F1:PRINT D$;
7830 COLOR C0,C1,C
7840 PRINT LEFT$(P3$,F4-LEN(D$)+1);
7850 IF LEN(D$)=F4 THEN LOCATE 24,1:PRINT P3$;
7860 RETURN
7900 REM **** SET CURRENT CURSOR COLOR SUBROUTINE (TOGGLE - BLINK)
7910 P2$=CHR$(SCREEN(F2,F1+LEN(D$),0))
7920 P0=SCREEN(F2,F1+LEN(D$),1):REM - READ CURRENT COLOR
7930 P1=P0 MOD 16:REM - GET FOREGROUND VALUE
7940 IF P0>127 THEN LET P1=P1+16:REM - ADJUST IF BLINKING
7950 IF P1 = C6 THEN COLOR C4,C5,C
7960 IF P1 = C4 THEN COLOR C6,C7,C
7970 LOCATE F2,F1+LEN(D$):PRINT P2$;
7980 RETURN

8000 REM ***** ERRORS
8010 RESUME
8020 8020 COLOR C14,C15,C
8030 LOCATE 23,1:PRINT P3$;
8040 LOCATE 24,1:PRINT P3$;
8050 COLOR C14,C15,C 8040 LOCATE 23,1:PRINT "ERROR AT LINE";ERL;
8050 LOCATE 24,1:PRINT E$;
8060 INPUT "",X$
8070 GOTO 1000
8100 REM ***** STOP
8110 ON ERROR GOTO 0
8120 COLOR 15,0,0
8130 STOP

9000 REM ***** EXIT
9010 CLS
9020 RUN "MENU"

10000 REM ***** PROGRAM VARIABLES
10010 REM **** PROGRAM TITLE
10020 LET G$="P.A. SPEAKER ANGLE LOSS CALCULATIONS"
10030 REM **** ERROR MESSAGE
10040 LET E$="CONFIGURATION IS NOT POSSIBLE. ANY KEY TO RESTART: "
10050 REM **** BELL AT AFTER MASK DISPLAY
10060 LET BELL1$="N"
10070 REM **** BELL AT FIELD FULL PROMPT
10080 LET BELL2$="N"
10100 REM **** DIVIDING LINES X,Y,LEN
10110 LET G0=1:LET G1=8:LET G2=80
10120 LET G3=0:LET G4=0:LET G5=80
10200 REM **** COLORS
10210 REM *** BORDER
10220 C = 0
10230 REM - ALL OTHERS TEXT & BACKGROUND
10240 REM *** INITIALIZE AND BACKGROUND TEXT
10250 C0 = 7 : C1 = 0
10260 REM *** PROMPTS
10270 C2 = 15 : C3 = 0
10280 REM *** MASK
10290 C4 = 15 : C5 = 0
10300 REM *** CURRENT CURSOR
10310 C6 = 31 : C7 = 0
10320 REM *** CURRENT INPUT
10330 C8 = 15 : C9 = 0
10340 REM *** FOREGROUND TEXT (ACCEPTED INPUT)
10350 C10 = 15 : C11 = 0
10360 REM *** FOREGROUND TEXT (OUTPUT DISPLAY)
10370 C12 = 15 : C13 = 0
10380 REM *** ERROR TEXT
10390 C14 = 15 : C15 = 0
10999 RETURN

11000 REM ***** DATA
11005 REM FIELD DESC,AUTO NO.,DEFAULT,MASK CHR,X,Y,IN X,Y,MIN,MAX,PROMPT
11006 REM F$,F0$,F1$,F2$,F,F0,F1,F2,F3,F4,F3$
11007 REM AUTO N0.: N= OMIT NUMBER, 0= OMIT LEADING ZERO, 1= 2 DIGIT NO., 2= 3 DIGIT NO.
11008 REM MASK CHARACTER (F2$) = TO "&" WILL DISPLAY A BOX - CHR$(254)
11010 DATA "H SPKR TO LAST LIST: ft"
11011 DATA "N","0","&",1,4,22,4,0,4
11012 DATA "HORIZ. DISTANCE FROM SPEAKER TO LAST LISTENER (FEET + INCHES OR INCHES ONLY)"
11020 DATA "in"
11021 DATA "N","0","&",37,4,30,4,0,6
11022 DATA "HORIZ. DISTANCE FROM SPEAKER TO LAST LISTENER (INCHES ADDED TO FEET)"
11030 DATA "H LAST LIST TO WALL: ft"
11031 DATA "N","0","&",1,5,22,5,0,4
11032 DATA "HORIZ. DISTANCE FROM LAST LISTENER TO BACK WALL"
11040 DATA "in"
11041 DATA "N","0","&",37,5,30,5,0,6
```

```
11042 DATA "HORIZ. DISTANCE
FROM LAST LISTENER TO BACK
WALL "
11050 DATA "H 1ST TO LAST
LISTENER:    ft"
11051 DATA "N","0","&",1,6,22,6,0,4
11052 DATA "HORIZ. DISTANCE
FROM THE FIRST TO LAST LISTENER
(FEET + INCHES OR INCHES ONLY)"
11060 DATA "in"
11061 DATA "N","0","&",37,6,30,6,0,6
11062 DATA "HORIZ. DISTANCE
FROM THE FIRST TO LAST LISTENER
(INCHES ADDED TO FEET)"
11070 DATA "V FLOOR TO AV LIST :
ft"
11071 DATA "N","0","&",42,4,63,4,0,4
11072 DATA "VERT. DISTANCE FROM
FLOOR TO AVG. LISTENING HEIGHT"
11080 DATA "in"
11081 DATA "N","0","&",76,4,71,4,0,4
11082 DATA "VERT. DISTANCE FROM
FLOOR TO AVG. LISTENING HEIGHT"
11090 DATA "V FLR TO CENT SPKR :
ft"
11091 DATA "N","0","&",42,5,63,5,0,4
11092 DATA "VERT. DISTANCE FROM
FLOOR TO CENTER OF SPEAKER"
11100 DATA "in"
11101 DATA "N","0","&",76,5,71,5,0,4
11102 DATA "VERT. DISTANCE FROM
FLOOR TO CENTER OF SPEAKER"
11110 DATA "ON AXIS SPKR TO REF:
ft"
11111 DATA "N","0","&",42,6,63,6,0,4
11112 DATA "DISTANCE FROM
SPEAKER TO ON AXIS db SPL REF.
MEASUREMENT"
11120 DATA "in"
11121 DATA "N","0","&",76,6,71,6,0,4
11122 DATA "DISTANCE FROM
SPEAKER TO ON AXIS db SPL REF.
MEASUREMENT"
11130 DATA "SPKR -6dB COVR
ANGL:    degrees"
11131 DATA "N","0","&",42,7,63,7,0,6
11132 DATA "ENTER THE
SPEAKER'S RATED -6 dB COVERAGE
ANGLE (LESS THAN 31.05 DEG.)"

20000 REM ***** VALIDATIONS -
USER SUBROUTINE
20010 IF D$="" THEN LET D$=F1$
20020 FOR J3=1 TO LEN(D$)
20030 IF
INSTR("0123456789.-+",MID$(D$,J3,1))
=0 THEN LET FLAG$="REENTER"
20040 NEXT J3
29999 RETURN
30000 REM ***** SLOT DATA - USER
SUBROUTINE
30010 IF J=1 THEN LET A1$=D$
30020 IF J=2 THEN LET A2$=D$
30030 IF J=3 THEN LET A3$=D$
30040 IF J=4 THEN LET A4$=D$
30050 IF J=5 THEN LET A5$=D$
30060 IF J=6 THEN LET A6$=D$
30070 IF J=7 THEN LET A7$=D$
30080 IF J=8 THEN LET A8$=D$
30090 IF J=9 THEN LET A9$=D$
30100 IF J=10 THEN LET A10$=D$
30110 IF J=11 THEN LET A11$=D$
30120 IF J=12 THEN LET A12$=D$
30130 IF J=13 THEN LET A13$=D$
39999 RETURN

40000 REM ***** CALCULATIONS -
USER SUBROUTINE
40010 REM **** CONVERT TO
NUMBERS AND INCHES
40020 REM *** SPEAKER TO LAST
LISTENER
40030
D1#=(12*VAL(A1$))+VAL(A2$)
40050 REM *** LAST LISTENER TO
WALL
40060
D4#=(12*VAL(A3$))+VAL(A4$)
40070 REM *** FIRST TO LAST
LISTENER
40080
D6#=(12*VAL(A5$))+VAL(A6$)
40090 REM *** FLOOR TO AVERAGE
LISTENING HEIGHT
40100
H6#=(12*VAL(A7$))+VAL(A8$)
40110 REM *** FLOOR TO CENTER
OF SPEAKER
40120
H7#=(12*VAL(A9$))+VAL(A10$)
40130 REM *** DISTANCE FROM
SPEAKER FOR db SPL REFERENCE
MEASUREMENT
40140
R#=(12*VAL(A11$))+VAL(A12$)
40150 REM *** SPEAKER'S RATED
-6dB COVERAGE ANGLE
40160 A4#=VAL(A13$)
40200 REM *** CONVERSION
FACTORS
40210 REM ** RADIANS TO
DEGREES
40220 RD#=180/3.1415927
40230 REM ** DEGREES TO
RADIANS
40240 DR#=3.1415927/180
40300 REM **** CALCULATE
SIMPLE HORIZONTAL DISTANCES
40310 D3#=D1#-D6#
40320 D5#=D1#+D4#
40330 REM **** CALCULATE
SIMPLE VERTICAL DISTANCES
40340 H1#=H7#-H6#
40350 H2#=H1#
40360 H3#=H1#
40400 REM **** CALCULATE
ANGLES
40410 REM *** 0 db ON AXIS - LAST
LISTENER
40420 A1#=ATN(D1#/H1#)*RD#
40430 REM *** -6 db AXIS
40440 A2#=A1#-(A4#/2)
40450 REM *** FIRST LISTENER
40460 A3#=ATN(D3#/H3#)*RD#
40470 REM *** FIRST LISTENER
COMPLEMENT COVERAGE
40480 A5#=(A1#-A3#)*2
40490 REM *** SPEAKER TILT FROM
VERTICAL
40495 A6#=90-A1#
40500 REM **** CALCULATE
SPEAKER AXIS THROW DISTANCES
40510 REM *** 0 db ON AXIS THROW
DISTANCE
40520 T1#=H1#/COS(A1#*DR#)
40530 REM *** -6 db AXIS THROW
DISTANCE
40540 T2#=H1#/COS(A2#*DR#)
40550 REM *** FIRST LISTENER
AXIS THROW DISTANCE
40560 T3#=H1#/COS(A3#*DR#)
40570 REM **** -6 db HORIZONTAL
DISTANCE
40580 D2#=SIN(A2#*DR#)*T2#
40800 REM **** CALCULATE BACK
WALL REFLECTION
40810 REM *** -6 db
40820 REM ** ANGLE FROM
HORIZONTAL IN DEGREES
40830 A9#=A4#+A2#-90
40840 REM ** THROW DISTANCE
40850
T4#=D5#/COS(ABS(A9#*DR#))
40860 REM ** BACK WALL HEIGHT
40870
H4#=H7#+((ABS(A9#)/A9#)*((TAN(A
BS(A9#*DR#)))*D5#))
40880 REM ** ANGLE FROM
VERTICAL
40890 A7#=A1#+A1#-A2#
40900 REM *** FIRST LISTENER
COMPLEMENT
40910 REM ** ANGLE FROM
HORIZONTAL IN DEGREES
40920 A9#=A5#+A3#-90
40930 REM ** THROW DISTANCE
40940
T5#=D5#/COS(ABS(A9#*DR#))
40950 REM ** BACK WALL HEIGHT
40960
H5#=H7#+((ABS(A9#)/A9#)*((TAN(A
BS(A9#*DR#)))*D5#))
40970 REM ** ANGLE FROM
VERTICAL
40980 A8#=A1#+A1#-A3#
```

```
41000 REM **** CALCULATE
DISTANCE SPL LOSS FOR AXIS
41010 REM *** 0 db ON AXIS TO
AUDIENCE
41020
L1#=20*(LOG(R#/T1#)/LOG(10))
41030 REM *** -6 db AXIS TO
AUDIENCE
41040
L2#=20*(LOG(R#/T2#)/LOG(10))-6
41050 REM *** FIRST LISTENER
AXIS TO AUDIENCE
41060
L3#=20*(LOG(R#/T3#)/LOG(10))
41100 REM *** -6 db AXIS TO BACK
WALL
41110
L4#=20*(LOG(R#/T4#)/LOG(10))-6
41120 REM *** FIRST LISTENER _#0
0!_f $_f f9 f_##fé1130
L5#=20*(LOG(R#/T5#)/LOG(10))
49999 RETURN
50000 REM ***** DISPLAY RESULTS
- USER SUBROUTINE
50010 REM **** TEXT FORMAT
50020 REM *** DISPLAY FRAME
50030 LOCATE 9,1:PRINT "
VERTICAL     HORIZONTAL     THROW
THROW     VERTICAL"
50040 LOCATE 10,1:PRINT "
HEIGHT     DISTANCE     DISTANCE
dB LOSS   TO THROW"
50050 LOCATE 11,1:PRINT
"SPEAKER TO:   Feet  In.   Feet  In.
Feet  In.   decibels   degrees"
50060 LOCATE 12,1:PRINT "--------
------------------------------"
50070 LOCATE 19,1:PRINT "--------
------------------------------"
50100 REM *** DISPLAY DATA
50110 REM ** LONG LINES
50120 REM * SET MASK
50130 P1$=":#####  ##
#####  ##   #####  ##
####.##    ###.##"
50140 X=1
50200 REM * DATA
50210 Y=13:P$="LAST LISTENER
":P0#=H1#:P1#=D1#:P2#=T1#:P3#
=L1#:P4#=A1#:GOSUB 51000
50220 Y=14:P$="-6dB LISTENER
":P0#=H2#:P1#=D2#:P2#=T2#:P3#
=L2#:P4#=A2#:GOSUB 51000
50230 Y=15:P$="FIRST
LISTENER":P0#=H3#:P1#=D3#:P2#=
T3#:P3#=L3#:P4#=A3#:GOSUB
51000
50240 Y=17:P$="-6dB BACK
WALL":P0#=H4#:P1#=D5#:P2#=T4#
:P3#=L4#:P4#=A7#:GOSUB 51000
50250 Y=18:P$="1ST LIST BK
WL":P0#=H5#:P1#=D5#:P2#=T5#:P
3#=L5#:P4#=A8#:GOSUB 51000
50260 LOCATE 15,68:PRINT "+AXIS"
50270 LOCATE 18,68:PRINT "+AXIS"
50300 REM ** SINGLE DATA LINES
50310 REM * BOTTOM OF SCREEN
50310 X=1:Y=20:P$="-6dB
COVERAGE ANGLE   : ####.##
deg.":P0#=A4#:GOSUB 51200
50320 X=1:Y=21:P$="1ST LST
COMP COVG ANGL: ####.##
deg.":P0#=A5#:GOSUB 51200
50330 X=38:Y=20:P$="SPEAKER
TILT FROM VERT: ####.##
deg.":P0#=A6#:GOSUB 51200
50340 D#=H1#:GOSUB 51500
50350 X=38:Y=21:P$="AVG LIST TO
CENT OF SPKR: #####
ft.":P0#=DF#:GOSUB 51200
50360 X=75:Y=21:P$="##
in.":P0#=DI#:GOSUB 51200
50400 REM * REFRESH TOP OF
SCREEN (INPUTS)
50410 REM HORIZ DIST FROM
SPEAKER TO LAST LISTENER
50420 D#=D1#:GOSUB 51500
50430
=22:Y=4:P$="####":P0#=DF#:
GOSUB 51200
50440 X=30:Y=4:P$="
##":P0#=DI#:GOSUB 51200
50450 REM HORIZ DIST FROM LAST
LIST TO BACK WALL
50460 D#=D4#:GOSUB 51500
50470
X=22:Y=5:P$="####":P0#=DF#:GO
SUB 51200
50480 X=30:Y=5:P$="
##":P0#=DI#:GOSUB 51200
50490 REM HORIZ DIST FROM FIRST
TO LAST LISTENER
50500 D#=D6#:GOSUB 51500
50510
X=22:Y=6:P$="####":P0#=DF#:GO
SUB 51200
50520 X=30:Y=6:P$="
##":P0#=DI#:GOSUB 51200
50530 REM VERT DIST FROM FLOOR
TO AVG LISTENING HEIGHT
50540 D#=H6#:GOSUB 51500
50550
X=63:Y=4:P$="####":P0#=DF#:GO
SUB 51200
50560 X=71:Y=4:P$="
##":P0#=DI#:GOSUB 51200
50570 REM VERT DIST FROM FLOOR
TO CENTER OF SPEAKER
50580 D#=H7#:GOSUB 51500
50590
X=63:Y=5:P$="####":P0#=DF#:GO
SUB 51200
50600 X=71:Y=5:P$="
##":P0#=DI#:GOSUB 51200
50610 REM REFERENCE DISTANCE
50620 D#=R#:GOSUB 51500
50630
X=63:Y=6:P$="####":P0#=DF#:GO
SUB 51200
50640 X=71:Y=6:P$="
##":P0#=DI#:GOSUB 51200
50650 REM SPEAKER'S RATED -6dB
COVERAGE ANGLE
50660
X=63:Y=7:P$="###.##":P0#=A4#:G
OSUB 51200
50700 REM *** WARNING
50710 IF A4#<A5# THEN RETURN
50720 LOCATE 19,1:PRINT
"WARNING !!! — ENTIRE AUDIENCE
INSIDE -6dB COVERAGE ANGLE !!! "
50999 RETURN
51000 REM **** PRINT LONG LINE
SUBROUTINE
51010 REM *** SET VARIABLES
51020 D#=P0#:GOSUB
51500:PF0#=DF#:PI0#=DI#
51030 D#=P1#:GOSUB
51500:PF1#=DF#:PI1#=DI#
51040 D#=P2#:GOSUB
51500:PF2#=DF#:PI2#=DI#
51110 REM *** PRINT 51110 LOCATE
Y,X:PRINT USING
P$+P1$;PF0#,PI0#,PF1#,PI1#,PF2#,P
I2#,P3#,P4#
51120 RETURN
51200 REM **** PRINT A SINGLE
LINE
51210 LOCATE Y,X:PRINT USING
P$;P0#
51220 RETURN
51500 REM **** CONVERT TO FEET
AND INCHES
51510 DF#=INT(D#/12)
51520 DI#=INT(D#-(DF#*12)+0.5)
51530 IF DI#<12 THEN RETURN
51540 DF#=DF#+1
51550 DI#=0
51560 RETURN
65535 END
```

db

LAB REPORT

LEN FELDMAN

Crown Macro Reference Power Amplifier

GENERAL INFORMATION

• One look at the power line cord of this amplifier tells you immediately that you are not about to run out of audio power with this one. The cord is heavy enough to easily handle the 26 or more amperes that this unit draws from an AC outlet when delivering full power. But high levels of audio power—760 watts per channel in the stereo mode, driving 8-ohm impedances—are really just a small part of the story here. As Crown puts it, the Macro Reference amplifier offers more than 120 dB of dynamic range—enough to reproduce the full dynamic range of a 20-bit digital program source.

To maintain tightest possible bass, this amplifier claims a damping factor of over 20,000! We say "claims" simply because there was no way we could confirm such a high damping factor in our otherwise well-equipped audio measurement lab. As for its power supply, the Macro Reference amp uses a custom-designed, tape-wound, low-noise torroidal supply with an extremely high power density. Crown's IOC circuitry (the acronym stands for Input/Output Comparator) compares the waveform at the input signal to that of the output signal, and if the difference equals or exceeds 0.05 percent, an indicator light illuminates.

With yet another acronym, "ODEP", Crown has developed a unique overall protection circuit for this amplifier. The initials stand for Output Device Emulator Protection circuitry. With it, the real-time operating environment of the power transistors is simulated and compared to their known safe operating area. If ODEP "predicts" that they are about to exceed their limits, their drive level is proportionately reduced. The last acronym used to describe a special feature of this amplifier is PIP, which stands for Programmable Input Processor—an expansion system that lets you add various modules to accomplish such modifications as electronic crossover, daisy chaining of several amplifier balanced inputs together, isolate source from inputs (using balanced 1:1 transformers) and much more. PIP modules install easily through the rear panel of the amplifier and connect by means of a card edge connector located inside the amplifier.

The cooling system of the Macro Reference amplifier consists of an infinitely variable speed fan. It draws air across the power transformer and main circuit board and pushes it across the power transistors and out diffuser exhaust vents. We found the fan, which turns on only when needed, to be one of the quietest we have encountered in this type of cooling arrangement.

CONTROL LAYOUT

The all black front panel of the Crown Macro Reference amplifier is equipped with individual level controls for each channel. An "enable" pushbutton switch applies power to the amplifier. Between the two input level controls are ODEP, IOC and signal indicators for each channel. Two banks of five LEDs each serve as either dynamic range meters (that's how they are configured at the factory) or as level meters. To alter the function of these LED meters to that of level meters, it is necessary to remove the front panel. However, once this is done, a two position slide switch is simply switched to its alternate setting.

The rear panel of the amplifier has a reset switch at the upper left, near the heavy power cord referred to earlier. This reset switch is essentially a 30 ampere circuit breaker. Just below the power cord and to its right is a three position switch. As supplied, it is set to its center position, which provides stereo operation. A *Parallel Mono* setting can be used to drive single loads with an impedance of less than 4 ohms while a bridged mono mode can be set for driving a single load having an impedance of 4 ohms or more. At the center of the

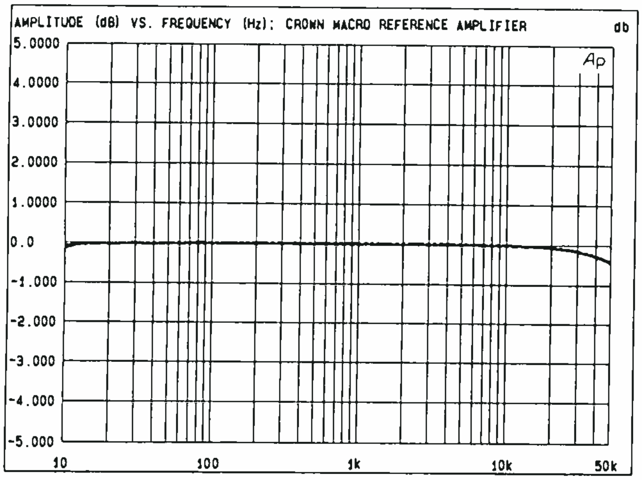


Figure 1. Frequency response at 1 watt per channel nominal output level, 8-ohm loads.

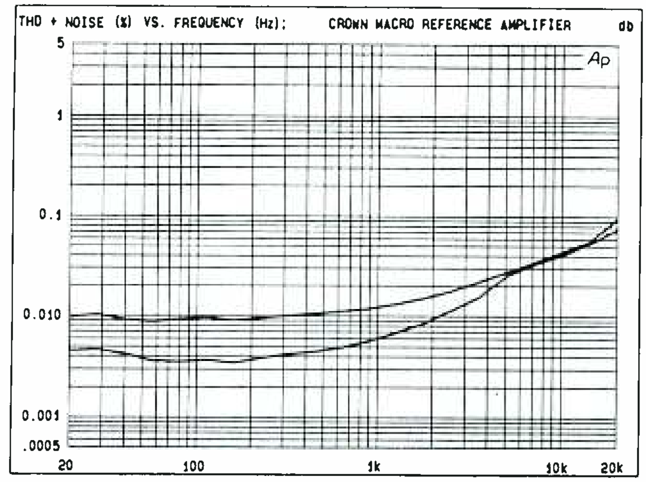


Figure 2. Harmonic distortion plus noise versus frequency, at rated output (760 watts/channel at 8-ohm load)s.

panel is a removable PIP-FX module, included as a standard feature to provide balanced XLR inputs. The XLR inputs are parallel to the phone jack inputs located just below the PIP module. The two sets of inputs, as supplied, can be used to facilitate "daisy chaining" multiple amplifiers. Between the two phone jacks is a *Ground Lift* switch that allows you to isolate the input signal from the AC ground. Often, this sort of isolation can help to eliminate the hum created by unwanted ground loops. Output gold plated banana jacks are located at the right end of the rear panel and will accept banana plugs, bare wire, or spade lugs.

LABORATORY MEASUREMENTS

The owner's manual supplied with the Crown Macro Reference amplifier is extremely detailed, and provides complete instructions for installation of the unit as well as for operation of the amplifier in its various stereo and mono modes. One of the first important notices we came across was the one concerning power requirements.

Calculating the equivalent distortion percentage from these components, we came up with a figure of THD of only 0.00335 percent!

While our lab is equipped with adequate power sources, no single circuit in the lab can handle 30 amperes of current. Furthermore, our variable voltage transformer certainly could not handle that amount of current. That transformer, commonly referred to as a Variac, is normally used to maintain a line voltage of exactly 120 volts while testing amplifiers. Under the circumstances, we had to bypass the Variac and connect directly to a wall outlet. Thus, when heavy current was drawn, voltage dropped to a bit below 120 volts. In other words, had we been able to maintain exactly 120 volts of AC voltage, distortion levels at rated output might have been even lower than they turned out to be.

We relate these details not just to indicate the thoroughness with which products are tested in our lab, but also as a cautionary note for those planning to buy and use this remarkably powerful amplifier. In the table of specifications near the end of the owner's manual, under the heading "Required AC Mains," there appears the following: "60 Hz, 120 VAC (10 percent). Draws 70 watts or less at idle. Draws as high as 26 amps with a continuous 1 kHz sinewave output of 760 watts into 8 ohms in *stereo* mode. (The next sentences also appear in italics in the manual, for extra emphasis.) *It is extremely important to have adequate AC power available to the amplifier. Power amplifiers cannot create energy—they must have the required voltage and current to deliver the undistorted rated wattages you expect."*

That having been said—and noted—we first measured the frequency response of the amplifier. Results are shown in *Figure 1*. Response was virtually flat down to 10 Hz and was down only 0.4 dB at 50 kHz. Next, we measured total harmonic distortion plus noise versus frequency at the rated stereo power output of 760 watts per channel for both channels. Bear in mind that the power line voltage was slightly lower than the standard 120 volts during the course of this measurement. Nonetheless, at 1 kHz, THD plus noise measured only 0.006 percent for the left channel and 0.012 percent for the right channel. A slight rise in THD was noted at higher frequencies while the amplifier was delivering its full rated power in stereo, but even at 20 kHz, THD plus noise was still less than 0.1 percent. Results of these tests are shown in *Figure 2*.

Since the measurements of *Figure 2* represent the sum of THD plus noise, we wanted to isolate the harmonic distortion from the noise. To do this, we used the spectrum analysis function of our test equipment and, with the FFT function, we acquired a sample at 1 kHz some sixteen times, in order to average out the random noise, so that actual harmonics could be seen and measured. Results are shown in *Figure 3*, and the only prominent harmonics visible are at 2 kHz and 3 kHz. They are at –92.5 and –93 dB respectively, with respect to the reference level of 760 watts, represented here as

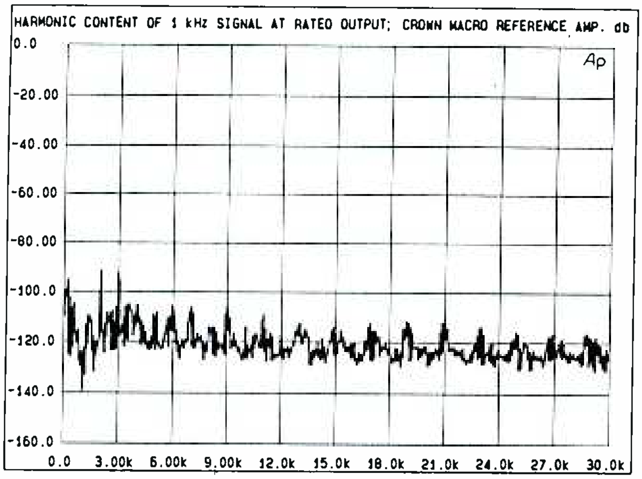


Figure 3. Spectrum analysis of the harmonics of a 1 kHz signal at rated output (760 watts/channel, 8 ohm loads) averaged result of 16 acquisitions, to reduce displayed noise and improve display of coherent signals.

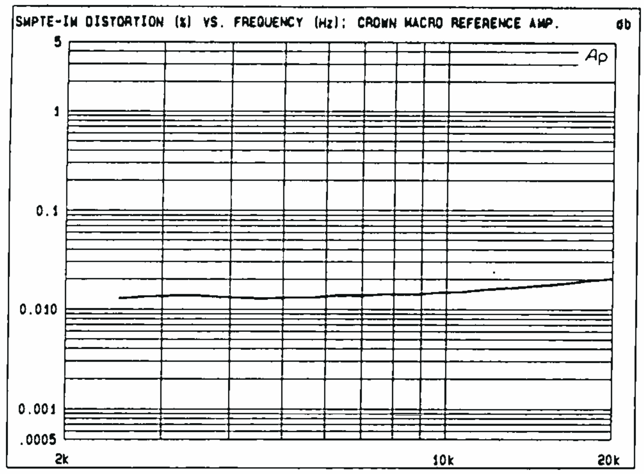


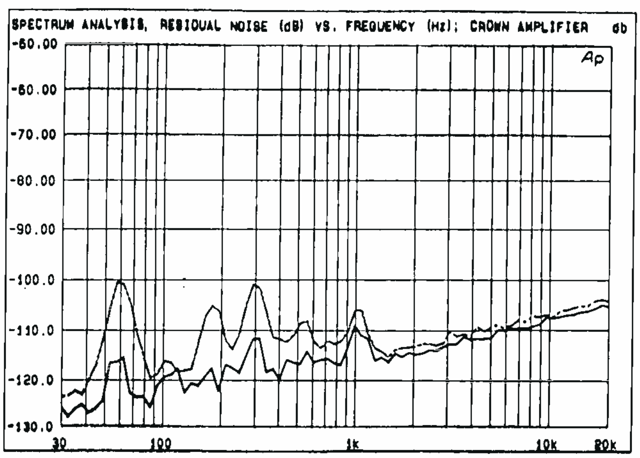
Figure 4. SMPTE-IM distortion versus frequency at rated power output (760 watts/channel equivalent, 8-ohm loads).

0 dB. Calculating the equivalent distortion percentage from these components, we came up with a figure of THD of only 0.00335 percent!

Figure 4 represents a plot of SMPTE-IM distortion versus frequency, in which the standard 60 Hz component is progressively modulated (in a 4:1 ratio) by frequencies ranging from over 2.0 kHz to 20 kHz. Readings ranged from 0.015 percent to 0.02 percent. While these results are somewhat higher than the 0.005 percent claimed by Crown, they may once again be due to the fact that we were unable to maintain a constant 120 volt power line voltage during this test.

A-weighted signal-to-noise ratio with input level controls adjusted to produce 1 watt per channel output, while applying an input of 500 mV measured 98.7 dB for the left channel and 96.3 dB for the right channel. If we were to relate those S/N readings to rated output (760 watts per channel), the overall S/N ratio would be nearly 29 dB greater, or 127.5 dB for the left channel and 125.1 dB for the right channel. Crown's claim of a S/N ratio of "greater than 120 dB" is easily confirmed. Again, referred to 1 watt and an input of 500 mV, we did a ⅓-octave spectrum analysis of the residual noise generated by this amplifier. Results shown in *Figure 5* show that, indeed, right channel noise was a bit higher than that of the left channel. The difference is attributed largely to power supply frequency hum components and their harmonics. Even at that, without using any weighting, the worst of these components is still more than 100 dB below the 1 watt reference level established for this measurement. Additional measurement results will be found in our usual table of *Vital Statistics* at the end of this report.

Figure 5. Spectrum analysis of residual noise at 1 watt output reference level.



CONCLUSIONS

Clearly, this amplifier is intended primarily for sound reinforcement applications. Loudspeaker systems to be used with this amplifier should be chosen with care, since few, if any, studio monitoring speakers can handle the power that this amplifier can deliver. What listening tests we were able to perform in our lab and listening studio were confined to levels that our domestic, high-fidelity loudspeakers could handle without being damaged. Under such conditions, the amplifier ran extremely cool; the fan never came on. (It had come on often when we were conducting the static signal bench tests.) Sound was clean and bass was as tight as we have ever known. If we had used speaker cable heavier than the #12 gauge available to us when this amplifier was tested, we might have achieved even greater damping and, perhaps, even tighter bass than we did.

Crown offers detailed data in their owner's manual that will enable you to calculate the proper gauge speaker wire to use for a given distance and a given damping factor that the manufacturer of your loudspeakers may deem as optimal. Crown's engineers have obviously expended a great deal of energy and thought to make this amplifier as reliable and rugged as current technology permits. It is not an inexpensive product, but for those who require high, clean power and cannot afford to have an amplifier failure in the field, the extra cost of such an amplifier can be well worth paying. db

VITAL STATISTICS

SPECIFICATION	MFR'S CLAIM	dB MEASURED
Frequency Response	20 to 20 kHz, ±0.1 dB	Confirmed
S/N Ratio	>120 dB	127.5/125.1 dB
Bandwidth	3 Hz to 100 kHz	Confirmed
IM Distortion	0.005%	0.015% (See Text)
Damping Factor	1800 @ 1 kHz	N/A
Power Bandwidth (–3 dB)	4 Hz to 30 kHz	4 Hz to 38 kHz
Rated Power Output/Channel		
8 ohm loads	760 watts	>800 watts
4 ohm loads	1160 watts	N/A
2 ohm loads	1500 watts	N/A
Load Impedance Range	2 to 16 ohms	Confirmed
Dimensions (WxHxD, in.)	19 X 7 X 18.75	Confirmed
Weight (lbs.)	56.5	Confirmed
Price:	$3,500.00	

db BUYER'S GUIDE

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

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

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

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

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

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

On to the charts...

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
ALESIS CORPORATION															
RA100	2	75	100		20-20k			.03 at 1k	.05 at 1k	20-20k 1	0.5V RMS	5.5 19 8	14	$349.	Clip lights on each channel. Passive cooling requires no fan. Short circuit protected.
ALTEC LANSING CORPORATION															
9444A	2	200	300		10-70k	0.05	0.05	0.05	0.01	10-70k 1	.812 .932	5.25 19 12.75	39	$990.	Balanced XLR/barrier strip inputs, level controls on back panel, 300W bridged into 8 ohms.
9442A	2	100	150		10-50k	0.1	0.1	0.1	0.1	10-50k	.775 .890	5.25 19 11	32	$770.	Balanced XLR/barrier strip inputs, level controls on back panel, powered accessory sockets.
9446A	2	400	600		20-20k	0.05	0.05	0.01	0.1	20-20k	.790 .910	5.25 19 15.75	52	$1,890.	Balanced XLR/barrier strip inputs, level controls on back panel, powered accessory sockets.
1407A	1	75	75		20-20k	0.01	0.01	0.01	0.1	20-20k	.775	5.25 19 12.5	24	$599.	Monaural amplifier with direct and transformer output. Direct=8 ohm.
1415A	1	150	150		20-20k	0.01	0.01	0.01	0.1	20-20k	.775	5.25 19	31	$735.	Monaural amplifier with direct and transformer output.
ASHLY AUDIO INC.															
FET-2000C	2	300	500	675	20-20k	.004	.01	.004	.01	20 20k	1.7	5.25 19 16	60	$999.99	Barrier strip inputs,MOSFET,UL listed. FT2000M same with peak reading meters—$1099.99.
FET1500C	2	200	300	360	20-20k	.004	.01	.004	.01	20 20k	1.7	3.5 19 16	42	$799.99	Barrier strip inputs,MOSFET,UL listed. FT1500M same with peak reading meters,XLRs—$889.99.
FET1000C	2	120	190	225	20-20k	.004	.01	.004	.01	20-20k	1.7	3.5 19 16	37	$699.99	Barrier strip inputs,MOSFET,UL listed. FT1000M same with peak reading meters,XLRs—$789.99.
BGW SYSTEMS															
GTA	2	360	625	1000	20-20k	less than 0.05	0.03	less than 0.1	less than 0.1	3-85k +0,-3	1.73	5.25 17.5 16.3	78	$2,199.	Twin power supplies, balanced inputs w/looping XLRs and 1/4 in. TRS.
GTB	2	300	450	425	20-20k	less than 0.05	0.03	less than 0.1	less than 0.1	3-85k +0,-3	1.54	5.25 19 14.3	50	$1,539.	Balanced inputs w/looping XLRs and 1/4 in. TRS, fan cooled, large LED status indicators.
750F/G	2	300	450	425	20-20k	less than 0.05	0.01	less than 0.1	less than 0.1	3-85k +0,-3	1.5	7 19 15	55	$1699(F) $1849(G)	LED status indicators, thermostat fan control, 850 w/2 ohm single channel, (G) stereo 50 dB range metering.
350/350A	2	200	325		20-20k	less than 0.05	0.03	less than 0.1	less than 0.1	3-85k +0,-3	1.18	5.25 19 11.8	34	$1199 $1349(A)	LED status indicators, convection cooled, balanced inputs w/looping XLRs and 1/4 in. TRS.
200	2	100			20-20k	less than 0.05	less than 0.03	less than 0.1	less than 0.1	1-85k +0,-3	0.92	1.75 19 11.9	14	$999.	100W/Channel in one rack space, balanced XLR and 1/4 in. TRS inputs, LED status indicators.
6500T	2	100	150		20-20k	less than 0.1	less than 0.05	less than 0.05	less than 0.05	3-100k +0,-3	0.9	3.5 19 12.9	28	$599.	Barrier strip inputs/outputs w/ 1/4 in. TRS, optional XLR balanced inputs, rearmounted level controls.
7500T	2	200	300		20-20k	less than 0.1	less than 0.05	less than 0.1	less than 0.1	3-85k +0,-3	1.18	5.25 19 12.4	36	$849.	Barrier strip inputs/outputs w/ 1/4 in. TRS, optional XLR balanced input, rearmounted level controls.
BRYSTON/BRYSTON VERMONT															
2BLP	2	50	100		1-100k	less than 0.01	less than 0.01	less than 0.01	0.01	1-100k	0.75	1.75 19 10	18	$775.	Full twenty year warranty.
3B	2	100	200		1-100k	less than 0.01	less than 0.01	less than 0.01	0.01 20-20k	1-100k	1.0	2.25 19 9	30	$1,375.	Modular construction.

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
4B	2	250	400		1-100k	less than .01	0.01	less than .01	0.01 20- 20k	1-100k	1.25 volt 50k	2.25 19 13.5	45	$1,995.	All triple good plater contacts.
7B	1	500	800		1-100k	less than 0.01	0.01	less than .01	0.01 20- 20k	1-100k	1.0V 50k	2.25 19 13.5	45	$2,195.	All discrete.
CARVER CORPORATION															
P1200	2	450	600		5-80k		0.1		0.5	20-20k +0,-.5	1.5	3.5 19 12.75	21	$1,250.	Bridged mono operation—70V direct drive operation, remote/sequential power on—LED power meters—XLR, TRS, inputs—clipping eliminator circuit.
PT1250	2	465	625		5-80k		0.1		0.5	20-20k +0,-.5	1.5	3.5 19 10.75	11	$1,500.	Has 70V direct drive operation, LED power meters, XLR, TRS inputs clipping eliminator circuit.
PT1800	2	600	900	1100			0.1		0.5	±5	1.5	5.25 19 12.75	46		Bridged mono operation, 70V direct drive, operation-dual detachable power cords, fully modular.
M120	2	40	60		2-120k		0.1		0.5	20-20k +0,-.5	1.5	1.75 19 12	10	$560.	Bridged mono operation, headphone jack—XLR, TRS, barrier strip in-inputs—clip indicators.
M300	2	110	150		4-70k		0.1		0.5	20-20k +0,-.5	1.5	1.75 19 12	11	$680.	Bridged mono operation, headphone jack—XLR, TRS, barrier strip in-puts—LED power Indicators.
M600	2	200	300		5-80k		0.1		0.5	20-20k +0,-.5	1.5	3.5 19 11.56	23	$820.	Remote/sequential power on/off circuit—convection cooling—bridged mono operation XLR.
M900	2	350	450		5-80k		0.1		0.5	20-20k +0,-.5	1.5	3.5 19 11.56	24	$995.	Same, with fan cooling.
CARVIN —See our ad on page 8															
FET 1000(W)	1	300	500		20-20k			.05	.05	20-20k 0.5	1.0	5.25 19 10	37	$679.	Mosfet circuitry, speaker guard protection, short circuit current limiting thermal shut off switch.
FET 450	2	125	450		20-20k			.05	.05	20-20k 0.5	1.0	5.25 19 10	32	$499.	Same
FET 401(W)		150	300	400	20-20k			less than 0.1	less than 0.1	20-20k 1	1.0	5.25 19 10	31	$399.	Has 9 band graphic equalizer, thermostatic protection.
CREST AUDIO, INC.															
8001	2	720	1100	1400				less than 0.25		20-20k	1.75	5.25 19 15	84		
7001	2	550	715	850				less than 0.02		20-20k +0,-3	1.4	3.5 19 15	53.5		
6001	2	400	600	720				less than 0.025		20-20k +0,-3	1.2	3.5 19	53.5		
4801	2	300	480	600				less than 0.025		20-20k +0,-3	1.1	3.5 19 15	50		
FA601	2	120	225					less than 0.25		20-20k +0,-5	.775	3.5 19 13	26		
LA601	2	120	225					less than 0.25		20-20k +0,-5	.775	3.5 19 11.5	30		

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
LA901	2	225	300					less than 0.25		20-20k +0,-5	.775	3.5 19 14	40		
LA1201	2	280	450					less than 0.25		20-20k +0,-5	.775	3.5 19 14	40		
CROWN INTERNATIONAL, INC.—See our ad on page 5															
MA600	2	235	340	410	20-20k	.05	.05	.05	.05	20-20k 0.01	.725	3.5 19 16	39	$1,295.	
MA1200	2	320	495	200	20-20k	.05	.05	.05	.05	20-20k 0.01	.775	3.5 19 16	44	$1,595.	
MA2400	2	520	820	1100	20-20k	.05	.05	.05	.05	20-20k 0.01	.775	3.5 19 16	51	$2,095.	
Macro Refer-ence	2	260			20-20k	.05	.05	.05	.05	20-20k 0.01	.775	7 19 16	62	$3,500.	
CT200	2	100	155		20-20k	.05	.05	.05	.05	20-20k 0.1	.775	3.5 19 16	21	$790.	
CT400	2	210	230		20-20k	.05	.05	.05	.05	20-20k 0.1	.775	3.5 19 16	31	$1,050.	
CT800	2	305	410		20-20k	.05	.05	.05	.05	20-20k 0.1	.775	5.25 19 16	46	$1,550.	
CT1600	2	540	850		20-20k	.05	.05	.05	.05	20-20k 0.1	.775	7 19 16	57	$1,990.	
DBX BX1	4	100	200	350	20-2ok	.05	.05	.05	.05	20-20k .075	1	7 19 25	85	$3700.	Configurable amp- 2,3,4 ch. Load-invariant response. Will drive .5 ohms. 100A current pk.
ELECTRO-VOICE															
AP2300 SA	2	100	150		10-50k	less than 0.1	less than 0.1	less than 0.1	less than 0.1	20-20k 1	.775	5.25 19 11	32	$810. $914.	Available with precision stepped attenuators. Has rear mounted level controls and octal crossover sockets.
AP2600 SA	2	200	300		7-85k	less than .03	less than .03	less than .05	less than .01	20-20k ±1	.775	5.25 19 12.75	39	$1,040. $1,146.	Same
AP3200	2	400	600		10-90k	less than .05	less than .05	less than .02	less than .01	20-20k ±1	.775	5.25 19 15.75	52	$1,890.	
7300	2	200	300		7-85k	less than .03	less than .03	less than .05	less than .01	20-20k ±1	.775	5.25 19 12.75	39	$1,060.	Each amp individually measured and certified for power and distortion.
7600	2	400	600	850	10-90k	less than .05	less than .05	less than .02	less than .10	20-20k 1	.775	5.25 19 15.75	52	$1,800.	Same
FOSTEX CORPORATION OF AMERICA—See our ad on page 9															
AP1020	2	75	100		10-35k				.05		.70	1.75 19 16.6	17.8	$995.	
AP2130	2	450	650		10-35k				.05		1.72	3.5 19 17.4	50.6	$2,000.	

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
HAFLER PROFESSIONAL/DIVISION OF ROCKFORD CORPORATION															
P1200	1 or 2	60	85		10-40k		.005		.01	10-40k -.5	1.1V	3.25 19 9.5	18	$500.	Has 60 watts per channel, balanced 1/4 in. and XLR inputs, lateral mosfet outputs, level controls.
P2400	1 or 2	120	200		4-40k		.005		.025	4-40k -.5	1.2	5.25 19 10.5	27	$630.	Has 120 watts per channel, balanced 1/4 in. and XLR inputs, lateral mosfet outputs. I
P5000	1 or 2	325	450		10-40k		.01		.025	10-40k -.5	1.5	3.5 19 14	40	$1,200.	Yields 325 watts per channel, balanced inputs, lateral mosfet outputs, front panel level controls.
HILL AUDIO															
DX1000A	2	500	800		10-40k	.002	.003	.003	.02	20-20k ±0	+6 dBm	3.75 19 18	40	$1,999.	All "DX" series amps feature sliding bias 'super A' configuration circuitry,.
DX2000	2	400	650	1000	10-40k	.002	.003	.003	.02	20-20k	+6 dBm	3.75 19 18	78	$2,499.	Same
DX3000	2	550	900	1500	10-40k	.002	.003	.003	.02	20-20k	+6 dBm	3.75 19 18	81	$3,099.	Same
CHAMELEON	2	400	600		10-40k	.005	.005	.003	.01	20-20k	+4 dBm	1.75 19 22	29	$1,599.	Single rack space, fan cooled, exponential heatsinks, "Head Lok" limiting circuitry, full protection circuitry.
INDUSTRIAL RESEARCH PRODUCTS, A KNOWLES DIVISION															
DH4020	2	100	140		20-20k			less than .01	less than .01	-1.0 rated power	1.75	13.5 19 14		$1,342.	Passive cooling, 100kHz switching power supply, MOSFET design, thermal prot., mute on/off, clip ind.
INNOVATIVE ELECTRONIC DESIGNS, INC.															
6208	1	200			15-24k	2.0	2.5	0.7	0.5	20-20k +0, -1.1	890 mv	2.38			Transformerless balanced floating output.
6270	1		200		20-27k	0.35	1.5	0.2	0.25	50-20k +0, -1.0	880 mv	2.38			Same
5201-8	2	8			10-100k	less than	less than	less than	less than	15-100k 0.1	150 mv	less than 1 lb.			Modular design, available as 54 different versions of mixer amps.
JBL PROFESSIONAL															
6290	2	300	600		20-20k	less than 0.1	less than 0.1	less than 0.1	less than 0.1	20-20k +0,-1	1.1	7 19 14	63	$1,650.	Balanced bridging input circuitry, full complementary driver and output circuitry.
SR6615	2	75	150	250	20-20k	less than 0.1	less than 0.1	less than 0.1	less than 0.1	+0,-1	1.1	3.5 19 17.5	32	$645.	Has 2 rack space unit, variable speed fan, rear to front coling system,.
SR6630	2	150	300	500	20-20k	less than 0.1	less than 0.1	less than 0.1	less than 0.1	+0,-1	1.1	3.5 19 17.5	34	$895.	Has 2 rack space unit, modular power supply and amp channels, balanced XLR and 1/4 in. phone input.
SR6650	2	300	500	650	20-20k	less than 0.1	less than 0.1	less than 0.1	less than 0.1	+0,-1	1.1	3.5 19 17.5	42	$1,195.	Has 2 rack space unit, bridgeable to 1000 watts into 8 ohms,.
ES150	2	75	150		20-20k	less than 0.1	less than 0.1	less than 0.1	less than 0.1	1	1.1	3.5 19 14.5	32	$645.	Has 2 rack space, individual channels removeable as single units.

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ES300	2	150	300		20-20k	less than 0.1	less than 0.1	less than 0.1	less than 0.1	1	1.1	3.5 19 14.5	34	$755.	Has 2 rack space unit, continuous variable speed fan, provides quiet, cool operation.
ES600	2	300	600		20-20k	less than 0.1	less than 0.1	less than 0.1	less than 0.1	1	1.1	3.5 19 14.5	40	$990.	Has 2 rack space unit, gain setting lock out function, variable speed rear to front cooling.
ES1200	2	300	600		20-20k	less than 0.1	less than 0.1	less than 0.1	less than 0.1	1	1.1	5.25 19 14.5	39	$1,595.	Has 3 rack space unit, turn-on surge limiter, rear to front forced air cooling system.
PANASONIC PROFESSIONAL AUDIO SYSTEMS															
WP-9055	2	50			10-85k	less than 0.05	less than 0.05	less than 0.05	less than 0.05	20-20k	+4 dBv	1.75 18.9 13.13	19	$540.	U.L. listed and carries five year limited parts and labor warranty.
WP-9110	2	100	150		10-85k	less than 0.05	less than 0.05	less than 0.05	less than 0.05	20-20k	+4 dBv	3.5 18.9 15.06	28.6	$699.	Same
WP-9220	2	200	300		10-85k	less than 0.05	less than 0.05	less than 0.05	less than 0.05	20-20k	+4 dBv	5.25 18.9 15.06	38.6	$899.	Same
WP-9440	2	350			10-60k	less than 0.06	less than 0.06	less than 0.06	less than 0.06	20-20k	+4 dBv	5.25 18.9 19.13	75	$1,810.	Same
PASO SOUND PRODUCTS															
Alpha 600	2	180	260		5-80k		.004			5-100k	1	5.5 19 14	25	$600.	Rack mountable, high current capability.
Alpha 800	2	250	420		5-80k		.004			5-100k	1	7.5 19 18	40	$900.	Rack mountable, dual output meter, 4-way protection circuit.
PEAVEY ELECTRONICS CORPORATION/AUDIO MEDIA RESEARCH—See our ad on Cover III															
Deca 528	2		200		20-20k			0.1	0.1		1V	1.75 19 13.5	12	$799.99	Digital energy conversion (all digital), fan cooled, stereo lightweight, rack space efficiency.
Deca 1200	2		600		20-20k			.06	0.15		1.3	3.5 19 18	37	$999.99	Same
CS1200	2	350	600		10-50k				.05	5-60k 1	1.4V +3 dBv	7 19 18	71	$1399.99	Stereo, DDT compression, fan cooled, calibrated level controls, crossover module port, bridge capability, line balancing capability.
CS1000	2	300	500		10-50k				.05	5-60k	1.4V +3 dBv	5.25 19 14.25	53	$999.99	Same
CS800	2	240	400		10-50k				.05	5-50k	1.4V +3 dBv	5.25 19 14.375	45	$799.99	Same
CS400	2	120	200		10-50k				.05	5-50k	1.0V 0 dBv	5.25 19 14.375	40	$699.99	Same
PMA701	2	30			20-20k			.01	.01		.775	5.25 7 8.25	10	$249.99	Dual channel, 100 watts instantaneous power per channel, half rack width, rear mounted levels.
PMA200	2	100			20-20k			.008	.008		1.0	5.25 19 12.125	29	$429.99	Has 100 watts per channel at 8 ohms, calibrated LED metering, 3 rack spaces high.

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QSC AUDIO PRODUCTS INC.															
MX 700	2	150	225	350	5-65k	.01	.025	0.1	0.1	20-20k 0.25	1.0	3.5 19 12	25	$598.	Has 1/4 in and barrier strip inputs, fan cooled, compact package.
1100	2	50	70	90	5-100k	.01	0.01	0.1	0.1	20-20k +0,-1	1.0	1.75 19	12	$568.	Has 1/4 in., XLR, and barrier strip input—headphone jacks, front.
1200	2	100	150	250	5-60k	.01	.025	0.1	0.1	20-20k 1	1	5.25 19 9.5	24	$598.	Has 1/4 in. XLR, and barrier strip inputs, optional fan cooling, rear gain controls.
1400	2	200	300	450	5-65k	.01	.025	0.1	0.1	20-20k 1	1	5.25 19 9.5	34	$798.	Has 1/4 in. XLR and barrier strip inputs, fan cooled, rear gain controls.
MX 1500	2	330	500	750	8-300k	.01	0.02	0.1	0.1	20-20k 1	1	3.5 19 17.9	47	$1,098.	Dual mono design, fan cooled, 1/4 in. barrier strip inputs.
3500	2	300	450	700	8-300k	.01	0.02	0.1	0.1	20-20k 1	1	3.5 19 15.9	50	$1,488.	Dual mono, front removable channel modules, convection cooled.
3800	2	375	600	850	8-300k	.01	.025	0.1	0.1	20-20k 1	1	5.25 19 15.9	75	$1,958.	Same
EX 4000	2	720	1100	1400	8-100k	.01	0.05	0.1	0.1 20-10k	20-20k	1.0	5.25 19 17.9	64	$2,298.	Open input architecture, speaker connectors, advanced thermal mgmt. system..
RANE CORPORATION															
MA6	6	100	150		5-80k	0.1	0.1	0.2	0.2	5-80k 3	.775	5.25 19 11.5	44	$1,499.	Built-in 15dB limiter for each channel; 25/70V output transformers available for distribution systems.
RENKUS-HEINZ, INC.															
P2000	2	300	500	600	20-20k		.015		.025 +0,-.5	20-20k	1.1V	3.5 19 16	45.3		Clip gard protection, thermal protection, front panel channel status indicators, variable speed fans.
P2500	2	900	600	700	20-20k		0.5		.025	20-20k +0,-.5	1.2	3.5 19 16	48.5		Same
ROSS SYSTEMS															
MEGA AMP 400	2	150	200		10-40k	.05	0.1	.05	0.1	20-20k 0	+2 dBu	3.75 19 15.13	40	$599.95	Features short circuit and thermal protection, current limiting, passive cooling with massive heat-sinks.
MEGA AMP 800	2	245	400		10-40k	.05	0.1	.05	0.1	20-20k 0	+2 dBu	5.25 19 12.63	38	$799.95	Same
SHURE BROTHERS, INC. —See our ad on Cover IV															
210	1	6	10		100-15k		1	3	100-	40mV 15k	2.75	2.13 9.5 5.63		$125.	Balanced mic input, unbalanced line in-put, ext. 12V power.
SOUNDCRAFTSMEN															
PM860	2	210	315	450	20-20k	.05	.05	.008	.05	20-20k	1.2	5 8.5 14	20	$599.	Has high current design to allow stability with 2 ohm loads.
450X2	2	210	315	450	20-20k	.05	.05	.008	.05	20-20k	1.2	5.25 19 11.75	28	$849.	High current MOSFET amp with balanced or unbalanced inputs.

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
900X2	2	375	675	900	20-20k	.05	.05	.008	.05	20-20k	1.22	5.25 19 16.5	59	$1,599.	Same
RA7501	2	275	420	320	20-20k	.05	.05	.05	.05	20-20k	1.21	7 19 15	47	$949.	Class H signal tracking design for maximum efficiency.
300X4	2, 3 4	600 205	900 300	450	20-20k	.05	.05	.008	.05	20-20k	1.0	5.25 19 14	60	$1,399.	Multi-channel MOSFET, 2, 3 or 4 channel mode indicators, front panel-mounted circuit breakers.
DJ600	2	125	190		20-20k	.05	.05	.008	.05	20-20k 0.1	0.95	5.25 19 14	30	$599.	MOSFET output, 1/4 in. input and output jacks.
DJ900	2	205	300		20-20k	.05	.05	.008	.05	20-20k 0.1	1.0	5.25 19 14	30	$749.	High current MOSFET, 1/4 in. input and output jacks.
SOUNDTECH															
PL150	2	55	75		15-30k			.007			1.23	1.75 19 8.5	17	$419.90	Single rack space, clip, protect, power and bridge/mono LED indicators, protection circuit.
PL250M	1	113	200	300	18-31k			.007			1.23	3.5 19 16	30.8	$449.90	Built-in 9 band graphic EQ, clip, protect, power and temp. LED indicators protection circuitry.
PL500	2	165	250		20-20k			.007			1.23	3.5 19 16	39	$599.90	Clip, protect, power, temp. and bridge/mono LED indicators, protection circuity,
PL1000MP	4	165	210		20-20k			.007			1.23	3.5 19 16	59	$1099.90	Four channels, clip protect, power temp. and bridge/mono LED indicators. Protection circuitry.
A300	2	124	169		10-34k			0.01			1.5	3.75 19 15.5	18.7	$629.90	Status indicator LEDs for clip, protect and level indication. Balanced/unbalanced inputs.
A600	2	240	324		10-50k			.008			1.5	5.25 19 15.75	38.6	$819.90	Same
A1000	2	312	480		11-28k			.005			1.5	5.25 19 15.75	47.4	$1199.90	Same
SPECTRA SONICS															
701	1-	33	58	86	20-20k	less than .05	.075	less than .025	.025	DC-20k	+5	2.5 10 1.88	.881	$108.00	A modular amp for bi, tri, qual and multi-way amplification, used in broadcast and recording.
701BP	1-	122	172	200	20-20k	less than .05	.075	less than .025	.025	DC-20k	+5 dB	5 10 1.88	1.76	$216.00	Is two 701s bridged together with the same qualifications.
712B	2	30	50	80	20-20k	less than .05	.075	less than .025	.025	DC-20k	0 dBv	5.5 19 14.5	22	$595.00	Is a 3.5 inch rack mount, self-contained power amplifier.
712	2	100	100	100	20-20k	less than .05	.075	less than .025	.025	DC-20k	0 dBv	5.5 19 14.5	24	$760.00	Same physical features as above.
STEWART ELECTRONICS															
PA50U	2	25	50	50	5-100k		0.02		0.07	20-20k	1	1.7 8.5 6.2	3.5	$249.	Switch mode power supply design reduces size and weight. Set up for unbalanced signals.
PA50B	2	25	50	50	5-100k		0.04		0.03	20-20k	1	1.7 8.5 6.2	3.5	$279.	Same, and balanced version also features level controls and clipping indicators.

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, $	Features
PA100U	2	50	90	100	5-100k		0.08		0.04	20-20k	1	1.7 8.5 10.2	5.5	$324.	Switch mode power supply design reduces size and weight. Set up for unbalanced signals.
PA100B	2	50	90	100			0.07		0.04	20-20k	1	1.7 8.5 10.2	5.5	$359.	Same, and setup for balanced or unbalanced; balanced version features level control and clipping indicators..
PA1200	2	250	400	600	3-180k		less than 0.1		0.06	20-20k 1	750 mV	3.5 19 12.53	15	$999.	Fully dual-monaural design utilizing switch mode power supply.
STUDER REVOX AMERICA, INC.															
B242	2	200	300		20-20k +0,-0.3		0.01		0.01 +0,-0.3	20-20k	1.55	18.4 6 14.2	40	$2,900.	MOSFET drive and special bipolar power transistors, two power transformers, mono bridgeable.
B150	2	80	100		20-20k +0,-0.2		.006		.006	20-20k +0,-0.2		17.7 4.3 13.1	26.6	$895.	
A68	2	150	250		30-15k +0,-0.5		0.1			30-15k +0,-0.5		19.5 5.5 13.5	46	$995.	Fully complementary from input to to output, mono bridgeable.
SUNN															
SPL7250	2	150	250		10-50k	less than 0.05	1	0.03	1	5-50k +0,,-3	+4 dBv	3.5 19 14	25	$749.	Two speed fan, switchable limiter, male and female XLR input connects and balanced 1/4 in. 500 watts at 8 ohms.
SPL7450	2	275	450		10-50k	less than 0.05	1	0.03	1	5-50k +0,,-3	+4 dBv	5.25 19 15	38	$899.	Two speed fan, compressor, male and female XLR input connectors and balanced 1/4 in. 900 watts at 8 ohm bridge mode.
SYMETRIX															
220	2	20	20		20-20k			.02 at 1k	.03	20-20k	.5	1.75 19 12	9	$349.	Stereo, 2-channel or mono-bridged operation. Balanced XLR/balanced and unbalanced 1/4 in. inputs.
TOA ELECTRONICS, INC.															
P-1030D	2	100	150			0.02	0.05	0.03	0.01	20-20k +0,-2	+4dB	5.25 19 13.75	39.8		
P-1060D	2	200	300			0.02	0.05	0.03	0.01	20-20k +0,-2	+4dB	5.25 19 13.75	44.2		
P-1090D	2	300	450			0.02	0.05	0.03	0.01	20-20k +0,-2	+4dB	5.25 19 13.75	48.5		
YAMAHA CORPORATION OF AMERICA															
P2700	2	350	500		10-50k	less than 0.03	less than 0.05		less than 0.05	10-50k	1.23	5.25 18.88 17.25		$995.	High-power stereo operation with 500W/channel, or 1,000W in bridged mono operation-forced air cooling.
P2350	2	175	250		10-50k	less than 0.03	less than 0.05		less than 0.05	10-50k	1.23	5.25 18.88 11.25		$795.	XLR and 1/4 in. input jacks, binding post and 1/4 in. output jacks-forced air cooling.
P2075	2	50	75		10-50k		less than 0.05		less than 0.003	10-50k	1.23	3.878 18.88 14.38	9.8	$395.	XLR and 1/4 in. input jacks, binding post and 1/4 in. output jacks-compact and lightweight.
P2160	2	80	125		10-40k	less than 0.03	less than 0.05		less than 0.05	10-50k	1.23	3.44 18.88 14	23	$595.	Same
PC4002M2	4	30	700		10-100k	less than .005	less than 0.01		less than 0.01	10-50k	1.23	7 18.88 18	95	$2,795.	High power "audiophile" monitor amp with calibrated meters.

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AD•VENTURES

BRIAN BATTLES

Homemade "Digital" Effects

• What do you do if you can't afford all those special digital signal processors and hard-disk-based editing systems everyone seems to be installing these days? Don't give up! Many of the things they do used to be done manually. The digital gear just makes it a bit easier.

PRE-ECHO

Record whatever you need normally on deck A. Flip the tape over. Dub it (backwards) from A to deck B and mix a little of B's tape output while recording to make an echo. Now flip the tape over to B and play it back. Voila!

You can experiment with different decay times and levels to get unusual effects. For example, try adding echo to just the first word or sound on tape (the last sound or word, when dubbing the backward version).

PITCH CHANGE OR TIME COMPRESSION

Carefully wrap a piece of masking tape around the deck's capstan. Use one or more turns. Be sure each turn is even; that is, wrap the tape and cut it off with a razor blade so that it makes exactly one full 360-degree layer around the capstan. If it's longer or shorter, you'll get a warbly sound—which could be useful as a special effect. A variation on this is to wrap several turns around the capstan and leave the end loose. Hold the loose end and start the tape, pulling it off so it unravels as the capstan spins. This technique creates a gradual slowing down or speeding up effect, depending on whether you do it during record or playback.

Make sure you painstakingly scrub any tape-adhesive residue off the capstan when you're done, and don't scratch the metal when you trim the tape with a razor blade.

FLANGING

You'll need three tape decks for this. Make an exact duplicate of your source material and load copies on deck A and deck B. Play them back in unison, recording on deck C. Slight variations in the audio waveforms caused by mixing the outputs of two decks with small speed differences will create a "swooshing" sort of effect. This will vary in depth and duration, depending on how close decks A and B hold their speeds.

What happens is that, as the signals are almost precisely matched up, audio waves get 180 degrees out of phase and begin to cancel themselves. Lower frequencies (longer wavelengths) are audible first, and you'll hear a sweeping noise as one deck gradually overtakes the other. You can manually control the effect by varying the output level on one of the two playback decks, and by rubbing a finger gently against the capstan or reel flange (which is how the technique got its name).

CREATE A CROWD

An unusual effect can be created with a bizarre setup I tried one day in the studio. Echo can be created by mixing a tape deck's playback-head output with the source (input) signal while recording. The echo delay time is determined by the distance between the record and playback heads, and by the tape speed.

I wasn't satisfied with the brief interval provided by the inch-or-so of distance between the heads on one deck. Since I had two identical tape deck models side by side in the studio, I used the left-hand deck as the "feed-reel" machine and threaded the tape through both decks' head and capstan assemblies, then onto the takeup reel on the right-hand deck.

Starting the right-hand (playback) deck a fraction of a second first, I recorded a source on the tape via the left-hand deck. There was now a couple of feet between that deck's record head and the second deck's playback head. This made the "echo" effect sound more like a several-second repeat. Switching the tape speeds (simultaneously) lengthened or shortened the delay. At 3¾ or 7½ in./sec., with the decks' heads about 2½ feet apart, this gave enough time to get a complete spoken phrase on tape before it "came back." Longer and shorter phrases can build up to the sound of a mob (especially if you speak in different voices). I used this effect "live" on the air one day to simulate my studio filling up with a murmuring crowd. Having several people on mic at one time sounds more authentic and makes the crowd's "size" multiply faster.

BE CREATIVE

These are just a few simple ways to make the most of ordinary analog recording equipment, the stuff most low-budget or home studios may have available. You can sound like a big-time production company, with a shoestring budget, if you're creative with the equipment you own. This provides three major benefits:

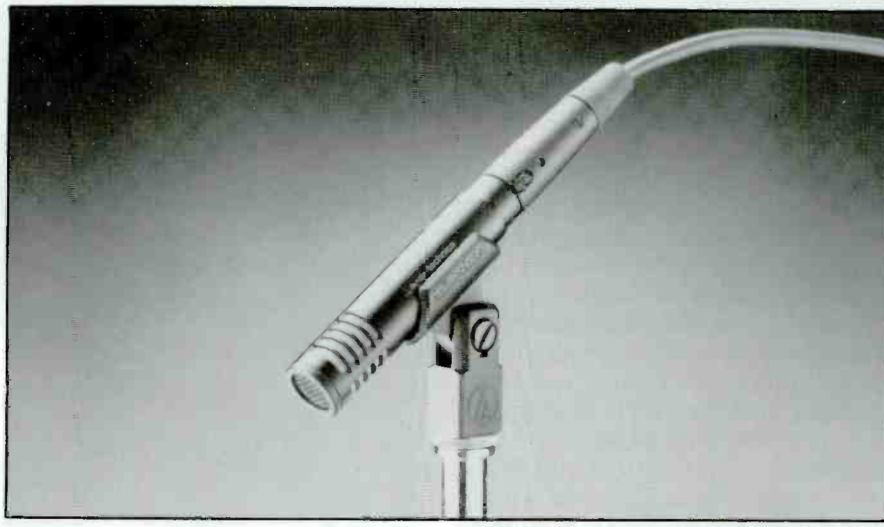
(1) You can capture accounts and earn the money you need to purchase the fancy digital stuff;

(2) You'll appreciate the new gadgets more when you finally get them, and;

(3) Playing around with recording gear like this is fun! Send me your creative, crazy, handy or ingenious radio-commercial production tips and we'll publish the best ones in **Ad Ventures**. db

CONDENSER MIC

• The PRO 37R remote-power condenser microphone is suited for digital sampling, studio recording and line recording where remote phantom power is available. Featuring a very flat frequency response and high sensitivity, it also provides a distortion-free signal in sound fields as loud as 141 dB. The mic can be powered with any phantom supply of 9 to 52V DC. The PRO 37R offers a 30 to 15,000 Hz frequency response, –41 dBm sensitivity, and 141 dB SPL maximum input sound level at 1 percent THD. The mic is 3.9 in. long and weighs just 2.1 oz. The housing has a low-reflectance matte finish, and a foam windscreen and protective carrying case are included.



Manufacturer: Audio-Technica
Price: $159.00
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MONITORS

• A new acoustically transparent grill design compliments the DD6, DD161 and DD261 monitors. Each features improved woofer design, a more efficient motor, higher temperature voice coil, more accurate response, vibration and sound absorbent front and rear gaskets, lead-free silver solder, high strand 15awg. wire, gold-plated binding posts, Watco oil-resin finish, solid state tweeter protection and improved cabinet acoustics.
Manufacturer: Digital Designs
Prices: DD6—$336.00
DD161—$530.00
DD261—$740.00
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SYNCED R-DAT

• The Model D20 is the first to ship with IEC Format SMPTE time code abilities. It has the ability to read and write SMPTE Time code in either the IEC format or the Fostex format. The user can access them both through a simple front panel command. The D20 comes complete with the 8310 system expansion card, which includes the Sony VTR emulation capabilities, as well as the IEC time code format. The D20 eliminates the need for 'middle-man' black boxes. The user can choose BVU-950, BVH-2000, or BVH-3000.
Manufacturer: Fostex Corporation of America
Price: $8000.00
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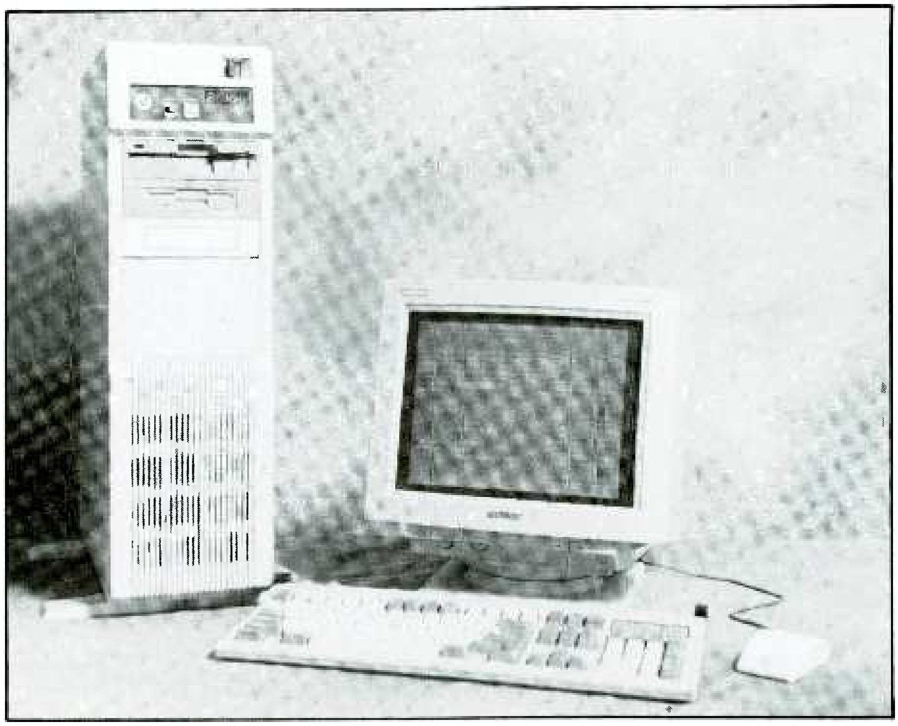
DAT EDITING WORKSTATION



• The ASR digital audio editorial system allows the editing and SMPTE time-code synchronization of stereo sound files recorded to hard disk. The digital I/O option allows DAT material to be loaded, edited, sequenced and transferred back or archived, without leaving the digital domain. Selectable sampling rates, along with time displays in time code, footage/frame, and beats, tailor the system for any application. Sampling rates of 48, 44.1 and 32 kHz are supported in addition to an external sync capability. A built-in MIDI interface allows the use of existing sequencer software. The system can be purchased as a stand-alone or rack-mount unit.
Manufacturer: AQ Design, Inc.
Price: Professional—$1,999.00
Desktop—$1,399.00
Software prices vary.

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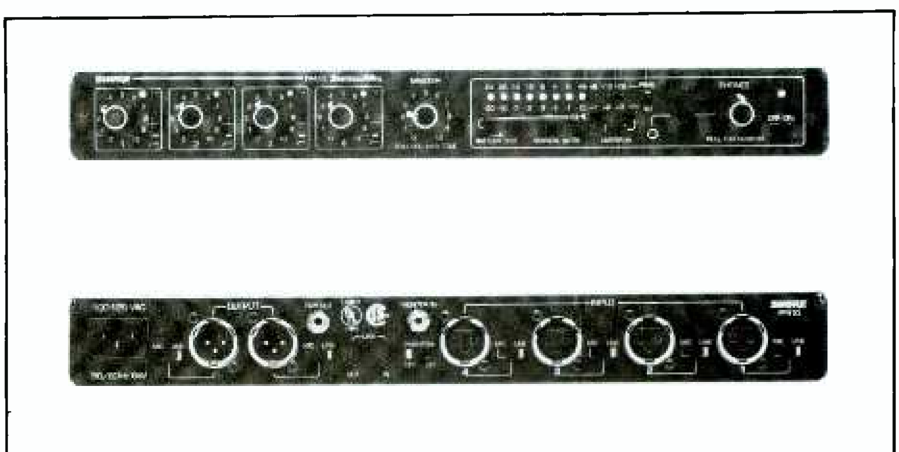
MIC MIXER



• The world's first portable automatic mic mixer, the Model FP410, will find primary applications in the corporate television, broadcast and field production environments. IntelliMix, the patented operational concept behind the FP410, is manifested in three key features of the automatic mixer circuitry: the Noise Adaptive Threshold activates mics for speech but not for constant room noise, such as air conditioning; Max Bus Circuitry limits the number of activated mics to one per talker; and Last Mic Lock-On keeps the most recently activated mic open until a newly activated mic takes its place. The FP410 has four transformer-balanced inputs that will accommodate a variety of mic types. All four inputs can also be switched to line level. Additional FP410 Mixers can be connected via the rear-panel link jacks by using the supplied link cable.
Manufacturer: Shure Brothers Incorporated
Price: $1,595.00

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1991 Editorial Calendar

JAN/FEB The Professional Electronic Cottage.
Winter NAMM Show issue.
• GUIDE: Speakers: Performance & Monitor.

MAR/APR The Broadcast Picture in the U.S.–Applications of the Electronic Cottage to broadcast.
NAB show issue.
• GUIDE: Consoles and Mixers.

MAY/JUNE Audio in Houses of Worship/Sound Reinforcement In Fixed Venues.
NSCA show issue.
• GUIDE: Power Amplifiers.

JULY/AUG Concert Sound–Producing it and/or Recording it.
• GUIDE: Tape, Tape Recorders and Accessories, Microphones.

SEPT/OCT The Recording Studio–What's happening, what's ahead for the 90s.
AES in New York Show issue.
• GUIDE: Signal Processing Equipment, Part I.(delays, reverbs, crossovers, equalizers.)

NOV/DEC The World of Post-Production–Television and Film.
SMPTE Show issue.
• GUIDE: Signal Processing Equipment, Part II, (noise gates, noise reduction, limiters, compressors), Spectrum Analyzers.

NAMM–Winter Jan 18-21 (Anaheim)
NSCA–May 19-21 (Cincinnati)
SMPTE–October 26-30 (Los Angeles)
NAB–April 15-18 (Las Vegas)
AES–October 12-18 (New York City)
NAMM–Winter 1992 Jan 17-19 (Anaheim)

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PEOPLE, PLACES & HAPPENINGS

• **Roland Corporation** has announced the formation of a new **Pro Audio/Video Division**. **Curtis Chan**, formerly with **Ampex Corporation**, has been named general manager for the division. To complement their existing professional product line, Roland is introducing their Sound Space Processing System, a self-contained Digital Multi-track disk-based recorder, and a SBX-1000 MIDI Cueing Box.

• **Disney Entertainment** asked **Aphex** President **Marvin Caesar** to help with the sound during the half-time show of the Super Bowl XXV, and he insisted that ABC put a Dominator II Model 723 with pre- and de-emphasis in front of the uplink to prevent satellite "splatter." Caesar also helped "tweak" the audio for the entertainment portions of the Super Bowl half-time entertainment, including Whitney Houston's rendition of the Star Spangled Banner and President Bush's message. The **Aural Exciter Type III** was used on the voice track to make the hottest Star Spangled Banner in years, and the president's message sounded natural and unprocessed.

• **The Showplace** announces the grand opening of **Showplace Studios** in Dover, NJ. The studio will be the first in the New York Metropolitan area to install the automated **Amek Mozart** console fitted with all **Rupert Neve** modules. The Showplace Studio has also installed the all new **Otari MTR 90** III tape machines and the MACIIci computer with **Digi Design's** soundtools which provides hard disc digital recording and editing.

• **The Society of Motion Picture Engineers** has formed a second section in Australia to cover the states of Victoria, South Australia and Tasmania. New South Wales, Queensland, West Australia and the Northern Territories are already covered in the first SMPTE section.

• **The Sound Recording Technology Program** at the **University of Lowell** in Massachusetts expanded their facilities with the opening of a new 24-track studio. It will be used by advanced students of the SRT program.. The equipment list is too long to mention everything, but representative companies' equipment are a Soundcraft TS24 console with Audio Kinetics Master Mix automation, Studer/Editech Dyaxis (with digital I/O processor and dual 320 meg drives), Adams-Smith Zeta Three synchronizer with Zeta remote auto locator, Lexicon 480L with LARC and SME board, Yamaha REV7, Yamaha NS-10Ms, E-mu Emulator II HD and E-mu SP1200.

• In a management restructuring, Vice President of Planning **Andreas Koch** has been appointed to vice president and general manager at **Studer Editech Corporation** of Menlo Park, CA, effective immediately. Koch will be responsible for all operations at SEC.

• **Sound Impressions** of Portland, OR, announced that they are the first studio in the Portland area to offer Dolby SR processing on all analog recording tracks. The first client to make use of this technology with be **Craig Carothers** on his project for **Atlantic Records**.

• Twelve **Gauss 3285** coaxials have been installed in Universal's Collapsing Bridge attraction at **Universal Studios** in Hollywood, CA. In addition, there are four Gauss 4843B 18-in. subwoofers in Gauss-designed cabinets to "make the earth move when the bridge collapses." The attraction has been redesigned in an effort to achieve sound realism.

• **John Storyk**, of **Walters-Storyk Design Group, Inc.**, has designed new post production suites and lobbies for **Howard Schwartz Recording** in New York City. Storyk has also completed a major renovation of the **Manhattan School of Music's** recording facility, lobbies, and various student/teacher dining and lounge areas.

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The **PRM™ 308S** is being applauded by top professionals — like ***Don Was, Kenny Loggins, Joe Hardy, and Walter Afanasieff*** — for their uniform frequency response, minimum coloration of signal, a selectable switch for reference and EQ playback, and most of all just for "THE SOUND!"

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- Impedance 8 ohms
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PRM™ 310A

- Mirror-image (left & right) pairs
- High-accuracy three-way system
- Switch-selectable response modes (equalized & reference)
- Acoustic foam blanket reduces baffle reflections
- Impedance 8 ohms

PRM™ 312A

- Mirror-image (left & right) pairs
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- Independent variable control of mid and high frequency levels
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- Acoustic foam blanket reduces baffle reflections

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