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PERFORMANCE/LIVE SOUND

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July/August 1990 VOLUME 24 NO.4

db magazine

serving: recording, broadcast and sound contracting fields

the sound contracting engineer

the recording engineer

THE ELECTRONIC COTTAGE

the broadcast engineer

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—See page 33

About the Cover

• Our *Buyer's Guide* in this issue includes one on Microphones. Accordingly we assembled a quantity of mics from the respective manufacturers and had photographer *Richard Lobell* stage them. We wish to thank *Audio Technica, Crown, Electro-Voice, Fostex, Neumann (Gotham Audio), Sennheiser, Shure, Telex,* and *Yamaha*. See page 51.

Calendar

• **The Society of Motion Picture and Television Engineers' 132nd SMPTE Technical Conference and Equipment Exhibit** will be held Oct. 13-17 at the Jacob K. Javits Convention Center in New York City. Conference Vice President L. John Spring, Jr., of Eastman Kodak Co. in Rochester, NY, also announced SMPTE's 134th Technical Conference and Equipment Exhibit will be held in Toronto, Canada, at the Metro Toronto Convention Centre Nov. 10-14, 1992. The technical sessions, equipment exhibit, Coffee Club and Honors and Awards Luncheon will be held at the convention center, and the Annual Banquet and the Fellows Luncheon will be held in an area hotel.

SMPTE's 133rd conference will be held Oct. 26-30, 1991, at the Los Angeles Convention Center, Los Angeles, CA.

• *Keyboard* and *EQ* magazines have joined forces to present CyberArts International, the first professional conference and performance showcase of emerging interactive and multimedia technologies as they apply to the arts. The event, to be held Sept. 6-9 at the Los Angeles Biltmore Hotel, will bring together computer hardware and software developers, computer animators, videographers, musicians, producers, directors, choreographers and theater artists.

CyberArts International will include presentations by some of the biggest names in the field. In addition, there will be numerous hands-on workshops, as well as art and technology exhibits focusing on computer animation techniques, virtual reality, user interface design, interactive video, multimedia systems, authoring, digital sound and image processing, fund raising, legal issues and more. There will also be expert conference presentations featuring such luminaries as Chick Corea, George Coates, Andy Moorer, Mark Cantor, Brian Eno, Myron Kruger, Jaron Lanier, Hans Zimmer and Bill Buxton, to name a few.

For more information about CyberArts, please contact **Bob Gelman, CyberArts International, Miller Freeman Expositions, 500 Howard Street, San Francisco, CA 94105; Tel.: (415) 267-7646; FAX: (415) 995-2494.**

• The first-ever **Mexican Electronics Design and Production Exposition and Conference—PRODISEN '90**—is scheduled Sept. 18-21, 1990 at Centro de Exposiciones de la Ciudad de Mexico, Mexico City. Covering the electronics manufacturing disciplines of design, fabrication, assembly, soldering, inspection and testing, this first annual meeting in Mexico will offer

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engineering managers from Latin America a convenient opportunity to update themselves about the latest trends, techniques and developments in the industry.

Qualified individuals interested in contributing technical papers outlining insights or advances in the field of electronics design and production to the conference program are invited to submit a 100-200 word abstract. For additional information, please contact: **Harry C. Lepinske, president, International Marketing Services Ltd., 1000 Jorie Blvd., Suite 42, Oak Brook, IL 60521 Tel.: (708) 990-8808 FAX: (708) 990-7706 Telex: 28-7423.**

• **The National Association of Broadcasters** will hold its annual radio convention Sept. 12-15 at the Hynes Convention Center in Boston, MA, covering management, programming, sales/marketing, promotion and engineering. The engineering portion of the conference will start Sept. 11.

The conference, which will feature long-form seminars on AM directional antennas, digital audio broadcasting and radio data systems (RDS) will also feature the 1990 Marconi Radio Awards and a session led by commentator Paul Harvey. Billionaire businessman H. Ross Perot will be the keynote management speaker at the convention during NAB's Radio Management Luncheon Sept. 14. To register by phone, call **NAB toll-free at 1-800-342-2460.**

• The 18th Annual Regional Convention of the Society of Broadcast Engineers, Central New York Chapter 22, will be held Friday, Sept. 14 from 9 a.m. to 5 p.m. at the Sheraton Inn Convention Center in Liverpool, NY (Exit #37 on the New York State Thruway). Technical papers will be presented, in addition to equipment displays and meetings with manufacturer's representatives. For more information, contact **John Soergel, Convention chairman, 25 Cotty Drive, East Syracuse, NY, 13057. Tel.: (315) 437-5805.**

• **Duquesne University** is offering a five-day Recording Arts and Science/Music Technology Seminar from July 23-27 at the Duquesne University School of Music. Workshop topics will include recording theory, acoustics and physics of sound, microphone placement, use of synthesizers and sequencers, studio operation and more. Participants will receive hands-on training in Duquesne's fully-equipped recording studios and music technology laboratory. Guitar enthusiasts of all styles and ages can experience a wide variety of expert instruction during Duquesne University's fourth annual Summer Guitar Workshop, to be held in Pittsburgh, PA Aug. 6-10. The program is open to all amateur or professional guitar and electric bass players interested in gaining additional knowledge and skills in an atmosphere which simulates a university guitar or bass curriculum. Class offerings include studies in classical, rock, jazz and fusion styles. Topics covered include basic theory, performance development, guitar maintenance, recording techniques and music technology. Other courses provide insights into the business of music and promotion, offer tips for a college audition and teach exercises which promote finger fitness and dexterity. The latest instruments and accessories for the guitar enthusiast will also be on display. Among participating suppliers and sponsors are Fender, Roland, Shadow, C.F. Martin, St. Louis Music, Hot Licks, Mesa Boogie, Midco International and Vestax. **For more information, contact the Duquesne University School of Music at (412)434-6080.**

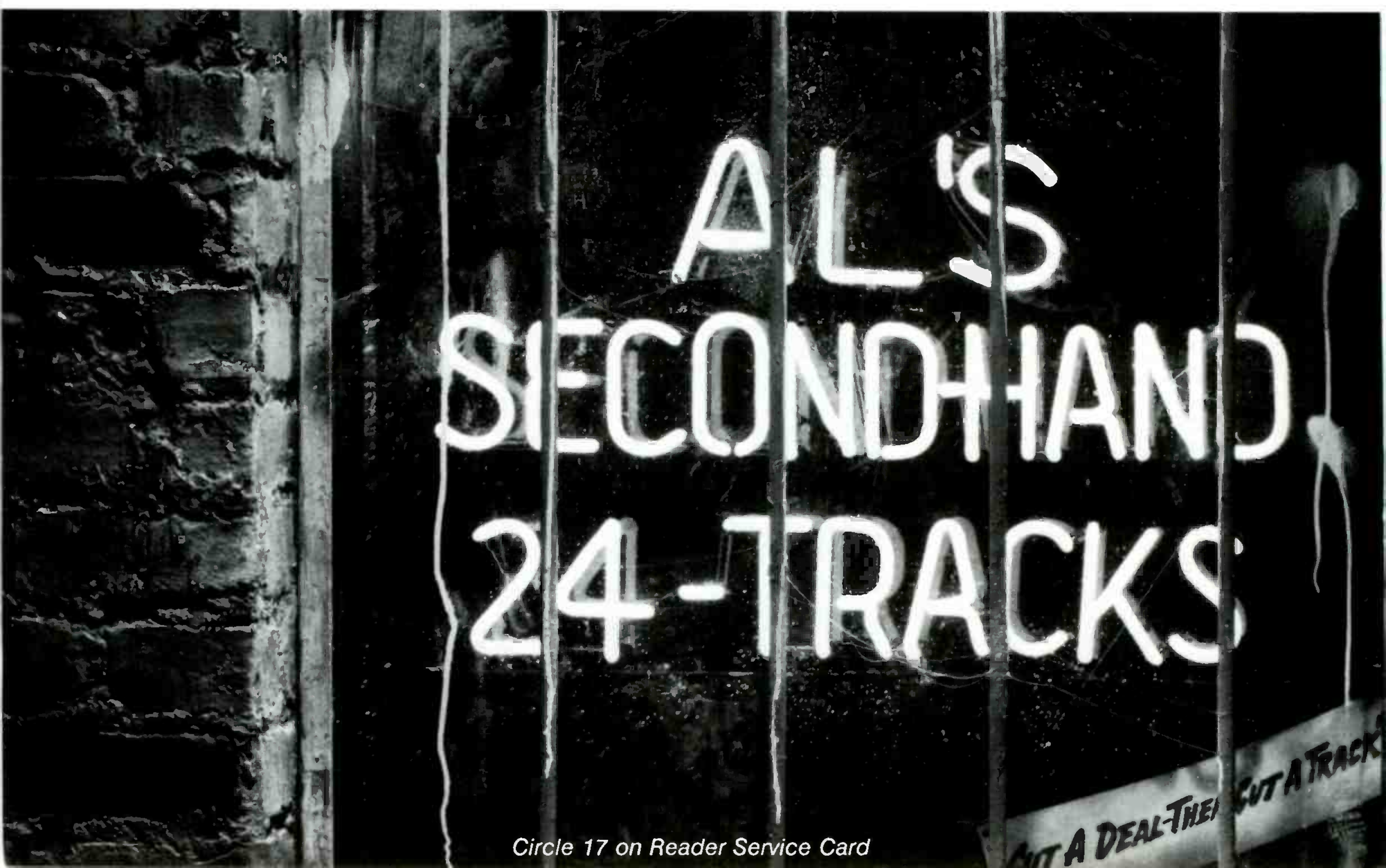
ED LEARNED

Troubleshooting: A Step-By-Step Guide

There are those things that go bump (or buzz, hum, crackle) in the night. Our perapatatic live sound man explores what to do when Murphy surfaces.

In a perfect world, the system cabling meticulously tested in the shop would work perfectly throughout a five-day multi-act outdoor music festival. The amplifiers, control electronics, effects and consoles laboriously checked and lovingly tweaked would continue to meet spec throughout the job. Today's audio equipment is more reliable than at any previous time in the history of modern sound reinforcements. Work long enough in this business, however, and eventually you'll be required to deal with an equipment failure on the job; fate and Murphy will not be denied.

How you respond to a problem, and how long it takes for you to solve it, says a lot about your ability as a system engineer. If there was an ideal way to troubleshoot quickly and efficiently, while maintaining grace under pressure, someone smarter than me would have written a book on the subject and retired by now!



There is no substitute for experience. Familiarity and facility with specific equipment can help narrow your choice of "where to start looking first" quickly.

Knowledge of the signal path, and the operation of each device inserted in it, is the basis of rapid troubleshooting. The subject of failures is fascinating for engineers: I've been part of many discussions where we all traded "war stories," trying to top each other's biggest disaster! Funny, isn't it?

Instead of the successful jobs, the problems are usually what you talk about. At the request of **db**, I'd like to share a few of these, along with their consequences and solutions. I also want to stress my concept of "offensive troubleshooting." I try to anticipate failures, and formulate how I might deal with one. I also believe in eliminating as many variables as I can. The importance of preshow prep and testing is never encouraged enough, both from the equipment and operators perspective. Some of these ideas may sound a lot like common sense, but if that's the case, why don't I see everyone doing them?

ELECTRICITY

You can't cut loose without that juice! Regular readers of my articles on international touring for **db** have noticed I spend a lot of time talking about electrical service at venues. The reason: everything starts from there; without A.C., nothing happens. A show I mixed in Norfolk, VA in the late 70s is indelibly remembered as my most catastrophic A.C. story.

The sound company at this job had an elaborate 3-phase power distribution system with a capacity of 200 amps/leg. This system was tied into a drop down the rear hall from the stage. The neutral wasn't tightened all the way, and somehow came loose during the show. The entire sound system died in mid-performance, and stayed dead for 15 minutes while the problem was traced. The crowd was very impatient, which made for a tense situation. This mess could have been prevented by checking the connection at the drop when it was made. Wise men learn from others' mistakes. Whenever I'm in charge of power distribution, I give all my A.C. feeder wires a tug to insure a tight connection.

I find many of today's engineers are very careless about checking A.C. voltage. Part of the troubleshooting process involves knowing what your parameters are, so get as much information about the A.C. system in your venue (be it stadium, concert hall, or club) as you can. I never plug in anything until checking *all* the A.C. outlets I plan to use with a voltmeter. I measure voltage on the hot, voltage on the neutral, and confirm the equipment ground is operational and the outlet is wired in phase.

Several incidents reinforce this habit we should all have. I once worked a bus-and-truck tour of the Broadway play "Grease," using a sound company out of the Midwest. I was responsible for power distribution, and dutifully checked everything every day. Eventually, budgetary considerations dictated I look for other employment, so I came home while the systems engineer finished the last three weeks. I found out later he learned about checking electricity the hard way. One day an electrician wired his neutral to a hot leg by mistake; when he plugged in some equipment, it fried. *Remember: the time it takes to check A.C. is less costly than your system components.*

Grounding and phasing errors are often the cause of mysterious buzzes and hums that plague sound systems, especially with instrument DI lines. I've worked many music festivals where it's very common for groups to share rental equipment. At a festival in Louisiana, a Yamaha CP-80 piano DI was buzzing badly. Several groups had used it, all with the same results. When I got up there, I started investigating A.C.. The sound crew assured me their ground was good, and it was. However, no one ever thought to check the extension cord feeding the keyboards. While the ground was good at the main box, it disappeared at the end of this cord. We replaced this with a new cord, and our buzz problem vanished along with the bad cable, which had (surprise) a broken ground wire. Everyone called me Houdini, but there was no magic; I just took the time to check it out.

I'm often called upon to provide sound systems and engineer shows in the Michigan Union Ballroom, on the University of Michigan campus,

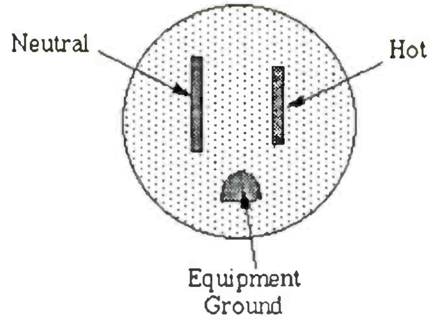


Figure 1. The standard U. S. grounded plug.

near where I live. These systems are small, and run off several 20-amp circuits on wall outlets. One of these is wired backwards: that is, hot and neutral are reversed (*Figure 1*).

I always avoid this outlet, but I've seen several of my competitors come into this room and use the bad outlet for a stack on one side of the stage. They then scratch their heads over why one side of the PA is more noisy than the other. All it takes is the time to check.

ONE MORE A.C. STORY

My last A.C. story involves generator power, and took place while travelling along with the Kinks' 1979 United States tour. We played McDonough gymnasium at Georgetown University.

This building is very old and didn't have enough power to handle the needs of the sound and lighting systems. Showco, the sound company, tied into the house service while Showlites, the lighting company, tied into a generator located inside a truck that was rented for the occasion and parked out back.

The lights began to experience strange problems late in the day, such as ghosting (lights on with the board channel turned off).

The show was delayed while the light guys searched for the cause. It was discovered there was a good 25 volts on the neutral. Electrical code in the United States states that the neutral should be earthed (grounded) before it enters a building, making the voltage upon it zero when referenced to ground. It turned out the generator's neutral was bonded to the truck's box. Physical contact to the earth wasn't possible here, since the box was up in the air, nicely insulated from the ground by four rubber tires. Of course, no one from the generator company was around, and it was pouring rain outside.

As monitor engineer for John "Cougar" Mellencamp, the opening act, I had an interest in starting the show on time, so I suggested a wire be run from the neutral of Showlites' main disconnect (a special breaker where the generator feeds and Showlites power distribution feeds were tied together) to a bathroom water pipe, which was already being used as an equipment ground.

A lively debate began as to who was responsible for this, but eventually it got done, and the problems cleared up with a clean neutral.

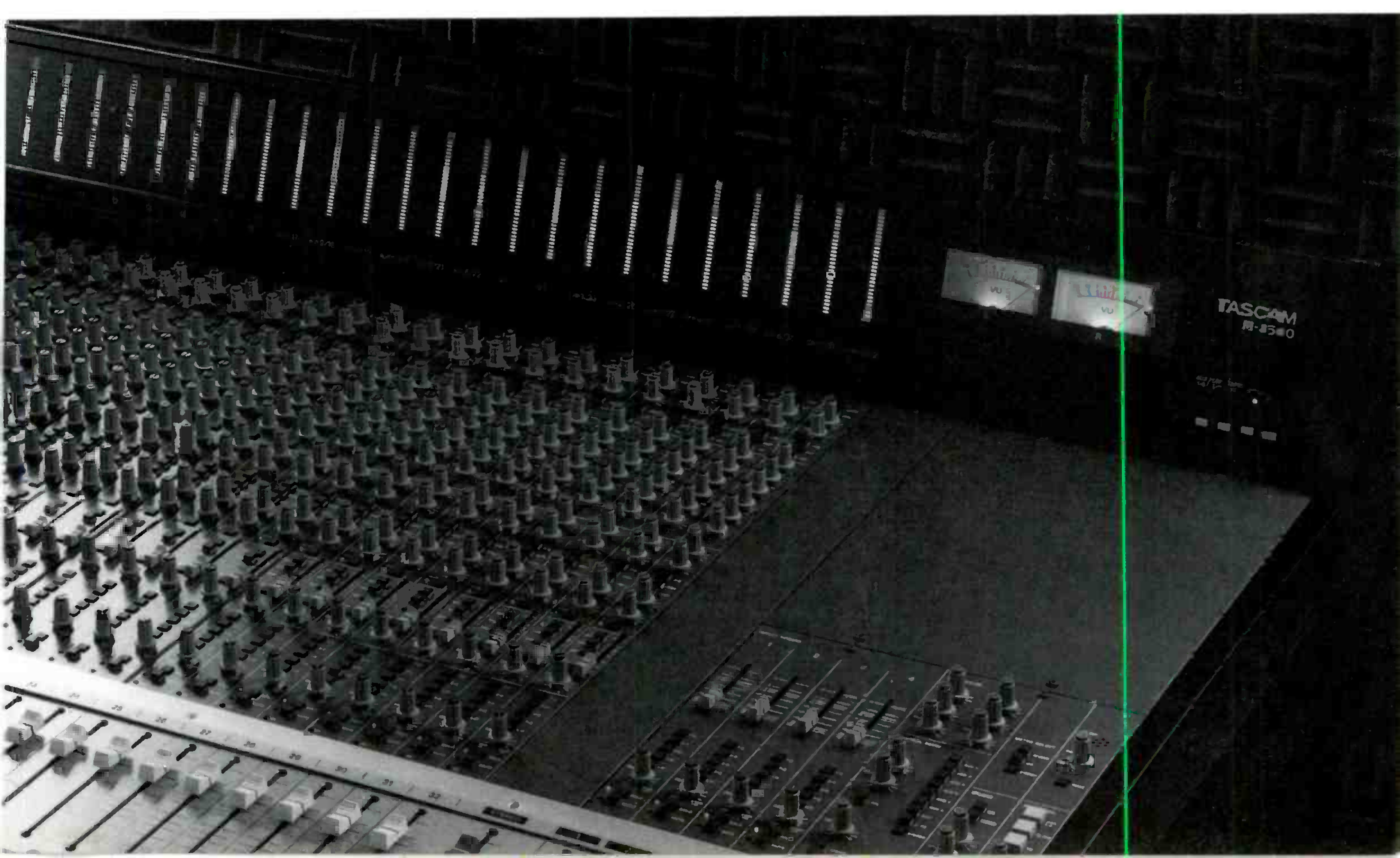
My lesson: insure that both neutral and ground are earthed when using a generator, and do it before the generator people leave.

SOUND SYSTEM COMPONENTS

It's become very unusual these days to find a stereo PA system. I'm aware of the coverage and balance arguments, but one thing no one seems to mention is the protection you get from failure in key system components, such as overall system limiters, graphics, or crossovers. I recall mixing a show on a mono 3-way system in Detroit when, in mid-concert, their UREI crossover failed and I lost everything but the lows.

The PA engineer turned white as a sheet and just walked away, never to return while I was out there. I was left with a system I'd never seen, broken at that, and no help. I could sense the entire audience turning to look at me while I struggled with the sound. I took the wire feeding the lows and plugged it into the left output of the console. This resulted in full-range to the 15-inch woofers, bypassing the single graphic EQ. After a moment to quick mix the vocals to barely audible, I took the right output of the console to the graphic. Using an XLR Y-cord from my shoulder bag (I always carry a few odd adapters with me; I was *very* glad to have this one), I tied the mids and highs together off the graphic output. I rolled off all the lows up to 800 Hz, then mixed in the combined mid/high section using the pan pots. I hoped I took out enough lows not to have blown up any horns, but with the house guy gone, all I cared about was saving the show, which I was able to do. I look back on it now and laugh, but believe me, it wasn't much fun at the time. The band and promoter were very grateful for my efforts; the local music that provided the PA is no longer in the PA business.

In my role of staff engineer for Aerial Enterprises, a major PA company in southeastern Michigan, I've also worn the hat of PA house engineer. When I have a problem, I don't run away from it. Of course, the solution to my crossover failure was much different thanks to stereo and its redundancy factor. Aerial's main system drive rack is fully stereo; a mono system is created by either mono-ing all signals at the console or using half the drive rack. Whenever I'm house engineer, I set up both graphics, limiters and crossovers



identically, even if I'm only using half the rack. Should there be a failure, the unused side can act as a spare.

About two years ago, I encountered a problem working at Chene Park, an outdoor facility on the Detroit River in the heart of the city. I had the system in stereo for a John McLaughlin/Weather Report show when the left side crossover died in the middle of sound check. All I had to do was unplug three system feed wires from the left side and plug them into the right side crossover outputs (Aerial's rack is wired with two receptacles for each crossover output) and I was back in business. The whole process, from failure to solution, took about 15 seconds. This incident inspired the company to install new crossovers, but maintain the same patching system. Contrast this with the changes necessary on the mono system, and you have a compelling argument for either spares or stereo.

CHECKING THE SPEAKER SYSTEMS

Troubleshooting speakers and drivers is a job best performed *before* the gear leaves the shop. I can't even count the number of jobs I've done where I walk up to the stacks and find speakers that don't work, or some that sound radically different than others. At that point, it is too late to do anything about it. I can recall a jazz date I did in Washington, D.C. where, when the bass player hit one particular note, half the bass speakers in the house right stack would buzz most egregiously.

In the summer of 1986, I worked for another major mid-Atlantic sound company as house and monitor engineer. I'd used their system featuring 18-inch folded-horn subwoofers for the first time at an R & B show, and noticed a slight difference in low end between the two sides while tuning the system and during the show. Since the system was mono, the side to side response shouldn't have been that different, so I decided to look more carefully at these speakers in the shop the following week.

My preferred technique for general testing of cone speakers is to apply a moderate level 10 hz tone to the speaker while still in the enclosure. Since 10 Hz is well below the loading of any enclosure I've ever encountered, the sound you hear reveals any abnormalities in the speaker itself, such as rubbing voice coils, tears in the surround or cone paper, or loose dust covers. When I tested these speakers, six out of eight turned up bad, with radial tears in the surround, loose paper where the speaker came unglued from the basket and a few rubbing voice coils. There is no telling how long these speakers were used in this condition, with degraded audio the result. Perhaps other operators noticed it, but couldn't quite put their finger on the problem. In sub-woofer mode, the frequencies were probably too low for anyone but an experienced operator to detect. If all speakers that went out on a job were tested in this manner when they came back, you'd be sure that the speakers you took out next time would be in good working condition. I'd also recommend a frequency-sweep test of each cabinet at least once a year, using the frequency range in which each component will be used. This can help pinpoint harmonic distortion in speakers from cone fatigue, not to mention loose hardware and cabinet rattles that could degrade overall system performance. Speaker and driver phase should also be checked once a year.

These days, most amplifiers have some sort of on-board fan to facilitate cooling during heavy use. Most of these fans also have a dirt or dust filter. Despite the obvious need to clean these filters regularly, I still encounter systems that shut down because of amps overheating due to clogged dust filters.

Another cause of amplifier failure is blown circuit breakers. A high-powered local rock band I know complained to me about their bad experiences in a local club. "The A.C. stinks," they told me. "The voltage is only 105 when we play, we usually pop a circuit at least once and our system sounds bad," they said. I visited them when they played the club again, and discovered they were trying to run two Crown PSA-2 and two Crown DC-300A amplifiers off a single 20-amp circuit. No wonder the voltage was dropping! Today's era of high-power amplifiers require real draw at stable, higher voltage. The venerable Crown DC-300A, once the standard of the industry, may have been shoved aside by newer higher-power amps, but it *does* sound reasonably solid and clean on 105 volts.

Not so the newer amps. Offensive troubleshooting requires you to insure that each amp is supplied with appropriate A.C. power according to draw. Even a dedicated 20-amp circuit is insufficient for bigger amps such as a Crest 8001. If voltage fluctuations in the venue are unavoidable, balance the load as evenly as possible over the maximum number of circuits available. Only then have you put yourself in a position to garner dependability with low-distortion amplification.

Everyone's ultimate nightmare is to have the console go down in the middle of a show. There usually aren't spare 40-channel desks lying around, so you'd plug in your spare power supply and hope that did it. If not, think fast. This tale of desperation comes from my good friend Steve Fisher, current house engineer for the Winans (see **db**, May-June 1990). Fisher was working for Al Jarreau as monitor engineer with Electrotech, a sound company out of Canoga Park, CA. During a concert, the house console died, and the spare power supply did not rejuvenate it. House Engineer Lars Broggard talked to Fisher on the intercom while Fisher created a mix, using headphones for monitoring, on a spare mix buss of the monitor desk. This was sent back down the snake, where Broggard used it to feed the house drive electronics and get the mains working again. They managed to limp by on this for the rest of the evening. Maybe not high-tech, but a great example of quick thinking under pressure.

Checking system cabling is another job that should be done *before* gear leaves the shop. Lets face it: all cabling has a shelf life, and cables are going to break on the job. Going out to a job with broken cables in the inventory, however, is just asking for trouble. There are many cable checkers on the market today, and they are definitely a good investment. Mic cables deserve particular attention. A mic cable with a broken shield wire will work fine on a dynamic mic, but show up totally dead on a condenser mic or phantom-powered active DI. Just because you talk-test a cord with a dynamic mic doesn't mean you are totally safe. Check the shield, too—talk-test with a *condenser*. How often do you see sound companies do a snake check on the job? At the festivals I work in Michigan, it's done at every show. When you start dealing with more than one act a day, the smart play is to insure that all your snake lines work before the first soundcheck. The buzz in that DI might not be an A.C. problem—perhaps the shield wire in that snake channel is broken. Remember to include any sub-snakes that are patched into your main box when doing the test. This practice can help you narrow options should problems arise later. With half-hour set changes between 32-channel bands that want 10+ monitor mixes, the last thing you need is time spent chasing bad wires!

OPERATOR ATTITUDE AND PREPAREDNESS

I really believe an "attitude check" is every bit as important as an equipment check. Confrontational attitudes between crew members and band personnel do not make for an enjoyable experience in a situation where days are long, yet time is short. As far as I'm concerned, no amount of technical brilliance is enough to offset a bad attitude. Diplomacy and respect are two qualities I encourage in crew members. I'm fortunate to be part of the best festival audio staff in Michigan, where personalities mesh well and work is done competently and in a professional manner. Part of the key here is having a plan and sticking to it. Each guy has a specific assignment. For example, one guy is responsible for stage mic'ing and snake patching, one guy does monitors and another does house. Before the day begins, we get together and agree on procedure, things like a generic stage plot designed to fit our biggest band of the day. We fit other bands into that plot, so drum channels stay in the same place, vocals always go from house left to house right in our patch, that sort of thing. It's helpful to designate one person to deal with bands and procure all pertinent information. This should include stage plot, location and number of monitor mixes, a rough idea of the program in each monitor mix and channel assignments to fit the "plot du jour." We try to be understanding of artists' microphone preferences, and hope they are understanding of ours.

I absolutely insist on a line check before each act, so imagine my surprise when I travelled to Atlanta in 1988 for the Democratic National Convention. I was working with the Winans as house engineer, subbing for Fisher. The group was to close out the gospel program of a special concert/benefit/testimonial to the Reverend Jesse Jackson, held at the Fox Theater. I witnessed several gospel groups go on before us, including Al Green. None had line checks; sound for each was a disaster. I watched as Green's engineer searched in desperation for Green's vocal mic while he sang, unheard. I'd say at least 10 channels were mis-patched. I was determined this would not happen to me, and actually held up the show while I line-checked the Winans personally. When they went on stage, all vocals could be clearly heard, and instruments could be located and mixed. Don't cut corners; *always line check*! Talkback from the house through the stage monitor system is essential here; intercoms are nice, but quite often the crew is too busy to notice the call from the house. We assign a channel of the monitor desk for house talkback—it is always on, assigned to all mixes. The house engineer is responsible for turning this mic on and off.

Usually, the vocal mics are the first thing set up in every stage change, which gives the monitor engineer a chance to work on the next act immediately. The house guy can listen in on the stage change, via headphones, over these mics, and converse with the stage guys, via talkback. The entire mix can be walked through very quickly. All this is done with the house subgroups muted, so the audience hears only program music. When I'm in the house, I do the line check for the touring engineer, who is usually only too happy to step to a board where everything works as labeled.

Remember my experience with the left crossover that died? The reason I handled that so quickly was I had pictured those circumstances in my mind and planned how I'd deal with it. I knew what cords to look for, and where they would have to be repatched. I'd even made the patching as neat as I could. It would pay for all of us to sit down and consider just what we'd do if X or Y broke. I also take my shoulder bag to every show I do. It contains tools, headphones, a voltmeter and an assortment of odd audio adapters. As I've learned, you never know what you might need to come up with at the last minute. db db 108 TEKTEXT

DANIEL C. ROGERS

Plotting the PA Angle—A Program

The author has created a computer program that has been written in and runs in basic. It calculates the correct aiming angle for a sound reinforcement speaker.

• First, let's define the problem. Anyone who has ever installed or set up a speaker has faced three questions:

1. *Where do I put it?*
2. *Where do I aim it?*
3. *Will it cover the area I need?*

Unfortunately, most speakers are installed by educated guesswork. The speaker itself is often chosen because a friend or "expert" recommended it, the installer had used it before and was familiar with it, the installer was a dealer for that brand, or because it was all that was available. Often, the only specifications considered are frequency response, power capability, size, weight, color and cost.

The problem is that calculating the required coverage specs for a given application is not very easy. You either use scale drawings and a protractor, or a lot of trigonometry. And to make matters worse, if you did calculate it, most speaker manufacturers either do not provide coverage specs or they are printed on a tiny, unlabeled and illegible graph.

Throw in a little bit of room acoustics and a lack of basic knowledge of how sound travels, and it is little wonder that technicians argue over where the speakers should be placed and aimed. The most common solution to a poor sound system is to turn up the power. If that fails, the speakers are automatically branded as "no good" and "should be replaced without question." The truth is, the speakers are usually working just fine. They may have the wrong coverage angle for the application, or be located and aimed improperly (Throw in some bad EQ, wrong hookup or mic'ing if you need to). More often than not, a proper application of the equipment will correct the problems. Speakers are too expensive to replace for "trial and error" guesswork.

As an example, here are two of the most common errors in speaker setups. The first is the speaker located on the stage floor for lack of a speaker stand. It is aimed straight at the front row. Because the floor slopes up, the people in the front rows are blasted out of their seats while the poor sound technician tries to get enough sound to the people in the back row. The other error is the infamous speaker stand. They make a lot of lightweight fancy stands for big bucks these days, but no one (that I know of) makes one that will allow you to tilt the speaker into the crowd. There appears to be a philosophy that thinks it is better to bounce sound around the room above people's heads than to let them hear it.

A peculiar irony is that acoustical engineering has been around since the days of the Greek amphitheater. Today we have all kinds of pink noise generators, sound pressure level meters and third octave analyzers to test our mistakes. We have third octave parametric equalizers, digital sound, computers and men on the moon, but we can't scientifically select a speaker or its location.

THE THEORY, SOUND DISSIPATION AXIAL LOSS

The first thing you need to understand is sound dissipates with the square of the distance. More simply, it dissipates logarithmically. Even more, you lose 6 dB every time you double the distance. This means if you have a speaker producing 96 dB SPL at 4 feet, at 8 feet you will have 90 dB SPL. At 16 feet, the SPL will be 84 dB. This can be calculated as follows:

dB loss = 20 Log(distance 1/distance 2) –12 dB = 20 Log(4/16)

POLAR PATTERNS

The two primary concerns are the horizontal and vertical response patterns. These are usually graphs (in either circular or linear form) that plot the sound pressure levels at all positions around the speaker in a single axis. Ideally, they will include several frequencies covering 20 to 20 kHz.

The angles of interest are the –6 dB and –12 dB. These are the angles from 0 degrees on-axis where the sound pressure level drops by 6 or 12 dB from the on-axis readings. The -6 dB angle is usually considered the "coverage angle" of the speaker.

Speakers, like microphones, can vary from virtually omni-directional to very directional. Unfortunately, better spec sheets are available on microphones.

SELECTING THE VERTICAL PATTERN

The ideal application for a central overhead system is for the on-axis sound to be aimed at the last listener. The distance from the speaker to the listener on the –6 dB axis should be half the on-axis distance. This causes 6 dB less attenuation by virtue of distance, but that is balanced by the off-axis loss. The –12 dB axis should be half the distance of the –6 dB axis, or one-fourth of the on-axis distance. The overall result is the loss due to axis plus the loss due to distance should be the same for all listeners.

This establishes a ratio of 4 to 1 for the distance from the speaker to the last listener compared to the distance from the speaker to the first listener. The speaker can be placed anywhere on an arc from near 0 degrees straight in front to 90 degrees directly above the first listener. This gives a theoretical range of coverage angles from near 0 to 31 degrees for the -6 dB angle and 150 degrees for the -12 dB angle. See *Figure 1*.

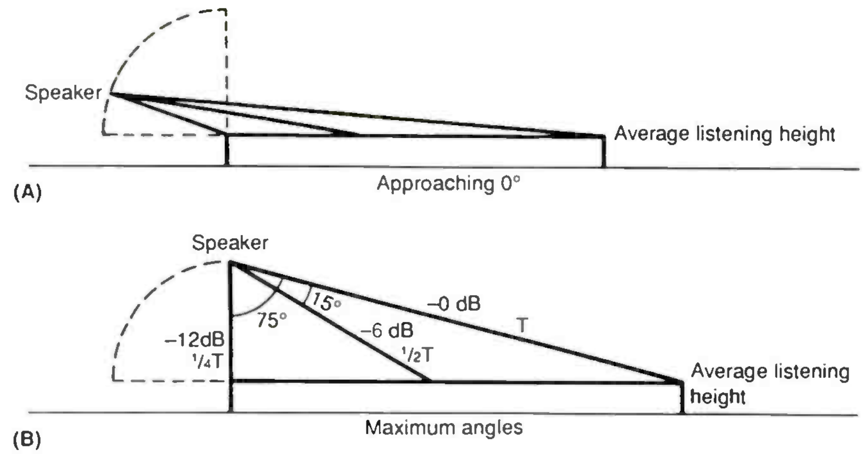


Figure 1. The speaker can be placed within these angles.

SELECTING THE HORIZONTAL PATTERN

The horizontal pattern is most easily selected by taking blueprints for the floor plan and measuring with a protractor.

This should be done after the vertical angles are selected to determine the distance from the speaker to the first listener.

It is helpful to check the vertical angles at several horizontal angles. Room shapes, horizontal coverage patterns and frequency response variations can have radical effects on side angles. When calculating the dB SPL on a side angle, remember to subtract the horizontal off-axis loss from the vertical losses to get the total loss. Typically, a vertical response chart is required for each horizontal angle.

AIMING THE SPEAKER

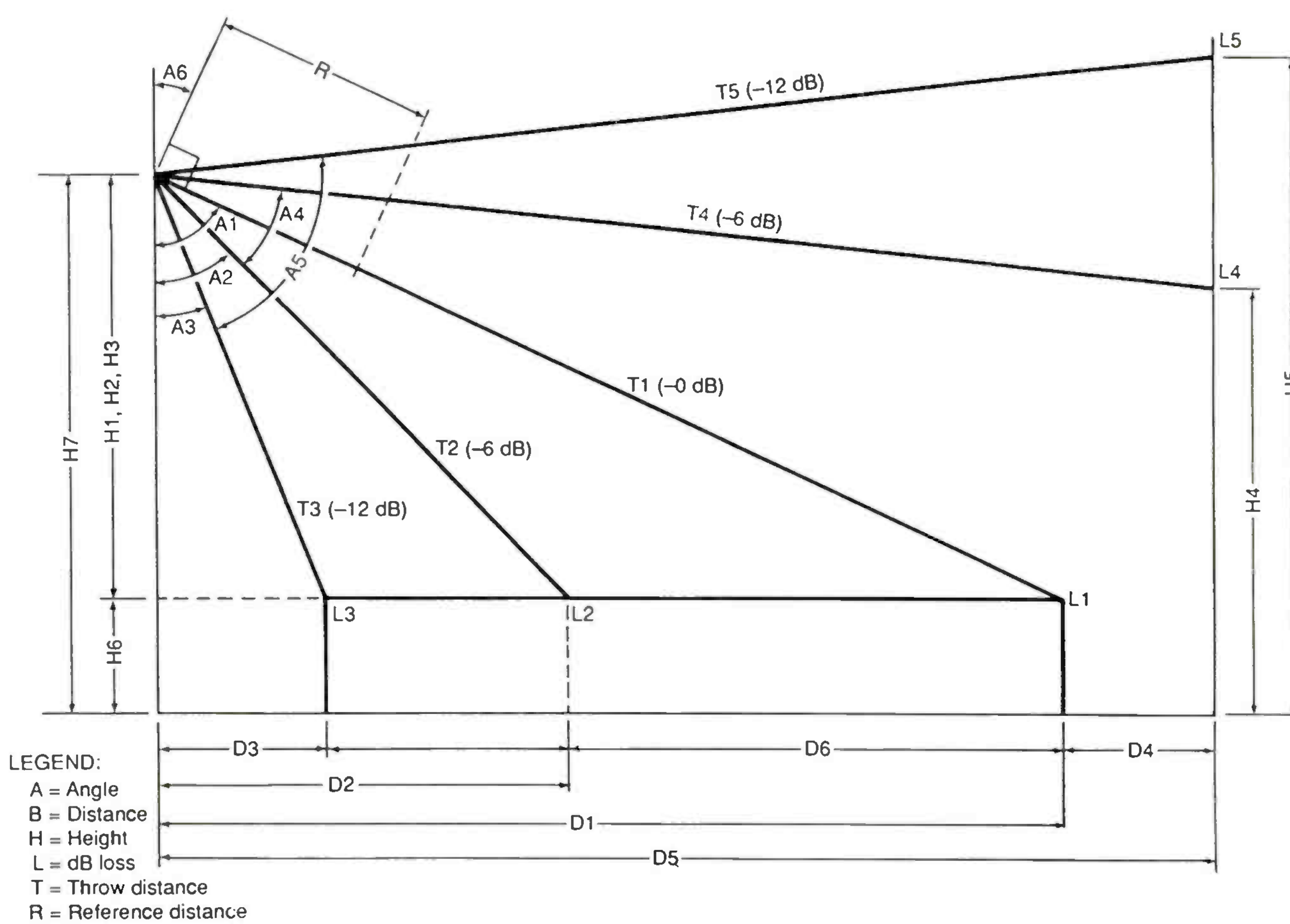
A simple technique for the actual physical aiming of the speaker is to fasten a small mirror to the front of the speaker cabinet (not the cone!). Then take a strong beam flashlight and mount it where the desired on-axis meets the average listening height (A mic stand works good for this). Aim it at the mirror and find where the beam goes. Re-aim the speaker until it flashes back directly at the light. When it does, you have the speaker aimed where you want. I suppose someday someone will put a small laser beam in the box to simplify this crude, but effective technique.

THAT ANNOYING BACK WALL

Two other considerations involve the back wall. First, a direct reflection path from the speaker to the microphone will probably exist. This is of prime importance for acoustic padding to reduce feedback.

The other area of general concern is the other half of the coverage pattern. If the on-axis is aimed at the last listener, and the –6 and –12 dB

Figure 2. The variable names used in the program.



angles are providing the power to the audience below the on-axis, what do you think is going on above the on-axis? The same amount of power you are dumping into the audience (the other half) is heading straight for the back wall. Enter increased reverberant levels, lower intelligibility and increased tendency for feedback. As a rule of thumb, the upper -6 dB angle will hit the back wall at about the same height as the speaker, so the lower the speaker is mounted, the less back wall padding you will need.

ADDITIONAL NOTES

If all that isn't complicated enough, this does not take into account the effects of room reverberation or the Q of the speaker (Q is roughly equivalent to the front to back ratio of a directional microphone). If you shoot for a speaker with a tight pattern and follow the above parameters, you will be surprised at the improvement of the uniformity of SPL and intelligibility. Even if you don't have your own anechoic chamber and can't calculate the room reverb to one percent, these basic principles will go a long way.

THE PROGRAM

The following program was written in an attempt to simplify calculating required speaker angles. It is the first of four program tools to help design and troubleshoot a sound system.

1. Find Ideal Angles From A Given Location
2. Find Ideal Location From Given Angles
3. Find Actual Coverage From Given Angles And Location
4. Find Direct Reflection Path

GENERAL DESCRIPTION

The program is written in GWBASIC. It uses extended precision variables for accuracy. When you type it in to your computer, be careful not to make typos. Switching periods and commas, colons and semicolons are common errors. If you have a different type of basic interpreter, you may need to translate some commands or syntax.

The program has a lot of REM notes. This is to help you figure out what is going on. They have a hierarchy. The most significant sections have five asterisks. Subdivisions are grouped by the number of asterisks.

P.A. SPEAKER ANGLE CALCULATIONS

H.SPKR TO LAST LIST: 100 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: 25 ft 0 in.
		ON AXIS SPKR TO REF: 4 ft 0 in.

SPEAKER TO:	Vertical height Feet	In.	Horizontal distance Feet	In.	Throw distance Feet	In.	dB loss Decibels	Vertical to throw Degrees
LAST LISTENER:	20	6	100	0	102	1	-28.14	78.41
-6 dB LISTENER:	20	6	46	9	51	0	-28.12	66.32
-12 dB LISTENER:	20	6	15	2	25	6	-28.10	36.55
-6 dB BACK WALL:	25	12	110	0	110	0	-34.79	90.51
-12 dB BACK WALL:	89	3	110	0	127	4	-42.06	120.28

-6 dB COVERAGE ANGLE: 24.19°	HOR.FIRST TO LAST LISTEN: 84 ft 10 in.
-12 dB COVERAGE ANGLE: 83.72°	AVG.LIST TO CENT OF SPKR: 20 ft 6 in.
SPKR TILT FROM VERT: 11.59°	

/CR/ TO START AGAIN:

Figure 3. A sample printout of the program.

TWO MAJOR DIVISIONS

The program is divided into two major sections. The first half (through line 9999) is a standard entry module. It is rather generic and will not appear to have anything to do with calculating speaker angles. The REM notes will help if you want to figure out how it works.

The second half of the program is the custom application. It has the program parameters and data to tell the first half how to work, and the actual calculations for the angles.

Figure 2 has the variable names used in the program.

RUNNING THE PROGRAM

The program may be started any way a normal basic program may be run. It may be invoked when basic is started, or loaded and run after basic is started.

Some general rules for operating the program:

1. You may exit to DOS by entering any one of the following letters at the first entry: q Q e E x X t T
2. Entering an "ESC" followed by a carriage return (first field only) will stop the program and return to the command level of basic.
3. A backspace will delete the previous character.
4. An exclamation "!" followed by a carriage return will back up to the previous entry.
5. Entry of an asterisk "*" will restart the program from any entry.

Some rules for entering data:

1. Except for the above, all entries must be numbers.
2. Feet and inches will be added. For example, if you enter 10 feet, 50 inches, it will automatically adjust to 14 feet, 2 inches.
3. A carriage return will default to 0.

If you enter a combination that is not feasible, the program will give an error message stating the configuration is not possible and give the line number of the error. This is a standard error message and will occur even if you have a typo. Look at the program line to find out what the error might be. See *Figure 3* for a sample printout to test your program.

For those of you with color monitors who want to play around, the colors can be changed in lines 10200 through 10390. See your Basic manual for the color options. If you like to have your computer beep at you once in a while, change the "N" to "Y" at lines 10060 and 10030.

The "print screen" function on your computer is an excellent way to get a hard copy of your final results.

STAY TUNED

The next program should be ready for the next issue. This means you should do several things. First, save this issue for reference, because 99 and 44/100 percent of this article applies to the next program and won't be repeated.

Second, make sure your subscription is up to date so you will get the next program. Lastly, if you have any problems, questions or suggestions, please contact me through **db magazine**.

The Basic Program

```
10 REM SPEAKER ANGLE
CALCULATIONS FOR P.A. SYSTEM
20 REM "PANGLE.BAS",A
30 REM V3.5
40 REM 05-11-90
50 REM DCR

100 REM ***** INITIALIZE
110 REM **** SYSTSEM 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 READF$,F0$,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 SYSTEM

10000 REM ***** PROGRAM
VARIABLES

10010 REM **** PROGRAM TITLE
10020 LET G$="P.A. SPEAKER
ANGLE 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=7: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
11015 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
11025 DATA "HORIZ. DISTANCE
FROM SPEAKER TO LAST LISTENER
(INCHES ADDED TO FEET)"
11050 DATA "H LAST LIST TO WALL:
ft"
11051 DATA "N","0","&",1,5,22,5,0,4
11055 DATA "HORIZ. DISTANCE
FROM LAST LISTENER TO BACK
WALL"
11060 DATA "in"
11061 DATA "N","0","&",37,5,30,5,0,6
11065 DATA "HORIZ. DISTANCE
FROM LAST LISTENER TO BACK
WALL "
11070 DATA "V FLOOR TO AV LIST :
ft"
11071 DATA "N","0","&",42,4,63,4,0,4
11075 DATA "VERT. DISTANCE FROM
FLOOR TO AVG. LISTENING HEIGHT"
11080 DATA "in"
11081 DATA "N","0","&",76,4,71,4,0,4
11085 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
11095 DATA "VERT. DISTANCE FROM
FLOOR TO CENTER OF SPEAKER"
11100 DATA "in"
11101 DATA "N","0","&",76,5,71,5,0,4
11105 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
11115 DATA "DISTANCE FROM
SPEAKER TO ON AXIS db SPL REF.
MEASUREMENT"
11120 DATA "in"
11121 DATA "N","0","&",76,6,71,6,0,4
11125 DATA "DISTANCE FROM
SPEAKER TO ON AXIS db SPL REF.
MEASUREMENT"

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$=D$
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 *** FLOOR TO AVERAGE
LISTENING HEIGHT
40080 H6#=(12*VAL(A5$))+VAL(A6$)
40090 REM *** FLOOR TO CENTER
OF SPEAKER
40100 H7#=(12*VAL(A7$))+VAL(A8$)
40110 REM *** DISTANCE FROM
SPEAKER FOR db SPL REFERENCE
MEASUREMENT
40120 R#=(12*VAL(A9$))+VAL(A10$)

40200 REM **** CALCULATE
HORIZONTAL DISTANCES
40210 REM *** SPEAKER TO BACK
WALL
40220 D5#=D1#+D4#

40300 REM **** CALCULATE
VERITCAL DISTANCES
40310 REM *** AVERAGE LISTENING
HEIGHT TO CENTER OF SPEAKER
40320 H1#=H7#-H6#
40330 H2#=H1#
40340 H3#=H1#

40400 REM **** CALCULATE
SPEAKER AXIS THROW DISTANCES
40410 REM *** 0 db ON AXIS THROW
DISTANCE
40420 T1#=SQR(H3#^2+D1#^2)
40430 REM *** -6 db AXIS THROW
DISTANCE
40440 T2#=T1#/2
40450 REM *** -12 db AXIS THROW
DISTANCE
40460 T3#=T1#/4

40500 REM **** CALCULATE HORIZ.
SPEAKER TO OFF AXIS DISTANCE
(THIRD SIDE OF TRIANGLE)
40510 REM *** -6 db AXIS HORIZ.
SPEAKER TO LISTENER DISTANCE
40520 D2#=SQR(T2#^2-H2#^2)
40530 REM *** -12 db AXIS HORIZ.
SPEAKER TO LISTENER DISTANCE
40540 D3#=SQR(T3#^2-H3#^2)

40550 REM **** CALCULATE
USABLE AUDIENCE - FIRST LISTENER
TO LAST
40560 D6#=D1#-D3#

40600 REM **** CALCULATE
ANGLES FOR AXIS FROM VERITCAL
40610 REM *** 0 db ON AXIS
40620 A1#=(180/3.1415927)*ATN
(D1#/H1#)
40630 REM *** -6 db AXIS
40640 A2#=(180/3.1415927)*ATN
(D2#/H2#)
40650 REM *** -12 db AXIS
40660 A3#=(180/3.1415927)*ATN
(D3#/H3#)

40670 REM **** SPEAKER TILT FROM
VERTICAL
40680 A6#=90-A1#

40700 REM **** CALCULATE
SPEAKER COVERAGE ANGLES
40710 REM *** -6 db ANGLE
40720 A4#=(A1#-A2#)*2
40730 REM *** -12 db ANGLE
40740 A5#=(A1#-A3#)*2

40800 REM **** CALCULATE BACK
WALL REFLECTION
```

```
40810 REM *** FIND -6db ANGLE FROM HORIZONTAL IN DEGREES
40820 A7#=(3.1415927/180)*(A4#+A2#-90)
40830 REM *** -6 db THROW DISTANCE
40840 T4#=D5#/COS(ABS(A7#))
40850 REM *** -6 db BACK WALL HEIGHT
40860 H4#=H7#+((ABS(A7#)/A7#)*((TAN(ABS(A7#)))*D5#))

40910 REM *** FIND -12 db ANGLE FROM HORIZONTAL IN DEGREES
40920 A7#=(3.1415927/180)*(A5#+A3#-90)
40930 REM *** -12 db THROW DISTANCE
40940 T5#=D5#/COS(ABS(A7#))
40950 REM *** -12 db BACK WALL HEIGHT
40960 H5#=H7#+((ABS(A7#)/A7#)*((TAN(ABS(A7#)))*D5#))

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 *** -12 db AXIS TO AUDIENCE
41060 L3#=20*(LOG(R#/T3#)/LOG(10))-12
41100 REM *** -6 db AXIS TO BACK WALL
41110 L4#=20*(LOG(R#/T4#)/LOG(10))-6
41120 REM *** -12 db AXIS TO BACK WALL
41130 L5#=20*(LOG(R#/T5#)/LOG(10))-12
49999 RETURN

50000 REM ***** DISPLAY RESULTS - USER SUBROUTINE
50010 REM **** TEXT FORMAT
50020 REM *** DISPLAY FRAME
50030 LOCATE 8,1:PRINT " VERTICAL    HORIZONTAL    THROW VERTICAL"
50040 LOCATE 9,1:PRINT " HEIGHT    DISTANCE    DISTANCE dB LOSS    TO THROW"
50050 LOCATE 10,1:PRINT "SPEAKER TO:  Feet In.  Feet In.  Feet In. decibels  degrees"
50060 LOCATE 11,1:PRINT "---------------------------------------"
50070 LOCATE 18,1:PRINT "---------------------------------------"

50100 REM *** DISPLAY DATA
50110 REM ** LONG LINES
50120 REM * SET MASK
50130 P1$=": ####  ##   ####  ##  ####  ##   ####.##  ###.##"
50140 X=1

50200 REM * DATA
50210 Y=12:P$="LAST LISTENER ":P0#=H1#:P1#=D1#:P2#=T1#:P3#=L1#:P4#=A1#:GOSUB 51000
50220 Y=13:P$="-6dB LISTENER ":P0#=H2#:P1#=D2#:P2#=T2#:P3#=L2#:P4#=A2#:GOSUB 51000
50230 Y=14:P$="-12dB LISTENER":P0#=H3#:P1#=D3#:P2#=T3#:P3#=L3#:P4#=A3#:GOSUB 51000
50240 Y=16:P$="-6dB BACK  WALL ":P0#=H4#:P1#=D5#:P2#=T4#:P3#=L4#:P4#=A1#+A1#-A2#:GOSUB 51000
50250 Y=17:P$="-12dB BK WALL ":P0#=H5#:P1#=D5#:P2#=T5#:P3#=L5#:P4#=A1#+A1#-A3#:GOSUB 51000

50300 REM ** SINGLE DATA LINES
50310 REM * BOTTOM OF SCREEN
50310 X=1:Y=19:P$="-6dB COVERAGE ANGLE  : ####.## deg.":P0#=A4#:GOSUB 51200
50320 X=1:Y=20:P$="-12dB COVERAGE ANGLE  : ####.## deg.":P0#=A5#:GOSUB 51200
50330 X=1:Y=21:P$="SPEAKER TILT FROM VERT: ####.## deg.":P0#=A6#:GOSUB 51200
50340 D#=D6#:GOSUB 51500
50350 X=38:Y=19:P$="HOR FIRST TO LAST LISTEN: #### ft.":P0#=DF#:GOSUB 51200
50360 X=75:Y=19:P$="## in.":P0#=DI#:GOSUB 51200
50370 D#=H1#:GOSUB 51500
50380 X=38:Y=20:P$="AVG LIST TO CENT OF SPKR: #### ft.":P0#=DF#:GOSUB 51200
50390 X=75:Y=20:P$="## in.":P0#=DI#:GOSUB 51200

50400 REM * REFRESH TOP OF SCREEN (INPUTS)
50410 REM HORIZ DIST FROM SPEAKER TO LAST LIST
50420 D#=D1#:GOSUB 51500
50430 X=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
50470X=22:Y=5:P$="####":P0#=DF #:GOSUB 51200
50480 X=30:Y=5:P$=" ##":P0#=DI#:GOSUB 51200

50500 REM VERT DIST FROM FLOOR TO CENT OF SPKR
50510 D#=H6#:GOSUB 51500
50520 X=63:Y=4:P$="####":P0#=DF#:GOSUB 51200
50530 X=71:Y=4:P$=" ##":P0#=DI#:GOSUB 51200

50540 REM VERT DIST FROM FLOOR TO CENT OF SPKR
50550 D#=H7#:GOSUB 51500
50560 X=63:Y=5:P$="####":P0#=DF# :GOSUB 51200
50570 X=71:Y=5:P$=" ##":P0#=DI#:GOSUB 51200

50600 REM REFERENCE DISTANCE
50600 D#=R#:GOSUB 51500
50610 X=63:Y=6:P$="####":P0#=DF#:GOSUB 51200
50620 X=71:Y=6:P$=" ##":P0#=DI#:GOSUB 51200
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#,PI2#,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#=D#-(DF#*12)
51530 RETURN
65535 END
```

JIM PAUL

Breaking Into Concert Sound in L.A.

The author, Jim Paul, is a sound engineer in Southern California. Jim teaches Concert Sound at Orange Coast College, and Recording Engineering at the Audio Recording Technology Institute in Anaheim, CA. He is also owner of Reel Time Productions, an audio/video production company based in Orange CA. In this 1st installment of a three-part article, Jim takes a look at getting started in concert sound in Los Angeles.

Southern California is rich with music venues ranging from smaller, more intimate clubs like The Whiskey in Hollywood or The Strand in Redondo Beach, to larger, more mainstream concert venues such as the Pacific Amphitheater or the Hollywood Bowl. Then there are the huge mega-venues where very large outdoor concerts are held including three stadiums which hold over 70,000 people (the Rose Bowl, Anaheim Stadium, and L.A. Coliseum).

And don't forget the cavernous interiors of the Great Western Forum (home of the Lakers, the Kings and many major concerts), the Shrine Auditorium where the Grammy Awards are held each year and the Dorthy Chandler Pavilion, host to the Academy Awards.

There are literally hundreds of other clubs, theaters and concert venues where music thrives and which require qualified sound engineers.

Each of these venues represents a unique challenge to the sound engineer and together they create a wealth of opportunity that draws talented engineers from all over the United States. But how does one go about "breaking in" to concert sound in L. A., and what will one find when one "breaks in"?

In this three-part series, we shall explore the pathways others have followed into the field, and from their experience we will learn the ups and downs of breaking in to concert sound.

Figure 1. This simple, rustic exterior gives no hint of the the concerts and quality sound system found inside.

About 45 miles south of L.A. is San Juan Capistrano, home to the Coach House, a one-of-a-kind concert hall. With its rustic exterior nestled in the hills of southern Orange County and listed as a 380-seat venue, the Coach House is not, at first glance, a very imposing structure. But upon entering, there seems to be an exciting air about this place which begins with the entrance hallway lined with dozens of pictures of major recording artists who have performed here in the last four years. Each picture is signed with a personal note from the artist. Continuing down the hall, we enter into the "house" area. The stage is small (about 25-feet across by 20-feet deep) and thrusts right out into the audience. The house speakers are permanently flown 12 feet above either side of the stage with a second set further off stage right and stage left.

Looking around, I am struck by the warmth and intimacy of the place. The floors are of wood, and the seats

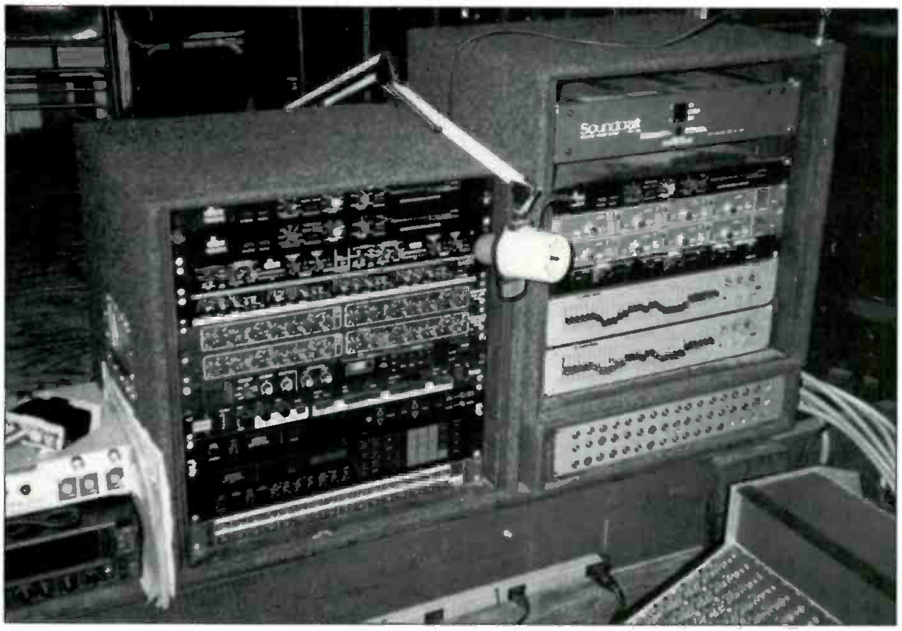


Figure 2. Gates, compressors, reverbs, etc. The outboard signal processing racks.

are gathered in groups of four or six around country style tables (they serve dinner here with the concerts!). Lots of old pioneer paraphernalia hangs from the walls and the overall effect is a recreation of a saloon right out of the Old West. But don't be fooled by appearances. The sound system here is no joke and the concerts are big business. There are two 16 X 8 Yamaha mixers at the monitor position and a large Soundcraft console at the rear in the mixing position (I would later find out it is a 40-input Model 500). Two racks full of signal processing gear are at the side of the mixing console and there are at least three patch bays (see the complete equipment list at the end of this article). There is even a grand piano on a hoist which can be lowered onto the stage as needed!

Scott Yockey, the chief sound man and resident electronics wizard for the Coach House, greets me and begins to unfold the story of his rise from a high school trombone player to head sound engineer here.

"I started off in high school meeting bands and offering to mix their sound," he says. "Most of those early gigs were for free, but then occasionally I would get paid a few bucks. I remember getting paid $40.00 for one gig and that seemed like a lot of money!"

After graduating from Aviation High School in 1972, Yockey went on to community college to take an electronics class. Discovering he really liked the subject, he continued to take electronics classes and studied electronics for two years.

During this time, Yockey made a critical move that would begin to propel his career quickly forward. "While I was still in college, I bought my very first PA system so I could begin renting it out to bands and mix their sound for them," he said. For $60.00 to $80.00 per night, a band got a mixing board, good quality speakers and Yockey to set up and run the whole thing.

"I bought 2 JBL 4560's with radial horns, 2 Crown 600-watt power amps and a 12-channel Tangent mixer. That was real good stuff for those days," he added.

Yockey continued mixing for as many bands as possible and as word about him began to spread, the phone began to ring more often. The most significant work he did during this period was at a local rock club called Club 88, and once he even mixed sound for a larger concert at the Santa Monica Civic Auditorium.

Around 1981-82, he met up with another young entrepreneur who owned a small company called *Rainbow Sound*, and who was looking for a partner. Yockey seemed a natural, and with this partnership, he was on his way to becoming a professional. It was during this time that he began to mix sound in larger and more noted clubs such as *Gazarri's* and

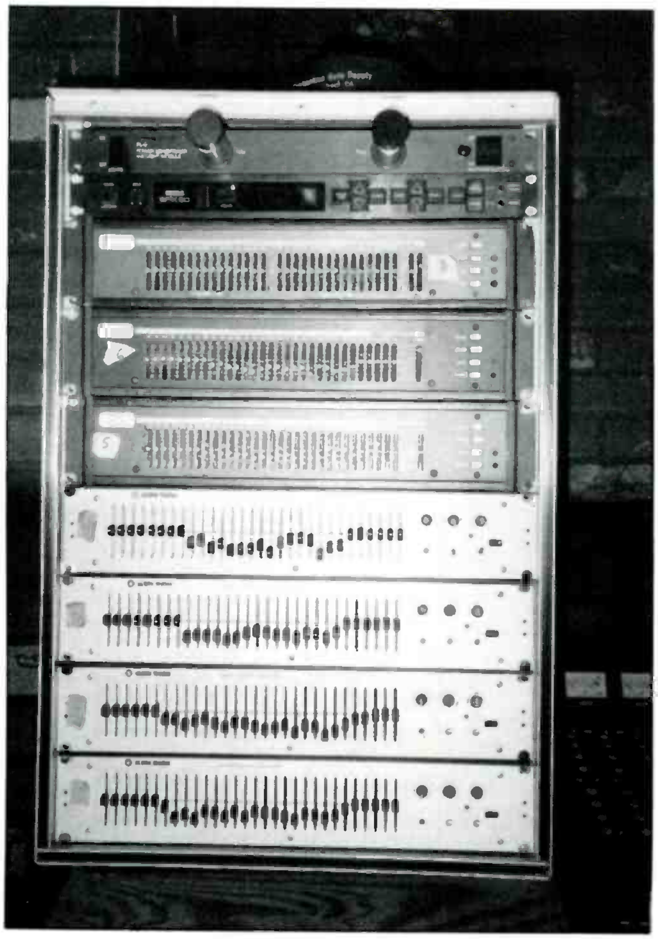


Figure 3. The house and monitor graphic equalizer rack.

Florentine Gardens, both in Hollywood.

"Things didn't work out well for Rainbow Sound and it came to an end, but its ending led me to meet the person who would turn out to be part of the reason I became involved with the Coach House," Yockey said. "I met a guy named Jeff Kaylor and together we formed another sound company called *J & S Sound*."

For the next three years, Yockey and Kaylor continued to rent out PA gear and themselves as sound engineers, but in 1986, an event occurred that would change Yockey's plans. J&S Sound got a call from a Mr. Gary Folgner who said he owned a club in San Juan Capistrano called the *Coach House*. Coach House was switching from a top-40 club format to a concert hall format. Folgner needed to buy equipment for his PA system, so the duo sold the Coach House some PA gear, most of which is still in use today. As months went by, the Coach House kept getting bigger and bigger names to play concerts, and Folgner kept calling J&S Sound to rent or buy more gear.

"This led to Gary Folgner getting to know us and to trust us," Yockey said, "and soon he was calling us up to mix sound at a lot of the concerts at the Coach House. Around this same time my partner Jeff left, and so I was mixing the sound at the Coach House concerts by myself."

This brings us to the present, and today, Yockey is the chief sound engineer here, and works virtually all of the concerts. It's been over four years, and he says he loves doing the job and plans to continue. Yockey and the Coach House have no formal contract, but Folgner is very happy with Yockey's work, and Yockey says the feeling is mutual.

For those trying to "break in" to this business, Yockey dispensed a few "pearls" from the wisdom he has gained in his job. "It's really not as glamorous as it might seem," he said. "I am the equipment loader, house electrician, stage manager and sound mixer. Sometimes I work 16 hour days which can include setting up and moving a lot of equipment, and sometimes the sound is really loud! But you know, I really love the job and plan to continue to do it for a long, long time," he said.

I then asked Yockey if he had any "war stories" to share about his work with the many bands and recording stars who have played at the Coach House. After a few thoughtful minutes, he laughed and said there was a band (name undisclosed) that played here and brought an entire semi-truck stacked to the ceiling with equipment (remember the 25-foot by 20-foot stage!).

"They brought their own speaker cabinets and even their own mixing console!" he said. "Their cabinets blocked part of the audience view...

Yockey and his crew had to unpack the entire thing, set up for one show, mix at around 126 dB SPL for two hours (measured on the house SPL meter), then tear down the entire set-up again, pack it back into the truck and send them on their way.

"They brought their own speaker cabinets and even their own mixing console!" he said. "Their cabinets blocked part of the audience view, and they used special lights, a color changer and a large smoke machine that filled the room so full of smoke that you couldn't even see across the stage! The show was great, but I'll always remember that as production overkill!"

Looking over the impressive list of artists that have played The Coach House, I picked out a few and asked Yockey to share his impressions of each one while working with them in this small, intimate environment.

SHARED IMPRESSIONS

Chick Corea—

"He had a *very* large stage set-up. Chick brought his own 32-channel Studiomaster mixer just for his keyboards. He also had a Yahama MIDI Grand piano and a large MIDI rack. He's incredible, a great musician!"

B.B. King—

"This was a simple set-up. Seems like he just brought his amp and Lucille (B.B. King's famous guitar)! He used all acoustic drums, no MIDI stuff, and a real hammond B-3 organ! He seems quiet, reserved. He just comes in, does the show, then leaves again."

Bonnie Raitt—

"She is someone who likes to work with her own equipment. She brought her own vocal mic- a Beyer M-88, a Fender guitar amp, and she *always* brings her own monitors! Bonnie is friendly, easy to set up for and easy to work with."

Miles Davis—

"Another huge stage set-up. Keyboards and a large percussion set-up stick out in my mind, including MIDI percussion. He seems like a very quiet guy. What a tight sounding band!"

Robin Trower—

"This was a real interesting stage. Robin used a large Marshall stack (very loud!), and we built a little drum cage out of plexiglas baffels to get some isolation on the drums. Robin comes in every year. A very nice person."

Al Diemola—

"His set-up was pretty easy. The band was all acoustic, and incredible. All he used for his guitar was a Lexicon 200 reverb and a guitar preamp! He has great ears. He can detect the smallest changes in sound or volume."

Michael MacDonald—

"Another artist who brought his own microphone for vocals, another Beyer M-88. He had three keyboard players including himself right at center stage, and one on each side. He is a very friendly person. He seems to love to play here at the Coach House."

My final question to Scott Yockey was somewhat philosophical in nature.

Here are major recording artists capable of filling up 10,000-seat and larger venues, and here we are in L.A. with all these *great* places to play, and yet these people are coming to out-of-the-way San Juan Capistrano to play a homey little western style, 380-seat concert hall! *WHY???*

"I think that they continue to come here because of the way we treat them," Yockey said. "We treat our artists *very* well. Everyone who works here is a professional and really knows their job. I think that the artists just feel relaxed and comfortable here, and they really like that, so they keep coming back!" he said.

It could be that his closing comment was the highlight of the day.

All the glamour, equipment and recording stars aside, the amateur just breaking in, and the seasoned pro alike can take Yockey's words to heart and put them to use in our own work.

Artists just seem to like going where they can feel relaxed and comfortable!

Thanks Scott, we'll try to remember!!

Equipment List for the Coach House

THE MAIN MIX

The main mixing console is a Soundcraft Model 500 with 40 inputs and 6 aux buses, and contains a four-band semi-parametric EQ. The outboard rack is pretty standard fare. Dynamics processors include three dBx 160 compressor/limiters, one dBx 166 compressor/limiter, one Gatex quad gate, and two Drawmer DS-201 dual gates. Effects include a Roland SDE-1000 digital delay, a Lexicon PCM-60 digital reverb, and two Yamaha digital reverbs, the venerable SPX-90 and REV-7. The stereo mix is then fed through a BBE 802 Sonic Maximizer, a Klark-Technik 30 band graphic EQ and finally to a Brooke Siren crossover for splitting into four stereo bands for the quad-amped system. The power amps total 6,000 watts and include four QSC 3800's and one QSC 3500, along with a Crown DC 300A.

THE MONITOR MIX

The monitor mix is done on two Yamaha 1608 mixers operating in tandem and provide eight separate stage mixes. Each mix is sent to a bank of Crown and Electro Voice power amps which provide 300 watts per send. The onstage monitors include ten Woodworks floor monitors with one 12-inch speaker and a one-inch horn each, and two Woodworks floor monitors with dual 12s and one 2-inch horn each. The two drum/side fill monitors contain one 15-inch plus a one-inch horn each.

THE HOUSE SPEAKERS

The house speakers are a custom design by Yockey himself and are comprised of high-quality components. Each side is serviced by an 18-inch subwoofer mounted underneath the left and right sides of the stage (sub-lows). Above the stage hang four identical cabinets, two positioned immediately stage right and stage left, with another pair further off to each side. Each cabinet contains the following components: two 15-inch JBL 2240s (lows); two 12-inch EV Proline 12-Ls (mids); one 2-inch JBL 2482 horn and one JBL 2402 bullet tweeter (highs).

THE MIC LOCKER

The microphone selection at the Coach House is varied enough to meet their artist's needs without being extravagent and Scott reports that many artists bring their own mics for certain purposes, in particular for vocals. Their selection includes nine Shure SM-57's, eight SM-58's, five Ramsa S-5 clip-on mics, four AKG C-451's (a favorite of mine), two Sennheiser MD-421's, and one EV PL-20. db

JOHN BARILLA

North Forty Music— Truth in Advertising

The world of advertising appears to be full of hype. Thc tendency of inflationary rhetoric usually extends not only to the ad agency, but to the musical production houses as well.

An interview with a commercial producer may dwells on a litany of instantly recognizable clients with whom the production company seeks to be associated, and the realities of this tough business, the philosophies and frustrations, the whole inner dialogue lying just beneath the surface, are usually never broached.

Hence, it is a rare moment when one can transcend the high-profile client list or glitzy studio, and get a glimpse at the unseen creative currents. This, to me, is the real "truth in advertising," and it's exactly what was captured in this insightful interview with Ric Kallaher of North Forty Music.

Located on New York City's east side, North Forty is not a large company. With a staff of five composer/producers (including Kallaher) and two useful (but decidedly non-glitzy) recording studios, North Forty's espoused aim is to concentrate on the creative process and avoid getting side-tracked on fancy items.

Their creative team is evidence of a wide diversity of musical backgrounds. While Kallaher makes no claims to being a musician (his background being in film and theater), he often comes up with rough musical ideas which are later hammered into a finished product by the staff. His function is one of crafting a concept, much as an executive producer would; overseeing the entire creative process and getting involved with clients.

Figure 1. Ric Kallaher in his production studio.

TWO PRINCIPALS

Kallaher, apart from being North Forty's mouthpiece, is one of the two principals of the company. Jim Todd, the other principal, has a long history of actualizing Kallaher's musical concepts.

While North Forty has only been in existence for five years, Kallaher and Todd have had a working relationship for over a decade. In their early days, Kallaher and Todd were employed by major advertising agencies and seeking a creative-entrepreneurial outlet.

"I got really bored being on the agency side," Kallaher said. "It got to be more administrative than anything else and I have sort of a creative slant." In time, he and Todd joined talents to form North Forty (a company name which has absolutely no significance except for its catchiness).

Todd, a trained musical composer, was later joined by Allen ("Woody") Smallwood, a musician/arranger who had previously been the keyboard mainstay for several rock 'n roll luminaries. Smallwood also brought with him some solid technical skills in recording engineering

Figure 2. Allen "Woody" Smallwood also is posed in the production studio.

which helped North Forty improve their "in-house" sound. Associate Producer Larry Geismar added his musical training and conceptual ideas to the creative stew, and rounding out the talent-base was Composer Paul Gottlieb. Typical of the new generation who grew up writing on electronic keyboards and sequencers, Gottlieb brings a youthful, hip approach to the North Forty sound. Despite their musical individualism, each North Forty member submits their compositional ideas to each other for criticism and fine tuning.

"We will sometimes write entirely separately on a project," Kallaher said. "But even in that framework, we eventually bring the work together and try to shift the focus of one piece to give the clients more of a choice—that is, not just one songwriter versus another, but one directed, focused concept (as a solution) to their problem, which is, how do we sell their product," he said.

SUBMIT MORE THAN ONE

North Forty will often submit up to five different song demos to an ad agency; ostensibly, one "seed" idea from each of the staff mutually honed to meet the client's needs. The ad agency will sometimes have a tightly structured format in mind, including a predetermined slogan and timed copy. Even so, North Forty always goes somewhat "beyond the call of duty" by offering the client a few additional creative perspectives—even to the extent of inventing an alternative slogan. The reason for North Forty's "go-the-extra-mile" attitude is two-fold:

First, it gives the client a more complete view of all the creative possibilities for a particular campaign. The second reason is pure pragmatism. If, for example, a competitor submits only one or two demo ideas and North Forty submits five, (provided all are of equal creative value), it is likely North Forty will get the account.

The formula apparently works well for North Forty. Their video soundtrack reels indicate their subjects are as prestigious as they come—NBC News, HBO, Kodak, Mercedes-Benz and so on, yet despite their successes, the nature of the business never allows them to feel safely beyond the competition. The reality is, North Forty never ceases to be battling against other production companies for choice clients. Very often, the company finds itself investing large amounts of time and creative energy competing for an account with precious little consideration out front.

"You have to be able to succeed, if you succeed," Kallaher says. "It's like acting. There are no middle-class actors. You're either doing really well—meaning a millionaire superstar—or you're a starving artist. It's the same way with what we do. If you really succeed, you're going to get a lot of work and make a lot of money; and if you don't succeed, you're not going to be able to stay in business—especially now," he said. "That's the unfortunate side for generations to come."

The "brain" of the room is an Atari Mega 2 computer running Notator 3.3.

Kallaher added that part of the present ad game seems to lean toward acquiring very expensive technology to remain competitive.

"It's so expensive now with the technology that some of these (ad agency) people are into—without really understanding that it's not always necessary," Kallaher said. "We have competitors who invest incredible amounts of money—like Synclavs up the wazoo. (But) if we need a Synclav for a job, we can book one," he said. We know what it does, but it's not right for every job. (In our studio) we have all different kinds of keyboard modules, and we really try to mix it up, because each one of the manufacturers has a great axe. Each one has a little bit of a different type of sound—a different specialty, and it's great to have a lot of those little things to play with," said Kallaher. "If you know what's out there and how you can use it and when it's right to use it, that's the best way.

"If you have this overhead (Synclaviers and other upscale digital systems) that you have got to justify somehow, I think it cripples you creatively, because you have to use it; you've got to pay for it," Kallaher said. "How do you get away from that? I always want our emphasis to be creative, not top-of-the-line studio 'bells and whistles'."

MIDI ROOMS

In almost constant use at the North Forty studios are two MIDI production rooms, the larger of which houses an Otari MX-80 24-track and a Trident Series 65 console (modified to a 24 X 24 configuration). The "brain" of the room is an Atari Mega 2 computer running Notator 3.3. The Notator program has received many plaudits for its usefulness and real-time music notation. Scoring to picture involves locking a 3/4-in. video tape with SMPTE time code to the Atari. When tracks are finished, they are sometimes printed to the 24-track and taken out-of-house to a larger studio for extensive overdubs. Layback to video-tape is frequently done out-of-house as well, where more extensive synchronization gear is available.

STUDIO DESCRIPTION

The main studio contains a good selection of sound modules with analog, digital and L/A synthesis represented, as well as sampling and sample playback units. The keyboards are all run through a Tascam MM-1 20- channel mixer before the main console. This unit has programmable MIDI muting, allowing the user to shut off channels when they are not playing—which helps keep a clean mix. The smaller studio is really a cubbyhole-wedge of a room doubling as a vocal booth for the larger studio. It houses a smaller rack of synthesizers and a computer, and is used more for off-line production.

Because of the pressures of competition, Kallaher notes that it has become important for each musical production company to have their own studio. If live musicians are required, it pays just to "test" the charts on synthesized instruments prior to booking time in a larger studio. The balance point is how extensive (and expensive) a studio is needed to turn a profit?

While owning a studio can make certain projects accessible that would otherwise be cost-prohibitive, going too far into hock can certainly put an entire business at risk. This stampede to have the "ultimate" studio, rather than simply an "appropriate" studio, is fueled, according to Kallaher, by the rather superficial trendiness of the advertising industry. The tendency to want to use a gadget simply because it's there is ironic; these gadgets (samplers, sequencers, etc.) don't end up saving any time at all; they simply drive the expectation for novelty even higher—often in very arbitrary, anti-creative ways.

"The bad part is that time is still at a premium," Kallaher says. You just don't push a button and get a new approach. It has to be programmed, and that is not magic," he said.

"We all participate and bounce ideas off one another and are able to bring a little bit of an extra ingredient to somebody else's basic concept,"

"The big catch-word these days is 'sound design', which is a very nebulous term, and I think it is incredibly misused," Kallaher said. "I think if you were to believe half the stuff that you read that talks about sound design, you would think all these people invented such a thing as sound design in the last couple of years. As if John Cage was born yesterday and nobody ever did sound effects for radio and theater 50 years ago. I think somebody decided it was a good way to describe the kind of things you can do with computers and sampling," he said

LIVE VS SAMPLED

Kallaher also addressed that, for example, a sampled string section does not replace a live one, nor does a drum machine replace a live drummer. The nuance in the synthetic version is simply not there, and the effort it would take to program it in—even with sophisticated vehicles like the Synclav—are cost-prohibitive. Looking at the contemporary obsession with novelty, Kallaher commented that the obsession is a reflection of how foolish seemingly sophisticated people are.

"We did this thing for Mercedes—for the new SL sports car, and we used what was the equivalent of a 70-piece orchestra," Kallaher said. "We had 40 pieces in the studio and then double-tracked (at The Power Station Studios in New York City) to add extra texture and parts. And we actually had someone ask us, 'Why would anyone want to spend all that money with all those players when you could do it with keyboards'. Yeah, it would have sounded fine," he said, "but this was a true orchestral piece, you know, and if you're going to have an accelerando, you really want to like, dig in to one part—you know, have it human—you can't do that.

DO COMPUTERS MAKE MUSIC?

"You spend the same amount of time, same amount of money trying to get your Synclav to do it. And why? You know, there's an expression: 'God is in the cracks—in the small places', and computers are made to have 0's and 1's," Kallaher said. "There's something missing a lot of the time somewhere in between.

"Listen, there's a reason why we all still love those songs from the 50s and 60s. A lot of it, yeah, it sounds crude, but you listen to the stuff today and it all sounds the same," he said. "It has a real harsh edge to it because everybody's using this great-new-drum-sample-and-great-new-bassline (he measures his words here in a machine-like fashion, mocking the uniformity of a digital sampler). Everybody uses it. It's two guys in a studio producing a song as opposed to five or six," he said.

"There's all these missing elements. That's why we like working the way we do, because there's five of us," Kallaher said. "We all participate and bounce ideas off one another and are able to bring a little bit of an extra ingredient to somebody else's basic concept," he said. "There are times when we're all working together on one piece—and that's a lot of fun."

North Forty's notable success has undoubtedly been a function of their ability to work together, synergistically, as a team, without the egotism and internal competitiveness that often wreaks havoc in the advertising world. Likewise, their ability to make prudent business decisions—like outfitting their studios with useful and affordable equipment—helps them stay profitable. Measured by Kallaher's own statements, it would seem that his production company is indeed, an unqualified success. db

RANDY HOFFNER

Broadcast Audio

HDTV Video Tape Recorder: A State-of-the-Art Audio Recorder as Well

• Get ready world: digital high definition television production is coming! As we know, HDTV requires production equipment capable of far greater video bandwidth than the NTSC equipment we are familiar with. Factor in the digital domain, and the recorded bandwidth requirements become truly astronomical when compared to 4.5 MHz video and 20 kHz audio.

The digital video section of the recorder uses a sample frequency of 74.25 MHz and 8-bit quantization to realize a published video signal-to-noise ratio of 56 dB.

Sony has manufactured the first digital HDTV recorder to hit the marketplace, and it has digital audio as well as digital video. This machine is called the HDD-1000, and with its companion processor, the HDDP-1000, you too can produce digital HDTV tapes, provided you can afford the freight. To give you an idea of the price neighborhood we are in, I'll just say that a one-hour reel of tape for the HDD-1000 costs over $1,000. Obviously at this time, digital HDTV is not for the faint-hearted!

I thought it would be interesting to describe this machine. First, let me throw in a little disclaimer; *This is not a product review!* It is a discussion of a piece of equipment that is on the cutting edge of television recording technology.

The HDD-1000 transport is an evolutionary descendent of the venerable BVH series of one-inch machines. In fact, except for the color, it looks, at first glance, quite similar to the BVH-3000. The high-density digital recording requirements placed on it, however, have resulted in a souped-up version; we might call it the "turbo" version of the transport. It is indeed a one-inch machine, but instead of conventional ferric oxide tape, it uses metal tape. One-inch type C tape travels at slightly over nine inches per second, but this speedster boasts a tape speed of about 31.7 inches per second. The head-to-tape writing speed is about 169 feet per second. In case your calculator isn't handy, that is about 115 miles per hour! You will probably not find it difficult to believe that this machine definitely makes noise when operating.

Figure 1 is a drawing showing the location of the heads on the scanner drum. It may be seen that there are 18 heads on the drum. *Figure 2* is a drawing of the recording's footprint on the tape. The astute observer will note the digital audio is not written by the helical scanner, but is rather recorded on longitudinal tracks in the Digital Audio Stationary Head

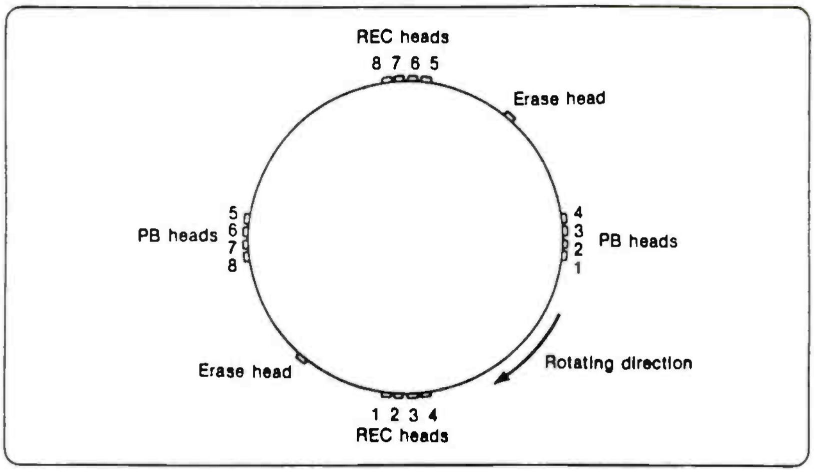
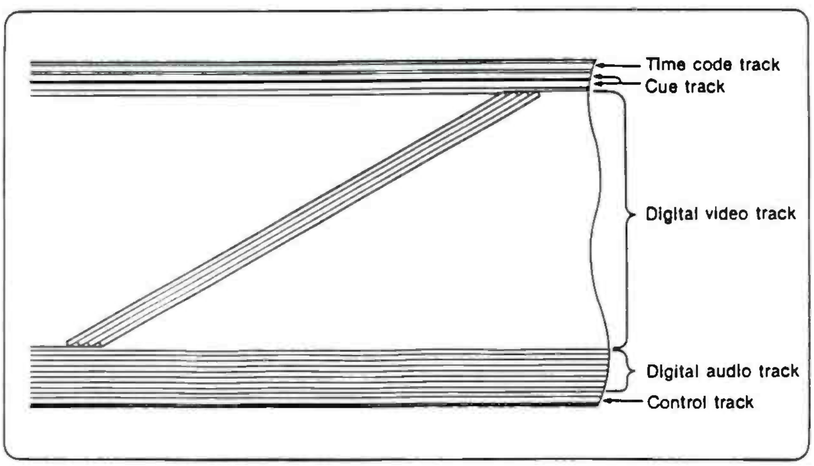


Figure 1. (Above) Video head placement on the scanner.

Figure 2. (Below) The recorded footprint that is recorded on the tape



(DASH) format. In addition to the helical video track, there is a longitudinal time code track, two longitudinal cue or analog audio tracks, the digital audio tracks and a control track.

The digital video section of the recorder uses a sample frequency of 74.25 MHz and 8-bit quantization to realize a published video signal-to-noise ratio of 56 dB. It has the 30 MHz luminance bandwidth required for the HDTV video signal.

The audio section has eight channels of digital audio, plus the two analog audio cue tracks. The audio is sampled at 48 kHz with 16 bit quantization, and it is synchronous with video. Its format conforms to the AES/EBU digital audio standard, and the machine has both analog audio inputs and outputs and AES/EBU serial digital input and output interfaces.

This machine differs from its conventional digital NTSC predecessors, D-1 and D-2, in that the NTSC machines record digital audio as well as video in the helical tracks. The use of stationary head digital audio recording may have been necessitated by the large number of tracks, twice as many as D-1 and D-2 have.

It is interesting to note the audio level indicators, which are unique in their appearance. They are LED segment strings with labels marked "infinity", -42, -32, -24, -20, -16, -8, and "over." "Over" means the digital Full-Scale point (0 dB FS, in the new AES terminology) has been exceeded, and there are no more bits left. When this point is reached, no greater analog amplitude may be digitally represented when that threshold is exceeded and hard clipping results. There is one unmarked segment between each marked one. The greatest resolution, 2 dB, is on either side of the -20 dB point. The fact that the scale's greatest resolution falls immediately above and below the -20 mark indicates the intended operating point is 20 dB below Full-Scale. There are some who would opine that this is operating too conservatively, and greater advantage of the medium's potential signal-to-noise ratio may be taken by operating at a level higher than -20 dB FS.

The audio section of this digital HDTV recorder is certainly several steps beyond the audio section of any previous video recorder. In fact, this is actually a digital HDTV video tape recorder with a built-in 8-track digital audio recorder...

There is general belief that when using a peak program meter with a 10 millisecond integration time, there are unregistered peaks up to 15 dB above setup level, that level of sine wave tone that causes the EBU ppm to indicate at its midpoint or "test" position. This would put those peaks about 6 dB above the "permitted maximum level" point, that level either 8 or 9 dB above "test", where program operating level should reside. If this is accepted, the daring operator would operate this machine so the indicators reached the -8 point regularly, but seldom went above that point, and in any event, the "over" indicator should never light. The instructions state that the normal mode of operation is as a "normal peak meter," but the actual ballistics are not specified. It is also not stated whether these meters truly indicate the digital level, or are actually indicators of analog levels. The meters have a calibration mode in which precise calibration within 0.1 dB is possible.

The audio section of this digital HDTV recorder is certainly several steps beyond the audio section of any previous video recorder. In fact, this is actually a digital HDTV video tape recorder with a built-in 8-track digital audio recorder. It is an affirmation of the statement that sophisticated, high quality audio is a necessary adjunct to high definition video. db

The New Wave of CD Referencing

BOB LUDWIG

• Major improvements in existing and new CD optical disc recording systems have recently been announced.

With the unfortunate demise of the vinyl disc in the record industry, the compact disc has become the only ultra-high quality medium left for the dissemination of music. Mastering facilities have always wanted to offer CD references to artists and producers to take the place of the vinyl reference, but until recently, this was not possible. Presently, DAT tapes have been supplied as a pseudo-reference for checking the CD master that will be used for CD production. I refer to the use of DAT as a pseudo-reference because, while the actual digital data stored on each medium is identical on playback through the same analog to digital converter, this is where the similarity ends.

A true CD reference means being able to check the PQ coding, which is often an artistic decision. Additionally, hearing the final product on various consumer CD players, from fairly brittle and dry-sounding portables to state-of-the-art systems that use Wadia or Krell D/A playback computers costing many thousands of dollars, is important to true CD reference.

The manufacturing of mass produced CDs is an expensive and difficult process. The actual recording of the CD, the glass mastering, takes place in Class 100 clean rooms which are much more clean than a surgical operating room. The undertaking of a CD plant is a multi-multi-million dollar venture. Mass production is essential to making CDs cost effective to the consumer. Making single copies of a CD using this system has been simply prohibitive.

Bob Ludwig cuts his master discs at New York City's Masterdisk Corporation.

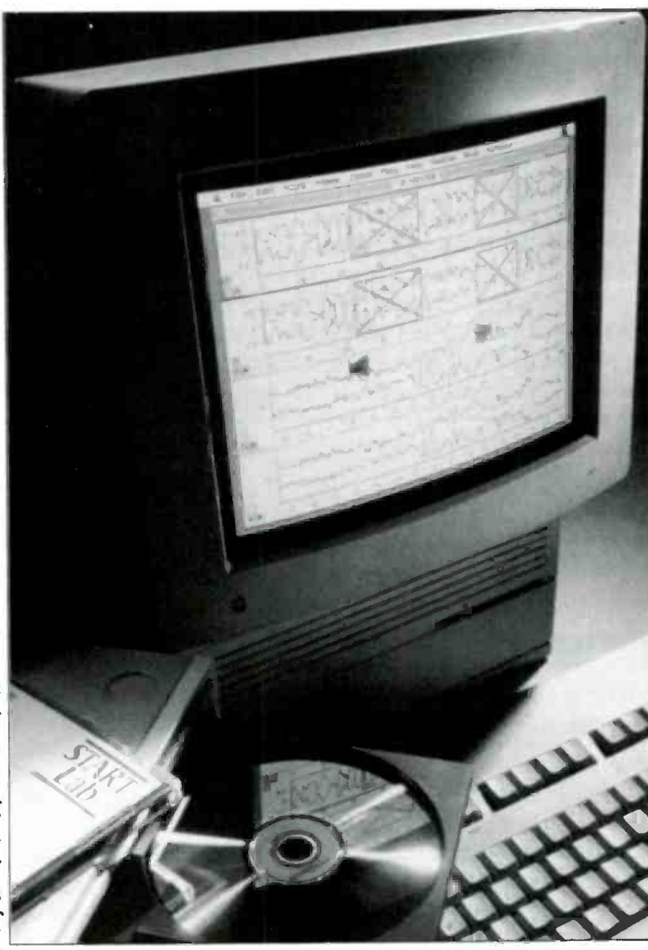


Figure 1. The START Lab's CD Maker and Sonic Solutions' Digital Editing System.

For over six months CD references have been a reality. Yamaha has offered the CRD-90 and now announces the improved YPR-201 CD recording unit as part of that system. It was now possible to make a true CD reference that could be identical in every way to the final CD pressing. The first version of this unit used optical blanks that gave 58 minutes of CD audio (they can also be used for making CD-ROM). Recording time was soon increased to 61 minutes. This system is offered by Gotham Audio with software and computer interface for the Yamaha Recorder provided by Harmonia Mundi Acustica. The aluminum-coated optical blanks are made by Fuji. These CDs were compatible with about 90 percent of the CD players on the market.

During this time, SONY and Taiyo Yuden offered a CD recording service in Tokyo to make one-off CDs, but their system was not for sale. To announce the availability of their system, Masterdisk Corp. of New York City recently hosted a press party given by Sonic Solutions to demonstrate their optical recording system, the Recordable CD Maker.

Available in May 1990, The Recordable CD Maker involved the special cooperation of Sonic Solutions of San Francisco, CA and **START** Lab Inc. of Tokyo. START Lab is a joint venture of **Taiyo Yuden** who invented and manufactured the latest CD-R63/CD-R74 "That's"-Compact Disc Recordable and **SONY** Corporation, Japan, who make the CDW-E1 Compact Disc Encoding Unit and the CDW-W1 Compact Disc Recording Unit. The-'"That's"-Compact Disc Recordable' is also a write-once optical recording disc for producing audio CDs or computer CD-ROM discs.

Using this SONY/START Lab system on projects prepared on the Sonic System can offer several operating advantages in creating the PQ code. The PQ code tells the CD player where each track starts and ends, in addition to other functions.

Each CD master requires frame accurate PQ codes. Because the material is prepared on the Sonic Solutions hard disk editing system, the computer can automatically detect start and end times of each selection if there is "digital black" between the selections as is true for many CD masters. The making of the PQ sheet, which is fairly laborious to do, becomes almost instantaneous!

In the near future, it will be possible to record double time ("cutting" an hour-long CD in a half hour) because the source is on SCSI disk drives instead of a DMR-4000 video-based machine. This will help bring the cost down to the point where it will be about as cheap to make a CD reference as it is to make a DAT copy at present.

Unlike the Yamaha system, the Sonic System requires loading the CD audio information on to the hard disk and then via SCSI to the optical format. Sonic promised a future system which would allow direct recording from the original master tape without going through the hard disk loading and unloading.

Presently, each CD maker includes one encoder and one or more recorders—up to 32 recorder units can be added to allow simultaneous recording of up to 32 discs.

Fuji has announced the YOD-201, a new optical blank, which also meets the CD Red Book standards as does the Taiyo Yuden disc. The new discs from Taiyo Yuden or the new Fuji gold optical discs have played perfectly on every player that has been field tested with them to date. In fact, owners of the Yamaha optical recorders are presently being persuaded to record on the Fuji gold and Taiyo Yuden discs. The competition should bring the high cost of these blanks down.

The price of the blank is a significant factor in determining the final price of the CD reference because, as with vinyl discs, several discs can sometimes be wasted in the process of creating a perfect reference. Also, with the earlier versions of the write once system, the recording time was limited, often necessitating the use of two blanks to represent a single consumer CD (i.e.: if the playing time was between 62 and 74 minutes, two write-once blanks would be required to contain the program material). This will no longer be necessary for Red Book standard CDs. All manufacturers of the optical blanks make them "pre-tracked" and are made with an organic dye which is chemically changed with the recording laser. This creates the CD pits dictated by the 8-to-14 modulation scheme. This recording laser is **much** more powerful than the laser in your CD player. Unlike normal CDs, these organic dye CD discs previously had a shelf life of about five years; this has now been increased to over 10 years.

It was announced the CD reference discs could be used as the CD master, replacing the video-based machines presently used. Because the block error rate of an optical CD can be very low compared to any tape storage system, it is desirable to have the CD optical disc *become* the CD master. With an extended table of contents, it would be possible to mail the optical disc to the CD plant to use as the master for creating the actual glass master with a high degree of reliability.

Finally, it was announced that in the future one could do assemble edits on the CD optical discs which would make it a very reliable and fast access recording medium. db

BRIAN BATTLES

Ad Ventures

Telephone BBSing

• Using a personal computer to exchange message and data files via bulletin board systems (BBSs) is an exciting way to keep up with the latest in news and software. Even if you don't (yet) use a modem with your personal computer, you can enjoy many benefits by using it to access an enormous number of landline bulletin board systems (BBSs) that carry recording, music, broadcasting and electronics-related message bases.

In these conferences you can ask and answer questions, present your opinions or observations, debate FCC rules, share technical hints, locate parts, list or find used equipment for sale or swap and talk about MIDI software. Occasionally some good-natured bickering breaks out when participants get fired up, but the crowd is usually well-behaved and offers a wide variety of knowledge and tips. You may even run into some celebrities on-line, folks who are closely associated with well-known studios or radio/TV stations, answering technical questions or holding forth on some philosophical point.

Some local BBSs also provide access to UseNet, an international system that mostly serves large corporations and universities through their mainframe computers. As a small PC user, though, you might be able to get into a local private BBS that carries some of the UseNet conferences ("newsgroups").

IT'S REALLY SIMPLE

In case you've never attempted data communication, it's simple to understand. Your computer may well be equipped with serial port that allows data to flow into and out of your PC when it's configured to act as a simple terminal. Until it's connected to some external device, however, there is no data to flow in or out. A modem (short for "modulator/demodulator") is a simple device that takes digital signals from a PC or terminal and converts them to audio (and vice versa). In audio form, they can then be transmitted back and forth over telephone lines.

You simply hook up a cable between your computer and modem, plug the modem into a nearby telephone jack and boot some kind of communication program.

Some inexpensive software programs to use with your modem are Telix, ProComm, QModem and Boyan (These commercial titles are in a category called *shareware*—unlike retail-store software packages, these programs may be legally copied from other computer hobbyists' collections. You can use them for a limited time for free and if you like a program enough to use it permanently, you pay a nominal registration fee to the author/publisher of the package).

Once you've connected a modem to your personal computer and learned a telecommunications program, it's time to log on to a bulletin board system. Just about any computer hobbyist who owns a modem can help you find a local BBS system. You can get started by asking around. Check with other local recordists or MIDI users, or talk to any friends who use personal computers with modems. Ask the folks at the office or in school. If you can't find an answer from someone nearby, browse through a copy of *Computer Shopper* (Coastal Associates Publishing, One Park Ave., New York, NY 10016), which lists User Groups in a special directory section. A call to one of the local groups will probably yield a couple of numbers to try. You can ask me a question by dialing into the Board at my workplace by calling the ARRL HQ BBS at 203-665-0090 (1200 or 2400 bauds, 8 bits, 1 stop bit, no parity).

Once you're logged onto your first BBS, things get easier. You'll quickly grasp how to maneuver your way through a succession of menus or screen prompts to get to what you're interested in. If you ever get "lost," most BBS software offers the option of requesting a help screen, paging the system operator (he or she can "chat" with you live via computer if he or she happens to be available when you page), downloading an instructional file to your computer's disk for later study, or leaving a private message for the SysOp (you can read his/her reply the next time you log on).

Most BBSs require that you configure your terminal for 1200, 2400, or 9600 bauds, 8 bits, 1 stop bit, no parity (see your telecommunication software documentation). Some provide flashy color graphics screens that can be viewed on personal computers that support color displays and ANSI screen drivers. You may find that several BBSs are local to you, in which case you can choose whatever one you prefer as your "home" BBS. You aren't rigidly restricted to using only one particular BBS, but it makes it easier for you to send and receive mail if you stay put. You might wish to have a "backup" BBS that you check into occasionally, just in case your home BBS has a long-term hardware failure or "goes out of business."

It's easy to find other BBSs in your local calling area because most BBSs provide a file someplace that lists other area BBSs users may wish to access. Nearly all that carry FidoNet Echoes maintain a listing of other FidoNet BBSs, complete with telephone numbers, so you might be able to find a half dozen within your local calling area.

Turning your personal computer into a communication terminal and connecting it to an on-line bulletin board will bring you right into the 1990s.

Stop wondering what to do with that telephone modem. You can spend some money on commercial on-line services, or get on a freebie hobbyist's board and join the fun on the recording, music and broadcasting network conferences.

If you're not careful, you may become a BBS addict like me!

Fire Up Your Modem

Dialup data communications services have been with us for several years, and commercial operations such as CompuServe, GEnie, Delphi, Prodigy and Quantum Link offer a variety of on-line services and conferences for hundreds of areas of interest.

These commercial services cost money, though, and many owners of personal computers and modems shy away from them because potentially hefty monthly bills accumulate rapidly if you spend much time on-line. But did you know there is a cheaper alternative?

All across the country a revolution has been going on for quite some time, as owners of well-equipped computers have created a multitude of private local BBSs. These range from simple, single-line, small-scale setups to massive multi-line systems offering megabytes of file and message storage. Most offer access to users absolutely free. Some private BBSs do charge a nominal fee for membership to help offset the cost of telephone bills, hardware, etc., although these fees are generally quite small.

So who runs these things? Is this some kind of mysterious, subversive, "underground" faction of phone hackers and techno-anarchists? Hardly! BBS system operators (SysOps) are mainly computer hobbyists who set up boards to share their enjoyment of computers with others. SysOps are therefore a special breed of often under-appreciated volunteers, who devote large amounts of time, effort and money to maintain complex BBSs and "ride herd" on hundreds of users. Many SysOps are college students, computer programmers, consultants, engineers and businesspeople who view operating a BBS much the same way you view recording and music.

There are probably at least two or three good BBSs in your local telephone calling area that may contain an interesting blend of special interest forums, public domain and shareware software programs, as well as games, live multiuser "chat" sessions and more. A few BBSs run by radio amateurs act as gateways between the telephone lines and packet radio. Beyond the local activity, though, you may also be fortunate enough to locate a nearby BBS that is a member of a wide-area network, such as FidoNet, InfoNet, or UseNet.

FidoNet is an international BBS system that links boards from all over North America and many other countries. Individual FidoNet BBSs take messages posted by local users and, through an extensive relay chain of local telephone calls, forward the "mail" to thousands of other BBSs all over. A message you type in on a BBS in Skowhegan, ME, will appear on BBSs in all 50 states, Canada, and even as far away as Australia, normally within just a couple of days. Similarly, replies to your comments can arrive from a multitude of originating BBSs, allowing you to indirectly "converse" with people over a widespread area...all at minimal or no cost to you!

The FidoNet structure has a system-wide "backbone" that carries messages in a variety of individual conferences that are called "Echoes" internationally, and individual SysOps may select the Echoes they wish to carry on their BBSs a la carte. A few of the popular Echoes include Telecommunications, Adoptees, MIDI, Substance Abuse, New SysOps, Jewish Issues, Musician's Services, Firearms, Politics, Media, Shareware, Men's Issues, C Programmers, Help Wanted, Graphics, dBase Users, Aviation, Entrepreneurs, Home Office and IBM PS/2.

My favorite Echoes are Broadcasting, Media, Home Office, Entrepreneur, Ham Radio, Packet and Ham4Sale. Look through your local BBS and see what conferences you wish to participate in...then, jump right in and join the fun!

Ed. note: **db** Contributing Editor Brian Battles is the Moderator of the FidoNet Ham Radio and Packet Echoes. Send him E-mail via FidoNet at node 1:142/670 or leave a message on the ARRL HQ BBS. Send him Amateur Radio packet mail to WA1YUA @N1API.

db

JOHN BARILLA

THE ART OF EQUALIZATION: Part 3

• Having touched on the "tools" and "theory" of equalization in Parts 1 and 2 of this series, we are now ready to discuss some of the "tricks" of the trade. Perhaps tricks is too strong a word; what will follow amounts to the basic techniques accomplished recording engineers use all the time. If it's a technique you are not aware of, it might seem like a trick! There is, undeniably, a sort of magic present in a great drum sound, a tight, punchy bass line or a smooth, warm vocal. But even if you're just plugging in some hot, new drum sample, I'd wager that somebody, somewhere had to do some radical EQ to get it to sound so good. Equalization is undoubtedly a fundamental practice that must be mastered, for no amount of reverb, compression or delay can ever compensate for a deficient or harsh EQ. Let's now examine some of the many techniques used in the art of equalization.

FINDING THE SWEET SPOT/SORE SPOT

Just as in tennis or baseball there is a "sweet spot" on the racket or bat which guarantees a good transference of power to the ball, so is there a sweet spot in equalization (There may even be more than one sweet spot, but finding one is usually the key to finding others). Effective equalization involves finding a sweet or sore spot and manipulating it to good advantage as would a massage therapist scan the back to find a sore spot and then press on it, aggravating it until releasing into a relaxed muscle.

By my definition, a sweet or sore spot is simply a frequency band of resonance—a naturally occurring peak in the sound—which can benefit from being boosted or cut. As we discussed in Part II of this series, each sound has a series of peaks caused either by harmonics (natural overtones) or formants (steady-state frequencies related to the structure of the instrument) which contain more energy than the surrounding frequencies. Once found, a little EQ will go a long way to change the shape of the sound (Here we should note a recognized axiom: "less is more." The minimal amount of EQ to do the job is usually the best, because excessive EQ is difficult to undo, and it adds a certain phase-related "cloudiness" to the sound. Each dB of boost or cut is much more effective if applied at an area of resonance).

HOW DO YOU FIND AN AREA OF RESONANCE?

It's simple. Turn up the gain all the way (on the section of the equalizer you are using), and then sweep the frequency control until you hit a resonant band (If you are using a parametric equalizer, you can increase your accuracy in pinpointing the resonance by adjusting the "Q" control for a very narrow bandwidth). If there is a sore spot to be found, this is how to find it. Stop sweeping the frequency control when your ears say 'ouch'! Once you've found the sore spot, it's just a matter of deciding how much of it you want to get rid of. Return the gain to zero, then start rolling off the gain until the harshness disappears.

Sweet spots are found by the same method, except your subjective reaction won't be the same; instead of "ouch," you'll be saying, "I found the power!" Just as salt is to food, the power frequencies you tune into can bring to life an otherwise dull sound. Nonetheless, like salt, a minimalistic approach is best. So, once again, having found the resonant frequency, return the gain to zero, then gradually boost it until the appropriate level is reached.

NEGATIVE EQ

It's interesting to look at the EQ curve of a mix done by a novice sound recordist; if there are three bands of EQ per channel, at least two, or maybe all three bands will be boosted. I'm not saying this is an intentional policy, but that's the way things frequently end up at the end of a session. It's incomprehensible that every track would need boosting in the highs, mids and lows, but once a pattern is established (of boosting everything that sounds good to you on a given track), you are almost locked into treating the other tracks in a similar way. What happens is that the first EQ'd track (with everything boosted) sounds brilliant and everything else sounds dull by comparison. The tendency is to perk up the remaining tracks in a similar fashion. But having done that, the first track is no longer sitting where you want it to sit in the mix, so boost those frequencies a bit more. Eventually, a war ensues between the channels. What was intended to be a mix, instead turns out to be an uneasy truce between the various tracks; every instrument screams at the same time, competing rather than cooperating, and each striving to be heard above the din of the crowd.

If you find yourself in this situation, you need to normalize all your EQ channels and try achieving your desired effect by negative equalization, which is achieving an equivalent frequency contour by rolling-off, rather than by boosting.

For example, suppose you find yourself radically boosting both highs and lows on a given sound. Imagine the graphical representation of your EQ. It would undoubtedly look like a somewhat flattened letter U. Boosting the highs and lows is tantamount to scooping out the midrange. If you can achieve the same curve by midrange cut, then you are using negative EQ.

Why bother thinking with such inverse logic? Well, there are some distinct sonic advantages. First, any time you can cut, rather than boost, you are automatically reducing noise as a beneficial side-effect. Second, if you can achieve the same effect as two bands of boost with one band of cut, you are retaining more of the original phase-coherency of the signal, making it sound more clear and clean, and easier to place in the context of the mix. While negative EQ is not applicable to all situations, it can be used frequently—if you remain aware of this valuable concept.

DEALING WITH THE MIDRANGE

No frequency range affects the perceived power of a sound like the area (roughly) between 500 and 5,000 Hz. Several factors coalesce here, making this midrange band a determinant factor. For one, much of the fundamental (1st harmonic), the 2nd harmonic (one octave above the fundamental) and all important formants are found in this range. Hence, it is obviously an area replete with resonance—fertile with sweet and sore spots. Coupled with this physical property is, of course, the subjective, for the ear is maximally sensitive to midrange frequencies. Since these factors synergistically work together, it is apparent that midrange equalization can be the critical factor for placing a sound in the context of a mix. Midrange EQ can actually make a sound proceed or recede relative to the other tracks. In other words, midrange EQ figures heavily in whether a track is perceived as "right in your face" or at somewhat of a distance from the listener. Hence, midrange frequencies are sometimes called "presence" frequencies because they do much to determine the attention a track will receive.

As one might suppose when using such a power tool as midrange equalization, it's very easy to miss the optimum setting. Getting it right takes a bit of experience and experimentation, but bearing in mind the follow characteristics of the midrange spectrum can be helpful. For most midrange instruments (especially voice), a boost in the area of 500-1,000 Hz will accentuate the low formant as well as fundamental and low harmonics. This is a powerful region for effecting the apparent size of the instrument. For example, if you boost a male vocal at 500 Hz, the vocalist will sound like he has a bigger head. Roll-off this frequency and he will become rather diminutive. The fact is, most sounds have a large energy component in this area, so equalizing here can seriously effect the sound's magnitude. While an excess might make the track sound a bit "boxy," fortunately, you can't really get anything harsh sounding under 1 k.

But the complexion radically changes less than an octave higher. Starting about 1.5 k and to 2.5 k, we encounter a midrange band which is also quite powerful, but less forgiving in its administration; boosting in excess will impart a distinctly "telephony" sound. It's actually an interesting and often used effect, but if you push it too much, it might become annoying. The next midrange area is perhaps the most useful. If you want to make something sit a little hotter in the mix without pushing up the gain, the area from about 3 k to 5 k is just the ticket. This region works psychoacoustically on the listener, tickling him/her in that area of maximum sensitivity. Subtle amounts of cut or boost in this region can really help find niches for the tracks in question, so that one recedes a bit, while another is made to move forward (This is the "presence" band spoken about previously).

WHEN TO EQ WHILE TRACKING

In the early days of multi-track recording, it was assumed that one should record everything flat (no EQ), and then, if absolutely necessary, apply equalization during the mixdown. That was the way it was done, end of story. During the 1970s, a great trend in experimentation hit the recording industry (and the rest of society, as well). The bottom line was to come up with a novel sound, and plenty of EQ—upon tracking and again upon mixing—became the rule. While this experimentation was very fruitful, many horrible, peaky, noisy records were also released. The 90s seem to herald a more balanced approach to recording, thankfully. While no one will be burned at the stake for using EQ while tracking, the predominant sentiment echoes the old farmer's proverb: "If it ain't broke, don't fix it!"

A few corollaries can be inferred from that proverb. First, if EQ is an integral factor in the development of a sound, then by all means use it. For example, when going for a certain quintessential drum sound, EQ becomes the essential ingredient. If, after poking around for a couple of hours you finally arrive at the sound you are looking for, it would seem prudent to capture it on tape just the way you are hearing it. Second, if EQ is necessary as a prophylactic measure, then use it (An example here would be rolling-off low-end rumble or high-end noise, if such extraneous trash becomes a distraction. Here, you would simply be filtering out unessential frequency bands, not actually doctoring the sound). The third corollary—and this would be the ultimate last resort—is if a sound was so unpleasant as to be unrecordable (because, for example, certain midrange components were making the sound harsh), then it might be well to improve the sound by EQ (This, however, should not be your first method of remedy. Usually, such a problem is best treated at the source by changing microphones, mic position, or in the case of a line-level signal, changing the sound itself).

Notwithstanding the above circumstances, it would seem that it is still wiser to record tracks without EQ whenever possible. This becomes especially true in the light of digital recording. Here it is irrelevant and even harmful to play unnecessary games with frequencies, for the quietness of the medium will reveal all peakiness in a very unflattering light. With special regard to vocals, recording with EQ can put unalterable spectral peaks in a vocal that will come back to haunt you later. For example, an extra dollop of upper mids or lower highs (say 7k to 10k) titillates the ear, but can wreak tremendous havoc by building up sibilance (excessive "s" sounds). If you don't catch this problem upon recording, it will come back to haunt you when you mix. EQ should be used, with discretion, on recorded tracks.

USES OF EQ WHILE MIXING

It may seem foolish, but it pays to ask this question periodically: "For what reasons do I use EQ as mixing

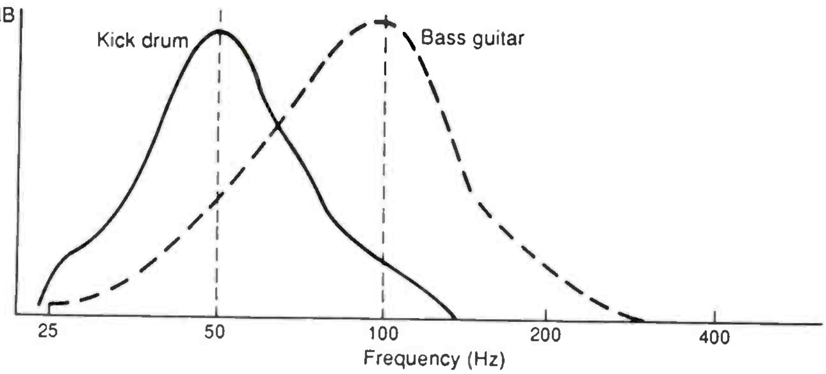


Figure 1. Complementary eqalization for bass instruments.

tool?" I like to ask myself that question every time I mix, because it keeps a rolling list of immense possibilities squarely in my consciousness. Think about it and make your own list. Here are a few from mine:

I use EQ to define a sound—to make it what it is, but even more so. It's like the difference between a photograph and a caricature of Richard Nixon. They are both clearly him, but the caricature has distilled his being into an essence that is more Nixon than Nixon himself.

EQ also allows me to add realism to a sound that may be accurately recorded, but lacking animation. EQ can also be used to increase stereo imagery. Instruments appearing as virtual images in both left and right channels can often sound more spacious if differences in equalization are introduced in addition to the obvious time and phase differences. The list could go on and on, but by far, the most universal use of EQ during mixdown is to help a track find a sonic niche relative to the other tracks. In other words, tracks need to be blended together, yet individual enough to be audible in the context of the mix. This is a factor that goes beyond mere definition. It is an interactive consideration where the frequency spectrum of each track affects (and is affected by) the spectral characteristics of all other tracks.

In dealing with this phenomenon, there is no concept more valuable than the notion of complementary equalization. The rationale for complementary EQ is a simple law of physics (and common sense): "No two things can occupy the same space at the same time." It stands to reason that if there are, say, a total "*x*" number of decibels allocated for a certain band of frequencies, if two or more instruments are all sitting in the identical range, they will either have to share the range and hence, each be pushed down to a lower level in the mix or have their range modified so they don't step on each other's toes, thereby allowing each to sit a little higher in the mix.

A few classic examples of this will make this point clear. The first is the case of the kick drum (a bass instrument) and bass guitar—essentially, the low-end of a standard rhythm section. Since both instruments, in a sense, compete for a share of the low-end terrain, they can either share the land or divide it. If, say, both instruments have their predominant concentration of bass energy at 100 Hz, one instrument will mask the other, making them sound like one big hybrid kick-guitar.

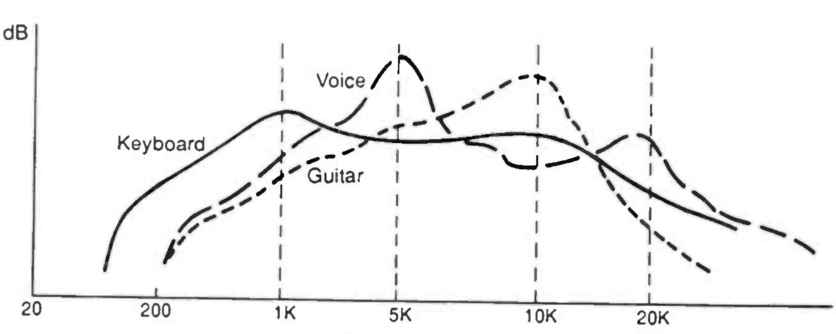
Sometimes this may be what works best, but most often you'll want to hear distinct low-end energy coming from each instrument. The way to deal with this is to simply shift the low-end frequency band for one of the instruments down an octave to 50 Hz, for example. The upshot of this will be two different kinds of bass energy which will be more easily distinguishable from each other (See *Figure 1*).

Another example is the case of placing a vocal in the mix. When EQ's are out-of-whack, it will be a nightmare trying to place the vocal in the mix. One moment it will seem buried by the music, and the next moment it will seem far too prominent. You might find yourself constantly changing fader level even though recorded level appears to be constant. You might try squashing it with a compressor and it still does not sit right. The problem might not be the voice at all. Don't forget that besides the above-mentioned bass guitar and kick drum, virtually all other instruments in a standard rhythm section (keyboards, guitars etc.) are all essentially midrange instruments. If they are all strong in the same frequency band, you literally have a sonic war on your hands. There is only one solution. Some elements must be weakened so the most important one will achieve dominance. Dominance does not mean "louder than..."; it simply means "more present than..."—in terms of midrange.

With a slight midrange roll-off on keyboard pads and other instruments that are constantly playing, the vocal track will sit comfortably nestled amongst them—always audible, but never overbearing. Why? Because you have actually carved-out a piece of the midrange terrain where the vocal can sit without competition (See *Figure 2*). Does this mean the keyboard and guitar tracks have to sound meek? Of course not. Let them have their own area of dominance shifted either up or down from the vocal EQ.

This then, is the concept of complementary EQ. Needless to say, it can be applied (when necessary) to all areas of the frequency spectrum. The goal is to create an evenly textured mix where the total sum of lows, mids and highs are in a pleas-

Figure 2. Complementary equalization for midrange instruments.



ing relationship, and each track has a well-defined sonic space.

MASTERING WITH EQ

Mastering is commonly thought of as "the process whereby a master tape is prepared for commercial release"; but in actuality, the principles of mastering can be applied to many everyday situations. For example, every time you make a safety copy or cassette dub, there exists an option to perform some typical mastering operations (this can be more broadly defined as "a second chance to get it right").

If most mixes were perfect, the mastering engineer would be out of work. He could easily be replaced by an automated device which senses input level and adjusts output accordingly. The truth is most mixes need a little help or at the very least, could be improved by (among other things) a little extra EQ. Many a mix has been pulled out of the fire by a person with an equalizer and a good set of ears. The most common problem (and the easiest to solve) is the case where all the tracks can be heard and are in good relationship, but the mix just seems to lack emotional impact. Typically, such a mix may be found to be accurate in the midrange, but lacking in sufficient warmth or brilliance. In these cases, shelving equalizers come in especially handy. By gently boosting the entire low-end or high-end, an element of sensuousness can be added to an otherwise dull, but clinically correct mix.

Other, more subtle problems can sometimes be successfully remedied in mastering. For example, an overall peakiness and harshness can usually be ironed out by rolling-off some resonant midrange frequency. Quite often, a narrow band of cut on a parametric equalizer will be all it takes to make a mix more listenable. At other times, a bit of select midrange boost will add power to a mix that does not seem to translate well in duplication. Even a vocal that seems to be slightly obscured by the instrumental tracks can often be made more intelligible by carefully finding and boosting the narrow band of the vocal formant. All in all, even a troublesome mix can often be revived by judicious use of EQ prior to duplication.

THE LANGUAGE OF EQUALIZATION

No treatise on equalization would be complete without offering some sort of a lexicon of EQ'ese. More than any other aspect of recording, the process of equalization seems to be replete with adjectives. Rarely do engineers, producers and artists communicate using specific technical terms like, "try x number of dB boost at frequency y"; most often, it is more like "let's warm up the guitar and add a little sparkle to the keyboards." While there might be some regionalism in this vocabulary and perhaps some variety in interpretation, it is still quite amazing how universal many of these terms are and how intuitively the human mind seems to grasp even unfamiliar terms. Perhaps they reflect a simple word/frequency association, but it seems there is a reality behind these words automatically grasped by sonically sensitive people.

While no attempt has been made to be comprehensive or authoritative, the following list does reflect some sort of a consensual opinion on the location between certain commonly used terms and their associations in the realm of frequency. You are encouraged to modify and add to this brief lexicon according to your common usage.

(My apologies to those who speak a slightly different dialect. I also realize these designations are subjective, and frequency ranges may shift depending on program content).

This vocabulary is offered in the interest of spurring better communication among practitioners of the audio arts.

Barillanguage

Boomy—applied to a sound overabundant in low lows (in the region of 40-60 Hz). These waves move a lot of air, hence, boomy.

Fat—generally applied to the octave above boominess (say 60-150 Hz). Makes things sound big, but not earth-shaking.

Woofy—a somewhat nebulous term for sounds that are sort of "covered"—masked by low-end energy (typically in the region of 125-250 Hz).

Puffy—is like an octave above woofy (say 250 to 500 Hz). It's still sort of a cloud, but not as big.

Warm—obviously a positive characteristic often found between 200 and 400 Hz. Could easily degenerate into woofiness or puffiness if overdone.

Boxy—seems to remind one of the sound in a small box-like room. (Usually found between 500 and 1 kHz).

Telephony—accentuating the limited bandwidth characteristic commonly associated with telephones (A concentration of frequencies around 1.5-2.5 kHz with a roll-off both above and below).

Cutting—Here, "cut" means to put an incisive "point" on the sound (2.5-4 kHz does this very effectively).

Presence—Anywhere from 3-6 kHz can be used to make a sound more present.

Sibilance—Dangerous "s" sounds and lots of other trashiness can often be found at 7-10 kHz.

Zizz—refers to a pleasantly biting high-end resonance (think of a "harpsichord"-type brightness found around 10-12 kHz).

Glass—A very translucent, but palpable brilliance associated with 12-15 kHz.

Sparkle—A real smooth stratospheric brilliance almost beyond hearing, but can certainly be sensed (Found at 15-20 kHz).

Brightness—Most generally achieved by a global (shelving) EQ of everything above 10 k.

Darkness—The opposite of brightness (a general lack of highs at 10 k and beyond).

Muddiness—Actually a compound problem: woofiness plus puffiness (excess low end and also low mids).

Thinness—The opposite of muddiness (a deficiency of lows and low mid frequencies).

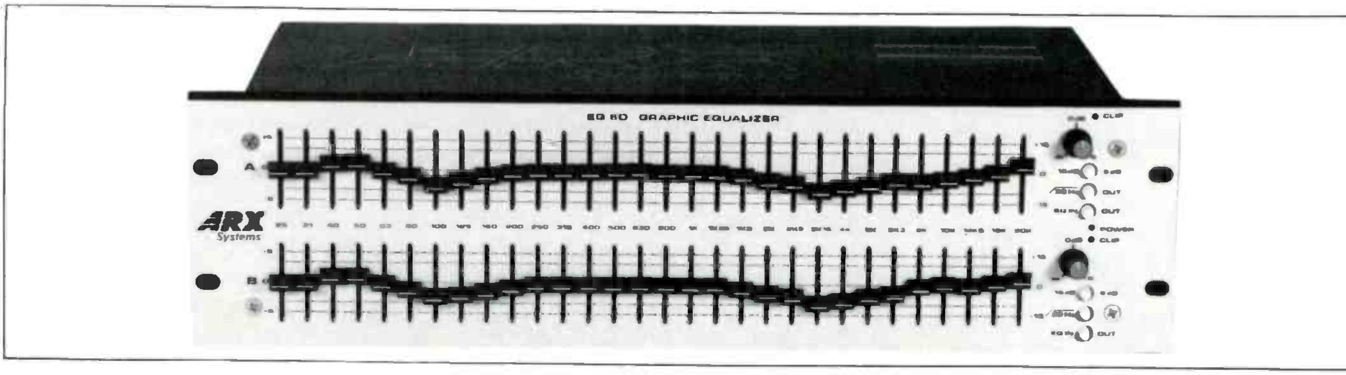
Openness—The quality of having sufficient highs and lows.

In conclusion, while no one can impart automatic discretion in the use of equalization, it should be clear that the art of equalization is one that can be studied, practiced and learned. Hopefully, this series of articles will be helpful to those looking for some light along the path. db

LEN FELDMAN

Lab Report

ARX Systems EQ 60 ⅓rd Octave Graphic Equalizer

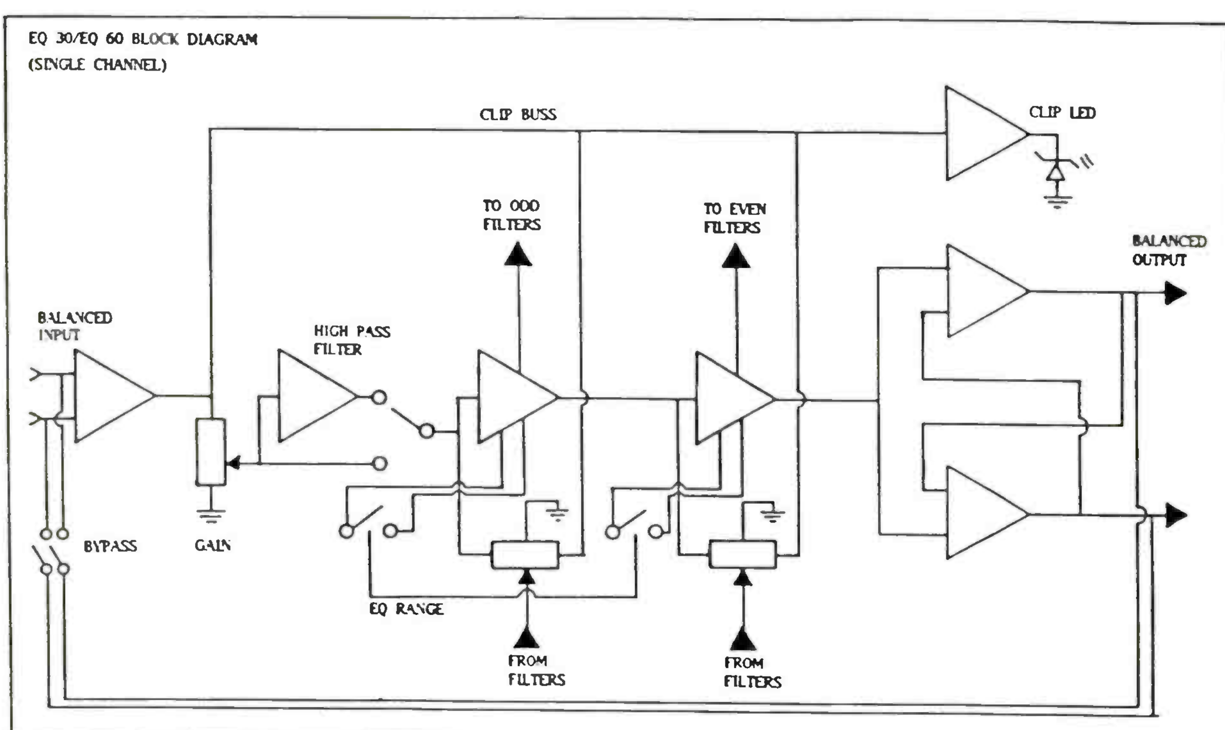


GENERAL INFORMATION

If you've always thought of Australia as a land filled with nothing but Koala Bears, Kangaroos and Wallabies (plus a few million people who speak English with a strange accent), you may be surprised to learn the land-down-under has a thriving professional and consumer audio industry. One company that has made its mark both below and above the equator is ARX Systems, whose excellent ⅓rd-octave dual channel graphic equalizer we have just tested and used. The ARX EQ 60 stereo graphic equalizer provides two channels of ⅓rd-octave equalization in a compact rack-mountable package that occupies only three rack-units of height. The company also makes a single-channel model, known as the EQ 30, which is identical in all other respects to the dual channel EQ 60 we tested. As we learned soon after hooking up the EQ 60 to our test equipment and plotting a few curves, the EQ 60 features innovative "Constant Q" circuitry developed by ARX engineers so that relative slopes of each ⅓rd-octave band are identical.

This feature makes it possible to tailor your final EQ curve much more precisely than would otherwise be the case. The maximum boost and cut range of each of the EQ controls can be set to 6 dB (for precise resolution) or,

Figure 1. Block diagram of a single channel of the ARX EQ 60. There is also an otherwise identical EQ-30 version that is single channel only.



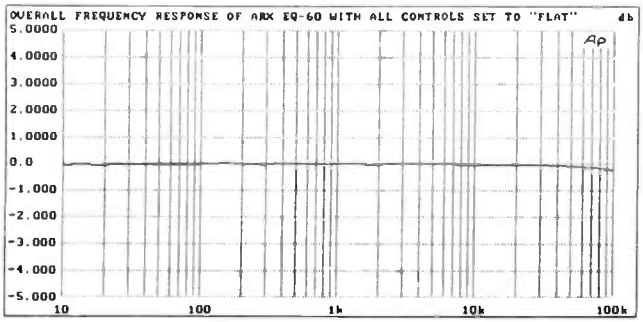


Figure 2. Frequency response, all controls set for "flat."

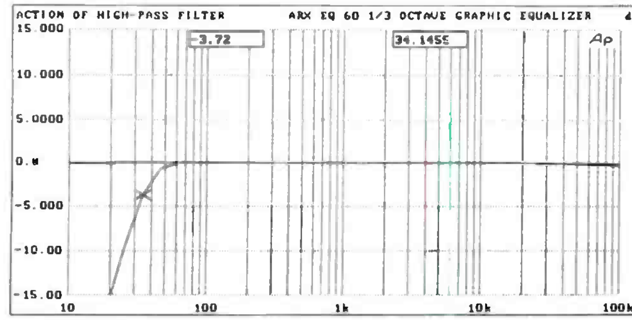


Figure 3. Action of the high-pass filter. The nominal cutoff for a –3 dB point is listed by ARX as 30 Hz.

when necessary, to 15 dB where extreme cut or boost is required.

The EQ 60 features balanced inputs and outputs, and when we took the cover off and examined the internal construction of the unit, we were pleased to note the extensive use of low-noise components as well as the intelligent layout of the p.c. boards. Of the three p.c. boards used, one houses all of the 60 smooth-acting slider EQ controls (30 per channel) and is mounted vertically, behind the front panel. A second board houses the equalizer filter circuitry while the third, smaller board houses level controls, high-pass filters and bypass switches as well as other input and output components.

The signal circuits are well-shielded from the power transformer by means of a steel plate that runs across the entire rear section of the EQ 60. The EQ 60 is housed in an all-steel chassis with a satin-finish brushed-aluminum front panel. Should something go wrong with this or any other ARX product (which is highly unlikely, judging by the care with which this unit has been assembled), you won't have to ship it back to Australia. ARX products are distributed in the United States by ARX Systems, P.O. Box 842, Silverado, CA. Interestingly, the sample we tested was shipped to **db Magazine** directly from Australia, but seemed none the worse for its long journey.

CONTROL LAYOUT

Most of the EQ 60's front panel is taken up by two identical horizontally laid out rows—one below the other—of 30 slider controls each. The controls have center detent positions. Although the detents don't feel quite as positive as some others we've used, once you are at a detent, the control is center-grounded so flat response for that ⅓-octave of the audio spectrum is assured. At the upper right corner of the front panel is a red LED that illuminates at approximately 2 dB before clipping occurs. The clip-indicating circuit is arranged so it detects any clipping at all vital stages throughout the equalizer circuitry and not just at the output stages.

Each channel is also equipped with an overall level control. When set to its center position, the level control provides unity gain (0 dB), while when set clockwise, up to 6 dB of gain is provided. Rotating the control fully clockwise provides infinite attenuation. Each channel is also equipped with three push-buttons. The first button is used to select either 6 dB or 15 dB of maximum boost or cut for each EQ control. The second button inserts a high-pass filter that attenuates response sharply below 30 Hz to reduce the amplitude of any stage rumbles or other subsonics. The third button is used to bypass the equalizer entirely, as shown in the block diagram of *Figure 1*. A green LED acts as a power indicator. There is no on/off switch.

The EQ 60's rear panel is equipped with XLR input and output connectors for each channel as well as with ¼-inch phone jacks. Either the XLR connectors or the jacks, when used with ring-tip-sleeve plugs can be used for balanced inputs and outputs. If, however, you use tip-sleeve phone plugs, the jacks can be used to feed unbalanced inputs to the equalizer. With so much confusion out there concerning the "standard" connections of an XLR 3-pin connector, we were pleased to see that ARX labels the pins in a diagram screened right onto the back panel. Specifically, pin 3 is *hot*, pin 2 is *cold* and pin 1 is

Figure 4 (A). The range of all equalizer controls versus frequency at the 6 dB settings.

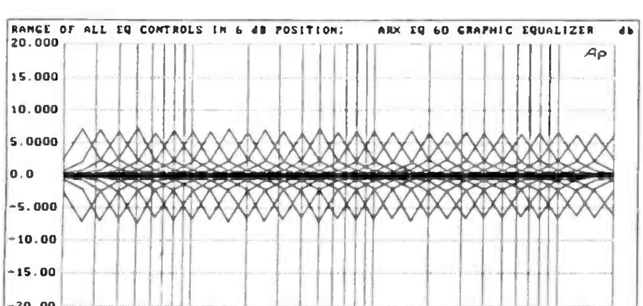
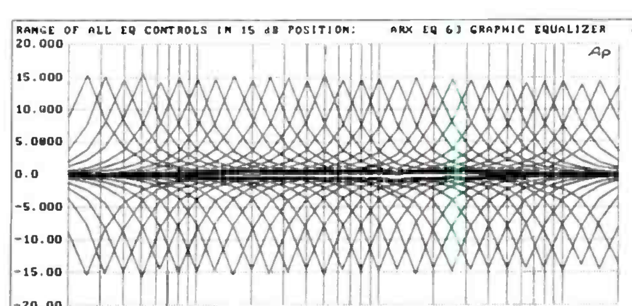


Figure 4 (B). The range of all equalizer controls versus frequency at the 15 dB settings.



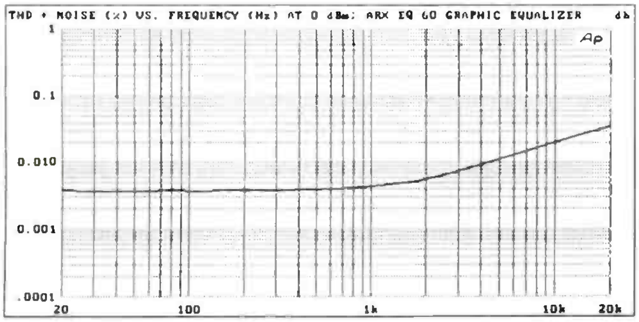


Figure 5. Harmonic distortion plus noise-versus-frequency with level controls set for unity gain.

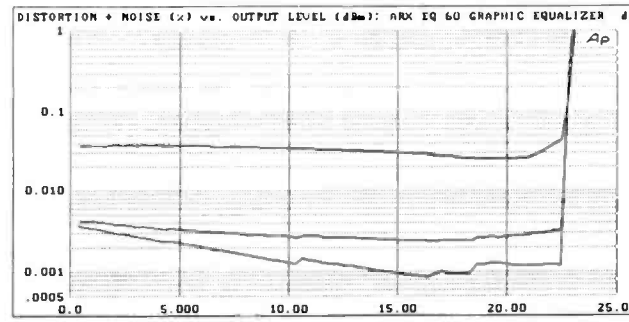


Figure 6. Harmonic distortion plus noise (%-versus-output (in dBm). Lowest TDH curve is for 20 Hz, middle is for 1 kHz, highest TDH curve is for 20 k

ground or, as they say down-under (and in the UK) *earth*. A fuseholder containing a 1 ampere fuse (for U.S. operation at 120 volts A.C.) and, of course, the power cord (terminated in a three-prong, grounded-type plug), completes the rear-panel layout.

LABORATORY MEASUREMENTS

With all slider controls set flat, but with the equalizer circuitry active, overall frequency response was extremely flat from 10 Hz to beyond 100 kHz, as shown in *Figure 2*. At 100 kHz, response was down by only about 0.2 dB! Our earlier concerns about the "softness" of the detent setting were baseless. Once a control is seated in the detent, flat response is assured. For *Figure 3*, we switched in the high-pass (low-cut) filter. Our test equipment is able to display actual data at any point in a plot, and the "X" cursor seen in *Figure 3* corresponds to a frequency of 34.1455 Hz, at which time the high-pass filter attenuated response by 3.72 dB. In other words, the cut-off point of this filter was slightly off, since the "specs" call for a –3 dB cut-off point at 30 Hz. The difference is hardly worth quibbling about, however.

The ARX EQ 60 is one of the most-precise graphic equalizers we have ever tested.

You begin to appreciate how well-designed this equalizer is when you examine *Figures 4(A)* and *4(B)*. Here, we plotted the maximum boost and cut of each of the 30 1⁄3-octave EQ controls for the 6 dB maximum settings (*Figure 4(A)*), and for the alternate 15 dB maximum boost or cut position of the equalizer range switch *(Figure 4(B))*. Notice how precise the maximum boost and cut range is for each of the 30 center frequencies. Those center frequencies, incidentally, are a standard as set by the ISO, and range from 25 Hz to 20 kHz. Notice, too, that the slopes of all the bands are virtually identical as is the spacing across the band. The graphs of *Figures 4(A)* and *4(B)* were made by running successive response sweeps, each with one of the 30 slider controls in either its maximum boost or its maximum cut position. Thus, it took 60 response plots to create the graphs of *Figures 4(A)* and *4(B)*.

Using an input level of 0 dBm, and with the level controls set for unity gain, we plotted harmonic distortion plus noise-versus-frequency for the EQ 60. Results are shown in *Figure 5*. At low- and mid-frequencies, THD-plus-noise varied from about 0.0038 percent to around 0.006 percent. THD rose slightly at higher frequencies, reaching a level of 0.036 percent at 20 kHz. Unweighted signal-to-noise ratio, referred to 0 dBm input and output, measured an impressively high 113 dB. When an A-weighting filter was added, the S/N reading improved to nearly 120 dB below reference level. We don't know whether ARX's published S/N specs of 93 dB and 98 dB are simply the result of extremely conservative specification writing or the result of a different method of measurement. In any case, ARX's contention that the EQ 60 is an ideal equalizer for use in digital-audio applications is certainly well-justified.

To check the signal-handling capability of the EQ 60, we next plotted THD-plus-noise versus output, in dBm, for test frequencies of 20 Hz, 1 kHz and 20 kHz. There was good correlation between these measurements and those of *Figure 5*, and we found that overload occurred at an output level of approximately 23 dBm. Worst-case THD was measured, as before, for the 20 kHz test signal.

CONCLUSIONS

The ARX EQ 60 is one of the most-precise graphic equalizers we have ever tested. Ergonomically, the control layout makes it easy to use and adjust, and because of the uniformity of slopes or "Q's" of each band, you can be certain the shape of the response curve you "draw" with the slider controls is the exact shape of the overall response you actually get when this component is installed in the audio signal path of a sound-reinforcement system or recording studio.

For precision-equalization tasks, the availability of the lower boost/cut range of 6 dB is a welcome feature, since in today's studio environment, a full 15 dB of cut or boost would seldom be needed. It would have been nice if each band were given the option of 6 or 15 dB maximum cut and boost ranges. As it stands, the switch alters the maximum boost and cut range of all the sliders at once.

Obviously, such a refinement would have added considerably to the unit's cost, not to mention the profusion of extra switches that would have had to clutter up the panel. So, all things considered, ARX has managed to provide the maximum amount of EQ flexibility in a well-engineered, well-executed and reasonably-priced product. db

VITAL STATISTICS

SPECIFICATION	MFR'S CLAIM	db MEASURED
Frequency Response (±0.25 dB)	10 Hz to 20 kHz	10 Hz to 100 kHz
Center Frequency Accuracy	±2%	Confirmed
Maximum cut or boost	±6 or 15 dB	Confirmed
High-Pass Filter	–3 dB @ 30 Hz	–3.7 dB @ 34 Hz
Maximum In/Out Level	+23 dBm	Confirmed
Distortion, 0 dB Gain		
100 Hz	0.004%	0.0038%
1 kHZ	0.0035%	0.0042%
10 kHz	0.01%	0.02%
S/N Ratio		
Unweighted	93 dB	113 dB
A-weighted	98 dB	120 dB
Input Impedance	20 k ohms	Confirmed
Output Impedance	150 ohms	Confirmed
Dimensions (WxHxD, inches)	19 X 5-1/4 X 9	Confirmed
Price: EQ 60–	$1,349.00	
EQ 30–	$899.00	

BRENT HARSHBARGER

Audio for the Church

• In the past few months, it has come to my attention that I need to go over some facts on the best way to purchase or upgrade a church sound system.

THE MISCONCEPTION

First of all, some churches complain that the local music or sound company has "taken them to the cleaners" because the sound system they just purchased doesn't work or sound good. Some church sound committees are so worried about being cheated that they become their own worst enemy by trying to do the right thing and getting the most for their money. For example, a church in my community decided they needed a new sound system. The sound committee called all of the sound companies and music stores in town, telling them the church wanted a sound system. They also called a "sound consultant" from the west coast to come and help, but he was also going to bid on the sound system installation. In response to the request, everyone contacted sent a proposal based on the equipment brand they were authorized to sell and what they thought would work. The differences in the system varied greatly in price, and in what was needed. This confused the sound committee even more, so they had everyone come in by appointment and explain why their system proposal was the best. The church is now more confused because the salespeople talked in technical terms, and a decision will probably be based on price.

Purchasing a sound system is an involved process. Therefore, I will first give a better understanding of what makes a good sound system and then how to qualify who, what and how to make your purchase.

A sound system is more than electronics; it is more accurately called an *electro-acoustic system*, because speech is being converted (which is an acoustic form) into electrical via a microphone and back, through the speakers. How a sound system sounds largely depends on how well the microphone sampled the speech, how well the speaker can reproduce what was sampled by the microphone and the conditions of the room it is installed in. The way a room affects the system is comparable to taking a speaker out of its enclosure, sounding terrible, but by placing it back into its box, it sounds great. The box dimensions change how the speaker sounds because the dimensions tune the box and make it resonate at certain frequencies. Similarly, by putting a speaker in a room, you are doing the same thing—putting a speaker in a bigger box.

The biggest misconception seems to be the mixing console and/or equalizer is what makes the system sound good, and that it can fix anything. They can enhance the sound, but they are not what makes a good sound system. The mixing console is the most impressive and fun part of a sound system, but it's the least important part. You can take the most expensive state-of-the-art mixing console, run it through an inefficient loudspeaker, and it is going to sound terrible, but if you take the cheapest mixing console and an efficient high-quality loudspeaker, you can have a good-sounding system.

Churches require two different types of systems because some people lean more toward speech (or the sermon), while others prefer music, and the requirements are different for each. The speech would need a central cluster and the music would use a split, or what some call stereo, even though only a few seats in the house can actually perceive any kind of a stereo image. A split cluster can reduce speech intelligibility because the two speakers' audio arrives at any given point in the room at different times, which causes phase cancellations and alters the frequency response. Phase cancellations in music can be a welcomed phenomena, but they destroy speech intelligibility. A well-designed and installed central cluster should have minimal phase cancellations.

Every worship environment is different and as such, the sound system has to cater to the type of worship. For example, a Baptist church would require a shorter reverberation time than a Catholic church, and a contemporary Baptist church would require less reverb time than a conservative Baptist church.

A contemporary Baptist church would probably use several electric instruments and heavy electronic processing, while the conservative Baptist church may have a large pipe organ and rely more on speech (sermon). You can see by now that the sound system's bandwidth and the sound pressure level will vary to the type of worship, making it impossible to use a one-system-fits-all design approach that many "box" houses would have you believe.

In designing any church sound system, you need to decide what your worship environment is and then base your system design around those requirements. If you worship at the contemporary Baptist church, your service would most likely have an electrified rhythm section and probably a brass section.

In my quest for making a better sound system design for churches, Brian Scott, once a representative for Electro-Voice in Buchanan, MI, told me I should meet John Murray, an audio engineer. We got together a few times and shared ideas. He impressed me, not only with his vast knowledge of electro-acoustics, but with his simplification of a good sound system's design into five parameters. This is how you start to determine what your system requires. The five parameters are level (which we have already discussed), bandwidth, coverage, gain before feedback and intelligibility.

BANDWIDTH

Bandwidth is the mathematical difference between the upper and lower cut-off frequencies of an audio system. For example, the bandwidth needed for the conservative Baptist church would only be the frequencies required for speech. The upper cutoff frequency for speech is approximately 12,000 Hz and the lower cutoff frequency is approximately 100 Hz. Therefore, the system only has to reproduce the frequencies from 100 Hz to 12,000 Hz. A more progressive church service would have to reproduce a wider bandwidth—frequencies from 40 Hz to 18,000 to 20,000 Hz. Bandwidth is also a factor of price, because the wider the bandwidth, the higher the cost of transducers, and usually more power required by the amplifier, which can result in *X* amount of dollars per watt of power.

COVERAGE

An ideal sound system has a uniform coverage throughout the house which is not more than ±3 dB in sound pressure level at any seat in the church. This assures that the person in the front pew has the same sound pressure level as the person in the last pew.

The choice of electroacoustic transducer, and the placement and aiming of the transducers, has a larger effect on the sound than some might believe. In selecting the transducers, or in this case, the speakers, they need to meet the demands you are asking the system to perform, i.e. the bandwidth, coverage and level required.

The cone, driver and horn are the devices that make up speaker systems as we know them today. The cone is a voice coil-type speaker commonly referred to as a woofer. The driver is used for mid to high frequencies and is coupled to a horn. The horn is a vital component in a sound system's design because it is directly related to the coverage. The most widely used coverage pattern is 90 degrees horizontally and 40 degrees vertically. Others include 120 x 40, 60 x 40 and 40 x 20. These coverage angles allow you to select long, medium and short throws, giving you control of making sure the entire seating area is covered, leaving no dead spots, and also achieving our ±3 dB of SPL throughout the church.

GAIN BEFORE FEEDBACK

How much gain or level that can be obtained before the system starts to feed back is known as Potential Acoustic Gain, or PAG. The mathematical equation for PAG is the following:

If more than one microphone is being considered as part of the PAG, then you would have to subtract PAG to the logarithmic equation of the Number of Open Microphones (NOM). Remember: every time you double the amount of microphones being used, you lose 3 dB of level before the system starts to feed back.

Now that we know what our PAG is, we need to find out what our Needed Acoustic Gain (NAG) is. NAG is gain in decibels needed to get the desired sound pressure level to the farthest listener from the talker in order to achieve the same sound pressure level as if these two people were having a face-to-face conversation (approximately 4 to 12 feet apart). If the NAG is greater than the PAG, you will not be able to reach the desired level before the system feeds back.

INTELLIGIBILITY

As the pastor speaks, the level is great, but can you understand what he is saying? This is one of the most challenging tasks for the architect, acoustical consultant and sound system designer—making the speech intelligible. Without going into great detail, I will explain the factors that affect speech intelligibility. The first can be an acoustic or electronic problem, signal-to-noise ratio (S/N). The second is in the acoustic domain that is RT60 (reverberation time). Next is the distance from the source. Others include, but are not limited to, speaker misalignment and strong reflections (Intelligibility will be covered in more detail in the next issue of **db**).

One of the most frequent questions asked by a church sound committee is "who do we purchase a sound system from?" Some may purchase a sound system from the music store the organ was bought from. After all, music is sound, and if you can hear and balance musical pitches, you make the sound system sound great. Right? Maybe, but as you've read, sound system engineering is more physics than music.

Some church committees may go to the local electronic systems contractor and purchase a sound system from them. After all, sound systems are electronics, right? Remember: sound systems are actually electro acoustic systems. A sound system is purchased from the person and/or company that understands, and has experience designing a sound system with the parameters that have been described. Some music stores have someone on staff who is well-equipped to handle this task, but unfortunately, most don't. On the flip side, most sound and communication companies have the personnel to deal with designing a church sound system as we have described.

The person and/or company you choose should also have the proper design and test equipment available. The equipment needed for system design should include a computer-aided design program that will show you not only sound pressure levels, but NAG, PAG, coverage, intelligibility (%alcons, or rasti measurements), reverberation time and reflections. There are several system design programs available that give a 3D view of the room and the sound pressure level.

The system test equipment should include a sound pressure level meter, real time analyzer, oscilloscope and an acoustical measurement system, such as Techcron's TEF 12, or TEF 20 system, or even the DRA-MLSSA system for the IBM.

The last item in selecting a company to design and install your sound system is to ask for references on some of the sound systems they have done in your area, preferably several the same size as yours, some larger and some smaller. Go to these facilities and talk to the pastor, the minister of music and the people who operate the system. Ask lots of questions.

If you would like to take a deeper look into sound-system design, I would suggest several books: *Sound System Engineering* by Don and Carolyn Davis, The *Handbook of Sound System Design* by John Eargle and Handbook for Sound Engineers (The New Audio Cyclopedia) edited by Glen Ballou.

By the way, this column is already one year old. I have enjoyed writing for you this past year, and trust you have, too. From the comments I have heard, I am reaching your needs, and I would like to thank those who have written and shared their questions and comments. If you would like to write, I will respond. Address all correspondence to Brent Harshbarger, P.O. Box 1702, Springfield, OH 45501. db

db magazine
serving: recording, broadcast and sound contracting fields

Book Review:

Tonmeister Technology
by Michael Dickreiter, published by TEMMER ENTERPRISES INC. 1989, 141 pages.
$21.95, paperback; $34.95, hardcover.

• The author, who has been associated with the teaching facilities maintained by the German Broadcasting Systems at Nuremberg (F.R. Germany) as a department head, concentrates his book on the application of microphone techniques for the recording of primarily classical music. Michael Dickreiter received his degree as a *Tonmeister* (German for "Sound Master," the approximate equivalent of a sound engineer) from the sound engineering institute associated with the *Musicademie* in Detmold (F.R.G.), one of the most renowned Tonmeister academies in Western Europe.

According to Mr. Dickreiter, proper microphone techniques can only be applied with a thorough understanding of the Recording Environment and the Sound Source. Therefore, one-half of his 141-page book has been devoted to each of these topics. The book has been translated from German by Stephen F. Temmer, former co-owner and chief-engineer of Gotham Recording Corporation and Gotham Audio Corporation. He recommends this book be used as a text for short-course seminars and as a component text in regular undergraduate courses. Other books by Mr. Dickreiter include *Handbook of Studio Technology* (in German) and *Score Reading* (in German and Japanese).

The book is divided into three main sections: Recording Environments, Sound Sources and Microphone Techniques. The first chapter begins with a comprehensible discussion of sound waves, sound propagation and fundamentals. This is also the only occasion where the author utilizes a few, although very basic, mathematical formulas. In general, Tonmeister Technology avoids mathematical expressions and attempts to treat every topic verbally. The first chapter also discusses sound diffraction and reflection, and includes informative pages on absorption, direct sound and reverberation. The definition of a "reverberation radius," which is said to be dependent on the directivity of the sound source, is unique. However, the phenomenon of sound masking, which the chapter completely omits, should have been included.

The early part of chapter two (Sound Sources) provides the reader interested in classical recording with a basic understanding of the most common orchestral and chamber ensemble seating arrangements, and then goes into detail about specific acoustics of various musical instruments. Every major orchestral instrument, such as members of the woodwind, brass and string families, is discussed in their dynamic range, notation, sound generation and sound spectrum. I took particular interest in the radiation patterns supplied for each individual instrument. Percussion instruments, as well as the singing and speaking voices, are also analyzed in that fashion. Criticisms may be directed towards the lack of a thorough piano sound analysis.

The third chapter contains a description of different microphone patterns (directional characteristics) and their frequency response. The text proceeds with the obligatory section about special purpose microphones, spatial hearing, stereophony and stereo signal monitoring. Stereo signal monitoring incorporates directions of how to interpret Lissajou patterns and the phase meter (correlation coefficient meter).

All information about stereo microphone techniques is condensed to 20 pages, discussing time-of-arrival stereo technique, intensity stereo techniques (M-S,X-Y), support microphones, dummy-head binaural technique, multi-microphone techniques and so on. The remaining pages deal with microphone placement for individual instruments such as woodwinds, strings, brass, drums, percussion and keyboard instruments such as harpsichord and piano. The book ends with a discussion of aesthetic principles in musical recording.

From a graphic point of view, Tonmeister Technology is neatly and professionally produced. The graphics and illustrations are consistent in style, layout and density, and are in proportion to the written text. However, the idea of having written text and graphics on opposite pages is disadvantageous because the average line length is too long to ensure an ideal eye comprehension level. The book could be much better if the author used a clearer writing style instead of a severe academic approach. Mr. Dickreiter's use of words in place of mathematical expressions did not make the text easier to understand.

Part of chapter one and all of chapter two are unique, and truly supplement the education of the future engineer. They resemble in many details, parts of J. Meyers' book on *Acoustics and Musical Performance Practise*, (Frankfurt 1972),since some drawings (with credit) come from that source. The author obviously tried to focus on certain topics more than on others. The problem with this is the information supplied does not suffice to enable students to pick the appropriate microphone for their recording.

If the book is, as suggested by the translator, intended as a textbook for short-course seminars, it will need to provide more practical knowledge about microphones. For example, Mr. Dickreiter gives a detailed explanation of the proximity effect, but he does not mention the influence of diaphragm size on the microphone sound. In discussing soloist microphones (a hand-held microphone, according to the author), he fails to mention that the bass rise due to the proximity effect is not necessarily always unwanted, but is often consciously used by vocalists to create a feeling of intimacy.

While concentrating almost exclusively on classical music recording, this chapter should have included the most popular microphone techniques used by major classical recording labels (the book only mentions the ORTF method and the JECKLIN DISK). The section about mic placement for individual instruments could have been omitted; there is no discussion on which microphone to use, just as there is no mention about minor details like phantom power or how to use an M-S matrix. The intelligent reader will certainly be able to find proper microphone placement through the knowledge derived from chapter two.

Another problem with Tonmeister Technology is that it was originally written for the German recording industry, and is not tailored to fit the American recording business. This becomes apparent when the author discusses "real peak" metering versus "quasi peak" metering. The author should instead have included the various types of metering (such as IEC Type I+II, BBC, VU and others), their ranges and response times (these are listed in "The New Recording Studio Handbook" by J. Woram and A.P. Kefauver).

References and supplementary readings in each chapter would have been useful, especially for students. I would recommend spending a little extra than $20 for Tonmeister Technology to obtain the excellent book *Acoustics and Musical Performance Practice*" by J. Meyer (Das Musikinstrument Publishing Co., Frankfurt 1972) in English. (This book,allso distributed by Temmer Enterprises, sells for $95.00!)This book contains more detailed information for the engineer interested in classical recording, including the here omitted sections about masking and piano sound. Mr. Dickreiter's book can be seen as a supplement to the sound engineer's library; it is a helpful and informative guide to classical music recording.

Bernd Gottinger is a senior Recording Arts major at the Peabody Institute of Johns Hopkins University. He is also a double major in Double Bass performance, studying with E. Levinson (Principal Bass N.Y. Philharmonic). Mr. Gottinger completed his Abitur (German High School Diploma) in Munich (F.R.G) in 1985, and then continued his musical education at the L. Mozart Konservatorium in Augsburg. In 1986, he was admitted into the highly-competitive sound engineering program of the music academy in Dusseldorf (F.R.G), one of the five major sound engineering schools in Europe, but chose to come to the United States for his studies.

• To order Tonmeister Technology see the classified ad on page 65.

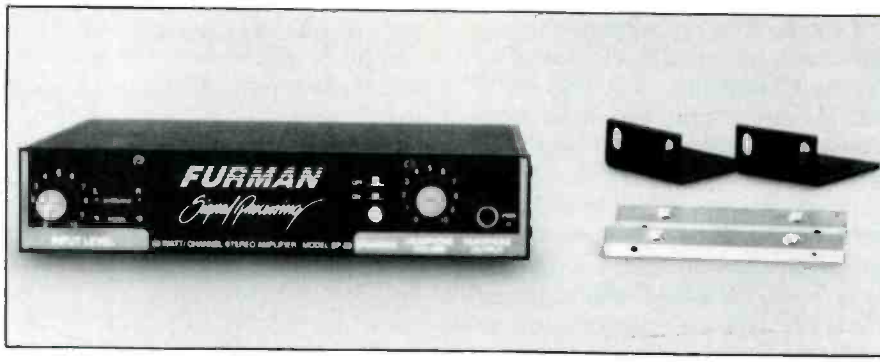
New Products

HIGH POWER AMP

• The MX 4000 has power ratings per channel of 750 watts at 8 ohms, 1125 watts at 4 ohms and 1500 watts at 2 ohms. In addition the amplifier incorporates "open architecture" via input connectors mounted on a removable module. This module allows for interface with control systems as they develop, and will also be compatible with a second generation of signal processing devices that the company is currently developing. New circuit designs include a built-in limiter that prevents excess clipping and also acts to smoothly reduce power if amplifier temperature becomes extreme.

Mfr: QSC Audio
Price: $2,798.00.
Circle 52 on Reader Service Card

POWER AMPLIFIER



• The SP-20 Stereo Power Amp is a compact, half rack 20-watt per channel stereo power amplifier. It may be switched for normal stereo, dual-channel mono, or bridged 40-watt mono operation as needed. It includes a stereo Input Level control, and Signal Present and Overload LEDs for each channel. It also features a headphone output with its own volume control and speaker mute switch, and if desired, the SP-20 can be used as a multi-station headphone amp by connecting a chain of HR-2 Headphone Remote Stations. The SP-20 offers low .01% THD at full rated output at 1 KHz, and .05% THD from 20 Hz to 20 KHz. It is fully protected against thermal overload, and can withstand a short-circuit on any or all outputs for an indefinite time without damage. It is available with optional XLR balanced inputs as model SP-20B.
Mfr: Furman Sound
Price: $289 for the SP-20 and $309 for the SP-20B.

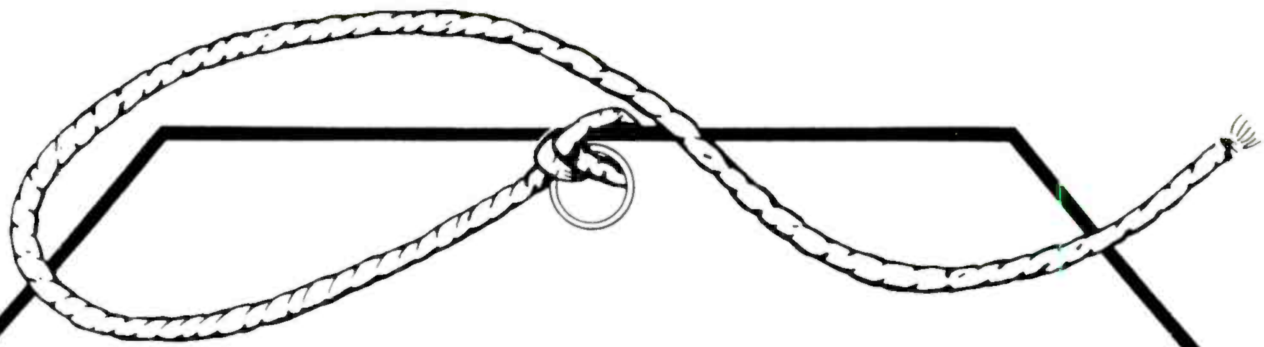
Circle 53 on Reader Service Card

TAPE RECORDER HEADS

• Immediate delivery of factory-equivalent Record and Playback Heads for use with Studer A-80 and A-800 Series 24-track professional studio recorders are available. The heads are made of longlife Permalloy to meet or exceed Studer electrical and mechanical specifications in all essential characteristics, and are interchangeable with original heads with no wiring modification.
Mfr: Saki Magnetics
Price: for each head is $2,850.
Circle 54 on Reader Service Card

NEW MICROPHONE

• This first model in a new line of video production microphones is a single-point stereo condenser microphone incorporating two independent mic elements to produce a classic Mid-Side (MS) stereo signal. As a self-contained stereo mic'ing system, the VP88 will find use in professional stereo applications. The VP88 incorporates two condenser microphone cartridges mounted in a coincident fashion to produce a stereo signal that is fully mono compatible. The Mid capsule faces directly forward, utilizing a cardioid polar pattern, while the Side element is perpendicular to the Mid element and employs a bidirectional pickup. The outputs of these elements are available to the user in either stereo or MS modes. In stereo mode, the VP88's on-board matrix produces separate left and right stereo signals with three distinctive, switch-selectable stereo images. In its MS mode, the VP88 sends discrete, fully separated Mid and Side cartridge signals to its output for external processing. The VP88 can also be used as a single-output cardioid or bidirectional microphone in this mode.
Mfr: Shure Bros.
Price: $995.00, which includes battery, carrying bag, foam windscreen, swivel adapter and Y-splitter cable.
Circle 55 on Reader Service Card



db

serving: recording, broadcast and sound contracting fields

Buyer's Guide—Microphones (including Wireless Mics), Recording Tape and Tape Accessories

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

As usual, please be aware that we attempt to contact every manufacturer, but not all are cooperative or prompt enough for our necessary deadlines. Accordingly, one or more manufacturers in these listings may have the statement *1989 Information*, meaning that no new information was received and we elected to provide the previous years' but with the warning that the information may not be not current.

Model	Type	Pattern	Frequency Response ±dB	Impedance, Ω	Sensitivity, 1k	SPL at % distortion	Dimensions-in., L, D, W	Weight, oz	Finish	Connector	Price, $	Features
model	type	ptrn	freq	imp	sens	spl	dim	wt	fin	con	$	Features
AKG ACOUSTICS, INC.												
The Tube	cond	multi	30-20k	200	10mV	128 0.5	1.7 8.9	24oz	brown	XLR	2,295.00	One of many models, both condenser and dynamic plus accessories available.
C414 ULS	cond	multi	20-20k	180	12.5	140 0.5	5.6 1.8 1.4	11oz	black matte	XLR	1,045.00	
C410	mini cond	card	20-20k	200	3mV	123 0.5		4.6oz	black	9V	225.00	Battery pack included, headset style.
C426B	stereo cond	multi	20-20k	200	11mV	132 0.5	1.7 9.3	1 lb	black	XLR	3,395.00	
C535EB	cond	card	20-20k	200	7mV	132 0.5	1.8 7.2	1.1oz	black	XLR	350.00	
C1000S	cond	multi	50-20k	200	6mV	137 0.5	1.3 8.7	9.7oz	black	XLR	325.00	
D112	dyn	card	20-17k	210	1.8mV-		4.5 2.9 5.9	13.4oz	gray metal	XLR	225.00	
D321	dyn	hyper card	40-20k	300	1.4mV		1.9 7.3	11.7oz	gray	XLR	210.00	
ALTEC LANSING CORPORATION 1989 Information												
D645	dyn	card	80-15k	150	56		6.6 1.4	6	beig matt		380.00	Offers an outstandingly smoooth frequency response with great naturalness and accuracy of inflection.
D658	dyn	card	80-13	155	61		6.6 2.16	8.9	grey		100.00	Moving coil dynamic, designed for the performer. Head design provides wide linear frequency response allowing greater gain-before-feedback.
C644	cond	card	50-18k	150	50		7.5 1.95	12.	black	A3M	220.00	Ideal for speech or singing reinforcement systems where there is a tendecy towards feedback.
C649	cond	card	40-18k	150	45		6.94 1.06	8.0	beig		390.00	Wide range,high input Z without distortion. A true cardiod pattern. Designed for voice.
D90P	dyn	omni page	50-15k	150	58		7.5 1.6	16	satn chrm		236.00	
D91P	dyn	omni close	180-10k	150	60		3.6 2.5	4.5	char grey		164.00	
D646	dyn	super card	80-15k	150	57		7.0 1.9	8.0	slvr	A3M	410.00	
654A	dyn	card	50-15k	200	56		7.25 1.875	8.0	satn nckl	A3M	240.00	
AMS/CALREC												
Sound Field-MK4	cond	multi	20-20k	100	115	140	9.5 0.5	18 2.5	black	XLR alum	5,850.00	True coincident stereo mic. Totally steerable horizontal and vertically.
ST250	cond	multi	20-20k	100	47	135 0.5	-	-	black alum	XLR	3,800.00	True coincident stereo, X-Y or MS. Remote control of patterns and capsule angle.
AUDIO-TECHNICA —See our ad on Cover II												
AT4031	cond	uni	30-20k	200	46	145 1	6.3 .81	4.9	black chrm	XLR	325.00	Pressure gradient capacitor, low self noise,high SPL capability. 9-52 V phantom.
ATM25	dyn	hyper card	30-15k	600	57	150 1	4.65 1.5	13		XLR	238.00	Large diaphragm high SPL capability,good low-end response. Suited for instrument mic'ing.
AT857	elect	card	30-20k	200	45	130 1	1.78 0.49	0.3	flat black	XLR	285.00	Specially suited to pulpit use because of unobtrusive appearance.
AT853	elect	card	30-20k	200	45	130 1	1.78 0.49	0.3	flat black	XLR	237.00	This is fast becoming the standard "choir mic" because of its small size and natural sound.

Model	Type	Pattern	Frequency Response ±dB	Impedance, Ω	Sensitivity, 1k	SPL at % distortion	Dimensions-in., L, D, W	Weight, oz	Finish	Connector	Price, $	Features
BEYER DYNAMIC,INC												
HM560	dyn	card	20-20k 3	200	58	138 1	6 0.5	6.8	black nickel	XLR	299.95	Lightweight headworn model.
MC742	stereo cond	multi	40-20k 3	150	46	144 1	9.5 3.5	11.5	black anad.	XLR	2,795.00	For MS,X-Y, and ORTF stereo, low freq. rolloff.
MC740	cond	multi	40-20k 3	150	46	144 1	8 3.5	10	black anad.	XLR	1,350.00	3 pos. rolloff,switchable 10 dB atten.
MC734	cond	card	30-20k 3	150	46	140 1	6.5 3	13	black or nickel	XLR	799.00	3 pos. rolloff,switchable 10 dB atten.
TG-X 180	dyn	hyper card	40-16k 3	290	54	140 1	6 2.5	4.5	black anad.	XLR	159.00	Small, high output.
TG-X 280	dyn	hyper card	30-16k 3	290	54	140 1	6 2.5	4.5	black anad.	XLR	199.00	Vocal instrument applications.
TG-X 480	dyn	hyper card	40-18k 3	280	50	140 1	8 3	6	black anad.	XLR	269.00	Vocal mic with pronounced proximity effect.
TG-X 580	dyn	hyper card	30-18k 3	280	50	140 1	8 3	6	black anad.	XLR	349.00	Fast transients and extended frequency response.
BRUEL AND KJAER												
4011	cond	card	40-20k 2	180	40	110 0.5	6.75 .75	5.8	black chrm	XLR	1,497.00	Transformerless,phantom powered.Switchable 0,20dB attenuator. Flat response both on and off-axis.
3529/30	cond	omni	10-20k 2	30	24	135 1.0	6.5 .63	5.3	black chrm	4pin XLR	5,08.00	Matched stereo pair (4003/4006),phase matched,amplitude, true omni nose-cones.
4003	cond	omni	10-20k 2	30	24	135 1.0	6.5 .63	5.3	black chrm	4pin XLR	1,337.00	Dynamic range from 15 to 154dB SPL(A)typical. Uses 2812 power supply.
4006	cond	omni	20-20k	30	36	135 1.0	6.5 .63	5.3	black chrm	XLR	1,226.00	Dynamic range from 15 to 143dB SPL(A)typical. Uses 2812 power supply.
4004	cond	omni	10-40k 2	30	38	148 1.0	6.5 .63	5.3	black chrm	4pin XLR	1,337.00	Dynamic range from 24 to 168dB SPL(A)typical. Uses 2812 power supply.
4007	cond	omni	20-40k 2	30	50	148 1.0	6.5 .63	5.3	black chrm	XLR	1,337.00	Dynamic range from 24 to 155dB SPL(A)typical. Phantom powered.
CARVIN CORPORATION												
CM68	dyn	card	45-15k	600	74		6.7 2.1	10.25	grey enamel	XLR	99.00	Vocal hand held mic.
CM67	dyn	card	40-15k	250	77		6.5 1.5	10	grey enamel	XLR	99.00	Studio or concert instrument mic'ing.
CM90	cond	card	30-20k	600	71		7.5 1.03	6.2	grey enamel	XLR	139.00	Flat response and wide frequency range.
COUNTRYMAN ASSOCIATES												
Iso max 2	cond	omni hyper card	20-20k 3	600	57	150 1	.81 .43	2.8	matt black	XLR	293.09	Multi-purpose mini-mic,four patterns,wide dynamic range.
Iso max 3	cond	omni hyper fig 8	20-20k	600	57	150 1	.81 .43	4.0	matt black	XLR	324.09	Same as Isomax 2 except provided on variable-length goosneck.
Iso max4	cond	card	70-18k 3	270	52	120 1	.81 .43	4.4	matt black	XLR	478.31	Podium/handheld,optimized for voice,variable-length goosneck, high-pass filter, electronic vibration isolation.
Iso max TVH	elec cond	hyper	70-18k 3	270	52	120 1.0	.81 .43	3.2	matt black	XLR	447.38	High gain before feedback.Rejects background noise and phase cancellation from multiple mics. Active vibration isolation.
EM-301	elec cond	omni	50-15k 3	600	52	130 3.0	.75 .34	3.0	matt black	XLR	384.19	Waterproof for mounting into conference tables,pulpits, baptistries. Internal isolation minimizes mechanical noise transmission.
Head set	elec	card/	50-15	600	57	150	.31	3.4	black	XLR	338.87	Low profile headset mic. Equalized for warmth and clarity.
CT MARKETING												
Gig ster	cond		35-25k	10k	77		3 0.05	2oz 0.75	brown plastic	phone jack	145.00	Also avail. in 8-in.length. Contact mic, popular for piano mic'ing.
CX	cond		35-25k	600	79		3 0.05	2oz 0.75	brown plastic	XLR phone	266.00	Also avail. in 8-in.length. Contact mic, popular for piano mic'ing.

Model	Type	Pattern	Frequency Response ±dB	Impedance, Ω	Sensitivity, 1k	SPL at % distortion	Dimensions-in., L, D, W	Weight, oz	Finish	Connector	Price, $	Features
CROWN INTERNATIONAL —See our ad on Cover III												
PZM-30 R	elec cond	hemi	25-20k 6	240	65	150 3	5 6	6.5	black alum.	XLR	349.00	Pressure Zone Microphone©. Smaller models available.
PZM-6-R	elec cond	hemi	20-20k 3	240	67	150 3	6 5	6.5	slvr alum	XLR	349.00	Pressure Zone Microphone©. Smaller models available.
GLM-100	elec cond	omni	20-20k 3	240	71	150 3	.75 .31	1.0	black PVC	XLR	199.00	Miniature clip-on mic for voice and instruments. Model GLM-100F for wireless transmitters.
LM-200a	elec cond	super	80-13k 6	150		120 3	16 .31	10.2	black PVC	XLR	289.00	
PCC-160	elec cond	half super	50-18k 6	150	53	120 3	6.7 3.2	11.5	blac steel	XLR	295.00	For stage-floor pickup of drama,musicals,opera,lecterns,news desks. Bass-tilt switch. Model PCC-200 is gated.
CM-200	elec cond	card	80-15k 6	200	73	151 3	7.5 1.8	7.0	black alum steel	XLR	209.00	For hand-held stage vocals and instruments.Wood handles available. Model CM-100 is PZM omni hand-held mic.
CM-310	elec cond	card	60-18k 6	200	77	151 3	7.3 2.0	7.0	black alum	XLR	259.00	Differential cardiod has outstanding gain-before-feedback. Wood handles available.
SASS-P	elec	omni/ uni	20-18k 3	240		150	11.56 5.28 5.7	17	black/ grey steel	XLR	899.00	PZM© stereo mic, mono compatible, all accessories included.
ELECTRO-VOICE, INC. (A MARK IV COMPANY)												
N/D 408A	dyn	card	60-22	150	51		2.75 2.85	6.7	blk	XLR	258.00	Pivoting instrument mic with special element for wide freuency response and high output.
N/D 457A	dyn	card	55-21	150	51		7.12 2.05	7.05	blk	XLR	256.00	Hand-held vocal mic with hypercardioid pattern for very high gain before feedback.
N/D 757A	dyn	card	50-22	150	51		7.12 2.05	7.7	blk	XLR	330.00	Hand-held vocal mic with extended frequency response, switchable low-frequency roll-off filter and special element.
N/D 857A	dyn	super	50-22	150	50		7.40 2.05	7.9	blk	XLR	450.00	Acoustical path corrector provides increased sens. and a uniform polar pattern.
N/D 357A	dyn	card	55-20	150	53		7.12 2.05	7.05	blk	XLR	206.00	Works well wiht both live and studio applications.
N/D 257A	dyn	card	65-19	150	53		7.12 2.05	7.05	blk	XLR	152.00	Offers 3 dB more output than others in its class.
N/D 308A	dyn	card	65-19	150	53		2.75 2.85	6.7	blk	XLR	222.00	Ideal for kick drum and other percussion applications as well as guitar amps.
FOSTEX CORPORATION OF AMERICA												
M77RP	ribbon	card	40-18k	250	56	147 2	1.75 6.8	12	black alum	XLR	460.00	Excellent mic for kick and snare. Has three position EQ switch.
M88RP	ribbon	bi	40-18k	600	58	55-2	5.5 1.75 2	2.0	black alum	XLR	650.00	Bi-directional mic can be used to produce a subtle ambience. Has useful deep notch. Response to 90 degrees off axis.
M20RP	ribbon	M/S	40-18k	600	54	148 2	6.87 2.75	25	black alum	XLR	695.00	Trimmed-down less expensive version of M22RP.
M11RP	ribbon	uni	40-18k	600	54	148 2	5.5 1.5 2.75	20	black zinc	XLR	595.00	Smoothest cardiod pattern in the RP series. Offers flat-frequency response.
HM ELECTRONICS,INC												
EM43-4	elec	omni	20-20k	2.2	65		.29 .78	1.5	black alum	TA4F	75.75	Built in by-pass capacitor to eliminiate RF interaction, high gain-before feedback and high SPL capability.
HM58	dyn	card	80-14k	600	75		6.6 2.0	10	grey	XLR	133.25	Designed for high quality professional applications.
RM77	elec cond	card	150-15k	600	74		7.5 2.0	11	grey	XLR	133.25	Built-in reverb with variable control. Mute switch.
NEUMANN(GOTHAM AUDIO)												
TLM-170	cond	multi	40-18k	150	8mV	140 0.5	6.0 2.4	22	nickel matt	XLR	1995.00	Advanced studio microphone.Transformerless.
KM130	cond	omni	40-20k	50	12mV	140 0.5	3.6 0.8	2.8	black brass	XLR	640.00	Miniature condenser mic. Transformerless output.Dynamic range of 124 dB. Capable of remotely powered capsule.
KM140	cond	card	40-20k	50	15mV	138 0.5	3.6 0.8	2.8	black brass	XLR	640.00	Similar to KM140, but with dynamic range of 122 dB.
KM143	cond	hyper card	40-20k	50	10mV	142 0.5	3.66 0.83	2.82	dark	XLR	640.00	Wide cardioid, new polar pattern.
KM145	cond	card	40-20k	50	14mV	138 0.5	3.6 0.8	2.8	black brass	XLR	640.00	Similar to KM140, but with dynamic range of 121 dB.Built-in low-frequency roll-off.
KM150	cond	hyper	40-20k	50	10mV	142 0.5	3.6 0.8	2.8	black brass	XLR	795.00	Similar to KM140, but with dynamic range of 124 dB and hyper-cardiod capsule.
RSM-1915	cond	M/S	40-20k	50	23mV	134 0.5	8.4 1.2	6.0	matt	XLR -5	3,050.00	Transformerless M-S/X-Y stereo shotgun mic with active matrix.

Model	Type	Pattern	Frequency Response ±dB	Impedance, Ω	Sensitivity, 1k	SPL at % distortion	Dimensions-in., L, D, W	Weight, oz	Finish	Connector	Price, $	Features
NUMARK ELECTRONICS												
UD940	dyn	card	50-17k	250	76		6.38 2	11.82	black alum.	XLR	154.95	
UD925	dyn	card	60-15k	600	76		6.56 2	10.87	slvr alum.	XLR	70.95	
UC935	elect	card	30-16k	600	68		7.87 0.93	4.75	slvr alum.	XLR	107.95	
UD9200	dyn	card	50-12k	600	74		7 2	16	slvr alum.	XLR	45.95	
PANASONIC/RAMSA												
WM-S1	cond	card	50-18k	600		148	0.5 1.93	0.5	black alum	XLR	210.00	Miniature condenser includes acoustically damped mounting system.Optimized for cymbals,hi-hats and strings.
WM-S2	cond	card	120-15k	250		138	0.5 1.93	0.5	black alum	XLR	170.00	Miniature condenser similar to WM-S1, but optimized for toms and brass.
WM-S5	cond	card	70-16k	600		158	0.5 1.93	0.5	black alum	XLR	280.00	Miniature condenser similar to WM-S1,but optimized for brass,snare and other high-SPL sources.
WM-S10	cond	card	120-15k	250		138	0.5 1.93	0.5	black alum	XLR	220.00	Headset mounted miniature condenser. Ideal for vocalists who play drums or keyboards. Also good for flutes and harmonicas.
PASO SOUND PRODUCTS												
M501	dyn	card	50-15k	250	20		6.2 1.7	15	grey	XLR	90.00	Unique dual shock mount to eliminate handling noise. Non-reflective finish.Includes 18ft. cable.
M601	dyn	card	50-15k	250	20		6.5 1.7	9	grey	XLR	104.00	Same features as M501.
M701	dyn	card	40-16k	250	20		6.5 2.0	11	grey	XLR	134.00	Same features as M501.
M800	dyn	card	40-18k	250	20		6.5 2.0	11	grey	XLR	160.00	Same features as M501.
M506-U	dyn	card	50-15k	250	20		6.25 1.75	15	grey		96.00	Gooseneck mount.Also available with gooseneck and flange as Model M506GX, $106.00.
PEAVEY ELECTRONICS												
PVM 520TN	dyn	card 3	45-19k	400	52 1	145 1.95	4.8	10 alum.	black	XLR	299.99	Yoke mount with three swivel points.
PVM 580TN	dyn card	hyper 3	60-16k	400	52 1	140 1.87	5.75	9.25 zinc	black	XLR	219.99	Titanium laminated diaphragm.
PVM 535N	dyn	card	40-16k 3	400	52	140 1	5.87 1.87	9.25	black zinc	XLR	219.99	New lightweight diaphragm.
PVM 38	dyn	card	60-15k 3	400	56	140 1	5.75 1.87	14	grey	XLR	199.99	The PVM workhorse.
SENNHEISER ELECTRONIC CORPORATION												
MD431	dyn	super	40-16k	200	57	120 1.0	7.8 1.2	8.8	black alum	XLR	429.00	High gain-before-feedback, handles high SPL. Triple-layered steel mesh grill. Magnetic reed on/off switch.
MKE 4032	elec cond	super	70-20k	200	46 1.0	140 1.9	8.1	7.5	black alum	XLR	595.00	12-48 V phantom or AA battery operation. Built-in blast filter, shock mount. Rugged, handles high SPL.
MKH20	cond	omni	25-20k	150	32	142 0.5	6.0 1.0	3.6	black alum	XLR	925.00	High frequency boost switch, and 10dB pad. Ideal for concert, acoustic and M-S recording.
MKH40	cond	card	40-20k	150	32	142 0.5	6.0 1.0	3.6	black alum	XLR	925.00	Versatile mic with switchable 10dB pad and bass attenuation. Transparent response.
MD518	dyn	card	50-16k	200	58	120 1.0	7.0 1.2	6.5	black alum	XLR	219.00	Versatile hand-held microphone. Uses include vocals, rack toms and sax.
ME80	elec cond	shot	50-15k	130	46		12.3 0.8	12	slvr alum	XLR	239.00	On-camera microphone or handheld interview mic for ENG/EFP. Increased gain-before-feedback for podium or lecture use.
MKE-2	elec cond	omni	40-20k	200 1000	46	126	.43 1.0	0.1 .23	black flesh	XLR pig	254.00	Two impedance options, fleshtone color option. Small, ultra light for broadcast, church and theatre in wireless system.
MD421	dyn	card	30-17k	200	54	175 1.0	8.4 1.2	13.6	black	XLR	399.00	Versatile, durable. Handles high SPL. Five position roll-off provides equalization up to 1000Hz.
SONY PRO-AUDIO —See our ad on page 10 and 11												
C-48	cond	multi	30-16k	150	39	128 1.0	2.2 9.1	20	satin nickel	XLR	1,050.00	Selectable patterns,10 dB pad,lo-cut switch, 9 V battery or phantom power. Vibration-proof structure.
C-535P	cond	card	30-16k	200	40	138 1.0	0.8 6.1	4.9	black alum	XLR	495.00	Slim-line design with 10 dB pad. Rejects SCR, TV and other electronic noise. Excellent transient response.
ECM-MS5	elec cond	stereo card	70-20k	150	40	130 1.0	1.9 8.4	7.6	alum	XLR -5	1,250.00	Three capsule design for M-S recording. Built-in M-S matrix field-rugged construction. 12-48 V phantom powered.
F-730	dyn	card	50-11k	300	60		1.7 6.5	8.8	black alum	XLR	120.00	For vocal recording, offers extra punch in low range.

Model	Type	Pattern	Frequency Response ±dB	Impedance, Ω	Sensitivity, 1k	SPL at % distortion	Dimensions-in., L, D, W	Weight, oz	Finish	Connector	Price, $	Features
SHURE BROTHERS, INC. —See our ad on Cover IV												
BETA58	dyn	super card	50-16k	150 290	71.5		6.38 2	9.3	slvr blue	XLR	258.00	Three-stage directional tuning network,advanced shock isolation system,humbucking coil,rugged steel grille.
BETA57	dyn	super card	50-16k	150 290	71		6.18 1.5	9.2	slvr blue	XLR XLR	258.00	same as the BETA 58 but designed for musical instrument mic'ing,features smooth wide response.
VP88	stereo cond	multi	40-20k	150 100	66	129 1	11.43 1.56	14.7	black alum.	5-pin XLR	995.00	MS stereo, mono compatible,built-in left-right stereo matrix adjustable side level switch, internal or phantom power.
SM7	dyn	card	40-16	150 150	57		5.8 3.8	27 7.5	grey	XLR alum	550.00	Independently switchable bass roll-off and presence-boost switches. Internal air suspension. Heavy gauge storage case included.
SM81AC	cond	card	20-20k	150 85	40	146 1.0	8.3 0.9	8	steel chpgn steel	XLR	398.00	10 dB attenuator, 3-position bass roll-off. Omnidirection cartridge(R104) available. Pop filter, swivel adaptor included.
SM91	cond	hemi	20-20k	150 90	45	144 0.1	0.6 3.7 5.0	9.3	black cast steel	XLR	310.00	Low profile boundary effect mic. External pre-amp with12 dB per octave roll-off switch. Accepts battery ot phantom power. Omni capsule available in SM90.
SM98	cond	card	40-20k	150	54	153 0.1	1.2 0.5	0.4	black brass	XLR	250.00	Full-range response in miniature unit. Optional A98SPM adaptor for super-cardioid pickup. Many optional mounting accessories.
SM99	cond	super	80-20k	150	48	130 1.0	1.2 14.7	5.8	black steel brass	XLR	240.00	Miniature gooseneck microphone with lownoise pre-amp built into base. RFI protection.Surface or stand mounting. 5-52 V phantom power. 102 dB dynamic range.
TELEX COMMUNICATIONS												
TE10	cond	card	30-20k	150-200	75	140		7.4	matt black	XLR	135.00	Natural sound, suspended by flexible "fingers" which isolate it from mechanical vibration.
TD11	dyn	card	50-16k	100-250	77			9.2	matt black	XLR	115.00	Die-cast case, reinforced steel mesh windscreen Multi-stage pop filters.
LM-100	cond	omni	20-20k	150	74		.75 0.4	1.0	matt black	XLR	220.00	Lapel mic system includes LM101 mic with 3-foot cable, and PS-10 power supply. Various mounting clips.
LM-300	cond	card	100-15	150	82		0.7 0.3	1.0	matt black	XLR	220.00	Lapel mic system includes WLM60 mic with 3-foot cable, and PS-10 power supply. Various mounting clips.
XCEL DYNAMICS CORP.												
512Li	dyn	card	80-16k	150	76		6.8 2.1	8.4	grey metl.	XLR	120.00	High sensitivity unidirectional mic, riveted mic ball. Also available as 512XL, $80.00.
516XL	dyn	card	80-12k	600	70		7.1 2.1	9.8	black	XLR	60.00	Suitable for general sound reinforcement, buillt-in pop filter.
YAMAHA MUSIC CORPORATION												
MH100	elec	card	10-10k	1.6k	70					phone jack	59.00	Headset and microphone in one unit. Uses lightweight pads that are easy on ears even with extended use.
MZ101	dyn	card	40-17k	250	76		6.2 0.9		mett brown	XLR	110.00	Noted for it's clean mid-range and high-end. Poly-laminate diaphragm. Unique 3-point suspension. Gold-plate connectors.
MZ 102BE	dyn	card	40-18k	250	76		6.2 0.9		mett brown	XLR	220.00	Deep,lower mid-range quality. Beryllium diaphragm for tight reponse. Die-cast zinc body. Gold-plated connectors.
MZ 103BE	dyn	card	40-18k	250	76		6.1 0.9		mett grey	XLR	140.00	Wide range mic with resistance to off-axis sound. Beryllium diaphragm, 3-point suspension and gold plated connectors.
MZ104	dyn	card	30-17k	250	77		7.0 1.4		mett brown	XLR	100.00	Good instrument mic. Good bass response. Lowered sensitivity to avoid hig SPL overload.Gold-plated connectors.
MZ 105BE	dyn	card	40-18k	250	77		6.0 1.4		mett brown	XLR	120.00	Designed to avoid unwanted bass buildup with close mic'ing. Beryllium diaprhrgm. Gold-plated connectors.
MZ106B	dyn	card	40-18k	250	77		7.2 2.0		mett grey	XLR	110.00	Ideal for vocal use. On/off switch with switch lock for lock-On. Two-layer laminated polyester film diaphragm.
MZ 205BE	dyn	card	40-18k	250	77		4.3 1.3		mett grey	XLR	250.00	Vocal microphone with right-angle XLR connector. Beryllium diaphragm. 3-point suspension. Gold-plated connectors.

Wireless Microphone Systems

AKG ACOUSTICS

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

Price:

$3,000.00 to $5,000.00

AUDIO TECHNICA —See our ad on Cover II

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

Price:

$650 and up depending on mic element.

COUNTRYMAN ASSOCIATES

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

Price:

On request

HM ELECTRONICS

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

Price:

$70.00 to $144.00

NADY SYSTEMS

Numerous wireless configurations are offered featuring proprietary system with optional mic heads from Shure, AKG, E-V and Audio Technica. 1200HT (top-of-the-line model) is a true diversity system allowing up to twenty units to work on high-band frequencies, utilizing built-in companding noise reduction. Other systems are of medium price and also can be configured for lavalier systems.

Price:

Dependent on model

PASO SOUND PRODUCTS

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

Price:

$700 to $1,380.00

SAMSON TECHNOLOGIES CORP.

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

Price:

$2,195.00

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

Price:

$1,995.00

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

Price:

$1,225.00

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

Price:

$1,050.00

SENNHEISER ELECTRONIC CORPORATION

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

Price:

$2,940.00

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

Price:

$3,515.00

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

Price:

$3,330.00

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

Price:
$3,905.00
UHF2H is similar to VHF2H system, but utilizing UHF carrier.
Price:
$6,430.00
UHF2B is similar to VHF2B system, but utilizing UHF carrier.
Price:
$7,105.00
UHF2EH features hand-held microphone transmitter SKM4031TVH, and diversity receiver EM2003TVH, in a portable canvas bag with battery. Utilizes UHF carrier.
Price:
$6,070.00
UHF2EB features body-pac transmitter SK2012TVH, lavalier mic MKE2, and diversity receiver EM2003TVH in a portable canvas bag with battery. Utilizes UHF carrier.
Price:
$6,876.00

SONY PRO AUDIO —See our ad on page 10 and 11

Sony has a large variety of both non- and full-diversity tuners, synthesized transmitters and microphone heads for every type of professional application.
Price:
on request

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

Model LS13 consists of one each of the L1 Body Pack Transmitter, L3 Receiver, and WA300 Instrument Cable. Adjustable squelch, ¼-in phone connectors.
Price: with 839W omni lavalier—$445.00
Model LS14/839 is a low-cost diversity system. The receiver uses a proprietary diversity system that monitors both RF signals and combines them for increased gain and improved reception.
Price:
with 839W omni lavalier—$580.00
LS24/96 is a professional diversity system featuring the L4 MARCAD receiver and the L2/96 transmitter. The Shure SM96 electret condenser is included. Mic cartridges are interchangeable when an SM58 head is required by the application.
Price:
$707.00
LS24/Beta 58 is similar featured to the unit directly above but comes with a Shure Beta 58 cartridge for smoothest, super-cardioid characteristics.
Price:
$748.00

TELEX COMMUNICATIONS, INC.

HT-400 is a two channel wireless microphone with integral transmitter and antenna. Models available with microphone heads: Telex TE-10 condenser, Shure SM-87 condenser or Shure SM-58 dynamic. Format is interchangeable.
Price:
$1,020.00 to $1,1250.00 depending on head.
$885.00 to $1,095.00 (depending on head)
HT-100 is a single channel wireless microphone with integral transmitter and antenna. Includes on/off switches for both audio and mic. Includes models with Telex TE-10 condenser, Telex TD-11 dynamic, Shure SM-58 dynamic and Shure SM-87 condenser heads.
Price:
$370.00 to $620.00
$370.00 to $620.00 (depending on head)

VEGA (A MARK IV COMPANY)

R-42 hand-held system consists of the R-42 Pro- Plus true-dual-diversity receiver and the T-86 Pro Plus transmitter. The system features 108 dB (typical) signal-to- noise ratio and wide RF dynamic range. 16 poles of IF filtering give high adjacent channel rejection. Transmitter features an EV N/D757 element.
Price:
$3,198.00
R-33B Pro plus portable system consists a miniature portable receiver and portable transmitter. The receiver is ideal for camera mounting.
Price:
$1,299.00
66B and 67B Pro Plus portable system consists of the 67B Pro Plus true-diversity receiver and the body pack transmitter. The 67B operates on four 9-volt alkaline batteries or external power from a 12-volt camera belt pack or other +10.5 to 18 volt d.c. source.
Price:
Non-diversity—$1,092.00, Diversity—$1,635.00
Pro 2HE hand-held diversity system consists of the R-32 Pro true-dual-diversity receiver and the T-36 Pro hand-held transmitter. System operates on any crystal-controlled frequency from 150 MHz to 216 MHz, at a range of up to 1200 feet.
Price:
$2,110.00
Pro Series body pack system consists of a non-diversity receiver and the body pack transmitter. Has DYNEX II audio processing. Transmitter accepts virtually all electret lavalier mics. Price:
$1,866.00
Ranger true diversity systems include eight body pack and hand-held configurations, including systems for applications not requiring diversity. All feature wide dynamic range and operate on VHF high-band frequencies.
Price:
$1,089.00 to $1,335.00

Magnetic Tape

AGFA CORPORATION

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

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

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

PE619i and 919i is designed specifically for high-speed cassette duplication, providing excellent durability under the stress of bin-loop applications. Also features excellent high- frequency retention after numerous passes and extremely low print-through. Available in ¼-in., ½-in. and 1-in. widths.

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

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

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

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

R-DAT digital audio tape conforms to all published standards for rotary head recording as well as real-time duplication.

AMPEX CORPORATION

1989 Information

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

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

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

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

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

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

BASF

LH Extra I cassette tapes utilize high-performance ferric tape and are available in C-60,C-90 and C-100.

LH Maximag I cassette tapes utilizes a double-coated ferric-cobalt tape with enhanced low and high frequency MOL values. Available in C-60 and C-90.

Chrome Extra II cassette tapes are pure chrome tape with extra high and low MOL, low distortion and noise. Available in C-60, C-90 and C-100.

Chrome Maxima II cassette tapes have a double-coated, high-density coating for extra dynamic range. Available in C-60 and C-90.

TDK

SM Sound Master are premium quality Type II music cassettes available in 10,- 20,-, 30- and 60-minute lengths.

AM Acoustic Master are audio-visual cassettes in 30,- 46,- 60,- 90,- and 120-inch lengths.

AL Acoustic master are instant starting cassettes with a pure ferric coating, jam-proof casing. Available in 60-minute and 90-minute lengths.

ZM Duplicate Master unlabeled audio cassettes are Type I and with jam-proof mechanism. 30,- 46,-, 60,- 90, and 120-minute lengths.

3M

250 Audio Mastering Tape incorporates a 1.5 mil thick back- coated polyester backing. Delivers high output/low noise performance with the widest possible dynamic range of analog mastering tapes. Ideal for high quality music mastering. Available in ¼-in., ½-in., 1-in. and 2-in. widths. 806 Audio Mastering Tape has a 1.5 mil thick backcoated polyester backing. Developed to give good compromise in print-through and maximum output level characteristics. Best tape for applications where both music and speech are being recorded. Available in ¼-in., ½-in., 1-in. and 2-in. widths.

808 Audio Mastering Tape has a back-coated polyester base 1.5 mil thick. Has extremely low print-through characteristics. Ideal for speech, sound effects, and other applications where low print-through is required. Available in 1/4-in. width only.

AUD Digital Audio Cassettes are engineered to deliver state- of-the-art performance in the production of CDs, record albums and cassettes. Available in 30, 60, and 75 minute lengths these units incorporate anti-static system. Available in both the standard album box or exclusive hanger/shipper box.

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

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

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

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

Tape Accessories

POLYLINE CORPORATION —See our ad on page 14

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

Prices:

Bulk prices available on all products

SHELEX ENTERPRISES —See our ad on page 31

CAT is a heavy duty hand-operated quarter-inch tape splicer that is fully automatic . It is a standard of the BBC.

Price:

$168.00

LYNX is pocket-sized version of the CAT splicer for quarter-inch or cassette tape.

Price:

$81.00

TAC is a plastic half-inch splicer for audio or video tapes.

Price:

$31.12

Quarter-inch splicing tape for the CAT are BBC standard Formula 250/Cartridge. A box of ten cartridges is $76.00

Quarter-inch or cassette for the LYNX are 125/cartridge, a box of ten cartridges for $58.00.

Half-inch for the TAC digital or audio thickness 100/cartridge, five cartridge box for $36.90.

TENTEL CORPORATION

1989 Information

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

Price:

$325.00

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

Price:

$725.00

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

Price:

$345.00

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

Price:

$49.00

XEDIT CORPORATION

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

Price:

$50.00

S-2 Editall Compact ¼-in. splicing block provides two cutting angles.

Price:

$36.00

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

Price:

$50.00

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

Price:

$65.00

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

Price:

$160.00

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

Price:

$145.00

EC-D1 fits all 1-in. digital formats such as Otari and Mitsubishi.

Price:

$300.00

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

Price:

Depending on size and quantity

TAPE RECORDERS

FOSTEX

X-26 is a 6-channel/4-track cassette mixer/recorder. It is a 2-head machine running at 1-7/8 in./sec. with a frequency response of 40 Hz to 12.5 kHz, a flutter rate of 0.15%, and a S/N ratio of 58 dB (A-weighted).

Price:

$449.00

160 is a 4-channel/4-track cassette mixer/recorder. It is a 2-head machine running at 3-3/4 in./sec. with a frequency response of 40 Hz to 14 kHz, a flutter rate of 0.1%, and a S/N ratio of 70 dB. THD is 1.5% at 1 kHz.

Price:

$ 840.00

260 is a 6-channel/4-track cassette mixer/recorder. It is a 2-head machine running at 3-3/4 in./sec. with a frequency response of 40 Hz to 14 kHz, a flutter rate of 0.1%, and a S/N ratio of 70 dB. THD is 1.5% at 1 kHz.

Price:

$1,295.00

R8 is a 8-channel/8-track reel-to-reel recorder. It is a 2-head machine running 1/4-in. tape at 15 in./sec. with a frequency response of 40 Hz to 18 kHz, a flutter rate of 0.06%, and a S/N ratio of 78 dB. THD is 1.0% at 1 kHz.

Price:

$2,800.00

E2 is a 2-channel/2-track reel-to-reel recorder. It is a 3-head machine running 1/4-in. tape at 15 in./sec. with a frequency response of 30 Hz to 20 kHz, a flutter rate of 0.05%, and a S/N ratio of 74 dB. THD is 1.0% at 1 kHz.

Price:

$3795.00

E8 is an 8-channel/8-track reel-to-reel recorder. It is a 2-head machine running at 15 in./sec. with a frequency response of 40 Hz to 18 kHz, a flutter rate of 0.05%, and a S/N ratio of 80 dB. THD is 1.0% at 1 kHz.

Price:

$4,495.00

E16 is a 16-channel/16-track reel-to-reel recorder. It is a 2-head machine running 1/2-in. tape at 15 in./sec. with a frequency response of 40 Hz to 18 kHz, a flutter rate of 0.05%, and a S/N ratio of 80 dB. THD is 1.0% at 1 kHz.

Price:

$7,995.00

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

Price:

$7,995.00

MITSUBISHI/NEVE

X-880 is a 32-channel digital recorder with a frequency response of 20 Hz to 20 kHz (+0.5/–1.0 dB). Running at 30 in./sec. (±10%), it uses 1-in. tape and weighs approximately 550 lbs. Its power consumption is 2.0 kVA.

Price: $157,500.00

X-86 and X-86HS are both 2-channel digital recorders which utilize sampling at 96 kHz. X-86C has a frequency response of 20 Hz to 20 kHz, +0.5/–1.0 dB. X-86HS has a frequency response of 20 Hz to 40 kHz (+1.0/–3.0 dB). Both use 1/4-in. tape, weigh about 220 lbs., and consume about 450 VA.

Price: X-86—$17,300.00, X-86HS—$33,000.00

OTARI CORPORATION

DTR-900-32 is a 32-channel digital tape recorder utilizing 1-in. tape with an up to 14-in. reel-size. It has 4 heads, 0 wow and flutter, and a frequency response of 20 Hz to 20 kHz (+0.5 dB/–1.0 dB). It has a 9600 Hz PLL capstan motor and 2 servo 1/2 H.P. DC reel motors.

Price:

$150,000.00

MTR-100A is a 24-track recorder utilizing 2-in. tape. It features automatic alignment, a quartz PLL DC brush-type, direct-drive capstan motor. Wow and flutter at 30 in./sec.. is 0.04%, frequency response is 50 Hz-25 kHz, ±2 dB, and S/N ratio is 70 dB at 1040 nWb/m.
Price:
$59,950.00

MX-55N is a 2-channel, 4-head compact recorder utilizing ¼-in. tape. It features a DC servo-controlled capstan, a frequency response of 30-22 kHz (±2 dB) at 15 in./sec. and an unweighted S/N ratio of 69 dB at 1040 nWb/m. Mic input impedance is 10 k ohms.
Price:
$3,895.00

MX-70 can be variously configured as an 8-track, 8 pre-wired for 16, and 16-track recorder using 1 in. tape. All feature DC servo reel motors and brushless a DC capstan motor (crystal referenced). Frequency response is 50-22 kHz (+2/–3 dB). Unweighted S/N is 70 dB.
Price:
$17,200.00 to $21,650.00

MX-80 comes in both 24 and 32-track versions. Both utilize 9600 Hz PLL capstan motor, microprocessor controlled, 2 servo ⅓ HP DC reel motors, and 3 heads. Signal to noise at 30 in./sec. is 67 dB (1040 nWb/m), frequency response is 60 Hz to 22 kHz (±2 dB).
Price: $33,850.00 to $39,150.00

MK-III-8 is an 8-channel recorder utilizing 1/2-in. tape. It has a DC servo-controlled capstan motor, 2 induction reel motors and 3 heads. Wow and flutter measures 0.04% at 15 in./sec. Frequency response is 40 Hz-22 kHz (±2 dB).
Price:
$5,495.00

MTR-90-II is available in various configurations: 1 in. 8- channel, 2 in. 16-channel, 2 in. 16-channel pre-wired for 24 -channel and 2 in. 24 channel. Frequency response at 250 nW/m is 45 Hz-29 kHz (30 in./sec.). S/N ratio at 1240 nW/m is 78 dB (30 in./sec.) All configurations are 3-head machines.
Price:
$39,950.00

SONY PRO AUDIO —See our ad on page 10 and 11

APR-24 is a 24-channel analog recorder utilizing 2-in. tape. It features amorphous steel heads and "DC constant tension design." Frequency response at 30 in./sec. is 48 Hz - 25 kHz (+0.75 /–3.0 dB). S/N is 70 dB at 30 in./sec. and 66 dB at 15 in./sec. Includes remote control with stand.
Price:
$45,500.00

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

PCM-2500 is a 2-channel, 2-head DAT recorder whose running speed is 8.15 mm/s. It has a S/N ratio of 90 dB and 0.05% THD. Frequency response is 2 Hz to 22 kHz. It is driven by a servo type motor, has LED record indicators and measures 17 x 4 x 16.6 in, weight: 24 lbs.,11 oz.
Price:
$3,550.00

PCM-2000 is a 2-channel, 2-head DAT recorder whose running speed is 8.15 mm/s. It has a S/N ratio of 90 dB and 0.07% THD. Frequency response is 2 Hz to 22 kHz. It is driven by a servo type motor, has LCD record indicators and measures 8.4 x 3 x 7.6 in, weight: 8 lbs.,13 oz.
Price:
$5,000.00

TCD-D10 PRO is a 2-channel, 2-head DAT recorder whose running speed is 8.15 mm/s. It has a S/N ratio of 85 dB and 0.08% THD. Frequency response is 20 Hz to 20 kHz. It is driven by a servo type motor, has LCD record indicators and measures 10 x 2.3 x 7.6 in, weight: 4 lbs.7 oz.
Price:
$3,550.00

STUDER/REVOX —See our ad on page 12

A820 series includes 24-, 16-, and 8-track analog recorders which feature automatic and simultaneous audio alignment for all channels with alignment parameters stored in non-volatile memory. Has a "menu-programmable" transport and optional Dolby SR or Telcom C4 noise reduction.
Price: $64,900.00 (24-track w/o noise reduction.)

A827-MHC is a 24-track microprocessor controlled recorder featuring phase compensated MDA controlled amplifiers with switchable Dolby HX Pro and menu-programmable transport functions. Parallel and serial RS232/422 ports for easy integration into editing systems.
Price:
$47,900.00

A820 2-track master recorder featuring the same processor control for audio and tape transports and the same transport and drive assembly as the A820-24. Frequency response is 40 Hz to 22 kHz (±2 dB).
Price:
$15,900.00

D820X DASH format 2-channel digital audio recorder offers "twin recording" at 15 in./sec. Transport design features DC-driven spooling and capstan motors. Frequency response is 10 Hz to 23 kHz (±0.4 dB.)
Price:
Available upon request.

A812 2/2 Time-Code (option) VUK is a compact recorder especially suited for broadcast application. Featuring the same processor control

for audio and tape transport as the A 820, it is available with or without overbridge. Unweighted S/N ratio is 70 dB.
Price:
$14,750.00
A807 series of 2- and 4-channel recorders are especially suited for broadcast and post-production environments. Features include: tape shuttle wheel, reverse play, right hand edit, tape dump, multi-function tape time and autolocator with programmable "soft-keys."
Price: $7,995.00 (2 channel)
Revox Pr99 MK3 is a 2-track production recorder featuring a real-time counter that reads both plus and minus time, autc search-to-cue for any preselected address, and auto repeat for continuous replay. Also available in "playback only" configuration.
Price:
$2,495.00
Revox C270 series 2-, 4-, and 8-channel recorders all feature microprocessor-based control logic (including precise search- to-cue), and "one-hand" editing under full servo control. Dolby HX PRO and RS 232 interface is standard in this series.
Price:
From $2,995.00 (2-channel)

TASCAM PROFESSIONAL DIVISION (TEAC CORPORATION OF AMERICA)—See our ad on pages 6 to 9
MSR-16 is a 16-channel, 16-track recorder/reproducer using ½-in. tape. It is a 2-head machine featuring a phase-lock looped DC drive capstan motor with a ceramic shaft. Tape speed is 7.5 and 15 in./sec. S/N measures 108 dB A-weighted with built-in dbx Type I NR.
Price:
Available upon request.
ATR-60/16 is a 16-channel, 16-track recorder/reproducer using 1-inch tape format. It is a 3-head machine featuring a phase-lock looped D.C. drive capstan motor with a ceramic shaft. Tape speed is 7.5 and 15 in./sec. S/N measures 108 dB A-weighted with built-in dbx Type I NR.
Price:
$15,999.00
MSR-16 is a 16-channel, recorder/reproducer using half-inch tape. It is a 2-head featuring a phase-lock looped D.C. drive capstan motor with a ceramic shaft. Tape speed is 7.5 and 15 in./sec. S/N measures 108 dB A-weighted with built-in dbx Type I NR.
Price:
Under $7,499.00
ATR-80 series include a 24-track, 2-inch tape machine and a 32-track 2-inch tape machine. Both are three-head units with phase locked loop capstan drive and frequency response of 450 25 kHz ±2 dB at 30 in/sec and 35-20 kHz ±2 dB at 15 in./sec.
Price: ATR-20 24-track—$34,999.00, ATR-20-32-track—$44,000.00
DA-800/24 is the DASH digital 24-track recorder. There are also 2 analog tracks, 90 dB signal-to-noise, and switchable sampling rates of 48, 44.1, and 44.056. The machine is fully compatible with the Sony 3324/3324A series machines.
Price:
$99,000.00
DA-30 is an R-DAT recorder that records at sampling rates of 48, 44.1 and 32 kHz with a better than 94 dB signal-to-noise ratio. A/D convertors: 64 k oversampling, 1-bit; delta-sigma D/A convertors, and 18 bit 8X oversampling.
Price:
$1,,899.00
MSR-24 is a one-inch 24-track with built-in dbx Type 1 noise reduction. Gapless and noiseless punch in and out is included, Response is 40-20 kHz ±3 dB at 15 in/sec and 40-16 kHz ±3 dB at 7.5 in/sec. A full-function remote is an available option.
Price:
$13,999.00

UHER OF AMERICA
CR1600 is a portable stereo cassette 4-track recorder. Features include: auto-reverse operation, 2 speeds, 3 heads, Dolby B, switchable ALC, full remote control, built-in voice activation system. Dimensions: 9 x 2 x 7 in, weight: 7 lbs.
Price:
$1,899.00
4400 Report Monitor is a portable open-reel 2-track stereo recorder. Features include: 4 speeds, 3 heads, belt drive, mic inputs (200 ohms), switchable ALC and LED function indicators. Dimensions: 11 x 3.5 x 9, weight: 3 lbs.
Price:
$1,867.00

YAMAHA
C300 is a "professional quality" 2-channel, 3-head cassette recorder featuring 12-layer laminated amorphous heads, a double-gap ferrite erase head and a closed-loop dual capstan transport. With dbx noise reduction "on," S/N is 95 dB.
Price:
$1,095.00
MT100II is a 4-channel/4-track cassette mixer/recorder with self-contained mixdown capability. Motors are DC servo type and top speed frequency response is 40 Hz-18 kHz. With dbx noise reduction "on," S/N is 85 dB.
Price:
$495.00
MT3X is a 6-channel/4-track cassette mixer/recorder with self-contained mixdown capability, "comprehensive" monitor system and programmable auto punch-in. Motors are DC servo type and top speed frequency response is 40 Hz -18 kHz. With dbx noise reduction "on," S/N is 85 dB.
Price:
$995.00
DRU8 has a 20-bit 8-track S-DAT recorder, an 8 X 2 digital mixer with panning for monitoring, digital crossfade punch-in with adjustable crossfade time, synch via SMPTE (all formats) or MIDI. Selectable sampling frequencies of 44.1 or 48 kHz.
Price:
Available upon request.

ADDRESSES

Agfa-Gevaert
275 North Street
Teterboro, NJ 07608

AKG Acoustics
77 Selleck Street
Stamford, CT 06902

Altec Lansing
10500 West Reno
Oklahoma City, OK 73128

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

AMS Industries, Inc.
1180 Holm Road, Suite C
Petaluma, CA 94954

Applied Research and Technology
215 Tremont Street
Rochester, NY 14608

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

BASF Corporation
19 Crosby Drive
Bedford, MA 01730

Beyer Dynamic, Inc.
5-05 Burns Avenue
Hicksville, NY 11801

Bruel and Kjaer Instruments
185 Forest Avenue
Marlborough, MA 01752

Carvin Corporation
1155 Industrial Avenue
Escondido, CA 92025

Countryman Associates
417 Stanford Avenue
Redwood City, CA 94063

Crown International
1718 W. Mishawaka Road
Elkhart, IN 46517

C-T Audio Marketing
3050 SW 14 Place, Suite 3
Boynton Beach, FL 33426

Electro-Voice
600 Cecil Street
Buchanan, MI 49107

Fostex Corporation
15431 Blackburn Avenue
Norwalk, CA 90650

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

HM Electronics
6675 Mesa Ridge Road
San Diego, CA 92121

Mitsubishi/Neve
7 Parklawn Drive
Bethel, CT 06801

Nady Systems
6701 Bay Street
Emeryville, CA 94608

Numark Electronics
503 Newfield Avenue
Edison, NJ 08837

Otari Corporation
378 Vintage Park Drive
Foster City, CA 94404

Panasonic/Ramsa
6550 Katella Avenue
Cypress, CA 90360

Paso Sound Products
14 First Street
Pelham, NY 10803

Peavey Electronics (AMR)
711 A Street
Meridian MS 39301

Polyline Corporation
1233 Rand Avenue
Des Plaines, IL 60016

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

Sennheiser
P.O. Box 987
Old Lyme, CT 06371

SHELEX Enterprises
P.O. Box 3752
Hollywood, CA 90078

Shure Brothers
222 Hartrey Avenue
Evanston, IL 60202-3696

Sony—Pro Audio
1600 Queen Anne Road
Teaneck, NJ 07666

Studer Revox
1425 Elm Hill Pike
Nashville, TN 37210

Tascam-Teac Corp. of America
7733 Telegraph Road
Montebello, CA 90640

TDK Professional Products Division
1411 West 190th Street
Gardena, CA 90248

Telex Communications
9600 Aldrich Avenue South
Minneapolis, MN 55420

Tentel
1506 Dell Avenue
Campbell, CA 95008

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

Uher of America
7067 Vineland Avenue
North Hollywood, CA 91605

Vega
9900 Baldwin Place
El Monte, CA 91731-2204

Xedit Corporation
133 South Terrace Avenue
Mount Vernon, NY 10550

Xcel Dynamics Group
1610 Lockness Place
Torrance, CA 90501

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

Classified

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Quantity discounts are: 3X – 15%; 6X – 30%.

ALL CLASSIFIEDS MUST BE PREPAID

Send copy to:

db, The Sound Engineering Magazine, 203 Commack Road, Suite 1010, Commack, NY 11725.

People, Places... & Happenings

• **Mark IV Audio, Inc.** has acquired a majority interest of **Dynacord GmbH**, a West Germany-based manufacturer serving the music and commercial sound segments of the professional audio market...**Studer ReVox** of Nashville, TN has been very busy sending equipment to studios. **Mobius Music** is the first Bay area studio to install Studer's A827 multi-track recorder, and **Hummingbird Recording**, also of Nashville, has added to its ongoing expansion with the A827. **Studer's Dyaxis Digital Audio Production System** was selected by **Bernie Grundman Mastering** of Hollywood, CA for its sound quality. **Soundworks West** of West Hollywood, CA has completed their renovation with the recent acquisitions of three Studer A820 24-track analog recorders equipped with Dolby SR, an A820 2-TC-VU mastering recorder with time code interface, five A727 professional CD players and one A730 CD system...They're starting 'em early at the **South Mountain Center for the Performing Arts**. The **Phoenix Union High School** district recently acquired **Tascam's** M-600 and M-512 mixing consoles, MS-16 and two 34B open reels for its future audio recording professionals. Also purchasing Tascam equipment are **Dallas Sound Lab** with Tascam's 238 multi-track recorder and **New York Music** with Tascam's one-inch MS-16 recorder/reproducer...**ARX Systems** has sold Quadcomps and/or Sixgates to **New Studio Hawaii** in Honolulu; **Hillview Studios** in Gilroy, CA; **Ground Zero Music** in Indianola, IN; **J R Sound** in Burton, MI; **Motro Sound** in St. Paul, MN; **NO SHOW SOUND** in Los Angeles, CA; **Kean Sound** Engineering in Boulder, CA and **Freeworld Audio** in Buena Vista, FL...Thirty **Dolby** Model 363 SR Two-Channel Frames and an MT Series Multi-Track Unit have been delivered to **Atlantic Video** in Washington, D.C. by **Washington Professional Systems** of Wheaton, MD...**Metro Studio** of Minneapolis, MN has expanded with the addition of a second 24-track studio..

• **DePaul University's School of Music**, of Chicago, has graduated its first class with a Bachelor of Science in Music degree, with an emphasis in Electrical Engineering and Recording and Sound Technology. Besides focusing on core requirements in music and liberal studies, students worked with state-of-the-art equipment at Universal Recording, one of the nation's largest recording studios.

• Similarly, Pennsylvania's **Duquesne University** has announced a new Bachelor of Music Degree with Sound Recording Technology major from its School of Music. The program will combine a traditional music education with hands-on training and theoretical understanding of recording arts and sciences and music technology.

• **Reel-Tek Inc.**, a subsidiary of **JRF Magnetics**, announces its **AGFA Packout Program** for *The Electronic Cottage*, serving especially the 1/4 and 1/2-inch multichannel tape recorders of such companies as **Fostex, Tascam, Otari, etc.** They will repair those machines and align and calibrate them for the PEM 469 tape. The Packout Program includes a free reel of tape with each machine repaired.

• **Music Annex Recording Studios** of Menlo Park, CA, is celebrating its first anniversary of Studio C, a product of the joining of **Dragon Studios** with the Music Annex. Studio C is designed for mixing, overdubbing, MIDI and voice-over work. Studio III, a new audio for video mixing suite, has also been completed. Additionally, **Sound Recording Organization** has joined the Music Annex's San Francisco facility. SRO is the product of a combination of **The Sound Service and Studio C.**

• Promotions are abound at a number of companies including: **Nagra USA**...**Martin Gardner**, who joined **Kudelski S.A.** as general manager in 1989 has been named vice president and elected to the board of directors at **Nagra USA, Inc.**, a subsidiary of Kudelski, S.A...**TDK Electronics** has appointed **Ken Kihara**, a 20-year TDK veteran, to the new position of vice president of marketing...**Paul McGuire** has been named president of **Electro-Voice** of Buchanan, MI. He succeeds **Robert Pabst**, who will continue as president of **Mark IV Audio, Inc**... **Gauss** has appointed **Young Nak So Ri Sa** as exclusive representative in Korea of Gauss high-speed cassette tape duplicating systems and equipment...**Atlas Soundolier** has promoted **Bud Waters** to National Sales manager in charge of outside sales for the sound contractor, music dealer and National Account market-segments; **Walter Best** has been appointed Sales manager, Eastern region and **Jim Edwards** has been appointed Sales manager, Western region with expanded responsibilities in their respective halves of the United States including manufacturers representatives and promotional activities...**Neve** has promoted **Rich Hajdu** to vice president of Sales and Marketing, and **Lisa Vogl** to director of Advertising and Promotions...**Sony Professional Audio Division** has promoted **Clayton Blick** to Marketing manager and Gary Rosen to National Sales Manager...**Chuck Prada** has been appointed Field Sales manager/Music Products Group by **Tascam**, where he will be responsible for overseeing sales activities of the company's regional sales managers, sales representatives and dealers...**Renkus-Heinz** has promoted **Carl C. Dorwaldt** to National Sales & Marketing manager; **Mark Duncan** to Product manager and; **Graeme Harrison** to European Marketing manager for Renkus-Heinz's European Operation...**Illbruck, Inc.**, manufacturer of **SONEX**, has appointed **Eric W. Johnson** as National Sales manager of SONEX Acoustical Products for pro-audio markets.