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*Based on *Music and Sound Retailer's* monthly survey of 1,200 audio dealers

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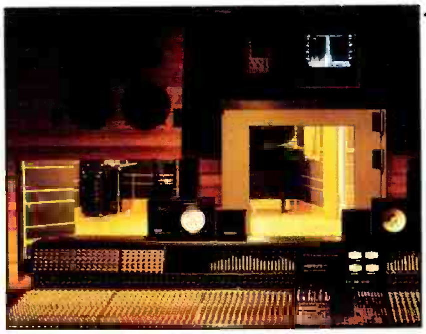

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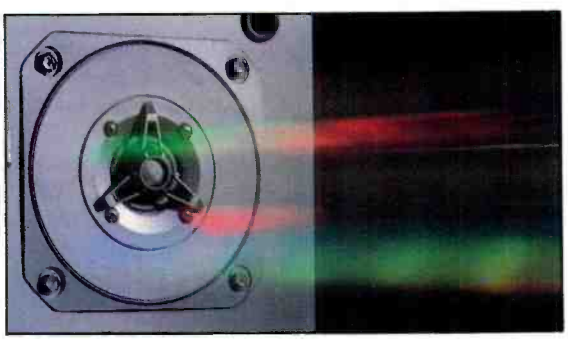

On the Cover: With the increasing move toward CD use, close field monitoring is becoming more important to the control room environment. TOA monitors courtesy of New World Audio, San Diego.

Photo by Gene Faulkner.

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EDITORIAL

Personal Choice Monitoring

There are at least two schools of thought concerning the selection of small monitors to be used in close field applications. A "generic" reference, which is used in many studios, or a "personal choice" reference, which is selected to complement *your* "ear."

Does one approach have an advantage over the other? Perhaps.

The generic or lowest-common-denominator approach is a safe route to follow, and has its advantages. Mainly, a great number of people (engineers and clients alike) are willing to accept these as universal references. It's no great secret that Auratone and Yamaha models have been accepted in this context, which highlights two other advantages: name recognition and the moderate price of most models.

While I have no argument with the engineer who selects these models as their personal choice system, selecting any monitor as a generic reference triggers a basic flaw inherent in how audio professionals approach critical monitoring: too many studios and engineers select their small monitors based on house, city or industry standards.

How can we expect to critically monitor any signal path if the reference point requires a significant amount of guesswork?

The electromechanical process of converting sound waves into electrical current and back into sound waves is a complex and cumbersome process. Because of this, audio engineers have to develop extraordinary "sonic-compensation" skills to deal with the flaws that have always been inherent in the process. This sonic compensation is usually done mentally, but if spoken aloud might sound like "...this monitor is a bit harsh in the 2kHz to 3kHz range (aarrgghh, I hate monitors that have this characteristic), and I know the producer wants the words to be clearly understood, yet I still need to leave some room for the lead vocal. Let's see if I cut a little here and boost a little there...".

During the course of a session, this process can burn up a tremendous amount of mental energy, which is a precious commodity in the engineering community.

Can you imagine a professional photographer working effectively with a viewfinder that didn't match what the camera lens was picking up? It would be extremely difficult to do precise work. The same can be said of the relationship between the engineer and the monitor. In simpler terms...just because you *can* get the shoe on, doesn't mean it fits!

Why own your own?

Choosing monitors can be both the most personal and the most critical purchasing decision facing the professional. Wouldn't it be nice if the next pair of monitors you worked with produced a sonic "picture" of exactly what was going on, as it relates to your own *personal* way of hearing? After all, we all hear things differently, and sometimes those differences are quite dramatic.

Think of personal choice monitors as an extension or continuation of the human hearing mechanism. For most engineers, there will be one brand or monitor model that complements their natural (subjective) hearing better than all others. This concept is not unlike the professional athletes who carefully select equipment to complement their game or particular style of play.

The ideal relationship between engineer and "personal choice" monitor system, should be one in which the engineer doesn't have to mentally compensate for any part of the audible spectrum. Once this relationship is solidified, the engineer can respond immediately and effectively if something needs to be done, and can tell *exactly* how much compensation is required without guessing.

In my active engineering days, I was able to experience this relationship, and can recommend it as a significant step toward a more consistent product overall. During the first few weeks of working with my personal choice monitors I had to consciously *stop* trying to out-guess the new monitors, and start responding directly to what I heard.

The original source is really the only true and accurate representation of any sound. That's not to say that it can't be altered (for better or worse), but even the original source sounds different to each individual. This may be the bottom line on subjectivity or perhaps the original source. This fact is often easily overlooked in the grander *subjective* discussions about what sounds good, what design is flatter or what system has the best overall specs.

Monitor manufacturers aren't trying to make their product sound better than the original source, but through research and development they are trying to design systems that induce an absolute minimum amount of coloration into the reproduction process. As this is accomplished the unit(s) is perceived to be better sounding.

We haven't gotten to the point where we custom-build a monitor to complement an individual's measured hearing response. But, with all the different models available, there is every reason for professional engineers to have and use an off-the-shelf system matched to their ears.

Working with personal choice reference monitors is a fairly new concept, and may require some client education. This shouldn't be too difficult as most experienced clients hire the person rather than the hardware. As an extension of the engineer's talents, the monitors are the most important tool. Also, investigating thoroughly the choices and investing your own money will demonstrate a high level of professionalism. We shouldn't have to apologize for saying "this is the way I hear it best, and this is the way I'm going to be able to give you my best, most accurate work."

Choosing this system doesn't come easily. First, you need the experience to know *what* you want to hear—that's the mental imaging I wrote about in the April *RE/P* editorial. But, once those mental "sounds" are firmly established, you'll be able to critically evaluate monitors in search of a pair that best matches the way you hear the original source. RE/P

By Michael Fay
Editor

TRUTH... OR CONSEQUENCES.

If you haven't heard JBL's new generation of Studio Monitors, you haven't heard the "truth" about your sound.

TRUTH: A lot of monitors "color" their sound. They don't deliver truly flat response. Their technology is full of compromises. Their components are from a variety of sources, and not designed to precisely integrate with each other.
CONSEQUENCES: Bad mixes. Re-mixes. Having to "trash" an entire session. Or worst of all, no mixes because clients simply don't come back.

TRUTH: JBL eliminates these consequences by achieving a new "truth" in sound: JBL's remarkable new 4400 Series. The design, size, and materials have been specifically tailored to each monitor's function. For example, the 2-way 4406 6" Monitor is ideally designed for console or close-in listening. While the 2-way 8" 4408 is ideal for broadcast applications. The 3-way 10" 4410 Monitor captures maximum spatial detail at greater listening distances. And the 3-way 12" 4412 Monitor is mounted with a tight-cluster arrangement for close-in monitoring.
CONSEQUENCES: "Universal" monitors, those not specifically designed for a precise application or environment, invariably compromise technology, with inferior sound the result.

TRUTH: JBL's 4400 Series Studio Monitors achieve a new "truth" in sound with an extended high frequency response that remains effortlessly smooth through the critical 3,000 to 20,000 Hz range. And even extends beyond audibility to 27 kHz, reducing phase shift within the audible band for a more open and natural sound. The 4400 Series' incomparable high end clarity is the result of JBL's use of pure titanium for its unique ribbed-dome tweeter and diamond surround, capable of withstanding forces surpassing a phenomenal 1000 G's.
CONSEQUENCES: When pushed hard, most tweeters simply fail. Transient detail blurs, and the material itself deforms and breaks down. Other materials can't take the stress, and crack under pressure.

TRUTH: The Frequency Dividing Network in each 4400 Series monitor allows optimum transitions between drivers in both amplitude and phase. The precisely calibrated reference controls let you adjust for personal preferences, room variations, and specific equalization.
CONSEQUENCES: When the interaction between drivers is not carefully orchestrated, the results can be edgy, indistinctive, or simply "false" sound.

TRUTH: All 4400 Studio Monitors feature JBL's exclusive Symmetrical Field Geometry magnetic structure, which dramatically reduces second harmonic distortion, and is key in producing the 4400's deep, powerful, clean bass.
CONSEQUENCES: Conventional magnetic structures utilize non-symmetrical magnetic fields, which add significantly to distortion due to a nonlinear pull on the voice coil.

TRUTH: 4400 Series monitors also feature special low diffraction grill frame designs, which reduce time delay distortion. Extra-large voice coils and ultra-rigid cast frames result in both mechanical and thermal stability under heavy professional use.
CONSEQUENCES: For reasons of economics, monitors will often use stamped rather than cast frames, resulting in both mechanical distortion and power compression.

TRUTH: The JBL 4400 Studio Monitor Series captures the full dynamic range, extended high frequency, and precise character of your sound as no other monitors in the business. Experience the 4400 Series Studio Monitors at your JBL dealer's today.
CONSEQUENCES: You'll never know the "truth" until you do.

JBL

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Northridge, CA 91329

Circle (6) on Rapid Facts Card

NEWS

NAMM sets summer dates

The summer NAMM International Music & Sound Expo is scheduled for June 24-26 at the Georgia World Congress Center in Atlanta. The show will run Friday through Sunday, with extended daily hours of 10 a.m. to 6 p.m. Pre-registration for badges should be completed before May 9. NAMM members pre-register free of charge; non-member pre-registration is $25 per person. Those registering on site must provide proof of employment with a retail or supplier company in the music products industry. For more information contact NAMM at 5140 Avenida Encinas, Carlsbad, CA 92008; 619-438-8001.

SPARS hosts business conference

SPARS will host a one-day marketing seminar at the UCLA Graduate School of Management on May 21. The seminar will feature marketing techniques and include information on logos, graphic presentations and demo reels. Speakers will include Nick Colleran, David Porter, John Rosen, Richard Trump and Dwight Cook. For more information contact Shirley Kaye, SPARS executive director, 4300 10th Ave. North, Suite 2, Lake Worth, FL 33461; 305-641-6648.

Obituary: Bob Liftin

Bob Liftin, a prominent audio consultant and engineer, died on Jan. 8. He began his career with CBS as an audio engineer for radio soap operas and live television. In the late 1950s, he launched Regent Sound Studios, New York. He was also a founding member of SPARS, and served as president and chairman of the board.

As a consultant and engineer, he worked with the 1986 Liberty Weekend, Tony Awards, Radio City Music Hall, Live Aid from Philadelphia, the Jerry Lewis Telethon, CBS' New Year's show from the Waldorf Astoria and "Saturday Night Live."

APRS schedules annual exhibition

The 21st annual APRS exhibition will be June 22-24 at the Olympia 2 Exhibition Centre in London. Hours are 11 a.m.-6 p.m. on Wednesday and 10 a.m.-6 p.m. on Thursday and Friday. For more information regarding the exhibition, contact Philip Vaughan, APRS, 163 A High St., Rickmansworth, Hertfordshire, WD3 1AY; 0923-772907.

News notes

New England Digital has announced that sales of the Optical Disk storage and retrieval system has passed the $1 million mark.

Klark-Teknik has acquired Midas Audio Systems Ltd. K-T will market, distribute and service Midas consoles. The Midas name will remain.

Focusrite U.S. has become the North American representative for Quested Monitoring Systems.

Audio Kinetics has been awarded a contract for the supply of three audio post-production control systems for the Finnish Broadcasting Company.

Neve has announced the opening of new facilities in New York and Nashville. The New York office is located at 260 W. 52nd St., Suite 25E, New York, NY 10019; 212-956-6464. The Nashville facility is located at 1221 16th Ave. South, Nashville, TN 37212; 615-329-9584.

ASR Enterprises has become a manufacturer's rep for TOA Electronics, PM&E division.

AMS has delivered AudioFile systems to: Producers Color Service, Detroit; Marsh Films, Los Angeles; Soundelux, Los Angeles; Hit Factory, New York; HBO Productions, New York; Multivision, New York; Frankford Wayne, New York; and MasterMix, Nashville.

RECORDING ENGINEER/PRODUCER is edited to relate recording science to recording art to recording equipment, as these subjects, and their relationship to one another, may be of value and interest to those working in the field of commercially marketable recordings and live audio presentation. The editorial content includes: descriptions of sound recording techniques, uses of sound recording equipment, audio environment design, audio equipment maintenance, new products.

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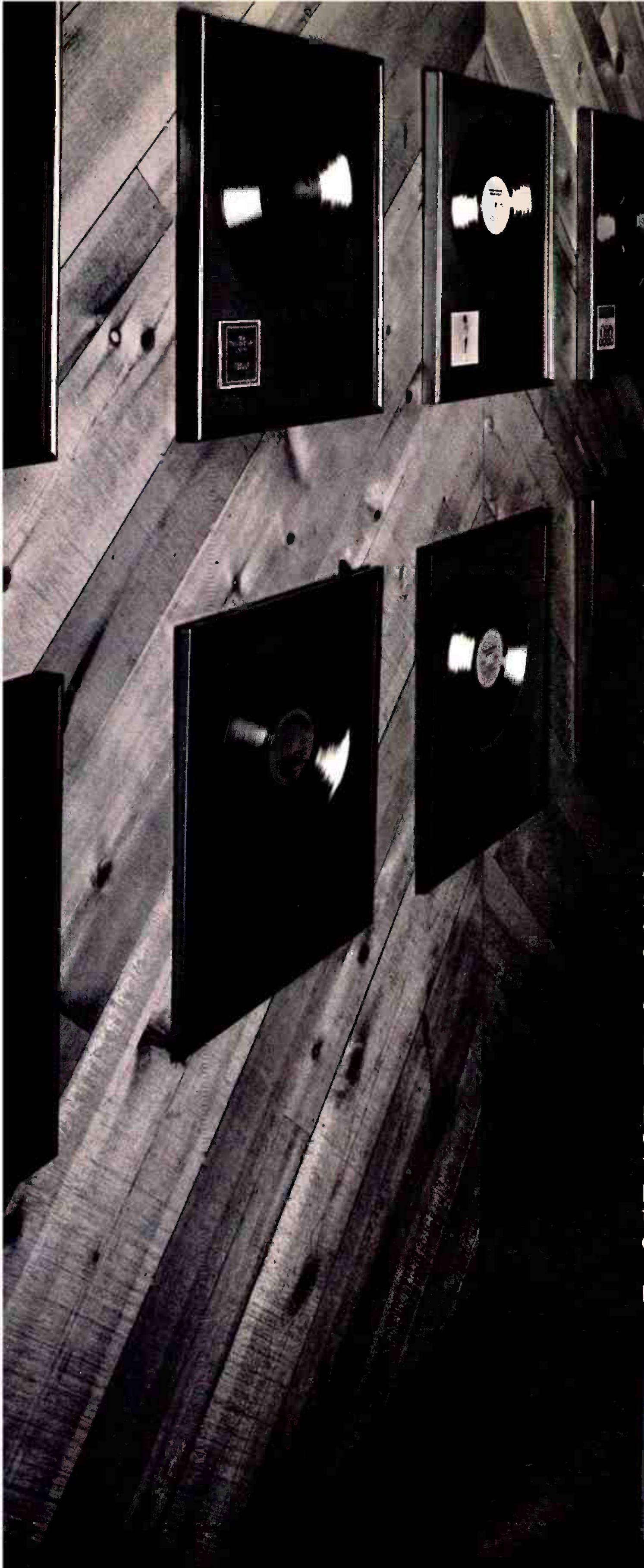

WHAT YOU DO WITH THE M-600 MIXER IS YOUR BUSINESS.

That's why we've designed it to meet or exceed your most demanding requirements. And made it the easiest, most flexible professional mixing console you'll ever work with.

The M-600 is modular. Which means you can custom configure the console to *your* audio or video production needs. The M-600 lets you choose up to 32 input channels, or you can start with 16 or 24 input channels and expand the board as your needs change. Optional stereo modules can also be added to provide even more line inputs for MIDI instruments and video production convenience.

Installation and wiring is exceptionally easy. The M-600 is the only modular mixer that's available with all the necessary finished cables and installation hardware. And that can eliminate a lot of installation hassles and expense. At the same time, no other mixer at its price gives you multi-pin, computer-type connectors for quieter, more secure connections.

But the real pleasures of the M-600 will only be evident after it's in your studio. Up to 64 stereo or 128 mono inputs can be accessed directly from the top panel. A patch bay can be added for fast, flexible routing. That's convenience.

The M-600 has all the features you'd expect in a professional mixing console. Like balanced insert patch points on all inputs, PGM busses as well as the stereo master buss for increased signal processing capability. Plus sweep-type parametric EQ, balanced inputs and outputs, phantom power, talkback/slate channel and all the audio performance you'll ever need. Without the exorbitant price you don't need.

So check out the M-600 modular mixing console. It's ready for fame when you are.

TASCAM

Circle (7) on Rapid Facts Card

NEWS

Apogee to provide filters for Sony digital machines

Apogee Electronics has been given approval by Sony to provide the series 944 anti-aliasing/anti-imaging low-pass filters for the Sony PCM model 3324. After conducting listening and measurement tests, Sony has approved of retrofitting the Apogee filters before delivery or to update machines already in the field.

Harris signs R-DAT OEM agreement with Aiwa

The broadcast division of Harris Corporation has announced distribution of the model XD-001UH Broadcast Use Digital Audio Tape DAT recorder/playback unit for AIWA Co., Ltd. of Tokyo. The standard unit, which will feature the Harris name, features high speed music search, tape scan, forward and back skip, timer recording and playback, high-speed cue/review and a multi-function counter that displays elapsed play time.

Audio recording seminar

The University of Iowa will present the 9th Annual seminar in Audio Recording on June 13-24. Topics will include stereophonic and ambisonic mic techniques, mic comparisons, digital-analog and digital-digital comparisons, noise reduction systems and preparing tapes for CD and LP mastering. Instructors include Jerry Bruck, Robert Ludwig and Lowell Cross. Tuition is $144 for undergraduates and $226 for graduate students. Two semester hours credit is awarded upon completion of the seminar. For more information contact Lowell Cross, Recording Studios, School of Music, the University of Iowa, Iowa City, IA 52242; 319-335-1664.

ITS and NATPE Teleproduction Conference

The International Teleproduction Society (ITS) will present the first International Teleproduction Conference and Exhibition in Los Angeles at the Los Angeles Convention Center, June 25-28. The theme "Teleproduction Today...Teleproduction Tomorrow," represents a joint effort by ITS and NATPE International, the organization of teleproduction programmers and producers. In addition to exhibitions, the conference will feature seminars, workshops and general sessions. For more information contact Susan Stanco, ITS, 990 Sixth Ave., Suite 21E, New York, NY 10018; 212-629-2366.

People

Tracy Crawford has been named professional sales engineer for Klipsch & Associates.

Don Morgan has joined the Mitek Group as product sales specialist.

Ron Ridderhoff, Dave Miller and **Matt Johnston** have formed MDR Sales. The company will represent HM Electronics in Michigan, and is located at 395 E. Elmwood, Troy, MI 48099; 313-585-3616.

Fred Ginsburg has joined Audio Services Corporation's marketing and product development department in North Hollywood, CA.

Spencer Burton has been named technical manager for Alpha Audio.

J. Michael Hughes has been elected vice president of marketing for HM Electronics.

Andrew Murray has been appointed field sales engineer for QSC Audio Products.

Karl Seglins has been appointed general manager of professional audio for Amber Technology Pty Ltd.

Diana Deutsch, a research psychologist at the University of California, San Diego, has been elected a governor of the Audio Engineering Society. She will serve for two years.

WaveFrame has announced appointments in New York, Los Angeles and Boulder, CO. Jon Eganhouse, sales representative, Susan Sloatman, sales coordinator, and Andy West, applications engineer, have joined the Los Angeles office. Liz Lockhart, sales coordinator and Steve Rossi, applications, have joined the New York office. In Boulder, Doug Wood has joined the staff as applications engineer.

Joseph Kempler has joined Sunkyong audio tape division as technical director.

Greg Speer has been appointed vice president and general manager of Electro-Sound, Los Angeles.

Bob Moses has joined the engineering staff of Rane Corporation.

Mark Miller and **Greg LoPiccolo** have joined LaSalle Audio Systems as sales representatives.

Steve Angel has been appointed hire and digital services manager for HHB.

Nancy Westbrook has been appointed regional sales manager for Mitsubishi's, Nashville, TN office. **Bob McNabb** has been appointed regional sales manager for Mitsubishi's, Los Angeles office.

Robert W. Fox and **Jeffrey Eiler** have joined Monfort Electronics Marketing as field sales engineers.

Adrian Weidmann, international manager of Bruel & Kjaer Pro Audio Group will now be based in the United States. The address is 185 Forest St., Marlborough, MA 01752; 617-481-7000.

John D. Johnson has been appointed vice president of software engineering for Innovative Electronic Designs.


AMPEX
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By Paul D. Lehrman

Who Owns MIDI?

A true story: A major company announces a new piece of MIDI-controlling gear. Great fanfare, great anticipation, great advance sales. It hits the market, and every unit in sight is snapped up. Great is their response.

A few weeks go by, and reports start to filter out about problems. Bugs. Inconsistencies. Unpredictable behavior. Non-functioning features.

The manufacturer assures everyone that it must be their imagination because the box was thoroughly tested by the best in the business. Users start to get concerned, then mad.

Finally, the truth comes out. The box was indeed thoroughly tested, but the testers only used MIDI equipment made by the same manufacturer.

All of the problems cropped up when users plugged the thing into equipment from *other* manufacturers. Oops.

As much as some companies might like to think otherwise, nobody owns MIDI, and no one has a lock on *the* right way to use it. There's lots of room in the MIDI spec for interpretation, expansion and just plain fooling around. Choose a task, and often as not, MIDI will give you several ways to accomplish it, whether it's turning a note off or making a crescendo.

If you want to try something new, there are plenty of undefined controllers, as well as "non-registered" or "general purpose" ones, "non-commercial" system-exclusive IDs, and various other undefined messages.

This flexibility is one of MIDI's most important assets, and it's only because MIDI is in the public domain that it is able to remain so flexible.

If some corporate bureaucracy were in charge of it, all those little loopholes would be closed up and MIDI would never change or develop. If it were held by a monopoly, no one would be allowed to experiment with it, and by now, less than five years after its birth, it would be dead, a victim of technological atrophy.

Working with MIDI is a constant balancing act between the tried and the untried.

If the designers of one of the first multitimbral samplers had slavishly followed the specification for MIDI modes, they would never have built an instrument that could handle polyphony on multiple MIDI channels.

But they went ahead and developed another mode. They made sure it didn't conflict with any existing parts of the spec, and now, hardly a synth manufacturer exists that does not use "Multi" mode in at least one of its models. If the makers of MIDI guitars didn't rethink the idea of pitch bend so that distinctions could be made between individual string bends and "whammy bars," and so that the latter didn't hopelessly choke up the MIDI data line, that field would never have gotten off the ground.

If manufacturers weren't willing to bend the definitions of notes and controllers just a little bit, there would be no MIDI-controlled mixers or signal processors.

The most dangerous thoughts any designer can have about MIDI are, "I'm doing it right, and everyone else is wrong." Barring extremely bizarre interpretations of the spec, there really is no right and wrong with MIDI.

As long as you're careful not to let it conflict with anything, you can use MIDI for just about anything you want and in any way you want.

Sometimes you'll want to use running status, sometimes you won't. Sometimes true note-offs will be appropriate, and sometimes note-ons with zero velocity will do the job. Most of the time, one byte of pitchbend will do what needs to be done, but once in a while that second byte that the spec provides comes in handy. Active sensing, which seemed like a good idea for only a very brief period of time, still has some uses.

MIDI is defined not by a piece of paper, but by how it's used. No matter what the rules say, if a manufacturer comes up with a MIDI implementation for a piece of gear that's way out in left field and he sells a lot of them, then he's changed MIDI, and everyone else has to go along.

Sometimes, of course, this goes too far, and when a manufacturer decides he wants to invent 256 additional MIDI channels, or change the clock rate, or throw in six extra bytes to increase a controller's resolution, it's time to call out the MIDI police.

Naturally, this high degree of flexibility increases the risk factor in fooling around with MIDI. Manufacturer A may put a lot of little software tricks and funny commands into his ABC-300 controller that he's convinced are going to save the world, but after he's committed to manufacturing 20,000 of them, he'll discover that Manufacturer B's XYZ-400 processor, which has recently become the hottest thing on the market, dumps all of its memory and shuts itself off whenever it reads data coming from the ABC-300.

It's only because MIDI is in the public domain that it is able to stay so flexible.

Sure, it's impossible to keep up with every little quirk of every piece of equipment out there (although organizations like the International MIDI Association and the Manufacturers' MIDI Association certainly make the effort), but it is any responsible manufacturer's duty to keep track of major developments, to keep those ambiguities in the MIDI spec—both the deliberate ones and the accidental ones—from developing into catastrophes.

Paul D. Lehrman is *RE/P*'s electronic music consulting editor and is a Boston-based electronic musician, producer and free-lance writer.

It's also incumbent upon all manufacturers to let the rest of the world know what they're doing, not to keep new interpretations of the MIDI spec a secret, lest someone else does something radically in conflict with it.

Again, the industry organizations are the best clearinghouse for this kind of information.

For the user, the task is twofold. Not only must you know how to make your equipment do what you need it to, but you have to be sure it does it in conjunction with everything else you've got in the studio. A digital processor that responds to MIDI controllers is going to be of very little use if you have a sequencer whose only editing function over controllers is to erase them.

MIDI is defined not by a piece of paper, but by how it's used.

The world's most sophisticated SMPTE-to-MIDI convertor will be wasted if it's used to drive a drum machine that can't read Song Position Pointer. And using a synthesizer that only transmits on channel 1 as a master keyboard is asking for trouble. None of these products violate the MIDI spec, but they can sure make life difficult if you try to make them work together.

Working with MIDI is a constant balancing act between the tried and the untried. There's always something new to be attempted, some way to bend or stretch the MIDI spec to accomplish a new level of expression or to make a task go faster and smoother.

And poking around in those little grey areas, those twilight zones between what's accepted and what's possible, those places where innovative engineers can go and say "Gee, no one's tried this before—let's see what happens," are what keeps MIDI vibrant and exciting, and ensure its longevity. MIDI is not perfect—and that's what makes it dangerous, and beautiful.

RE/P

SPARS ON-LINE

By Bruce Merley

A Personal Approach to Managing Personnel

Recently at Clinton Recording we have been writing, revising and updating a number of our operation and procedure manuals. This arduous task has led me to examine the very nature of documentation and its place in small business.

Oh, for the days when we had just started the studio—the days when we all knew our jobs and everyone worked for the common joy of seeing the company succeed. How do you keep your staff productive once the glitter of newness is gone, the founding team has moved on to new horizons and you've experienced several staff turnovers?

In many ways, the task of keeping a successful business running well is far more difficult than creating the company in the first place. Once the glamour of construction and new equipment installation is gone, you are left with the essence of any business: the people. How well they are directed and motivated will make the difference between long term success and failure.

Many of us who operate studios keep far more of our company's policies and procedures in our heads than we would care to admit. With a smaller staff, little more than overtime wages seem worth recording and controlling. When the business is just beginning, the staff is hand-picked, and each person is perfect for the job. The need for training and orientation is minimal because everyone knows their job and shares a common desire to be part of a new success story.

How do you keep your staff productive once the glitter of newness is gone?

Sooner or later, however, the receptionist leaves to take a position as an account executive or a neurosurgeon. So, who trains the replacement? Even if the new recruit is a former AT&T operator, how do they find out about company policy and the intricacies of the job? Often, the training is done by the departing employee, or by fellow employees in similar positions. Training becomes haphazard and is only as comprehensive as the knowledge, skills and interest of those doing it.

Frequently, the owner/manager is the trainer and must learn the necessary skills of *every* job. Imagine the situation when the chief technical engineer leaves after years of designing, building, wiring and tweaking, and you discover that there is no documentation.

How do you begin to train a replacement when you don't fully understand the job yourself? Suddenly, the need for manuals, procedures and documentation becomes apparent.

Suppose you dismiss an employee for excessive and unwarranted absences. This person may contest the firing and you could end up in labor court defending your actions. Can you support your case? Do your time sheets reflect the absences? Did you give the employee adequate warning? Did you put the warning and termination in writing? Was the employee aware of the consequences of his actions? All of these factors could affect your case. The need for proper policy and documentation is essential.

The goal is to develop personnel practices and operating procedures that simplify rather than complicate your business. You need accurate and comprehensive job descriptions. You also need concise and clearly written policies that will clarify the employer/employee relationships while protecting the rights and interests of both. Finally, you need systems that will support employee morale, cooperation, growth and development, so that relationships are mutually beneficial. In this way, the business can have maximum staff productivity.

One way in which an owner/manager can cultivate a creative staff is through a personnel manual. A skillfully written manual can be a powerful resource and can serve as a positive, solid foundation for the staff's overall happiness.

The manual should serve two purposes: to inform the staff of company regulations and changing policies, and to give managers the necessary support to enforce those regulations and policies. It also should clearly spell out salary reviews, holidays, benefits, leaves of absence and other details that affect morale. Omitting of these details can lead to legal problems.

A personnel manual can take a variety of forms and can be as large or as small as necessary. It may be a booklet or folder given to all employees, or a single master volume that is continually revised and updated. There are a number of sources for boilerplate manuals that serve as sample material. However, whether you use standard statements or write your manual as an original document, be sure to have your attorney review the document. Even simple innocent errors can be seen as discriminatory and lead to costly implications.

Descriptions for all classifications of employees are invaluable. Good job descriptions are very helpful for the employees' understanding of responsibilities, how one job relates to others in the organization and how each role is perceived by the employer. A useful exercise is to have employees write their own job descriptions and then compare them with the "official" versions. It's astonishing how the employer and employees can view jobs in different ways.

Bruce Merley is first vice president of SPARS and is president/general manager of Clinton Recording Studios, New York.

The personnel manual and job descriptions form the basis of interaction, both socially and professionally, within the business environment. Once these materials have been prepared, further specifications and standards tailored for the particular business can be written. The extent of such documentation is up to the discretion of the owner/manager.

Once again, the documentation should merely simplify and increase employee morale and productivity. Avoid casting in stone the things that are common knowledge and practice. Once you write it down, you may limit people's intuitive instincts for finding more efficient ways to get a job done.

Sooner or later, we are all faced with situations that manuals and specifications do not cover. For example, how do you deal with a key employee who develops serious personal problems? Operate by the book and you may lose a critical team member. But, if you fail to deal with the situation in a manner consistent with policy, you could incur the wrath of other employees.

Develop personnel practices and operating procedures that simplify rather than complicate your business.

The owner/manager's life is filled with exceptions to the rules. This is the real reason for developing administrative structures. Without first agreeing on the rules, you cannot know when you are confronted with an exception. The entire staff should be directed with fairness and consistency. To insure this, there must be a firm policy of operation. What will follow is healthy morale and higher employee productivity.

Firm policies will be welcomed by employees if they are written well, clearly stated and consistently enforced. Most people find a measure of security in knowing the rules, and will do their best to follow them. When the rules are unknown, unstated or constantly changing, the staff becomes unsure and are less willing to make a commitment to the business.

Many organizations offer materials, seminars and training for personnel management. The American Management Association, the U.S. Small Business Administration and local small business agencies can be sources of help. Professional seminar training organizations such as Career Trak and Keye Productivity Centers are also good sources for developing skills in personnel management.

This year, SPARS will sponsor business seminars on marketing and personnel management. For more information contact: Shirley Kaye, executive director, SPARS, 4300 10th Ave. North, Suite 2, Lake Worth, FL 33461; 305-641-6648. RE/P

UNDERSTANDING COMPUTERS

By Jeff Burger

Hardware and Software

In the computer world, most everything falls into the general categories of software or hardware. Software is the term used to describe programs and data, while hardware refers to the actual machinery that acts as a vessel or conduit to hold, manipulate and process the software.

While the analogy is a bit crude, you might think of your car's engine as hardware and the gas, oil, water and other fluids as software.

The major hardware components consist of input devices such as the keyboard, mouse and joystick. Output devices include monitor and printer; memory includes RAM and ROM; storage devices include floppy drive, hard disk and cartridge; interfaces include expansion slots, serial port, parallel port and perhaps most important—the microprocessor.

Let's begin our discussion with memory. There are basically two types: RAM and ROM. RAM stands for random-access memory and is where your program and data live while you are working with it. This memory is volatile, meaning that when you turn the computer off, the contents are lost. The term "random-access" refers to the concept that each memory location is uniquely addressable. Indeed, you can think of each one as a mailbox that holds one byte of data. One kilobyte (1k) of memory is 1,024 bytes. Looking back in the April issue, this apparent incongruity makes sense because 1,024 is a multiple of 256. One megabyte is one million bytes and one gigabyte is one billion bytes. Big time!

ROM is an acronym for read-only memory. ROM is also addressable, but as the name implies, you can only read information from it. You cannot write to it. ROM is used primarily for storing programs and instructions.

For example, the ROM cartridge for a DX7 can provide you with alternate sounds, but you can't store any alterations back to the cartridge. Conversely, a RAM cartridge will allow you to write to it.

ROM chips are also used to store "firmware" routines that a computer or peripheral needs in order to operate. For example, the BIOS chip in an IBM is a ROM, which contains its operating system or personality. So when you first turn the computer on, the BIOS tells it automatically to do a memory check and look for a disk that has additional instructions or personality traits. Let's now establish the distinction between memory and storage devices. Earlier, we established that while RAM is the computer's workspace it is also volatile. Disk drives provide a way to store information semi-permanently, and floppy disks additionally offer a medium where programs and data can be transported, exchanged and sold.

All disk drives have a circular surface that spins much as a record spins on a turntable. Rather than the spiral grooves in an LP, however, the disk consists of concentric rings, and the surface is further cross-sectioned into wedge-shaped sectors. (Picture a dart board with the metal wire separating the different sectors.) The surface is susceptible to magnetic fields just like a piece of recording tape. An electromagnetic head flies just above the surface of the spinning disk and can be positioned over any of the concentric tracks to access any sector on any track.

Before using a blank disk, you must first format it. Another way to think of a disk is as a valet parking lot. Imagine your data as a fleet of cars. An unformatted disk is a parking lot without delineated spaces, so the lot manager can't find your data/car. By formatting the disk, the available storage area is divided into defined spaces, each specified by a track and sector cross-reference. Each location can then hold, say, 256 or 512 bytes of information, depending upon the machine.

Now that we have places to "park" our data, the manager keeps track of what's parked where by creating a disk directory, which is a sort of map to where everything is. All this lower-level management is done for you. That is because all computers have a form of DOS, or disk operating system, that manages the everyday needs of summarizing, loading, saving, deleting and renaming disk files. So when you tell the computer to store your file, it finds a place for it. When you want to retrieve it, the location is automatically looked up and retrieved. I'll cover DOS in a future column.

Let's correlate the interaction between RAM, ROM and floppy disk by the word processing of a letter as an example. When you first turn the computer on there is nothing in RAM, but the internal ROM knows to proceed with a memory check and to prompt you to insert a disk if none is present. If the DOS is disk-based, the ROM will pull the operating system into memory and from then on, use it whenever the disk drive is needed. (On some systems, DOS is part of ROM, so this step is obviated.) Now you would insert the disk with your word processor program on it and instruct the computer to run it. This process is known as booting a program.

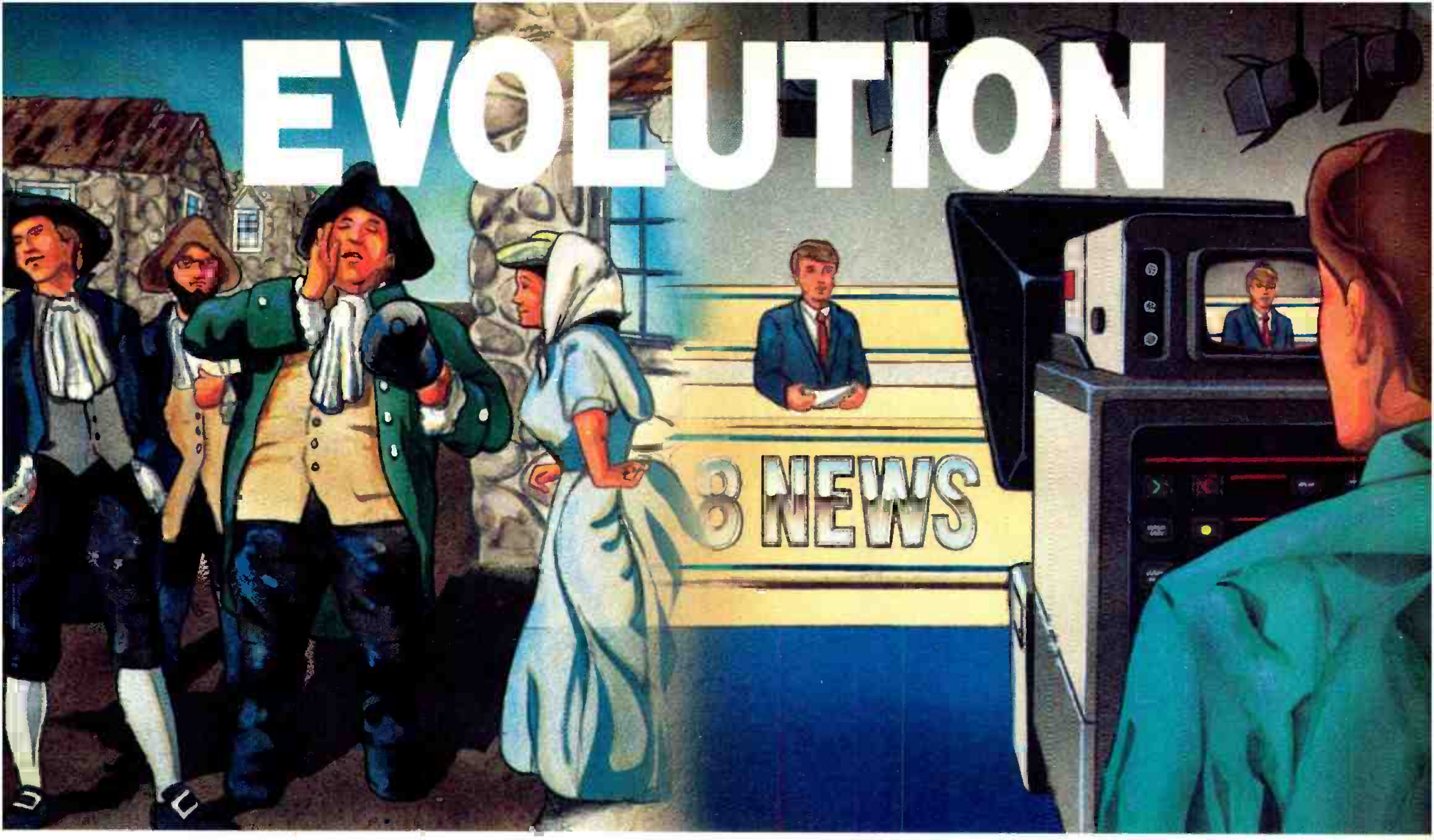
At this point, the computer pulls a copy of the program into RAM in order to be able to run it. Now you're ready to type your letter. Keep in mind that the original program has not been erased and is still intact on the disk.

When you type your letter, the characters go into a different section of RAM, separate from the program itself. When finished, it is necessary to store the document on a disk using a save command. If you simply turned the computer off after typing, your work would be lost, because of the volatile nature of RAM. Again, it is not necessary to store the word processor program back on disk because it lives there permanently unless you erase it.

Now you can turn the computer off without concern. When you come back later to make changes to your letter you go through the same procedure of booting the word processor with the added step of retrieving your document back into RAM using a simple load command. After making changes, issuing a save command will replace the original file with the revised one. If you wish to keep both versions, simply choose a new name or the revised version and a separate file will be created while the original remains intact and unchanged.

Next month, we'll take a look at the differences between floppy disks, hard disks and other storage media. RE/P

Jeff Burger is owner of Creative Technologies, a computer consulting company in Los Angeles, and is a technical writer.



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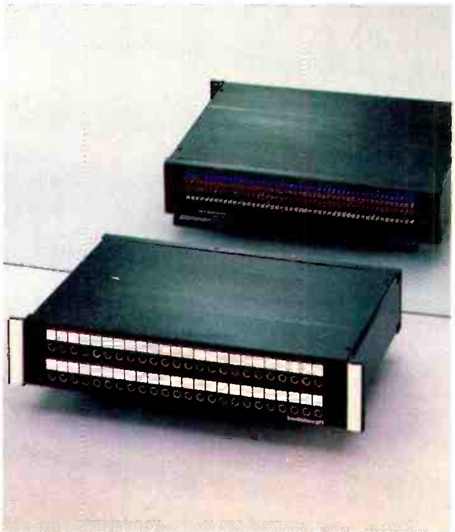

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Circle (11) on Rapid Facts Card

Close Field Monitors

By Jeff Blenkinsopp

Getting the most out of your close field monitoring system and looking at future trends.

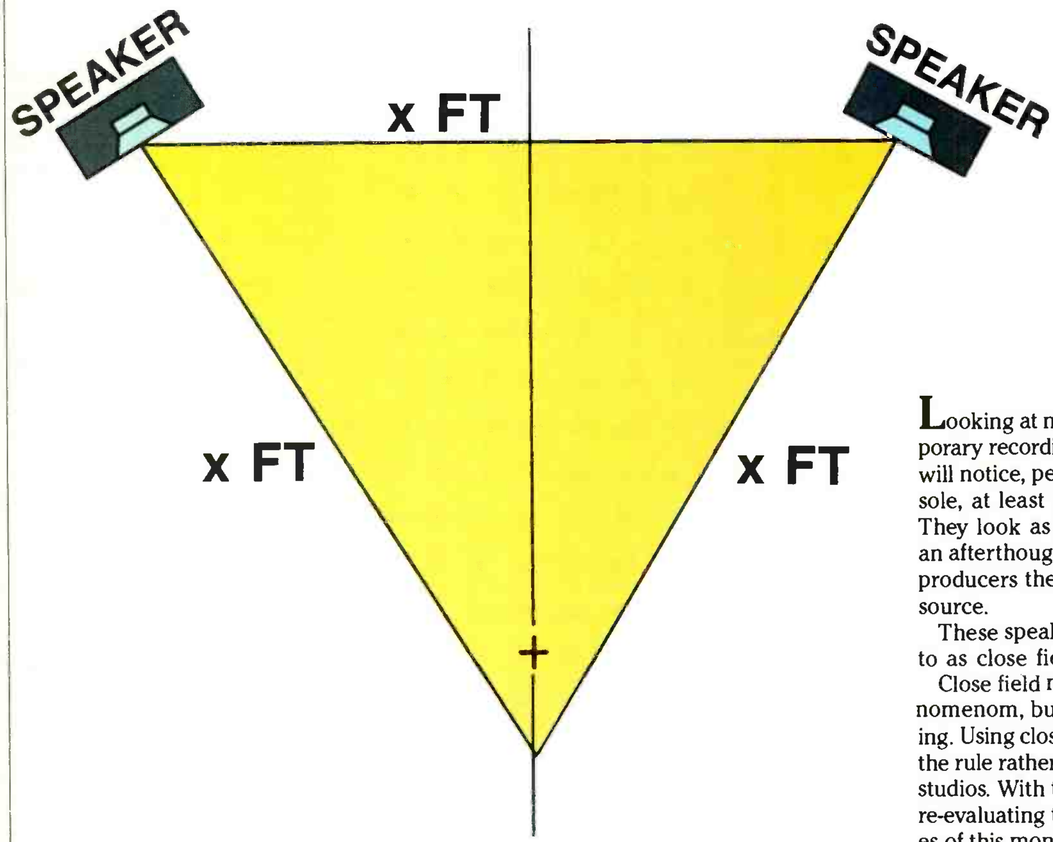


Figure 1. Example of height and spacing techniques.

Looking at most photographs of contemporary recording studio control rooms you will notice, perched on the top of the console, at least one pair of small speakers. They look as if they are placed there as an afterthought, but to most engineers or producers they are the primary listening source.

These speakers are commonly referred to as close field monitors.

Close field monitoring is not a new phenomenom, but it is an area that is changing. Using close field monitors has become the rule rather than the exception in many studios. With this evolution, engineers are re-evaluating the strengths and weaknesses of this monitoring technique...searching for optimum performance.

Consider that in a commercial recording studio, where numerous people will be working, a standard listening reference is a necessity. Without a reference, it is difficult to know what the sound really is. In today's studio, that is generally achieved

Jeff Blenkinsopp is an audio consultant in New York. He is currently designer/project engineer at Quantum Sound Studios, Jersey City, NJ.

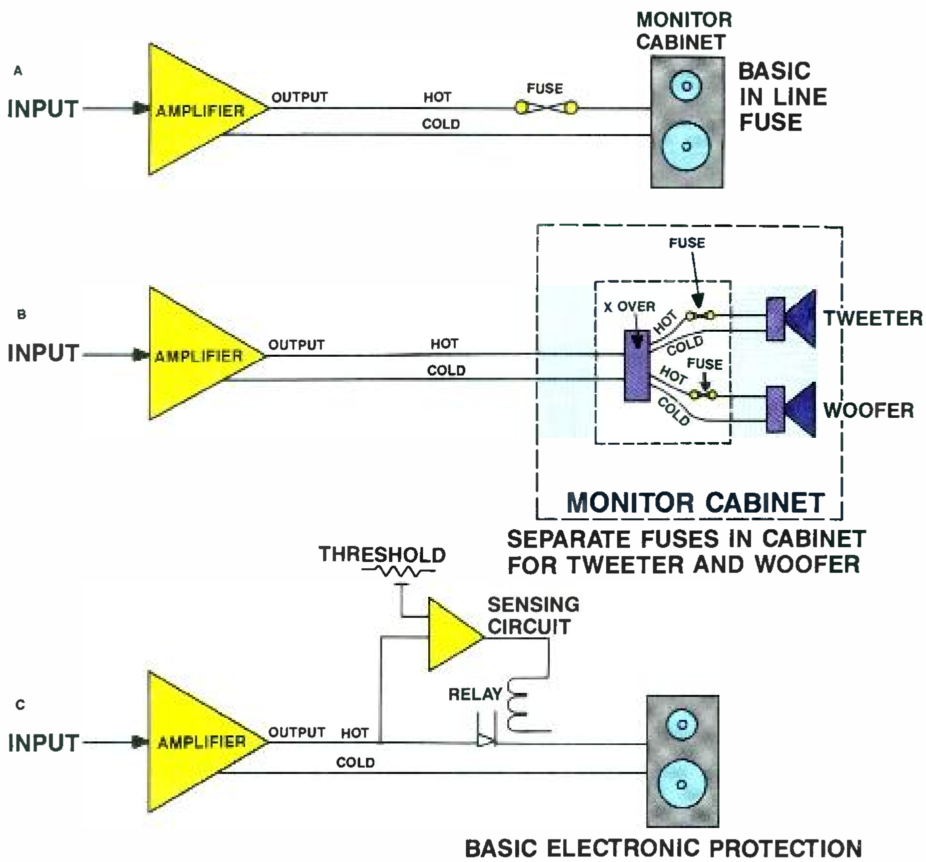


Figure 2. A: *Example of in-line fuse between the amplifier and the monitor.* ***B:*** *Example of fuses fit inside the cabinet.* ***C:*** *Example of electronic protection.*

by those small speakers sitting on the console.

The definition of close field monitor is a speaker that can be chosen *and positioned* by the listener, and the listening point is within the direct sound field.

The effect of any anomalies in the acoustics of the room (such as the boundary effect, resonances and reflections) are reduced by listening within the direct sound field. The closer you are to the speakers, the more direct sound you hear and the less the reverberant field affects the sound.

When wearing headphones, you only hear the direct sound, and the room you are listening in has no effect on the sound. It is this headphone effect, to some degree, that we are trying to achieve by having the speakers close.

The acoustic power generated by the speakers should also be optimized so the room is not driven (again to reduce the effect of room anomalies) and also to reduce the speaker distortion that can occur at high output levels.

I've also stated that the listener has control of the positioning. This may seem obvious, but it is a unique factor as the monitor *selection and positioning* can be controlled by the engineer—a detail that is often overlooked.

And, of course, size becomes a factor. If the monitors are too large, they become impractical, affecting the sight lines or interfering with the direct sound from the large monitors.

Basically, close field monitors are used to reduce the variables of the control room acoustics and the abnormalities of the main monitor speakers.

Reference speakers

The small size of most close field monitors make the monitors ideal for reference use. Engineers/producers can easily bring in whatever system allows the "reference" they feel is most accurate for their style. The flexibility to bring in your own "electro-acoustic ears," is making close field a popular and effective alternative. The ability to reference any job or environment to your known standard is a great advantage.

Pros and cons

The current generation of close field monitors are relatively inexpensive and easy to install and power. It's easy even for a small studio to have the same monitoring that the leading studios have.

You would think that close field monitoring is *it*. But there are still some deficiencies. The biggest problems are lack of bass and undistorted acoustic power. It is for these reasons that the large monitors are still very important in the control room. Getting enough low end out of a small monitor cabinet is difficult, and without knowing what is happening in those frequencies, your whole project could be ruined.

Also, getting the "feel" of the music is very important, and sometimes close field monitors will just not give you that, if, for example, you're doing a heavy-metal band or a dance-single re-mix.

Mounting and location

Close field monitors are usually freestanding and are not mounted in soffets or against walls. Even when they are sitting *on* the console, the effect of the acoustic coupling between the speaker and the console has to be considered. Generally, there is a slight increase in the low end response (which is often desirable), but it can be reduced by mounting the monitors on risers.

And, just like the main monitors, the high frequencies will tend to splash off the console, causing coloration and affecting the stereo image. The amount of this effect depends on the shape of the console and the speaker positioning. This is one area where the engineer can reposition the monitors to improve the effect of high frequency reflections off the console. The cabinets can be turned on their sides, or even upside down. It's up to the engineer; there is no definitive right or wrong. These positions are not so easy to achieve with the mains.

The monitors should be matched pairs, and the manufacturer often marks the cabinets R and L. If you blow a component in one cabinet, it is best to replace the same part in both cabinets simultaneously. The remaining good part can be kept as an emergency replacement.

Height and spacing

One certain guideline to follow is: the distance between the speakers should equal the distance from slightly behind the listener's head and the center of the cabinet. (See Figure 1.) The speakers should be the same height or slightly higher than the listener's ears (when seated approximately 42 inches from them.)

Wiring, powering and protection

The cable used should be given the same considerations as the large monitors. Don't skimp on the cable just because the monitors are small. Care should be taken that the termination of the cable at the cabinet is satisfactory. Often the terminals on the back of the cabinets are inefficiently designed. If the hole is too small for the wire to go through, don't cut the wire down but tin it and wrap it around the terminal. In other situations, spade adaptors could be used that fit around the binding posts.

High-quality amplifiers should be used with enough output headroom to allow for all transients. Often, especially during large transients, the *distortion* of the amplifier blows the monitor speakers; a larger power amplifier is more important than you would think. For example, a 250W amplifier connected to a 50W cabinet is often safer than a 50W amplifier that operates continuously at or near its rated output.

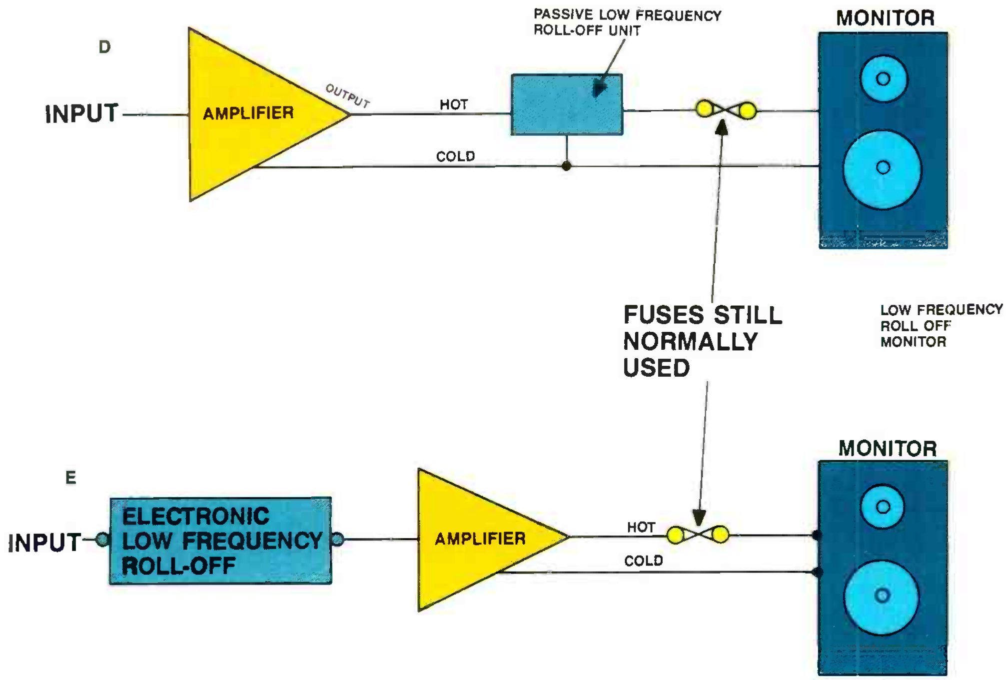
The speakers used for close field monitoring usually cannot handle a large power input, and because the trend is to connect them to high power amplifiers, the possibility of blowing the monitors exists. If you decide to add protection, the most common method is with an in-line fuse between the amplifier and the monitor. (See Figure 2A.)

The fuse should be a fast blow. A fast blow fuse will blow on surges and a slow blow fuse will not. A common fuse is the AG series; this would be marked 3AG, AG or AGC on the fuse. The inside of a fast blow fuse usually has a fine wire, while a slow blow fuse has a coil of wire or some unidentifiable mass. The fuse's AMP value is determined by the rating of the speaker and experience.

The easiest way to determine the fuse rating is by trial and error. Start by installing a very low fuse value (typically 1A) and then turn up the level. If the fuse blows before the listening level is comfortable, replace it with the next higher value. Fuses typically come in 0.25A increments so the next value would be 1.25A. Try replacing fuses until one does not blow.

Some studios fit fuses inside the cabinets and fuse the woofers and tweeters separately (see Figure 2B). This configuration causes no problems if the cabinet is sealed around the fuse holders and correct fuse values are used.

The type of fuseholder is also important. There are two basic types of fuseholders: screw top and quick release. Screw top fuses have proven to be more reliable because they are mechanically stronger, so the small amount of time it takes to change one is well worth it. As for mounting fuse holders, they can go anywhere—in a panel by the amplifier, on the back of the monitor cabinet or in an accessible box.



There are also electronic means of protection. With electronic protection, the input to the speaker passes through a relay and the circuitry senses the power going to the cabinet. (see Figure 2C). If too high, the relay is tripped. The relay is reset if the power is removed. Some hi-fi speakers have these protection devices installed internally. Unfortunately, because of the low setting of the trip point, they are annoying and are often bypassed if used in the studio.

The cabinets do not reproduce very low frequencies (typically anything below 40Hz-50Hz), but it's the presence of these frequencies that often blows the monitors. Some studios are fitting low-frequency roll-off units in their systems to eliminate these frequencies getting to the monitors. (See Figure 2D and 2E.) This provision will work providing it's inaudible to the engineer. The time, energy and cost to get the monitors protected is well worth it. Replacing drivers is costly and aggravating.

New approaches

The use of close field monitors has become so common and important that the development and improvement of this technology should be pushed. I'm sure all the speaker manufacturers are trying to develop new products that will improve the low-end frequency response and power handling capabilities, but perhaps the entire system needs to be re-evaluated. Below are some thoughts on systems I feel deserve more research and energy.

- Bass or sub-bass systems: One of the problems with close field monitor cabinets is the lack of bass response. One way to compensate for this lack would be to add a separate bass system. Because the low end is not as directional, these cabinets could be located away from the existing close fields—possibly under the console. The crossover (probably electronic) for this type of system would have to compensate for phase and time alignment, etc., but we have the technology. The use of separate bass/sub-bass systems is becoming more common in live sound systems and in home hi-fi designs, but, as yet, has not found much acceptance in the studio. I expect to see this change in the future. What about mono bass?
- Multiple cabinets: Why not use two close field cabinets next to each other with each cabinet featuring a different response and wired in such a way that the engineer can listen to one or both. One cabinet could be the standard reference and the other with characteristics to compensate for the reference.
- Multiple systems: In today's studios where it is common practice to have some musicians in the control room, especially keyboard players, more than one monitor system is becoming necessary and, yet, it is rare that a second or third pair of monitors is incorporated. There is really no reason multiple systems can't work in tracking sessions, especially if the overall levels are not saturating the room.
- Hi-fi systems: The use of home hi-fi cabinets as an engineer's own personal monitor is not uncommon. The common use of these cabinets in the control room seems to have been dismissed without real in-depth evaluation.

Room design

The standard control room environment can also be re-evaluated. With close field monitoring a whole new range of options becomes available. For electronic production, the division between control room and live room can disappear. The need for conventional soffets is outdated and so the control room can take on a more home hi-fi type environment becoming more comfortable, flexible and cost effective.

I hope you are now reassessing your close field monitoring. In order to improve the state of our art the people involved must push the bounds of the norm. It's only through open lines of communication that change can be affected. Your feedback is important to all of us. If you have any comments or suggestions on improving close field monitoring we would like to hear from you. RE/P

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The MIDI control information for the DMP7 can now be recorded in sync with the music. The DMP7 then automatically mixes everything down into a two-track master or demo. Until now, this level of mixed automation was only possible with more expensive consoles.

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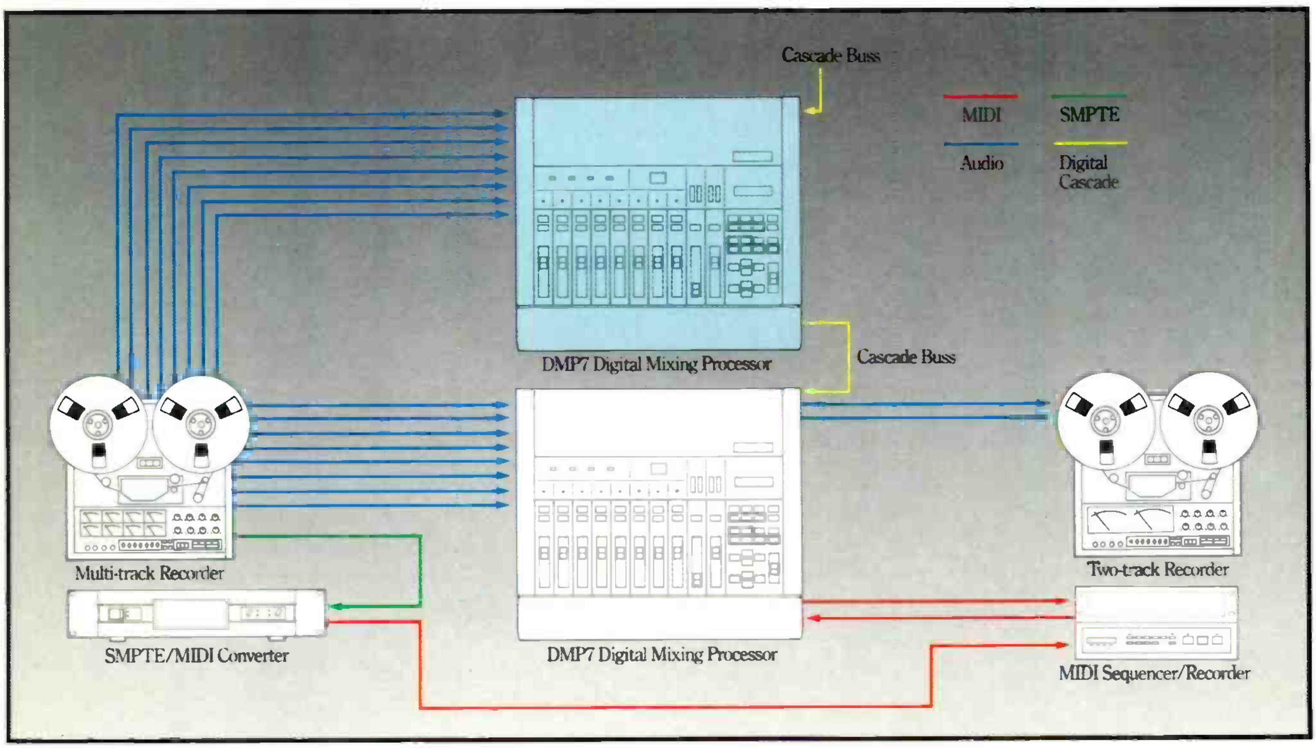
What's more, if you have more than eight tracks, you can cascade in the digital domain to another DMP7 for 16 tracks. Or another for 24. And yet another for 32 tracks.



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Circle (13) on Rapid Facts Card

Subwoofers in the Studio

By Mike Klasco

The subwoofer's role in the studio might be to supplement the low end response of the popular close field monitors.

Recording studio engineers may consider the subwoofer as a high-end consumer product, but rarely as a component in the mixdown process. The explosion of digital technology may soon transform the perception of the subwoofer into a necessary extension of close field monitoring techniques.

Recording studio monitors evolved from the A-7 Altecs of the early 1960s to the wall-mounted JBL vented 3-way monitors of the 1970s. Some studios also alternated their monitoring to a set of KLH extension speakers, which used a 4-inch extended-range driver, to simulate how their mixes would sound on small hi-fi and AM/FM car systems.

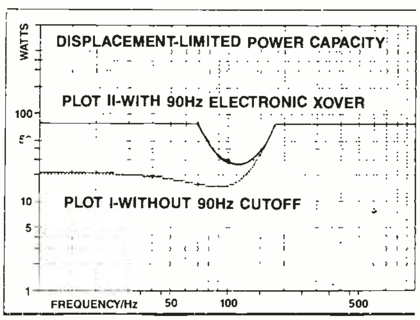
Auratone "cubes" followed, and speaker designs evolved to the 2-way close field monitors of the 1980s. The term near field is used in acoustics to define a condition close-in to the sound source, where a slight increase of distance results in a large change in sound level. Close field as a term to describe monitors relates to placing small speakers usually on the console meter bridge a few feet away from the recording engineer's head. The idea is that the direct sound from the monitors is sufficiently loud to overwhelm the reflected room sound. [Editor's note: See article on close field monitoring on page 16.]

It has been suggested that subwoofers' roles in the studio might be to supplement the low-end response of the popular near field monitors. Most large professional monitors have bass response to at least 40Hz, and some extend down to around 20Hz, but this bass comes at the expense of having to switch away from the spatial advantage and continuity of using near field monitors.

Mike Klasco is president of Menlo Scientific, an engineering and acoustic consulting company in Berkeley, CA.

Photo by Michael J. Gill

Cross-section view of the Velodyne Acoustics ULD-15 driver.



Driver Parameters:
FS (Hz): 51
Qes: .62
VAS (L): 38.4
Vd (ccm): 44
Pe (W): 75

Box Parameters:
Vb (Hz): 7
Filling: Y
Atc/Fc: 1.07/120.4

Figure 1. *Yamaha NS-10M power and handling model with and without high-pass electronic crossover.*

The concept is simple—use a subwoofer to provide the bottom octave that is missing from the near field monitor's response. But other benefits accrue from this configuration. The close field monitors and subwoofer would be bi-amped using a crossover point below 100Hz. The responsibility of low-frequency reproduction is lifted from the small monitor woofer resulting in higher dynamic range, lower distortion, and improved reliability and stability of performance.

And, for conventional operation, the electronic crossover and subwoofer are bypassed and the close field monitors are used by themselves, providing continuity of spatial perspective and spectral balance, without the disruptive effect of switching to and from the large monitors—in order to check low end response.

There are a number of popular close field monitors, but regardless of the specifics of any one design, the physics remain that these small monitors suffer from excursion-limited dynamic range. Scientific Design Software's computer-aided-speaker-design program was used to model the power handling of the Yamaha NS-10M, both driven full range (see Figure 1, plot 1) and when used with a subwoofer and electronic crossover (90Hz/12dB per octave slope). From the computer modeling we can see that even at 100Hz the power handling has doubled (see Figure 1, plot 2) when used in conjunction with the electronic crossover.

Single or multiple subwoofers

A single bass source eliminates the low-frequency comb filter effects of spaced enclosures. For example, woofers spaced 10 feet apart will have a peak at about 100Hz (one wavelength) and a notch at half wavelength. At the acoustical research company of Bolt Beranik and Neuman, a concert hall simulator was constructed to help predict the acoustical characteristics of halls under design or renovation.

The system was complex, involving many channels, but only a single low-frequency source was used in order to pre-

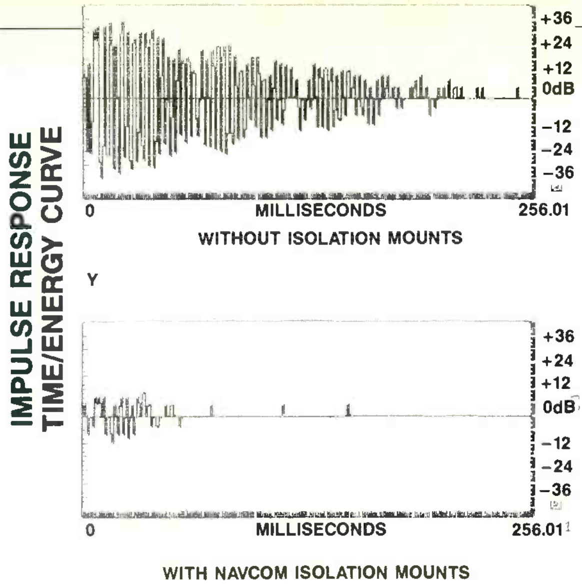


Figure 2. Relative energy/time plots for vibration leakage to the floor with and without isolation.

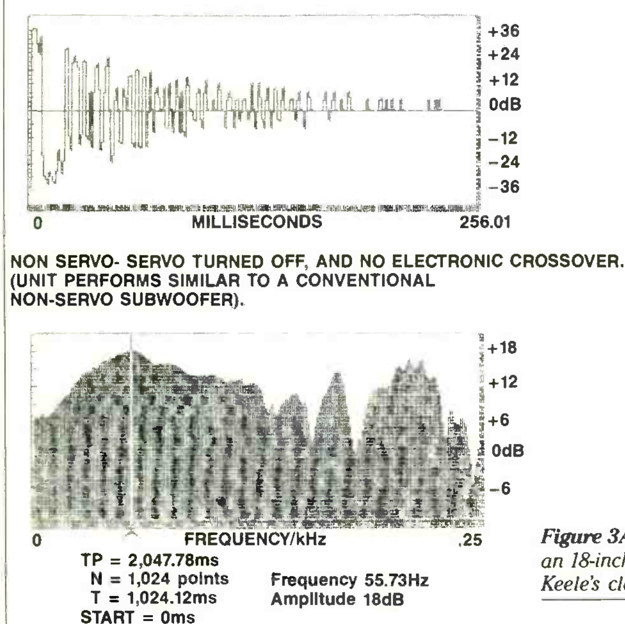


Figure 3A. Impulse response test for an 18-inch subwoofer featuring Don Keele's close field technique.

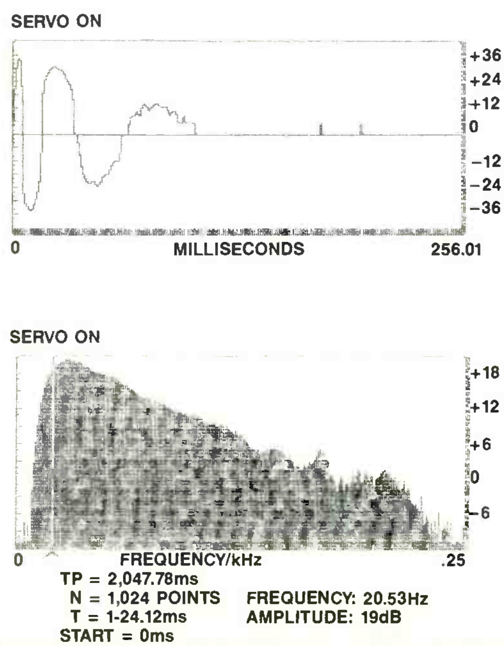


Figure 3B. Impulse response test for an 18-inch subwoofer with servo-control "on" using identical mic, time and gain settings.

serve accurate low-frequency reproduction. The research found that the peaks and dips of spaced woofers (a comb filter effect) could be best prevented with a single bass channel, although the current theories on the human localization mechanism assume progressively less sensitivity to stereo localization below 160Hz (and no localization below 80Hz). Studies a few years ago at the British speaker company, KEF, conversely pointed to the importance of multi-channel extreme low-frequency reproduction for accurate portrayal of "space" and ambience.

My own experiences of living with a monitoring system that extends to 15Hz, have found that regardless of whether single- or dual-channel subwoofers are used, the sense of spatial ambience is maintained when the response and the recording extends to below 30Hz. Possibly, this is the "acoustic room signature" of the original space, ie., low-frequency reverberation/standing waves, etc. Therefore, stereo may not be important at the bottom octave because the original acoustics may be perceived somewhat by the global reverberation characteristics and Eigen tones (standing waves) rather than localization cues.

In any case, the studio engineer's thoughts on these subjects of physical space, budget limitations and other criteria will determine whether single or multiple subwoofers will work best.

Increased definition and dynamic range

The bass driver in close field monitors vary from around six and one-half inches to eight inches. Reproduction of low frequencies, even at moderate levels, results in mechanical stress and fatigue on the driver. A weakness of many close field monitors is the loss of clarity in the vocal range when accompanied by instrumentals containing substantial bass energy. Physically, the problem is easily traced to cone over-excursion beyond the linear region of the driver. This situation results in harmonic distortion and intermodulation distortion (due to non-linearies of the magnetic system and suspension at large excursions), Doppler distortion (frequency modulation caused by the bass frequencies modulating midbass and midrange frequencies from the same driver), and cone breakup due to overdriving.

Time alignment

Accurate time alignment is now an accepted design refinement in a modern studio monitor. The small, closely spaced direct radiator (or coaxially mounted horn tweeters) used in most close field monitors are able to achieve reasonable time alignment over most of the critical mid-frequencies. This has been one of the side

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benefits of the system. Certainly, precise time alignment is critical for stereo imaging between 1kHz-5kHz, (where a distance of a few inches represents hundreds of degrees of phase shift). In the case of close field monitoring, smooth phase response is especially important as the listener is hearing predominantly direct sound with very little masking by the reverberant field.

At frequencies below 100Hz, the ear has a low sensitivity to phase shift, and the distances are fairly large before significant cancellation occurs (about five feet for 180° phase shift at 100Hz). To predictably match the subwoofers to the near field monitors, both should be minimum phase devices throughout the crossover band. This case requires that the near monitor not have a vent resonance, or be rolling off, in the crossover region.

Positioning

The subwoofer(s) should be positioned within a few feet of the monitors. But keep in mind the effect of each surface boundary (i.e., walls and floor) that will be reinforcing the bass by about 3dB. Many subwoofers' designs use a downward firing configuration with only an inch or two spacing from the floor. This approach provides some beneficial loading to the woofer cone, while reducing the audibility of harmonic distortion. Harmonic distortion is "lost" due to diffusion and floor absorption effects. Care should be taken that *cavity resonances* are not created if the subwoofer is placed behind the console.

Vibration

A well-designed subwoofer can provide the extended bass response and sound levels of the best full-size commercial monitors. Careful attention must be paid to isolate the transmitted vibration from the subwoofer enclosure to the floor, walls, or the building structure (this is to avoid leakage out of the control room). Isolation mounts should be used and the enclosure walls should not be in direct contact with any equipment racks, or other equipment. Figure 2 shows energy/time plots for vibration leakage to the floor without isolation mounts—compared to the energy reduction with effective materials.

The commercial success of digital disks has spurred on the popularity and sales of subwoofers in the consumer audio market. The need for the studio engineer to mixdown with a monitoring system with at least the clarity, definition, spatial and bandwidth performance of the better consumer audio speakers becomes self-evident. The symbiotic relationship between close field monitors and a subwoofer is one solution to achieve this performance in the control room. RE/P

Servo-controlled subwoofer design

By Mike Klasco

Subwoofers come in a variety of design configurations. One such design incorporates the use of a servo-controlled motor in place of the more traditional voice coil. See Figure 3A and 3B.

Velodyne Acoustics manufacturers subwoofer systems that use high-gain servo technology to extend the low-frequency response and reduce distortion. The system consists of the bass enclosure, rack mount amplifier and a small control unit that can be positioned by the near field monitors to switch-in the subwoofer for use in conjunction with the close field monitors or to bypass the subwoofer and use the monitors' full range.

High-gain servo subwoofers are an advanced development of motional feedback transducers. Motional feedback is the process of feeding back, to the driving amplifier, an output signal for comparison to the input signal and error correction circuitry.

A servo-controlled loudspeaker requires some sort of sensor to supply the output signal (feedback to the amplifier) for comparison to the "command" signal for error correction. While the sensor itself must be accurate, the greatest challenge is making sure the sensor is providing an accurate indication of the speakers output signal.

In previous attempts at servo speaker designs, various locations and types of sensors have been used. The obvious way to sense the woofer's output would be to use a microphone. Unfortunately, the delay in the sound traveling to even a closely coupled mic would cause enough phase shift, between the command signal and the signal picked up by the mic, to significantly reduce the usable bandwidth and stability of the system.

Dave Hall, president of Velodyne, developed and patented a solid-state integrated device in a configuration directly coupled to the bobbin and coil. A pre-amp located within the driver provides a signal back to the amplifier (see photo). This technique requires the construction of a loudspeaker having a high signal-to-noise ratio. The parameter signal-to-noise is not generally applied to loudspeakers, but this is a critical factor in the design of reliable high-gain servo systems.

The "noise" can consist of air turbulence in the voice coil gap, or from the vented pole piece, bobbin resonances, frame rattles, the dust cap buckling during large excursions, standing waves (reflections) back from the cone to the bobbin because of a wide cone body angle, or even vibrations between the laminations of the voice coil.

Although the general idea of feedback is for distortion reduction, it is not a short-cut approach to low distortion design. Keeping the driver free from rattles and buzzes requires an optimum size dust cap with special pressure release vents in the cone behind the cap, flat wire coil techniques to minimize lamination rattle and to maintain upper band phase response (important for stability), hyperbolic curvilinear cone flair to minimize standing waves back to the bobbin and numerous other critical design techniques.

To the platform of a rattle-free and low-distortion driver, the sensor is bonded to the edge of the voice coil. This technique provides precise transconduction to almost 10,000Hz, and allows high-gain feedback to a few hundred hertz. Non-linearities in the magnetic system and suspensions are directly sensed, along with the precise location of the coil within the magnetic field. As the sensor "knows" the location of the coil, the system monitors against damage by limiting the maximum excursion through control of the gain in the feedback circuit.

An optimally dampened sealed box will be free of vent noises and typically have a shorter decay time at its natural resonance. With a servo-controlled driver in a sealed design, the in-box resonance can be brought down to 5Hz, well below the audible range, and the decay time is shortened as well.

Another benefit of servo-control is the wide range of enclosure volumes that can be used, without altering the performance. The exception is that the amplifier must work harder with smaller volumes. If equipment space is at a premium, servo-control allows the use of a smaller "sealed" box other than a vented box with a conventional driver.

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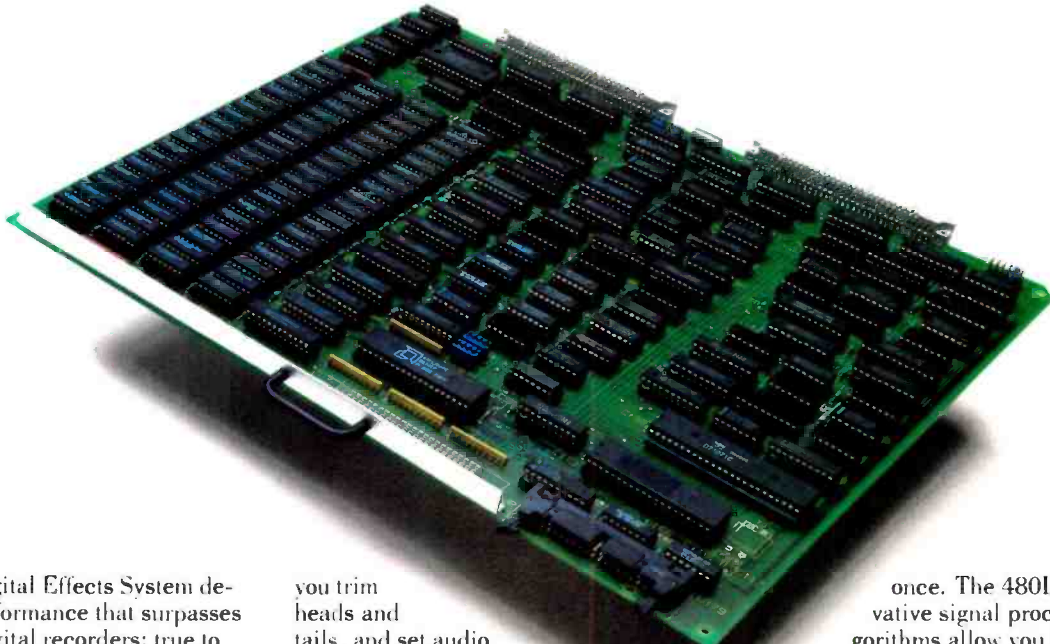

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Studio Monitoring: Design Perspectives

Photo by Loisjoy Thurstun

Boxer monitors in Red Bus Recording Studio, London.

By Richard Elen

Though expensive, a new generation of all cone/dome main monitors is becoming increasingly popular where improved bandwidth and linearity are demanded.

There are few aspects of studio life guaranteed to cause more controversy than a discussion of studio monitoring systems. Horns, cones, domes, subwoofers and near field...all have their devotees and detractors. In this article, we look at developments in all cone/dome monitor technology, how digital technology is forcing studio designers and monitor manufacturers to rethink approaches to controlling what we actually hear in a control room, and the importance of designing the monitor and control room package as one.

Richard G. Elen is a recording engineer, producer, technical journalist and owner of Creative Technology Associates in Somerset, England.

Facing the eternal challenge

Monitoring is possibly *the* most subjective area of sound recording. In earlier times, it was microphone technique—something that seems to occupy less of an engineer's time these days.

But monitoring never goes away (at least not in the foreseeable future). With the ever-increasing concentration on quality, particularly the impact of digital audio techniques, production teams need to have an audio reference that is not only sufficiently detailed to pick out all the dynamics and subtleties of sound that digital audio makes possible, but is also in some way definitive in representing what is *really going on* throughout the signal path and from sound source to ear.

Superior dynamic range, subtle detail and broadband linearity is difficult to achieve, both technically and—perhaps more to the point—conceptually. Loudspeakers and transducers are highly subjective instruments. Unlike a tape machine, the design criteria for successful control room monitors are riddled with personal preferences.

Studio monitoring, however, necessitates compromise. On one hand, we need the detail of a top-end hi-fi system. On the other, we require 24-hour-a-day professional reliability in the industrial environment of the recording studio. Different companies have made the inevitable compromises in different ways.

Near field vs. main

There are a number of viewpoints when it comes to the business of monitoring. Main monitors vs. near field is one of the primary polarities. Studio design consultants may be at fault here, because throughout much of the 1970s studio control rooms were so inconsistent, and the principles on which their designs were based so diverse, that it was difficult to achieve any degree of room-to-room reference. It was not at all surprising that the engineer ended up reaching for something that was more of a known constant, and portable enough to carry around. But at the same time, some people are still baffled by the fact that the same pair of loudspeakers sound entirely different in two rooms, even within the same recording complex.

It is surprising, perhaps, that it took so long for designers to realize what people heard in the control room was in fact the result of a highly complex interrelationship between room and speaker system—and that the two are inextricably linked, incapable of separate consideration except in the most mundane cases.

An increasing number of designers have come to the conclusion that monitor speakers should be subject to the same degree of control as the room acoustics. Instead of merely buying-in a stock monitor system that may have sounded good in another room, a system should be built from scratch. Obviously, it is impractical to custom-design a monitor system for every room that is proposed, but at least the consultant can specify a system that complements the acoustic work being done on the room.

Neil Grant of Harris Grant Associates, British studio design consultants, had a background in transducer design, and was

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involved in some of the first direct-radiating studio systems. A custom monitor design was one of his company's first decisions.

"We would develop a dome from scratch as the center of a completely new system," he says. "Because, if we were going to control the room acoustics and we were going to control the monitor system, there could be no excuses."

The Integrated System Theory

The theory is simple: if the room works properly, and the speakers work properly, there should be no *need* for a plethora of alternative monitors on top of the console itself other than the perfectly valid desire for an engineer to listen to an alternative or consumer system. This provides a variable reference while giving some indication of what the rest of the world would hear at home, in an office or car.

A large monitor could theoretically be designed, Grant and others argued, that had a competent amplitude and phase responses, and sounded good both on and off axis. And it should be possible to design a control room around such a system that would have stereo perspective across the entire rear of the room *and* was capable of generating sound pressure levels sufficiently loud to satisfy any customer, yet which was not fatiguing or painful to listen to.

The horn dilemma

Ten years ago, speaker designers were pointing out the deficiencies of horn-loaded systems. They are designed to provide high sensitivity from small devices over a relatively limited bandwidth and in a specific way. But, in a control room, we are looking for a system that will spread a broad bandwidth across a relatively broad area—a low-Q device.

The two aims are antithetical. When you achieve peak sound pressure levels in excess of 130dB unweighted behind the console, you have sound pressure levels far in excess of that between the diaphragm and the phase plug in a compression driver.

"The pressure is so high," Grant says, "that the air ceases to behave in any way linearly." And that's only one problem. The whole air path and diffraction around the edge of the horn are difficult to allow because while still balancing the performance factors you need quality-conscious monitoring. "It's surprising," Grant says, "that it took us so long to realize there might be a another way."

But, in another sense, perhaps it is less surprising. Until recently, there have been problems associated with getting the required sound pressure levels out of commercial direct-radiating devices—power handling and physical parameters to name but two. An obvious first step to finding a better way, once new materials arrived, was to examine the requirements of the mid-range driver, particularly its bandwidth.

All commercial horn-loaded systems tended to cross over in the 800Hz to 1kHz range. "Right in the middle of everything that we consider interesting," as Grant puts it. The first dome radiators covered a mid-band wide enough to push the crossover points to either side of what can be regarded as a *musical* interval. Grant adds, "When we designed the SD-100 midrange driver for the Boxer system, we saw a musical interval as a decade running from 250Hz to 2.5kHz.

These points weren't chosen to be suitable for the drivers that existed, but what we felt would be nice from a musical point of view. The two octave bands that are considered critical for intelligibility are the 1kHz and 2kHz bands, yet there is a great deal of fundamental information between 250Hz and 1kHz.

Intermodulation problems made it difficult to push the HF crossover point any further up, and one decade seems to be about as much as one can expect a single driver to do on its own.

The Boxer was designed around a midrange driver with a 100mm (4-inch) voice coil, in order to provide the power handling level that was necessary for the kind of sound pressure levels required, and an extremely large motor unit (magnet/voice coil assembly) to keep the sensitivity up. Once the crossover boundaries were out of the way and the midrange could be handled smoothly, LF and HF drivers could be added to provide the full frequency range.

Harris Grant designed its own HF unit, because it couldn't find a commercially available alternative that had the power-handling.

"It's one of the lowest thermal impedance tweeter assemblies to be developed," Grant says. It used commercial bass drivers because of the prohibitive considerations of manufacturing them themselves. To complete the integrated system, a crossover and tri-amp system was designed. This type of monitor is capable of producing a high enough sound pressure level to satisfy the requirements of the larger rooms that are now becoming more common.

The other important consideration in a monitoring system is power response, which is the integration and dispersion, into the half-space of the entire device. In the past, many people have simply used a cabinet and put up a microphone—giving a specific curve on-axis.

Another realistic approach, which some manufacturers have been doing for some time, is to examine both the horizontal and vertical axes, measuring the entire half-space in front of the speaker and integrate the response.

"You'll find that you get a trailing response [a response that falls off as the frequency rises]," Grant says, "because when you measure on-axis, you get this nice flat response, and as you go off-axis that increasingly falls, because you're dealing with devices that get more and more directional as the frequency increases."

Many designers are trying to produce a system with a smooth power response, with no nasty holes at the crossover points. As you go off axis with many conventional systems, especially those which are horn-loaded, you find that the horn driver is effectively moving backward.

As a result, there starts to be a characteristic notch in the power response at the crossover point in non-symmetrical systems. "We're pleased that ours doesn't do that," Grant adds, "but there's still a trailing response, because Q is rising with frequency."

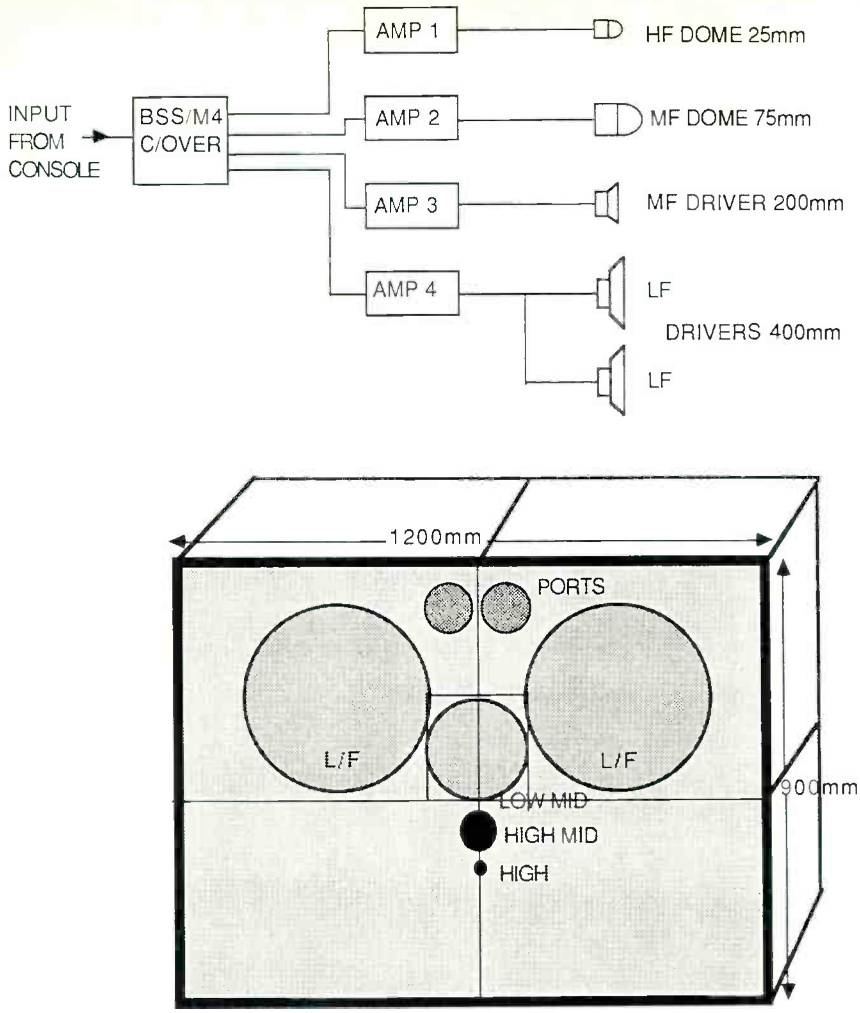
The trick is to cross over while using the directional characteristics of the speaker, so that the rising Q of the midrange matches the Q of the HF driver, for example. This crossover ensures you don't end up in a situation where one driver becomes increasingly directional while the next is less directional.

This tends to happen with some monitor systems featuring horn drivers in the midrange. An especially noticeable example might exist if a pair of bass drivers crosses over to a horn at about 1kHz—where the horn is actually less directional than the bass drivers. You'll find as you go off axis, the midrange becomes increasingly pinched.

Large control rooms

Swanyard Studios in London is one of the first of a new breed of spacious control rooms. You can walk across 10m (about 30 feet) or so of back wall and still hear the furthest of the pair of Quested monitors. The two keys to achieving this involves controlled dispersion and careful overall acoustic design. Peter Gabriel's new control room at Box Mill in England is another example of a very large control room using all cone/dome monitoring. The room measures 60' x 56' with a height of about 23 feet. Their aim was to have a stereo image *wherever you were* in that space.

Central to the argument is that a step in quality is required to meet the challenge of digital audio and the almost paranoid, yet entirely justified, growing concern among engineers to hear everything that is going on, no longer masked by



System layout of the M4(5). This view represents the front panel with all units flush-mounted.

analog noise or sonic coloration from the monitors. Suddenly, studios began to find, with embarrassment, that audiophiles at home had a better idea of what was ending up on tape than they did.

And one of the most important reasons, it could be argued, was many of the monitors in use at the time of the first serious influx of digital recorders were in reality more suited to stage monitor applications. They owed a lot to compromises required to achieve required sound pressure levels and durability. But now new developments in materials and transducer technology have made it possible for direct radiators to make as much sound as horn-loaded systems.

And there are psychoacoustical problems here also. While there is a woeful lack of agreement on such issues in general—and those concerning transducers in particular—there is at least one school of thought that believes the human brain is traditionally unaccustomed to dealing with distorted sound in the natural world.

As far as the brain is concerned, distortion is something that is generated in the ear by loudness. So, distortion sounds subjectively loud. If you then listen to a system that has an order of magnitude less distortion, you are likely to be unimpressed by its level (the distortion that tells the brain its loud isn't there)—and turn the gain up. This can be dangerous and a great deal of education is necessary here to perform this successfully.

The question then becomes: is it apparent loudness or smooth response that is desired from a studios' main monitors? And, though the rational voice is for smooth response, the lure of the apparent loudness remains addictive.

But, if the main studio monitoring system is doing its job, some engineers may consider a move away from the use of close field monitors, because, at last, they can trust what they hear from the main system. For a while there are several good arguments for close field monitoring—and especially "real world" monitors—the engineer should not be forced to depend on close field systems simply because the combination of control room and main monitors is not good enough to trust.

The close field monitor has its place in giving the engineer and producer a guide as to what the consumer may hear at home on a typical audio system. However, for a well-designed room and main monitor setup, close field units shouldn't have to be used to get a decent stereo image, minimize unwanted acoustical effects in the control room, or produce results that sound decent elsewhere.

If the main monitors and the control room acoustics do not give you accurate results, there is something wrong with one, the other, or both. (These are examples of less than perfect room and monitor design that have produced a good and successful product.)

When the acoustic design and main monitors are right, you'll make use of them a good deal of the time. You simply won't need auxiliary monitors to make up for main system deficiencies, because those deficiencies have been minimized. The speaker is only there to monitor the way through the recording chain. Like all technology in the studio and the control room, it is a tool—a point often missed.

But certainly, today's tools must be more sophisticated to meet the challenge, and so it is with monitors. You must be able to listen accurately through the chain with sufficient headroom and bandwidth, as we are now concerned with audio below 40Hz.

Because a monitoring system cannot reproduce frequencies below 30Hz to 40Hz doesn't mean that those frequencies are absent from the program. Therefore, it is critical, especially in the latter stages of production, that what LF information ends up on the digital master is sufficiently attenuated so as not to destroy the increasing number of hi-fi systems that are capable of responses down to 20Hz. And, of course, it is absurd that you should be able to monitor more accurately at home than you can in the studio.

Another philosophy

Andy Munro, of Windmill Munro Designs in England, goes along with much of Neil Grant's philosophy, especially regarding the enthusiasm for specifying cones and domes in the digital age. But, he has taken a slightly different approach.

Acknowledging that commercial top-end hi-fi drivers may offer the quality that's required, but generally lack the power-handling capacity, Munro has adopted a very sophisticated multi-driver design. His design incorporates a larger number of commercial drivers resulting in a 4-way rather than a 3-way system, and one in which five drivers are used in all. They are linked in a design that ensures each driver is running well below its rated power-handling capacity.

Munro is particularly concerned—as one might expect in view of the large number of drivers—with phase linearity, and has put a lot of effort into this area. As a result, despite the physical size of the monitors, they behave very much like point source systems.

The BSS Audio FDS 360 crossover used in Munro's monitors drives a 82-foot soft-dome, latex-doped tweeter that features ferrofluidic cooling to increase damping and enables the unit to handle more power and decrease thermal compression; a similarly specified 1-inch high/mid driver (both of these drivers have their own critically-damped chambers in their rear housings to optimize transient response, and both are rated at 1kW peak); an 8-inch JBL cone driver for low/mid applications; and a pair of 15-inch JBL high-compliance bass drivers with long linear excursion capability.

So much for loudspeaker design. But what about the room? During the past few years there has been a significant trend toward high-diffusion room designs. This is in contrast to the heavily "trapped" designs of the 1970s and early 1980s. Grant has developed a technique, in conjunction with RPG, that relies primarily on geometry. The angles of the walls are of primary importance. Rather than absorbent techniques (to avoid early reflections reaching the listener), wall diffusers return the energy into the room within a given time-window where it is reintegrated with the direct sound from the monitors.

But, as it is entirely diffused, it does not negatively react with the direct sound, so there is no comb filtering and no interference with the stereo image. As a result, stereo is excellent (helped by the fact that the speaker cabinets are mounted so as not to move relative to each other) and relatively unaffected by listener position, and the rooms are loud because little of the energy is absorbed. Grant uses absorption to tailor the decay time rather than to remove reflections.

Munro expresses his design principle in different terms, but the results are no less valid. He points out that monitors are placed all too often in the top corners of the control room. Here, in a perfect world, acoustic loading would generate a 9dB increase at the bass end—assuming right-angled corners—or 3dB per boundary. To begin with, Munro tries to mount speakers tangentially, making the wall angles oblique to minimize this effect.

But, what is so often neglected, according to Munro, is that the rear corners are loaded, too, by the sound arriving from the monitors. This can result in major problems. Even if there's an inverse square law reducing rear-wall effects, you can easily end up with signals coming back from the rear corners that are just a few decibels down. It is this, he feels, rather than the conventional explanation of standing waves, that causes trouble. Like Grant, Munro is concerned about a control room having "oomph" or "wellie," as some call it. He finds that phase linearity makes a major contribution here to the overall "punch."

However the philosophy is implemented, this approach to design, which considers the monitors and the control room as an integrated system rather than as two disconnected entities, is expensive. And, we have essentially looked at only a few examples aside from expressing the general theory. But, even so, it's expensive only in terms of what one might usually expect to pay for monitoring, rather than in absolute terms. Even if the monitors/crossovers/amps combination alone costs $20,000, it's still only a tenth of what your console is likely to cost. And, after all, isn't selecting good monitoring probably the most important hardware decision in the studio? Without it, how can you accurately evaluate the work being done.

RE/P

Headphone Distribution Systems

By Gary Davis and John Windt

Cue headphones should present a clear accurate portrayal of the sound at comfortable levels for each performer.

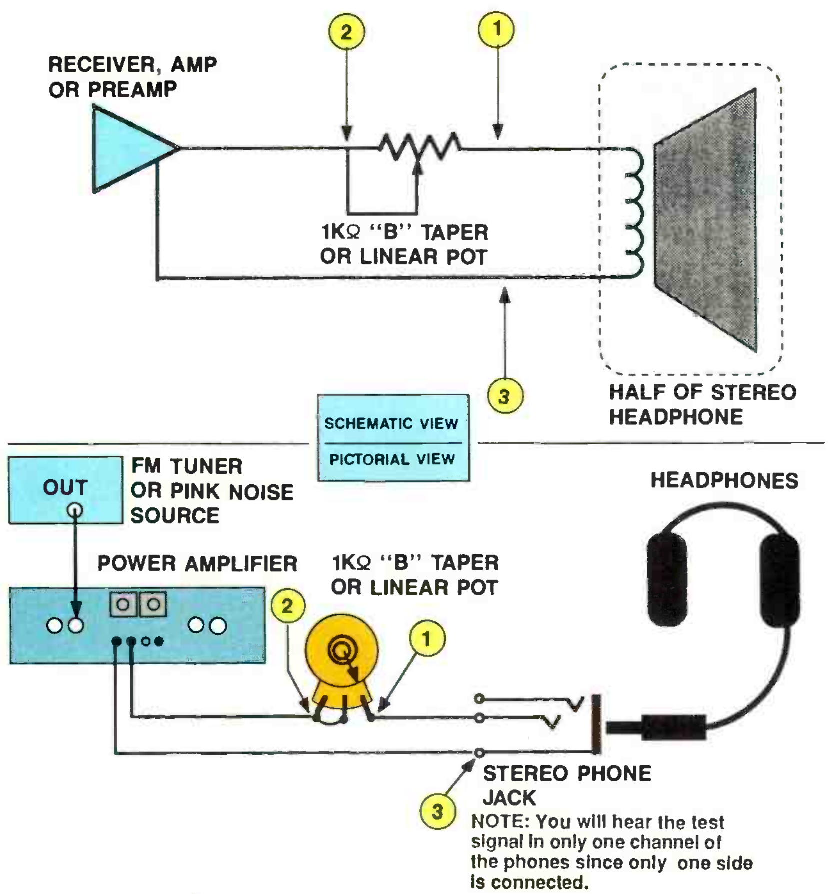


Figure 1. Test setup for measuring headphone impedance.

Every studio is faced with the job of getting sound to performer's cue (or foldback) headphones. It seems pretty simple...until you actually try to wire up such a system.

Why not just connect as many phones as you need directly to the output of a power amp? It's more complicated than it seems. For one thing, you will usually have to accommodate different types of headphones, each with a different sensitivity. Some headphones will be too loud, and some too soft at any given amplifier setting. If low and high impedance headphones are used together, by the time the amp level is turned up high enough for the needed level in the high-Z phones, the low-Z phones will probably burn out from too much power. (Most low-Z phones are inexpensive, made for use with hi-fi receivers and capable of dissipating only about 1W.)

If musicians rely upon the acoustic sound field in the studio to synchronize their playing with the other performers, their timing will be delayed by the period it takes for the sound to travel across the studio. (Sound moving across a 25-foot space takes about 22ms, based on the nominal speed of sound, which is 0.885ms per foot at sea level, 70°F.) Using headphones, the musician hears the sound from the mic at the other side of the studio in just a fraction of a millisecond, allowing much tighter tracks to be performed.

In some situations, headphones are essential for the performers to hear certain

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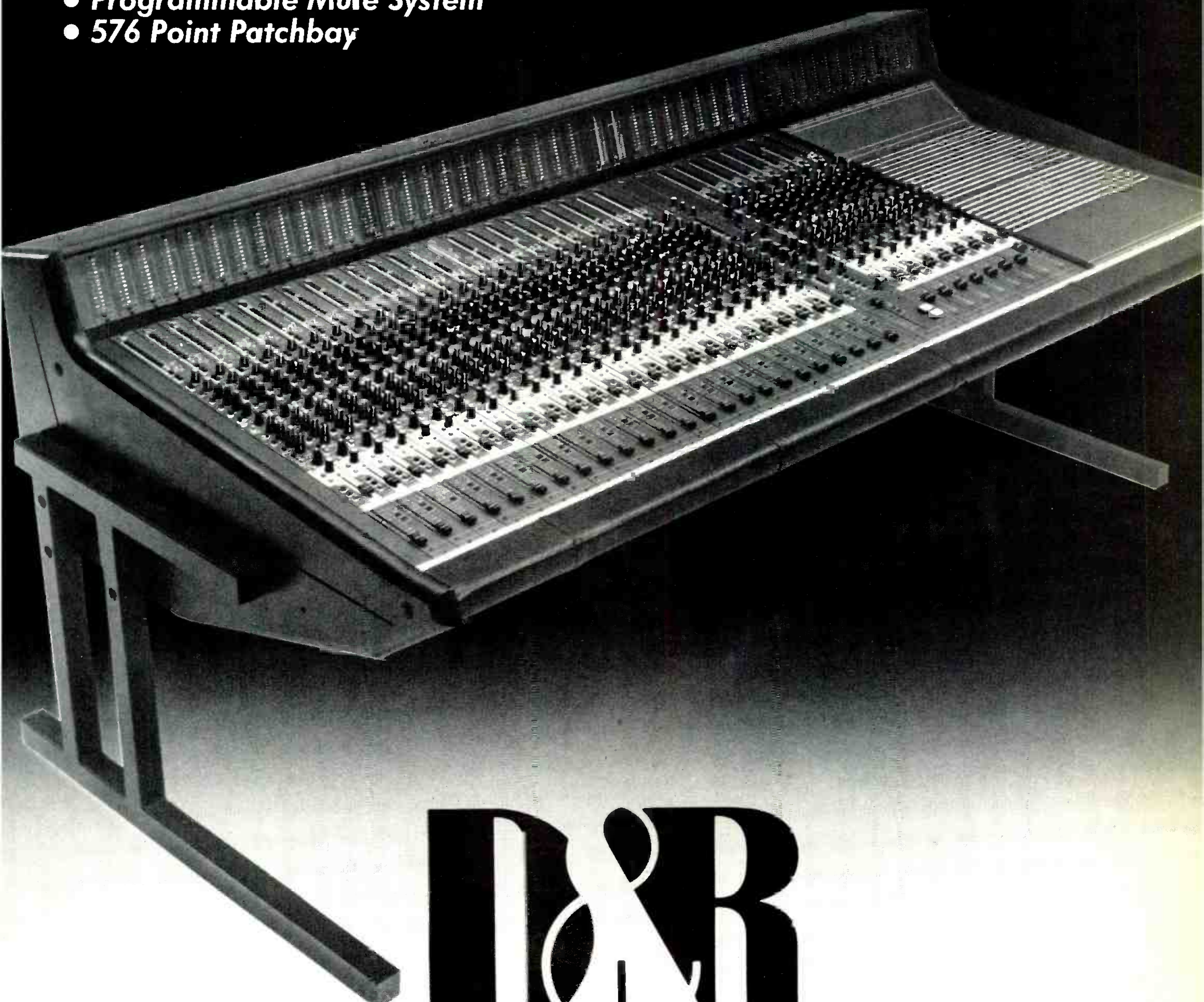

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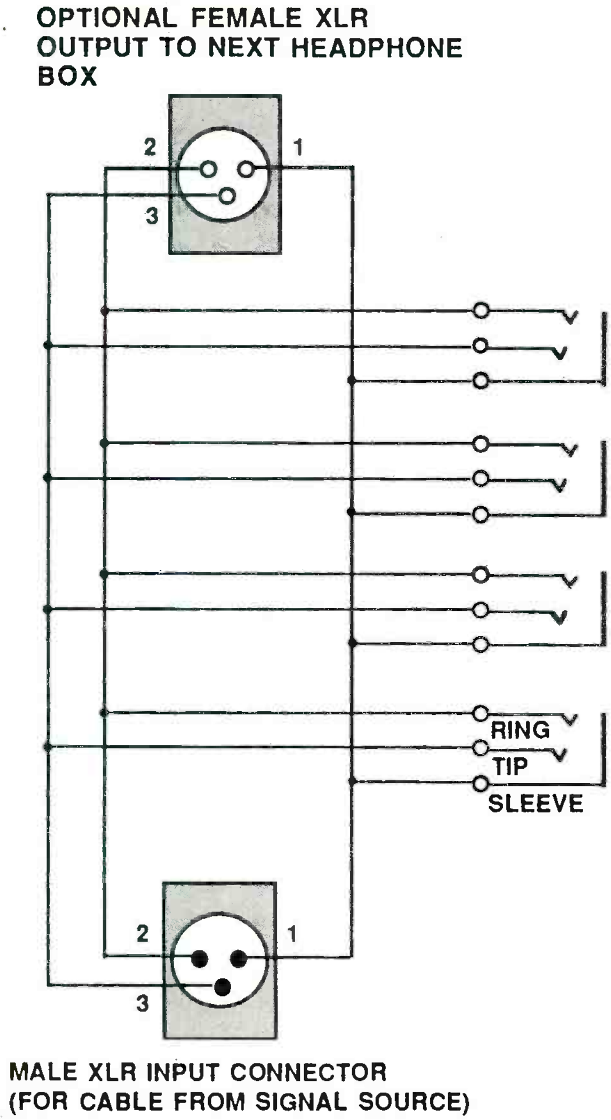


***Figure 2.** A simple, parallel-wired headphone distribution box.*

instruments, such as drums that are acoustically isolated in a special booth and electronic instruments that are directly connected to the mixing console or previously recorded tracks.

Regardless of the reason for their use, the studio's cue headphones must present a clear, accurate portrayal of the sound at a level that is comfortable for each individual performer. The headphone system also must be reliable if the studio is to function efficiently.

The use of individual headphones and power amplifiers does not guarantee that the overall headphone system will be reliable. It is always possible for someone to abuse or inadvertently misuse the headphone distribution system.

For example, if a mono phone plug is inserted in a stereo phone jack, one "hot" side of the circuit will be shorted to ground. Unless the system is protected against this common mistake, the interconnecting wiring, and possibly the amplifier can be damaged. If you believe that because you stock only stereo phones in your studio you're immune to this type of problem, think again.

Some musicians, particularly string players, like to carry their own (mono) phones. They use one ear for hearing their own instrument and a single phone for the track. Besides, it's always possible for a guitar player to inadvertently insert a mono phone plug in your headphone distribution jack. This is one of the potential pitfalls in a system used by many different people of varying degrees of technical knowledge of, or concern for, your studio equipment.

A good headphone distribution system will alleviate the potential problems involved in using different headphones, enable performers to hear what they're doing better, and improve the tracks recorded in the studio, as well as the reliability and efficiency in the studio. Once it is installed, the distribution system can be used for other tasks too.

Types of headphones

To understand the requirements of your headphone distribution system, you should be aware of the types of headphones used. There are three general categories.

1. Low impedance phones (which are truly low impedance) present about an 8Ω load to the source.
2. Medium impedance phones, which may be labeled "low impedance," can be rated at 8Ω, but actually present several hundred ohms load impedance.
3. High impedance phones that are rated at 600 to 10kΩ (typically 2kΩ load impedance), may also be mislabeled "low impedance."

From these definitions, it is clear that what is written on the headphone box is not always what you are getting in a headphone.

True low impedance phones are less desirable because of low quality construction, which justifies the low cost—often less than $50. These headphones offer poor sound quality and questionable reliability.

These low impedance phones may work with hi-fi pre-amps, where they generate plenty of volume with relatively little power present at the preamp's "headphone" output jack. Nonetheless, these "hi-fi" type, low-Z headphones do sometimes find their way into the studio, so your studio should be equipped to deal with them.

In contrast to low impedance headphones, medium impedance headphones generally feature high-quality construction, are reasonably efficient and reliable and the best choice for the studio.

High impedance phones often have good fidelity, and are lightweight and thus preferred by string players. But they require a very large power amp to obtain enough voltage swing (not power output because the high-Z phones don't draw much power) to get the desired level in the phones. A problem arises when medium- or low-impedance phones are connected in the same system with high-impedance phones. The lower impedance units, which draw a lot of power as the input voltage is turned up, will become excessively loud or even fail before the players using the high impedance phones hear a high enough level.

There are means to "even out" the level differences although these methods increase the cost to the headphone distribution system. The first step to do, therefore, is to determine the actual load impedance of any headphones you'll be using. You can make a simple impedance tester as shown in Figure 1.

To do so, wire a 1kΩ B-taper (linear) potentiometer to a stereo phone jack and connect it to the output of a power amp as shown. The amplifier should be driven with a typical program source (or pink noise generator). *Do not use a sine wave source.*

The reason you don't want to use a sine wave (or other single-frequency) test signal is that the impedance of any pair of headphones will tend to vary tremendously with frequency. A single frequency test would not yield a representative average impedance. You would have to make numerous tests at many different frequencies and tediously average the results. What you really want to know is the result of the average impedance as it affects sensitivity, so a program signal or wide-band pink noise signal is ideal.

Plug the headphones into the tester's jack, set the 1kΩ pot at full clockwise rotation (i.e., the pot shorted out) for the loudest output and turn up the power amp output until you hear a comfortable level in the phones. You'll only be listening to one side if they are stereo headphones, but that's all you need here. Now rotate the pot to the other extreme. If the phones volume goes to zero, you have low impedance phones. If the volume does not change, you have high impedance phones. If there is some change in level, they are medium impedance phones.

If you want to know the exact impedance of your phones, hook them up as shown in Figure 1, with a pink noise signal.

Follow the steps below:

1. Connect a voltmeter across the headphone (test points 1 and 3) and adjust the pot for maximum voltage.
2. Turn the pot down to obtain half that maximum voltage. Verify that this is the mid setting by now measuring the voltage drop across the pot (test points 1 and 2), which should be the same as that across the phones (test points 1 and 3).
3. Disconnect the pot from the headphones and the amp. Measure the resistance across the pot (test points 1 and 2). This resistance is equal to the impedance of the headphones.

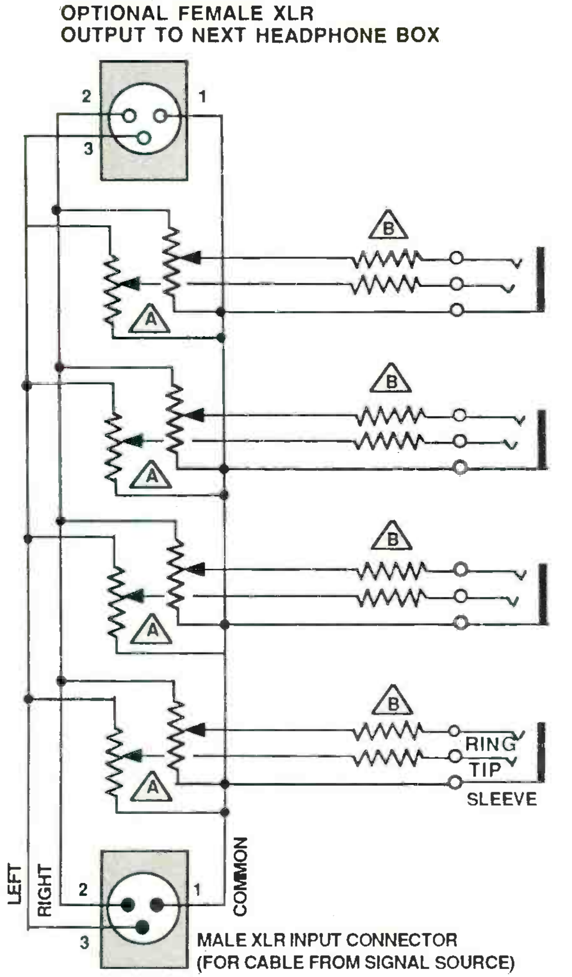
It is recommended that you test each different model of headphones using this technique. A given manufacturer will often build similar looking phones, sold for similar prices, yet the different models will have different impedances. Once you know what you've got, then you can examine your options with regard to how you arrange them.

Types of systems

The three different headphone distribution systems are outlined below.

1. If you have all medium or all high impedance headphones (but all the same), you can use a simple parallel-wired box with multiple phone jacks (see Figure 2). (Note: This box offers no short-circuit protection for the amplifier or the interconnecting wiring. It should be used with headphones all having the same actual impedance. Use caution to avoid amplifier overload with low impedance headphones.)

Caution: *If a mono (tip/sleeve) phone plug is inserted in the simple parallel-wired system illustrated in Figure 2, it will short the right channel of the power amp to ground. This will kill that channel in all the headphones in the system. Depending on the make and model of power amp, the short also may damage the amplifier.*



A: HEAVY DUTY VERSION: USE TWO 100 OHM, 12 WATT POTS PER OUTPUT JACK (L, R); Ohmite 0112 or equivalent.
STANDARD VERSION: USE TWO 1 KOHM, 1/2 WATT POTS, OR DUAL-GANG POT PER OUTPUT JACK.

B: HEAVY DUTY VERSION: USE 10 OHM, 5 WATT RESISTORS; Ohmite 4561 or equivalent.
STANDARD VERSION: USE 510 OHM, 1/2 WATT RESISTORS.

Figure 3. *Schematic for headphone distribution box with adjustable level for each output.*

If you have a combination of different impedance phones, want to be able to handle these at some time in the future, or if you want individual control of the volume in different phones, you'll need a distribution system with potentiometers for each pair of headphones. These adjustable distribution systems, in turn, can be built in two versions:

2. Heavy-duty construction for low impedance phones. This system is also capable of driving monitor loudspeakers.

3. Standard construction for general-purpose use.

Both of the adjustable systems (2 and 3) have the same overall design, but use resistors and potentiometers with different values. This is illustrated in Figure 3. The heavy-duty version is much more costly to build, but you get what you pay for. If you opt for the heavy duty version, be sure you use the recommended, high-wattage resistors and pots. Low wattage components may work for a while, then burn out when they're needed most (in the middle of a hot session with performers "cranking up the level").

If you're using the headphone distribution system with low impedance phones, use the heavy-duty components and all metal boxes, preferably diecast. The pots must dissipate a lot of power in order to avoid "cooking," and rely upon the box to serve as a heat sink. In this box, individual left and right channel pots must be used because dual pots are not available in a high enough power rating.

For medium to high impedance phones, almost any type of box may be used, although we still prefer metal for durability. Remember, the box sits on the studio floor, and it is likely more than one person will step on it.

Amplifier power ratings

The choice of a power amplifier(s) will be dictated by the type of headphones in use. If you're using medium- or low-impedance phones, you must guard against use of too large an amplifier that could easily burn out the phones. Generally, an amplifier rated at 50W to 100W per channel is sufficient. On the other hand, if high impedance phones are being used, a 200W per channel amp may be necessary simply to obtain an adequate level in the phones, regardless of how few or how many phones are connected to the amp.

Wiring

Never use shielded cable for the long wire runs because the center conductors tend to be too small in diameter, and there is too much reactance. Instead, use "zip cord" (standard lamp wire) or a similar insulated, 2-conductor wire. With a medium impedance system, 18- to 22-gauge wire can be used. For a low-impedance system, 16-gauge or heavier wire should be used. The longer the cable and the more phones in use, the larger the wire diameter you'll need. High impedance phones draw little current, so almost any gauge wire can be used.

Build out resistors

Figure 4 is an illustration of a typical headphone distribution system. Notice a build-out resistor shown in series with each power amplifier output. Should the system accidentally be shorted out, or simply overloaded from too many headphones (particularly with the type of box illustrated in Figure 1), the build-out resistor on each amp channel output will maintain a minimum load impedance so the amplifier does not overheat or shut down. This serves not only to protect the amplifier, but also to keep the interconnecting wiring from overheating.

These resistors, however, will run warm and become very hot if there is a short, which is why they should be mounted on a metal panel or heat sink in the equipment rack. Choose a location where they won't melt insulation or ignite anything. Each build-out resistor should be equal to the amplifier's lowest rated load impe-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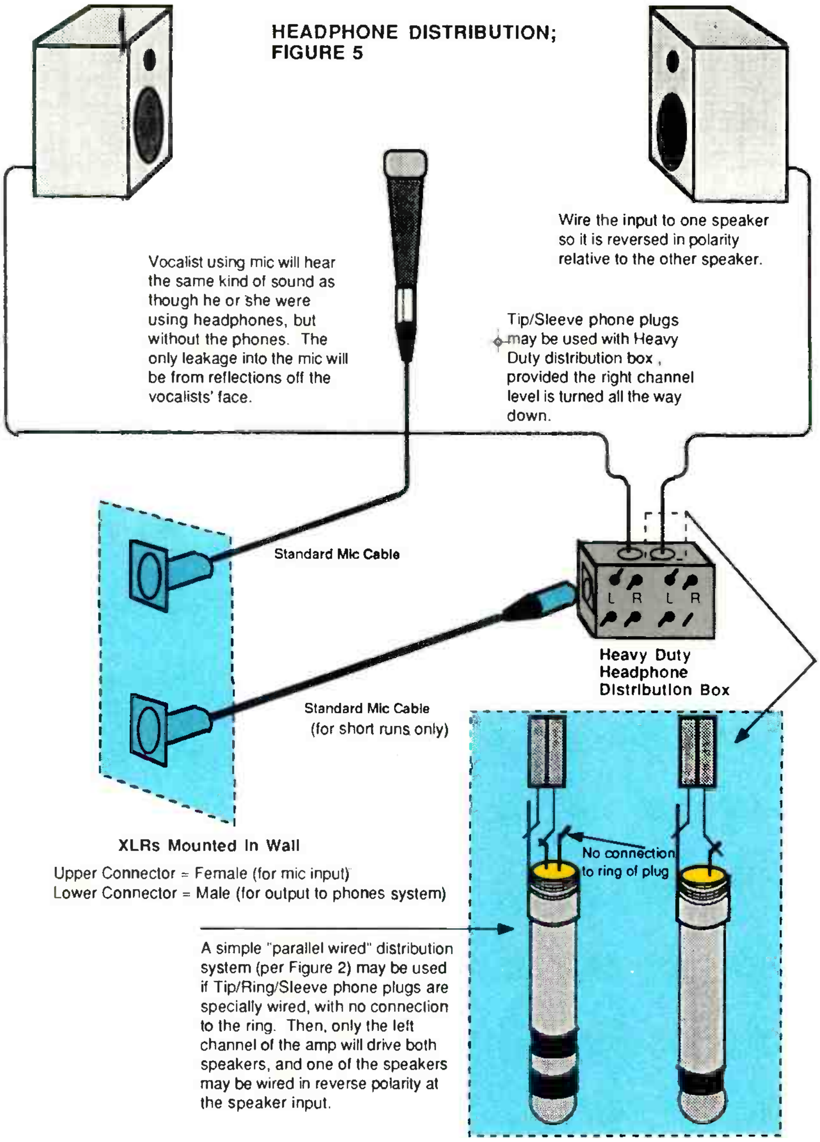

Figure 5. *Pictorial of headphone distribution system used to drive reverse-polarity speakers to eliminate cue phones.*

dance, and should be capable of dissipating a third of the amplifier's rated power output at that impedance. For example, an amp rated at 200W per channel into 4Ω will require a 66W, 4Ω resistor on each channel.

Connectors

Any standard, chassis-mount stereo phone jacks may be used for the headphone connections. For connection to the amplifier (and interconnection to additional headphone distribution boxes), we prefer XLR connectors.

XLRs avoid the chance of accidental shorts (as can happen when a tip/sleeve phone plug is inserted in a tip/ring/sleeve jack) or disconnections (XLRs are locking connectors). However, to avoid feeding the output of the power amp into a microphone, the wall mounted XLRs should be male. Standard mic cable may then be used to interconnect the distribution boxes to the wall feed, and to one another (if the optional output connectors are installed in the boxes).

Headphone connections to amps

With the heavy-duty adjustable level boxes (Figure 3), regardless of the type of headphones used, the impedance presented to the amp is such that a maximum of 25 pair of phones may be used with a 4Ω rated output (or 12 pair of phones with an 8Ω amp). And with the standard version adjustable level boxes, up to 10 times that number of headphones may be used.

If you use the simple parallel-wired boxes (Figure 2), the actual headphone impedances will determine the number of phones that can be used with a given amp. If all phones are low impedance, no more than five pair should be connected to a single 4Ω amp. If all phones are medium impedance (assuming 250Ω), then about 35 pairs may be used. With high impedance phones, there is no practical limit on the number which may be used.

Other applications

If you need a spot monitor, or several small monitor speakers, you can simply plug them into the headphone distribution system. Of course, the speakers should be rated at at least 8Ω, and the same precautions that apply to the use of low impedance headphones apply here. *Don't plug in too many speakers.*

Some vocalists (or musicians) feel constrained with cue headphones. You can substitute small speakers instead. Select a well-matched pair of small, spot monitor speakers that can be mounted on mic stands. Place them about 4 to 5 feet apart, on either side of the microphone, aiming the speakers at each other (see Figure 5). Locate the mic and speakers near the middle of the studio, away from walls or other large objects. Connect the cable on the back of one speaker so it is reversed in polarity relative to the other speaker.

There are optional methods to plug the speakers into the headphone distribution system. The signal fed to the speakers must be a mono program. You can use standard tip/sleeve (mono) phone plugs *if and only if* you are using the heavy-duty distribution box with separate pots for each side of each jack. Turn down the right channel all the way for the jack feeding each speaker, and turn up the left channels for both speakers to the same setting. The resistance in the pots will prevent the dead-short across the ring and sleeve connection of the jack from actually shorting the amp and other headphones in the system (worn by other performers) and can continue to be used normally.

It is possible to use a simple, parallel-connected distribution box (Figure 2) if you use tip/ring/sleeve phone plugs. Of course, the feed you want is mono, so connect each speaker to only one side of the line (the tip is preferred). In this case, you'll be loading one side of the stereo phones circuit more heavily than the other, so it may not be possible to use other headphones at the same time.

Cue headphones must present a clear, accurate portrayal of the sound at a comfortable level for each performer.

Once the speakers are connected, drive them with a test tone or program source, activate the mic, and carefully move the mic back and forth between the two spot monitors until the engineer in the control room indicates there is minimum sound from the speakers leaking into the mic (there should be nearly none).

What you have done is to locate the mic in the acoustic null point between the two identical, reversed-polarity sound sources. The sound fields from the two speakers cancel out at the mic element, which therefore, does not respond. However, the performer can stand slightly off-center from this null, or even if standing directly in front of the mic, the distance between one's ears will place each ear sufficiently off center that the speakers can be heard for monitoring; sound from the performer will excite the mic with no cancellation. This situation is similar to wearing cue phones, without the phones.

It is also possible to drive a guitar or instrument amplifier from the headphone distribution system or a "Leslie" type rotating speaker system. Use one of the models of the distribution system with local volume controls for these applications.

RE/P

Editor's note This article was adapted from Pantechnicon, the Ramsa newsletter.

Functional Equations for Recording and Production Engineers

By Jay Petach

A basic familiarity with the various equations that convert tempos to timings and delay settings can be useful to engineers working on a variety of audio projects.

If you've ever wanted to set a delay line in rhythm with a song; if you've ever needed to know the number of measures required to fill a scene of film, or the proper tempo to make jingles last 28 seconds; if you've wanted to know how varispeeding the multitrack a half-step will affect the tempo or how many inches of tape are in a beat of music, here are some simple calculations that can be made to quickly provide such information.

Delay times

Setting delay line in rhythm with a piece of music can be accomplished by first detemining the length of time per

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beat. Delay time is normally expressed in milliseconds, while tempo is usually indicated in beats per minute. Since there are 60 seconds in a minute, the formula becomes:

$$\text{Delay time, } t = 60/T$$

Where: t = delay time in seconds, T = tempo in beats per minute.

For example, in a tempo of 144 bpm:

$$t = 60/144 = 0.417 \text{ seconds, or } 417/\text{ms}$$

If $T = 160$ bpm, then:

$$t = 60/160 = 0.375 \text{ seconds, or } 375/\text{ms}$$

Note that in the two previous examples setting a DDL, to these delay times would produce a delay equal to a *single* beat. To determine the delay time equal to two beats, simply multiply the single beat delay time by two.

In the first example, with $T = 144$:

$$t \text{ for two beats} = 2 \times (60/144) = 833 \text{ ms}$$

To determine the delay time equal to half a beat, halve the single-beat delay time.

Similarly, for a delay time equal to a third of a beat (triplet), divide the single beat delay time by 3.

Delay time determinations for a full measure

To determine the delay time for a full measure of music, simply multiply the time for one beat by the number of beats per measure:

$$t_1 = 60 \times b/T$$

Where: t_1 = delay time in seconds for a single measure
b = the number of beats per measure
T = tempo in beats per minute

Therefore, for a tempo of 144 bpm, and time signature of 4/4, a measure would last:

$$t_1 = (60 \times 4)/144 = 1.67 \text{ seconds}$$

In another example, if the tempo is 138 bpm and the time signature 3/4, a measure would last:

$$t_1 = (60 \times 3)/138 = 1.30 \text{ seconds}$$

Note that it is the top number of the time signature that indicates the number of beats per measure. The bottom number of the time signature indicates the note value that receives a beat: 8 = eighth-note; 4 = quarter-note; 2 = half-note, etc.

Determining scene times

To determine the time required for a number of measures, simply multiply the time required for one measure by the total number of measures (assuming all measures are at the same tempo).

$$t_m = (60 \times b \times M)/T$$

Where: t_m = time for M measures, in seconds
b = number of beats per measure
M = total number of measures
T = tempo in beats per minute

Therefore, for a tempo of 152 bpm, and a time signature of 6/8, 15 bars of music would last:

$$t_m = (60 \times 6 \times 15)/152 = 35.5 \text{ seconds}$$

For a tempo or quarter-note = 144, 27 measures of 4/4 would last:

$$t = (60 \times 4 \times 27)/144 = 45 \text{ seconds}$$

Calculating the number of measures

If the length of time and the tempo are known, the equation can be inverted to calculate the number of measures needed to fill that particular length of time at that tempo:

$$M = (T \times t)/(60 \times b)$$

Where: M = number of measures
T = tempo in bpm
t = time in seconds
b = the number of beats per measure

Therefore, 59 seconds of 4/4 at a tempo of 160 bpm would require:

$$M = (160 \times 59)/(60 \times 4) = 39.3 \text{ measures}$$

If the tempo was 120 bpm, the length of the scene three minutes and 50 seconds, with a time signature of three/four, the number of measures would be:

$$M = (120 \times 230)/(60 \times 3) = 153.3 \text{ measures}$$

Determining tempo

Sometimes it may be necessary to calculate how the alteration of other parameters will affect the tempo. The equation can again be inverted:

$$T = (M \times 60 \times b)/t$$

For example, in order to be completed in 29 seconds, a TV underscore having 16 measures of 4/4 would have to be performed at a tempo of:

$$T = (16 \times 60 \times 4)/29 = 132.4 \text{ bpm}$$

If the same piece had to be performed in 28 seconds, the new tempo would be:

$$T = (16 \times 60 \times 4)/28 = 137.1 \text{ beats/minute}$$

Frame beat metronomes

Many studios use metronomes that express tempo in film frames per beat, instead of beats per minute. In order to use the equations that have been discussed, it is necessary to convert the tempo to bpm. Since frame per beat metronomes express tempo in tens, units, and eights of a film frame, there is a 2-step conversion procedure.

Step one involves converting the eights of a frame to a decimal equivalent. This can be done by multiplying the number in the display (0 thru 7) by 0.125. Step two is simply to add this number to the tens and the units, and divide 1440 (24 fps × 60) by the result:

$$T = 1440/F$$

Where: T = tempo in beats per minute
F = tempo in frames per beat, with decimal equivalent

For example, if the frame per beat metronome setting is 1/4/7, what is the equivalent tempo in bpm?

Step one: Take the eights reading and convert to a decimal equivalent:

$$7 \times 0.125 = 0.875$$

Step two: Add the result from step one to the tens and units to get the tempo in frames per beat with a decimal equivalent:

$$F = 0.875 + 10 + 4 = 14.875$$

Next, divide 1440 by this sum to get the tempo in beats per minute:

$$T = 1440/14.875 = 96.8 \text{ bpm}$$

Sometimes it may be necessary to work the other way. For example, what is the proper setting on a frame per beat metronome to produce a click of 132 bpm?

When solving for the proper setting on the frame per beat metronomes, step one is to solve for F. In the current example with a tempo of 132, F = 1440/132 = 10.9.

Step two then involves converting the decimal equivalent to eights. This can be done by dividing the number to the right of the decimal point by 12.5. In this case, 9/12.5 = 0.72. Then add this number to the tens and the units: 0.72 + 10 = 10.72 or 1/0/7 on the frame per beat metronome. (Note that this tempo is actually 132.4 bpm, because the 0.02 had to be dropped. However, this is the closest setting to 132 using a frame per beat metronome.)

Changing time signatures

Quite often a piece of music will have an odd bar with a different time signature. The best way to handle this situation is to convert the odd bar to an equivalent bar of the main time signature.

For example, in a piece of music with 20 measures of 4/4 and one measure of 2/4, b would be 4, and M 20.5. The half a measure represents the two-beat bar, which is half of a typical four-beat bar.

In another example, if there were 30 measures of 4/4, two measures of 5/4, three measures of 3/4, and four measures of 2/4, the value of b would again be 4, since 4/4 is the predominant time signature. The value of M would be:

$$\begin{aligned} M &= 30 + ((2 \times 5) + (3 \times 3) + (4 \times 2))/4 \\ &= 30 + (10 + 9 + 8)/4 \\ &= 30 + (27/4) \\ &= 36\tfrac{3}{4} \text{ equivalent measures of } 4/4. \end{aligned}$$

Accelerando and ritard

The equations provided so far assume that the tempo remains constant; for much of today's commercial music this is a reasonable assumption. If there are several tempos within a piece, however, each section must be treated individually.

For example, the verse section of a song has 16 measures of 4/4 at a tempo of 120. The tempo suddenly changes on the chorus to 138 for 10 measures of 4/4. How many seconds do the 26 measures take to be performed?

First the verse:

$$t = (60 \times 4 \times 16)/120 = 32 \text{ seconds}$$

Then the chorus:

$$t = (60 \times 4 \times 10)138 = 17.4 \text{ seconds}$$

Next, add the two times together:

$$t = 32 + 17.4 = 49.4 \text{ seconds}$$

Sometimes, however, there will be a gradual transition from one tempo to another. If the tempo accelerates, the transition is called an accelerando; if it slows down, the transition is called a ritard.

In the previous example, if the verse had sped up during the last two bars in order to prepare for the tempo change into the chorus, the 26-bar phrase could be thought of as having three distinct sections.

First, the initial 14 measures of the verse at 120 bpm:

$$t = (60 \times 4 \times 14)/120 = 28 \text{ seconds}$$

Then the next section would be the last two bars of the verse as the tempo is accelerating from 120 to 138 bpm.

The easiest way to deal with this transition is to assume a linear acceleration, and calculate an *average* tempo for the transition measures. To obtain the average, simply add the two tempos and divide by two:

$$^{T}\text{Average} = (120 + 138)/2 = 129 \text{ bpm}$$

Therefore, the time for the two-bar transition is:

$$\begin{aligned} t &= (60 \times b \times M)/T_{Average} \\ &= (60 \times 4 \times 2)/129 \\ &= 3.7 \text{ seconds} \end{aligned}$$

Then add these two section times to the time for the chorus:

$$t = 28 + 3.7 + 17.4 = 49.1 \text{ seconds}$$

The procedure would be the same if a ritard were encountered instead of an accelerando.

Tempo and pitch

Whenever tape speed is altered, tempo, pitch and timbre, among other things, are affected. Whatever parameter is most noticeably affected depends upon the program material. However, despite all subjective interpretations, there is a mathematical relationship between tempo change and pitch change in varispeed mode.

$$T/T^* = P/P^*$$

Where T = original tempo
T* = tempo after varispeed change
P = original pitch or frequency (Hz)
P* = pitch or frequency after varispeed (Hz)

For example, assuming that the original pitch was A = 440Hz, how will changing the tempo from 152 to 180 bpm affect the pitch?

Inverting the equation to solve for the new pitch, P*, the equation becomes:

$$P^* = (440 \times 180)/152 = 521\text{Hz}$$

The new frequency of 521Hz is very close to a C (523.25Hz). Therefore, the tonality of the piece was raised approximately three half steps from an A to a C.

Sometimes it is useful to varispeed a multitrack in order to transpose the pitch up or down by a half step; again there is a mathematical relationship between tempo and pitch.

In order to move the pitch a half-step higher for a piece that's already recorded on tape, the tempo is increased by 5.9% to become 105.9% of the original. To move the pitch a half-step lower, the new tempo is 94.3% of the original.

Expressed as an equation:

$$T^* = (1.059)^S \times T$$

Where: T* = tempo after varispeed to raise pitch s number of half steps
T = original tempo
s = number of half steps of pitch change

If the tempo was originally 176 bpm, in order to raise the pitch on tape by one half-step, the new tempo would be:

$$T^* = (1.059)^1 \times 176 = 186.4 \text{ bpm}$$

In order to lower the pitch by any number of half steps the equation becomes:

$$T^* = (0.943)^S \times T$$

If the original tempo was 208 bpm, in

order to lower the pitch by one half-step, the new tempo would be:

$$\begin{aligned} T^* &= (0.943)^1 \times (208) \\ &= 196.1 \text{ bpm} \end{aligned}$$

If the original tempo was 208 bpm, in order to lower the pitch two half-steps or one whole-step, the new tempo would be:

$$\begin{aligned} T^* &= 0.943 \times 0.943 \times 208 \\ &= 184.9 \text{ bpm} \end{aligned}$$

Tempo and tape speed

The relationship between change in tempo and change in tape speed is identical to the previous tempo/pitch relationship. It's logical that both sets of parameters have this same direct relationship.

If you speed up the multitrack, you also increase the tempo and increase the frequencies recorded on the tape.

Expressed as an equation:

$$T/T^* = S/S^*$$

Where T = original tempo
T^* = new tempo
S = original tape speed (ips)
S^* = new tape speed (ips)

For example, how would a tempo of 120 bpm be affected if the speed changed from 15ips to 13.75 ips?

Solving the equation for the new tempo, T^*, the equation becomes:

$$\begin{aligned} T^* &= (13.75 \times 120)/15 \\ &= 110 \text{ bpm} \end{aligned}$$

Tempo in inches per beat

There are some applications where it's helpful to know how many inches of tape there are to a beat of music. The relationship between tape speed and tempo can be expressed as follows:

$$L = (60 \times S)/T$$

Where: L = length of one beat (inches)
S = tape speed (ips)
T = tempo (bpm)

For example, at a tape speed of 7.5ips and a tempo of 176 bpm, a single beat of music would represent a tape length of:

$$\begin{aligned} L &= (60 \times 7.5)/176 \\ &= 2.55 \text{ inches} \end{aligned}$$

One possible application for the above example might be to determine the correct amount of tape to insert for a pause equal to a certain amount of beats.

In another example, if the tempo was 160 bpm and the distance between the record and playback heads is 1.81 inches, what speed would the machine have to be running to produce a quarter-note slapback?

Solving for tape speed, the equation becomes:

$$\begin{aligned} S &= (1.81 \times 160)/60 \\ &= 4.83 \text{ips} \end{aligned}$$

For the above example to be of practical application, the tape machine would have to be equipped with a velocity indicator; most newer machines incorporate this feature.

Of all the material presented, at least some of the equations should be useful in the work you're doing. In all cases the calculations involve simple arithmetic, and can be easily done on a pocket calculator. We can all use a few of the formulae presented here to add a few tricks to our repertoire. RE/P

Circle (23) on Rapid Facts Card

Interconnecting Audio Equipment

By Cal Perkins

In system interfacing, what works in theory may differ from what works in practice.

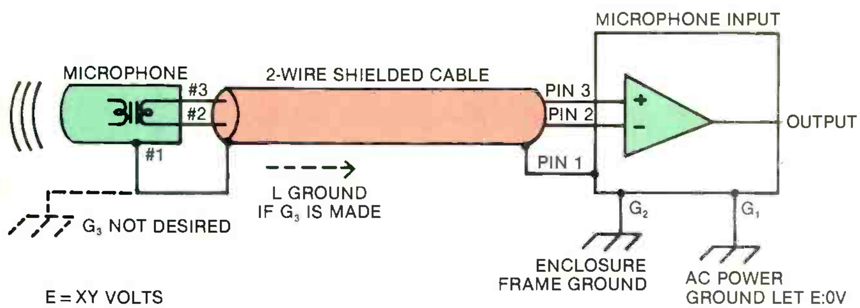

Figure 1. *Interconnections for a floating source such as a microphone. Note that the microphone shield is bonded to the microphone case. It also should be bonded to the microphone input chassis.*

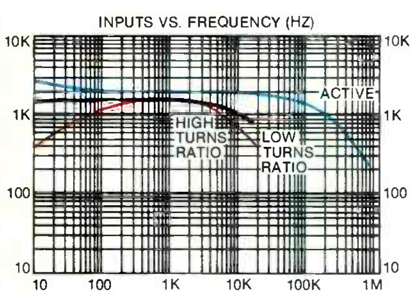

Figure 2. *Typical input characteristics of the three most commonly used input-stage circuit realizations.*

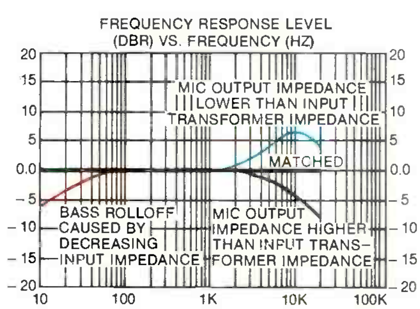

Figure 3. *Frequency response of a typical microphone connected to a high turns ratio input transformer as a function of the microphone's output impedance.*

If you've never had problems interfacing audio equipment into a studio, you've never installed any equipment. When it comes to designing and building a modern facility, theory and practice often go in divergent directions.

A system design may look elegant on paper, yet turn out to be a nightmare when all of the equipment is installed and turned on. That beautifully wired patchbay may contain more hum than a power transformer. The neatly bound cables running along the edge of the rack may look great, but might end up acting like the secondary of a transformer-coupling, ac-induced hum into other lines, equipment or even the technical ground. In short, when you're faced with building or remodeling an audio system, theory alone may not help you prevent problems.

Following a signal through the entire system begins with the most commonly used transducer to convert sound to an electrical signal—the microphone—and continues through to the final electrical component—the power amplifier. For purposes of this discussion, disc systems, tape decks, musical instruments and non-me-

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chanical signal-producing components will be considered signal processing devices.

Any system depends on the following four major elements:

- the components, which include microphones, mixers, signal processors, amplifiers and speakers;
- the component interconnections, cables connectors and wiring methods;
- the physical mounting of the component equipment racks, consoles and road cases; and
- the power company.

Rarely will theoretically correct approaches and cookbook methods yield the desired results, because each group of products has a unique set of interfacing problems by design and by implementation. Products A and B may test perfectly when evaluated on a stand-alone basis; yet when connected to each other, they may hum, buzz, hiss or oscillate.

Connecting the microphone

Because the microphone is considered to be a floating source, its shield should be connected to the microphone case and input chassis and nowhere else, unless you want a significant amount of hum and buzz in the system (see Figure 1). However, there are exceptions. Take, for example, the grounded-shell XLR connector, which has an additional shell ground pin that can be connected to the shield. The shell ground provides complete RFI shielding at the connection point where two cables join together. This allows you to serially connect several microphone cables and maintain complete shielding.

The connector shell idea, however, fails when the XLR shells are allowed to come in contact with any grounded metallic fixtures in the building, such as a water pipe or electrical conduit.

Interfacing mic, cable and pre-amp

A microphone's sonic performance is a function of the cable's electrical characteristics and the type of load presented by the pre-amplifier. When you plug microphone A into cable B, which is connected to input C, each component interacts and affects the others.

All microphone cables have stray capacitance between the signal carrying conductors and between the conductors and shield. The longer the cable, the larger the capacitance. The microphone/cable system formed can have a major influence on the frequency-response performance of the microphone pre-amplifier.

Traditionally, microphone pre-amplifiers use an input transformer to couple the floating microphone to the amplification stages. Transformerless input stages are becoming popular. (See Figure 2.)

In less expensive equipment, the input transformer generally has a turns ratio of about 1:8 (15Ω to 10kΩ) so that the transformer has a voltage gain (18.2dB in this case), and is a better match to the input transistors or ICs for lowest noise performance. When used in the configuration mentioned, the secondary often is not terminated with a 10kΩ load.

Figure 3 shows the effects of three source impedances of a microphone on the frequency response of the input transformer. With some microphones, the decreasing input impedance of the transformer at low frequencies also causes a low-frequency rolloff of some microphones. Additionally, the lowered transformer input impedance can cause a considerable increase in distortion, especially with some condenser-type microphones. When the microphone's output impedance is matched to the input transformer specifications, a flatter response often is obtained.

With transformers, you get what you pay for. Cheap transformers create a host of interface problems, most of which are clearly audible. Electronically balanced transformerless circuits eliminate this phenomenon, but they too have performance limitations. Surprisingly, inexpensive electronically balanced transformerless input circuits can show the same increasing distortion at low frequencies, often seen with input transformers. Although active circuits do not exhibit the core saturation problems that transformers do, they are limited in their *common-mode rejection* capacity. The CMR is the maximum common-mode signal that can be applied to both inputs.

Clearly, the common-mode signal plus the input signal cannot exceed the input stage's power-supply capacity. For most IC circuits, the power-supply voltage is from ±15Vdc to ±24Vdc, the lower voltage being the most popular. The maximum common-mode signal handling of an input transformer is limited primarily by the insulation resistance of the transformer's

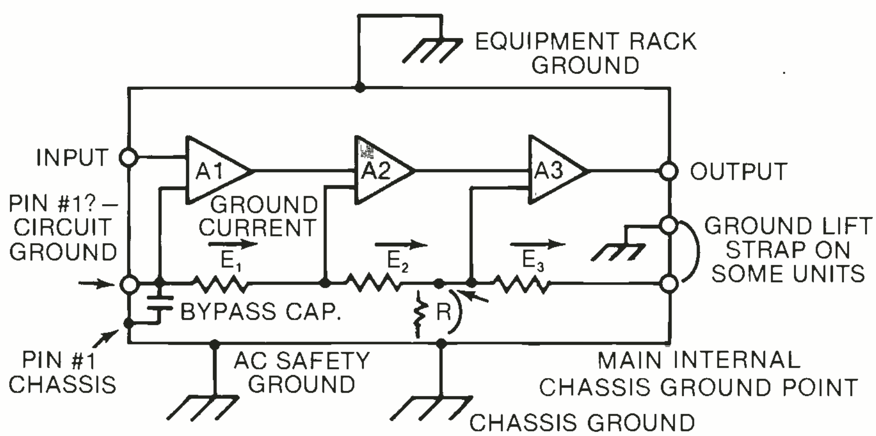

Figure 4. *A shield connected to the circuit ground, shield or ground currents (at one end only) will cause the voltage drops e_1, e_2 and e_3, which are amplified by A_2 and A_3. Sometimes a small resistance, R, connects the chassis and the circuit ground, reducing these voltages.*

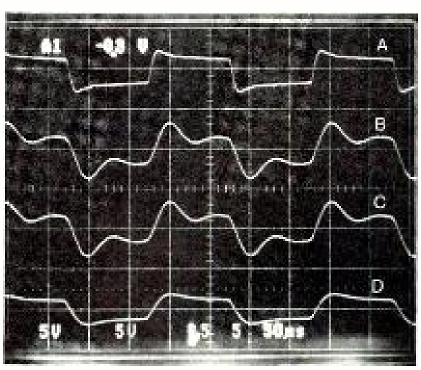

Figure 5. *Square waves showing the effects of cable loading on an electronic crossover.*

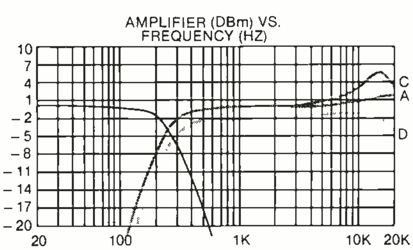

Figure 6. *Swept frequency-response curves showing the effects of cable loading on an electronic crossover. The frequency response in curve C shows that ringing outside the audio band does affect what happens inside the audio band.*

wire and, therefore, is (usually) much greater than ±15V.

Component interconnection

Input stage design and proper system ground techniques inside all signal-processing equipment (even consoles) play major roles in determining the final performance of the equipment when integrated into a total system. (See Figure 4.) The unit's susceptibility to external magnetic fields, line-current leakage to the chassis, input stage characteristics, output stage aberrations under load, system-grounding philosophy and susceptibility to RFI often are not specified by the equipment manufacturer.

Some products have a mixture of balanced and unbalanced I/O. In some consoles, for example, although the microphone, aux inputs and program outputs might be balanced and/or floating, most of the channel patch points are unbalanced.

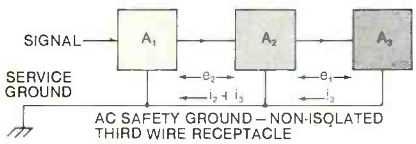

Figure 7. Serial ground implementation. Equipment-leakage currents flowing in the safety ground create small potential differences among the equipment. Amplifier A multiplies the ground current by its gain.

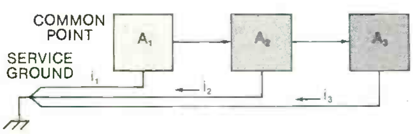

Figure 8. In a star or unipoint ground, with isolated safety ground, the ground currents do not add up because all the grounds are referenced to one point. To function properly, the point cannot be 50 feet long, but should be a proper ground bus in the power-service panel.

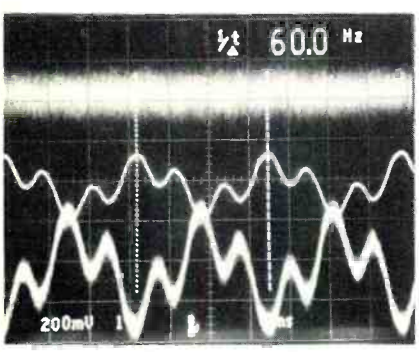

Figure 9. The top trace is the residual program noise output and measures −88.5dBu. With 1.2mA of ground current flowing (center trace), the output noise floor (bottom trace) rises to −74.5dBu, an increase of 14dB.

Usually, the input device's impedance is greater than 10kΩ. However, what is the input impedance of 10 2-channel units, all connected in parallel, when the manufacturer has shunted the inputs with capacitors of 500pF or more to reduce the RFI problems or to stop the unit from oscillating? These capacitors, totaling 10nF, represent a 795Ω capacitance load at 20kHz connected in parallel with the 1,000Ω combined input impedance. What could this capacitance do to the output performance of the device feeding it? In some cases, plenty.

Output connections

Currently, five generic types of output stages are used:

- the transformer-coupled output (usually floating);
- electronically balanced (circuit-ground center-tap referenced);
- electronic floating;
- single-ended (unbalanced); and
- 3-wire ground current-compensating circuit (which is essentially an unbalanced output).

Each of the various methodologies has its merits in systems operation. Unfortunately, about the only redeeming quality of the simple unbalanced output stage is the lower cost of the unit. The lower component cost often is offset substantially by the inordinate amount of on-site labor costs generated trying to de-hum and debuzz a system.

One of the least-published specifications is a device's capability to drive the interconnecting cable capacity. Figure 5 demonstrates how an additional 200 feet of standard cable can affect an electronic crossover's performance.

Trace A is the 15kHz square wave response measured at the crossover's output terminal. Trace B is the measured response, again at the output terminal, but with 200 feet of standard broadcast cable attached. In trace C, the response is measured at the end of the unterminated cable. Finally, trace D shows the resulting square wave response at the end of the cable when terminated in a 600Ω load.

Frequency response also can be affected by the addition of cable to a device's output. In Figure 6, the crossover's frequency response is plotted in the same test conditions listed for Figure 5. Curve A represents the device's response without the cable attached. Curve C represents the response with 200 feet of unterminated cable attached. Curve D shows the crossover's frequency response when connected to 200 feet of cable terminated with a 600Ω load.

This example shows that although a product may have stellar specifications and excellent slew rate when measured on the test bench, the addition of a few hundred feet of cable may seriously degrade its overall performance. This phenomenon is true especially if the output stage is an integrated circuit.

Because slew rate is defined as the time rate of change of voltage with respect to time ($dv/dt = I/C$), if you know both the maximum peak current and the presented capacitance, you can calculate the actual slew rate of the system. The data sheets for a 5532 specify the typical maximum peak output current at 38mA at 25°C. If this theoretical output stage design is required to drive 200 feet of cable with a lumped cable capacitance measured at $0.016\mu F$ between the leads for a balanced configuration, the actual calculated slew rate will degenerate from the specified $8V\mu/s$ to just $2V\mu/s$. So much for data sheet numbers when you're working in the real world.

The power company

Common to all of the components used in a system is the ac power line. Regardless of how simple or how complex the system is, attempt to keep all the building machinery motors, appliances, ballasts and lights off the audio-video ac service. When using 3-phase power, try to keep all of the signal-processing equipment connected to the same phase, thus minimizing the amount of 60Hz leakage current flowing among the various pieces of equipment. This clean audio service often is referred to as *technical power.*

Many times, the technical power service is carried to an extreme when a completely separate audio ground-grid earth electrode subsystem is used to ensure that the technical power is clean and has a good RF ground. In most cases, unfortunately, engineers are handed an ac service after the fact and are left to their own devices to sort things out. If you have any influence on the building wiring, insist that all the ac conduit is steel and not plastic, because steel is a good electromagnetic shield, whereas aluminum and plastic are not.

Make sure that at least two feet separate the power and the signal conduits. The last thing you want is a 500-foot conduit run of ac power and mic lines in plastic conduit located two inches from each other. When designing and/or connecting the power-distribution system, try to think of the system as a simple series circuit where any current flowing will cause a voltage drop between two pieces of equipment. Depending on how the system is grounded, the voltage drop can and does appear as an input signal that is amplified by the downstream system gain and is heard ultimately as hum. (See Figure 8.)

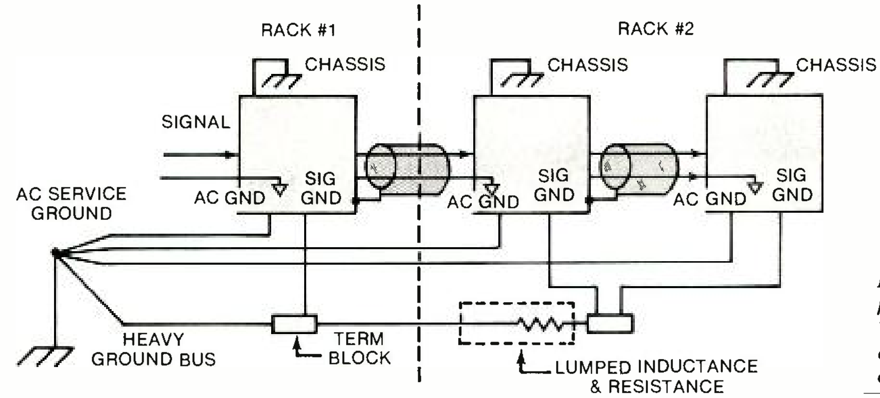

Figure 10. *Concept of a technical ground implemented with a low-impedance ground bus. The theory holds only if the technical ground's actual impedance is considerably below the rest of the signal return paths.*

A star or unipoint ground system, shown in Figure 9, prevents the ground currents from adding up, as they do in the serial ground system. To function properly, the common ground point should not be excessively long, but rather, be a proper ground bus located in the power-service panel.

Grounding, shielding and safety

Equipment interconnection inevitably brings up the subject of grounding and shielding. The term grounding, when applied to audio, often refers loosely to the interconnective wiring and shielding practices used by manufacturers, consultants and contractors. Technically *ground* is defined as the zero potential point in a system. The trouble is that, in practice, a ground is just as defined: a point with a zero potential that the unfortunate contractor must attempt to find.

A common misconception is that the only way for an audio system to be free from hum and buzz is to secure a good earth ground. Anyone with a portable radio/cassette player knows that this simply is not true. You will have much more success if you view the system's grounds as the signal returns. If you invert the system and call the signal hot outputs/inputs ground, you will quickly observe that the new system hot wire has far more going on than you previously thought.

In the United States and Europe, a third-wire safety ground wire is commonly employed to connect the chassis to the building or service ground. Any leakage currents from any components in the chassis are, presumably, shunted harmlessly to ground, rather than through you.

It is significant to note that the commonly used 0.01μFd line bypass capacitor can contribute as much as 452μA of leakage current to the chassis in addition to whatever leakage current is caused by the power transformer's winding capacitance leakage to the core. In practice, only a few microamps of leakage current are necessary to cause an audible hum and buzz, as shown in Figure 10.

From a safety standpoint, never lift the third-wire safety ground and depend on

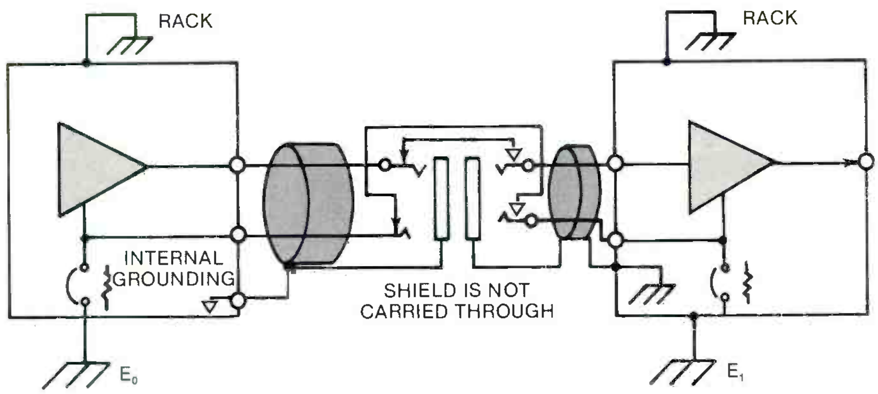

Figure 11. *Unbalanced I/O through a patchbay. How the grounding is handled inside the units determines the overall result. The larger the difference between E_0 and E_1, the worse the problem. To ensure success, both units should be plugged into the same duplex outlet, operating from the same phase. Often the unbalanced I/O is a normalized send/receive loop on a console remote to a patchbay.*

the signal wiring for fault protection. You may be in for an expensive surprise. A considerable amount of equipment in the marketplace has pin No. 1 of the XLR connector (shield) and the sleeve of the phone plugs connected to the circuit ground rather than to the chassis. The circuit ground is then often tied to the chassis at the power supply. If a fault occurs in another component that also is improperly safety grounded, then the full 120Vac potential may return to chassis via the internal printed circuit boards, destroying everything in its path. (See the related story "Pin No. 1, where are you?" on this page.)

The third-wire safety ground plays havoc with any type of rational system interconnection involving any type of unbalanced input/output, because the safety ground is a secondary and/or tertiary signal return. If you want to avoid grief, avoid unbalanced I/O. Regardless of the many claims, there are almost always problems with large multiple-patched unbalanced I/O systems.

This leads us to the concept of the unipotential or single-point *technical ground* in which all the signals are, in theory, referenced to a single low-impedance ground, as shown in Figure 11. In theory, all shields and equipment chassis grounds should be connected to the technical ground, which is supposed to be clean.

If the system is large and complex, then the technical ground should be at a much lower impedance than all the other grounds. However, with typical leakage currents and typical system gains, the technical ground rapidly can approach the dimension of a copper 2′ x4′, not a piece of No. 4 wire, if the system is to function properly. The inclusion of the technical ground can almost guarantee that you will

Pin No. 1, where are you?

Compounding and confounding the grounding issue are the various ways in which equipment manufacturers internally ground their equipment. For a shield to be functional, it should be connected directly to the chassis at the output or input port, depending on the system wiring configuration. Pin No. 1 and the phone jack sleeve in a balanced system, logically, are then tied to the chassis.

But, what happens when the equipment uses a 2-wire unbalanced connector in which, unfortunately, the shield is often the signal return? Should the shield (sleeve on a phone plug) be allowed to float from the chassis or be grounded to the chassis?

For unbalanced systems it is best to use a 2-wire scheme in which the ground connection is made via one of the internal wires, and the shield is bonded to the chassis at one end only. Floating the shield from a direct connection to the chassis can eliminate ground loops, but the shield becomes an effective antenna for transmitting RF energy inside the chassis (where according to Murphy's Law, the RF always is detected at the wrong time).

Floating phone plugs on consoles almost guarantees that the patches will hum and buzz, because the shield currents modulate the console's audio grounds. For example, a 0.063-inch wide trace of 1-foot long, 2-ounce copper has a resistance of 0.045. A shield current of only 0.1mA can create a signal of 104.7dBv. A 32-channel console's summing amplifier will increase this signal by 20log (32+1), plus the gain after the summing amplifier (typically, +10dB). What was quiet at −95dB is now degraded to −64dB at the output of the console.

Some equipment manufacturers float the entire circuitry on the enclosure and depend on a chassis ground strap to tie the shields to the chassis. In most cases, however, the PC board resistance between the shield's grounds is quite high; rarely do you find the necessary RF bypass capacitor connected directly at the shield ground to the chassis.

One popular technique commonly used by power amplifier manufacturers, because of the large currents present inside the chassis, is that of raising the ground of the RCA or phone jack connector a few ohms above the chassis. By referencing the unit's small signal internal grounds to this point, ground loop currents are dropped across the resistor and appear as a common-mode input signal. Any fault currents, however, will smoke the resistor at the wrong time.

have ground loops, because now the system has the following ground returns: technical ground, safety ground, physical rack-frame ground and signal grounds in the equipment that may or may not be chassis-referenced.

Additionally, if the shields are to be effective near power lines, they need to be terminated at the component chassis, not several feet away at the technical-ground terminal block. For the technical-ground concept to work, all the safety grounds must be returned to the main service via isolated U-ground receptacles, where they join the technical ground. Independent safety-ground isolation should keep some leakage currents from flowing in the technical ground.

For every type of interconnection rule, there is an exception, especially if the system has a large variety of balanced, floating and unbalanced equipment I/O all showing up on patchbays. In the case of an unbalanced output connected to an unbalanced input, a 2-wire plus shield cable should be used, and the shield should be connected to the zero-signal reference potential at the signal output.

If there is a difference in potential between the two pieces of equipment, howver, connecting the heavy-gauge shield at both ends actually may reduce the noise; its resistance is now in parallel with the signal return wire, thus lowering the total resistance. As shown in Figure 12, the larger the difference between E_0 and E_1, the worse the problem will be. To help ensure success, make sure that both units are plugged into the same duplex outlet and that both receptacles are fed from the same phase.

Many times, the unbalanced I/O is a normalized send/receive loop from console to patchbay. Incomplete theory can lead you astray. For instance, an in-between patch point further complicates the issue and requires a slightly different approach.

In general, the shields should connect to the signal ground at the chassis tie point of the signal-source end. The shield should be connected at one point only. High RF areas may require that the receive end of the shielded signal cable be bypassed to the chassis with a $0.01\mu F$ ceramic disk capacitor.

Equipment mounting

The equipment racks sometimes cause interfacing problems. Different amounts of ac line-to-chassis leakage currents in the rack-mounted equipment often can cause a potential of several millivolts from the top to the bottom of the rack. Sometimes, if there are enough line cords all neatly dressed along the length of the rack, the rack can become a good enough transformer core to magnetically couple the ac line into the clean technical ground.

If the equipment has unbalanced I/O, the only way to avoid the rack-induced ground loops may be to tally isolate the equipment from the rack with insulating washers and screws. If you must do this, don't cut the safety ground.

Whatever you do, try not to wire and terminate the entire system at one time. Rather, build the system by terminating equipment progressively and checking the results as you go. As you connect the system, all units must be physically installed and turned on. If some of the equipment is off, it is difficult to get an accurate representation of stray equipment fields and power-line perturbations.

Also, test the equipment in as many operational configurations as possible. One test setup may not identify a potential problem. The time to find out that device A will not work properly with device Q is not when you need the combination for important production work. RE/P

Interfacing Monitor Amplifiers

By Richard C. Cabot, P.E., Ph.D.

A power amplifier's specifications tend to lose importance once the amp is connected to other components.

Amplifier manufacturers would have you believe their units are the ideal black boxes with lots of gain. (This probably explains the basic black cosmetics many of them use.) However, as you know, that isn't true. The equipment that is connected to the amplifiers, and how it is connected, determines the performance of power amplifiers in the real world. This article will explore some of the areas in which system interconnection can affect performance significantly.

If you made a simple model of the black box amplifier, it might look similar to the one shown in Figure 1. The model in the example will produce both linear and nonlinear effects, which vary with the devices connected to it. However, it is highly simplified and will include only the effects that will be described here.

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The input signal is applied to the terminals on the left. The terminals are shown as if the amplifier is electronically balanced, which is common for professional units. The output signal appears on the right, unbalanced. This is the case except for bridged-mono amplifiers. The power line enters at the top of the box on three leads, including the chassis/safety ground. Let's examine the effects of connecting this model to a real system.

Input loading

The input of the amplifier may be viewed as a set of impedances between the input terminals and from each input terminal to ground. With many amplifier designs, the source driving the amplifier sees a different load on each input terminal. This unbalances the lines and creates *common-mode voltage,* the voltage present in phase on each conductor of a balanced line with respect to ground. When many amplifiers are paralleled, the difference could become significant. Because the amplifier responds to the difference between the two voltages, this is not supposed to be a problem.

However, some balanced source circuits misbehave when presented with an unbalanced load. A subtle difference between the behavior of these drivers occurs at large amplitudes. Suppose that the source is delivering 14V peak to the amplifier. Each side of the balanced output would have to produce 7V peak if the load is balanced. However, if the load is unbalanced, one output has to deliver more while the other output delivers less. With a highly unbalanced load, the source clips even though it easily could drive the balanced case.

The amplifiers' input capacitance combined with the interconnecting cable causes a roll-off in the high-frequency response by working against the source impedance of the driving device. It also creates an imbalance of the signal on the line at high frequencies if the capacitances are not balanced.

Although most manufacturers specify the performance of amplifiers at rejecting the common-mode voltage (the common-mode rejection ration, or CMRR), many do not specify the maximum voltage allowed before overload will occur and the amplifier will not be able to reject the signal. This specification is called the *common-mode voltage range.* If this value is exceeded, the input ceases to be linear, even though the signal may be well within the allowable level before clipping. If the amplifier and signal source are located far

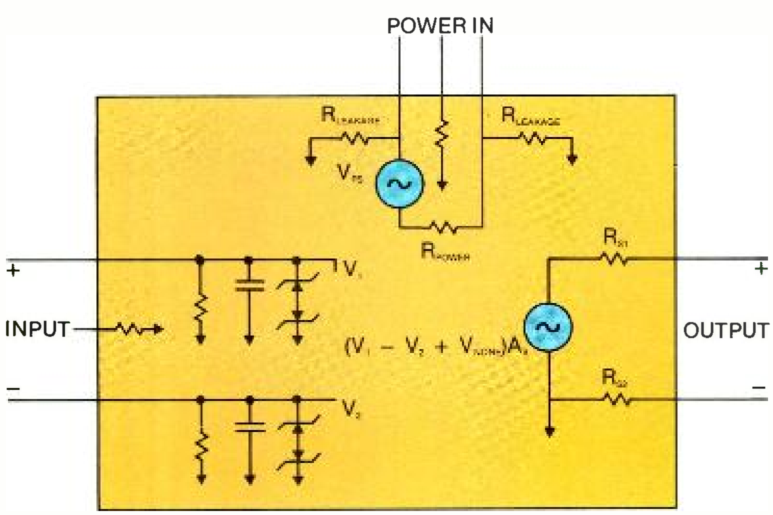

Figure 1. *Inner workings of a basic amplifier that can produce linear and non-linear effects.*

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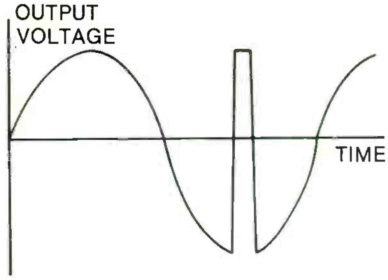
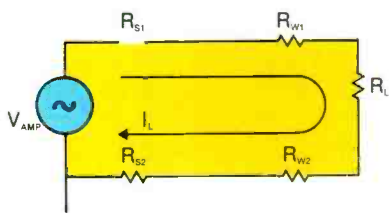


Figure 2. When an amp that uses a Bi-FET op-amp at the input receives too much common-mode voltage, the output reverses itself and swings in the opposite direction.



$V_L = I_L R_L$

$V_{AMP} = I_L(R_{S1} + R_{W1} + R_L + R_{W2} + R_{S2})$

Figure 3. An example of how the resistance in the output of the amplifier causes a voltage decrease. This voltage drop can be calculated by applying Ohm's law (V=IR) to the connection loop.

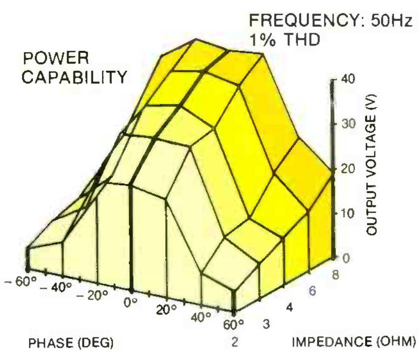


Figure 4. As the load becomes reactive instead of resistive, the curve tends to drop quickly, generating less output current.

apart or on different power sources, there may be significant common-mode potential between their chassis.

Some amplifiers use an op-amp input buffer circuit to perform a differential to single-ended conversion. The output of this op-amp drives the power-amplifier circuits. Other amplifiers employ a fully differential topology that provides a differential input on the power-amplifier circuit itself. Both approaches can result in a limited common-mode voltage range on the input before that amplifier distorts.

However, the op-amp designs more often exhibit undesirable behavior when this happens. Figure 2 shows the effect of too much common-mode voltage on an amplifier that uses a Bi-FET op-amp at the input. When the signal gets within about 3V of the negative supply, the output reverses itself and swings in the other direction.

Output loading

The output of an ideal amplifier is a voltage spruce. No matter how much current you draw from it, the voltage should not change. However, because of resistance in the output of the amplifier, the voltage goes down. Figure 3 illustrates this effect. The amplifier output voltage is developed across the internal resistance, the cable resistance and the load. It is a simple matter to calculate the voltage drop in each of these elements by applying Ohm's law (V=IR) to the connection loop. If the load current is known, the voltages are found by multiplying the current and the resistance.

Amplifier output resistance is often specified as a *damping factor* rather than a resistance. Damping factor is the ratio of the load resistance to the amplifier output resistance. This computation assumes a nominal load value and tends to disguise the real information of interest, the output resistance. If you know the damping factor, you can easily recompute the output resistance as:

$$\text{Amplifier output resistance} = \frac{\text{load resistance}}{\text{damping factor}}$$

In typical installations, the amplifier resistance is a negligible part of the power loss before the load. The majority of the power is lost in the wiring.

If long cable runs are required, the cost of wire large enough to keep the power loss down becomes prohibitive. Transformers may then be used to raise the voltage and to lower the current levels in the wire. The lower current levels reduce the voltage drop across the wire.

Most real-life loads that a power amplifier handles (such as a speaker) are reactive as well as resistive. This means that the load looks like a combination of a resistor, an inductor and a capacitor. The impedance changes with frequency, making the current vary with frequency. This also results in a phase shift between the output voltage and the current through the cable.

Two major problems can occur with significant variations in the impedance as a function of frequency. The varying current causes a variation in the voltage drop in the cable. This variation introduces frequency-response changes that follow the same shape as the impedance curve. At frequencies where the impedance rises, as happens at resonance, the response rises. When the impedance drops, the response decreases.

For example, if the wire resistance is 10% of the minimum impedance of the speaker, this results in a voltage loss of approximately 10% in the wire. This loss does not occur at resonance or at very high frequencies where the impedance rises dramatically. There will then be a 1dB variation in response.

The energy storage behavior of inductors and capacitors causes the other effect of a real-life load on an amplifier. When current is developed through an inductor, energy is stored in the magnetic field. When the direction of current flow is changed, which happens every cycle with an ac signal, the stored energy must be removed and the field restored in the opposite direction. For sinusoidal signals, this results in the familiar 90° phase shift between voltage and current through an inductor. Transient signals, such as square waves or pulses, require large current pulses at the same time the polarity reverses because the energy must be withdrawn and replaced fast.

Matti Otala (of TIM fame) has shown that loudspeakers can require current peaks as much as twice those expected from the minimum value of the impedance. In other words, if the speaker impedance dips to 4Ω at some frequency, that current draw on a 40V sine wave at this frequency would be 10A. However, it is possible to devise a pulse waveform that will make the speaker draw 20A at the transitions. This 2-to-1 difference easily can clip an amplifier that was specified into the system based on the 10A value. These worst-case transients aren't likely to happen often, but others that draw substantially more than 10A can be expected.

Worse yet, speaker manufacturers usually quote a nominal impedance value that is sort of the average impedance value over the frequency range of use. Designing a monitoring system with this value can produce even worse overloads than the example cited here.

Most power amplifiers contain limiting circuits that control the maximum output current. The current limit usually is a function of the phase shift between the current and the output voltage, which limits the output into a reactive load, such as a speaker, to a lower value than can be supplied to a resistor. Because most amplifiers are used to drive speakers, not resistors, this can impact seriously on the system's design.

Figure 4 shows the available output power from a commercial amplifier as a function of the phase angle of the load. The height of the curve down the center indicates the power available into a resistive load. It is quite respectable. However, as the load becomes reactive, the curve drops off quickly. The amplifier can deliver only a fraction of its resistive output current when the load becomes sufficiently reactive.

Power-supply effects

The power feed to an amplifier rack can produce some unusual problems when the system is connected together. The most common problems occur because of leakage currents in the amplifier from power line to ground. These are indicated in the black box models as a pair of capacitors from the power input to chassis. Although current will be returned to ground through the power cord ground lead, the resistance of the ground causes a voltage difference between the chassis and others in the system. If signal ground leads are connected between the amplifier and other devices, some of the current splits and returns through signal grounds. This current can then induce hum in the signal source as it travels through the chassis heading for true ground.

Several amplifiers on the market today have special power supplies called *back-slope regulators, shark-fin regulators* or whatever the marketing department has dreamed up. The basic idea is that a power transformer generates flux in its core based on the voltage applied to the primary. If it is applied only when it is needed to charge the filter capacitor on the secondary, the flux may be reduced and the core can be made smaller. This is accomplished by turning the voltage to the transformer on and off with an SCR at the appropriate time.

The waveform of the applied voltage looks like a shark's fin, hence, the name. This design has the side effect of increasing the peak current drawn from the power line. The increased peak current and the menacing-looking voltage waveform can turn a once tame and friendly sine wave power line into a noise generator extraordinaire.

If a rack full of these amplifiers is used in a studio, the results can be disastrous. The current peaks will introduce huge voltage drops in the power feed as if something had chewed on the waveform (something did). No matter how carefully filtered the other devices on the power line, they will get jealous and buzz loudly to let everyone know it. A separate power feed for the power amplifiers will be required just to reduce the interference to something manageable.

With shark-fin power-supply amplifiers and conventional designs, the current in the line cord may couple into signal leads through mutual induction. Two wires placed near each other create a small transformer that can couple ac current from one to the other. Although the coupling generally is small, the power-line currents are so large that the result still can be a fairly large hum current.

Attention to proper lead dress can eliminate this problem. Keep power wires away from signal leads, and when they must cross, do so at right angles to each other. Similar problems can occur if the speaker feedlines are routed too close to signal leads. This wiring error can produce crosstalk of one signal into another and, in extreme cases, can cause oscillations by coupling the output of an amplifier back into its input.

Features that can help

Many power amplifiers are now equipped with compressor circuits as standard items. These usually sense clipping in the output stage and reduce the input signal level appropriately. If designed correctly, they can prevent serious clipping of the amplifier. However, most circuits sense only voltage clipping and ignore current limiting, which is merely another form of clipping. Therefore, highly reactive or low-impedance loads still can cause problems. Under no circumstances should the compressor circuit be viewed as overload protection for the speaker driver. Unless the power output of the amplifier is matched exactly to the driver, the compressor circuit will not offer much protection against overdriving the speaker.

In most circumstances, an active differential input on the amplifier is adequate to reject ground noise and induced signals. When this fails, transformers often provide a significant improvement at low frequencies. If the interference is at high frequencies, capacitive imbalance in the transformer windings limits CMRR to the same or worse performance than a high-quality active input.

If line-related interference is suspected, filters can be placed on the power-line inputs of the offending equipment. RE/P

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Phase Shift...

Should We Worry?

By Terry Pennington

Should we be wearing small cardboard badges sensitized to phase shift to evaluate our daily dosage?

Phase shift. What is it? Should we be concerned about it? Should we be wearing small cardboard badges sensitized to phase shift in order to evaluate our daily dosage?

Phase shift had better not be a bad thing, because we are exposed to large doses of it every day. Every sound you hear is so heavily laden with the stuff that all sound would be hard to hear if it were a problem. Sound cannot be transmitted through the air without vast amounts of phase shift. This applies not only to recorded or electronically reinforced sound, but also to all audio signals, naturally produced or otherwise.

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It takes time for sound to travel through the air, and this time delay imparts phase shift to audio information. Applying the same time delay to all frequencies produces a "linear phase" condition, but nothing this important can be that simple. The following is a scenario that all audio folks should be aware of.

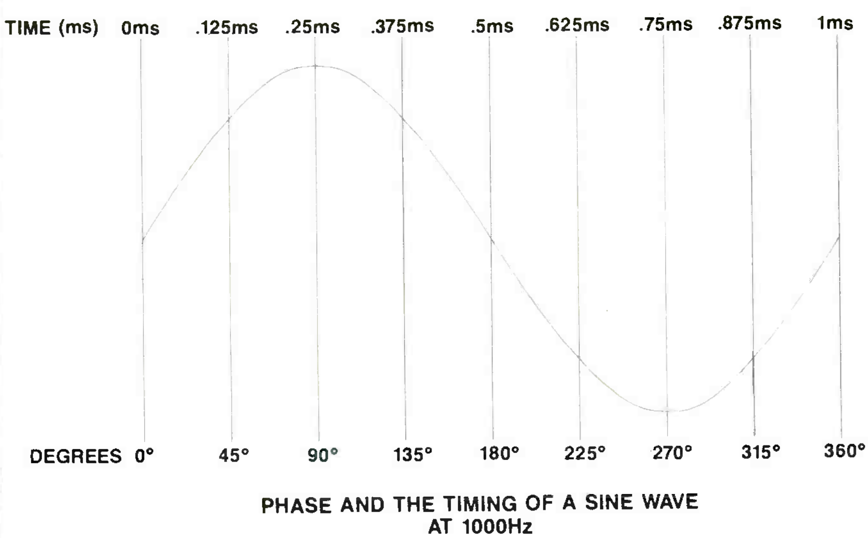

Figure 1. *Phase and the timing of a sine wave AT 1,000Hz.*

The mechanics

By now, you are probably wondering why all of this is true. It is relatively straightforward. However, it does take a bit of basic technical savvy to understand it. Acoustic-related phase shift phenomena relates to the time it takes for sound to travel through the atmosphere. If sound traveled at the speed of light, there would be no problem. Unfortunately, that is not the case. The speed at which sound travels is extremely slow, relative to the wavelengths of sound, and can cause a lot of difficulty in certain situations.

Phase shift is introduced into an audio signal as a function of the *period* of the audio signal and the *time* it takes the audio signal to pass through the air. Because time is a direct function of the distance that the signal must travel, distance becomes the primary variable in the equation. The period of the waveform is also time—the frequency of the signal divided into one second. For example, 1,000 cycles per second. (1,000Hz divided into one second produces an answer of 1/1,000 of a second, or 1ms.) Thus, the phasing of the signal is a function of how much delay is imparted to the signal as it passes through the air in its acoustic form. The actual phase shift relates to how the *amount* of delay lines up with the signal.

If there is one full cycle of delay as the 1kHz tone passes through the air, there is then 360° of phase shift applied to it. If the signal is delayed by exactly one-half cycle, there is 180° of phase shift. The illustration in Figure 1 represents how degrees of phase line up with time during one cycle of a 1,000Hz.

To understand time delay and phase as it relates to audio signals, consider the following. Seat yourself 10 feet from your favorite sound source, acoustic or otherwise. Now assume that this sound source is producing a nice, clean 20Hz . At a speed of sound that roughly equals one foot per millisecond (1/1000 second), it would take 10ms for the 20Hz to travel from the source to your ear. In terms of phase shift, the transmission medium (air) is introduc-

Linear phase shift with changing frequency is only possible when no reflections are present.

ing a 72° phase shift because the 10ms delay across the air is equal to 1/5 of the time it takes for 20Hz to achieve one cycle of its total period. (See Figure 1.)

Also note that the 10-foot distance is 1/5 of a 20Hz wavelength. At any rate, 1/5 of the total 360° of a 20Hz cycle is 72° (360 ÷ 5 = 72). Fine. So much for 20Hz.

At a higher frequency, such as 20,000Hz, the previously discussed 10ms delay is equal to 200 full cycles. The phase shift across this 10-foot space is now equal to 72,000° because of the fact that it takes 200 cycles at 20kHz to span 10 feet. (200 x 360 = 72,000.)

That is all there is to calculating phase shift through acoustic space. This example makes it seem silly to worry about a leading or a lagging phase shift response in a piece of electronics under evaluation. It should be obvious that the phase shift at frequencies in between these extremes will be a minimum of 72° and a maximum of 72,000°. The phase, as it appears at the ear of the listener, will always be off in a relative manner, one frequency to another. This example is just as true in a live room as it is in an anechoic chamber.

The difference between live acoustics and anechoic environments is that these phase deviations may have only one path or several paths to travel. It is when the phase deviations have several paths that frequency response abberations due to phase deviations are created.

Can you hear it?

Now, you say that you have heard phase shift? If you have heard examples where you thought you could hear it, what you really heard was frequency response changed by phase shift. A lot of opinion regarding the audibility of phase shift surfaced in the 1960s, when it came into use as a recording effect. This effect yielded a noticeable change in the sound of such groups as the Boxtops. It was not the phase itself that was audible but the combination of swept phase with "dry" signal that created the audible sound. If phase is changed and is not allowed to mix with "straight" signal, no perceptible change in sound will occur.

The human ear is a bit like a spectrum analyzer in the way that it ascertains the frequency content of an acoustic input. It is, under normal circumstances, only the frequency components of the signal that are discriminated from the cacophonous elements of sound. Timing, as in phasing, is of little or no importance.

As an example, if the harmonics of a tone are scattered about in time with respect to the fundamental, the sound will be the same. Experiments have been constructed to show that this is not always the case. However, these were usually done with some sort of circuit that had other attributes as well as its ability to modify the phase of the signal. A great deal of care must be taken to ensure that other more audible elements (such as frequency response) are not changed when attempting such experiments.

The delay associated with the transmission of sound through the air produces a "linear phase" characteristic. That is, the change in phase shift vs. a change in frequency is linear. This characteristic is usually accepted as ideal in a situation where phase shift is unavoidable. Most individuals, if given a choice, would opt for no phase shift at all. This, as demonstrated previously, would be impossible to achieve in acoustic transmission due to the delay caused by the speed of sound through air. [See "Effects of the Speed of Sound" in the April issue of *RE/P*.] Unfortunately, there is no way to really achieve the linear phase characteristic in the real world of acoustics.

Linear phase shift with changing fre-

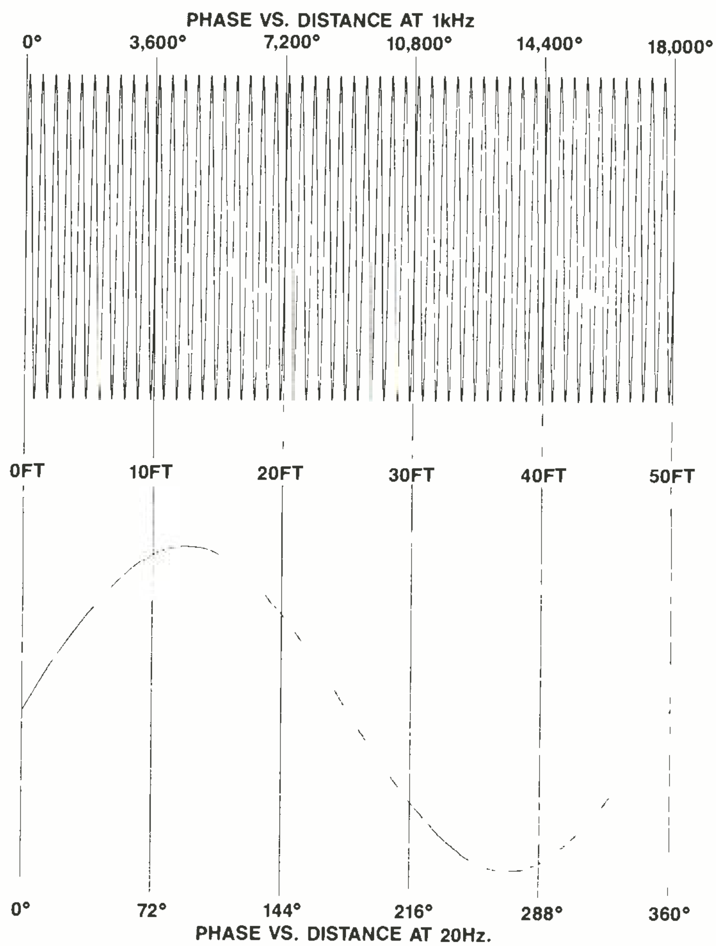

Figure 2. *Demonstration of two identical waveforms with opposite timing added together. The result is zero.*

quency is only possible if there are no reflections present, which is the case in an anechoic chamber. (Anechoic means no echoes, therefore, no reflections.) When any reflections are present, the direct signal is combined with the reflections. These reflections must always travel a different distance to the listener, and, therefore, have different delay times applied.

Different delay times result in different phase angles. What we end up with is a conglomeration composed of direct sound and all the reflected sounds (of which there is almost an infinite number in any space). The phasing of the resultant sound is impossible to predict due to the innumerable variables that exist. It is also impossible to maintain and equally difficult to speculate on the nature of the broadband frequency response of this altered signal.

If you were sitting at the mixing desk in the average studio control room and moved your head only a matter of inches to reach a control on a distant part of the console, the phasing and frequency response reaching the ears would vary dramatically. A movement of such a small difference can completely change the relationship of the direct sound from the monitors and the reflected sound coming from the glass, walls, ceilings, floors and so on. Most control rooms and studios have acoustic treatment in place to control this effect; however, it is rarely eliminated completely by such measures.

Filtering with phase

Electronic filters use phase shift to produce the frequency sensitivity required by the circuit. This task is achieved in much the same way as the phaser discussed earlier. The difference is that there is no sweeping of phase normally occurring so that the frequency response stays fixed. If a variable filter such as an equalizer or tone control is desired, manually adjustable elements may be placed in the filter.

You already know that when you connect two speakers out of phase with each other and play the same material through both, something gets lost. Figure 2 demonstrates that when two identical waveforms of opposite timing are added together, the result is zero. If one would add more or less of the opposing waveform to the original or to change the timing of the opposing signal, more or less of the original would appear at the output of the example.

Recent advances in the area of digital signal processing have made it possible to construct filters with no apparent phase shift at their outputs. However, at this time they have their limitations and a rather expensive price tag. There may come a time when most all electronic filters are of the digital variety, but is this necessary? Why should we remove phase shift from electronics?

A great deal of our collective time and energy goes into the quest for minimum phase shift. The question is, should we bother? The common-sense answer is no. There is so much phase shift in nature and little we can do to reduce it. If one were to employ digital signal processing techniques to rid our signal paths of phase shift, what would we accomplish? We would then have such a small improvement at the expense of the majority of the country's GNP that it would seem senseless. If we could clean up all of the phase shift present in active and passive signal processing and recording devices, what would we do then? The only way to remove phase shift from audio is to change the speed of sound. I doubt anyone is working on that. RE/P

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Circle (45) on Rapid Facts Card

Personnel Management

By Foote Kirkpatrick

How managers can assist engineers through the beginnings of a recording career.

I always ask the same question of any aspiring engineer: "Why do you want to be a recording engineer?" The answer is always the same. "I love music."

Because I've never met anyone who hated music, there have to be other reasons why people want to spend the rest of their working lives holed up in a windowless room trying to interpret someone else's creative values. They may already know what some of those other reasons are or may discover them after starting to work. It's part of a manager's job to help with that discovery. With a large and diversified studio, there are many different opportunities available.

As "second engineers," they will have the opportunity to taste it all, from video sweetening, Synclavier, radio and TV post, record projects and jingle recording. After a year or more, second engineers will have a good idea of what their strengths are, and whether or not they are in the right career or studio.

Interviewing a prospect

During the interview, I look for someone who is college-educated and, preferably, has some computer knowledge. I also like to know about the college programs: do they have hands-on 24-track studios and what electronic courses are required? I am interested in an engineer's grade average and also impressed with people who have worked their way through school.

If applicants have worked in another studio, I'm interested in what they learned, how far they moved in the pecking order and why they left. But I never accept tapes of the "here's what I can do" genre. Given plenty of time and the proper supervision, I'm convinced that just about anyone can create an acceptable tape.

Another important aspect is character. They must possess a high level of self-confidence. Without high self-esteem, it's hard to make it in this industry, where so much of what's expected of you is subjective. They have to possess a certain toughness to make it through a long apprenticeship (assistant engineers may not sit at the board for two years, maybe more). They have to be unflappable and crisis-oriented. Never walk if you can run to get something done!

Someone may seem to fit all the specifications but, the proof is always in the pudding. If there are some serious character flaws, they are bound to surface during the long hours and pressure of those first months. At the beginning, it is the desire and thirst for knowledge that keeps them hanging on.

Apprentices are exposed to so much and work with many different staff engineers in the beginning. They may work on jingle sessions all day and switch to seconding a record session at night. Each session requires a special discipline, and what is acceptable conduct for a record client may not be for a jingle producer.

Jingles are good training ground for engineers. In one session lasting several hours, engineers lay the basic tracks, do all the overdubs and record the vocals and mixes. All the tasks they have to perform for a month-long record project are telescoped into that one jingle session. After a few weeks of that kind of pressure, a record session can feel like a vacation.

At the beginning, apprentice engineers may not be very productive. It may take quite a while before new staff members get the feeling anyone really cares about their success. This is where a manager's role begins. Good management is simply good leadership. The success of a talented employee means success for the company. Managers should lead the engineers toward that success.

The stages

The development of recording engineers involves two stages. The first year or two is the learning period; the desire to learn is what motivates. As they learn, they are given more freedom to develop their own ideas and techniques. At Universal, apprentices are allowed to use any room on downtime to experiment and learn.

The first stage ends when second engi-

Foote Kirkpatrick is studio manager at Universal Recording, Chicago.

neers become convinced they're ready to engineer sessions alone. They believe that nobody appreciates how good they really are and that they are not being paid what they are worth.

The manager's role increases at this point. As manager, I can't make anyone's career, but if I see that an engineer has the right ingredients to make it, I'll do everything I can to help.

My management style is direct and I shoot from the hip. During meetings, I'll point out strengths and weaknesses I see in the engineer. Here is the opportunity for a manager to help smooth over the rough edges and help the engineer move on to the next stage.

Guiding the engineer

At this point, a manager should help second engineers get a handle on what they do best and lead them there. If I think someone is suited for music mixing, I will select a project that will give him an opportunity to show his talents. I will watch closely and give advice if asked. I'm especially interested in his sense of confidence and people skills.

Sometimes if they have shown great promise, I may toss them into something a little over their heads. Often, that challenge brings the best results. Fear of failure will often motivate success. By the end of this second stage, a studio has invested a lot of time and money in these engineers, and should want to help them to develop completely.

As skills develop, reward these engineers with salary increases as well as recognition for performance. But I don't believe that money motivates anyone to do a better job. It's up to the leadership in a company to recognize superior performance and reward success. If money was the big motivator, performance would automatically improve with every raise. This has proven not to be the case.

At Universal, we have a number of talented engineers on staff with great personal followings—clients that would follow them anywhere. Why do they remain? Because they are given a lot of freedom to develop their own styles—both technical and personal, because they're part of a company that believes in supplying them with the latest tech "goodies," and because they believe they can progress as far as their talents will carry them.

Universal created a subsidiary company called "Studio Consultants" in response to requests from studio owners to share the facility's management and buying strategies. Through this forum, I am often asked the question, "What do I do with this engineer—he is in business for himself and could care less about the studio."

I usually say that if an employee has reached that stage, you've already lost him, and he no longer belongs there. But the secret is not to let him reach that stage. By the time an engineer is at the top, the degree of confidence, trust and respect he feels for management can be measured in the desire he has to be a part of the company. It goes back to morale and management's belief in itself and the staff. In the recording industry, you are dealing with artistic people with high ego needs. The manager should reinforce the staff's confidence, expect high performance and share in their successes.

Managers don't have to run a popularity contest and hang facile, "rah-rah" teamwork slogans on the wall, but they should be enthusiastic about the company. A manager's ultimate responsibility is to keep the studio at the top of the heap and should expect everyone who works at the studio to try to do the same. When you focus on expecting something from someone, you can usually get it.

RE/P

Engineer Interview: Andy Heermans

By Brian Lee

One of this engineer's most recent sessions points to a new direction in MIDI: all the musicians playing live in the studio.

Andrew Heermans has been the favorite engineer of Carlos Alomar, David Bowie's guitarist, since 1985. Alomar says, "Andy keeps me in line, always adding a bit of freshness to an otherwise dull production."

Before Andy met Carlos, he divided his time between playing bass with John Cale and engineering at Planet Recording in New York. Planet co-owner Jon Grossbard describes Heermans as "one of the most meticulous people I know."

RE/P: *How did you start engineering?*
AH: I had been working around New York as a bass player and one of the places I often recorded at was called Planet Sound. One day, the owner Jon Grossbard, called me up and asked me if I was an engineer. I remember he would always catch me fooling around with the console while the engineer wasn't looking. I had exactly $2 that day, so I stretched the truth a little and said yes. Jon invited me down to the studio and put me on staff right away. From that day on I was an engineer. I was lucky. I always managed to stay 2ms ahead of my needs. So technically I learned as I was going along.

RE/P: *When did you switch from bass playing?*
AH: I engineered on and off for three years in between tours with Cale. Jon Grossbard would always welcome me back when I was off the road. But after three years, almost exclusively at Planet, I pretty much stopped playing the bass professionally. I decided it was more fun to be in control than to be just a sideman. But I still have a very fond place in my heart for that spectrum of the audio frequency band. I love that bottom end.

RE/P: *When did you start free-lancing?*
AH: Carlos Alomar and I met when I was still at Planet. Carlos and I worked on a couple of things, some demo tapes and some records for some other people. Things went really well, and he and I have been working together ever since. Carlos was really the one who was responsible for breaking me out in a major way. Just being able to work with him gave me the clout (because of the experience I gained through working with him on various albums) to be able to make a living.

RE/P: *How do you deal with monitoring in different studios?*
AH: Well, I used to be really uptight about it. I used to go into a studio and get five pairs of NS-10Ms out of the back room and listen to every pair to try and find one that sounded matched. Studios always mix them up by the time you get there, and you never have the original matched set. You get one speaker with a tweeter that was replaced last week and the woofer on the left side was replaced a month ago. Now I've learned to just fly by the seat of my pants. It's sort of like back to the way I learned to engineer. Now I go into a room and suss it out using my ears (see Figure 1). Sometimes I take a record I've mixed before to listen to, but that's about it.

RE/P: *Do you carry your own speakers with you?*
AH: I have a set of monitors, but I don't carry those around because I don't want to blow them up—they're too expensive! The set I use at home is my standard. So, every night when I come home I take a cassette—no matter what I'm doing—I listen just to see how it was in that room. Then if I ever go back or if I am going back the next day I know what the room really sounds like.

RE/P: *What is your reference home setup?*
AH: I have Miller-Kreisel satellites and a MK sub-woofer. The MKs are really neat because they have two different EQs. They have what they call a British EQ, which is the BBC standard, and is very flat. And then there's a German EQ that is a much more excited mid-range. It makes them sound light in the bass and heavy in the high end. They are especially handy if you work on a project for six weeks. When your ears start to go, you can switch over to the high end version.

RE/P: *What about close fields with subwoofers in the control room?*
AH: I think it's a great concept, and I wish people would do it. I have a subwoofer in my system. I'm totally into the concept, especially now that control rooms are needing to have multi-point monitoring.

RE/P: *Have you tried it in the studio?*
AH: No, not yet, but I hope to in the near future.

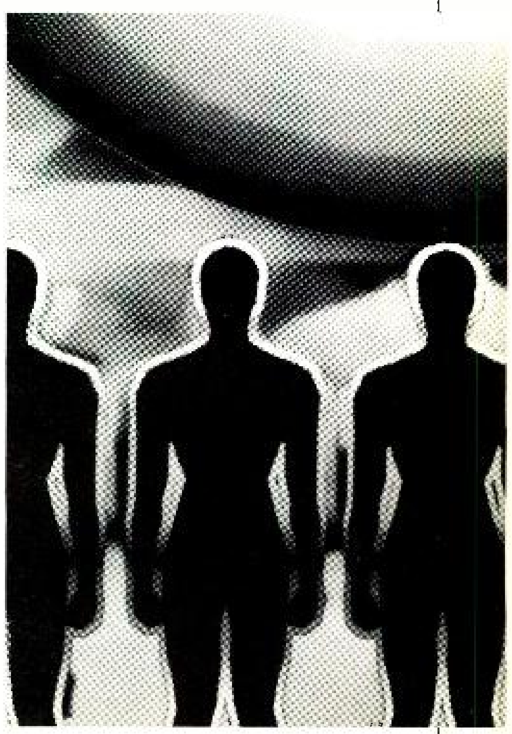
RE/P: *Do you have any favorite studio speakers?*
AH: I can mix on any speakers as long as they're not blown. It's just a matter of sussing it out. Every engineer has to do that because all the rooms are different, and you have to be able to hear through the discolorations of the control room.

RE/P: *How would you do it?*
AH: I would choose more transparent monitoring systems. I think some point source time align systems distort the reality of what's going on tape. The Quested monitors at Greene Street (in New York) were to me the most transparent. They sound more like my home stereo than anything. They're real flat.

RE/P: *A point source speaker doesn't impress you from the standpoint of phase and image?*
AH: Well, they have six drivers in each cabinet at Greene Street and I didn't hear any phase problems. I think they should try to make control rooms sound flat not "good." I think a speaker should be an honest reproducer. I don't have any schooling in engineering—I just learned by experiment and observation.

RE/P: *What about monitoring level? Do you monitor loud?*
AH: Not really. The monitors usually get up loud when the artist comes in the room, but I listen at all levels. Occasionally, I need to purge myself with crushing volume, but that is an exception.

RE/P: *Do you monitor differently when you are tracking and mixing?*
AH: In a regular tracking situation I listen pretty dry. I like to keep it very natural. The artists can have reverb in their cans—that's usually my assistant's job. I will have certain kinds of reverbs up that I'm thinking of using in the mix. For me, mixing starts when I'm doing the basics. My mix ideas are all inspirations. I don't say: well here we are with band type 163; therefore, we need to add reverb from Group C. Some of these mix ideas are pretty weird



Brian Lee is co-owner/producer and engineer of the Sync Tank, a post-production scoring studio in New York.

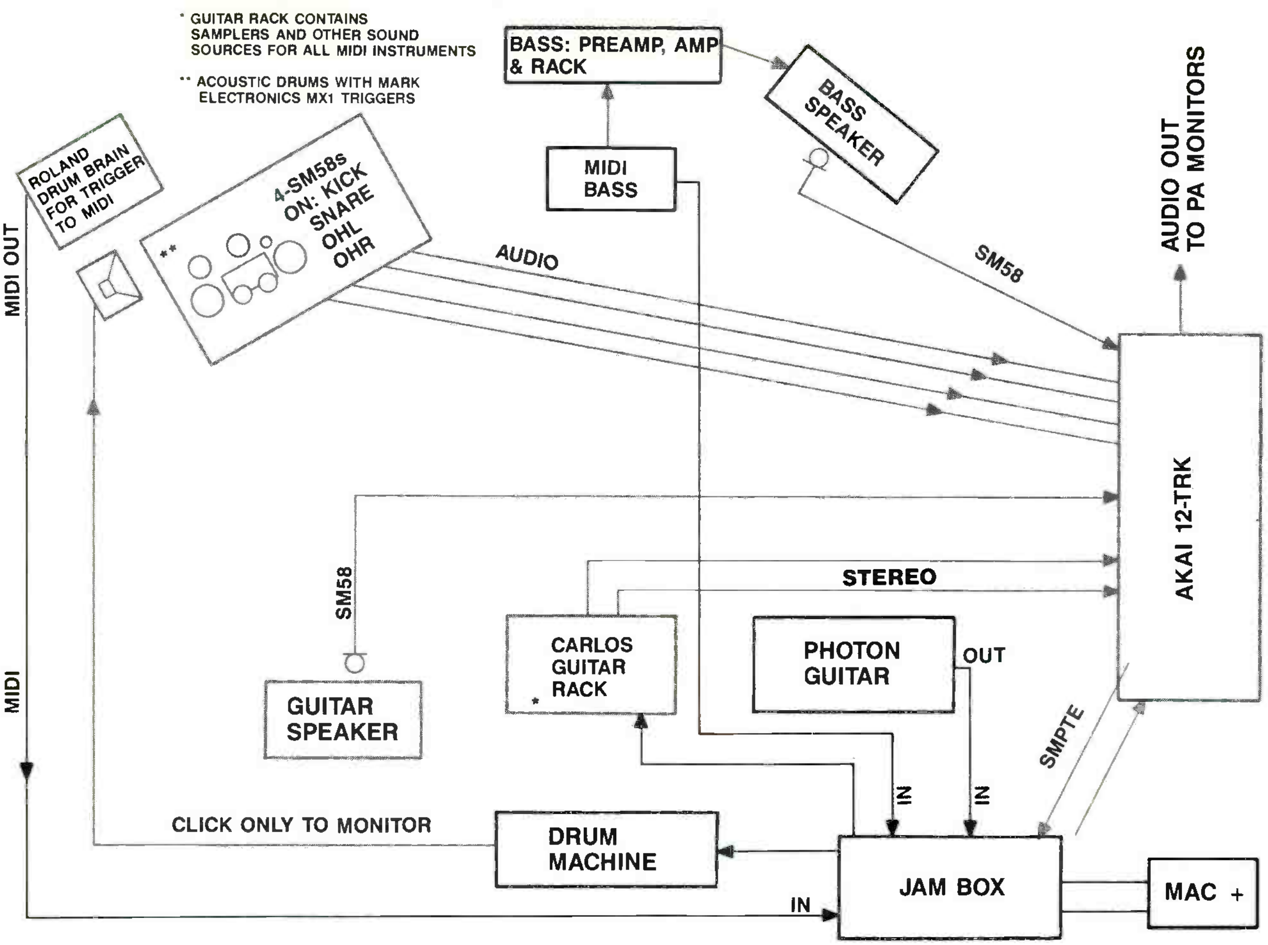
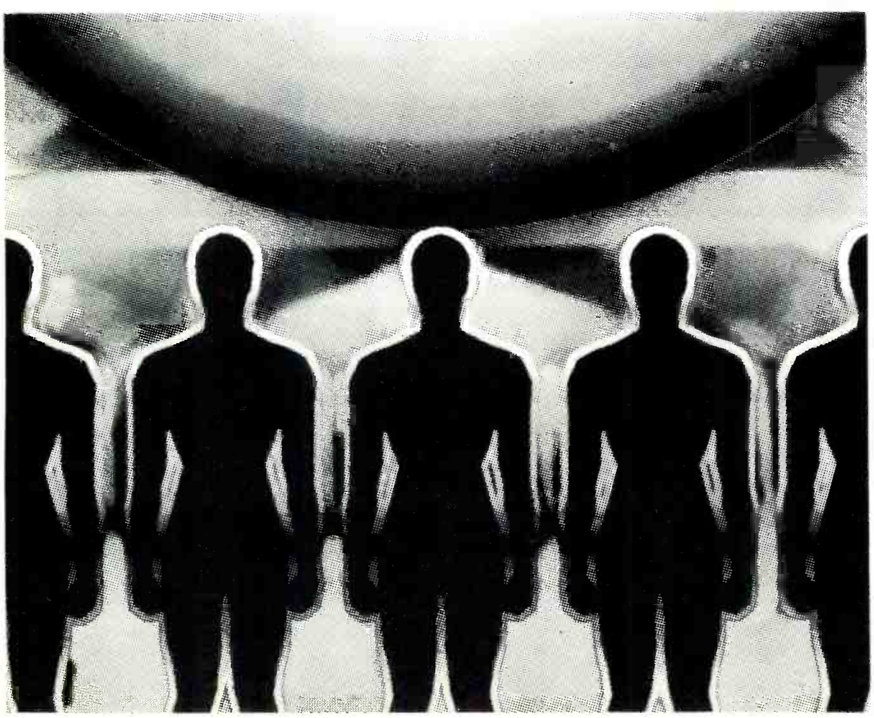


Figure 1. *Rehearsal setup for the upcoming Carlos Alomar album.*



and unless you have them in the back of your mind when you're tracking, they'll never come off. But, the most important thing to me in recording the music is the song.

RE/P: *Is the song more important than the sounds?*

AH: Yes, the song is more important than the sounds. Sounds to me are just things that I do. The responsibility of the engineer is to make a true recording of the sounds and *not* change it unless asked to. We only talk about sounds if we want something like a special effect.

For example, make my voice sound like a radio or something. If I'm inspired to make a drum go *Pgkk*, or just *Dmmm* or sound like a Broadway snare drum or a Power Station rip-off or whatever, that's just a special effect, an engineer lick. I think a recording engineer's job is to make an accurate recording.

If the musicians want to have a special effect or want their sound treated a special way, the producer or the artist will tell you.

You can't make a bad sound good, you can make a bad sound into a very interesting sound that someone might call a great sound. But I don't think that the engineer should be a guy that feels like he has the actual power to make a decision about what the sound should be like.

***RE/P:** Doesn't the engineer have a responsibility to reach for or suggest alternative sounds?*

AH: An engineer used to be hired as a scientist to make a clinical recording of what was happening. When I was a musician, the engineers I worked with would not offer any assistance. The producer's job is to be the philosopher of sound. I have enough to do trying to accurately reproduce—without coloring—what the musicians are playing.

***RE/P:** At what point would you make a suggestion?*

AH: I just present them with the best sound I can. Usually it is the most true reproduction of the instrument. On record dates (professionally) the musicians I have been blessed to work with have great experience. They usually present me with a sound that is perfect for the situation. Tommy Mandel always has an endless variety of bizarre and wonderful sounds. He and I once drove a producer to distraction using a pad layered with moaning girls, burping toilets and sherpa priests.

***RE/P:** How are you recording Carlos' new album?*

AH: We're using live MIDI. The whole drum kit (we have Alan Childs from the Bowie band) is MIDI, the bass (Carmine Rojas) is MIDI, Carlos' guitar is MIDI, and the only guy who's not is Ronnie Drayton, who uses a Marshall.

The album is going to be recorded into the computer and onto the 2-inch tape simultaneously. The last album was recorded all digitally but we decided to go analog on this record because we like the effect of pushing the tape to the limit. We use MIDI Paint by Southworth and we have the Jam Box, which has four MIDI inputs. Each musician gets to plug into the computer which gets SMPTE from the 2-inch. The bass gets four MIDI channels—one for each string, the guitar gets six MIDI channels—one for each string, etc. We're not using any quantized sequencing—everything is live playing.

Everybody's part is going on tape as a real instrument. A real bass and guitar but, at the same time, we're recording into the computer. We can turn around and say "Oh gee, Carmine's D string in the choruses would make a great flute part or a great conga part." Then Carmine's D string in the choruses becomes a conga part, too. The band is going to have the image of a rock band. It's going to be all guitar players and drums on stage. But everybody is MIDI, so they all have to be thinking split mind. Everyone will be playing several sounds at once.

That's the direction I think MIDI should be going, because we're getting back to musicianship. I happen to be blessed with some very talented friends who I get to work with. These are all *real* musicians who've studied and have a life history in music.

***RE/P:** Any particular problems?*

AH: Well, the Photon guitar pickup counts frequency vibrations. With a bass the time delay is much worse than with a guitar. Gibson, which distributes Photon, lent Carmine a bass with a Photon pickup and regular pickups. But the strings have to be thinner for the Photon pickup to read properly. As a result, the bass is normal size but it is one octave higher. For monitoring I used an Ibanez pitch shifter to get it down to the correct octave and compress it to get the glitchyness out. Then, when we play back the recorded MIDI, we have to track slide in the computer to compensate for the delay.

***RE/P:** How do you feel about MIDI in the control room?*

AH: I think the MIDI stuff is great. But I think people get too hung up on sequencing. I hate quantizing. Someday they'll all get beyond 16th notes and maybe get into triplets, which might make life interesting. But still, it will all be quantized and still it will all be this perfect square time, which is great for robots and computers, but for human beings I think it's boring. As I said, the album I'm doing now with Carlos is the direction I think MIDI should go.

***RE/P:** What about dance music?*

AH: Even for dance music I think its boring. I think music should be played. That's the difference between a musician and a technocrat who knows how to make things go *di-di-di-di-dit.* I think people are going to finally get tired of this fake music stuff. Even some rap records now are sampling real human drum beats and even though its the same bar being repeated, at least there's a human groove in that bar.

***RE/P:** What is M?*

AH: M is made by Intelligent Music. It is not really a sequencer but a compositional tool. I wrote a song called No Pain, No Gain with the random factors on 100%—nothing gets repeated. You can randomize note order, velocities, rests or anything.

***RE/P:** Isn't using an algorithmic composer even worse than quantizing?*

AH: Not really. There is a lot to be learned from breaking a composition down. For example, I like to transcribe minuets from "Don-Giovanni" into my computer and then play with the random tables to change the mathematics of the piece. I listen to Mozart played upside down or backward or whatever way I can make it sound interesting. The secret is to start with a good piece of music.

***RE/P:** Isn't that a conflict with your philosophy in the studio? With Carlos you take all these samples and try to capture the most human performances. Then you take Mozart—a sublimely human composer—and computerize it.*

AH: Well, it's self indulgent, but to me it sounds wonderful. We are all self indulgent. Engineering is not a profession where many people make a lot of money. Someone told me once if you want to make money, be a lawyer; if you want to be a musician—and engineers *are* musicians—be a musician.

***RE/P:** Do you want to be a producer?*

AH: Well, I always have a pet project that I'm doing on the side—like a musician or band that usually has no money but a lot of spirit and desire. Sometimes they have lousy sounds and I take an active role in writing, arranging and the overall sound. RE/P

Facility Spotlight:

Studio Jive

By Katsumi Otsuka

Located in a residential neighborhood, Studio Jive's design and equipment selection is designed to provide the best sound in the smallest space possible.

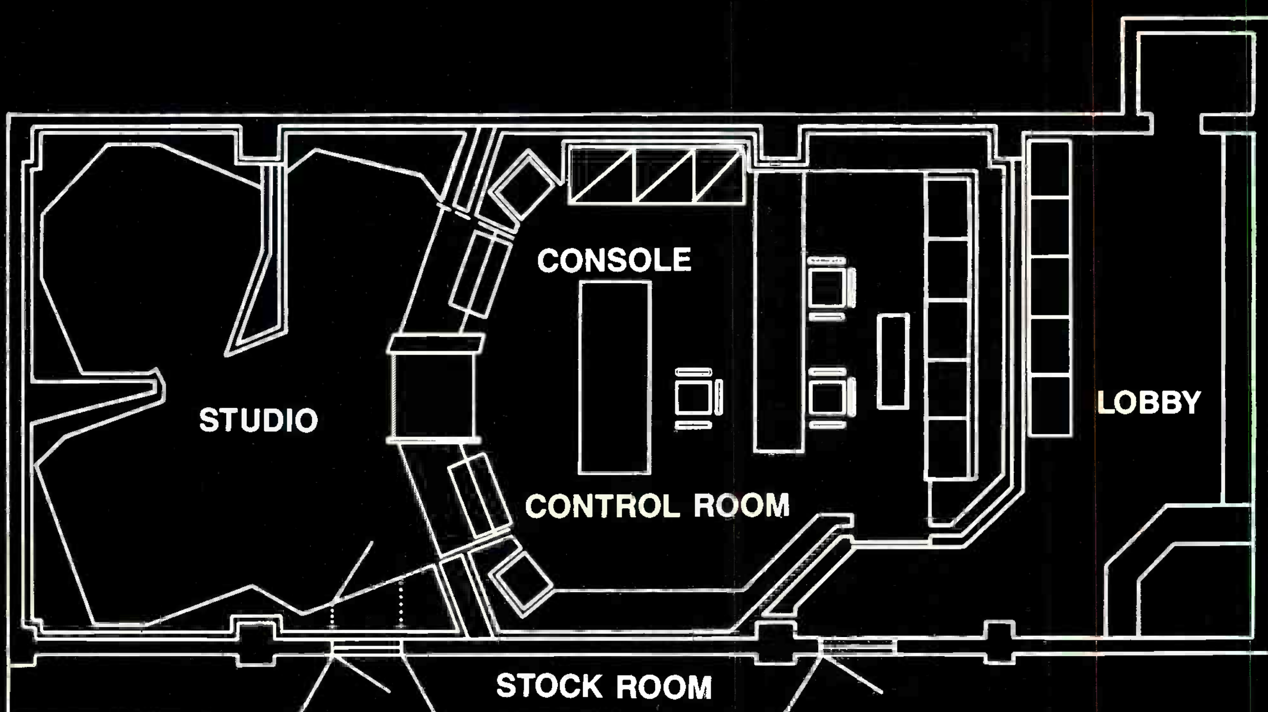


Figure 1. *Floor plan of Studio Jive.*

Studio Jive occupies the first floor of a three-story apartment building located in a quiet residential area in the western part of Tokyo. The building is owned by Tadashi Nomura. Originally, he had no relationship with the music business, but after he met a noted Japanese keyboardist, they hit if off and he became part of the recording world.

The two first considered opening a private studio. Soon, other engineers joined them and began planning the studio. In the planning phase, there was a problem concerning the level and quality of studio hardware. Eventually, it was decided to operate a commercial studio instead of a personal-use facility. The initial plans had to be changed drastically to accommodate professional clientele.

Jive began operation in October 1984. Clients are primarily rock or popular music artists, and the studio is used primarily for short projects.

Studio design

The design was commissioned to Masami Toshima, of the JVC Technical Lab. Toshima has designed many studios in Tokyo including Hupajam and Town

Katsumi Otsuka is an engineer at Studio Jive, Tokyo.

Studio Jive features an Amek APC 1000 console with GML Moving Fader Automation.

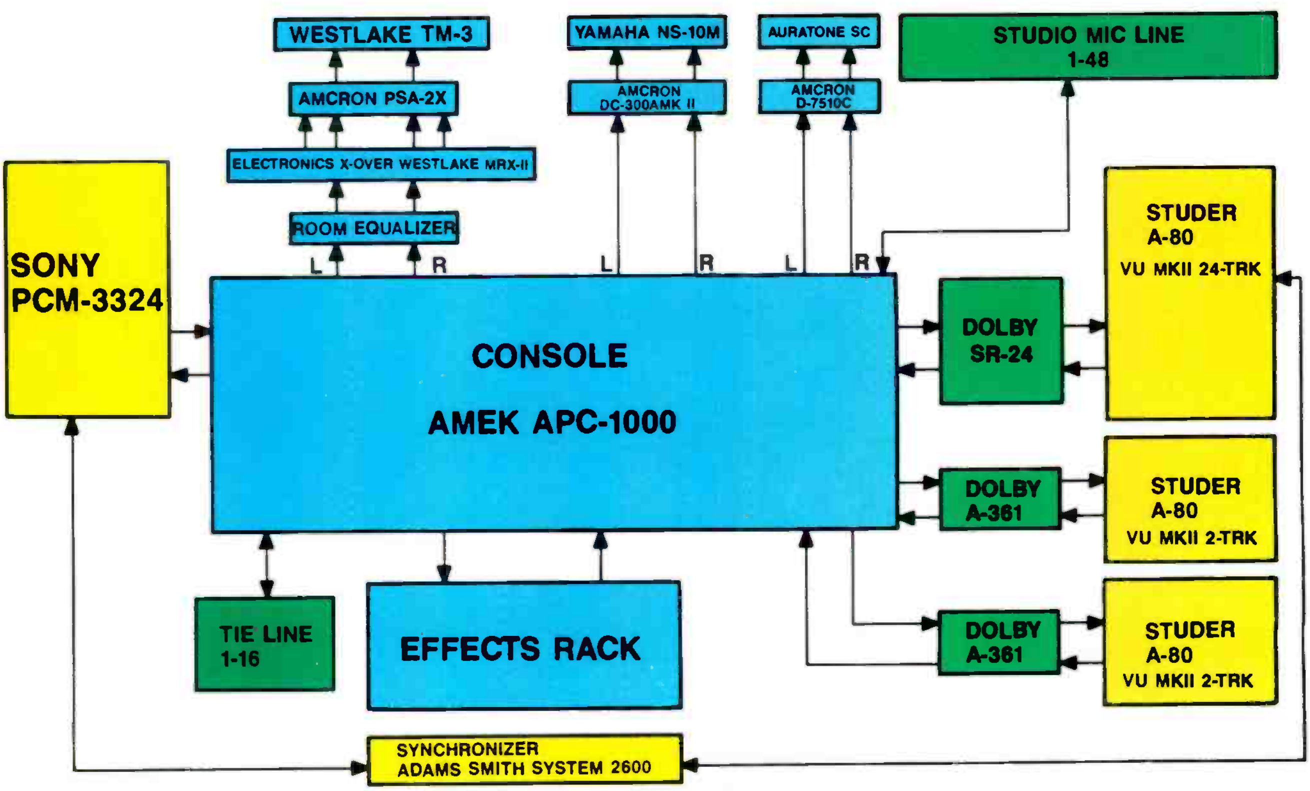
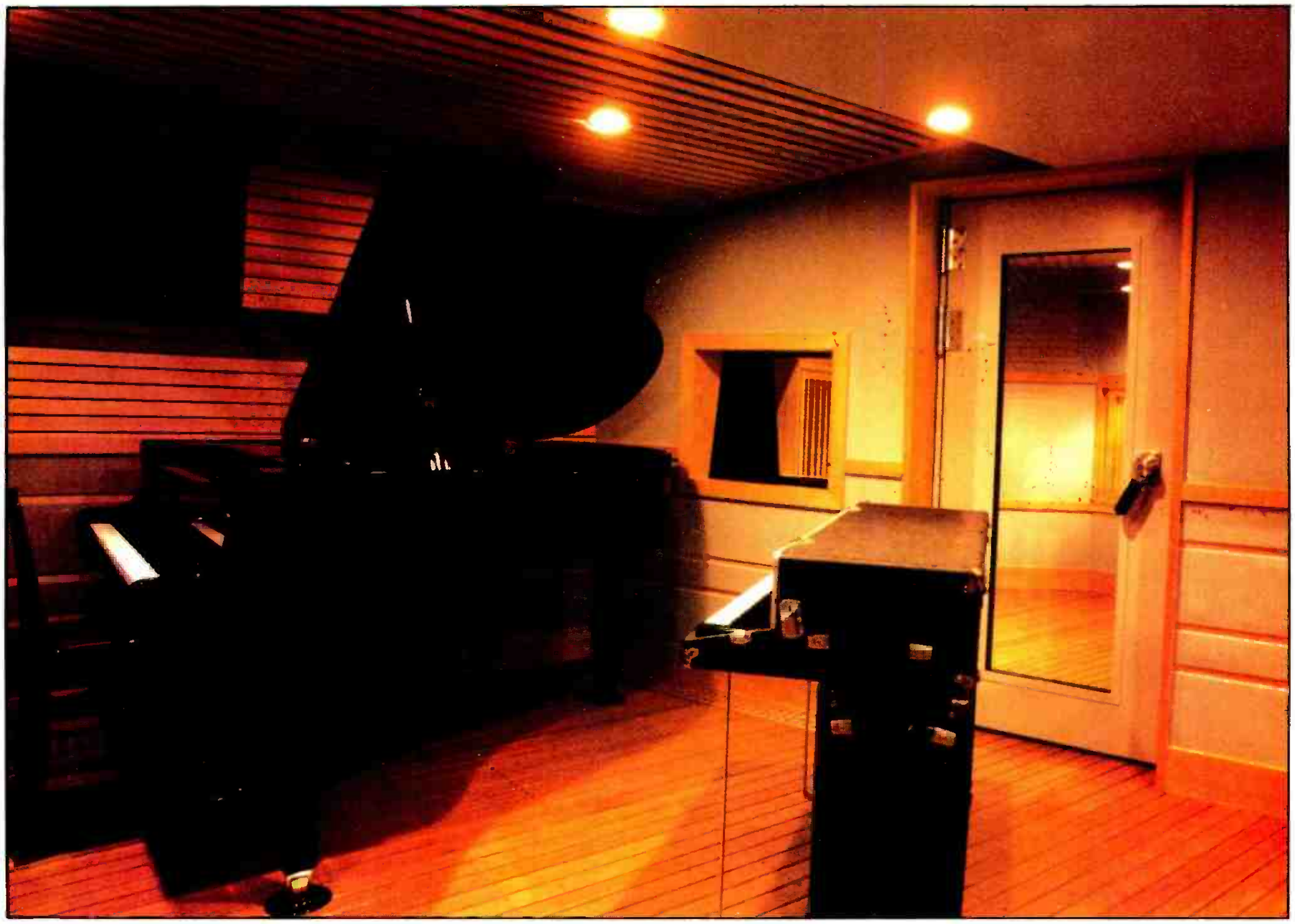


Figure 2. *Equipment layout for Studio Jive.*



Studio Jive's control room to studio space ratio is 6:4.

House 4 Studio in London.

Designing Jive included special problems. Because the site was on the first floor of the apartment building, space was restricted.

The first problem was how to split the space into a control room and a studio. The available area for both (not including a lobby) was about 330 square feet (100 square meters), thus allowing for only a small studio.

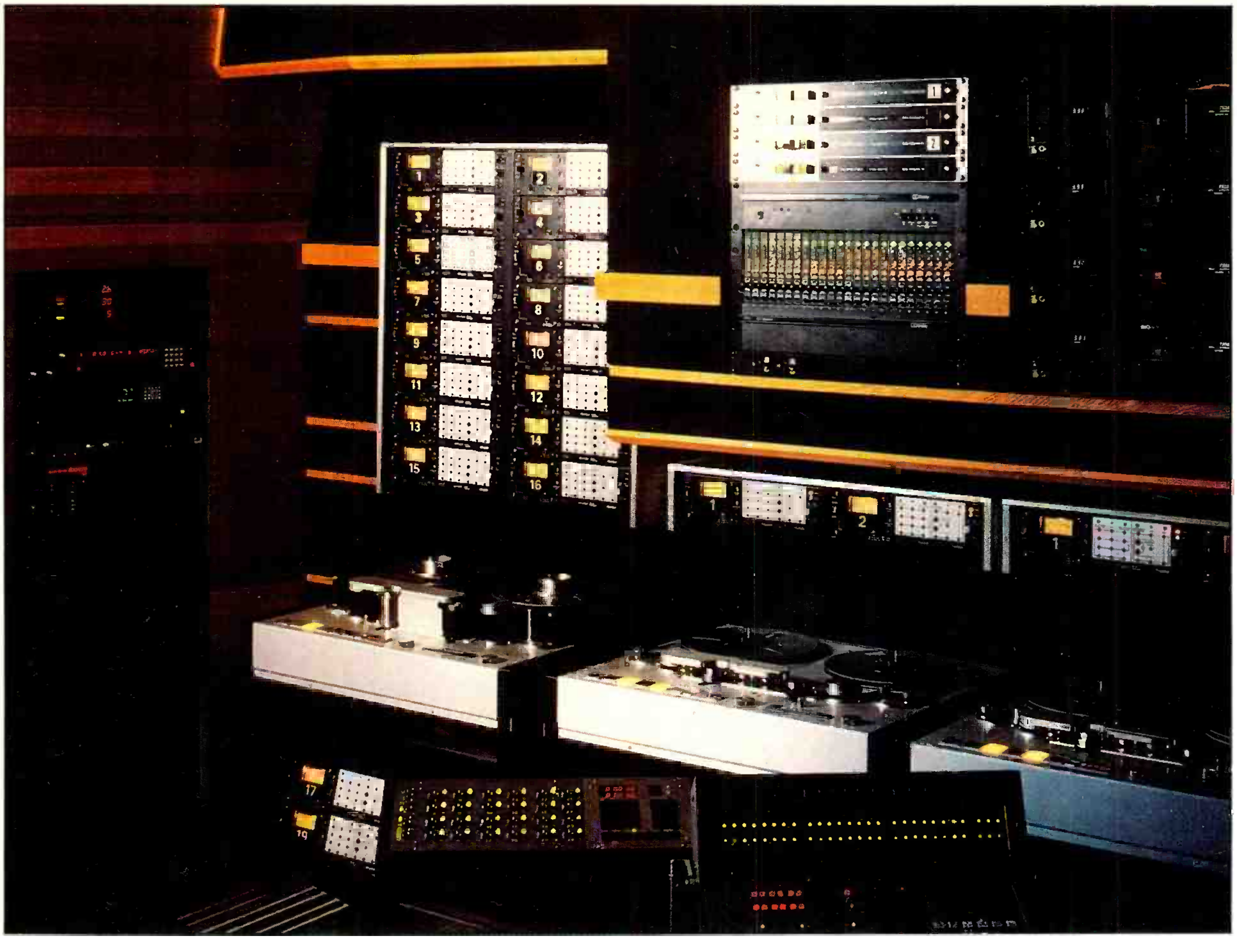
Recently, recording work has required a lot of desk room in the control room, for sequencer processing and setting synthesizers and drum machines. Although the staff also wanted room to record a 4-piece rhythm section, the staff agreed to give priority for the control room, in view of comfort as well as working efficiently.

The control room to studio space ratio was determined to be about 6:4. Because the second floor was used for housing, the ceiling had to be extremely soundproof.

The door from the control room to the studio is located between large speakers, and the passage between the control room and the studio can be used as an isolation booth. The door on the studio side is a sliding door to ensure space for opening and closing.

Monitors include Westlake TM-3s. With this system, acoustic measurements were taken several times when furnishing the interior of the monitor speakers. A structure the size of a console was temporarily installed to check the conditions so that interior materials could be changed if necessary.

At the same time, the studio owners requested that Toshima ensure a somewhat live sound. This was achieved using a suit-



Tape machines include a Sony PCM 3324 and Studio A-80 24-track and 2-track.

Detail of the Sony recorder.

Selected equipment list

Console: *Amek APC 1000 console with GML Moving Fader Automation; 56 in, 48 out.*
Tape machines: *Sony PCM 3324, and Studer A-80 24-track and 2-track.*
Adams Smith 2600 synchronizer.
Dolby SP-24 noise reduction.
Control room monitors: *Westlake TM-3s; Yamaha NS10Ms and NS10M Studios; Auratone 5Cs.*
Studio monitors: *JBL 4312s; Boss 301s.*
Power amps: *Amcron PSA-2s; DC-300A MkII; D-7510C.*
Reverbs: *AMS RMX-16; EMT-140 stereo; Yamaha REV-1, REV-7 and SPX-90; Sony MU-R201; Roland SRV-2000; Korg DRV-3000.*
Delays: *AMS 15-80S; Yamaha D-1500; Roland SDE-2000.*
Compressor/limiters: *Neve 33609; UREI 1178, 1176 and LA-4; BSS DPR-402; dbx 160X.*
Noise gates: *Kepex II, Dramer DS-201.*

able quantity of plates (tiles) on the floor in front of the control room. For the small monitors, Yamaha NS10Ms and Auratone 5Cs were selected.

A dead structure was selected for the acoustic design because of the limited space available. Under the floor, about 23 inches was ensured to provide a bass trap, which permitted a clear and low sound range to be produced.

Wiring design

The wiring design was commissioned to Yasuo Kita of Studio Equipments Co. Ltd., and features an orthodox structure, with a 48-bus output and 48 inputs for the console, and 48 inputs/outputs for the two multitrack tape machines. Canare and Mogami cable in various makes are also used. In the control room, pin 3 is hot. Musical interfaces such as MIDI were not included.

The original console selected was an Amek M2500, with VCA faders, 48 inputs, 24 outputs and the Optimix computer system. In August 1987, the studio upgraded to an Amek APC 1000 with GML Moving Fader Automation.

Virtual mixing

With the APC 1000, all switch setting and assigning can be performed on the center keyboard. The module width is reduced to one inch (30mm), so the console size is reduced, permitting more inputs to be included in a small space.

Virtual mixing within the scope of the console design philosophy attracted the studio's staff. The traditional recording technique is a two-stage process of recording signals onto multitrack tape and then remixing to stereo and adding sound effects processing during mixdown. Here, there is no operational difference between the monitor mix during track recording and the final mixdown.

If possible, two-step recording should be eliminated. From the phase of rhythm recording, it is preferred that the target is always the stereo mix. With virtual mixing (mix as you go) the goal is to create the end product from the commencement of recording.

With the APC 1000, the monitor mix fader in recording and the stereo mix fader in trackdown (mixdown) are the same process. Thus, the fader automation is used from the beginning. The console is consistently in the mixing mode—the normal setting for the APC 1000.

The APC is an all-input console lacking the concept of a setup mode that is common in most conventional consoles. Yet the channels can be set up individually.

In addition, the rough mix ordinarily carried out at every (over) dubbing process will be refined toward perfection through repetition. This makes it easy to obtain the final image and eliminates the necessity of starting from scratch in trackdown, enabling workload and time to be reduced.

The studio features Studer A-80 MkIII and a Sony PCM 3324 recorder with the Adams-Smith 2600 synchronizer being used for analog/digital synchronization.

In today's world, it is not unusual for "sounds" to lead music. Investment in hardware is essential to meet the needs of idea-rich creators. To meet these needs, recording studios can be thought of as the frontiers of music production. RE/P

Staff engineers

Masashi Kudo
Yuji Kuraishi
Katsumi Otsuka
Koji Sugimori
Hiroyuki Shimura

Computer Power Protection

By Mark Hill

Don't let your computer-driven audio equipment suffer damage from power-line disturbances.

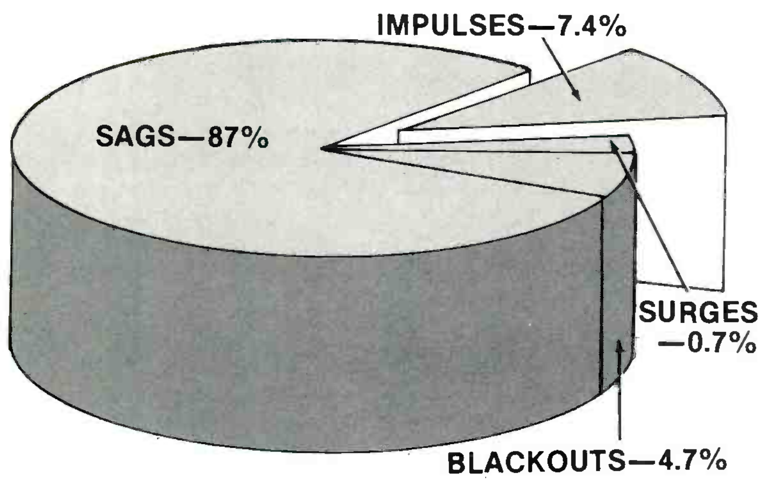


Figure 1. The pie chart depicts the average percentages of outages by category, measured in multiple locations over a two-year period.

Pure, raw power. We all need it and use it every day. But more than any other external factor, the quality of power provided by the local utility determines how well a piece of electronic equipment will operate. Disturbances in the power line, no matter how brief, can cause improper operation, permanent circuit damage, excessive heating, data loss and shortened component life.

Unconditioned power from the local utility may run 5% or more above or below the stated levels of 120Vac and 240Vac. In addition to voltage variations, a given power line may suffer from a number of other disturbances. The seven deadly sins of ac power are electrical noise, spikes, voltage surges, voltage sags, glitches, outages and frequency deviations. To combat these problems, you must understand the cause of each type of disturbance and its

Mark Hill is a technical service representative for International Tapetronics/3M, Bloomington, IL.

effect on electronic equipment (see Table 1, page 00).

Disturbances

Electrical noise generally manifests itself as hash or a series of spikes on the power line. Noise is usually much lower in amplitude than the single spike.

Electrical noise generally can be broken down into two types: RFI (radio frequency interference) and EMP (electromagnetic pulse). Any device that generates an arc or spark can radiate RFI ENERGY. This interference often originates in motors and motor-control devices, switches, delays, static and atmospheric discharges and even automotive ignition systems. Broadcast transmitters also are possible culprits. Electrical noise is not as destructive as a sudden high-voltage spike, but it can cause poor performance, accelerated component deterioration and altered data in computer circuits.

Voltage spikes, or transients, are sudden, short-term increases in line voltage, typically lasting 100ns or less. They may reach an amplitude of thousands of volts, with frequencies in the kilohertz or megahertz range.

Spikes or transients may be generated from many sources, but usually result from some type of inductive "kick." Switching heavy loads onto or off of the power line can cause large transients.

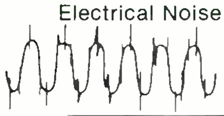
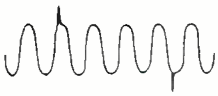
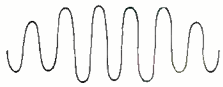
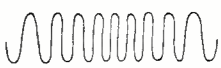
Even an electric typewriter, for example, when switched on and off can induce a 1,000Vac spike onto the power line. Power utilities often generate spikes when switching from one feeder to another. Severe spikes can cause permanent component damage, erratic operation and lost or altered data.

Voltage surges are increases in line voltage above the normal level for more than one-half cycle. Surges may last several cycles, seconds or minutes. They can be caused by removing a heavy load from a power line or switching feeder lines at the power utility station. Although a surge typically will not reach the magnitude of the single spike, surges can cause erratic operation, overheating, reduced component life and damage to input circuitry.

Sags usually are caused by switching heavy loads, either at the power utility station or within your own facility. Lightning also may cause short-term sags. The low voltage often causes power supplies within equipment to fall out of regulation, which, in turn, causes noise, improper operation or complete shutdown. Excessive heat probably is the most severe result of low-line voltage, in addition to lost or altered data.

Voltage sags, or brownouts, are defined as drops in voltage below the normal level

Table 1. These seven disturbances account for most of the power-line-induced damage to equipment.

Electrical Noise 	Atmospheric conditions Radar, Radio signals, Arcing equipment Switching apparatus	Individual data bits changed, data altered
Spikes 	Lightning Power line feeder switching Power factor capacitor Switching Turn-off of heavy motors	Data altered, wiped out Circuits damaged
Surges 	Sudden load decreases Switching of feeder lines	Data altered Equipment shuts down
Sags 	Lightning Turn-on of heavy loads	Data altered Equipment heats up excessively, leading to early failure Computer shuts down
Glitches 	Power line feeder switching Circuit breaker re-closing Brief short-circuits	Disk heads crash Data altered Equipment shuts down
Outages 	Accidents involving power lines Transformer failures Generator failures	Disk heads crash Data altered Equipment shuts down
Frequency Deviations 	Generator instabilities Huge load changes	Data altered Disks shut down

for one-half cycle or more. In extreme cases, sag can last for several hours and cause permanent damage to electronic equipment.

The infamous glitch is actually a power outage or near-outage lasting less than one cycle. Glitches often occur when the power utility switches feeder lines. The glitch also may be accompanied by a voltage spike generated at the completion of the switchover. Glitches often result in data loss, injected errors in computer systems and possible head crashes on disk drives.

Although uncommon, a power outage, or blackout, can be the ultimate disturbance, resulting in an unplanned equipment shutdown.

Frequency drifts are uncommon on U.S. power lines, but are inherent to motor-driven generators. Line-frequency variations can cause improper operation, speed variations in motors, data alteration or disk drive shutdown.

You may be familiar with the effects of these disturbances in severe cases, but you may not be aware of the damage that can be done to equipment when it is exposed repeatedly to seemingly minor disturbances. Capacitors subjected to spikes can break down and short. Resistors can heat up from power sags and electrical noise, changing value over time. Semiconductors and ICs, although appearing to function normally, can deteriorate in performance and eventually fail completely. Clocked-logic systems sometimes become confused by glitches, spikes and electrical noise. In general, dirty power lines are detrimental to the operation of all electrical equipment.

Monitor the lines

What can you do? Before running out to buy power-line protection, first deter-

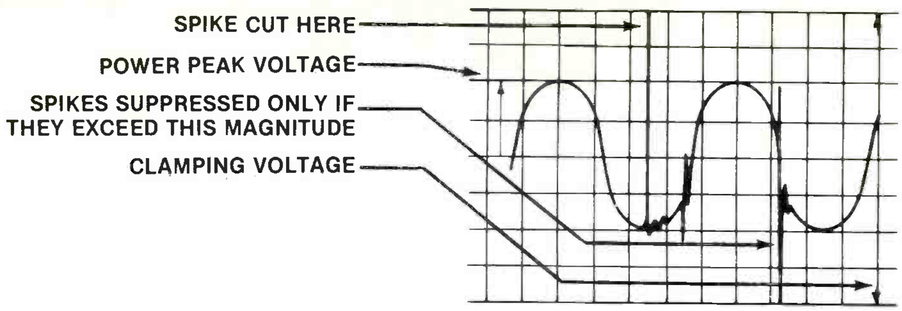
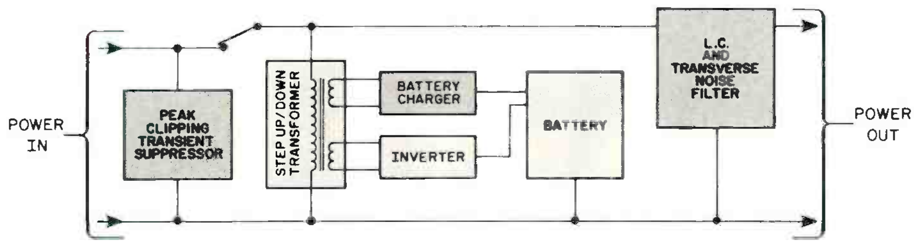
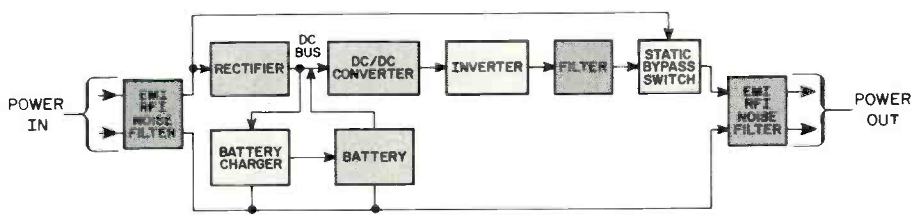


Figure 2. *Spike clippers suppress only voltage spikes that exceed the clamping voltage, which is customarily set at approximately 200Vac for a nominal 120Vac power line.*



The typical standby power system protects only against blackouts. Also, the transition time between modes can create problems for computer-based equipment.



The UPS system provides constant operation. There is no transition time outage between line and backup operation.

mine your requirements. What types of disturbances are occurring at your facility? How frequently do they occur? Is your area prone to frequent outages, electrical storms or ice storms? A chart of the relative frequency of common power-line problems is shown in Figure 1.

You might contact your local utility company to inquire whether it will rent or loan you a chart recorder, which can be connected to the power line for a few days. A local computer service company also might have a power monitor for rent. This type of monitor provides a graphic indication of the types of disturbances present on that line.

Also consider what areas to protect, and the resultant cost of a power outage in those areas. A disturbance in the studio may be catastrophic, while a shutdown of the office computer may just be inconvenient. Once you have identified the areas requiring protection, calculate the load requirements of each area. Consider more than just the present requirements. Try to project what your needs will be one or two years from how. Perhaps you will be adding a new recorder or console. Many digital tape systems are major loads to the power system.

Identify the noise source

Now that you have identified the types of disturbances and the areas needing protection, examine the situation carefully. There are several things to look for to help reduce the disturbances:

• Grounding: Proper grounding is essential. Many power-line disturbances can be reduced or eliminated by proper grounding. It is important to note that what may be an adequate ground for protection against electrical shock may not eliminate electrical disturbances. Voltage spikes and RFI can be capacitively coupled into equipment and improperly shielded audio or control lines.

Care should be taken to ensure proper grounding of equipment and all connecting audio and control lines. (See the January issue of *RE/P*.)

• Line load: Is the area requiring protection on a dedicated line? If not, what other equipment is on that line? Moving heavy inductive loads to different circuits may be necessary. Dedicated lines can be installed to protect an area, but this may provide protection only from internally generated disturbances.

• Isolation: Some pieces of equipment are, by nature, electrically noisy. Some use power in surges; isolating this equipment with transformers may result in a significant improvement. Another alternative might be to isolate the offending equipment to one phase of the 3-phase supply,

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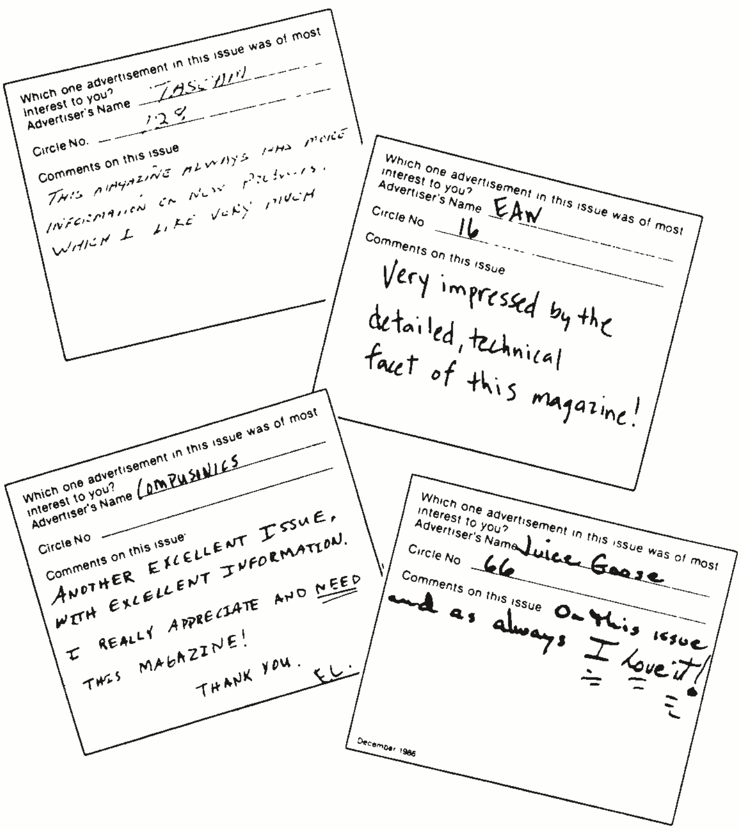

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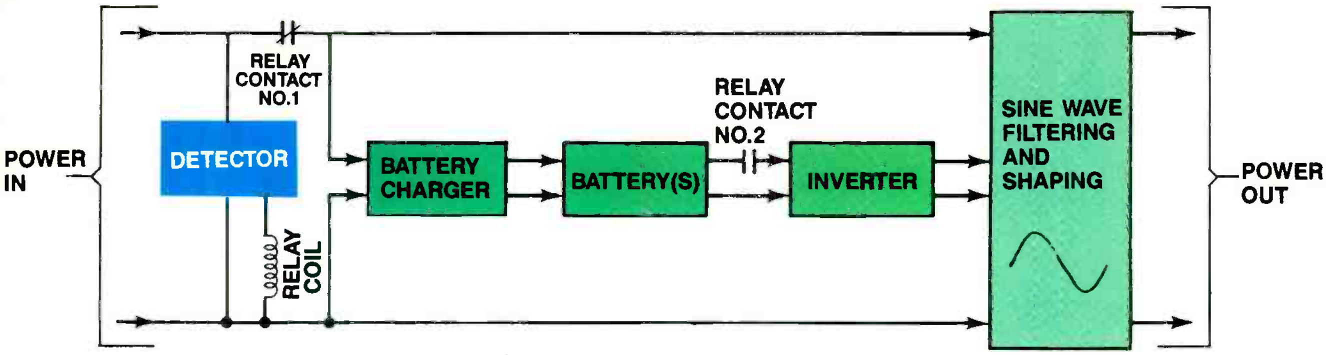
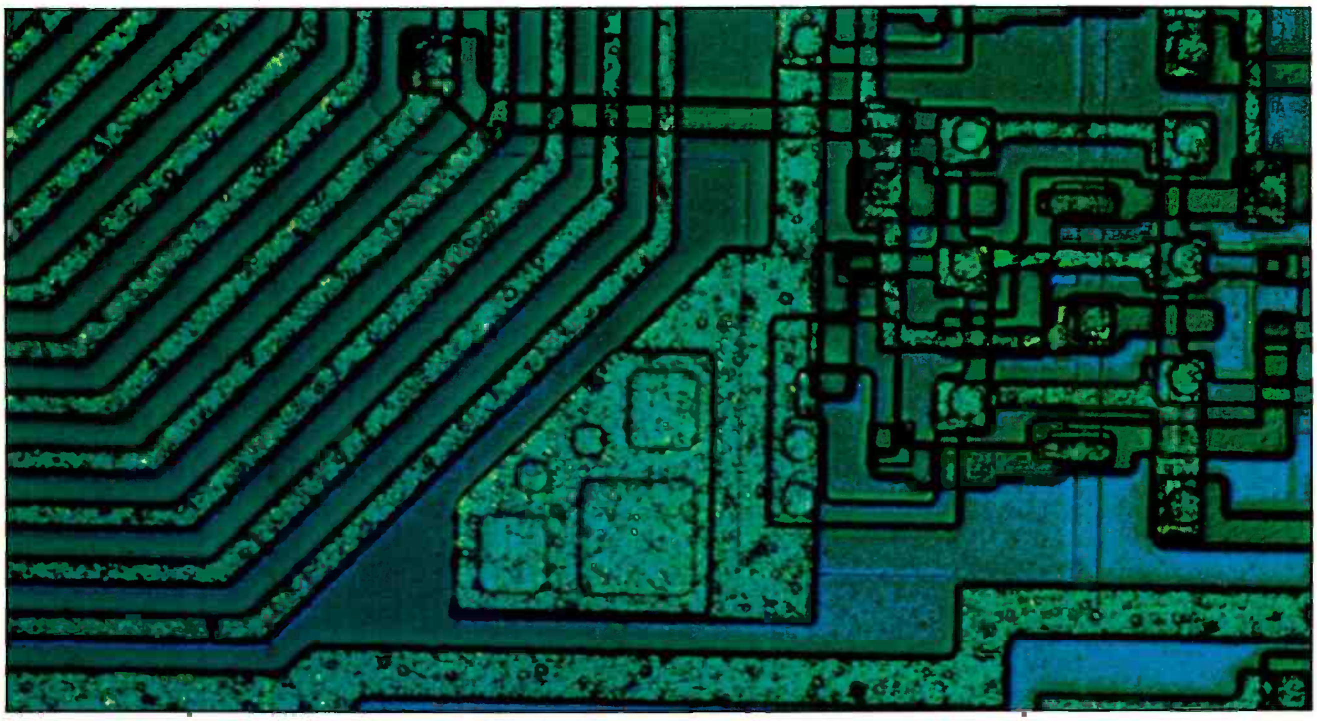


In addition to providing standby power protection, some power supplies also provide line-voltage regulation and noise filtering, while operating in the normal on-line mode.



The increased dependence on integrated circuits in hardware demands careful attention to suppression of transient disturbances on the ac power lines into a facility. Transient overvoltages represent the greatest single threat to proper operation of ICs and other sensitive components.

while running sensitive equipment from the opposing phases.

• Peak demand: If power disturbances seem to occur at the same time every day, try to identify the source of the disturbance. Usually a little detective work will lead you to the cause.

Install protection equipment

Once you identify the source, you can be relatively confident about the steps to take toward solving the problem. A wide range of power protection systems and devices is available. Most of them fall into one of five categories:

• Surge suppressors, or spike clippers, shunt voltage spikes that exceed the clamping voltage of the device (see Figure 2). Typically, this is set at 200Vac on a 120Vac line. Surge suppressors offer no protection from spikes less than 200Vac or from voltage reductions or electrical noise.

• Isolation transformers electrically isolate equipment from the power line. They protect against electrical noise and short-term voltage spikes. However, isolation transformers offer little or no protection against longer duration spikes, glitches or low-voltage conditions.

• Voltage regulators generally provide no isolation, but merely regulate the supply voltage. Regulation usually is accomplished by electronically switching taps on the transformer's primary winding. Equipment is left exposed to spikes and sags that occur too quickly for the regulator to react.

• Isolators/regulators combine the protection of an isolation transformer with a voltage regulator. This combination provides effective protection against most disturbances. There are two types of devices commonly available: tap-switching and ferroresonant. Tap-switching devices, like the voltage regulators, fall short in protecting against longer duration spikes and require one to three cycles to correct voltage fluc-

UPS vs. standby

Selecting the type of power protection to use requires careful consideration. Many facilities rely on computer-based equipment, which is less forgiving of power-line disturbances than some other devices. In many cases, the importance studios now place on computers may demand that the computers never suffer a brownout or power outage.

If this applies to you, a UPS or standby power system may be required. Both devices rely on an inverter to convert dc voltage from a battery into 120VAC for power during an outage. There are, however, distinct cost and performance differences between a UPS and a standby system.

A standby system consists of an inverter, battery, battery charger and high-speed transfer switch. When not activated, the inverter is at rest and the primary ac power passes straight through to the load. When the ac voltage drops below a preset threshold, the load will be transferred and the inverter switched on to supply ac power. When the ac returns, the load is switched back to the ac line, the inverter is switched off and the battery is recharged.

The standby system is sometimes called an off-line UPS because during a power outage there is a lapse during the switching process from ac to dc power. The switching time for a standby power system is typically from 2ms to 10ms. Standby systems protect only against blackouts; they do not protect against line voltage problems such as surges and sags.

UPS

The UPS system is similar to the standby power system, but doesn't have a transfer switch. The load is continuously supplied power developed within the UPS. The line ac is converted to dc to charge the battery and to power the inverter section. The inverter converts the dc back to ac, which powers the load.

A drop in the ac line causes the battery charger's dc output to decrease, but the battery automatically compensates and continues to supply dc power to the inverter. The inverter's ac output to the protected load continues with no interruption.

In addition to providing blackout and brownout protection, the UPS also serves as a power conditioner. Low voltages, spikes, surges, noise and most other power-line problems can be eliminated by a UPS. A UPS has the advantage of no switching time; the unit is always on-line. This feature is important for many computers that cannot tolerate the momentary outage produced by a standby system.

Other factors

The standby system often is less expensive than the UPS. The standby system can be useful for equipment that can withstand the switching time during an outage without suffering malfunction or component failure. More sensitive and sophisticated equipment may require the continuous operation provided by a UPS. In both systems, the length of time you can operate off-line depends upon battery capacity and load-current requirements.

tuations. Neither type can protect against power outages.

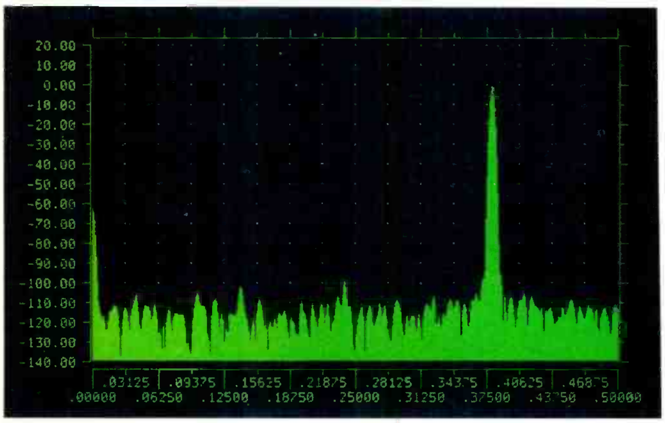
• Uninterruptible power supply (UPS) systems offer the utmost in power-line protection, guarding against all forms of interference, including power outages. However, depending on type and battery capacity, a UPS system can cost several times more than an isolator/regulator of comparable capacity. UPS systems are invaluable where a clean, steady and uninterrupted power supply is critical.

Countless hours are spent nationwide, tweaking and tuning equipment to achieve the utmost in performance. All too often, the power being supplied to that equipment is taken for granted. Power protection requirements vary greatly, and require a careful analysis of each facility.

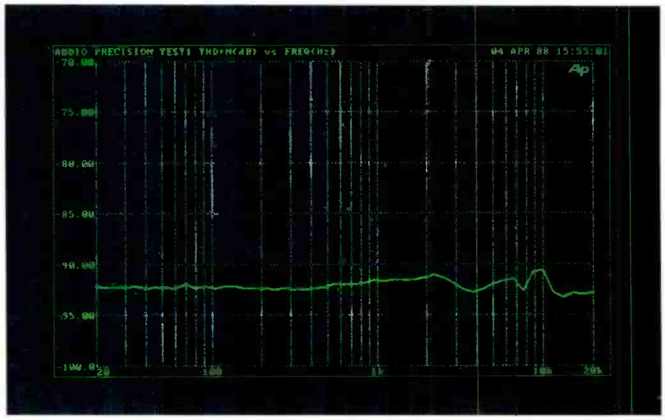
The cost of protecting your facility could vary greatly, depending on size, magnitude of the power problems and level of protection required. In nearly every case though, the cost of power-line protection is small compared with the cost of repair problems. Carry your good interfacing practices right down to the power line, and you are almost guaranteed success.

RE/P

Stop, look, listen.



FFT: Detailed distribution of noise and harmonic distortion vs. input signal. Measurement of sample and hold and ZAD2716 A/D converter only. Computer analysis receives digital information from A/D converter (to avoid DAC reconstruction distortion).



Using an Audio Precision system, a sine wave is processed through sample and hold, A to D, ZAD2716 A/D converter, and Apogee AP filter. Graph shows combined total harmonic distortion and noise vs. frequency.

Introducing the world's finest professional 16-bit stereo digitizing subsystem and 18-bit deglitched reconstruction DAC.

Want to position yourself at the leading edge of digital audio? Then hear this.

Analog Solutions gives you a new family of professional audio products with confirmed "Golden Ear" performance using the industry's two most powerful test criteria. The eye. And the ear.

If seeing is believing, you'll be impressed by the harmonic distortion and noise plots of our ZAD2716/ZSH202 digitizing subsystem (left) and ZDA1801 deglitched reconstruction DAC (right). Nobody's ever seen plots like those before.

But there's more to the story than a good plot.

The ZAD2716-2 coupled with the ZSH202 Dual Sample-Hold can provide stereo digitizing at a 50 kHz sampling rate per channel. The ZSH202 allows simultaneous sampling of both channels within ± 2 nsec with guaranteed –96 db/channel separation. And there's no sacrifice of phase linearity or sin x/x. In fact, we guarantee sin x/x performance of 0.01dB and phase linearity of 0.1 (degree) over 0 to 20 kHz. (Performance that brings a smile to any Golden Ear.) Cost savings are significant on a per channel basis, because you only need one converter for two channels.

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We maintain the same outstanding performance at the back end of your system with the ZDA1801. This 18-bit DAC, with a built-in deglitcher operating up to 200 kHz for up to 4X oversampling, offers guaranteed signal-to-noise plus harmonic distortion of 94 dB over the full audio range.

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STUDIO UPDATE

Contents

Attention Studio Owners

Studio Update has been expanded, making it the industry's most complete source of studio news. But to make it work, we need your input:

- Talkback: Do you have a tip on solving an audio problem or an unusual request? Send it to Talkback. If we print it, we'll pay you $50 and credit you and your studio.

- Studio News: The latest in equipment updates and personnel additions.

Mail to: Studio Update, RE/P, P.O. Box 12901, Overland Park, KS 66212.

Talkback

Solving sync problems

By Bill Ivie

The problem: The video is finished. They love it. But it is eight seconds too long and there is no reference audio on the video master. Because there was no audio on the master, there was no beginning reference point for the music.

Solution: First a tach reader was attached to the tach output of the 2-track. Then the drummer and I worked together writing down the tach frequency required to sync-up each section of music. In order to ensure that there wasn't a noticeable variation in pitch or tempo, we went through the entire five-minute piece eight bars at a time. Finally, with the drummer calling out the numbers, while I was watching the tach counter, I rode the varispeed through the whole song—in one pass.

What happened: Before taking their project to post production, the band realized they had a problem, but neither the band nor the staff at the first video house knew how to fix it. In this particular case, the Nagra copy was made improperly or was not running in sync in the field. When they edited the video using the incorrect Nagra music dub as their reference audio track, the finished edited 1-inch video master was eight seconds longer than the correct music master. To make things more fun, the engineer at the first editing house managed to erase the original reference audio on the 1-inch.

What they should have done: First of all, choose the post-production house before shooting begins. By using one facility for all transfers editing and sweetening, you minimize the chance of such disasters. At National, bands start by making a "rock video package." This begins with dubbing the music master to a 1-inch video, a 4-track with stereo music and time code, and a ¼-inch Nagra to shoot with in the field. All of these dubs are made simultaneously, with all machines referenced to house sync. Then, when the band shoots on location all the audio and video machines are referenced to video sync or 30 NSF and will assure lock-up in post production. RE/P

Bill Ivie is senior mixer at National Video Center, New York.

Have you encountered a problem or unusual request during a recent job that required a unique solution? We would like to share it with the industry.

Send it to "Talkback"; if we use it, we'll pay you $50. "Talkback " is a new forum for sharing your solutions to difficult production situations other engineers may encounter. In a continuing effort to educate, we feel this type of information is helpful and will display your professional abilities. This is not a tech tips column/rather, the focus is on solutions to problems—technical or non-technical.

Each month *RE/P* will select one or more "Talkback" pieces for publication.

To submit: In 1-2 pages, describe the job, what the problem was and what you did to solve the problem. Include any supporting documentation, such as diagrams or photos, that would help explain the situation. If we publish your entry, you and your company will be fully credited. Send material or inquiries to Michael Fay, Editor, RE/P, 8885 Rio San Diego Drive, #107, San Diego, CA 92108.

Spotlight

Universal Music & Post

By Dan Torchia

This post studio prospers in a smaller market by attracting nationwide clients and diversifying its operations.

Universal Music and Post's Studio A features a Soundtracs CP-6800 console.

In 1984, Rod Slane saw his studio business changing in the same direction as the rest of the industry.

He had just moved his 24-track studio from his house to a growing section of Tulsa, OK. Band business was declining, and opportunities were opening up in corporate/industrial work. But the studio's name suggested a music studio, and corporate clients were not forthcoming.

With an eye on changing the studio's image, Slane changed the name and embarked on a promotion campaign to attract corporate clients.

Today, Universal Music & Post does most of Tulsa's post-production work. In addition to four studios in a 5,000-square-foot complex, the facility has a music production library with more than 420 themes, intro/promos, stingers and news grids, a music/home video production company and a film company whose first produc-

Dan Torchia is staff editor of *RE/P*.

tion, a documentary, recently aired on the Discovery Channel.

If diversification is the general rule in the industry, it's mandatory in a smaller market such as Tulsa. With limited band business and in-town clients totaling only about 50% of business, Universal had to broaden its client base and diversify into new business ventures.

"About five years ago, before we moved to the present location, I saw a decline in what I call cash business, which is primarily bands," Slane says, "and we tried to figure out how to increase our business. We saw that the in-town market would not increase, but that there was an opportunity in nationwide business."

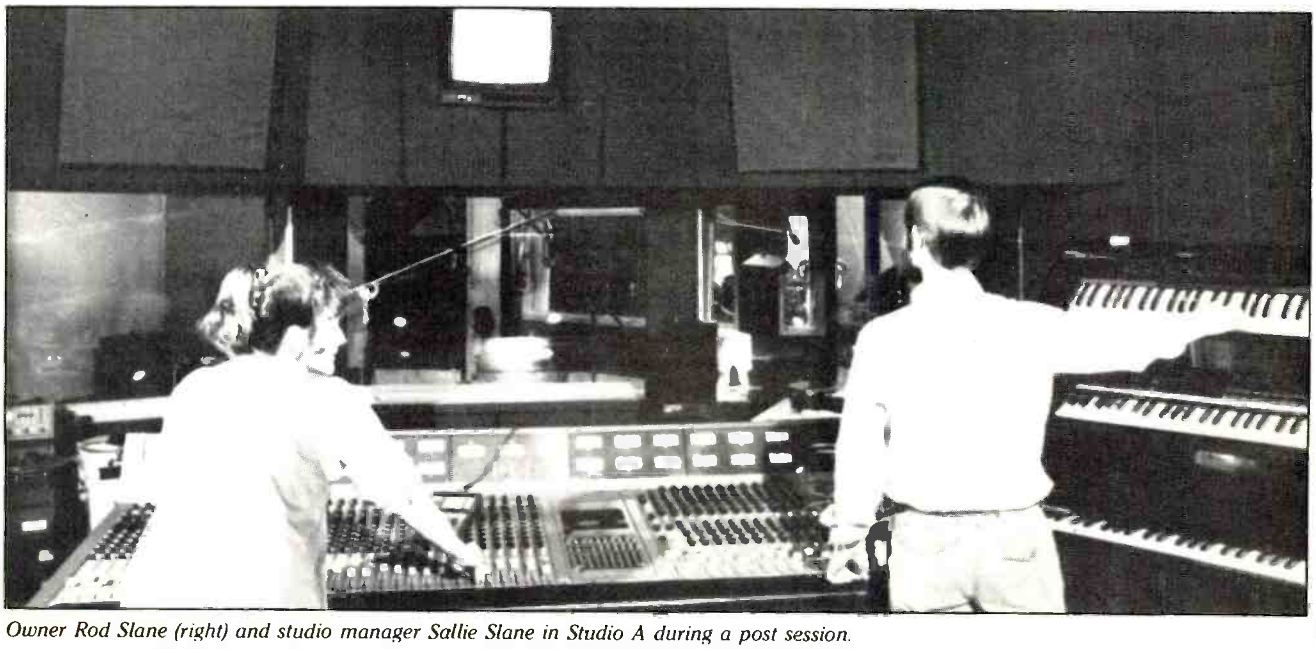
Studio background

Like many studio owners, Slane started his first studio on a shoestring budget and learned engineering along the way. A musician who became an engineer after becoming dissatisfied with the engineering work on his own sessions, Slane opened a 4-track studio, called Star Track Recording, in Little Rock, AK, in 1978.

He moved to Tulsa in late 1979. The jump to 16-track occurred in 1980, with 24-track following in mid-1981. In February 1984, Slane moved the studio out of his house to the present location.

Although Slane's intention was to get into corporate work, the studio's name was still Star Track Recording, which put the studio in a sort of limbo. The rates had in-

Owner Rod Slane (background) and studio manager Sallie Slane during a post session.



Owner Rod Slane (right) and studio manager Sallie Slane in Studio A during a post session.

creased, which caused a decline in band business. But Star Track suggested a music studio, and corporate clients were not booking time.

"We changed the name because we wanted to change the image," Slane says. "We asked ourselves, 'What is it that we do?', and we came up with the name."

Along with the name change, the studio embarked on a promotion campaign for the rest of 1984. It conducted open houses, seminars with potential clients, created direct mail pieces and completed a video demo tape. The campaign worked. And, beginning in 1985, corporate business picked up.

Universal attracted one of its biggest clients by redoing one of its recent corporate pieces for free. The client liked the results and is now a steady customer.

"It was basically an education process," Slane says of the promotion campaign. "We were able to do a lot of new things technologically, and we wanted to make them aware of that."

Studio layouts

Universal's four rooms each appeal to different client groups. Studio A is the facility's 24-track room, and is used mainly

At a Glance

Owner: *Universal Music & Post Inc. (Rod and Sallie Slane, owners).*
Studio manager: *Sallie Slane.*
Studio B engineer: *Keith Slane.*
Staff writer/arranger: *Joe Ninowski.*
Other services: *Musipak (music production library); Spectral Films (documentaries); Mellenium Films (music/home video).*
Studio design: *Paul Westbrook.*

Studio A

Control room: *16'x18'.*
Studio: *30'x24'.*
Drum booth: *10'x10'.*

Equipment:
Soundtracs CP-6800 32/24/12/2 console.
Otari MTR-90 24-track.
Otari MTR-10 2-track.
Otari MX-50/50 ½-inch 4-track.
MCI JH-110 1-inch layback recorder.
JVC CP5550U ¾-inch video player.
Adams Smith synchronizers.
Roland SBX-80 synch box.
Studer CD player.
EXR Aural Exciter.
UREI C-35 graphic EQ.
JBL 4430 monitors.
Fostex RM-765 reference monitors.
Lexicon PCM-70 digital effects processor.
Eventide Harmonizer.
EMT echo plate reverbs.
Lexicon PCM 41 digital delay.
Yamaha REV-7 digital reverb.
Omni Craft gates.
dbx 150 Type I NR.
Roland Compu-Editor.
dbx 160X compressor/limiters.
Scamp Rack.

Rates:
- *24-track recording, automated mixdown, 2- or 4-track mastering: $95/hour.*
- *Synchronization: $145/hour.*
- *Music scored to picture: quoted per project.*
- *Block rates available.*

Studio B

Control room: *13'x16'.*
Studio: *10'x12'.*

Equipment:
TAC/Amek 24/8/2 console.
Otari MX-70 16-track.
Otari MX-50/50 2-track.
Technics 1500 2-track.
Yamaha REV-7 digital reverb.
dbx 150 Type I NR.
Adams Smith synchronizer.
EXR Aural Exciter.
UREI C-35 graphic EQ.
Omni Craft GT-4 noise gate
dbx 160 compressor/limiters.
Yamaha E1010 analog delay.
JBL 4313B monitors.
Auratone speakers.
Roland Compu-Editor.

Rates:
- *2-track recording: $45/hour.*
- *16-track recording/mixing: $65 hour.*
- *Block rates available.*

Studio C

Dimensions: *12'x14'.*

Equipment:
JVC CR850 U-matic recorders.
JVC RMG850U controller.
Panasonic 14-inch video monitors.
Nagra recorder.
Crown amplifiers.
Fostex monitors.

Rates:
- *¾-inch video off-line editing with editor: $55/hour.*
- *Without editor: $35/hour.*

Studio D

Dimensions: *14'x16'.*

Equipment:
16x16x2 custom console.
JBL 4311 monitors.
Crown amplifiers.
Yamaha REV-7.
JVC ½-inch VHS hi-fi VCR.
E-Mu Systems Emulator II sampler.
Roland S-50 sampler.
Roland D-50 synthesizer.
Roland Jupiter 6 synthesizer.
Yamaha RX-5 drum machine.
Yamaha DX-7.
Korg Poly 61.
Korg EX-8000.
360 Systems MIDI Bass.
Apple Macintosh SE with Southworth Systems MIDI Paint Sequencer software and Jam Box software.
Pearl drum set.
Roland Octapad.

Rates: *$40/hour with programmer.*

Film production equipment:
Arriflex S 16mm camera.
Arriflex BL 16mm camera.
Mathews Doorway dolly.
Lowel Light package.
Mole Richardson spotlights, 2kW, 5kW.
Prime lenses: 10mm/40mm macro/90mm macro.
Zoom lenses: 12-120 Angenieux.

Address: *5840 S. Memorial, Suite 210, Tulsa, OK 74145; 918-622-6444.*

for music scoring, jingles, Foley work, ADR and dubbing. Studio B is a 16-track room used mainly for voice-overs, music sessions with two or three musicians, ADR and dubbing. Studio C is a ¾-inch off-line video editing suite, and Studio D is a MIDI room.

Although corporate and industrial clients make up the bulk of the business in the main room, the 16-track room and the MIDI room allow it to gain business it might not get in the 24-track room. The 16-track room will appeal to clients who might not afford the 24-track rates, and the MIDI room draws musicians who might otherwise go to demo studios. This sort of diversification is absolutely necessary in a market such as Tulsa, Slane says.

Music production library

Another example of Universal's diversity is its music production library, Musipak. Slane started the music library as a way of getting into the nationwide market. Clients who bought library music later became studio clients, and others have ordered custom music jobs.

"A good portion of our existence is working for ourselves," he says. "The studio stays busy almost every day doing production music for the library."

Throughout the years, Slane has observed two rules that he credits with staying in business: don't cut rates and keep busy doing projects that are sure to pay off. Not cutting rates ensured that the studio's reputation would be intact, and working on solid projects was better than giving time away or doing spec work with bands.

Universal's move toward documentaries was a result of the need for product in the cable and satellite TV industries, and Slane's evolving interest in film work. Through reading trade magazines and in talking with other engineers around the country, Slane says, Universal's direction is no different than from other studios in the country.

"I spent 10 years learning how to get a great kick drum sound, which quite frankly is now useless," he says. "That '70s type of engineer is gone. Now, the question is, how creative is the guy with new technology, and how much does he know about subjects other than recording music?

"The best thing I can do is keep my knowledge up, and I can stay in business."

RE/P

Studio News

Northeast

Lavskymusic (New York) has purchased a New England Digital Synclavier featuring 96 voices, 64Mb of RAM, a Direct-To-Disk system and a 2-billion-byte optical disc system. According to NED, this is the largest system the company has ever delivered, and is possibly the largest workstation in the world.

Atlantic Studios (New York) has added Steve Bramberg as studio manager. The studio has also installed a George Massenburg Labs Moving Fader Automation system on its Neve 8078 console in Studio A. *1841 Broadway, New York, NY 10023; 212-484-6093.*

Lion and Fox Recording (Washington, DC) has named Jim Fox president of the corporation. Former president Hal Lion, who co-founded the studio with Fox, has been named chairman of the board. New equipment includes a Studer A-80 24-track recorder, which interfaces with a Cipher Digital Softtouch synchronizer system. *1905 Fairview Ave. NE, Washington, DC 20002; 202-832-7883.*

Eastern Artists Recording Studio (East Orange, NJ) has appointed Michael Van Duser as studio manager. New equipment includes a TC 2290 delay/sampler, an Alesis drum machine and a rack of Alesis Microverbs. *36 Meadow St., East Orange, NJ 07017; 201-673-5680.*

Roar Productions and Musical Services (Columbia, MD) has taken delivery of an Otari MTR-10 2-track with center-track time code. The studio has also completed construction of a MIDI room with an IBM XT compatible computer. MIDI instruments include an Akai AX80 programmable analog synthesizer, Yamaha TX-81Z and TX-7 FM tone generators, Akai S-900 sampler and Oberhheim DMX digital drum computers. *6655-H Dobbin Road, Columbia, MD 21045; 301-381-1440.*

Balance Sound Studios (Bethesda, MD) has promoted John Biehl to chief engineer and added Steve V. Johnson as studio manager. New MIDI gear includes Cakewalk sequencing software, Roland MT-32 module, Yamaha SPX-90II and Eventide 910 Harmonizer update. *4917 Cordell Ave., Bethesda, MD 20814; 301-951-3900.*

Master Sound Recording Studio (Virginia Beach, VA) has upgraded from an Amek Angela and added an Amek G2520 40-input console with Audio Kinetics Master Mix automation. An Otari 24-track machine has also been added. *5249 Challdon Drive, Virginia Beach, VA 23462; 804-499-0000.*

HBO Studio Productions (New York) has ordered a Solid State Logic SL-6000 E series console for the audio post room now under construction. The console features 48-input mainframe with 32-input/output modules and is equipped with a G series fader automation system. Other hardware includes an AMS AudioFile and an Otari MTR-90. *120A E. 23rd St., New York, NY 10010; 212-512-7842.*

The Hit Factory (New York) has retrofitted its Sony 3324 tape machine with the Apogee Electronics anti-aliasing, anti-imaging low-pass filter.

Kajem Recording (Gladwynne, PA) has opened its second studio. Equipment includes two Otari MTR-10 2-track recorders, TimeLine Lynx synchronizers, Dolby 261 SR units, Kurzweil 250 RMX digital sampling unit and a variety of outboard gear. The facility will specialize in video sweetening and jingle production. *1400 Mill Creek Road, Gladwyne, PA 19035; 215-649-3277.*

The Power Station (New York) has retrofitted its Sony 3324 tape machines with the Apogee Electronics anti-aliasing, anti-imaging low-pass filter. *441 W. 53rd St., New York, NY 10019; 212-246-2900.*

Hip Pocket Recording Studios (New York), the new name of Blank Tapes Inc., officially opened for business on Jan. 1. The facility features three studios, all equipped for film, video and commercial work. Studio A contains an SSL-6056E console with Total Recall; Otari analog and Sony digital recorders. Studio B has a modified MCI 542-C console; Studio C features a complete MIDI environment and has a Synclavier with tielines to the other two studios. Jim Doherty is studio manager,

STUDIO UPDATE

Joe Arlotta and Butch Jones are engineers, and Rich Oliver is chief technical engineer. *37 W. 20th St., New York, NY 10011; 212-255-5313.*

Horizon Recording (Pittman, NJ) has expanded its operations. New equipment includes an Otari MX-80 24-track recorder, to be used with a Neotek series II console. A new second studio features a Tascam 85-16B recorder and an M-520 console.

Music Delli (New York) has purchased a Soundtracs CP6800 40-24 console with 20 normal inputs and 20 enhanced EQ inputs.

Sheffield Audio Video Productions (Baltimore) has installed a Neve 8068 MkII console in its audio remote truck. The board was first used on a classical recording for Erato Records. *13816 Sunnybrook Road, Baltimore, MD 21131; 301-628-7260.*

Third Story Recording (Philadelphia) has completed construction on a second recording room, and plans to upgrade to 24-track by mid-1988. *5120 Walnut St., Philadelphia, PA 19139; 215-747-1200.*

Manhattan Center Studios, (New York) now features two fully equipped recording facilities. Studio 7, known for ambient acoustics, features an SSL G series 56x48 console with Total Recall and an Otari DTR-900 32-track digital tape recorder. Dimensions are 94′ x98′ with an attached stage measuring 95′ x60′, which features full lighting capacity for live stage production. Studio 8 features a TAC Scorpion 28x8x24 console and an Otari MX-80 analog 24-track tape machine. *311 W. 34th St., New York, NY 10001; 212-279-7740.*

Sigma-Alpha Entertainment Group (Philadelphia) is the result of a merger between Sigma Sound Studios and Alpha International Recording Studios. The venture will include music recording, music synthesis facilities and a music production creative department. *212 N. 12 St., Philadelphia, PA 19107.*

Southeast

Key Recording Studio (Jacksonville, FL) has expanded its outboard gear and keyboard selection. Equipment now includes a Yamaha DX-7, RX-5 and two SPX-90s, two Roland SRV-2000s, SDE-3000 and DDR-30, and assorted compressors, limiters and equalizers. *2969 Edison Ave., Jacksonville, FL 32205; 904-388-8273.*

Alpha Recording (Elizabethtown, KY) has installed a Trident series 65 console. The studio is also in the process of adding an audio-video production, post-production and duplication facility. *207 S. Mulberry, Elizabethtown, KY 42701; 502-765-7899.*

Midwest

South River Recording (Indianola, IA) has installed a Soundcraft 600 24-channel console.

North central

Pearl Sound Studios Ltd. (Canton, MI) has purchased a Neve V series 48-input console with Mastermix automation. *47360 Ford Road, Canton, MI 48187; 313-455-7606.*

Universal Recording (Chicago) has named Joe Stopka as director of sales and marketing. *46 E. Walton St., Chicago, IL 60611; 312-642-6465.*

Southern California

Summa Music Group (Los Angeles) has taken delivery of a Solid State Logic G series console with a 64-channel mainframe, 56 inputs, VU metering, G series computer and Total Recall.

Encore Studios (Burbank) has recently completed a new control room, which houses a Solid State Logic 6000 console. Darryl Caseine has joined the staff as studio manager. *721 S. Glenwood Place, Burbank, CA 91506; 818-842-8300.*

Syngram Productions (Pacific Palisades) has purchased a Soundcraft 200B SEQ console. *16169 Sunset Blvd., No. 101A, Pacific Palisades, CA 90272; 213-459-3197.*

Red Zone Studios (Burbank) has expanded into video and film work, and has purchased a Sony 5800 ¾-inch deck, an NEC high-resolution monitor and a Fostex synchronization system, which consists of a 4010 generator/reader, 4030 synchronizer and 4035 controller. *623 S. Glenwood Place, Burbank, CA 91506; 818-955-8030.*

The Enterprise (Burbank) has added a New England Digital Direct-to-Disk system. *4620 W. Magnolia Blvd., Burbank, CA 91505; 213-935-5585.*

EFX Systems (Burbank) has installed four New England Digital post-production systems. *919 N. Victory Blvd., Burbank, CA 91504; 818-843-4762.*

South Coast Recording Studio (Santa Ana) has taken delivery of a Straube baby grand piano. *1818 N. Main St., Santa Ana, CA 92706; 714-541-2397.*

Skip Taylor Recording (Los Angeles), a new studio, has installed an SSL 4000E console with Total Recall. Other equipment includes a Studer A800 MkIII 24-track recorder, Ampex ATR 102 and 104, and TAD monitors. Andrew McCarl is the studio manager. *506 N. Larchmont Blvd., Los Angeles, CA 90004; 213-467-3315.*

Ignited Productions (Hollywood), a new production and publishing company, has opened its computerized MIDI and post-production facility. Equipment includes a Neve 8058 console; Studer A800, and an extensive collection of synthesizers. Jefferson Chitouras is the studio manager. *1645 N. Vine St., Suite 614, Hollywood, CA 90028; 213-461-0734.*

Capitol Studios (Hollywood) has appointed Leslie Ann Jones as staff engineer. *1750 N. Vine St., Hollywood, CA 90026; 213-462-6252.*

Secret Sound L.A. (Woodland Hills, CA) has purchased an Otari DTR 900 32-track digital recorder and a Sony PCM 2500 R-DAT recorder. *4836 Queen Victoria Road, Woodland Hills, CA 91364; 818-999-6160.*

Northern California

Fantasy Studios (Berkeley) has installed a 28-input Soundcraft 600 console in its new MIDI keyboard studio.

Robert Berke Sound (San Francisco) has renovated Control Room A with acoustics designed by Randy Sparks of RLS Acoustics. New equipment includes a Sound Workshop 34C console with disk-based automation, Kelly Quan disk-based audio-for-video editing system and Dolby SR noise reduction. *50 Mendell, No. 11, San Francisco, CA 94124; 415-285-8800.*

Different Fur Recording (San Francisco) has opened MIDIFUR, a computer and MIDI production room. Equipment includes an E-mu Systems Emulator III and Emax; Roland D-50, PD-31 multi-trigger drum pads and PM-16 pad-to-MIDI interface; and a Macintosh computer with Digidesign Sound Design and Performer 2.2 MIDI Sequencer software. *3470 19th St., San Francisco, CA 94110; 415-864-1967.*

Independent Sound (San Francisco) has added Kathy Braun as commerial work representative. *2032 Scott St., San Francisco, CA 94115; 415-929-8085.*

Sound Recording Organization (San Francisco) has added an Otari 32-track tape machine and a CMX audio editing system in Studio 2, to complement its surround sound mixing and 35mm 6-track recorder in Studio 3. *1338 Mission St., San Francisco, CA 94103; 415-863-0400.*

Dolby Labs (San Francisco) has added a Harrison MR-4 series console in its screening room complex, to be used for audio training and demonstrations, and music recording and screening room applications.

Northwest

Sound West Studios (Tocoma, WA) has installed a Soundcraft TS012 console and a pair of UREI 809 Time Align monitors. *2321 Tacoma Ave., Tacoma, WA 98402; 206-272-4251.*

Canada

Columbia School of Radio TV & Recording Arts (Vancouver, British Columbia) has purchased a Soundtracs CM4400 32x24 console. RE/P

Send studio news, including facility openings, equipment purchases and personnel additions, to Studio Update, RE/P, Box 12901, Overland Park, KS 66212.

NEW PRODUCTS

Panasonic SV-250, -3500 R-DAT recorders

The SV-250 is a portable recorder that weighs 3.2 pounds and can be powered from a rechargeable battery pack, external dc supplies or main power. XL-type mic connectors are included, as well as peak-level metering, headphone monitoring and switched 14dB mic attenuation. Maximum recording time with the battery pack is 2.2 hours.

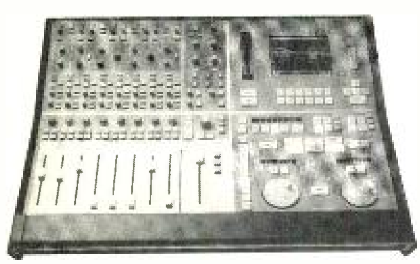
The studio version SV-3500 has record sampling frequencies of 32kHz and 48kHz, and replay sampling frequencies of 32kHz, 44.1kHz and 48kHz. Analog and IEC digital interface inputs/outputs are included.

Circle (100) on Rapid Facts Card

SSL 01 digital production center

Solid State Logic's 01 combines signal processing, storage, mixing and editing functions in a single integrated system. The system contains an 8-channel mixer, three stereo tape machines, a synchronizer and edit controller, time code reader/generator, A/D converters, a sampling rate converter and sync generator. Faders and EQs work like analog counterparts, and traditional concepts such as cut and splice are employed.

Circle (101) on Rapid Facts Card



Eneractive Group Energenius CS3200

The unit is an alternate power and electrical distribution system that backs up sound systems when the ac power is down. It can be used as an ac power distribution system. Specifically designed for audio applications, the unit features total isolation of ac common and chassis ground to eliminate ground loops. It operates from 12V storage and contains a 12V, 80A quick restoration automatic battery charging and maintenance system.

Circle (102) on Rapid Facts Card

Ampex 478 low-print mastering tape

The tape line is intended for radio broadcast customers, film and video post houses and recording studios where low-print mixdown is desired. A new high-speed backcoating process reduces edge damage, pop strands and the need for slow speed rewind. The tape is available in CCIR and NAB formats.

Circle (103) on Rapid Facts Card

Beyer Studio Group mics

The M260, M740 and M500 are part of Beyer's Studio Group of mics. The 260 is a ribbon mic designed for acoustic piano and stringed instruments, and has a hypercardioid polar pattern. The 740 is a condenser with five selectable polar patterns, enabling it to be used in a wide variety of applications. The 500 has a hypercardioid characteristic with a rising frequency response curve, and also has a built-in pop filter.

Circle (104) on Rapid Facts Card

New version of Baccus editing software

The company's TX81Z and TX802 graphic editing systems can now be used on IBM PS/2 models 25 and 30. The graphics systems run in the highest resolution available on the PS/2: 640x480. The system can also be used on the Toshiba 3100 lap-top computer and the AT&T 6300. Graphics resolution is 1280x800 on the Wyse WY-700 graphics subsystem.

Circle (105) on Rapid Facts Card

Winsted rack-mount cabinets

The cabinets feature a 30° sloping profile with 21 inches of rack space, and are designed for applications where a low profile is needed. The module can be combined with the company's System/85 modules in a variety of configurations.

Circle (106) on Rapid Facts Card

UREI 6210, 6211 amplifiers

The amplifiers are designed to convert JBL's series 4400 or any 8Ω speaker into a self-powered system. They fit on the back of the speaker enclosure and feature 40W output into 8Ω. Both have 3-pin XLR and ¼-inch phone jack input connectors wired in parallel, and active balanced inputs that accept balanced or unbalanced line level sources. Additionally, the 6211 has a switch-activated pre-amp for low impedance mic inputs and a user-selectable high-pass filter for reducing mic proximity effects and wind pops.

Circle (107) on Rapid Facts Card

Sony DAE-3000 digital audio editor

The unit replaces the DAE-1100A, and has been designed for use with the company's CD mastering recorders and DASH recorders. Editing resolution has been increased to 23μs; crossfade time variable is in the range of 1ms to 999ms, in 1ms steps, and can be preset individually for in and out points. A 16-bit digital gain fader controls the level of signal to be edited, and can be extended to 12dB. The unit can accept up to four players.

Circle (108) on Rapid Facts Card

Power Mountain Software filter design tools

PMSS Active Filter Design Tools 2.0 is an IBM PC software package for CAD filter design. Users enter the required parameters; the menu-driven format then leads you through each step to the final design. It also draws schematics with values for accurate documentation.

Circle (109) on Rapid Facts Card

Electro-Voice 7300 stereo power amplifier

The 7300 delivers 300W into 4Ω and 200W into 8Ω. In the mono bridge mode, which is activated by a mode switch on the back of the amplifier, the unit delivers 600W into 8Ω. An optional APX plug-in module provides on-board bi-amp capability. Other features include LED clip indicator and protect indicator for each channel, dented front-panel pots and balanced XLR connectors.

Circle (110) on Rapid Facts Card

Studer A727 CD player

The professional CD player is the first product from the Studer/Philips joint venture. Features include multiple cuing modes, autocue mode, self-illuminated display panel, AES/EBU outputs, balanced XLR outputs and BNC clock input and output jacks for varispeed and synchronized operation. It can be used either on a tabletop or in a rack.

Circle (111) on Rapid Facts Card

WaveFrame updates for the AudioFrame

Several additions and updates have been announced for the AudioFrame digital audio workstation. New software includes SoundProcessor, which includes a new user interface, cut and paste audio editing, stereo sampling and AudioTrigger for track replacement, and SoundStore, a sound sample storage system. Vertical Integral Time Code input synchronization has been added to the Studio Control Processor Module. Hardware additions include a user-programmable digital signal processor module, which occupies a single slot in the audio rack, and an 8-channel A/D converter module.

Circle (112) on Rapid Facts Card

Horita TG-50 time code generator

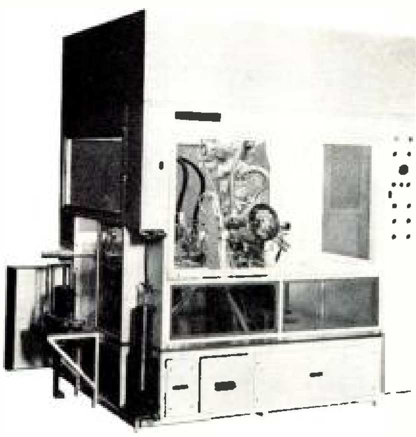
The TG-50 is a time code generator and time code window generator, allowing users to generate code and make "window dub" copies of SMPTE-based videotapes. Total manufacturing time is less then two minutes, and a disc is completed every eight seconds. The unit generates drop-frame or non-drop-frame code and can jam to the value of incoming code. It measures 4"x3"x2" and is powered by a dc power adaptor, which is included.

Circle (114) on Rapid Facts Card

Shape UDMS CD manufacturing system

Standing for Unit Disc Manufacturing System, UDMS is an integrated system that is compatible with all CD molding machines and molds. All stations are protected by class 100 HEPA filters, eliminating the need for a clean room. Total manufacturing time is less than two minutes, and a disc is completed every eight seconds. The unit measures 4'x8' and can be configured to run 3-inch CDs, CDV and CDI; future applications will include WORM and erasable optical discs.

Circle (113) on Rapid Facts Card



Carvin MX601 series rack mixers

The 6-channel mono mixers come in two models. The MX621 delivers 200W rms, while the MX641 delivers 440W rms. Both can be rack-mounted and have low-noise pre-amps. The 641 contains a 48V phantom power supply for condenser mics.

Circle (115) on Rapid Facts Card

NEW PRODUCTS

Monster Cable M1000 MkII cable

A successor to the M1000, MkII uses the company's Bandwidth Balanced technology, dual inner conductors and three separate multiple-gauge wire networks for lows, midranges and highs. According to the company, the cable reduces distortion and facilitates more accurate sound.

Circle (116) on Rapid Facts Card

QSC model 1100 stereo amplifier

Occupying one rack space, the 1100 delivers 65W per channel at 4Ω and 50W per channel at 8. Features include 5-way binding posts for speaker outputs and Octal sockets for use with other QSC products. A split power supply allows each channel to operate as an independent amplifier, and the company's Output Averaging Circuit protects the amplifier from indefinite short circuits without false triggering into reactive loads.

Circle (117) on Rapid Facts Card

Clear-Com W series wireless intercom

The system features full-duplex operation, which permits each portable unit to function like a wired intercom station. The WBS-6 base station connects directly to a standard Clear-Com intercom line. The WTR-1 portable transceivers operate on two 9V batteries for up to eight continuous hours. The system can support up to six units by adding WBS-R receiver modules to the base station.

Circle (118) on Rapid Facts Card

New interface for CompuSonics DSP 1000

The optical disk recorder/editor now has a digital audio interface that makes it compatible with the Sony 1610/30 format. The interface allows it to go from optical disk editing to CD mastering using the Sony system, without losing a generation from analog dubbing.

Circle (119) on Rapid Facts Card

Audio-Technica AT4031 unidirectional mic

Designed for a variety of studio and sound reinforcement applications, the mic has −44dBm sensitivity, 140dB SPL and a range of 30Hz to 20kHz. Features include a removable foam windscreen/pop filter and an integral high-pass filter with a recessed switched.

Circle (120) on Rapid Facts Card

NTI VOPEX-8M video port expander

The VOPEX-8M is designed for the MAC II video standard can be used for monochrome and color monitors. Up to eight monitors can be driven by a single MAC II up to 50 feet away from the computer. The unit connects directly to the MAC II video port by a 4-foot interface cable.

Circle (121) on Rapid Facts Card

Pro Co Sound PM-148 patchbay

A part of the Patchmaster series, the PM-148 provides 48 unbalanced ¼-inch phone jacks connected to 48 similar jacks on the rear of the unit. The rear panel jacks allow the use of pre-fabricated cable assemblies, permitting rapid installation. The Selecta-patch switch system allows users to determine whether a pair of jacks is full-normalled, half-normalled, paralleled or open.

Circle (122) on Rapid Facts Card

New interface for DSC-200 system

Digital Sound Corporation's DSC-200 now has a hardware and software interface option that links the audio conversion system to Sun Microsystem's Sun-3 and -4 series computer workstations/microcomputers. The interface also incorporates a SCSI interface, which will allow the unit to be compatible with other workstations on the market.

Circle (123) on Rapid Facts Card

Stewart Electronics PM-6 power supply

The PM-6 is a 6-channel, 48V phantom power supply that delivers 10mA of 48V dc to mics, direct boxes or other equipment requiring phantom power. The unit features short-circuit protection for each channel, isolated outputs, individual regulation of each channel, and low noise and crosstalk. The chassis design allows front or rear rack mounting.

Circle (124) on Rapid Facts Card

DigiTech DSP 32 Studioverb

The rack-mount unit contains 30 different reverb effects, including large room environments with long decay times, small rooms, reverse reverb effects and gated reverb. The unit uses 16-bit A/D/A linear PCM encoding and the company's DSP256 VLSI chip for wide dynamic range. Inputs and outputs are stereo, and an effect defeat switch jack is provided.

Circle (125) on Rapid Facts Card

Sphere Audio Products 1604A mixer

The 16-input, 4-output mixer is available in mono or stereo versions. Input modules have transformerless inputs, with signal present and overload indicators. Parametric EQ may be alternated with the input modules. The output module LEDs show signal present, normal level and overload on each of the four outputs and cue.

Circle (126) on Rapid Facts Card

Jensen Tools Pelican cases

The cases are made of cycolac resin and contain pre-scored ½-inch foam cubes. Users form the shapes they need by taking out appropriate cubes. Replaceable neoprene O-ring seals make the cases airtight, which protect equipment from dust and moisture and enable the cases to keep up to 50 pounds of equipment afloat. A pressure purge valve releases the seal for altitude changes.

Circle (127) on Rapid Facts Card

360 Systems Professional MIDI Bass

Professional MIDI Bass contains samples of bass instruments stored in resident EPROM chips. Extra sockets are provided, which allow users to customize their onboard libraries by inserting additional sound chips from the company. Features include two programmable zones with separate main and accent sounds, global and individual zone transpose, variable volume, decay, release, filter and velocity crossover.

Circle (128) on Rapid Facts Card

Amek/TAC automation system

A collaboration between Amek/TAC and Steinberg, a West German software company, this automation system will be available on Amek/TAC consoles, and as a retrofit to other manufacturers using the company's interface and hardware. The system runs on the Atari ST, and up to 128 faders and 15 switches per channel may be automated. The system will also be available for the Apple Macintosh.

Circle (129) on Rapid Facts Card

Pulizzi Engineering power distribution system

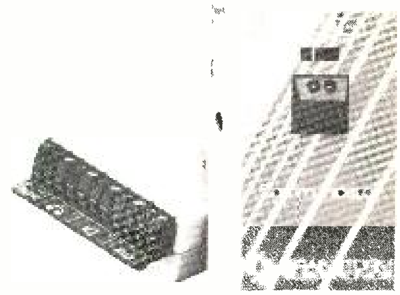
The TPC 115-8-A and -C are 120V, 12A and/or 24A units that contain a filter for protection against EMI and RFI. Transient voltage spike and surge protection is provided line-to-line and line-to-ground. A master 20A or dual 20A circuit breaker provides overload protection and acts as the master on/off switch.

Circle (130) on Rapid Facts Card

Brainstorm Electronics TB-4 system update

The TB-4 Communicator infrared talkback system has been updated with an EMI/RFI shield and does not require direct line-of-sight between the transmitter and the receiver. Owners of previous systems can trade in their units for the upgraded version.

Circle (132) on Rapid Facts Card



NEW PRODUCTS

HME DN100 antenna distribution system

The DN100 allows users to operate four of HME's RX520 switching diversity receivers in a rack with two antennas. The system consists of the DN100 antenna distribution unit, an ac adapter and locking clip, and eight TG58 BNC-to-BNC coaxial cables. A specially designed circuit guarantees that there is no signal loss because of antenna splitting, according to the company.

Circle (134) on Rapid Facts Card

Audio Services Corporation Sound Booth

The unit is a lightweight, portable enclosure for location recording. Wild lines and narration can be recorded without having talent in a studio. The booth consists of a tubular plastic frame and acoustic foam panels that roll up and are stored in their own case. The unit can be assembled in a few minutes.

Circle (133) on Rapid Facts Card

Additions to CMX CASS-1 system

Four features have been added to the sound sweetening system. CMX Gismo allows for search, job and mark in/out edit points from a hand-held device. Track-Select allows editors to arm up to 48 channels on a multi-track recorder through software. Intelligent Interface is a synchronizer that addresses machine types through the system. Improvements in the Keystroke Memory File include nine hot keys along with KSM editing.

Circle (135) on Rapid Facts Card

Sonosax SX-PR mixer

The portable unit is designed for ENG/EFP applications, and can be used for digital recording. It is available in two, four or six inputs, and weighs three, four and five pounds. An MS matrix and an HF transmitter or receiver will be available soon, according to the company.

Circle (136) on Rapid Facts Card

FM Acoustics 236/4 crossover

The linear-phase electronic crossover is an updated version of the 236, designed for 3- and 4-way mono applications. The 3-way unit has a low/mid crossover of 80Hz to 1kHz, and a mid/high crossover of 1kHz to 12.5kHz. For the 4-way unit, low/mid crossover is 80Hz to 1kHz, low-mid/high-mid is 250Hz to 3kHz, and high-mid/high is 1kHz to 12.5kHz. The 3-way models can be upgraded to 4-way.

Circle (137) on Rapid Facts Card

Prosonus Code Disc

Code Disc allows you to use a CD player as a time code generator. Audio cues on one track correspond to the CD player's digital display, with guaranteed dropout-free time code on the other. The disc is available in NTSC and EBU formats.

Circle (138) on Rapid Facts Card

Yamaha DEQ7 digital EQ

The dual-channel digital equalizer/filter system uses 44.1kHz sampling and has a 20kHz bandwidth. It contains 30 different types of EQs, tone controls and filters and can store up to 60 custom curves. Simultaneous or independent left and right channel program is available in either the graphic or parametric EQ programs. Up to 734ms of delay may be programmed per channel, and can be displayed in seconds, feet or meters.

Circle (139) on Rapid Facts Card

Electron Processing Interference Eliminator

Interference Eliminator is a filter that is designed to eliminate RFI from audio equipment. It provides more than 38dB of isolation from signals from 0.5MHz to 500MHz, and has no effect on signals below 30kHz. The filter is equipped with 3-pin XLR connectors, allowing it to be used in mic and line level applications. Passive components are used, eliminating the need for external power.

Circle (140) on Rapid Facts Card

Howard W. Sams "The Microphone Manual"

Written by David Miles Huber, "The Microphone Manual: Design and Application" introduces and explains microphone design, characteristics and theory, and is designed for intermediate to advanced users. Topics include basic operation, fundamentals of single mic and stereo techniques, speech and music reinforcement, and techniques for music and video/film production.

Circle (141) on Rapid Facts Card RE/P

CLASSIFIED

Advertising rates in Classified Section are:
$1.00 per word per insertion. Initials and abbreviations count as full words. "Blind" ads $25.00 additional. Minimum classified charge $20.00. Classified is also available at $122 per inch. Order must be accompanied by payment to ensure publication.

Classified columns are not open to advertising of any products regularly produced by manufacturers unless used and no longer owned by the manufacturer or distributor.

Classified Advertising should be sent to RE/P, Advertising Department, 9221 Quivira Rd., Shawnee Mission, KS 66215.

EMPLOYMENT

MIXING ENGINEER. Creative, personable engineer for new automated sweetening studio. Must be experienced with all phases of audio for video and film. Send resume and credits in confidence to Ruth Scovill, VP/General Manager, One Pass Film and Video, One China Basin Building, San Francisco, CA 94107. 5-88-1t

ATTENTION: STUDIO OPERATORS. Prolific award-winning composer producer w/L.A. midi studio SEEKS full-time PRO ENGINEER w/resources to co-manage/produce. Equal partners op/expansion a must. Call Bluefield Music Design, 213-463-7664/write Box 726, L.A., 90028. 5-88-1t

MISCELLANEOUS

WANTED: Macintosh, Marantz, Audio Research, Quad, Dynaco, Western Electric, Westrex, Langevin, equipments. Early spearkers by Tannoy, Altec, JBL, EV, Jensen, W.E. Audio City, P.O. Box 832, Monterey Park, Ca. 91754 Tel: 818/576-2642. EOM

FOR SALE

SOUNDCRAFT TS-24, 32 x 24, 3 years old, mint condition, $35,000.00. URSA Major 8 x 32 w/update, $1,500.00. Recordex stereo Pro-H, Hi-Spd. Cassette duplicator. $1,500.00. Inquiries call: (901) 525-5500 or (901) 363-3856, ask for John Fleskes. 5-88-1t

NEW 19" ADC tt patchbay panels, balanced/normalled, 144 points each, $125 each or 3 for $300. Roger Cordell, 10818 Hortense, North Hollywood, CA, 91602. (818) 763-7915 5-88-2t

DOLBY A-TYPE, MH SERIES, 3 RACKS of 16 channels, will sell in any configuration. New Mitsubishi Westar 36 x 32 automated console, free factory on site commissioning to buyer. MCI JH-542 42 x 32 automated console. Mitsubishi X-80 2 track digital recorder, low hours. All used equipment professionally maintained and in service at Ardent Studios, 2000 Madison Avenue, Memphis, TN: Contact Pat Scholes at 901/725-0855. 5-88-1t

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A bold statement. But no exaggeration.

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With the A807, Studer defines a new price point. If you don't believe us, check out the competition . . . It won't take you long, because there isn't any.

Broadcast, post production, or studio—in whatever audio environment you operate—make it your business to check out the Studer A807. Start with price . . .

High Tech, Low Price

Anyone who thinks "Studer" always means "expensive," please take note. With the A807, your dollars actually buy *more* now than they did five years ago—more technology to help you do your job better and faster.

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The Studer A807 is a fast, full-featured machine for making quick work of your production tasks.

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Studer Audio Performance and Reliability

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STUDER REVOX

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