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Recording

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Second Annual Salary Survey

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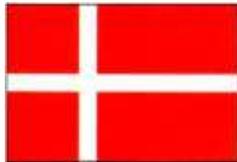
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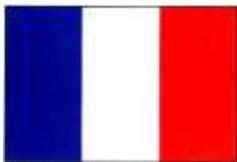
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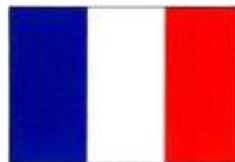
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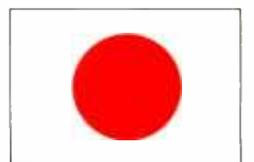
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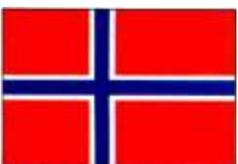
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CONTENTS

Microphone Applications

Microphone Calibration Techniques: An Overview, Part I

An internationally accepted calibration procedure and full documentation of studio microphones would allow direct product/performance comparisons.
By *Adrian Weidmann* 24 ▼



Mic Pre-amp Evaluation Methodology

Some thoughts on stand-alone mic pre-amp evaluations that can be applied generally to any piece of equipment in the signal path.
By *John Hardy* 30

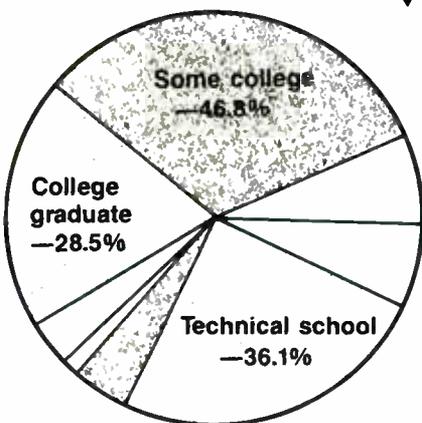
Phantom-Powering Precautions

A drop in phantom supply voltages may be what's causing distortion or level loss in your condenser mics.
By *Bruce Bartlett* 36

Other Features

Second Annual Salary Survey

Although benefits were down, salaries increased for most job categories in the pro audio industry. Get the details in *RE/P's* exclusive reader survey.
By *Dan Torchia* 42 ▼



Recording the "Tunnel of Love" Tour

A portable touring/recording system solved both the immediate needs of touring and longer-term studio recording needs once the tour was over.
By *the RE/P staff* 57

MIDI Control of Effects: One Designer's Viewpoint

Full MIDI implementation can be a tremendous help in streamlining effects creation if total flexibility and independent control over all parameters is provided.
By *Paul D. Lehrman*
with *Jeffrey Stanton* 62

Equipment Leasing Made Simple

Leasing is 100% financing, with the lessor buying the equipment from a vendor for cash and leasing it for a specified period of time to a lessee.
By *Tom Roche* 67



On the Cover

This month's cover features Audio Technica's new 40 series microphones, from left to right: AT4049 omni, AT4053 hypercardioid, AT4051 cardioid.

Volume 19, No. 11

Active Balanced Inputs and Outputs

Don't become unbalanced over hum and noise.
By *Richard Cabot* 69

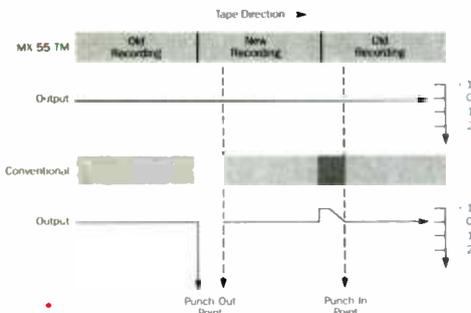
Departments

Editorial	4
"Microphone Technique"	
Letters	6
More comments on phase shift	
News	10
Goldmund acquires Stellavox	
Managing MIDI	14
A MIDI user's wish list	
SPARS On-Line	16-18
"Making Good Money in the Audio Business"	
By <i>Chris Stone</i>	
Understanding Computers	20
"Communicating with the Outside World"	
Studio Update	
Talkback	74
"Moving with Murphy"	
By <i>Todd Lockwood</i>	
Studio News	75-78
The Cutting Edge	79
Flying Faders automation system	
New Products	80-86
Equipment, literature, and hardware and software updates	
Classified	87
Advertisers' Index	88

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Finally! A two-track that knows the difference between radio and TV: Introducing Otari's new MX-55TM

We got respect, but we also got complaints. Post production people said, "Give us a compact, two-channel recorder with center track



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SMPTE time-code and all the goodies the big machines have." Or, "Why is it that when an audio machine really delivers what I need, it's always in my way and it costs me an arm and a leg?"

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Now you've got it. A compact, SMPTE time-code machine designed just for you. And we can prove it. (We knew you'd ask us to.)

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Which reminds us: Otari's standard interface connector is provided to make synchronizing fast and easy.

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And, its tape time display will also read out tape speed in ips or % difference from selected speed—very useful with the machine's $\pm 20\%$ vari-speed.



The MX-55TM's built-in three cue-point locator with LED display.

And what about HX-Pro* to give you increased headroom at high frequencies?

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Microphone Technique Part 1

There are a lot of things that make a good audio engineer, and with each new generation of technology, that list grows. But at the heart of the list, the very foundation, are two talents that separate the good from the great: a concise mental picture of the sound to be recorded and the knowledge and understanding of good microphone technique.

If you've been reading my editorials during the past year, you know I place great value in the mental processes that *should* take place before a microphone is ever plugged in. [See April editorial "Think Audio".]

Good micing technique is a process that includes gathering basic information about the project. This permits the engineer to consider and allow for all parameters and influences, including but not limited to: (1) the sound source(s), (2) the size and sound of the acoustic space, (3) the style of music, (4) the arrangement, (5) mic placement, (6) the available mics, (7) the number of mics being used simultaneously, (8) the abilities of the musician or vocalist, (9) the number of available channels and/or tracks, (10) and not least, the allotted time for the session. Armed with this information, a professional can now reach for the best available mic.

Unfortunately, all of this information is not always available before the session starts. Therefore, experience must also be a vital factor. Let's look at why these considerations are so important.

The sound source

On the surface, this is perhaps the simplest and most obvious parameter to deal with. Most intermediate-level recording engineers seem capable of reaching for those omni-popular mics (that just about every other engineer's been using for years) to record a common rhythm section. I'm willing to bet that most of you learned these basics by watching, or listening to what others *said* you should use on these various instruments. Don't get me wrong, this is often a good way to learn, but sometimes there are several mics in the locker that will do an excellent job of capturing the source in uniquely different (and useful) ways.

Somewhere along the way you have to judge the differences for yourself. Maybe,

because of the way you hear or work, other non-standard mics will be more useful or allow a broader range of creativity. Some careful experimentation can be very fruitful.

What do you reach for when the producer brings in a Diggerydoo? Experience comes into play when you are presented with an unfamiliar sound source. Here you can compare the similarities of the unknown source to those of an instrument that is more familiar. Careful though, because we've only considered the fundamental *sound* of the instrument, there is still the mic placement as well as many other variables to consider.

The size and sound of the acoustic space

Understanding the polar response characteristics of your available mics can be very helpful when trying to capture (or reject) the various anomalies of an acoustic environment. For a great-sounding "live" room you might want to capture a good portion of the room sound along with the source sound. This can be done several ways: using an omnidirectional pickup pattern, a Figure 8 pattern, and even cardioid patterns. If you can take the time to find the right placement, each of these patterns will give you a slightly different sound, perspective and source-to-room ratio.

The same holds true for drier rooms. Selection of the right pattern and position can "open up" the sound of an instrument (if you can back off a few feet) or you can "tighten down" on the sound by coming in on the source with a super- or hypercardioid pattern. This may be desirable in sampling work, for example, where any room-sound would detract from the raw sample.

Music style

This is one of the first things you need to know before doing your setup. There can be vast differences between mic selection and placement when doing rock 'n' roll vs. jingles, rap, orchestral, jazz or other types of sessions. Even within a certain genre, there may be vast differences. For example, capturing a traditional jazz-bop drum sound as opposed to a contemporary jazz-pop sound can be very different. This is a good example of why it's important to experience a wide variety of musical styles, allowing you to have the *right* sound to draw from in your memory.

The arrangement

"But wait, I thought that was the arranger's job". Well, it is, but you'd be amazed at how much clearer your tracks sound if you are recording a good arrangement and you have a some understanding of the overall instrumentation and voicings. Having worked in the jingle world during most of my engineering career, it was not uncommon to work with the same small group of musicians on many different projects—but with different arrangers. Same players, same studio, same engineer, same mics; dramatically different results. There is a critical difference.

Knowing the instrumentation gives you some idea of how many tracks you'll use and the number of stacks or layers you'll be recording. This is useful when calculating the amount of room-sound to capture with each track. Understanding the voicings will help when balancing the harmonic content of the various sections (such as in a large string section). Are the violins playing in unison or two-part harmony? What is the harmony; thirds, fifths, tenths? Are the basses and cellos playing in octaves? Knowing these answers will make blending the overall sound of the section much easier.

This information is also helpful for mic placement and pattern selection as it relates to the available quantity and type of mics. For example, sometimes it's just as effective to record a string section (regardless of size) using only two mics, while at other times one mic for every two chairs is best. I've often used both these techniques simultaneously, when tracking smaller sections in dry rooms.

Next month we'll look at mic placement, the available mics, the abilities of the musician or vocalist, the number of channels/tracks, and time.

Michael Fay
Editor

Special note:

All of us at *RE/P* would like to welcome Laurel Cash back as a regular contributor. As many of you may know, she was part of the *RE/P* team under its previous ownership.

Laurel will provide late-breaking news on hot/new products in a new column called "The Cutting Edge," which has been developed to expedite information on industry developments and technological breakthroughs that affect *your* immediate operational and/or purchasing decisions. We know you will find this information timely and helpful.

RE/P

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LETTERS

Automation clarification

From: Eli Slawson, technical support, JL Cooper Electronics, West Los Angeles.

This is a (somewhat) late reply to the March article by Erika Lopez "Automation: The Purchase Decision," which contained a few factual errors about our product, MidiMation (also called "the MAGI System"). The article contained a few bizarre misconceptions.

For starters, the article stated that MidiMation is a "snapshot" automation system. It is not. MidiMation is a dynamic (that is, continuously variable) automation system. All fader and mute events are remembered with respect to incoming SMPTE time code.

Also, I have never heard the term "MCA" before, and I am curious as to who coined it. It certainly isn't some industry-standard term. Although MidiMation does employ MIDI, it is only used as a communication protocol between two pieces of hardware. Using the term "MCA-based automation" seems to vaguely imply that some sort of external MIDI device is required to make the system operate at full efficiency. While this is true for *both* Twister and Iota, it is not true for MAGI. Those systems require the additional purchase of a MIDI sequencer, but MAGI works all by itself.

Also, the distinction between computer-based and non-computer-based systems is unclear. MAGI does not in any way require a computer, but an engineer may elect to hook up an optional graphics interface. This allows a Mac or ST to display fader levels, SMPTE time and other useful information. This computer in no way affects synchronization, which leads me to my final point.

MAGI runs directly off of SMPTE; no other low-cost automation package does this. A few computer-based automation systems lock to MIDI Time Code. Several lock to MIDI Sync with MIDI Song Position Pointer. In either case, *all* low-cost automation systems (except MAGI) require the use of some kind of external synchronizer.

One last concern: What does she mean when the article says that MTC-based systems are "slower" than SMPTE-based systems? MTC is just a way of sending SMPTE down a MIDI cable. A frame is a frame, whether it is buried in an 80-bit SMPTE message, or traveling down a MIDI cable at 31.25 Kbaud. Are "MIDI seconds"

somehow slower than "SMPTE seconds"? Where did she get this? Sorry to hammer about this, but I get a tad vexed because there is so much "buzzword garbage" flying around about MTC.

Phase shift, cont.

From: J.L. Guenther, audio engineer, University of Wisconsin-Stout teleproduction center, Menomonie, WI.

I would like to respond to the letters in the August issue regarding Terry Pennington's article on phase shifts.

Pennington made a comment stating that "phase shift, in and of itself, lends nothing positive or negative to the quality of sound." However, a phase shift cannot be considered in and of itself. The result or effect of a phase shift (in reality rather than on paper) is a change in amplitude, thus a deviation from the original signal.

It is true that phase shifts abound in the real world. They are the timbre, the color of a sound. However, I have always felt it to be the goal of the science and art of audio engineering to be able to store and reproduce a sound as close to the original waveform as possible. Phase shifts in the system are a deviation from that goal, therefore a concern.

From: Kristan D. Ellis, independent engineer, Northridge, CA.

In response to Terry Pennington's reply in the August issue, I don't think that Mr. Pennington has caught on yet. I wish to propose two exaggerated examples that will, I think, explain the basic underlying controversy in all this.

Situation 1: I have a synthesizer that generates pure sine wave notes. I play the notes C, D, E, F and G in that order exactly one second apart from each other. This signal is fed into a power amplifier that has gross frequency *independent* delay only. Three seconds later, the exact same tones in the exact same time relationship to each other is present at the output of the amplifier. Net result: no audible degradation to the actual signal or pattern, only a three-second delay *for all frequencies*.

Situation 2: I have the same above-mentioned synthesizer and send the exact same signal (C, D, E, F and G, one second apart) into an amplifier that has gross

frequency *dependent* delay. If C and D arrive at the output of the amplifier at the same time, F arrives before E and G arrives five seconds afterward, then all the frequencies have *not* been delayed equally and we have successfully destroyed any resemblance to the original signal.

This is an exaggerated example of how shifts in phase that are frequency dependent create an audibly distorted signal, whether the delay is in seconds or milliseconds. We don't have to imagine what effect bad phase response makes on a complex waveform that is rich in harmonics and overtones—we've all heard the harsh midrange smear. When the harmonics are scattered about in time different from the original signal, you no longer have the original signal. Mr. Pennington proposes that this very audible effect is just "faith." I say that "faith comes by hearing. . ."

From: Dennis Ciapura, president, Teknimax Telecommunications; vice president, Noble Broadcast Group.

As one who has spent nearly two decades striving for the ultimate in audio reproduction within the band-limited broadcast environment, I must say that my experience strongly supports the position taken by Terry Pennington in his now-infamous *RE/P* article. Broadcast engineers are critically concerned about the fidelity losses resulting from the myriad of phase shifts that are part and parcel of the 15kHz low-pass filtering necessary to prevent aliasing in the stereo generation process, as well as the 30Hz high-pass function commonly employed to prevent subaudible components from destabilizing the FM exciter carrier frequency.

Experiments personally conducted over the last 10 years include blind A/B comparisons with everything from direct-to-disk to studio master tapes and CDs monitored through cascaded seventh-order 15kHz low-pass filters using electrostatic headphones. Despite the fact that the experimental bias was always toward finding and documenting audible artifacts to rationalize conversion to digital anti-aliasing filtering and more sophisticated low-frequency protection, the results have always strongly correlated with work performed by Lipshitz, Pocock and Vandercooy, as well as Shanefield and Moir.

In a nutshell, despite the best efforts of a number of experimenters working in-



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LETTERS

dependently over a period of years hoping to illuminate the real facts regarding the ear/brain system's sensitivity to phase distortion, the experimental results continue to demonstrate that while certain synthetic test signals can indeed trigger phase audibility in the midrange, it is extremely difficult to hear with musical inputs through wideband headphones, and impossible to hear over loudspeakers in a normal echoic environment. This holds true until the non-linear delay is considerably in excess of the long-standing CCIF standards.

Shanefield has even reported experiments in which live vs. recorded music was compared through both phase-coherent and phase-staggered speaker systems to determine whether even the really gross phase distortions of loudspeakers (compared to electronics) would be audible with the most challenging musical input. Again, the results were negative. Shanefield found that the test subjects did not perceive the phase-coherent reproduction as sounding any more like the live input than the phase-staggered reproducers. Moir had similar experiences with amplifiers back in the '50s, when he applied wideband design techniques developed for radar pulse transmission to audio amplifiers in the hope of achieving improved transparency. He was more than a little surprised to find that nobody could hear the difference.

I suggest the following experiment to anyone interested in exploring the audibility of phase shifts introduced by anti-aliasing filters. All that is required is a sine wave generator, a square wave generator, a couple of 15kHz low-pass filters (Torotel makes some that are easy to work with) and a monitoring system that includes electrostatic headphones in addition to the most magnificent loudspeakers available. Start by comparing the sound of the sine and square waves up to about 5kHz. The square wave will, of course, have the characteristic edge, but if the experiment is continued beyond 5kHz, the reviewer will find that by 7kHz or 8kHz the sine wave and the square wave will sound exactly alike, as long as the levels are very carefully matched, even through electrostatic headphones with 30kHz bandwidth. This is a staggering experience for audiophiles and professionals alike trying it for the first time.

Next, send the square wave signal through two cascaded low-pass filters and compare the input to the output through

the electrostatic headphones. Once again, if the *audible* levels are very carefully matched, the bizarre anomalies that an oscilloscope will display at the filter output will be found to be inaudible.

The point of all of the above is not to say that phase distortion should be completely ignored, as it is always inaudible. Such a generalization would be absurd, and I am sure that it is not what Terry Pennington intended with his article. It is interesting to note that back in 1980, when Lipshitz presented the landmark paper on the audibility of midrange phase distortion to the New York AES, his closing statement expressed the hope that the results would not be taken out of context. Lipshitz clearly recognized the danger of advocates of one position or another hearing or reading only that which seemed to support their position.

Nevertheless, the paper drew comments from Shanefield, which were published in the June 1983 issue of the *AES Journal*, and Moir in the December 1983 issue of the journal. Then in the October 1985 issue, H.R.E. van Maanen commented on the comments of both Shanefield and Moir, who then commented on the comments of van Maanen! Aside from the ubiquitous notion of "digital sound," probably no other audio topic has been more controversial than the audibility of phase distortion.

As Terry Pennington quite correctly pointed out in his response to comments in the August "Letters," we must separate religion from science. The overwhelming body of evidence suggests that of all of the things we need to worry about in audio reproduction, phase distortion is probably one of the least significant. However, in the absence of blind A/B testing, it is inevitable that people will hear the effects that they have come to expect, either from industry rhetoric, or inappropriate application of the underlying science. How often do we hear people say that a certain superior aspect of audio performance can only be heard in extended listening tests and not in A/B or A/B/X comparisons? At this point, we are dealing with matters of faith and audio culture, and no amount of logical argument will suffice to resolve the controversy.

The brutal truth is that the psychogenic effects are every bit as audible as real ones! In short, if you think you will hear an improvement, you will probably hear it. Conversely, if you suspect that there will be a degradation, you will probably hear

that, too. If we invest in a new console that is known to have superior transient response, we really do "hear" the improvement after the new equipment is placed in service, whether the difference is really audible or not. After the novelty wears off, we sometimes wish that we had opted for a different unit with mechanical features offering more operational convenience, and I suspect that far less replacement audio equipment would be sold if it was practical to set up valid A/B testing.

In the final analysis, it is reasonable to conclude that audio science and audio religion will continue to enjoy a stimulating if uneasy coexistence. More important than the controversy itself is the role of professional journals like *RE/P* to provide an ongoing forum for discussions of this type.

Showpower clarification

From: Lori L. Perkins, production manager, Showpower, Compton, CA.

I am writing in regard to the June article "Computers on the Road." This article contained the following sentence: "To deal with this problem, the tour carries its own power generators with a UPS (Uninterruptable Power Supply) backup called 'Show Power.'" This statement is not entirely accurate. Showpower Inc. is a company, not UPS power.

Showpower provides power distribution and production services for the entertainment industry and is contracted by Michael Jackson to provide power services for his world tour. These services include the generators, cabling, distribution, transformers, line conditioners, the UPS system and three full-time electricians.

The Synclavier backup system was actually designed and implemented by John Camion, vice president of Showpower and crew chief on the Jackson tour. This particular application of the UPS evolved from a need to power the Synclavier in hotel rooms for rehearsals.

RE/P

Send letters to RE/P, 8885 Rio San Diego Drive, Suite 107, San Diego, CA 92108. Letters may be edited for length and clarity.

■

G R E A T C O M P A N Y



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Stellavox purchased, reorganized; convertible A/D machine announced

Stellavox, the Swiss tape recorder manufacturer, has been purchased by Goldmund, a Swiss manufacturer of high-end consumer audio components. The company has been reorganized and given a new name, Digital Audio Technologies.

A number of new products have been announced in conjunction with the acquisition, including synchronizing devices for film and television and the simplification of the portable recorders under the name of SP9.

Also announced was a program to upgrade the TD9 to a convertible analog/digital machine. New machines will be able to be converted from analog to digital recording by exchanging pluggable elements, including the headblock, tensioners and electronics. The digital elements will be available for different standards, and existing owners will be able to upgrade their machines.

Gentner acquires Texar

Gentner Electronics has acquired Texar, a manufacturer of audio processing equipment for broadcast facilities, in a cash transaction. Texar's Monroeville, PA, facility has been closed and all operations have been moved to Gentner's Salt Lake City facility.

President Russell Gentner said that the product line will continue to carry the Texar name, and several new products will be introduced in the near future.

Ambisonic operas broadcast

Four operas recorded digitally in ambisonic surround sound were broadcast in October by National Public Radio stations.

Recorded by KWMU-FM, the St. Louis public radio station that regularly airs ambisonic programs, the operas were distributed as part of the "World of Opera" series.

Ambisonic recording allows listeners with ambisonic decoders can hear the program with front/back information, in addition to the regular left/right stereo information. Listeners without decoders hear the material in normal stereo or mono.

The operas were recorded by Barry Hufker, with Mary Edwards and John Miller. Equipment used included the AMS/Calrec SoundField microphone and Series M mixing console; mics by Shure, Schoeps and Neumann; and Valley People

mic pre-amps. Additional ambisonic equipment was built by Audio & Design, including a UHJ transcoder, which converts an ambisonic signal into two, stereo/mono-compatible channels, and a pan-rotate unit, which allows a sound to be positioned at any point around the listener.

KWMU premiered the first ambisonic broadcast in May 1984.

Fostex equipment stolen

Fostex reports that equipment en route from a trade show was stolen from a Ryder van parked at 3760 Caheunga West on the evening of Oct. 2. All of the equipment was contained in individual Bobadilla gray ATA-style flight cases with the orange Fostex logo on the sides.

The equipment and their serial numbers are:

- 1 Fostex 160 recorder, #617611.
- 2 Fostex 260s, #0601 and #2584 (numbers are last four in series).
- 2 Fostex X-30s, #591696 (look out for #602861, #597843, #608015, #602875).
- 1 Fostex model 80 recorder, #0205009.
- 1 Fostex 4030 synchronizer, #1201 or #0250 (last four numbers).
- 1 Fostex 4035 synchronizer controller, #0119 (last four numbers).
- 2 Fostex 460 recorders, #0200153, #0002 (last four numbers).
- 1 Fostex 1840 mixing console, #004 (last three numbers).
- 1 pair of Fostex RM900 monitors, #805039 and #805040.
- 1 pair of Fostex 6302B monitors, #6134, 6135 (last four numbers).
- 1 Fostex M20 mic, #0200011.
- 1 Fostex M55 mic, no serial number available.
- 1 pair of Fostex T-20 headphones.
- 1 Sony VO 5800 video recorder, no serial number available, but has a special Fostex factory modification, which is a 20-pin connector labeled Shinji 8540 mod.
- 1 Fostex SPA 11 powered speaker with mic stand adapter.
- 1 Sony KV XBR 25-inch video monitor.

Anyone with information concerning the equipment should call Steve Teipe at Fostex at 213-921-1112. A reward is available for information that leads to the recovery of the stolen equipment and/or the arrest and conviction of any suspects.

UCLA plans winter engineering courses

The UCLA Extension Department of the Arts has scheduled four recording engi-

neering classes this winter.

"Recording Engineering Practice I" will meet on Wednesdays from Jan. 4 through March 22, and will be taught by Van Webster. The workshop will deal with recording studio procedure, giving each student experience as a mixer, recordist and setup person. The workshop will meet at Webster's studio, Digital Sound Recording. The fee is \$895.

"From Fixer to Mixer: Studio Operation and Maintenance" will meet on Thursdays from Jan. 2 through March 23, and will cover first-level servicing, alignment, diagnosis and repair of audio recording and related equipment. Duke Gee, a service technician with Fostex, will be the instructor. The course will meet at UCLA; the fee is \$450.

"Audio Signal Processors: Effects Devices—A Workshop" is on Mondays from Jan. 9 through March 20. The workshop will provide a detailed examination of outboard equipment, including lectures, discussion and hands-on exercises. Independent engineer Michael Braunstein will be the instructor. The workshop will meet at UCLA and the fee is \$295.

"The Merging Technologies of Audio and Video: A Production Workshop" will meet on Thursdays from Jan. 29 through March 16, and will deal with the new tools available to audio producers, engineers and musicians, and visual producers, editors and technicians. Topics covered include audio-for-visual production planning, film scoring in the computer age, synchronizing audio/audio, audio/video and MIDI/SMPTE, digital audio recording on videotape, music and sound effects for visual post-production and new techniques for stereo television. Lectures will be at UCLA, and production sessions will be at Metavision studios. The instructor will be Theo Mayer, president of Metavision. The fee is \$295.

For more information, contact UCLA at Box 24901, Los Angeles, CA 90024; 213-825-9064.

MacMusicFest 2.0 announced

The second annual MacMusicFest has been expanded to two days and is scheduled for Dec. 3-4 at the Paramount lot in Hollywood.

Hosted by Filmsonix, the Record Plant affiliate, and Apple Computer, the festival will include the latest in Apple systems, music software, synthesizers, and film and television products. Also included will be

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NEWS

a series of lectures and workshops, and hands-on demonstrations of MacMusic-Room, said to be the world's first desktop music publishing studio for hire.

Announced manufacturers and distributors that will attend include Apple, Coda, Passport, Graphic Notes, Future Lab, Optical Media, Max 3 and Audio Inter-visual Design.

Advance admission is \$25 for both days, \$30 the day of the show. Show hours are 11 a.m. to 6 p.m. For more information, call 213-653-0240.

News notes

Klark-Teknik has been named the U.S. distributor for Milab.

Audio Research and Technology, an Australian-based manufacturer also known as ART Australia, has changed its name to ARX Systems. The company made the change to avoid confusion with the American company Applied Research and Technology.

Apogee Sound has added seven distributors to its international network: Entertainment Services Australia, Australia; Titan Audio, Belgium; Paul Farrah Sound, England; Orthophono Professionale Audio, Italy; Focus Showequipment BV, Netherlands; Sony Espana, Spain; and Audio Rents B-AG, Switzerland.

Sunkyong Magnetic has raised prices on its professional audiotape by approximately 6%. The increase was effective on Sept. 1.

JBL served as the official sound system for this summer's Aspen Music Festival. John Eargle, a JBL consultant and sound engineer, was a guest lecturer on audio recording during the festival.

Fane Acoustics' B Series and ASS Series glass fiber horns and flares, previously distributed by ASS and Adam Hall, are now distributed exclusively by Fane.

Ashly Audio has presented Paul Ackel of Triad Marketing with its annual outstanding sales award. Honorable mentions were given to A/V Marketing, GMI, Zinskind Associates and Dick Bellew Sales.

Precision Monolithics has acquired SSM Audio Products, formerly known as Solid State Micro Technology for Music.

Media & Marketing has moved to 11288 Ventura Blvd., Suite 462, Studio City, CA 91604; 818-753-9510.

Effective Sept. 12, **Solid State Logic** has new phone numbers in the United Kingdom. Voice phone is 0865 842300; fax is 0865 842118; and telex is 837400 SSL OXG.

Fane Acoustics has moved to expanded warehousing facilities at 872 Thomas Drive, Bensenville, IL 60106; 312-766-6696.

Ampex audiotape and videotape was used by NBC for its coverage of the Summer Olympics in Seoul, South Korea.

Shure Brothers reports that its microphones were used at the podiums at the Democratic and Republican national conventions.

McKenzie Acoustics has moved to 286 Bradford Road, Batley, West Yorkshire WF17 5PW England; 0924 477102.

Agfa-Gevaert has launched the Agfa Pancake Award, which will be presented to duplicators whose facilities have achieved the highest production volumes with Agfa products, or have advanced duplication technology. The award will be presented to three audio duplicators and three video duplicators.

Trevor Cash has formed Trevor Cash International, a marketing consultant company for the pro audio industry. The company can be reached at 102-104 Golders Green Road, London NW11 8HB England; 01-455-00534.

Continued on page 86

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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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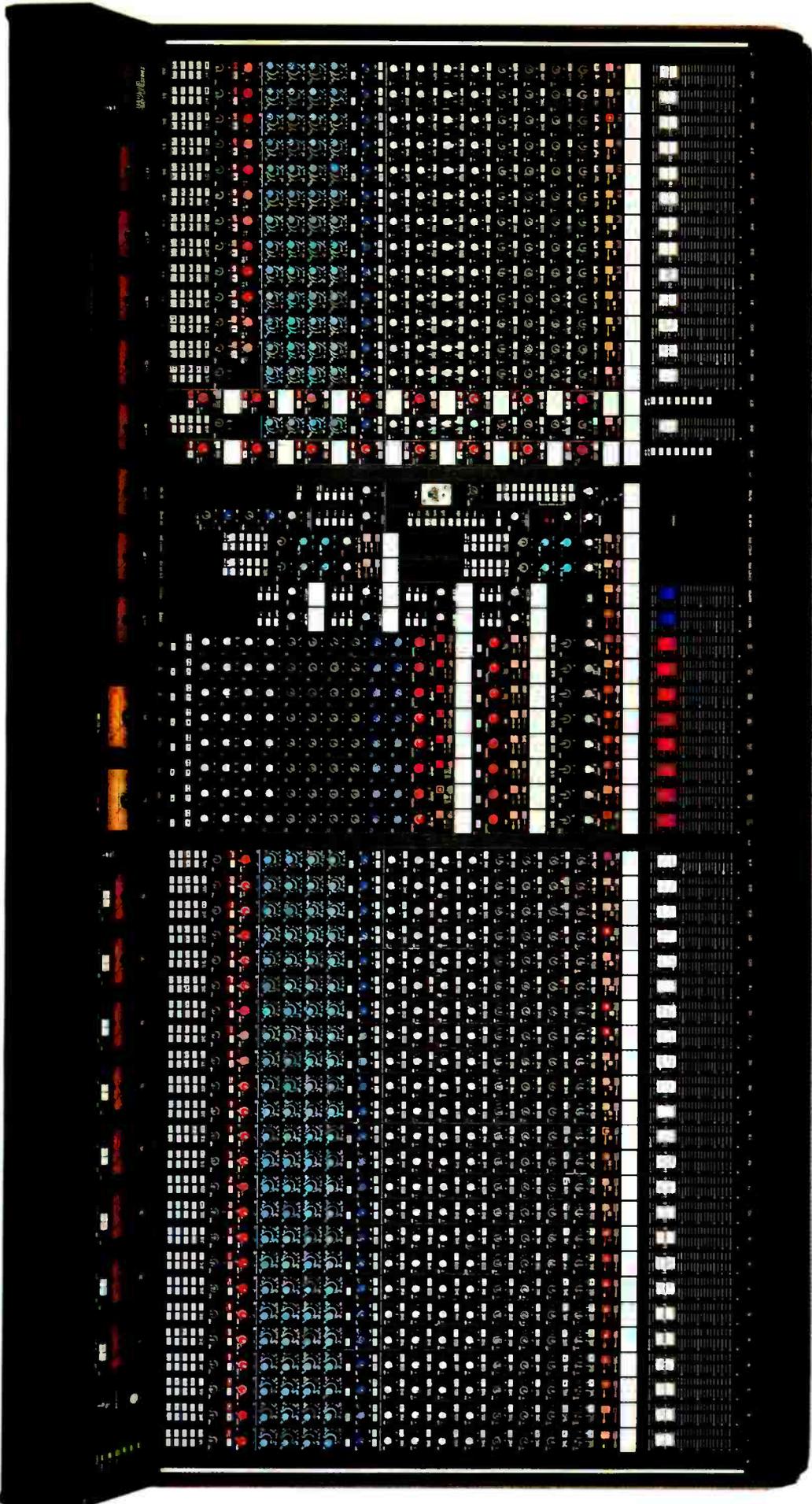
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MANAGING MIDI

By Paul D. Lehrman

Dear Santa...

Christmas is approaching, and it's time to petition the old guy with the white beard for those toys we can't possibly live without. So here's this year's wish list.

1. *True multi-tasking music software.* Most computers used for music production now have some kind of operating system available in which multiple programs can run simultaneously, so that you don't have to quit one program before you launch another. These were originally designed for desktop publishers so that they could run graphics and text-processing programs at the same time, but we can benefit from them, too.

They are invaluable for those times when you want to edit a sample while you're in the middle of putting together a tricky sequence. But these systems need to go a little bit further before they can be really useful.

In the case of one computer, the multi-tasking environment only supports programs from one manufacturer. This is OK unless you want to use a program made by someone else, in which case you're out of luck. On the other computers, these environments don't allow programs to *really* work simultaneously: you can't get a patch editor to send a burst of system-exclusive information out while a sequencing program is playing.

Instead, you have to stop the sequence, send the patch, and restart the sequence. In some cases, you even have to reconfigure the I/O ports of the computer before you can jump to a different program, because the ways the various programs handle I/O routines are not only incompatible, they actually negate each other.

The lack of true simultaneity is a particular drawback when it comes to doing studio automation over MIDI. A music sequencing program is not necessarily the best interface for handling fader motions, processing changes, or sound-effects syn-

chronization, so specialized software has been developed for those aspects of production. But you can't run that software at the same time you're running a sequencer, so you can't use it in real time to automate those production tasks that the sequencer can't handle!

To get one MIDI program to behave itself in the presence of another is no minor task; each program has to be aware of how the other is handling interrupts and other I/O routines. But this is by no means impossible—all it requires is for the software developers to get their heads together and come up with a common plan for doing it, or for the computer manufacturers themselves to prescribe ways of handling the situation that all of the developers can feel comfortable with.

2. *Manufacturers that pay attention to standards.* Of course, once you get a standard agreed to, enforcing it is the next problem. It's very nice that new additions to the MIDI spec like MIDI Files, MIDI Time Code (MTC), and the Sample Dump Standard (SDS) have been worked out, but if manufacturers don't include them in their hardware and software, they're worthless.

Sometimes standards are just ignored: one synthesizer manufacturer publicly hailed the new sample standard, and then came out with a new sampler that doesn't subscribe to it at all. Sometimes they're just implemented poorly: a program out almost a year now that makes a big deal out of being MIDI File-compatible has been shipping with a bizarre bug—if you import into it a MIDI File that has any programmed ritards (slowing of tempos), the program will exaggerate the ritard to the point where it seems to drop dead.

3. *Adoption of MIDI as a studio control standard.* Oh, for the day when I can use my computer to run my tape deck. Why can't I just have some buttons in a corner of my screen that say "Locate," "Record," and "Stop"?

At the last NAMM show I had a fascinating talk with a major tape recorder maker about how this kind of a scheme could be implemented. We realized that the hardware doing the controlling (the computer) would have to get some kind of feedback from the hardware being controlled (the tape deck), like the status of its controls or the location of the tape, but MIDI can't talk in two directions at once.

Using two MIDI lines, however, would solve the problem easily. As to the data

format, another universal system-exclusive message (which is what MTC and SDS use) could be devised. Now it's just up to the software and hardware manufacturers to meet and work it out.

After they get it to work with tape decks (video and audio), the next step would be to get the new standard working on CD players, hard disks, videodisc players, WORM drives, character generators, still stores, video switches, etc. etc. etc..

4. *A cheap DAT. Or better, two.* I hate tape. I've always hated tape, and now that I work in my home studio in an essentially tapeless medium, I hate it even more. It's noisy, it degenerates, and if you make a mistake handling it, it's all over. When I finish a project, however, it has to go on tape.

I use my trusty old 2-track only when I have to—most of the time, I record digitally with an F1-type converter and a half-inch video deck. It's reliable, it sounds good, and if you don't want to buy one, it's not hard to find one to rent inexpensively. But you can't edit with that kind of a system, unless you don't mind 3-second holes in your music and the occasional loud pop as the muting cuts out. On the other hand, I can't possibly justify buying a complete three-quarter-inch editing system, so I'm stuck.

But wait—here comes DAT! Or there went DAT... Thanks to the geniuses at the RIAA, the only way I'm going to get a digital deck into my studio this year is to pay someone to go to Europe or Japan and pick it up for me. And because the consumer market has been totally quashed in this country, the only decks out there are "professional" models that have twice as many features as I need, with price tags to match.

For the past year, you've been able to get a perfectly acceptable CD player for well under \$200—the same price as a good turntable. So now that the Copycode nonsense is out of the way, I'm hoping digital decks will soon be available for about the price of a good analog cassette deck. A DAT is the perfect complement to the home MIDI studio. You can do assembly edits accurate to within a couple of frames, and of course, there's no generation noise. With a couple of those things around, I might even get to like tape.

RE/P

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

SPARS ON-LINE

By Chris Stone

Making Good Money in the Audio Business

Making good money in the audio business has always been a problem. Why? Because the number of people trying to enter this industry has always been greater than the jobs available. Twenty years ago, the entry-level wage for an audio trainee in a recording studio was \$3 an hour. Today it is \$5, a disheartening figure when you consider that today's audio trainee probably has some college education and many even be a graduate of a four-year school specializing in audio. The audio grad goes off to a studio while fellow students land positions starting at \$30,000 a year (\$15 an hour).

What is the lure of working in audio? It's simple. People love music and want to be involved in the industry. They will settle for less money while they search for a niche that may lead to a better income. Being in an exciting, creative production environment is preferable to most workplaces. Participating in musical events provides a level of excitement that balances the salary struggle.

Obtaining an entry-level position is the same in most fields. You beat on the doors of studios or post-production houses until someone gives you a break at minimum wage. You now have that chance to prove yourself. You'll probably be treated like a slave, but there is opportunity. If you're lucky, you may find yourself working with the pros and finding out how things really work at the top of the music industry.

You start at the bottom and plan to spend a year or two paying dues to get a chance to get your hands on that expensive equipment. You get some practical experience, maybe even some formal training. Learn, learn, learn, and be prepared to put in 100 hours a week if asked. Benefits to go along with the slave wages usually include a portion of your medical insurance, a week sick leave and two weeks vacation per year. Beyond instruction in the operation of sophisticated

Chris Stone is president of L.A. Record Plant and is a former SPARS president.

equipment, you may meet that client who will someday lift you from an assistant position into a truly creative role.

Another path to take is technical maintenance. In today's solid-state, digital world, a maintenance technician needs to know more than how to repair a TV or a radio. It's best to have an electrical engineering degree with a digital minor if you want to land a job as a trainee (or "pup") in a good studio or post-production house. The only other entrance is from the practical side, perhaps as a roadie who learned to "fix" things. Once in the door,

Being in an exciting, creative production environment is preferable to most workplaces.

however, the entire picture changes. Starting wages today are approximately \$8 to \$10 an hour. Within a year, if you have any chops, it goes up to \$12 to \$15 an hour. With any luck, there is the potential of \$20 to \$25 an hour and lots of overtime. This translates as \$40,000 to \$60,000 a year.

Steps to advancement as a production trainee usually mean staying with the same studio throughout your training. If you move, you will have to start over at the bottom. The tech pup, on the other hand, has the ability to move from company to company because there are more jobs than qualified applicants. Salary increases are also different. For the production trainee, there should be a review every three to six months with an average 5% to 10% increase, topping at about \$10 an hour, unless you find a union position. The tech pup can look for more money at a faster rate, with a cap of about \$20 an hour dictated by the wages paid for comparable union work.

While the maintenance technician will hit a salary ceiling, the production person, once trained, can command up to \$100 an hour, plus points, credits and hanging out with the big guys. The glamour is all on the production side, as is the high earning potential. We've all heard of the superstar audio engineer who is in great demand because of production expertise and the "magic ears" that earn big bucks. But for every superstar, there are 10 qualified folks trying to find a gig.

The profile of this industry is changing. In the major markets, the accent is now on audio for visual production. This usually means union, and getting into the union is usually a Catch-22 hassle. You can't get into the union until you have a union job and you can't get a union job until you're in the union. Because unions live on dues, however, check with the local in your area and find out what the entry requirements are and who might be available to give you a helping hand. Once in the union, you can look forward to higher wages and better benefits if you are working in a major market such as New York, Chicago or Los Angeles. Unfortunately, you will be up against a lot of older union members whose seniority will give them the first chance at any job that comes up.

There are strategies for dealing with the union situation, but they differ from market to market. One plan is to gain experience in a non-union studio or post-production facility, get familiar with the latest equipment, and then find a union house to sponsor you because you have the talent to operate the gear that older employees are unfamiliar with. The industry is evolving slowly, but new doors are opening and there is progress.

Long-term success in the audio field demands advanced technical expertise in production, post-production and technical maintenance. You must know signal flow; digital, as well as analog, circuitry; tape machines and tapeless bit streams; computers, computers and more computers. The computers control the equipment, talk to each other and control many of the musical instruments used in recording and mixing, with or without picture. Continual-

For every superstar, there are 10 qualified folks trying to find a gig.

ly upgrading your state-of-the-art knowledge, whether it's gained through practical experience or formal education, is what will put you ahead of the pack and ensure a job. Without it you have very little chance of even getting an entry-level position.

Familiarity with picture, whether television or film, is an advantage. Until quite recently, there was little difference be-

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SPARS ON-LINE

tween the two because most TV shows were shot on film, posted on film, and then transferred to video. Today, many TV shows are shot on film, but the world of telecine has entered our field and most post-production is done in video because of the speed and relative simplicity. Sync tone is becoming a thing of the past. Most post-production operations are SMPTE time code-controlled, and much of the ed-

*Look at the big picture
when determining your
career goals.*

iting and audio preparation is done electronically. This requires more technical expertise, more training and more knowledge of how picture and sound are combined.

A distinct difference between the film/TV industry and the record industry is that most audio engineers in the record world are independents. This is also true of many maintenance technicians who service the smaller studios and tape duplication facilities. In the visual industries, however, the majority of employees are on staff. This is probably because of union influence and the fact that the larger organizations dominate the visual industry because of the extensive investments needed to remain competitive.

Look at the big picture, the entire world of audio when determining your career goals. Some people enjoy the smaller entrepreneurial environment, while others thrive in a large corporate atmosphere. The choice of being a staff employee or remaining independent once you are trained determines whether you will be paid according to a union contract, or whether you will be able to demand what the traffic will bear and choose the jobs you want.

Finally, endurance in this industry is a function of your ability to stay current with the technology and adapt to the needs of your client and your employer. Persistence and knowledge can open the door. Once inside, your chances of making good money depend on tenacity and, of course, luck.

REP

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UNDERSTANDING COMPUTERS

By Jeff Burger

Communicating with the Outside World

A computer's applications can be vastly expanded when it is connected to the outside world. In this way peripherals (such as printers, telecommunications devices and musical instruments) can be accessed and controlled by our silicon friends. Two different types of communication are in common use today, *serial* and *parallel*. Some computers come equipped with serial and/or parallel ports, ready to do business. Others require that circuit cards be placed in slots inside the PC to provide these interfaces. By and large, most users don't have to know that much about these protocols to make them work, but we'll take a more in-depth look because "inquiring minds want to know!"

ASCII—Before we talk about the differences between serial and parallel, let's cover a subject they have in common.

The computer industry has the same type of standards problems as the audio industry.

Let's assume that bytes of digital data are sent from one device to another. So what! What do those bytes mean? Most often communications are used to send letters of the alphabet to a printer or another computer. Since bits and bytes are equivalent to numbers and not letters, the ASCII (American Standard Code for Information Interchange) system was developed whereby each number, punctuation mark and letter of the alphabet is represented by a number code ranging from 0 to 127. (As you may recall, that's the range represented by seven bits; the eighth bit of an ASCII byte is used for error checking.) For example, ASCII code 65 always represents a capital "A" and 97 always represents a lower-case "a" to just about every computer and printer in the Western

Jeff Burger is RE/P's consulting editor and is president of Creative Technologies, Los Angeles.

world. So if a computer sends a code 65 to an ASCII printer, the printer will print an "A."

Serial communications—Serial refers to the concept of things happening in a series or sequence. Data flows to and from the computer one bit at a time. As we've discussed before, a single bit is a simple on/off, yes-no deal, and as a combination of eight bits or more is typically required to represent anything intelligent. So, in serial communications, bits are paraded down the wire single file into a sort of electronic corral called a UART chip (Universal Asynchronous Receiver/Transmitter) until there are enough to represent a byte. That byte is then presented to the computer's CPU for processing.

The other guidelines involve protocol. For example, RS-232C typically indicates a DB-25 (25-pin) connector and the types of signals that may be communicated. As always, different manufacturers bend the rules for their own reasons. For example, Apple Macintosh computers and their name-brand peripherals use a small, round connector with eight pins for serial communications. You can certainly convert this connection to facilitate communication with non-Apple equipment, but a special cable is required.

The other serial protocol you may be familiar with is MIDI. Here, a 5-pin DIN connector and the standardization of the pin-out is specified. The MIDI spec also standardizes the transmission speed and includes its own numeric codes, which represent different notes and controls, much in the same way that ASCII specifies textual characters. In this way, any MIDI device can talk to virtually any other MIDI device.

Most parallel communications protocols use eight sets of lines so that eight data bits can be transmitted and/or received at once.

Parallel communications—Parallel, of course, refers to things happening simultaneously in a side-by-side fashion. For example, most parallel communications pro-

ocols use eight sets of lines so that eight data bits can be transmitted and/or received at once. All things being equal, this obviously provides much faster processing of data.

Like serial communications, guidelines exist that help standardize the connections and required cabling. One of the more common printer protocols is the Centronics standard that specifies a universal 36-pin connector on the printer end.

The ASCII system was developed so that each number, punctuation mark and letter of the alphabet is represented by a number code ranging from 0 to 127.

While DB-25 plugs are prevalent, the actual connectors and pins used on the computer end are all handled differently by various manufacturers.

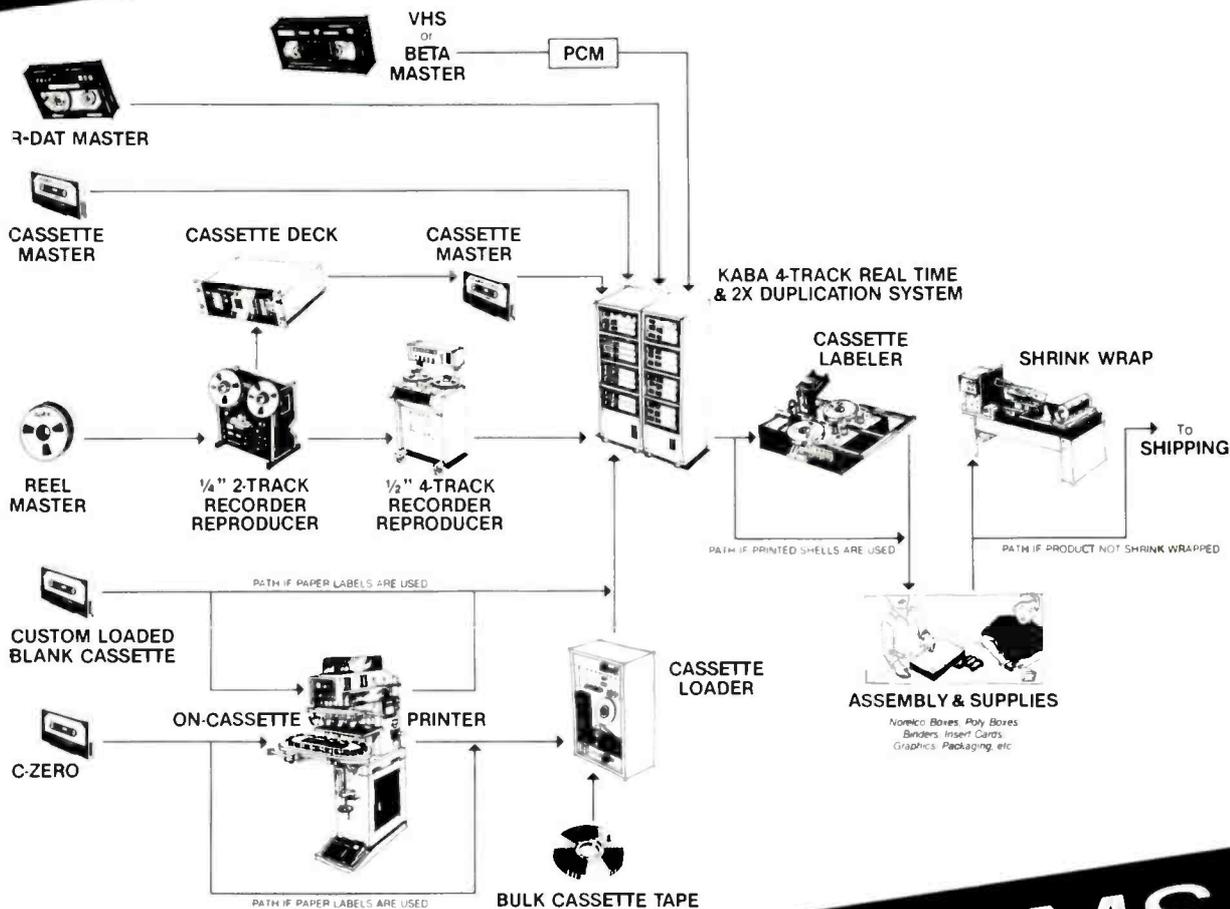
"But if serial and parallel use the same type connectors, how do I tell them apart?" Usually (yes, the computer industry has the same type of standards problems as the audio industry), serial and parallel ports will be of different genders as they appear on the computer side. Unfortunately you'll need to refer to the documentation for your specific equipment brand and type to establish proper connections.

In our next few columns, we'll take a closer look at what we actually connect to our serial and parallel ports—printers, modems and MIDI devices.

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NEWS

Syn-Aud-Con plans SR workshop

Synergetic Audio Concepts will sponsor a workshop on concert sound reinforcement in the Los Angeles area on Jan. 17-19.

Among the topics discussed will be power, rigging, house speaker systems, monitor systems, house mixing and microphone techniques. A live band will be on location to demonstrate the principles being discussed.

Don and Carolyn Davis of Syn-Aud-Con will host the workshop. Will Parry of Maryland Sound is the workshop chairman, and will be assisted by M.L. Procise of Showco, Mick Whelan of Electrotec and Albert Leccese of Audio Analyst. David Scheirman, president of Concert Sound Consultants and RE/P's live sound technical consultant, will be the facilities coordinator.

According to Syn-Aud-Con, the workshop is the first time that such important talent from the touring sound contracting industry have met to share what they know in a three-day workshop.

For more information, contact Syn-Aud-

Con at R.R. 1, Box 267, Norman, IN 47264; 812-995-8212; fax 812-995-2110.

DMM releases increasing

Since its introduction three years ago, Teldec's Direct Metal Mastering process is being used by an increasing number of U.S. pressing plants and mastering studios, according to Gotham Audio, the U.S. distributor.

More than one-third of albums on the Billboard pop album chart are DMM, with percentages higher among the top 50 and dance singles chart. According to Gotham, DMM is being used because producers, artists and promotion people still see the vinyl record as a primary medium for new releases. Other reasons include quick turnaround time, higher quality and fewer rejects than with a lacquer master.

Clarity, Lexicon complete new product

Clarity has completed the software and hardware design for a new product that will

be sold by Lexicon. The project is a part of a technology exchange agreement between the two companies.

The Lexicon MRC is a definable MIDI remote control for the Lexicon LXP and PCM 70 signal processors and for all general MIDI devices. Using the proprietary Kramer Synthesis Macros, the unit also acts as an intelligent FM programmer.

The product marks the first time that Lexicon has gone outside for product concept and design.

News notes

Intersonics has provided Michael Jackson's U.S. tour with 16 SDL-5 ServoDrive subwoofers, which are being used in conjunction with 72 Clair S-4 cabinets.

A Kurzweil K250 faced off in mid-August against Philadelphia's Wanamaker's Department Store pipe organ, said to be the world's largest. The Franklin Institute produced the event in conjunction with its "What Makes

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Music?" science exhibit to focus attention on the different characteristics and capabilities of acoustic and electronic instruments. The instruments were played separately and together. Sound reinforcement was provided by TekCom Corp. of Philadelphia. Equipment used included Boas Acoustamass cabinets and Wave Cannons, and EAW FR153 cabinets.

Sound Workshop Professional Audio Products has moved to 79 Express St., Plainview, NY 11803; 516-932-6570; fax 516-932-6573; telex 530-464.

Custom versions of **Apogee Electronics'** 944 filters have been incorporated into the Solid State Logic 01 Digital Production Center and the Harry Digital Sound Editor.

Shape Inc. increased the price of all video products by two cents, effective Nov. 1. The company cited the increase costs of raw materials, labor and packaging materials.

Canare Cable has relocated its entire West Coast operations to a complex that is five times larger than its previous location. The company's new address is 511 5th St., Unit G, San Fernando, CA 91340; 818-365-2446; fax 818-365-0479.

Burns Audio, which supplied the sound system for the Democratic National Convention in Atlanta, used Apogee Sound loudspeaker systems. The sound system included 17 Apogee 3x3s for the house, 17 AE-5s for down fill and monitors, four AE-3s for front fill and two AE-1s for obstructed seating.

Nimbus Records has opened a West Coast sales office and has appointed Sandy Richman as its western regional sales manager. The office is located at 4524 Tobias Ave., Sherman Oaks, CA 91403; 818-783-4250.

Ultra Sound as has added 10 Aphex 612 expander/gates to its equipment lineup. Six units are being used on the Grateful Dead tour, with the remainder being used on the

Bob Dylan tour.

Harris Audio, Miami, the Otari sales representative, has expanded its operation to include Georgia, and North and South Carolina. Harris will represent the DTR, MTR and MX machines in these states.

Engineered Packaging Inc. has opened a West Coast office, headed up by Ralph Hoopes. The office is located at P.O. Box 4489, 3040 E. Spaulding St., Long Beach, CA 90804-4489; 213-985-0062; fax 213-494-1713.

Gentner Electronics has consolidated its San Jose, CA, RF products division into its Salt Lake City operations. Gary Crowder, director of broadcast marketing and sales, will handle the RF product line.

Prince's European/American tour is exclusively using the **Electro-Voice** Mt-4 concert sound system. A total of 64 boxes comprise the system—32 MTH-4 high-frequency boxes and 32 MTL-4 low-frequency boxes. The system was supplied by dB Sound, Chicago.

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Microphone Calibration Techniques

An Overview, Part I

By Adrian Weidmann

An internationally accepted calibration procedure and documentation of studio microphones would allow direct comparison between different microphones.

A calibration is a measurement referenced to an accepted, known entity or standard. Conversely, it is important to note that a measurement is not a calibration unless the measurement complies with an existing standard or reference. Aligning tape machines with internationally accepted alignment tapes, setting Dolby tones, calibrating VU meters and recording reference tones on tape are examples and methods of calibrating various aspects of the recording chain. Transferring these tones via tape enables recording engineers and producers to travel from one studio to another while maintaining continuity. An internationally accepted calibration procedure and documentation of studio microphones would allow direct comparison between different microphones and allow one to calibrate a recording system, from microphone to output metering, using a known acoustical reference.

Adrian Weidmann is international manager for the Bruel & Kjaer pro audio group, Naerum, Denmark.

Without a rigid set of calibration standards, manufacturers would have little to reference when trying to design high-quality microphones. To benefit fully from such an international calibration standard, it may be helpful for engineers to have some background information on how studio microphones are calibrated and what the references are. If the signal path (using a microphone) is to impart a minimum of coloration on a sound source, it is important that the first link in the chain (the mic) maintain the integrity of the incoming signal.

The need for an accurate, verifiable calibration standard for professional recording microphones has never been more necessary. Many different microphones are used daily in facilities throughout the world. Because the microphone is often the first and most critical component in the audio chain, it would be useful to compare these different microphones directly from a "standardized" calibration and specification sheet.

Current professional microphone cali-

bration and specification sheets often obscure information to achieve the cliché "flat frequency response from 20Hz through 20kHz."

Any sound-measuring or -recording system, from the simple sound level meter to the complex, computerized monitoring or analyzing systems, or modern audio recording and storage devices, must be accurately calibrated to provide meaningful results. Regular calibration checks validate the stability of the system and ensure that the requirements of the standards, which apply to the measurements, are fulfilled. Measurements taken at timely intervals can be used for comparison purposes. The need for recording equipment "calibration" is imperative to establish a reference to allow specific sounds to be recorded/stored accurately.

Several different calibration methods are used and readily available to calibrate pressure microphones. The method of calibration to be used is directly related to the degree of accuracy required.

What to calibrate?

Basically, microphone calibration involves measuring the absolute sensitivity at one frequency and the pressure frequency response at a single amplitude level. From the mathematical results of the two known parameters, the operation of the microphone can be predicted for a wide range of applications. However, for some applications, other characteristics may be of interest. These include amplitude linearity at high sound pressure levels, phase response, transient response, and frequency response of the microphone in gases other than air or at pressures other than atmospheric. The main methods for sensitivity calibration are (1) the reciprocity method, (2) the comparison method and

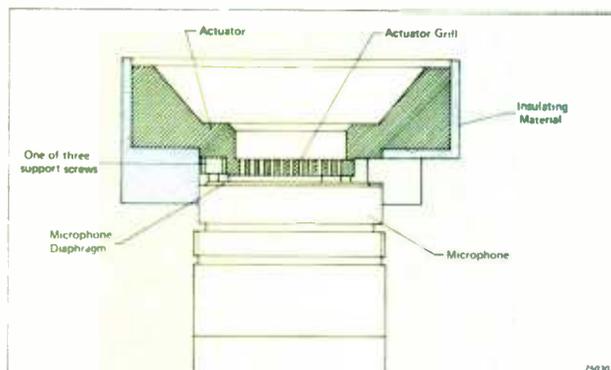


Figure 1. Principle of the electrostatic actuator method.

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(3) the *transfer* method using a calibrated sound source.

The "insert voltage" technique may be used in the three above calibration methods to obtain the microphone's open-circuit sensitivity. (The open-circuit sensitivity is the sensitivity of the microphone cartridge when it is not capacitively loaded by a pre-amplifier.)

The microphone's frequency response is normally measured using the electrostatic actuator method, which is also used for phase and transient response measure-

ments, as well as for supplying the microphone's pressure response. If the free-field response is of interest, the relevant free-field corrections can be added to the actuator characteristics. Alternatively, the free-field response may be measured in an anechoic chamber.

Electrostatic actuator measurements

Condenser microphones having a plain metallic or metalized diaphragm may be influenced by electrostatic forces similar to the action of sound waves. This is done by placing a rigid metallic grid close to the diaphragm and applying an alternating test voltage between the diaphragm and grid. (See Figure 1.)

The distance to the diaphragm is approximately 0.5mm, which provides an equivalent sound pressure level of 104dB with a 800Vdc component and a 100VRMS ac excitation voltage.

Because the magnitude of the force depends upon the distance between the diaphragm and the grid, absolute calibration is rather difficult and is not performed in practice. On the other hand, the electrostatic actuator method is a practical method for relative measurements such as frequency response, phase characteristics and pulse response measurements.

The B&K pre-amplifier Type 12627, together with a primary standard microphone and a precision voltmeter, allows a standard microphone system to be formed. Using the insert voltage technique, the system can be adjusted to indicate the microphone's open circuit voltage. Such a standard microphone system is used to calibrate secondary standards and transfer standards.

Although a microphone is always used with a pre-amplifier that does not have an infinite input impedance, the open-circuit sensitivity can be measured using the insert voltage technique, which will be described later.

Several degrees of accuracy

The sensitivity of a microphone may be calibrated with different degrees of accuracy, depending on the use for which it is intended. Basically, microphones may be divided into three groups, according to the accuracy with which they are calibrated: primary standards, secondary standards or measurement/studio microphones.

Primary standards are calibrated with the highest possible accuracy, using, for example, the reciprocity method. Primary standards are used to calibrate secondary standards, which, in turn, are used to calibrate measurement and/or studio microphones. Primary standards are also used to calibrate reference sound sources, such as the Pistonphone. The Pistonphone would then become a "transfer" standard. These transfer standards are then used for the calibration of secondary standard microphones or complete sound-measuring systems at a single frequency. (See Figure 2.)

Outside the calibration of secondary or transfer standards, primary standard microphones are seldom used for continuous calibration work. They are carefully protected against the rigors of routine work so that they develop a history of stable reference—confirmed by annual certifications by external authorities.

Pistonphone

The Pistonphone (B&K 4220) is a highly stable sound calibrator that operates at 250Hz and produces a sound pressure level of 124dB (re 20μPa). Calibration accuracy is ±0.15dB. (See Figure 3.) (Note: A Pascal is a unit of pressure corresponding to a force of 1 Newton acting uniformly upon an area of 1 square meter. Hence, 1 Pa = 1N/m² = 94dB SPL. The weakest sound a healthy human ear can detect has an amplitude of 20 millionths of a Pascal (20μPa), some 5 billion times less than normal atmospheric pressure. A pressure change of 20μPa is so small that it causes the eardrum to deflect a distance less than

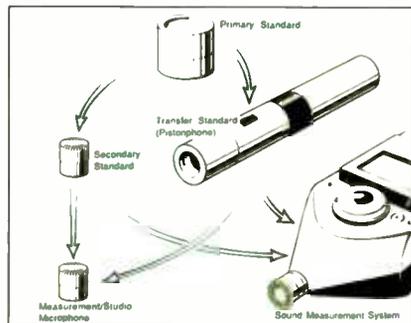


Figure 2. The calibration transition from a primary standard through a secondary or transfer standard to the sound measurement system or measurement/studio microphone.

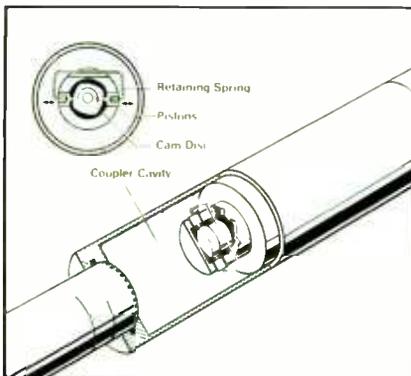


Figure 3. Principle of the Pistonphone.

Microphone sensitivity

Generally, a microphone is measured in terms of open-circuit sensitivity. This is defined as the ratio of the microphone output voltage to the sound pressure acting on the diaphragm when the microphone is working into an infinite electrical impedance. One advantage of this definition is that, when the microphone capsule is mounted on a pre-amplifier (usually the body of the mic), it is a simple matter to derive the overall sensitivity of the complete microphone assembly if the open-circuit sensitivity, the capacitance of the microphone capsule and the input impedance and capacitance of the pre-amplifier are known.

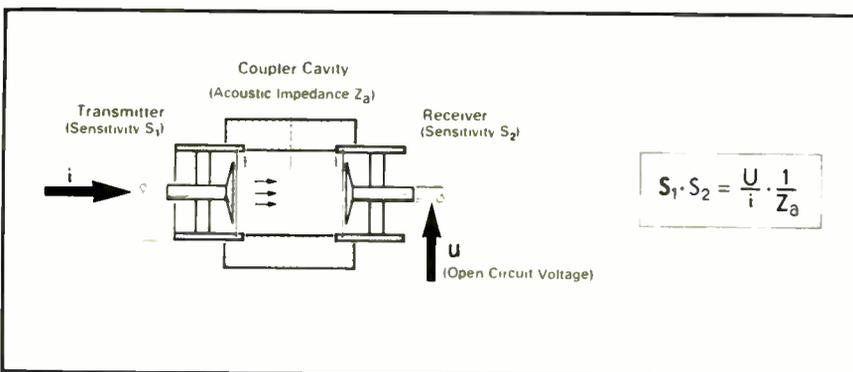


Figure 4. Principle of pressure reciprocity calibration.

the diameter of a single hydrogen molecule. The ear can tolerate sound pressures more than a million times higher. Thus, if we measured sound in Pa, we would end up with large, unmanageable numbers. To avoid this, another scale is used: the decibel or dB scale.) The sound pressure is produced in the cavity by two pistons moving in opposite phase. This arrangement, together with careful cam design and machining, ensures very high stability and very low non-linear distortion.

Primary microphones

The most commonly used primary standard microphones are 1-inch condenser microphones that meet the Type L requirements of ANSI A1.12 1967, "Specifications for Laboratory Standard Microphones."

These specifications, originating in the 1940s, were written around the only instrumentation condenser microphone then available, the Western Electric 640AA. In 1958, Bruel & Kjaer introduced the 4132, later redesignated as Type 4144. This microphone also meets or exceeds the ANSI Type L requirements. Recently developed, the B&K Type 4160 also meets the ANSI standards and is an exact equivalent of the 640AA, having the same nominal sensitivity and equivalent volume of air behind the diaphragm.

Reciprocity calibration

Reciprocity calibration is an absolute, highly accurate calibration method using three microphones. The calibration can be obtained without reference to a primary standard microphone or sound source. The method is described in IEC Recommendation 327 and 402 and in ANSI S1.10-1966. The accuracy obtained on the pressure sensitivity is estimated to be 0.05dB.

Two microphones, M1 and M2, are acoustically coupled in a small closed cavity where M1 is used as the transmitter and M2 as the receiver. From the current through the transmitter microphone, the output voltage of the receiver microphone and the cavity volume, the product of the microphone sensitivities may be calculated. Using a third microphone, M3, in turn with M1 and M2, three sensitivity products are obtained, from which the individual sensitivities are derived. Or, two microphones and an auxiliary sound source can be used.

Since reciprocity calibration is used to measure the open-circuit sensitivity, the insert voltage technique must be used to compensate for the loading effect of the pre-amplifier. (See Figure 4.)

Comparison calibration

Comparison calibration is the most widely known technique for transferring calibra-

tion from the standard to the unknown microphone. The microphone under test and a standard microphone are submitted to the same sound pressure, and the sensitivity of the unknown microphone is derived from the two resulting output voltages. In practice, the method can be implemented in three different ways: in a coupler, in an anechoic chamber, or using a transfer standard. (See Figure 5.)

Comparison calibration in couplers can be carried out in two ways. The first

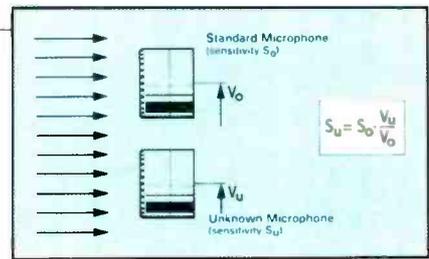
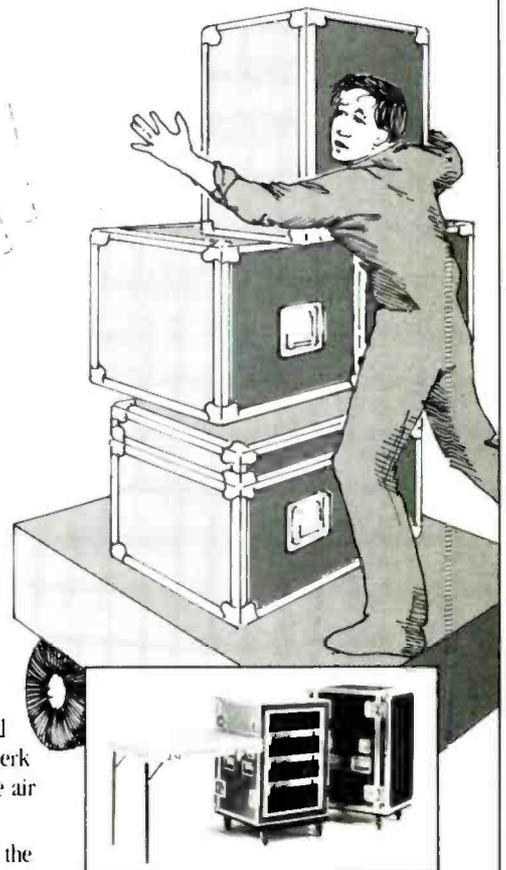
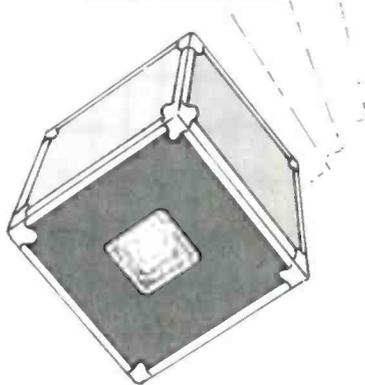


Figure 5. Comparison calibration. Two microphones, a secondary standard and the microphone being calibrated, are subjected to a known sound source. Since the sensitivity of the standard microphone is known, the unknown sensitivity can be derived from the sensitivity product.

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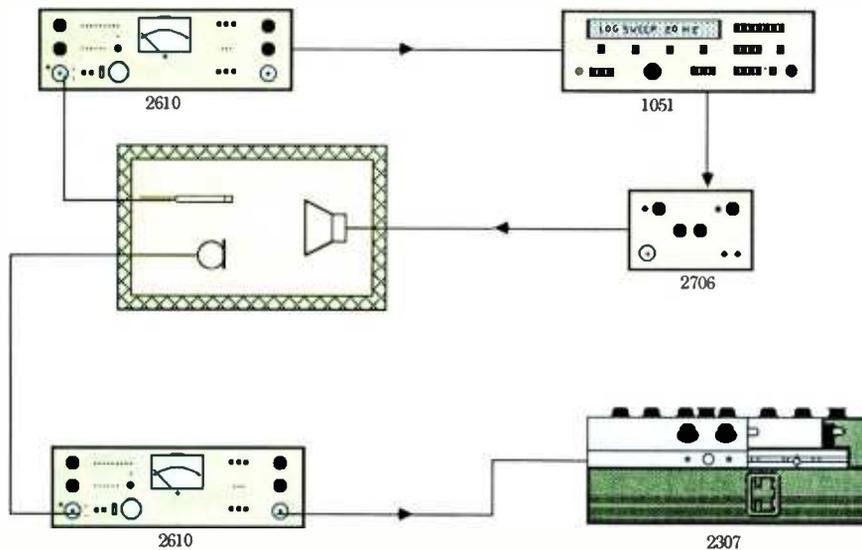


Figure 6. Setup for comparison calibration in an anechoic chamber using a compressor loop.

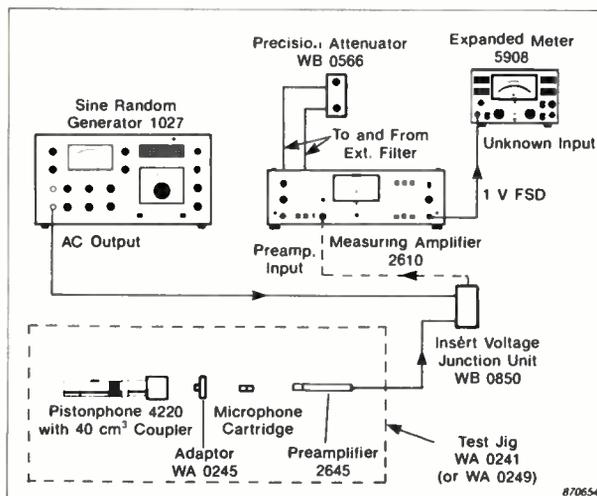


Figure 7. Comparison calibration set-up using a Pistonphone as the transfer standard.

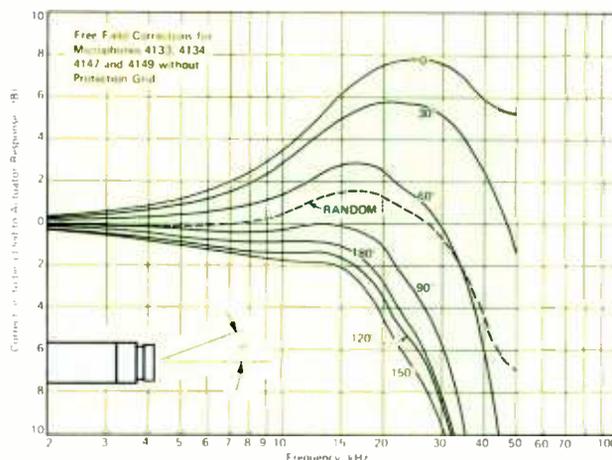


Figure 8. Example of free-field correction curves.

method uses both the standard and the unknown microphones as receivers in a coupler in which the sound pressure is created by some other source. The second method uses one microphone (for example, the standard) as sound transmitter and the other as the receiver. The microphones are set up in the same way as for reciprocity calibration, and their sensitivity product is measured. Since the sensitivity of the standard microphone is known, the sensitivity of the unknown can be derived from the sensitivity product.

Calibration in anechoic rooms allows measurement of the microphone's free-field response. If the standard microphone has a flat frequency response over the frequency range of interest, it can be included for automatic recording of the frequency response of the unknown microphone. If the microphone is rotated on a turntable, directivity diagrams may be obtained at fixed frequencies. (See Figure 6.)

In comparison calibration using a transfer standard, the standard and the unknown microphones are submitted in turn to the sound pressure delivered by a sound generating device that is a transfer standard, such as a B&K Pistonphone.

Transfer standards are used for comparison calibration of microphones. They also provide a simple, accurate means of calibrating complete measuring systems, both in the field and in the laboratory. (See Figure 7.)

Free-field corrections

Most microphone calibration methods give the microphone pressure response, that is, the response of the microphone to the sound pressure actually acting on the diaphragm. However, when performing free-field measurements, the original sound field is affected, at higher frequencies, by the presence of the microphone, which results in an increased sound pressure at the microphone diaphragm. This pressure increment is called the free-field correction and should be applied to the microphone pressure response to obtain the microphone free-field response at a given angle of sound incidence. (See Figure 8.)

Calibration and specification "standard"

The calibration techniques outlined above are part of and comply with world standards. In Part II of this article, the microphone calibration sheet itself will be looked at. To promote a "standardized" microphone calibration sheet, we will make a proposal to help audio professionals make an accurate comparison of two different microphones—before they "reach your ears."



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Mic Pre-amp Evaluation Methodology

By John Hardy

The following thoughts are directed at stand-alone mic pre-amp evaluation, but can generally be applied to any piece of equipment in the signal path.

Evaluating mic pre-amps can be simple, enjoyable, rewarding and accurate. Then again, it can be confusing, misleading, frustrating and impossible.

It seems so easy. Take the best microphones, power amps and speakers you can get, insert the mic pre-amp in the signal path, then listen. But there can be problems. (*Lots of problems.*) Whether you are looking for the most accurate mic pre-amp, or the mic pre-amp with just the right colorations for your application, you have much to think about. The following thoughts can generally be applied to any piece of equipment in the signal path.

Finding the weak link

You must intimately know the character and performance limits of every piece of equipment in the signal path. Otherwise you cannot be certain what to blame when things don't sound right. An unknown weakness or characteristic in one piece of equipment can mislead you about the performance of other equipment in the path.

Sooner or later it happens to all of us: You blame a problem on one piece of equipment, only to discover six months later that something else was causing the problem. Up to a certain point you can mentally compensate for a weakness or characteristic such as mild frequency response variations. Beyond that point you must upgrade the equipment if your evaluations are to be accurate.

Ideally you will be evaluating on the same equipment that you work with every day. That way, you will intimately know

the limits of the equipment and will have eliminated any serious problems. If you are forced to evaluate an unfamiliar system, you must get to know the system before you can do anything meaningful. Otherwise, you are wasting your time.

You must also eliminate as much unnecessary equipment as possible from the signal path. That way, there will be fewer sources of error, degradation and mud to mislead you. The phrase "minimum signal path" is very appropriate. Ideally, you should go straight from the microphone to the mic pre-amp, and straight from the mic pre-amp to the monitor amp.

You should avoid the console completely, but if you must use a console, do the most direct patching possible to avoid as many op-amps, coupling capacitors, switches, connectors and miles of wire as you can. They all degrade the audio signal to some degree.

Avoid all outboard equipment. Even if the outboard gear is in "bypass" mode, it is often not a complete bypass. The signal is still sent through additional connectors, switches, op-amps, coupling capacitors and other parts, which are all sources of signal degradation.

The record/playback process, whether analog or digital, adds too much coloration to be part of the optimum signal path evaluation. Yes, sooner or later many of us have to record something to make a living, and you need to find out how the sound quality holds up after the record/playback process.

Ultimately, you need to hear how it holds up at the final destination, whether that destination is an expensive home system, a Walkman, a juke box, a PA system, my favorite 1958 General Electric Musaphonic AM radio, or whatever. But, for the

most accurate evaluation of a mic pre-amp, the recording process should be avoided because it is a source of further error. It is a convenient way to do comparisons and has been used in many equipment reviews, but it is not accurate.

A popular analog tape machine sends the signal through a dozen op-amps just to get to the record head. Add the coupling capacitors, FET switches and EQ networks with their potential phase problems, and you will certainly have errors. And then there is the temperamental magnetic process involving the record head and tape. (Whew!)

Digital recording systems generally have their own problems with limited sampling rates, bits and anti-aliasing filters. Many consoles send the signal through a horde of op-amps, capacitors, FET switches and EQ networks too, just to get from a mic input to a bus output. Then it starts all over again when it's time to play the tape back. More errors. Avoid all of that circuitry whenever possible.

Whether you are working on your own system or an unfamiliar one, be suspicious of everything. Paranoia and skepticism are encouraged. Take detailed notes of the equipment being used, and the conditions under which it is being used. You might even want to consider the weather, and your personal health and mood. We all have days when nothing sounds good, and it's our own fault. A head cold, or head trip, can ruin everything. Start at the beginning and meticulously go all the way through the equipment until you are sure of what you are dealing with.

Microphones

What microphones are you using? What is their condition? Is this your first exper-

John Hardy is president of the John Hardy Company, Evanston, IL, a manufacturer and designer of professional audio products.



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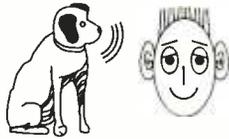


Figure 1. Direct sound.

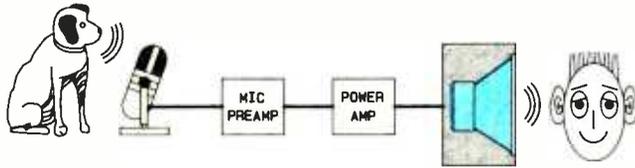


Figure 2. Minimum signal path.

ience with a particular model? If so, you can't be sure whether it is the mic pre-amp, or the microphone, or some combination of the two that creates the final sound. Pick-up patterns (on-axis and off-axis), capsule and electronics overload characteristics, noise, maximum input and output levels, and output impedance must be considered.

Mic cables

Use the shortest, best mic cables available. Go directly from the microphone to the mic pre-amp, avoiding the patch bay or any other connectors and cables. It does make a difference.

Mic pre-amps

One very important point: Many of today's console mic pre-amps leave much to be desired. They mask the full capabilities of your microphones. Some of them actually alter sound enough to force you to head for the EQ, hoping to re-create the sound as you heard it in the studio.

The problem may be inferior circuit design, poor execution of a design, or both. Perhaps there are budget limitations that cause unavoidable compromises in the choice of op-amps, transformers, capacitors and other components. Perhaps there is a lack of experience in basic mic pre-amp design. An expert in digital design may not be aware of the unique requirements of analog design. Improper grounding layout and circuit layout can seriously degrade the performance of even the best design. Whatever the reasons, the compromises are there.

This creates a serious problem: How can you make meaningful decisions about mic pre-amps when you've never properly heard your microphones? Do you *really* know what a U87 or SM57 is capable of? This is where your years of experience, talent, knowledge and intuition are really put to the test.

Regardless of what microphone you are using, you will eventually come upon a mic pre-amp that simply does a better job than the others, at least under certain conditions. If your goal is the highest accuracy, perhaps the better pre-amp will provide less distortion, or extended bandwidth, or a firmer low-end response, or a smoother mid or high end. Maybe it is more transparent. Cleaner. Warmer. Better focused. More detail. More air. Or maybe it's just incredible! No comparison! Descriptive terms can get pretty strange, but somehow, the meaning usually gets across. If your goal is a colored sound or special effect of some sort, you will recognize it when you hear it.

There are many variables to consider. Some pre-amps have transformer-coupled inputs, others have transformerless inputs. Some have transformer-coupled outputs, others have transformerless outputs. Some outputs are balanced, others are unbalanced or single-ended. Some mic pre-amps use monolithic op-amps, some use vacuum tubes, and some use combinations. There are many opinions on the advantages and disadvantages of these various constructions, and wide variations in quality within each type of construction. Decide for yourself.

Gain structure is very important. One engineer might be recording soft classical music from a distance, requiring a pre-amp with very high gain and low noise. Another might be recording a stack of screaming amplifiers at point-blank range, requiring a pre-amp with very low gain and the ability to handle extremely high input levels without distortion. With today's high-output condenser microphones, even routine vocal overdubs can drive some pre-amps into unavoidable distortion.

Some pre-amps sound better at certain gains than they do at other gains. Logically, you should be testing mic pre-amps

under circumstances as close to yours as possible, but you must also consider the performance of the pre-amps under other circumstances. That classical engineer might suddenly find himself having to record a screaming amp one day. Good results for one set of conditions do not guarantee good results under others.

Find out the maximum gain available from the pre-amps. Also find out the maximum signal level the pre-amps can handle without causing distortion, and determine whether the distortion is caused by overloading the input or clipping at the output of the pre-amp. Also related to this is the minimum gain the pre-amps provide, and the maximum output level the pre-amps can produce.

For example, if you have a signal coming in at +4dBV, and the minimum pre-amp gain is 20dB, the output will be at +24dBV. That is fine if the pre-amp can handle a +4dBV input and provide a +24dBV output, but the signal will be hopelessly distorted if the pre-amp is only capable of a +18dBV output. Even then, you must also consider the load impedances the pre-amps will have to drive. All pre-amps should be able to drive a 10kΩ load, but what if you are driving several tape machines at once, resulting in a 1kΩ or 2kΩ load? Play with the figures to see if the pre-amps can handle your requirements.

The method of gain adjustment is important. A multiposition gain switch might have the advantage of being resettable to exact gains, but what if the required gain falls in between two positions? Do you tell the singer to back up, or sing softer? Not likely. More engineers are going straight from the mic pre-amp to the tape machine these days, and there is no way to achieve an "in between" gain setting without going back through the console to use a fader. (And the whole idea was to avoid the console.) A continuously variable gain pot has an advantage in this case. It also allows you to ride gain. Everyone's situation is a little different, so you must decide what is best for you.

Contact plating is an area of great debate. Audio fanatics are sometimes called "golden ears," yet some of them prefer silver-plated switches and connectors.

The phantom power supply is another concern. Some condenser microphones consume relatively high levels of current from the phantom supply. Some mic pre-amps can't provide sufficient current to operate those microphones. Some mic pre-amps don't have a phantom supply at all. Others have the supply, but you have to disassemble the pre-amp to turn the supply on and off. Check your requirements. [See "Phantom-Powering Precautions" on page 36.]



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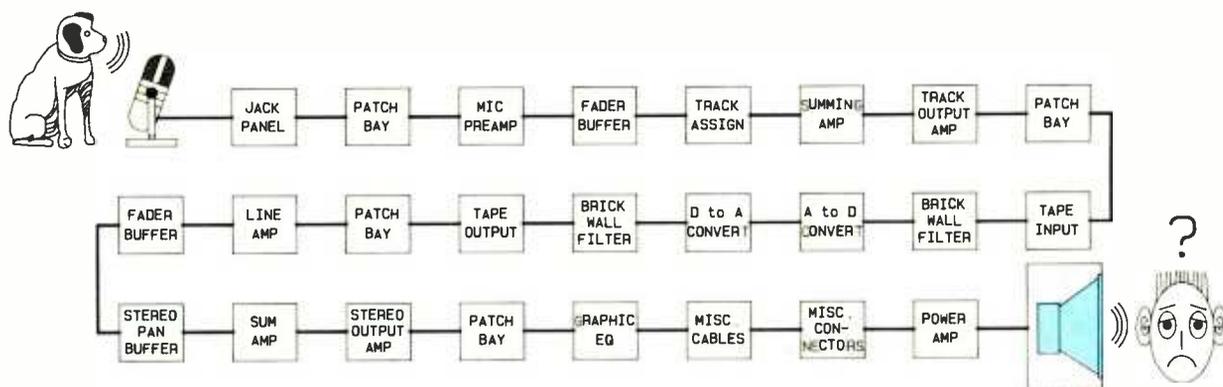


Figure 3. A typical studio monitoring signal path.

A microphone might sound better with some pre-amps than it does with others. Aside from the fact that some pre-amps are simply better than others, the differences in sound quality could also be caused in part to the matching of the microphone output impedance to the mic pre-amp input impedance. The output impedance of a "low impedance" microphone can vary substantially from one model to the next, and the impedance is not always linear. The input impedance of a mic pre-amp may also vary from one model to the next, and is not always linear. These impedances can interact like an equalizer to create subtle (and often not so subtle) alterations to the frequency and phase response of the audio signal. Distortion might also be increased. If the effect is desirable, great! If not, move along. You could conceivably end up with several pre-amps, each being best with a specific microphone.

Try to use the same listening levels when comparing equipment. This can be complicated by the fact that alterations in frequency and phase response, and differences in distortion may cause one pre-amp to sound louder than another, even when meters say the levels are the same.

Phase polarity is another critical item. Some mic pre-amps invert the phase, others don't. If you are comparing one pre-amp that is in-phase to another pre-amp that is inverted, your results will be faulty. Even two identical pre-amps will sound different if you reverse the phase of one of them. This problem is compounded by the fact that not all microphones are wired with the same polarity either. Throw in a mic cable that (unknown to you) has pins 2 and 3 wired backward at one end, and you have a serious problem. You must verify phase consistency in all of the equipment. A positive pressure on the microphone diaphragm should cause the monitor speaker cone to create a positive pressure by moving *out*. If there is a phase inversion in the system, the speaker will move *in* when a positive pressure is ap-

plied to the microphone diaphragm.

Features and parameters such as phase reverse switching, metering, common mode rejection ratio, equivalent input noise (and others) must be considered. You should also look at factors such as ease of operation, quality of construction, quality of components, serviceability and long-term reliability.

Interconnect cables

The cable from the mic pre-amp to the power amp may not be quite as critical as the mic cable, but then again maybe it *is* as important. Use the best cable you can get.

Power amps

A wide range of amplifiers is being used in studios today, and substantial differences in their sound quality exist. High negative feedback, low negative feedback, no negative feedback, tubes, field-effect transistors, bipolar transistors, class A, AB, C, D, and so on. You must know if the amps are causing any degradation.

Speaker wires

(Here we go again.) Not all wires are the same. Investigate.

Monitor speakers

What kind of monitors are being used? Big ones with response from dc to UV, or small close-field monitors with limited bandwidth? Do they sound good, or are they just popular? Do they have active or passive crossovers? Are they stock or modified? Who modified them? Did the modification make their performance better or worse? Lots of "great ideas" turn out to be not so great. Lots of great studios have their peculiarities.

Is a graphic equalizer being used? If so, check it out. It may be causing more problems than it is solving. Check the overall frequency response of the speakers. Different engineers have differing opinions on the ideal frequency response of a monitor system. Listen for phase problems at

the crossover points.

Your listening position relative to the monitor speakers is critical. Slight changes in listening position can cause major changes in frequency response and imaging. I have seen cases when a group of people gather around to listen, and you know they can't *all* be in the sweet spot. Take turns, do what you must, but find the optimum listening position. Anything less is probably a waste of time.

The room

Find out how the room affects the sound. Live ends, dead ends, standing waves, reflections and resonances can all affect what you hear.

It is common practice for an engineer to bring a favorite master tape along to find out how an unfamiliar system sounds. You could also play a favorite record or CD through the system. It must be source material that you are extremely familiar with, material that you have heard on a wide variety of systems to avoid the problem of "tunnel vision," aurally speaking.

The material must really challenge the unfamiliar audio system so you can find out where it is coming from. Discover the flavor and personality. Or discover a burned-out tweeter as I once did. You still face the potential errors of the tape machine, CD player, phono system or cassette deck, but you should get at least a rough idea of how the system sounds. You might even consider bringing your own cassette deck or CD player.

Perhaps a better approach would be to bring a reference microphone and mic pre-amp to help you find out how the system sounds. Listen to something live, using the minimum signal path. This would eliminate the errors in the tape and CD methods.

Mic positioning and ear positioning

If you are not planning on sticking your ears exactly where the microphone is going to be placed, you will not hear the

Phantom-Powering Precautions

By Bruce Bartlett

A drop in phantom supply voltages may be what's causing distortion or level loss in your condenser mics.

There's a new problem with phantom-powering condenser microphones, and it could affect you. It's becoming more common with today's multi-console systems, in which you "Y" or split a microphone signal to feed more than one console in parallel. If there are significant differences or weaknesses in the phantom supply design, the supply voltage may drop from, say, 48V to 5V—too low for the microphone to operate properly—causing distortion or level loss.

There are several ways to solve the problem of power-supply sag. But to understand them, we need a brief review of phantom powering itself.

Definitions

A remote microphone power supply is a device that provides the voltage that operates the microphone circuitry. In an externally biased mic capsule, the remote power also is used to polarize the condenser transducer. (Microphone capsules using the electret or RF principle do not need external polarization.)

Remote power can be supplied from a stand-alone unit or from the console. In the latter case, a dc voltage is applied to each mic connector in the console and is usually switchable, on or off, for each input.

Remote dc powering can be supplied in

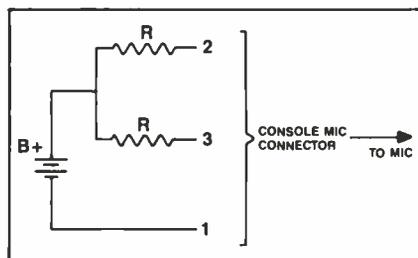


Figure 1. Basic simplex phantom-power supply circuit.

two ways: (1) through power wires in the mic cable that are separate from the audio wires or (2) through the audio wires. (In the setup described in method 2, the microphone receives power from, and sends audio to, the console along the same cable conductors.)

With method 2, two systems are used, phantom (also called simplex or multiplex) and AB or T (also called modulation-lead powering).

Phantom powering, the most common system, is a positive voltage (12V to 48V) on pins 2 and 3 with respect to pin 1. The cable shield is the supply return. There is no voltage between pins 2 and 3. This is DIN standard 45 596, which is called "multiplex powering." (Apparently the terms "simplex," "multiplex," and "phantom" mean the same thing in this context—we welcome letters clarifying this subject.) Pin 1 is ground; pin 2 is audio in-phase, and pin 3 is audio return. Although the audio

signal has polarity, simplex phantom power has no polarity because both pins 2 and 3 have the same positive dc voltage.

You can plug a dynamic microphone into a simplex phantom supply without damaging the mic. That's because the voice-coil leads are not connected to pin 1, so no current from the phantom supply can flow through them.

AB powering applies positive voltage to pin 2 and negative voltage to pin 3. This is DIN standard 45 595, modulation-lead powering. It's less commonly used because any noise in the power supply is amplified by the differential-input mic pre-amp. Also, AB power might destroy a dynamic microphone. The main advantage is that it avoids undesirable current in the shield. (In the United States, AB powering is used almost exclusively in the motion-picture industry, in which Nagra recorders supply AB powering for certain Sennheiser RF condenser shotgun mics.)

Most older tube-type microphones do not work on phantom power (12V to 48V). They require their own power supplies because their polarization voltages and filament voltages are relatively high.

A real world example

Let's return to the problem of the phantom supply sagging when a microphone feeds two consoles. Here's an example:

Say a house console supplies 48V between pins 1 and 2, and between pins 1 and 3 (measured using a good VOM with

Bruce Bartlett is senior microphone development engineer at Crown International, Elkhart, IN.

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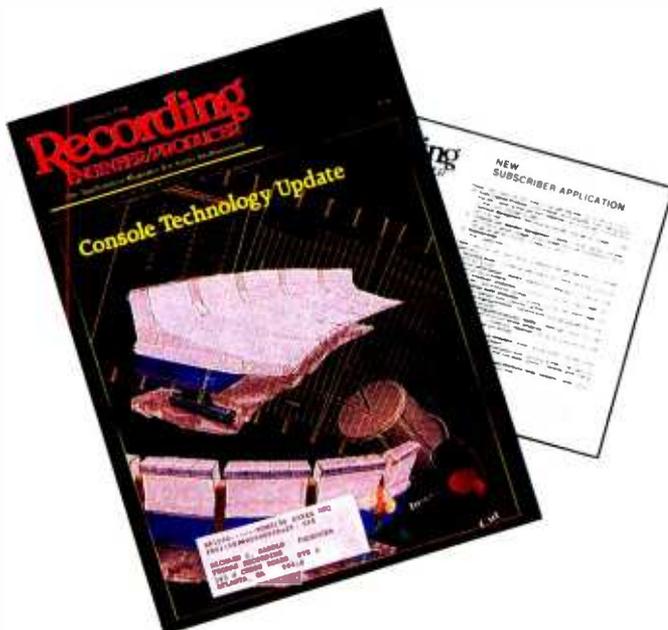
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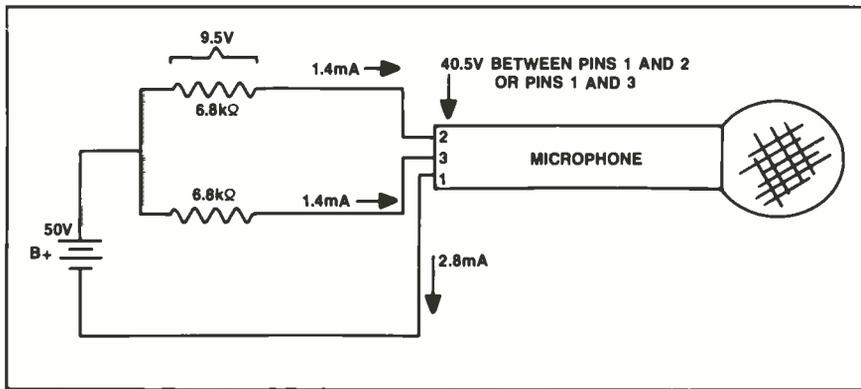


Figure 2. Phantom-voltage loss caused by microphone drawing current from the supply.

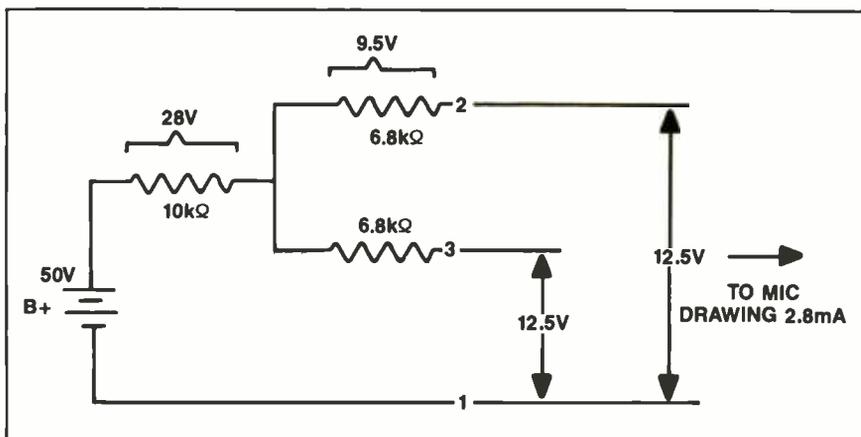


Figure 3. Phantom supply with an extra current-limiting resistor.

a high-impedance input). Now plug in a condenser mic with high current drain. The phantom voltage drops to, say, 13V. This is extreme, but it can happen. The microphone still works, but barely.

Next "Y" or parallel the mic output to feed a house console and a monitor console. The phantom voltage might drop to 5V, so the mic signal either dies or slowly fades out.

Explanation

Here's what's going on: In a typical simplex phantom-power supply (see Figure 1), a B+ voltage (48V, for example) is applied through two equal resistors to pins 2 and 3. Inside the microphone, two equal resistors in series are across pins 2 and 3. B+ for the microphone circuitry is taken off the center tap of the two equal resistors.

The power-supply resistors must be high enough in value to avoid loading down the microphone and isolating several microphones from each other in case one mic cable shorts the supply. These resistors also must be low enough in value so that when the mic drains current through them, they don't drop the phantom-supply

voltage excessively.

The voltage drop that occurs when you plug in a mic is given by Ohm's law: $E = IR$, where:

E = voltage drop in volts.

I = current in $R = \frac{1}{2}$ current drain of microphone in amperes.

R = resistance of each resistor R in ohms.

(See Figure 2 for an example.)

Suppose the supply is 50V, and the microphone draws 2.8mA. (Most condenser microphones draw from 0.4mA to 4mA.) Half that current flows through resistor R . So the voltage drop is $E = IR = (0.0014) \times (6800) = 9.5V$. The 50V supply drops to 40.5V when the mic is plugged in. That's adequate to power the microphone.

But now suppose we use the phantom-power supply shown in Figure 3. An extra 10kΩ resistor has been added to prevent pops when the mic is plugged in by allowing the phantom voltage to ramp up slowly. Unfortunately, the 10kΩ resistor drops 28V because 2.8mA is flowing through it. Plus, each 6.8kΩ resistor loses 9.5V. The total loss is 37.5V. $50V - 37.5V = 12.5V$. That 10kΩ resistor is a problem.

Only 12.5V appears at the microphone,

and this is barely enough to power it. Indeed, with some mics it is not enough. The microphone's headroom is reduced, so it distorts at a lower sound-pressure level. If you "Y" or split this mic to feed two consoles, the consoles might load down each other and reduce the phantom voltage to as little as 5V! With such little voltage applied, the mic will be distorted, low-level or dead.

Here's how the console not supplying phantom might load down the console supplying phantom: Some consoles contain a large-value capacitor across the phantom supply to filter out noise and improve short-term regulation. (See Figure 4.) Some engineers add a bleeder resistor, which can be switched across the capacitor when phantom is turned off. (See Figure 5.) This is done with the belief that any phantom voltage remaining after turn-off might damage a dynamic mic plugged into the input. But it won't, so there's no need for that bleeder resistor.

Solutions

There are several ways to prevent phantom-supply voltage loss (sag or deregulation) and the consequent microphone distortion or signal loss:

- If there's a resistor between B+ and the two equal resistors that go to mic-connector pins 2 and 3 in the console, remove (short) that resistor. The microphone may pop when it's plugged in, but at least it will work.

- If splitting a mic to feed two consoles results in severe deregulation, either: (1) use a transformer-isolated mic splitter (this costs a little more, but is effective and reduces signal loading), (2) supply phantom from the console that is better-regulated, or (3) if there is a bleeder resistor across the B+ filter capacitor, remove it (if you must have it, use a high value, say, 100kΩ).

Phantom power supply design

If you're adding phantom power to your own console, you can apply phantom power through two equal resistors to pins 2 and 3. These must be matched within 1%. What value resistors should you use? Table 1 answers this question.

Picking off a phantom voltage from the 15V circuit supply in the console causes problems. Instead, find a power-transformer winding at about 48V and filter it. Or, consider a high-quality, stand-alone phantom supply.

[Note, if you attempt to install phantom power to your own console while it is under warranty, you will very likely void that warranty.]

Needed: a new microphone specification

To prevent the phantom-loading prob-

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An Automatic Switchover for Redundant Power Supplies

By John Monforte

A simple automatic power supply switchover can be used to prevent even a momentary loss of signal. This circuit also indicates when the switching has occurred by illuminating a lamp or LED. Using this circuit, one of the power supplies may be turned off and removed for service, while keeping the devices "on air" at all times. This can be used for the phantom supply or any other power supply in a system.

Figure 1 shows the two positive voltage supplies where each is connected to diodes that are rated for the maximum current the supplies

can deliver. Also, the breakdown voltage rating of the diodes should exceed the supply voltage. For a phantom power supply, 1N4002 diodes are adequate. Most of the current is delivered by the supply that has the higher output voltage or has the diode with the smaller forward voltage drop. Since silicon diodes drop about 0.6V, the supplies can be adjusted to a slightly higher voltage to compensate. Just turn off the other supply and adjust for the proper voltage at the load.

A light bulb rated for the supply voltage is wired across the supplies' outputs. When both are off or both are on, the lamp is off. If only one supply is on, the lamp lights, indicating a failure. In place of the

lamp, a bidirectional two-color LED, or two LEDs wired in parallel but in opposite directions, can be connected here through an appropriate current-limiting resistor, as shown in Figure 2. Now the color of the LED will tell which supply is faulty. To calculate the resistance, divide the supply voltage by 20mA.

Bipolar supplies can be set up with more of these circuits or a bridge arrangement, using a single indicator lamp, can be used, as in Figure 3. The switchover diodes are designed to the current and voltage specifications as before. The four diodes in the bridge can be 1N4001 types. Failure of either voltage in either supply unbalances the bridge and lights the LED.

John Monforte is RE/P's technical editor and director of Recording Services at the University of Miami.

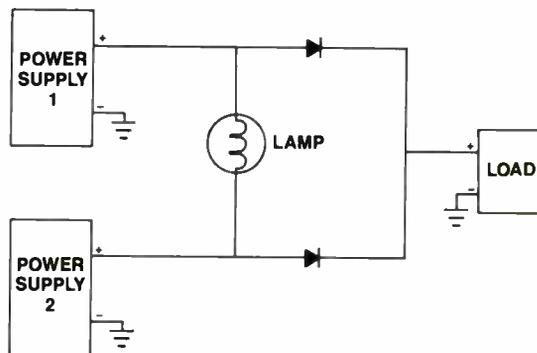


Figure 1. A simple automatic power supply switchover circuit.

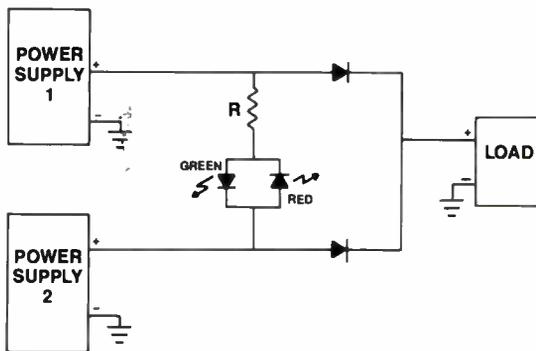


Figure 2. A simple automatic power supply switchover circuit with two LED's wired in parallel but in opposite directions.

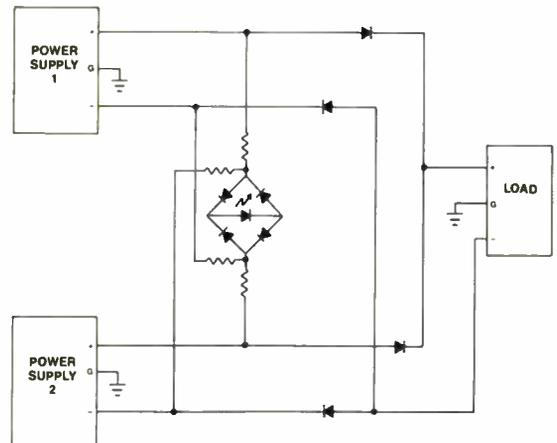


Figure 3. An automatic, bipolar supply switchover circuit.

lem, console designers need to know how much and how little voltage a microphone can work with. Most microphone data sheets state how much phantom voltage the mic requires (typically 12V to 48V). But they are unclear about whether this refers to the unloaded phantom voltage the console supplies or the voltage the mic expects to see when it is plugged in. Ideally, both would be specified. You can calculate the approximate voltage sag for your particular console and microphone by using the microphone current-drain spec as described earlier. Note: The current drain decreases slightly from the specified value as the loaded phantom voltage decreases, and vice versa. The microphone data sheet also should state how low the supply voltage can go without changing the phantom-supply resistors.

Other precautions

In addition to watching for supply sag, you might want to consider these suggestions:

- Don't plug a mic into an input with phantom already switched on, or you'll get a loud pop. If you have no choice (for example, during a live concert), try to have the mic's fader down when you plug it in.

- Have a spare 48V supply in case the main supply goes down. In the circuit shown in the sidebar, "An Automatic Switchover for Redundant Power Supplies" by John Monforte, a second supply takes over if the primary one fails, and a warning light comes on.

- Avoid running phantom in a patchbay, because someone is likely to patch in, causing a pop. If you must patch into a jack with phantom on it, mute the input module to which the mic is connected or turn down its fader. Sherman Keene, author of "Practical Techniques for the Recording Engineer," gives this advice: When making a mic-level patch, first plug into the mic pre-amp input jack, then plug into the mic out jack. When breaking a patch, first pull out the mic out plug, then pull out the pre-amp in plug.

Note: Mic-level patches should be avoided. The connectors are not rated for those signals and give substandard performance. Also, crosstalk with nearby line-level signals can result in feedback.

- Some phantom supplies cause a hum when you plug in a connector that ties the shell to ground. Float the shell. This is a good idea anyway to prevent ground loops.

- Since the cable shield carries the dc return, be sure the shield and solder connections are secure. Otherwise you can expect crackling noises, especially when the cable is moved.

Table 1. Appropriate resistor values for various voltages, based on DIN 45 596. Caution: 680Ω resistors may load down a microphone excessively.

Supply voltage	12	24	48
Recommended R for two-resistor configuration	680Ω	1.2kΩ	6.8kΩ

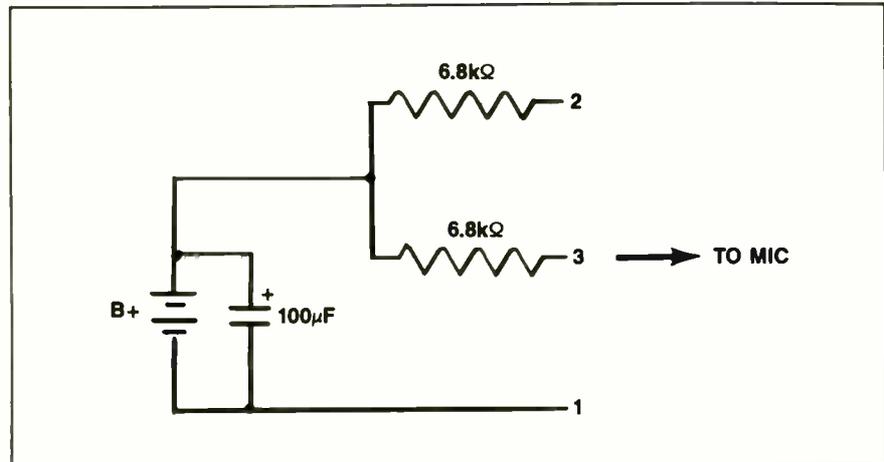


Figure 4. Capacitor added to phantom supply.

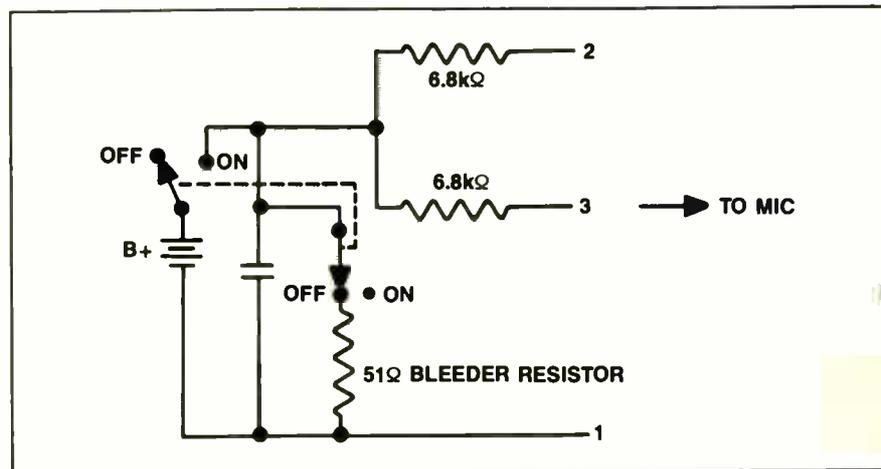


Figure 5. Bleeder resistor added across capacitor in phantom off position.

- Power supplies are rated in the total number of milliamps they can supply. Make sure that the total current drain of all the mics plugged into the supply doesn't exceed the supply's current rating.

- Some microphones work either on internal batteries or external phantom power. In most designs, connecting the mic to phantom automatically removes the battery from the circuit. Otherwise, the battery would severely load down the phantom supply. If this appears to be happening, remove the battery.

In case of mic failure caused by power-supply sag, don't guess what the problem is. Measure the voltages to determine

where you are and what solution is best in your situation.

If a condenser microphone doesn't work due to low phantom-supply voltage after the mic is plugged in, try these suggestions:

- Supply phantom from a better-regulated console.

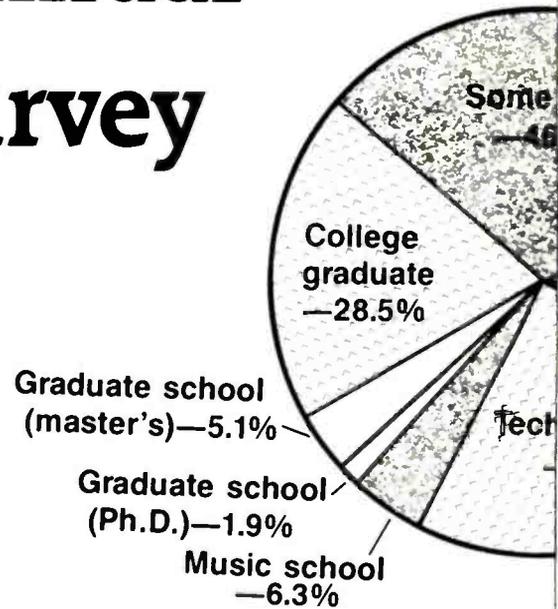
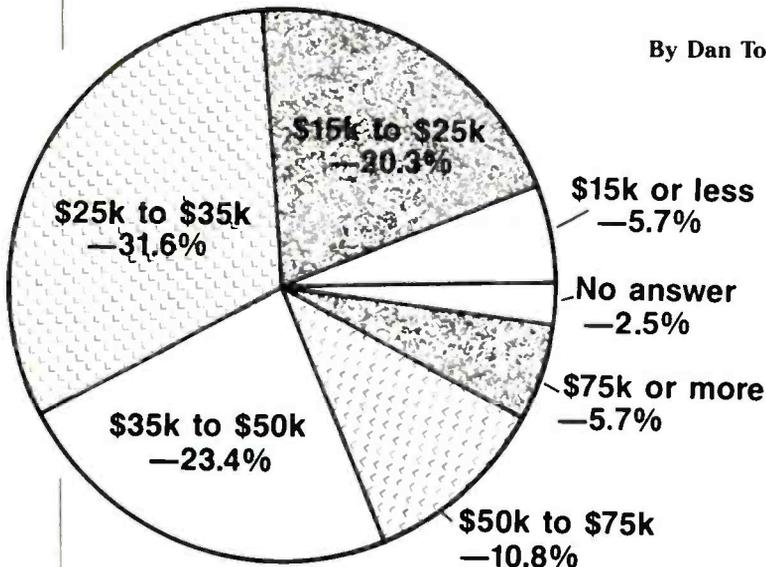
- Redesign the phantom supply for lower loss as described in this article.

REP

Acknowledgments: Many thanks to Gene Clair and Ron Barthaick of Clair Brothers and John Monforte for their help and suggestions.

Second Annual Salary Survey

By Dan Torchia



Although benefits were down, salaries increased for most job categories in the pro audio industry, according to *RE/P's* exclusive reader survey.

Look, we know that you didn't go into the pro audio industry for the money. If you really cared about money, you would have gone to business school, like your parents wanted you to. Or you would have gone to truck driver's school after years of waiting for your big break, like tracking the latest Primordial Ooze album, or perhaps posting "Son of the Return of the Saginaw Slasher."

But let's just say, for the sake of argument, that money matters just a little bit to you. How do you think your salary compares to the guy across town? What about across the country? Let's get to deeper questions. How much are you worth? And are you getting paid that?

Salary information is pretty hard to come by in an industry as diverse as pro audio, which makes it hard to get a definitive handle on such basic items as how much to ask for in a basic salary or how much you should be paid in relation to your experience and where you live.

But such information is invaluable. The only way to know if you're near your earning potential is to compare yourself to your industry peers. Although it's impossible to draw definitive conclusions across the entire industry, some general trends and observations can be made.

Hence the purpose of *RE/P's* second annual salary survey, a polling of our readership concerning salaries and related information in our industry job categories. Because this is the second time we've polled readers, we now can compare this year's data with last year's. We make no claims for industry-wide trends, but this random-sample survey gives some insight into you and your fellow readers.

Here are some of the results:

- Generally, most readers reported that salaries increased from last year.

- While salaries were up, most types of benefits were down in many job categories. The number of people reporting that they received no benefits at all also increased.

- The work force has remained stable, both in terms of the number of years in

the present job and number of years in the industry.

- The average age of the work force has essentially remained the same or has increased only slightly.

Survey basics

This scientific survey was conducted by Katy Smith, director of market research for Intertec Publishing, *RE/P's* parent company. Names were selected from *RE/P's* circulation list on an *n*th, or random basis. A total of 1,000 questionnaires were sent out, and 336 were returned, for a response rate of about 33%.

All individual job classifications were broken down into three main categories:

1. *Management*: owner, president/vice president, general manager, studio manager and acoustic consultant.

2. *Technical*: chief engineer and maintenance engineer.

3. *Production*: audio engineer, mastering engineer, live sound engineer, post production engineer, engineer/producer, arranger/producer and programmer.

Within these categories, individual jobs

Dan Torchia is *RE/P's* staff editor.

have been broken down into sub-categories that describe the types of audio production facility or business at which they work:

1. Recording studio: either fixed or mobile.
2. Live sound.
3. Broadcast audio production: either radio or TV broadcast audio production.
4. Post production: audio post-production/video sweetening, film sound, tape duplication and disc/CD mastering.
5. Corporate, educational and government.
6. Engineering and/or production company.
7. Acoustic/technical consulting.
8. Dealer/distributor, rental.

The sidebar "Survey Questions" details the type of questions we asked. The results of the survey are detailed in Tables 1, 2 and 3, corresponding to management, technical and production categories.

1988, 1987 comparisons

Tables 4 through 7 detail median response information for all job categories. Tables 4 through 6 are individual comparisons between 1987 and 1988, while table 7 compares median totals between the three main job categories.

Management respondents' median salary for 1988 was \$36,700. This represented a \$2,450 increase over 1987's median salary of \$34,250, or an increase of about 7%. The highest median salary within the individual job classifications were for post-production and dealer/distributor, each at \$43,800. The lowest was live sound, at \$32,200.

The average management respondent has spent about 7.5 years in his present job, a slight decrease from the 8 years averaged in the 1987 survey. Number of years in the industry also decreased slightly, from 16.6 in 1987 to 15.3 this year. Average age also dropped slightly, from 38.7 to 38.

Technical engineers received the smallest median salary increase, amounting to only \$250, from \$31,400 last year to \$31,650 this year. This amounted to a 0.7% increase. Techs working at post-production facilities scored the highest median salary, \$50,000. Lowest was at recording studios, \$28,350.

Years in the present job and years in the industry was essentially unchanged. Median age increased slightly, from 36.6 in 1987 to 36.9 in 1988.

Production staffers reported the biggest salary increase. Median salary increased 15%, from \$27,500 in 1987 to \$31,850 in 1988. The acoustical/technical consulting category had the highest median salary, at \$42,500. Lowest was broadcast audio

production, at \$26,400. Number of years in the job and in the industry increased slightly, and the average age increased about a year, from 33.2 last year to 34.3 this year.

The highest median salary among all job categories was technical staff at post-production facilities (\$50,000), while the lowest was production staff at broadcast audio facilities (\$26,400).

Benefits

Although the decrease in benefits varies between individual items, the overall benefit package was less attractive than last year's. And more respondents in all three main categories reported this year that they received no benefits at all. This ranged from a high of 29.8% of management respondents, an increase of almost 10 percentage points from last year, to low of 4.4% of technical respondents, an increase of about 1.5 percentage points. A total of 18% of production respondents reported receiving no benefits at all, an increase of 2.6 percentage points from last year.

Overall, most respondents receive basics such as insurance and vacation, but those receiving vacations and holidays dropped. Fewer respondents reported participating in such benefits as stock options or profit sharing.

Reader profiles

RE/P also included extra questions to get an accurate perception of audio professionals. Although these are secondary to the basic salary questions, they were extremely helpful in getting a clear picture of personnel working in the industry.

Some of these data's highlights:

- An overwhelming majority of respondents learned or received their initial audio experience on the job. (See Table 8.) This ranged from about 70% of management and technical respondents to 62.5% for production respondents. About 25% to 30% of respondents received experience from a college or university. Recording schools scored the lowest among the three main choices. More management respondents reported receiving initial experience at a recording school, 6.8%, while technical and production respondents totaled 3.8%.

- Respondents indicated that industrial/corporate and commercial production were their top income source. (See Table 9.) Record and demo work scored somewhere in the middle. Mastering and duplication received the lowest scores across all three main categories. A significant percentage of respondents did not list one of five main categories, but instead chose "other," indicating that facilities are indeed

Survey Questions

1. Job title.
2. Staff or independent contractor.
3. Primary type of business.
4. Top income source.
5. Years in present job.
6. Years in the audio industry.
7. Salary level.
8. Average hourly rate if an independent contractor.
9. Fringe benefits received.
10. Union membership.
11. How initial audio experience was acquired.
12. For owners/managers, years of engineering, production and technical experience.
13. Education level.
14. Age.
15. Membership in professional societies.
16. Trade show attendance.
17. State in which business is located.

diversifying into non-traditional client bases.

- The majority of respondents hold staff positions, totaling 54.8% for management, 88.6% for technical and 73.6% for production.

- Respondents who said they were independent contractors quoted hourly rates were \$46.85 for management, \$32.75 for technical and \$23.95 for production.

- Across the three main classifications, most respondents reported earning between \$25,000 and \$35,000 a year. (See Figures 1, 3 and 5.) Median salary figures for all job categories are well above government figures for the general population, for which the average annual pay is \$19,966.

- The industry is also highly educated as compared to the general population. (See Figures 2, 4 and 6.) The percentage of respondents with a bachelor's degree totaled 36.9% for management, 28.5% for technical and 48.3% for production. According to government data, 27.1% of the U.S. population has some college, while 7.2% have a college degree or beyond.

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TABLE 1. — MANAGEMENT STAFF PROFILE

Management	Total %	Recording Studio %	Live Sound %	Broadcast %	Post-Production %	Corp./Ed/Govt. %	Eng./Prod. %	Acoustical Tech. %	Dealers/Distr. %
Job Title									
Owner	52.7	70.2	69.4	42.9	60.7	15.6	72.7	45.9	25.6
President/VP	15.5	9.5	18.4	14.3	25.0	6.7	27.3	18.9	15.4
General Manager	19.0	13.1	14.3	36.1	14.3	22.2	—	2.7	59.0
Studio Manager	14.3	16.7	2.0	14.3	10.7	55.6	3.0	—	2.6
Acoustic Consultant	6.8	2.4	6.1	—	—	2.2	—	43.2	2.6
Job									
Staff	54.8	54.8	46.9	71.4	50.0	82.2	21.2	32.4	76.9
Independent Contractor	42.3	40.5	53.1	19.0	42.9	17.8	81.8	70.3	12.8
No Answer	5.4	7.1	4.1	9.5	7.1	4.4	—	—	10.3
Type of Business									
Recording Studio									
Recording Studio (fixed)	22.6	88.1	—	—	—	—	—	—	—
Recording Studio (mobile)	4.2	15.5	—	—	—	—	—	—	—
Live Sound									
Sound Reinforcement	19.0	—	100.00	—	—	—	—	—	—
Broadcast Audio Production									
Radio Broadcast Production	7.1	—	—	52.4	—	—	—	—	—
TV Broadcast Audio Production	5.7	—	—	52.4	—	—	—	—	—
Post-Production									
Audio Post-Prod./Video									
Sweetening	10.7	—	—	—	53.6	—	—	—	—
Film Sound	4.5	—	—	—	21.4	—	—	—	—
Tape Duplication	7.1	—	—	—	25.0	—	—	—	—
Disc/CD Mastering	0.9	—	—	—	3.6	—	—	—	—
Corp./Ed/Govt.									
Corp. Audio Prod.	8.6	—	—	—	—	31.1	—	—	—
Educational	12.5	—	—	—	—	66.7	—	—	—
Government	1.2	—	—	—	—	2.2	—	—	—
Eng. and/or Production									
Eng. and/or Prod. Company	16.4	—	—	—	—	—	100.0	—	—
Acoustic/Tech. Consulting									
Audio Acoustic Consulting	8.6	—	—	—	—	—	—	51.4	—
Audio Technical Consulting	12.8	—	—	—	—	—	—	67.6	—
Dealer/Distr./Rental									
Dealer/Distr./Mfr.	14.3	—	—	—	—	—	—	—	89.7
Equipment Rental	4.5	—	—	—	—	—	—	—	10.3
Top Income Source									
Records/Demos	14.0	39.3	4.1	—	3.6	2.2	24.2	5.4	—
Commercial Prod.	11.9	20.2	6.1	47.6	14.3	2.2	9.1	2.7	2.6
Corp./Industrial	22.6	14.3	18.4	9.5	21.4	40.0	45.5	21.6	15.4
Post-Prod.	9.2	10.7	—	14.3	32.1	2.2	6.1	13.5	5.1
Mastering/Duplication	7.1	14.3	—	—	25.0	6.7	3.0	—	2.6
Other	35.7	13.1	65.3	33.3	3.6	42.3	12.1	54.1	66.6
No Answer	4.2	2.4	10.2	—	3.6	4.4	—	2.7	7.7
Years at Present Job									
Less than 3	19.4	16.7	14.3	33.3	14.3	8.9	21.2	21.7	35.8
3 or 4	16.4	14.3	18.4	9.5	17.9	11.1	21.2	18.9	20.5
5 to 9	28.0	33.3	32.6	42.9	25.0	33.3	18.2	18.9	15.4
10 to 14	17.3	14.3	14.3	—	21.4	26.7	21.2	21.6	15.4
15 or More	18.0	21.4	18.4	14.3	21.4	17.8	18.2	18.9	10.3
No Answer	0.9	—	2.0	—	—	2.2	—	—	2.6
Median	7.5	7.9	7.5	5.9	8.6	9.4	7.1	7.5	4.3
Years in Professional Audio Industry									
Less than 5	4.8	6.0	6.1	4.8	10.7	—	3.2	5.4	2.6
5 to 9	16.7	25.0	14.3	9.5	10.7	17.8	6.1	13.5	20.5
10 to 14	26.2	21.4	28.6	23.8	25.0	15.6	42.2	32.5	28.2
15 to 24	36.2	33.3	40.8	47.6	28.6	46.6	39.4	27.0	30.8
25 or more	13.7	13.1	8.2	9.5	21.4	15.6	9.1	21.6	12.8
No Answer	2.4	1.2	2.0	4.8	3.6	4.4	—	—	5.1
Median	15.3	14.3	15.0	16.3	15.5	17.1	14.8	14.8	14.3

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TABLE 1. — MANAGEMENT STAFF PROFILE . . . continued

Management	Total %	Recording Studio %	Live Sound %	Broadcast %	Post-Production %	Corp/Ed/ Govt. %	Eng./Prod. %	Acoustical Tech. %	Dealers/ Distr. %
Salary Level									
Less than \$15,000	6.5	14.3	8.2	9.5	3.6	--	3.0	5.4	10.3
\$15,000 to \$24,999	17.6	23.8	18.4	14.3	14.2	15.6	9.1	8.1	15.4
\$25,000 to \$34,999	19.6	14.3	32.7	33.3	10.7	26.7	27.3	13.5	10.3
\$35,000 to \$49,999	26.5	21.4	24.4	23.8	25.0	35.5	21.2	37.8	20.5
\$50,000 to \$74,999	14.6	9.5	12.2	4.8	17.9	15.6	15.2	13.5	30.7
\$75,000 or more	12.2	15.5	4.1	4.8	17.9	4.4	21.2	16.2	12.8
No Answer	3.0	1.2	--	9.5	10.7	2.2	3.0	5.5	--
Median	\$36,700	\$32,900	\$32,200	\$34,500	\$43,800	\$38,150	\$38,000	\$40,700	\$43,800
Independent Contractor	42.3	40.5	53.1	19.0	42.9	17.8	81.8	70.3	12.8
Average Hourly Rate									
Less than \$25	6.3	4.8	16.3	4.8	7.1	--	9.1	8.1	--
\$25 to \$49	13.7	15.5	14.4	14.2	10.7	2.2	24.2	21.6	7.6
\$50 to \$74	8.0	9.5	6.1	--	3.6	2.2	21.2	16.2	2.6
\$75 or More	8.3	9.5	4.1	--	14.4	4.4	12.1	21.6	--
No Answer	6.0	1.2	12.2	--	7.1	9.0	15.2	2.8	2.6
Median	\$46.85	\$49.10	\$28.35	N/A	\$50.00	N/A	\$50.00	\$56.30	N/A
Fringe Benefits Received									
Med. Ins. (Paid)	58.6	39.3	49.0	66.7	60.7	86.7	57.6	51.4	84.7
Dental Insur. (Paid)	34.8	22.6	22.4	38.1	32.1	62.2	33.3	27.0	53.8
Vision (Eyecare) Ins. (Paid)	19.3	19.0	12.2	14.3	17.9	42.2	15.2	2.7	25.6
Short-Term Disability Plan	22.9	11.9	14.3	23.8	17.9	44.4	18.2	29.7	33.3
Long-Term Disability Plan	29.2	21.4	18.4	23.8	14.3	64.4	30.3	21.6	38.5
Sick Leave	41.1	23.8	24.5	61.9	46.4	77.8	24.2	32.4	64.1
Vacation	51.8	38.1	40.7	61.9	50.0	82.2	33.4	37.8	84.7
1 week	10.1	8.3	12.2	9.5	14.3	11.1	3.0	8.1	15.4
2 weeks	16.4	10.7	18.4	23.8	14.3	11.1	15.2	8.1	38.5
3 weeks	12.2	9.5	6.1	14.3	10.7	22.2	6.1	10.8	20.5
4 weeks	8.0	4.8	2.0	9.5	10.7	22.2	6.1	5.4	7.7
5 or More	5.1	4.8	2.0	4.8	--	15.6	3.0	5.4	2.6
Median	2.6	3.0	2.4	2.9	2.8	3.9	2.9	3.3	2.8
Holidays	47.3	27.4	32.7	61.9	50.0	82.2	33.3	37.8	79.5
Personal Business Days	20.2	15.5	14.3	14.3	14.3	44.4	18.2	13.5	25.6
Stock Purchase Plan	6.8	7.1	2.0	--	3.6	22.2	3.0	8.1	2.6
Profit-sharing Plan	18.5	14.3	10.2	33.3	14.3	17.8	9.1	21.6	38.5
Savings Plan	10.4	1.2	8.2	4.8	7.1	33.3	6.1	5.4	20.5
Pension Plan	25.9	14.3	10.2	23.8	10.7	80.0	15.2	21.6	33.3
Comp. Deferment Plan	8.0	3.6	2.0	--	3.6	20.0	6.1	8.1	20.5
Bonus	17.0	13.1	14.3	19.0	17.9	13.3	18.2	13.5	33.3
Assoc. Membership (Paid)	21.7	8.3	18.4	4.8	21.4	40.0	21.2	35.1	30.8
Trade Shows, Etc.	36.0	17.9	32.7	23.8	46.4	57.8	36.4	32.4	59.0
Tuition Refund Plan	14.3	8.3	4.1	--	7.1	51.1	3.0	13.5	20.5
Automobile Furnished	19.9	13.1	22.4	19.0	14.3	13.3	39.4	21.6	25.6
None	29.8	46.3	34.7	14.3	35.7	6.7	36.4	37.8	5.1
Company Unionized	8.9	4.8	10.2	14.3	10.7	17.8	6.1	2.7	10.3
Reader a Union Member	18.8	14.3	20.4	19.1	25.0	20.0	27.3	13.5	18.0
Union									
IATSE	6.3	3.6	14.3	9.5	14.3	--	--	5.4	7.7
AFM	5.7	8.3	4.1	4.8	7.1	4.4	6.1	--	7.7
AFTRA	2.7	3.6	--	--	--	4.4	12.1	--	--
IBEW	1.5	--	--	--	--	--	6.1	8.1	--
NABET	1.2	--	--	4.8	3.6	--	6.1	--	--
SAG	1.2	2.4	--	--	3.6	--	3.0	--	--
Other	3.0	--	2.0	--	--	13.2	6.1	--	2.6
How Initial Audio Experience Was Acquired									
On the Job	70.2	72.6	91.8	61.9	60.7	51.1	81.8	56.8	74.4
College/University Program	29.8	26.2	18.4	28.6	35.7	35.6	21.2	51.4	28.2
Recording School	6.3	11.9	6.1	--	7.1	6.7	--	--	7.7
Other	6.8	3.6	6.1	9.5	10.7	11.1	9.1	5.4	5.1
No Answer	2.4	4.8	--	--	3.6	4.4	3.0	--	--
Education									
High School	11.0	9.5	16.3	14.3	14.3	8.9	6.1	5.4	15.4
Some College	33.3	38.1	49.0	42.9	35.7	11.1	36.4	24.3	28.2
College Grad (Bachelor's Degree)	36.9	35.7	30.6	28.6	25.0	37.8	54.5	32.4	48.7
Grad School (Master's)	12.2	13.1	2.0	9.5	14.3	24.4	--	24.3	7.7
Grad School (Ph.D.)	4.8	3.6	2.0	4.8	--	15.6	--	10.8	--
Music School	12.2	16.7	12.2	9.5	10.7	15.6	12.1	2.7	10.3
Technical School	19.0	19.0	18.4	19.0	25.0	17.8	27.3	18.9	10.3
Armed Services School	6.0	2.4	8.2	4.8	14.3	6.7	6.1	5.4	5.1
No Answer	0.6	--	--	--	3.6	--	--	--	2.6

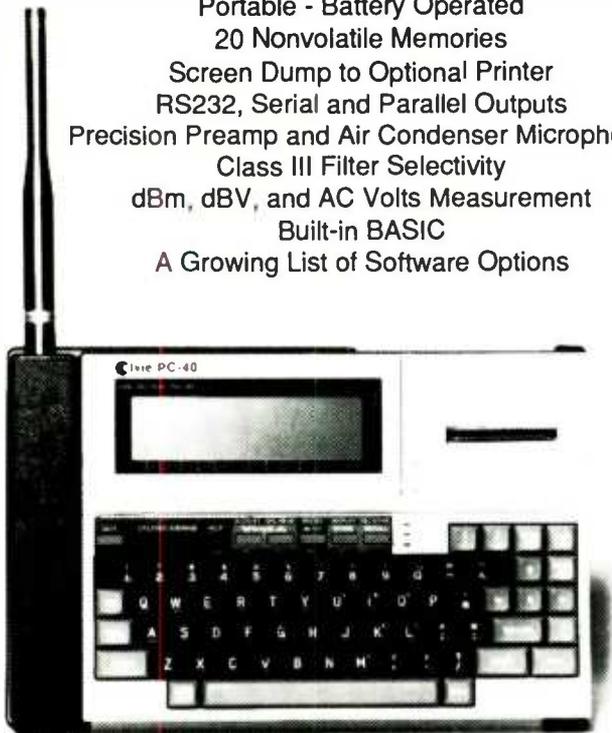
TABLE 1. — MANAGEMENT STAFF PROFILE . . . continued

Management	Total %	Recording Studio %	Live Sound %	Broadcast %	Post-Production %	Comp/Ed/ Govt. %	Eng./Prod. %	Acoustical Tech. %	Dealers/ Distr. %
Age									
Under 25	0.6	--	--	4.8	--	--	--	--	2.6
25 to 34	36.0	38.1	49.0	33.3	28.6	22.2	30.3	40.6	38.5
35 to 44	42.8	42.8	38.8	47.6	28.6	49.0	54.5	35.1	46.1
45 to 54	11.9	9.5	8.2	9.5	35.6	13.3	9.1	13.5	5.1
55 to 64	6.0	6.0	2.0	4.8	3.6	8.9	6.1	10.8	5.1
65 or Over	2.1	2.4	2.0	--	3.6	4.4	--	--	2.6
No Answer	0.6	1.2	--	--	--	2.2	--	--	--
Median	38.1	37.6	35.3	37.5	42.5	40.5	38.6	37.7	36.9
Societies/Organizations									
Hold Membership									
AES	35.4	23.8	34.7	4.8	17.9	44.4	27.3	75.7	48.7
SMPTE	13.1	11.9	4.1	9.5	14.3	17.8	15.2	24.3	10.3
NAB	11.3	8.3	8.2	38.1	--	11.1	12.1	5.4	20.5
NAMM	7.1	10.7	10.2	--	--	--	3.0	5.4	17.9
NSCA	3.9	2.4	--	--	3.6	--	3.0	10.8	12.8
SPARS	3.3	3.6	4.1	--	--	4.4	--	5.4	5.1
APRS	0.9	1.2	--	--	3.6	--	--	--	2.6
IMA	0.6	1.2	--	--	3.6	--	--	--	--
MMA	0.6	1.2	--	--	--	--	3.0	--	--
Other	19.9	4.8	8.2	23.8	17.9	24.4	36.4	54.1	15.4
None	42.9	54.8	53.1	52.4	60.7	28.9	30.3	18.9	35.9
Trade Shows Attended									
AES — United States	50.0	44.0	63.3	9.5	35.7	46.7	42.4	70.3	69.2
NAB	30.1	16.7	18.4	47.6	39.3	28.9	39.4	27.0	53.8
SMPTE	19.6	15.5	4.1	14.3	35.7	20.0	27.3	18.9	33.3
NAMM — Summer	15.8	16.7	26.8	--	10.7	6.7	6.1	13.5	30.8
NAMM — Winter	14.0	15.5	18.4	--	3.6	6.7	9.1	13.5	33.3
NSCA	7.7	1.2	8.2	--	7.1	2.2	--	27.0	20.5
AES — Europe	2.4	2.4	--	--	3.6	4.4	3.0	--	5.1
APRS	1.2	1.2	--	--	--	2.2	3.0	--	2.6
Other	8.0	3.6	4.1	9.5	14.3	8.9	3.0	8.1	20.5
None	27.7	36.9	26.5	47.6	21.4	28.9	24.2	21.6	10.3

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TABLE 1. — MANAGEMENT STAFF PROFILE . . . continued

Management	Total %	Recording Studio %	Live Sound %	Broadcast %	Post-Production %	Corp/Ed/ Govt. %	Eng./Prod. %	Acoustical Tech. %	Dealers/ Distr. %
Geographical Location									
New England	5.4	2.4	8.2	9.5	3.6	17.7	--	--	2.6
Middle Atlantic	18.6	22.6	20.5	19.0	17.9	11.1	12.1	16.2	23.1
East N. Central	14.0	13.1	16.3	14.3	3.6	17.8	18.2	8.1	17.9
West N. Central	5.4	2.4	8.2	14.3	10.7	6.7	3.0	2.7	2.6
South Atlantic	14.3	19.0	16.3	23.8	10.7	8.9	15.2	13.5	5.1
East S. Central	4.5	3.6	2.0	9.5	3.6	6.7	6.1	2.7	5.1
West S. Central	9.5	9.5	6.1	4.8	10.7	11.1	12.1	16.2	5.1
Mountain	4.8	4.8	6.1	--	--	8.9	3.0	2.7	7.7
Pacific	23.2	21.4	16.3	4.8	39.2	11.1	30.3	35.2	30.8
No Answer	0.3	1.2	--	--	--	--	--	2.7	--
Engineering Experience									
Less than 5 Years	58.7	72.5	69.4	38.1	53.5	28.9	78.8	67.5	35.9
5 to 9	5.7	8.3	6.1	--	7.1	--	9.1	5.4	5.1
10 to 14	12.5	20.2	16.3	4.8	10.7	2.2	21.2	13.5	--
15 to 24	15.2	16.7	18.4	4.8	7.1	8.9	21.2	27.0	10.3
25 or More	17.9	20.2	22.5	19.0	25.0	11.1	18.2	13.5	12.8
No Answer	6.6	7.1	6.1	9.5	3.6	6.7	9.1	8.1	2.6
Median	0.6	--	--	--	--	--	--	--	5.1
Median	13.6	12.3	13.4	18.4	15.9	18.8	12.2	12.8	15.0
Production Experience									
Less than 5 Years	54.9	64.3	42.9	81.0	67.9	44.5	84.9	43.2	23.1
5 to 9	6.6	3.6	6.1	4.8	10.7	2.2	18.2	10.8	2.6
10 to 14	11.0	15.5	14.3	14.3	14.3	4.4	15.2	8.1	--
15 to 24	15.5	20.2	4.1	23.8	17.9	15.6	24.2	16.2	5.1
25 or More	15.5	16.7	14.3	33.3	14.3	15.6	18.2	8.1	10.3
No Answer	5.7	8.3	4.1	4.8	10.7	6.7	9.1	--	--
Median	0.6	--	--	--	--	--	--	--	5.1
Median	13.1	13.3	11.3	14.5	12.5	15.0	11.9	10.9	N/A
Technical Experience									
Less than 5 Years	48.2	48.8	51.1	43.0	57.2	28.9	63.7	64.8	33.3
5 to 9	4.2	4.8	8.2	--	3.6	--	6.1	2.7	5.1
10 to 14	7.7	10.7	8.2	4.8	3.6	2.2	6.1	16.2	5.1
15 to 24	10.1	8.3	6.1	19.1	14.3	4.4	21.2	13.5	5.1
25 or More	17.6	17.9	18.4	14.3	25.0	15.6	21.2	18.9	10.3
No Answer	8.0	7.1	10.2	4.8	10.7	6.7	9.1	13.5	2.6
Median	0.6	--	--	--	--	--	--	--	5.1
Median	15.9	15.3	16.5	16.3	17.5	19.4	14.7	15.0	15.0

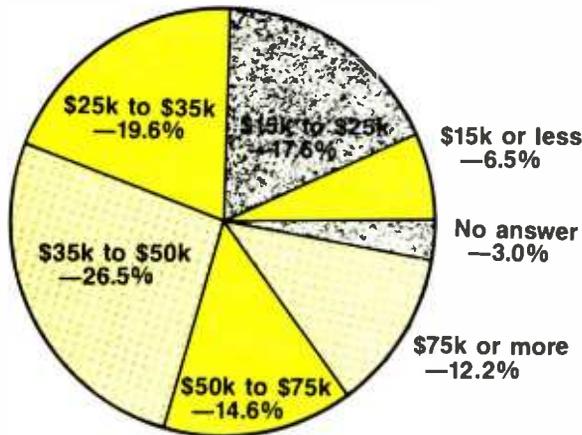


Figure 1. Salary level for management staff.

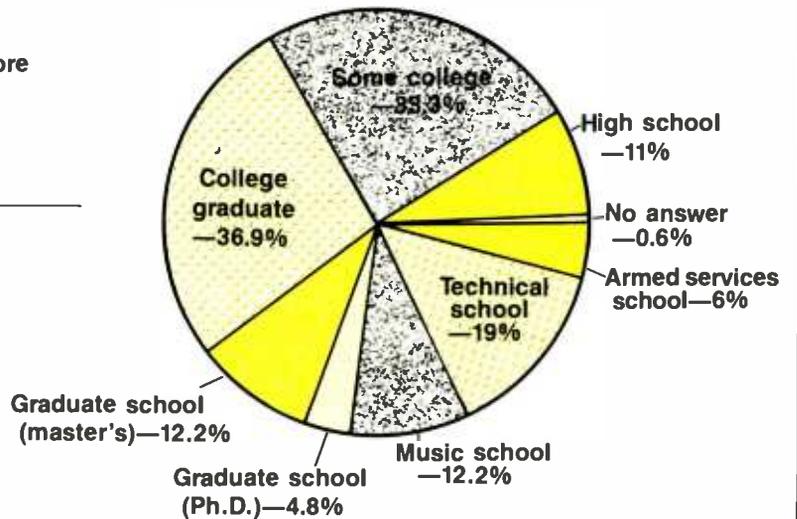


Figure 2. Education level for management staff.

TABLE 2. — TECHNICAL STAFF PROFILE

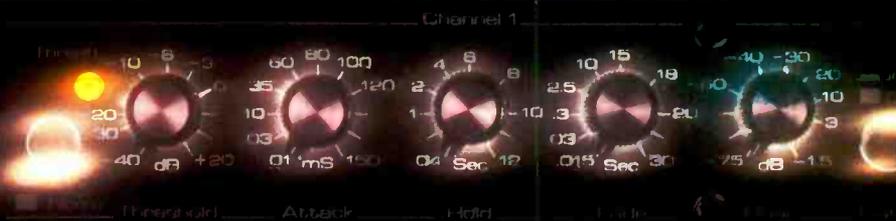
Technical	Total %	Recording Studio %	Live Sound %	Broadcast %	Post-Production %	Corp/Ed/ Govt. %	Eng./Prod. %	Acoustical Tech. %	Dealers/ Distr. %
Job Title									
Chief Engineer	65.2	76.0	--	69.2	50.0	64.0	--	--	33.6
Maintenance Engineer	34.8	24.0	--	30.8	50.0	36.0	--	--	66.7
Job									
Staff	88.6	84.0	--	96.2	100.0	92.0	--	--	100.0
Independent Contractor	12.0	28.0	--	3.8	--	8.0	--	--	--
Type of Business									
Recording Studio									
Recording Studio (fixed)	14.6	92.0	--	--	--	--	--	--	--
Recording Studio (mobile)	1.9	8.0	--	--	--	--	--	--	--
Live Sound									
Sound Reinforcement	6.3	--	--	--	--	--	--	--	--
Broadcast Audio Production									
Radio Broadcast Production	20.3	--	--	55.8	--	--	--	--	--
TV Broadcast Audio Production	16.5	--	--	46.2	--	--	--	--	--
Post-Production									
Audio Post-Prod./Video									
Sweetening	10.8	--	--	--	70.0	--	--	--	--
Film Sound	2.5	--	--	--	20.0	--	--	--	--
Tape Duplication	1.9	--	--	--	10.0	--	--	--	--
Disc/CD Mastering	--	--	--	--	--	--	--	--	--

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TABLE 2. — TECHNICAL STAFF PROFILE . . . continued

Technical	Total %	Recording Studio %	Live Sound %	Broadcast %	Post-Production %	Corp/Ed/ Govt. %	Eng./Prod. %	Acoustical Tech. %	Dealers/ Distr. %
Corp./Ed/Govt.									
Corp. Audio Prod.	4.4	--	--	--	--	20.0	--	--	--
Educational	12.0	--	--	--	--	68.0	--	--	--
Government	3.2	--	--	--	--	12.0	--	--	--
Eng. and/or Production									
Eng. and/or Prod. Company	6.3	--	--	--	--	--	--	--	--
Acoustic/Tech. Consulting									
Audio Acoustic Consulting	--	--	--	--	--	--	--	--	--
Audio Technical Consulting	3.8	--	--	--	--	--	--	--	--
Dealer/Distr./Rental									
Dealer/Distr./Mfgr. Equipment Rental	7.6	--	--	--	--	--	--	--	100.0
Equipment Rental	0.6	--	--	--	--	--	--	--	8.3
Top Income Source									
Records/Demos	10.8	60.0	--	--	--	--	--	--	8.3
Commercial Prod.	26.6	20.0	--	59.6	10.0	--	--	--	--
Corp./Industrial	15.2	8.0	--	5.8	5.0	--	--	--	41.7
Post-Prod.	13.9	--	--	9.6	55.0	--	--	--	--
Mastering/Duplication	4.4	4.0	--	1.9	15.0	--	--	--	--
Other	23.4	4.0	--	23.1	15.0	--	--	--	50.0
No Answer	7.6	8.0	--	3.8	--	--	--	--	--
Years at Present Job									
Less than 3	20.9	16.0	--	23.1	20.0	16.0	--	--	41.7
3 or 4	23.4	24.0	--	25.1	30.0	24.0	--	--	8.3
5 to 9	32.3	32.0	--	28.8	50.0	32.0	--	--	8.3
10 to 14	11.4	20.0	--	9.6	--	8.0	--	--	25.0
15 or More	12.0	8.0	--	13.5	--	20.0	--	--	16.7
Median	5.9	6.0	--	5.4	5.0	6.6	--	--	5.0
Years in Professional Audio Industry									
Less than 5	7.6	4.0	--	11.5	5.0	4.0	--	--	8.3
5 to 9	19.0	20.0	--	15.4	30.0	8.0	--	--	33.3
10 to 14	29.1	28.0	--	28.9	40.0	24.0	--	--	25.0
15 to 24	31.0	36.0	--	23.1	25.0	48.0	--	--	33.4
25 or more	12.7	12.0	--	19.2	--	16.0	--	--	--
No Answer	0.6	--	--	1.9	--	--	--	--	--
Median	14.0	14.7	--	13.9	11.9	19.4	--	--	10.0
Salary Level									
Less than \$15,000	5.7	20.0	--	--	5.0	--	--	--	--
\$15,000 to \$24,999	20.3	20.0	--	26.9	20.0	12.0	--	--	25.0
\$25,000 to \$34,999	31.6	12.0	--	38.5	20.0	48.0	--	--	16.7
\$35,000 to \$49,999	23.4	28.0	--	25.0	5.0	32.0	--	--	33.3
\$50,000 to \$74,999	10.8	16.0	--	9.6	10.0	8.0	--	--	8.3
\$75,000 or more	5.7	--	--	--	40.0	--	--	--	--
No Answer	2.5	4.0	--	--	--	--	--	--	16.7
Median	\$31,650	\$28,350	--	\$28,850	\$50,000	\$33,200	--	--	\$40,000
Independent Contractor									
Independent Contractor	12.0	28.0	--	3.8	--	8.0	--	--	--
Average Hourly Rate									
Less than \$25	4.4	4.0	--	1.9	--	--	--	--	--
\$25 to \$49.99	5.1	16.0	--	--	--	8.0	--	--	--
\$50 to \$74.99	1.9	4.0	--	1.9	--	--	--	--	--
\$75 or More	0.6	4.0	--	--	--	--	--	--	--
Median	\$32.75	N/A	--	N/A	--	N/A	--	--	--
Fringe Benefits Received									
Med. Ins. (Paid)	82.9	68.0	--	86.5	95.0	92.0	--	--	100.0
Dental Insur. (Paid)	50.0	40.0	--	48.1	65.0	64.0	--	--	58.3
Vision (Eyecare) Ins. (Paid)	22.2	16.0	--	21.2	30.0	36.0	--	--	25.0
Short-Term Disability Plan	34.8	16.0	--	53.8	40.0	32.0	--	--	41.7
Long-Term Disability Plan	36.7	12.0	--	50.0	45.0	48.0	--	--	41.7
Sick Leave	68.4	44.0	--	76.9	75.0	92.0	--	--	83.3
Vacation	84.9	68.0	--	94.2	95.0	92.0	--	--	100.0
1 week	11.4	16.0	--	11.5	--	16.0	--	--	8.3
2 weeks	36.7	28.0	--	44.2	50.0	20.0	--	--	50.0
3 weeks	18.4	8.0	--	21.2	30.0	24.0	--	--	16.7
4 weeks	10.8	12.0	--	9.6	10.0	20.0	--	--	8.3
5 or More	7.6	4.0	--	7.7	5.0	32.0	--	--	16.7
Median	2.8	2.6	--	2.9	3.0	3.4	--	--	2.8

TABLE 2. — TECHNICAL STAFF PROFILE . . . continued

Technical	Total %	Recording Studio %	Live Sound %	Broadcast %	Post-Production %	Corp/Ed/ Govt. %	Eng./Prod. %	Acoustical Tech. %	Dealers/ Distr. %
Holidays	77.2	48.0	--	86.5	95.0	92.0	--	--	91.7
Personal Business Days (Paid)	22.2	8.0	--	38.5	15.0	24.0	--	--	8.3
Stock Purchase Plan	8.2	4.0	--	7.7	10.0	8.0	--	--	16.7
Profit-sharing Plan	15.2	8.0	--	23.1	--	8.0	--	--	25.0
Savings Plan	13.9	4.0	--	17.3	10.0	24.0	--	--	25.0
Pension Plan	34.2	8.0	--	40.4	40.0	72.0	--	--	33.3
Comp. Deferment Plan	24.7	12.0	--	36.5	10.0	36.0	--	--	25.0
Bonus	19.0	24.0	--	17.3	25.0	8.0	--	--	16.7
Assoc. Memberships (Paid)	15.2	4.0	--	17.3	10.0	16.0	--	--	25.0
Trade Shows Etc.	35.4	20.0	--	36.5	60.0	32.0	--	--	58.3
Tuition Refund Plan	20.9	4.0	--	25.0	20.0	36.0	--	--	41.7
Automobile Furnished	10.1	8.0	--	5.8	--	4.0	--	--	25.0
None	4.4	--	--	1.9	--	4.0	--	--	--
Company Unionized	22.8	4.0	--	25.0	40.0	40.0	--	8.3	--
Reader a Union Member	25.9	20.0	--	25.0	60.0	32.0	--	--	--
Union									
IATSE	9.5	16.0	--	--	40.0	--	--	--	--
IBEW	5.7	4.0	--	11.6	10.0	--	--	--	--
NABET	2.5	--	--	5.8	5.0	--	--	--	--
Other	8.5	4.0	--	7.6	10.0	32.0	--	--	--
How Initial Audio Experience Was Acquired									
On the Job	69.0	72.0	--	65.4	65.0	68.0	--	--	66.7
College/University Program	25.9	24.0	--	28.8	45.0	24.0	--	--	16.7
Recording School	3.8	8.0	--	--	--	--	--	--	16.7
Other	9.5	12.0	--	7.7	5.0	16.0	--	--	8.3
No Answer	1.3	4.0	--	1.9	--	--	--	--	--
Education									
High School	10.1	4.0	--	13.5	5.0	--	--	--	25.0
Some College	46.8	52.0	--	50.0	50.0	40.0	--	--	41.7
College Grad (B.A., B.S.)	28.5	28.0	--	23.1	30.0	28.0	--	--	33.3
Grad School (M.S., M.S.)	5.1	4.0	--	3.8	5.0	12.0	--	--	--
Grad School (Ph.D.)	1.9	8.0	--	--	--	4.0	--	--	--
Music School	6.3	16.0	--	5.8	5.0	4.0	--	--	--
Technical School	36.1	16.0	--	38.5	30.0	32.0	--	--	58.3
Armed Services School	9.5	4.0	--	7.7	10.0	8.0	--	--	16.7
Age									
Under 25	3.2	--	--	3.8	--	--	--	--	8.3
25 to 34	38.6	28.0	--	48.1	55.0	24.0	--	--	25.0
35 to 44	39.8	52.0	--	26.9	40.0	52.0	--	--	58.3
45 to 54	12.7	12.0	--	13.5	5.0	24.0	--	--	8.3
55 to 64	4.4	4.0	--	7.7	--	--	--	--	--
65 or Over	--	--	--	--	--	--	--	--	--
No Answer	1.3	4.0	--	--	--	--	--	--	--
Median	36.9	38.8	--	34.6	34.1	40.0	--	--	37.9
Societies/Organizations Hold Membership									
AES	27.8	32.0	--	13.5	30.0	28.0	--	--	41.7
SMPTE	19.0	8.0	--	15.4	45.0	32.0	--	--	8.3
NAB	17.7	4.0	--	36.5	5.0	16.0	--	--	8.3
NAMM	1.3	--	--	--	--	--	--	--	--
NSCA	1.3	--	--	--	--	--	--	--	8.3
SPARS	1.3	4.0	--	--	--	--	--	--	--
IMA	0.6	--	--	--	--	4.0	--	--	--
Other	15.8	16.0	--	17.3	10.0	20.0	--	--	8.3
None	37.3	56.0	--	36.5	45.0	20.0	--	--	41.7

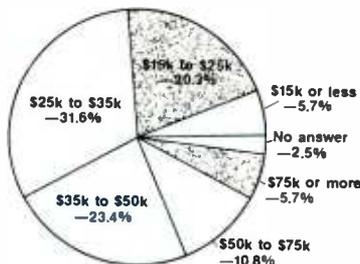


Figure 3. Salary level for technical staff.

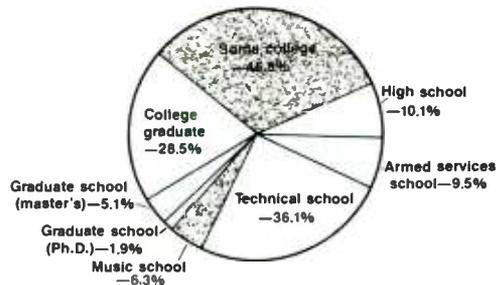


Figure 4. Education level for technical staff.

TABLE 2. — TECHNICAL STAFF PROFILE . . . continued

Technical	Total %	Recording Studio %	Live Sound %	Broadcast %	Post-Production %	Corp./Ed./Govt. %	Eng./Prod. %	Acoustical Tech. %	Dealers/Distr. %
Trade Shows Attended									
NAB	45.6	32.0	--	55.8	55.0	48.0	--	--	50.0
AES—United States	38.6	68.0	--	9.6	65.0	28.0	--	--	50.0
SMPTE	25.3	24.0	--	11.5	60.0	32.0	--	--	41.7
NAMM—Summer	2.5	4.0	--	--	5.0	--	--	--	8.3
NAMM—Winter	1.9	8.0	--	--	--	--	--	--	--
NSCA	1.9	--	--	--	--	--	--	--	--
AES—Europe	6	--	--	--	--	--	--	--	8.3
Other	3.8	4.0	--	7.7	5.0	8.0	--	--	16.7
None	27.2	16.0	--	28.9	20.0	40.0	--	--	25.0
Geographical									
New England	4.4	8.0	--	3.8	--	4.0	--	--	8.3
Middle Atlantic	15.2	16.0	--	13.5	15.0	16.0	--	--	8.3
East N. Central	18.4	12.0	--	25.0	5.0	28.0	--	--	--
West N. Central	4.4	--	--	7.7	--	--	--	--	--
South Atlantic	18.4	16.0	--	23.1	15.0	20.0	--	--	--
East S. Central	5.7	8.0	--	5.8	5.0	4.0	--	--	16.7
West S. Central	4.4	4.0	--	3.8	5.0	4.0	--	--	8.3
Mountain	3.8	--	--	5.8	5.0	--	--	--	--
Pacific	24.7	36.0	--	9.6	50.0	24.0	--	--	58.4
No Answer	0.6	--	--	1.9	--	--	--	--	--

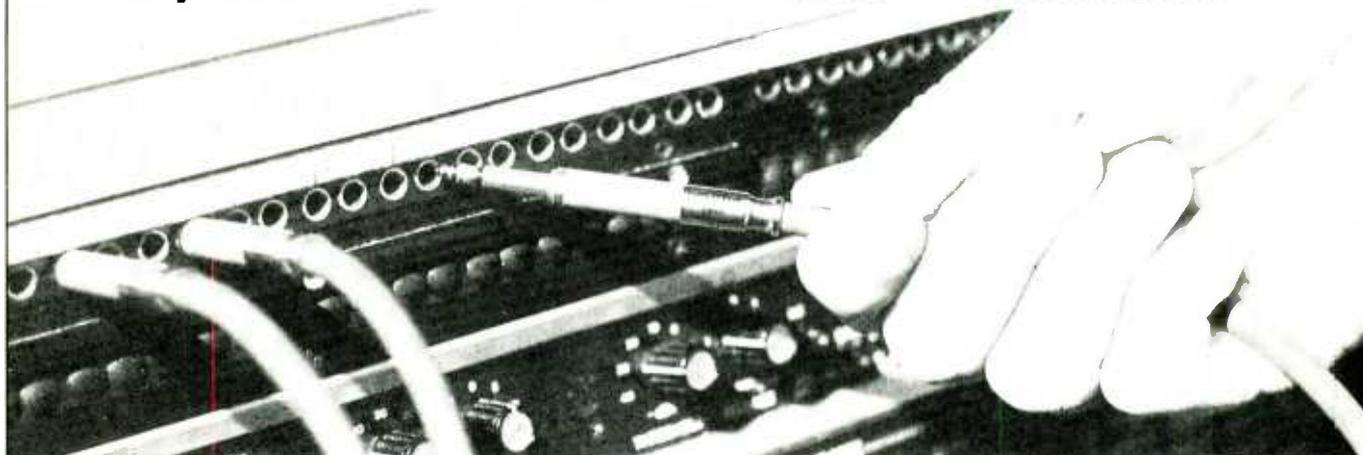
TABLE 3. — PRODUCTION STAFF PROFILE . . . continued

Production	Total %	Recording Studio %	Live Sound %	Broadcast %	Post-Production %	Corp./Ed./Govt. %	Eng./Prod. %	Acoustical Tech. %	Dealers/Distr. %
Job Title									
Audio Engineer	54.8	53.1	44.1	62.2	45.2	63.4	51.9	47.1	90.0
Mastering Engineer	2.3	3.1	62.7	--	12.9	--	3.7	11.8	--
Live Sound Engineer	16.5	3.1	--	6.7	3.2	3.3	--	--	--
Post Prod. Engineer	7.7	3.1	--	13.3	29.0	6.7	7.4	5.9	--
Engineer/Producer	16.5	50.0	1.7	20.0	6.5	23.3	22.2	23.5	10.0
Arranger/Production	3.4	3.1	--	6.7	3.2	3.3	11.1	5.9	--
Job									
Staff	73.6	56.3	61.0	88.9	67.7	93.3	74.1	52.9	100.0
Independent Contractor	26.0	50.0	37.3	11.1	32.3	6.7	25.9	47.1	--
No Answer	0.8	--	1.7	--	3.2	--	--	--	--
Type of Business									
Recording Studio									
Recording Studio (fixed)	10.7	87.5	--	--	--	--	--	--	--
Recording Studio (mobile)	1.9	15.6	--	--	--	--	--	--	--
Live Sound									
Sound Reinforcement	23.8	--	100.00	--	--	--	--	--	--
Broadcast Audio Production									
Radio Broadcast Production	8.4	--	--	44.4	--	--	--	--	--
TV Broadcast Audio Production	10.7	--	--	60.0	--	--	--	--	--
Post-Production									
Audio Post-Prod./Video Sweetening	9.6	--	--	--	64.5	--	--	--	--
Film Sound	3.1	--	--	--	19.4	--	--	--	--
Tape Duplication	1.1	--	--	--	3.2	--	--	--	--
Disc/CD Mastering	1.9	--	--	--	12.9	--	--	--	--
Corp./Ed./Govt.									
Corp. Audio Prod.	6.9	--	--	--	--	40.0	--	--	--
Educational	6.1	--	--	--	--	40.0	--	--	--
Government	4.2	--	--	--	--	20.0	--	--	--
Eng. and/or Production									
Eng. and/or Prod. Company	11.1	--	--	--	--	--	100.0	--	--
Acoustic/Tech. Consulting									
Audio Acoustic Consulting	3.1	--	--	--	--	--	--	35.3	--
Audio Technical Consulting	6.1	--	--	--	--	--	--	76.5	--
Dealer/Distr./Rental									
Dealer/Distr./Mfr.	8.8	--	--	--	--	--	--	--	100.0
Equipment Rental	1.5	--	--	--	--	--	--	--	5.0

TABLE 3. – PRODUCTION STAFF PROFILE . . . continued

Production	Total %	Recording Studio %	Live Sound %	Broadcast %	Post-Production %	Corp./Ed/ Govt. %	Eng./Prod. %	Acoustical Tech. %	Dealers/ Distr. %
Top Income Source									
Records/Demos	8.4	53.1	--	2.2	--	--	14.8	--	--
Commercial Prod.	18.0	25.0	11.9	42.2	9.7	3.3	26.0	5.9	5.0
Corp./Industrial	20.3	6.3	16.9	8.9	19.4	40.0	18.5	47.1	30.0
Post-Prod.	9.2	--	--	11.1	51.6	6.7	3.7	--	--
Mastering/Duplication	3.8	3.1	--	--	19.4	6.7	--	5.9	--
Other	32.6	18.8	55.9	26.7	3.2	20.0	29.6	41.2	60.0
No Answer	9.2	--	15.3	8.9	--	23.3	7.4	5.9	5.0
Years at Present Job									
Less than 3	21.1	25.0	17.0	13.3	29.0	10.0	33.3	29.4	25.0
3 or 4	28.0	28.1	23.6	35.6	19.4	30.0	29.7	17.6	40.0
5 to 9	26.3	18.8	39.0	20.0	38.6	23.3	18.5	11.8	25.0
10 to 14	11.9	9.4	10.2	22.2	6.5	20.0	3.7	11.8	5.0
15 or More	12.3	15.6	10.2	8.9	6.5	16.7	14.8	29.4	5.0
No Answer	0.4	3.1	--	--	--	--	--	--	--
Median	5.2	4.7	6.2	5.3	5.2	7.2	4.1	6.3	4.3
Years in Professional Audio Industry									
Less than 5	9.2	15.6	8.5	8.9	3.2	6.7	11.1	5.9	15.0
5 to 9	27.6	37.5	28.8	24.4	22.6	20.0	40.8	--	40.0
10 to 14	26.8	28.1	25.4	24.4	38.7	23.3	22.2	29.4	25.0
15 to 24	25.3	12.5	27.1	29.0	25.8	36.7	25.9	35.3	5.0
25 or more	9.6	6.3	8.5	11.1	9.7	10.0	--	23.5	15.0
No Answer	1.5	--	1.7	2.2	--	3.3	--	5.9	--
Median	12.3	9.6	12.4	13.2	13.2	14.7	9.8	17.5	9.4
Salary Level									
Less than \$15,000	5.0	18.8	5.1	6.7	--	3.3	--	--	--
\$15,000 to \$24,999	21.8	21.9	30.5	37.7	19.4	20.0	7.4	5.9	--
\$25,000 to \$34,999	33.8	21.9	33.9	31.1	32.3	40.0	33.3	29.4	55.0
\$35,000 to \$49,999	22.2	15.5	22.0	17.8	12.9	23.4	40.8	23.5	30.0
\$50,000 to \$74,999	10.7	12.5	5.1	6.7	16.1	10.1	11.1	29.4	10.0
\$75,000 or more	5.4	9.4	1.7	--	16.1	--	7.4	11.8	5.0
No Answer	1.1	--	1.7	--	3.2	3.3	--	--	--
Median	\$31,850	\$27,500	\$27,850	\$26,400	\$34,400	\$32,500	\$39,150	\$42,500	\$34,450

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TABLE 3. — PRODUCTION STAFF PROFILE . . . continued

Production	Total %	Recording Studio %	Live Sound %	Broadcast %	Post-Production %	Corp/Ed/ Govt. %	Eng./Prod. %	Acoustical Tech. %	Dealers/ Distr. %
Job Title									
Audio Engineer	54.8	53.1	44.1	62.2	45.2	63.4	51.9	47.1	90.0
Mastering Eng.	2.3	3.1	62.7	—	12.9	—	3.7	11.8	—
Live Sound Eng.	16.5	3.1	—	6.7	3.2	3.3	—	—	—
Post-Prod. Eng.	7.7	3.1	—	13.3	29.0	6.7	7.4	5.9	—
Engineer/Producer	16.5	50.0	1.7	20.0	6.5	23.3	22.2	23.5	10.0
Arranger/Producer	3.4	3.1	—	6.7	3.2	3.3	11.1	5.9	—
Programmer	3.1	3.1	1.7	8.9	—	6.7	3.7	17.6	—
Job									
Staff	73.6	56.3	61.0	88.9	67.7	93.3	74.1	52.9	100.0
Independent Contractor	26.8	50.0	37.3	11.1	32.3	6.7	25.9	47.1	—
Average Hourly Rate									
Less than \$25	12.2	25.0	20.3	8.9	12.9	6.7	7.4	—	—
\$25 to \$49.99	5.4	9.4	5.1	—	9.7	—	7.4	23.5	—
\$50 to \$74.99	3.5	12.5	1.7	2.2	—	—	3.7	5.9	—
\$75 or More	1.5	—	—	—	3.2	—	3.7	11.8	—
No Answer	4.2	3.1	10.2	—	6.5	—	3.7	5.9	—
Median	\$23.95	\$22.50	\$21.00	N/A	N/A	N/A	N/A	N/A	—
Fringe Benefits Received									
Med. Ins. (Paid)	73.6	46.9	64.4	84.4	77.4	80.0	74.1	76.5	100.0
Dental Ins. (Paid)	52.9	34.4	40.7	57.8	58.1	63.3	63.0	52.9	70.0
Vision (Eyecare) Ins. (Paid)	23.4	21.9	25.4	13.3	19.4	30.0	25.9	23.5	35.0
Short-Term Disability Plan	36.4	21.9	30.5	40.0	29.0	46.7	48.1	41.2	45.0
Long-Term Disability Plan	38.7	15.6	25.4	44.4	35.5	53.3	55.6	52.9	50.0
Sick Leave (Paid)	61.7	28.1	47.5	80.0	58.1	80.0	66.7	76.5	75.0
Vacation	73.2	43.8	57.7	88.8	77.5	86.7	74.1	82.4	95.0
1 week	9.2	12.5	8.5	4.4	22.6	10.0	—	—	15.0
2 weeks	29.5	15.6	28.8	33.3	29.0	16.7	40.8	29.4	50.0
3 weeks	21.8	9.4	10.2	28.9	19.4	30.0	25.9	41.2	30.0
4 weeks	9.2	3.1	6.8	17.8	6.5	16.7	7.4	11.8	—
5 or More	3.5	3.1	3.4	4.4	—	13.3	—	—	—
Median	2.9	2.6	2.7	3.2	2.6	3.6	2.9	3.3	2.7
Holidays (Paid)	68.2	37.5	47.5	86.7	77.4	83.3	66.7	76.5	95.0
Personal Business Days (Paid)	25.7	15.6	11.9	44.4	35.5	23.3	29.6	35.3	15.0
Stock Purchase Plan	12.3	6.3	6.8	17.8	6.5	20.0	25.9	5.9	10.0
Profit-sharing Plan	19.2	6.3	13.6	20.0	9.7	20.0	37.0	17.6	45.0
Savings Plan	17.6	3.1	6.8	28.9	12.9	30.0	29.6	11.8	25.0
Company Unionized									
Reader a Union Member	22.2	15.6	33.9	33.3	25.8	13.3	7.4	17.6	5.0
Union									
IATSE	7.3	6.3	20.3	—	9.7	—	—	5.9	5.0
IBEW	5.4	3.1	3.4	13.3	9.7	—	—	11.8	—
NABET	3.4	—	—	13.3	3.2	—	3.7	5.9	—
Other	6.9	3.1	6.8	11.1	9.7	16.7	3.7	—	—
How Initial Audio Experience Was Acquired									
On the Job	62.5	59.4	79.7	55.6	64.5	60.0	55.6	64.7	40.0
College/University Program	31.0	31.3	20.3	28.9	22.6	46.7	37.0	29.4	50.0
Recording School	3.8	6.3	10.2	—	3.2	—	—	—	5.0
Other	14.2	15.6	8.5	17.8	16.1	10.0	14.8	11.8	25.0
No Answer	.4	—	—	2.2	—	—	—	—	—

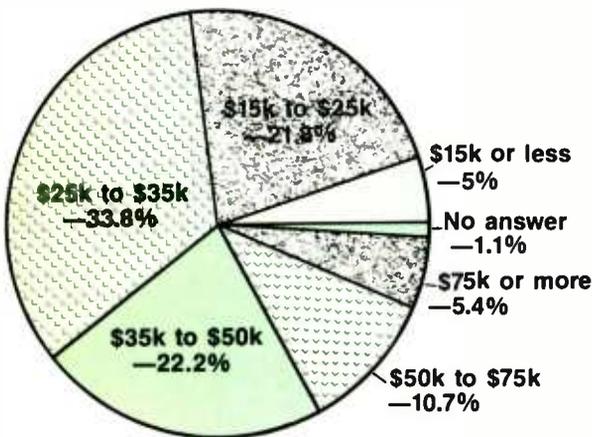


Figure 5. Salary level for production staff.

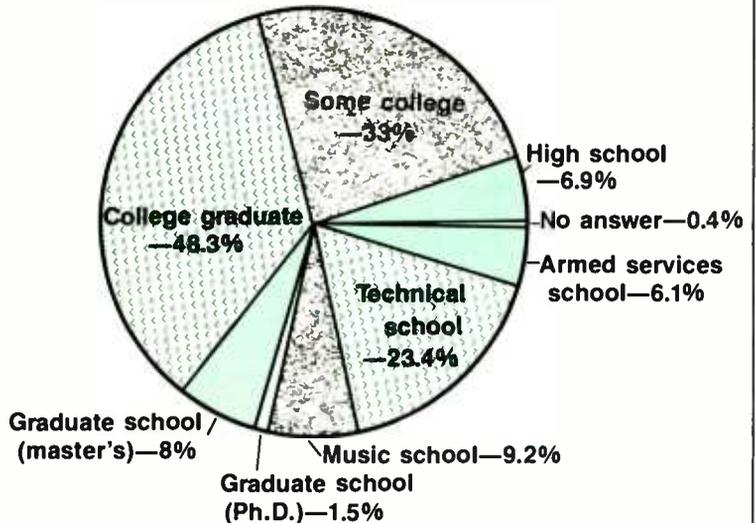


Figure 6. Education level for production staff.

TABLE 3. — PRODUCTION STAFF PROFILE . . . continued

Production	Total %	Recording Studio %	Live Sound %	Broadcast %	Post-Production %	Corp/Ed/ Govt. %	Eng./Prod. %	Acoustical Tech. %	Dealers/ Distr. %
Education									
High School	6.9	3.1	6.8	8.9	12.9	3.3	3.7	--	15.0
Some College	33.0	34.4	55.9	28.9	38.7	16.7	14.8	23.5	20.0
College Grad (B.A., B.S.)	48.3	59.4	33.9	44.4	41.9	60.0	59.3	58.8	50.0
Grad School (M.A., M.S.)	8.0	--	3.4	8.9	3.2	10.0	22.2	11.8	15.0
Grad School (Ph.D.)	1.5	--	--	--	--	10.0	--	5.9	--
Music School	9.2	12.5	10.2	6.7	12.9	10.0	7.4	--	10.0
Technical School	23.4	15.6	37.3	20.0	29.0	26.7	3.7	--	35.0
Armed Services School	6.1	--	11.9	--	6.5	10.0	7.4	5.9	5.0
No Answer	0.4	3.1	--	--	--	--	--	--	--
Age									
Under 25	5.4	6.3	8.5	4.4	6.5	--	11.1	--	--
25 to 34	47.8	56.2	47.4	44.6	48.4	43.3	48.2	29.5	65.0
35 to 44	29.9	31.3	33.9	33.3	25.8	40.0	25.9	17.6	15.0
45 to 54	10.0	3.1	3.4	13.3	9.7	10.0	7.4	29.4	20.0
55 to 64	4.6	--	5.1	4.4	3.2	6.7	3.7	17.6	--
65 or Over	1.9	3.1	1.7	--	3.2	--	3.7	5.9	--
No Answer	0.4	--	--	--	3.2	--	--	--	--
Median	34.3	32.8	33.8	35.3	33.7	36.7	33.1	46.0	32.7
Societies/Organizations Hold Membership									
AES	34.5	31.3	33.9	13.3	41.9	23.3	40.7	76.5	50.0
SMPTE	10.7	6.3	1.7	15.6	25.8	10.0	18.5	11.8	--
NAB	4.2	--	--	13.3	9.7	--	3.7	--	5.0
NSCA	3.1	--	5.1	--	--	--	3.7	17.6	5.0
NAMM	2.7	6.3	3.4	--	3.2	3.3	--	--	5.0
SPARS	2.7	6.3	1.7	--	3.2	--	7.4	5.9	--
IMA	1.1	--	--	--	--	--	3.7	--	10.0
MMA	0.4	--	--	--	--	--	--	--	5.0
Other	19.2	25.0	10.2	17.8	6.5	46.7	40.7	17.6	20.0
None	48.7	46.9	62.7	60.0	45.2	46.7	37.0	12.6	35.0

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TABLE 3.— PRODUCTION STAFF PROFILE . . . continued

Management	Total %	Recording Studio %	Live Sound %	Broadcast %	Post-Production %	Corp/Ed/ Govt. %	Eng./Prod. %	Acoustical Tech. %	Dealers/ Distr. %
Trade Shows Attended									
AES—United States	47.5	50.0	49.2	26.7	51.6	23.3	70.4	76.5	60.0
NAB	23.0	12.5	10.2	26.7	41.9	23.3	25.9	29.4	30.0
SMPTE	13.4	6.3	3.4	13.3	32.3	13.3	18.5	23.5	10.0
NAMM—Summer	11.1	15.6	18.6	2.2	3.2	3.3	7.4	11.8	30.0
NAMM—Winter	7.7	12.5	8.5	2.2	6.5	3.3	3.7	17.6	30.0
NSCA	4.6	—	8.5	—	—	—	7.4	—	10.0
AES—Europe	1.9	—	—	2.2	—	3.3	3.7	5.9	5.0
APRS	0.4	—	—	—	—	3.3	—	—	—
Other	12.6	9.2	6.8	6.7	6.5	36.7	7.4	17.6	25.0
None	28.4	25.0	35.6	48.9	19.4	30.0	14.8	11.8	10.0
Geographical Location									
New England	6.1	6.3	5.1	8.9	3.2	6.7	7.4	5.9	5.0
Middle Atlantic	19.2	18.8	18.6	13.3	29.0	16.7	26.0	17.6	15.0
East N. Central	21.4	15.6	23.7	24.5	16.1	23.2	18.5	23.5	25.0
West N. Central	8.0	15.6	8.5	6.7	9.7	6.7	3.7	—	10.0
South Atlantic	14.6	3.1	16.9	20.0	3.2	20.0	18.5	11.8	20.0
East S. Central	5.4	6.3	6.8	15.6	3.2	—	—	—	—
West S. Central	6.1	3.1	8.5	2.2	12.9	6.7	11.1	—	—
Mountain	2.7	—	1.7	4.4	—	6.7	—	11.8	—
Pacific	15.7	31.2	10.2	2.2	22.7	13.3	14.8	29.4	20.0
No Answer	0.8	—	—	2.2	—	—	—	—	5.0

Table 4.—1988, 1987 Median Comparisons—Management

	1988	1987	±
Salary	\$36,700	\$34,250	+\$2,450
Years in present job	7.5	8.0	-0.5
Years in industry	15.3	16.6	-1.3
Age	38.1	38.7	-0.6

Table 5.—1988, 1987 Median Comparisons—Technical

	1988	1987	±
Salary	\$31,650	\$31,400	+\$250
Years in present job	5.9	6.1	-0.2
Years in industry	14	14	—
Age	36.9	36.6	+0.3

Table 6.—1988, 1987 Median Comparisons—Production

	1988	1987	±
Salary	\$31,850	\$27,500	+\$4,350
Years in present job	5.2	4.9	+0.3
Years in industry	12.3	11.9	+0.4
Age	34.3	33.2	+1.1

Table 7.—1988 Median Comparisons Between Job Categories

	Mgt	Tech	Prod
Salary	\$36,700	\$31,650	\$31,850
Years in present job	7.5	5.9	5.2
Years in industry	15.3	14	12.3
Age	38.1	36.9	34.3

Table 8.—Acquiring Initial Audio Experience (%)

	Mgt	Tech	Prod
On the job	70.2	69	62.5
College/university	29.8	25.9	31
Recording school	6.3	3.8	3.8
Other	6.8	9.5	14.2
No answer	2.4	1.3	0.4

Table 9.—Top Reported Income Source (%)

	Mgt	Tech	Prod
Records/demos	14	10.8	8.4
Commercial production	11.9	26.6	18
Industrial/corporate	22.6	15.2	20.3
Post-production	9.2	13.9	9.2
Mastering/duplication	7.1	4.4	3.8
Other	35.7	23.4	32.6
No answer	4.2	7.6	9.2



Recording the “Tunnel of Love” Tour

By the RE/P staff

A portable touring recording system solved both the immediate needs of touring and studio recording needs once the tour was over.

The problem is common for any major touring act: If you are going to record any shows, which recording process is going to be the most economical, yet yield the desired audio quality?

The answer to the question becomes much more complex when dealing with an artist like Bruce Springsteen, whose penchant for recording almost all of his

shows is well known. The cost of hiring a mobile truck for that many dates was so high that it became obvious that buying the required equipment would be cheaper.

But finding gear that can straddle the demands of both installed and mobile use can be challenging. Not only are there sonic considerations, there is also the

question of how to store and transport the gear from venue to venue. For Springsteen, these questions were answered by his engineer, Toby Scott, who, in conjunction with Tim Wilson of Amek/TAC, designed a portable recording system that will be installed in Springsteen's project studio after the “Tunnel of Love” tour is completed.



Toby Scott, Bruce Springsteen's engineer, designed the Thrill Hill Recording System in conjunction with Tim Wilson, Amek's national sales manager.



Closeup of racks No. 2 and 4, containing a patchbay, monitor amplifiers and Lynx TimeLine synchronizers.

The system, dubbed the "Thrill Hill Recording System," is housed in 11 road cases (see Figure 1):

- No. 1: Amek Angela 56/48 console.
- No. 2: Patchbay rack with all connections via ELCO/EDAC connectors to effects rack, console and tape machines.
- No. 3: Four TAC SR-9000 power supplies

for the console, with automatic diode switch-over system in case of failure of one unit.

- No. 4: Four TimeLine Lynx time code synchronizers and three power amplifiers for monitor systems.

- No. 5: Cassette and R-DAT recorders, plus limiters and compressors, including

Sony TC-WR950 cassette and two Sony DTC-1000ES R-DAT decks, UREI LA-2A and LA-3A limiters, Drawmer stereo Tube Limiter and five Drawmer compressors. (See Figure 2.)

No. 6: Outboard effects, digital reverbs and samplers, including three Roland SDA-3000 DDLs, three Audio Engineering Services SX-303R delay-samplers, two Yamaha SPX-90s, a Yamaha REV-7, an AMS RMS-16 digital reverb, an AMS DMX-15-80 stereo DDL, a dbx FS900 frame with two de-esser modules and an Ursa Major Space Station. (See Figure 3.)

No. 7: Cable trunk for entire system harness.

There also were four other flight cases, two holding Sony PMC-3324 DASH-format digital 24-tracks and two housing recording tape, monitor loudspeakers and stands, video monitor and accessories.

"The primary reason for putting together this portable system," Scott said, "is so that we will have material for posterity, and to allow Bruce to check the sound of the shows. Although he hasn't changed the shows much over the course of this tour, he has put in four or five songs as substitutes. The digital tapes will let him hear how he is doing.

"We will also have plenty of material for a subsequent live album release from the 'Tunnel of Love' tour."

Cost factor

The decision to design a mobile system that could be permanent was borne out of Springsteen's studio recording requirements and the cost to record tour dates.

"By the end of the 'Tunnel of Love' album, we had outgrown our 32/24 console and were looking for a replacement," Scott said. "At the same time, Bruce was considering a new tour with the full band, and his manager considered that we might want to record certain dates to have a record of the tour.

"I was considering all the alternatives. Because we had gone to 48-track on certain songs, it was suggested that we buy another PCM-3324, regardless of whether or not we would be using a mobile truck to record the tour."

Cost was also a major consideration. Scott figured that it would cost \$8,000 to \$10,000 a day to hire a mobile truck, including travel and living expenses. With at least 10 dates to record, the tour would spend \$100,000 to \$150,000 for mobile recording and end up with nothing but multitrack tapes. Also factor is that Springsteen would still need a console to remix the tape, or do work on a new studio album, and it became clear that an alternative was needed.

The initial concept was a console, racks and umbilical to the stage. All the racks were built into foam-padded shells and can

be pulled out of the flight cases for the permanent studio system. Of particular importance was the choice of the console, which had to be compact, yet appropriate for live and studio use.

"The Amek Angela seemed to be the most compact console we looked at, yet the most flexible in terms of what it could achieve," Scott said. "I drew up a basic plan, talked with Tim Wilson, Amek's national sales manager [and systems designer for the project] and sent him my diagrams and sketches. He quoted a price for a turnkey system, which was far less than the cost of a mobile and crew."

It took about three weeks from the time Scott got the go-ahead to build the system to its arrival in mid-March at the Richfield Coliseum in Cleveland. The first show recorded was the second night at Richfield Coliseum. Also recorded were the second-night show in Cleveland, as well as shows in Detroit and the Nassau Coliseum on Long Island before the system was stored for three weeks.

It then went to Los Angeles for five shows at the Sports Arena. The final five dates of the U.S. tour were at Madison Square Garden. The system was not used for the band's European summer tour. Instead, Scott made final preparations for the system to be installed at Springsteen's house.

Ease of use

One of the system's advantages is that it easily plugs together. It takes Scott and Paul Dugre, the system technical engineer, about 90 minutes to unload the truck and prepare the system to receive a signal. It takes about 70 minutes from the last song to being ready to load into the truck.

The backstage mixing position connects to the PA system by means of an 80-way snake and splitter box. The house-mix position, manned by live-sound mixer Bruce Jackson, receives 49 inputs, plus feeds from six or seven additional audience mics and extras used on certain instruments. In most cases, the 49 inputs are submixes. For example, originally the system received feeds from 15 mics, including horn, horn/vocal and percussion mics. These are now combined at the house console and sent to the stage as a submix.

"Via a fader-reverse switch on the Angela, I can control the mix-bus levels going to the multitracks using the rotary monitor pots, and the long-throw channel input faders for the stereo mix," Scott said. "I monitor on Yamaha NS-10M speakers in my makeshift control room. Channels 1 through 24 are monitoring the outputs to Sony 3324 No. 1, while channels 25 through 48 monitor machine No. 2, to cover reel changeovers. That way I can have a reasonable running stereo mix of

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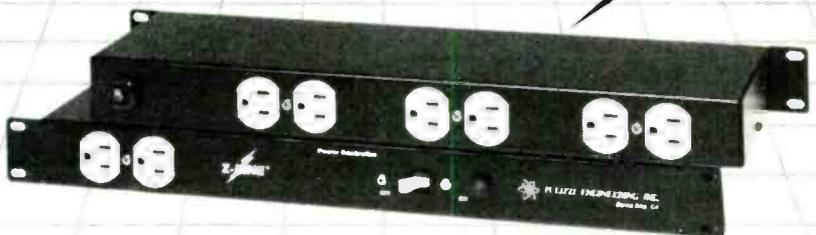
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The Carriage House

Bruce Springsteen is no stranger to recording at home. His 1984 album, "Nebraska," was recorded on a Tascam 8-track. And the carriage house where the remote system will be permanently installed was the site for the "Tunnel of Love" sessions; recording took place in the living room.

"The area measured about 12' x 20' without a separate control room; to record any acoustic instruments, we turned off the monitors and used headphones," said Toby Scott. "The Sony machine sat in the kitchen. It was a very comfortable, convenient working environment—we had enough outboard equipment to add special effects—and it worked real well."

Although Springsteen owns one Sony 3324, on some cuts the tracks were being filled up, and a second, rented machine was needed. In those situations, Scott took the multitrack tapes into New York, edited them on two 3324s and then made a slave copy.

He then brought the slave back to New Jersey, erased the unneeded tracks and overdubbed on this second tape. Both tapes were then in sync for remix.

"Most songs on that album involved fewer than 24 tracks, although on three or four, we had 30," Scott said. "One in particular required 38 or 40 tracks. Remixing from dual 24-tracks was problematic. The 32-input/24-bus Trident Trimix we had at the studio proved to be real good, but if we needed to work with more than 24 tape tracks, we would run out of line inputs for new tracks, or for effects returns. As a result, we needed to go into a studio for remix and complex overdub sessions."

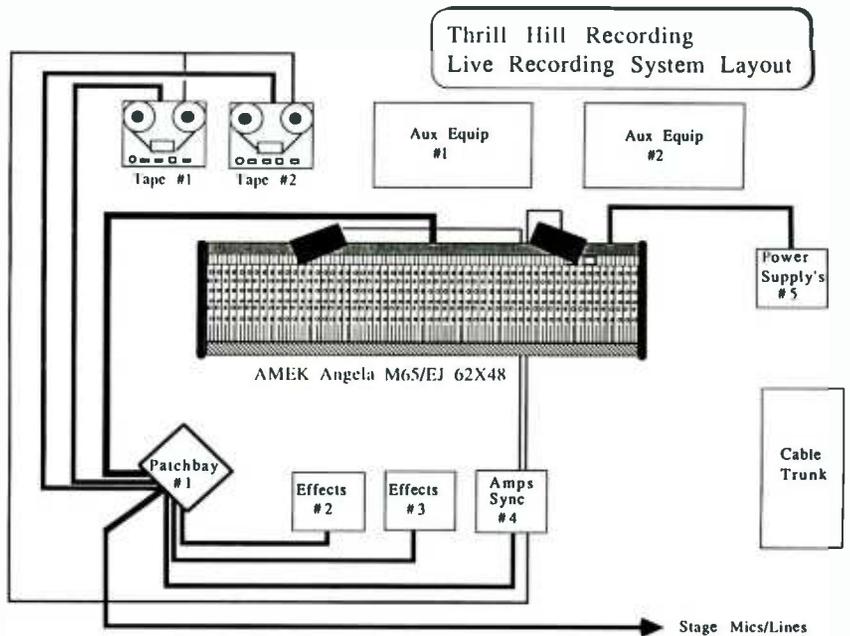


Figure 1. Block diagram of the Thrill Hill Recording System.

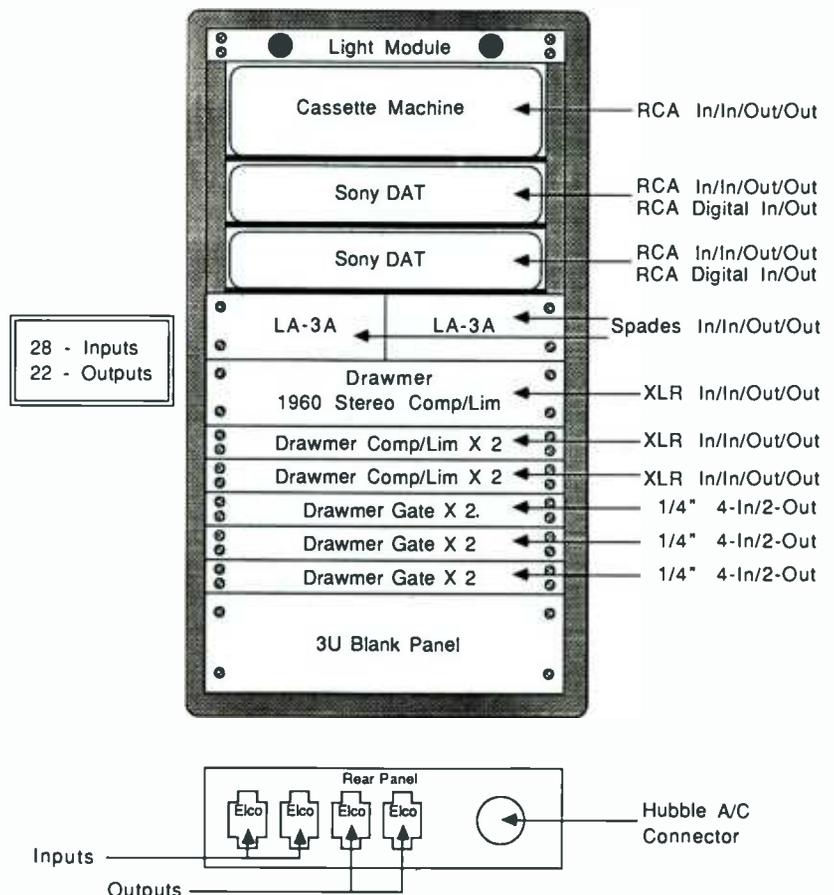


Figure 2. Diagram of Rack No. 5, holding part of the outboard equipment.

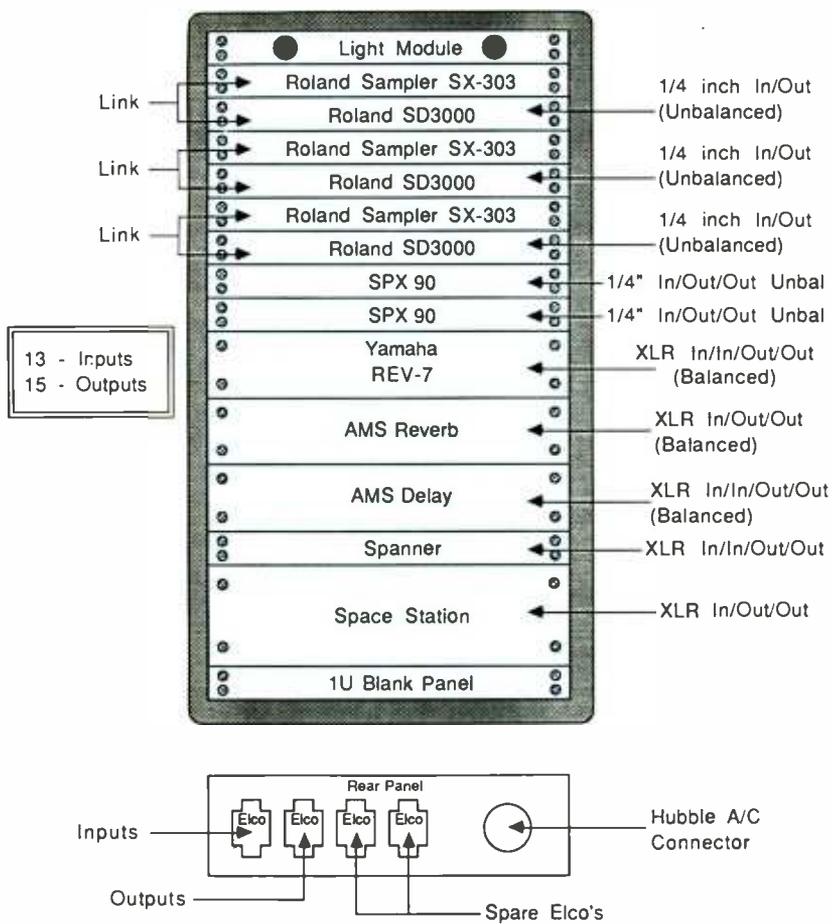


Figure 3. Diagram of Rack No. 6, containing the rest of the outboard equipment.

the show, which is recorded to R-DAT. I use the six auxiliary sends on the Angella for echoes, slaps and other effects I want in the mix—they aren't printed to multi-track, however."

Springsteen and crew have been using the Sony digital multitracks since the 1984 "Born in the USA" tour. Because there was a lot of talk of releasing a live album, Scott said, he believed it would be useful to own a machine rather than rent one for sessions. That machine was then taken from studio to studio.

"The money spent in the digital recording systems may seem like a lot, but it does improve our logistics," he said. "For example, using the PCM-3324 means that I walk out of here after a concert with four one-hour reels of 1/2-inch digital tape and three R-DAT cassettes.

"Before going digital—aside from the improved sound quality—it would have taken 17 to 18 2-inch reels of analog tape and usually 10 1/4-inch reels. Between 1980 and the live album of 1986, I had record-

ed dozens of shows for Bruce—that adds up to a great deal of analog tape. From these 10 Los Angeles and New York shows, I will have around 50 reels of 1/2-inch tape."

At the bottom line, the system is consistent with the way Springsteen has worked all his life. According to Scott, once Springsteen gets together a group of people and a system that work, he sticks with them. Using this system on tour and having it installed in his house simply goes along with that preference.

"He basically wants a studio for the comfort of a continuous environment—and knowing that when he goes into this place, his favorite chair will be there," Scott said. "He needs to work in that relaxed way. The system we are using to record his concert dates, and that will be installed in his home studio, will allow him to be even more creative in the future."



Photos by Elizabeth J. Annas.

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MIDI Control of Effects: One Designer's Viewpoint

By Paul D. Lehrman
with Jeffrey Stanton

Full MIDI implementation can be a tremendous help in streamlining effects creation if total flexibility and independent control over all parameters is provided.

In the modern recording and production studio, the role of effects continues to grow in importance. Music is relying more and more on synthesized and sampled sounds that cry out for processing. Improvements in the quality of audio for film and video demand similar improvements in the sophistication of voice and effects processing.

At the same time, advances in technol-

ogy have made more interesting and flexible effects devices available to the studio operator. In the past, a well-equipped facility might have a plate reverb, a spring reverb, a delay line and possibly a live chamber. Today, it is not uncommon to see more than a dozen digital effects processors, many of which can handle multiple tasks.

Making an effects device as flexible as possible is important if the device is going to see use in many projects and, at the same time, continue to sound fresh. Effects units are still called upon to produce na-

tural-sounding ambiances, but more and more they are required to produce totally unnatural results, or to create "spaces" that couldn't possibly be constructed by human hands.

In effects generation, the best way to achieve total flexibility is to offer independent control over all the parameters that make up a sound. A traditional plate reverb can have its decay time (RT_{60}) adjusted by applying dampers to the metal surface of the plate. When that is done, however, the high-frequency decay characteristic is damped disproportionately, so the sound of the plate at lower RT_{60} values is duller than that of the undamped plate. A modern digital reverb, however, can treat high-frequency decay and overall RT_{60} separately, so it is possible to create a plate that actually gets brighter as its RT_{60} is decreased.

In a visual setting, or when used as compositional elements, effects are often called on to change rapidly. Therefore, some form of automation can be of tremendous help when it comes to mix and assembly. The traditional methods of muting and fading effects sends and returns on a master console can prove inadequate when dealing with a dozen stereo effects devices, each changing its identity every few seconds.

Most modern digital effects devices have storage memories in which groups of settings, or programs, can be preserved and instantly recalled, but without some form of remote control over these, the mixing engineer might still have to grow several extra sets of hands.

MIDI can be a tremendous help in streamlining the processes described earlier. MIDI is now universally accepted and versatile enough to handle most of the

Paul D. Lehrman is *RE/P's* consulting editor for electronic music, and is a consultant to AKG Digital. Jeffrey Stanton is chief software engineer for AKG Digital.

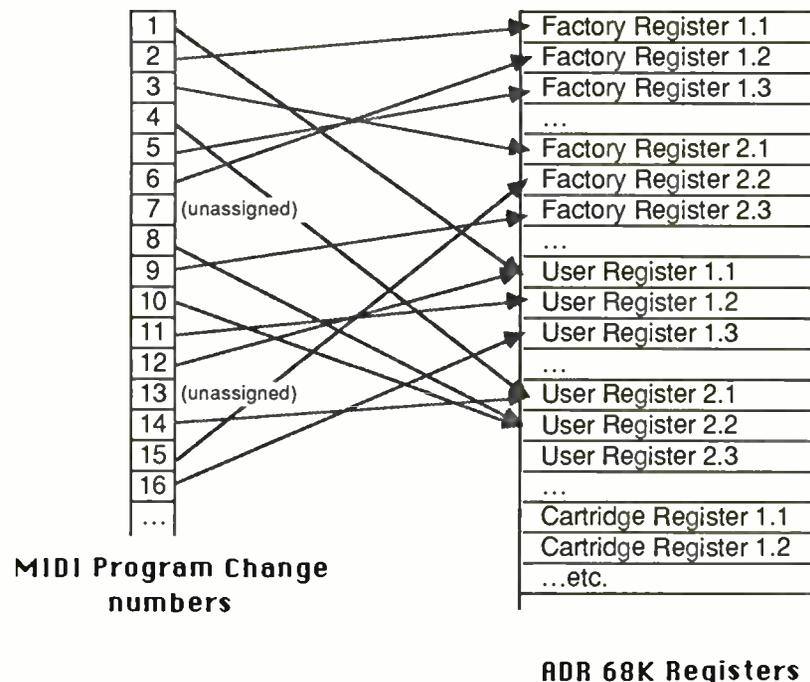


Figure 1. Program change map.

control functions in a typical automated studio. A single MIDI line contains 16 distinct data channels, and each one can carry hundreds of thousands of different commands, at a speed greater than 1,000 commands per second.

Inside the ADR 68K

The ADR 68K is a reverb and effects processor that was designed with MIDI automation in mind. Besides reverb and effects, the device also contains a large memory for sampling, using a 16-bit format, which can be divided into as many as 12 distinct sections, and is capable of recording in stereo. The entire device is a true stereo unit, capable of several different effects simultaneously, and has two independent sets of balanced stereo outputs.

It is equipped with 150 factory presets arranged in 11 banks, with each bank containing programs (which reside in "registers") that use the same basic algorithm, e.g., all the Plate programs are in Bank 1, the Hall programs in Bank 4, the sampling programs in Bank 6, and so on. There is also room for 50 user presets in non-volatile RAM, and another 50 presets can be stored on an external data cartridge.

Although the main electronics of the device are housed in a rack-mount "mainframe," all control functions are handled by a remote unit, which has six programmable faders, some three dozen momentary-contact membrane switches, three 12-segment LED displays for monitoring levels, and a four-line by 40-character LCD.

Within a program, a parameter is adjusted by moving the fader associated with it. The LCD shows which parameters are controlled by which faders. Most programs have more than six parameters, so "arrow" keys are used to change "pages" within a program. Fader 1, therefore, may adjust RT₆₀ in a reverb program when Page 1 is showing, but will adjust gate-trigger level when Page 3 is showing. The buttons and faders are also used for system routines, such as program storage and recall, MIDI setups, and help screens.

The ADR can generate various effects such as: four different types of basic reverb, split reverb, delay, chorusing, stereo simulation, sample editing, reverse reverb, and a Multi-Effects mode that combines many of the functions into one program. To ensure maximum flexibility, separate controls are provided within the reverb programs for low-frequency and high-frequency decay, overall bandwidth, apparent room size, punch, diffusion, randomness, gate trigger levels and closing time, and timing and level for six distinct early reflections. In the delay programs, two discrete digital delay lines are available, each with independently adjustable

delay time, feedback, high-frequency decay, and modulation depth and rate. Other modes are equally complex.

MIDI program changes

Many processing devices use MIDI program changes from a sequencer or "live" controller to recall stored programs. Some use a fixed "map," in which specific MIDI program changes call up specific internal programs, but in more advanced devices, this can lead to problems.

First of all, a device may have far more programs available than there are MIDI program change numbers. (In the ADR, the number of presets can be as high as 200.) Second, live performers may want a particular effect to be associated with a specific patch on their controlling synthesizer, or may not want to reconfigure all of the synth's patches to conform with the reverb's program numbers.

The ADR has a user-definable MIDI Program Change Receive Map, through which any MIDI number can be programmed to call any preset, whether it is in factory ROM, user RAM, or on a cartridge. More than one MIDI number can be assigned to an internal register, which allows the same effect to be used for more than one synthesizer patch. (See Figure 1.)

If a program change number is unassigned or mapped to an empty register, then that program change will be ignored. This can be useful in situations in which available channels are in short supply. The ADR can share a MIDI channel with another device if it is set, for example, to ig-

nore all program changes below 65, while the other device is set to respond *only* to program changes below 65.

The program-change map is stored in non-volatile RAM, and it can be disabled (but not erased) by a master switch at any time. A default map, which addresses the first 10 registers of each of the factory banks, is provided.

One problem with changing effects in real time is that the change is often accompanied by noise as the digital circuitry resets itself. Some more-sophisticated devices, although quieter, often leave a "hole" in the sound when the program changes, as the reverb or delay register for the old program empties out its audio data before the register for the new program begins to fill. The ADR addresses this problem by allowing one program to decay naturally at the same time audio starts being processed under the new program.

This works best when the two programs use the same basic algorithm. The programs don't have to be at all similar, they just need to use the same *type* of effect: a Plate program with a severely damped high end, heavy early reflections, and a 30-second RT₆₀ will fade smoothly into another Plate program with an accentuated high end, no early reflections at all, and an RT₆₀ of two seconds. When moving between programs based on different algorithms, the unit compensates for any level drop by momentarily mixing in a measure of dry signal as the first program decays, then fading it out as the second program begins to be heard.

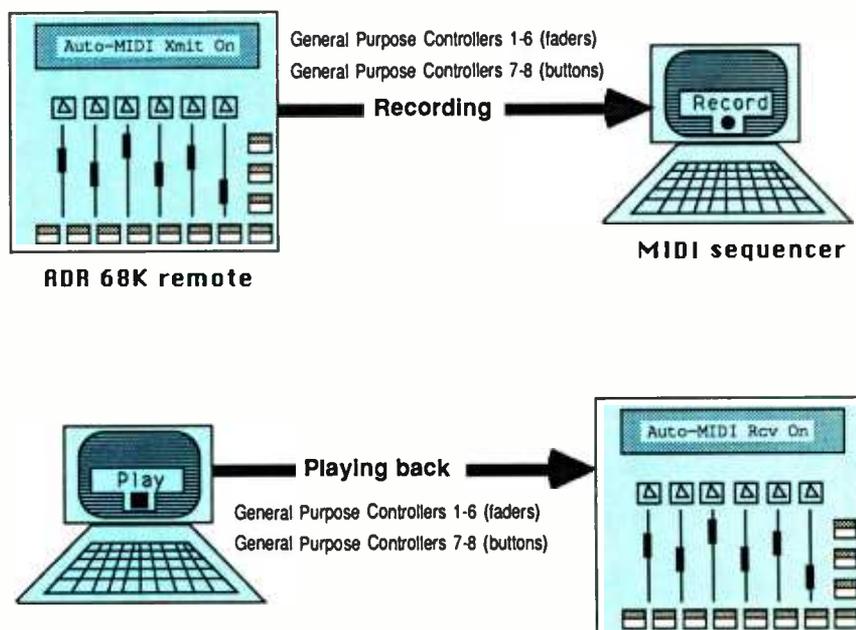


Figure 2. Transmitting and receiving auto MIDI.

"Auto-MIDI"

When effects are being used in a session creatively, engineers often like to "play" their controls, making adjustments in real time as they would on a musical instrument or mixing console. The ADR has a special mode that allows controller movements to be preserved as they are made.

When the "Auto-MIDI Transmit" mode is enabled, every movement of a fader or press of a switch on the ADR remote control is broadcast as a MIDI command. These commands can then be recorded into a sequencer. Playing back the sequence, and turning on the "Auto-MIDI Receive" mode, enables the original controller moves to be faithfully reproduced. (The physical faders themselves don't move, but the displays above them, showing their position, do change.)

As with any automation scheme, deciding how often to update faders is an important design consideration. In the case of the ADR, not only must the internal processing speed be considered, but so must the bandwidth of MIDI itself. The unit's faders are scanned by the software every 100 milliseconds, so that the MIDI commands are sent at a reasonable rate of no more than 10 per second, per fader, which should not tax the capabilities of any sequencer. On playback, the maximum update rate is the same, but parameter changes (whether they are generated locally or received via MIDI) are automatically "ramped," to provide smooth transitions.

Auto-MIDI takes advantage of the "General Purpose Performance Controllers" (GPPCs) described in revision 3.3 of the MIDI Specification. When Fader 1 is moved, it transmits a GPPC 1 command (which is the same as Continuous Controller number 16), followed by a value, or data byte, reflecting the fader's new position, on a scale of 0 to 127. The other five faders transmit similarly on GPPCs 2 through 6, which are Controllers 17, 18, 19, 80, and 81.

Pressing any button sends out a GPPC 7 (Continuous Controller number 82) command, but the *data byte* of the command is determined by which button is pressed. For example, pressing the "Plate" button (which normally calls up a Plate reverb program) sends out a GPPC 7 command with a data byte of 52 (decimal). Releasing a button sends out the same *data byte* as pressing it, but on a different *controller*: GPPC 8 (Continuous Controller 83). (See Figure 2.)

The Auto-MIDI mode is invisible in use and requires no knowledge of MIDI to operate. Anything the ADR can do can be automated, including changing programs, changing pages, accessing a cartridge, recording samples, editing program names

and even calling up the help screens. Control sequences can be as simple or complex as needed, so the mode can be used to automate a few program changes or to record complex sweeps of multiple faders.

There are two important limitations to this mode, however. First, if a sequence containing Auto-MIDI information is started from the middle, the ADR may well be in the wrong state (for example, have the wrong page showing), and therefore will respond incorrectly, inasmuch as Fader 2 may have a completely different function on Page 3 from the one it has on Page 1. Therefore, most sequencers always need to be started from the beginning when using the Auto-MIDI mode.

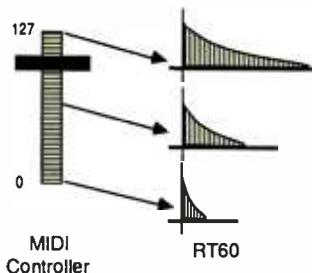
Second, Auto-MIDI information is difficult to change once it is recorded. You can only overdub moves on top of a previously recorded track if you're absolutely sure that the commands you're overdubbing don't contradict or interfere with the previously recorded moves. (Inserting a Page change into the middle of a fader movement, for example, would cause very strange occurrences.) You can safely perform punch-ins only if the punch continues to the end of the sequence, lest the unit be in the wrong state when you punch out. Editing recorded Auto-MIDI information is only possible in a sequencer that allows very precise controller editing, and because the buttons all represent different values of the same controllers, one small slip can have serious consequences.

Parameter control

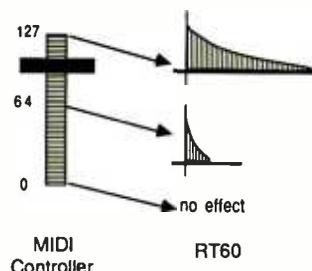
For maximum MIDI control flexibility in both live and studio applications, it is necessary to be able to address all of the unit's parameters individually. Ideally, any MIDI command should be addressable to any internal parameter, and a large number of such mappings should be usable simultaneously.

In the ADR, this is achieved through a third MIDI mode known as "Parameter Mode." It is set up by using parameter maps (not to be confused with the Program Change maps discussed earlier), each of which describes a MIDI source, which can be just about any MIDI command; a destination or "target," which is a program parameter within the processor; and a scaling. Each program can contain as many as 10 such maps.

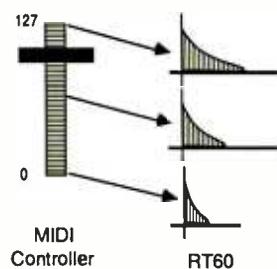
A typical MIDI Parameter map might take MIDI controller number 1, Modulation Wheel, and assign it to Fader 1, which in most reverb programs happens to be RT₆₀. Extreme movements of the modulation wheel would produce very short or very long reverb times, while intermediate settings of the source would be interpolated and produce intermediary RT₆₀ values. (See Figure 3a.)



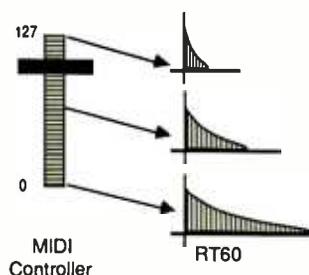
a) standard parameter mapping



b) parameter mapping with limited source range



c) parameter mapping with limited target range



d) "upside-down" parameter mapping

Figure 3. Mapping options.

Ranges for the source and target parameters are set up on the scaling page. Typically, a source range would be the entire range of a MIDI controller, 0 to 127. If the source scaling in the above example were set from 64 to 127, then all modulation wheel values below 64 would be ignored, while a value of 64 would give the shortest RT_{60} . (See Figure 3b.) The target range can also be compressed or expanded (see Figure 3c), and either range can be turned upside down, with a maximum value below the minimum value. Using this same example, higher values of modulation wheel would then produce a shorter RT_{60} . (See Figure 3d.) MIDI controllers with only two values, such as sustain pedal, are normally set up to toggle between two values of a parameter.

Complex MIDI control

The Parameter maps allow the ADR to respond to MIDI in very complex ways. For example, the high-frequency decay of a reverb can be tagged to the incoming note number, so that higher notes on a keyboard sound brighter than lower ones. The trigger level of a gated reverb can be keyed to note-on and note-off commands, so that the gate opens when a key is pressed and closes when it is released. The delay time, and hence the resonant frequency, of a delay line can be controlled by modulation wheel, so that the wheel can serve as a "real-time flanger," while a foot pedal can be used to increase feedback gain, thereby deepening the flanging effect.

Even MIDI clock can serve as an input source. A simple application would be a digital delay whose timing is related to the tempo of a sequence being played. A more unusual application would be a reverb whose RT_{60} is inversely proportional to

Ideally, MIDI parameter maps are stored as part of the program in which they are created.

clock speed, so that slower musical passages would have longer reverb times, and brisker ones would have shorter decay.

Output levels are available as targets within any of the programs, and in programs that use multiple effects, two effects can be balanced by assigning their output levels to the same controller, giving one an upside-down scaling. One interesting application of this might be to set up two delays, one with a rising volume envelope and the other with a falling one, and balance their outputs using a pitch wheel. Movement of the pitch wheel in one direction will now create an echo effect, while

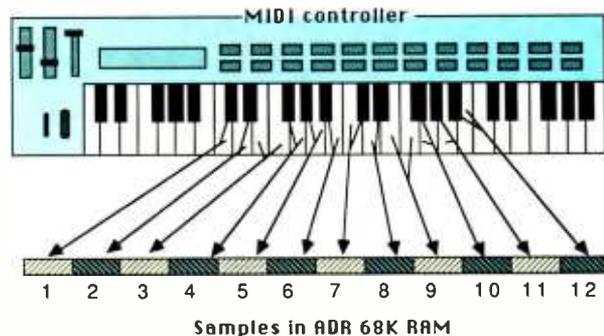


Figure 4. Complex sample triggering.

movement in the other will create a reverse echo.

The unit has a "Mute" button that shuts off the input to the processor, and it can be set up as a Target. By assigning a sustain pedal to set the RT_{60} of a reverb program to "Infinite" and to have it turn the Mute on, the pedal can serve as a "freeze" control. (The same effect can be produced in a delay program by having the pedal toggle the feedback gain to 100%.) MIDI parameter maps are stored as part of the program they are created in, in user RAM or on a cartridge, so when a program is recalled, so is its MIDI map. Two programs in memory can be identical except for their MIDI maps, so that a reverb program may have a constant predelay and then, on cue, can tag it to incoming MIDI clock timings.

In addition to the MIDI commands, the Parameter mode allows other types of information to be selected as Sources. One of these is input level, which allows "intelligent" effects based on the music such as, for example, the level of the incoming signal determining feedback gain in a delay line. For use in live performance, there are four 1/4-inch input jacks located on the back of the unit's remote control, which can be connected to a switch, a control-voltage source, or a simple potentiometer, and then assigned a Target.

Triggering samples

Users experienced with MIDI samplers need to be able to access the ADR's samples in a familiar way. Normally, multiple samples are "played" from the remote control by moving a "Select" fader to the number of the desired sample, and then pressing a "Play" button directly above it. To simulate this over MIDI, a special Source command called "Note Trigger" was devised. This consists of a Note-On command and its note number which, when it is mapped to a Select fader, "moves" the fader and presses the button above it,

thereby triggering the sample.

The Source scaling of the Note Trigger command is the range of MIDI notes. In practice, the distance between the minimum and maximum values of the source (the lowest and highest notes) should be set to an even multiple of the number of samples in use. For example, if there are 12 samples in memory, the note range could be set to C5-B5, so that each sample would be triggered by a different key in that octave. If the range were to be doubled, i.e. C5-B6, then each sample would have two keys to play it, like some drum machines.

Notice that the ADR has to have the ability to respond to this command backwards: The actual MIDI string places the note-on command first, followed by the note number, while the ADR must know the note number first, so it can move the Select fader to the appropriate place, and then execute the Play command. Another Source "command combination" is "Note Trigger/Release." When this is mapped to a Select fader, it moves the fader and triggers the sample, and then cuts off the sample when a Note-Off command is received.

Sample triggering can occur while reverb and effects programs are active, and it is with this combination that the large number of available Parameter maps can be used to great advantage. Consider a group of samples being played back through a multi-effects program: Key number and velocity determine the sample number and volume; modulation wheel turns on a chorus; portamento pedal turns on a repeating delay, while foot pedal sets its speed; sustain pedal turns on an infinite reverb; and aftertouch controls left-to-right movement in a stereo multitap delay. (See Figure 4.)

A different mode on the ADR allows one sample (up to 32 seconds long) to be recorded and played back, with control over the playback rate, i.e. pitch. The pitch is changeable up to one octave higher or

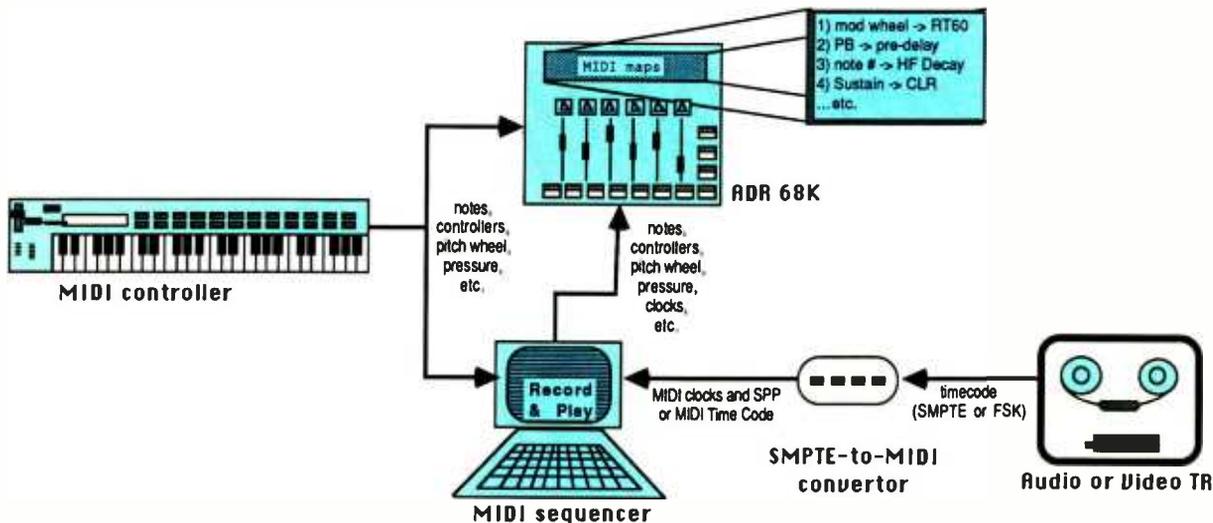


Figure 5. Using the parameter mode in the studio.

lower, and both coarse and fine adjustment faders are provided. Using the Parameter mode, the Note Trigger command can be assigned to play the sample back, with note numbers (typically two octaves' worth) corresponding to the pitch-change range. In addition, pitch wheel or another continuous controller can be assigned to the fine pitch adjustment, so that the device can be played like a real synthesizer. To add to the effect, MIDI velocity can be mapped to output level.

Parameter mode in the studio

Because so many different types of MIDI installations exist, MIDI-controllable processors need flexibility. Some MIDI keyboards have little or no capacity for generating continuous controller information, and some sequencers do not record and/or edit such information very well, and so MIDI note numbers become the only means of control. At the other end of the scale are keyboards that generate polyphonic aftertouch, wind controllers that generate breath-control information, and sequencers that allow editing, copying, and remapping of controller information. The ADR's Parameter mode is flexible enough to allow the unit to be used in any of these settings.

The Parameter mode is also more flexible when it comes to dealing with sequencers. Editing controller values is not nearly as finicky a process as it is in the "Auto-MIDI" mode. A sequence controlling the unit can be started in the middle, as long as the sequencer includes a "chase" function. Such a feature, which is now included in several high-end sequencers, allows the program to look backward through a sequence prior to the current starting point, and find and execute the

most recent program and controller changes—thereby bringing any device reading them "up to date"—before the music starts.

Even in a session that is not using MIDI instruments, a sequence can be set up that contains nothing but commands for the ADR and this sequence locked to the master tape with a SMPTE-to-MIDI converter. The timecode track then becomes a master automation track for the ADR, exactly like the computer in an automated console. (See Figure 5.)

At this point, it should be noted that we hope the ADR is a lot easier to use than it is to talk about. We understand that complex MIDI control of effects is a new concept for many, and to help, the authors have produced a cartridge containing 50 new presets for the unit that use MIDI Parameter maps, ranging from simple to complex. Thoroughly documented, the presets are both useful in their own right and serve as "starter" programs for users.

Sysex and dumps

Like a synthesizer, the contents of any or all program registers (i.e., the parameter settings of the programs) can be sent or received using MIDI system-exclusive commands. This can be useful for one ADR to communicate with another, or for storing programs in a dedicated "librarian" program on a computer. The system-exclusive loading and dumping procedure is straightforward with no complex handshaking, which allows use with "generic" librarian programs to be used easily.

Since the sample memory is volatile, a way of off-loading samples had to be devised. In fact, two such ways are included. The unit can send and receive samples using the MIDI Sample Dump Standard,

which allows it to exchange information with other samplers, as well as with computer programs designed to edit samples. But this method has drawbacks. Chief among them is that most existing sample-editing software can deal only with 12-bit samples, and if the ADR's 16-bit samples are run through such software, even if only for archival purposes, some fidelity will be lost. Another disadvantage is that the Sample Dump Standard has no provision for dealing with stereo samples.

Therefore, an alternative method that uses system-exclusive data is provided. It overcomes these problems and is also faster than the Sample Dump Standard. Its obvious disadvantage is that samples stored in this format can be used only by the same or another ADR.

What it means

MIDI is as an effective way to control and automate signal processing in both the studio and live performance. MIDI by itself gives us flexibility, precision, and repeatability. It is up to the designers, of both hardware and software, to add comprehensiveness, intelligence, and user-friendliness.

We hope that the work we've done on the ADR 68K will show the way for many more MIDI-controlled processing devices of all kinds, and will help MIDI to develop even further as a tool for tying professional audio equipment together, giving the artist and the engineer more creative power, and allowing the studio to work harder, and better.

RE/P

Equipment Leasing Made Simple

Leasing is 100% financing, with the lessor buying the equipment from a vendor for cash and leasing it for a specified period of time to a lessee.

By Tom Roche

Leasing is recognized as the fastest-growing and largest external source of capital for equipment in the United States. For most companies needing small-ticket equipment (under \$25,000), leasing is less a matter of cost than it is of convenience. In 1985 alone, 85% of American companies, including 65% of the Fortune 1,000 companies, used leasing to acquire capital equipment.

Simply put, today's studios are leasing because it is a practical and cost-effective method to acquire equipment. Leasing provides the use of the equipment for extended periods of time, and at the end of the term, provides the opportunity of ownership. And because lease payments are based on fixed, rather than adjustable, interest rates, the monthly payments are set for the term of the agreement. This process permits a company to become more flexible in its cash flow management.

At the maturity of a lease, the lessee's options are:

1. To buy the equipment at fair-market-value, often estimated to be 10% of the original purchase price, at the end of a 3- to 5-year lease.
2. To buy the equipment at fair-market-value, whatever fair-market-value is at the end of the lease. Typically, this purchase option offers the lessee lower payments throughout the term.
3. To return the equipment to the lessor.
4. To retain the equipment by renewing the lease for an extended term.

Additional provisions that can be added are irregular payment schedules, changing lease rates, purchase or renewal op-

tions, equipment upgrade options, early cancellation options and guarantees. Each condition will affect the transaction, but additional costs are justifiable if the services enhance the value of the equipment. A lease agreement usually covers a period of 12 to 72 months; occasionally, it is extended up to 10 years for certain high-ticket items, such as large consoles.

Leasing is a practical and cost-effective method to acquire equipment.

A lease is a contract. It might consist of a single transaction with a single piece of equipment, or it can be termed a "master lease" with multiple equipment transactions or a continued arrangement to acquire equipment. Any individual, sole proprietorship, partnership, corporation, Subchapter S corporation or trust can become a "lessee" under a lease agreement.

Most new or used property that remains useful after the final purchase year and can be depreciated for the purpose of income can be leased. It is the use, not the ownership, of equipment that produces revenue, although leasing is sometimes viewed as a contingency situation. The cash flow in and out of a business can be strengthened by reducing payments. Lease payments are frequently smaller than debt-financing payments. The overall "effective rate" is usually less than current loan rates if tax benefits are properly used. The "effective rate" is the interest rate.

Leasing is 100% financing, with the lessor buying the equipment from a vendor for cash and leasing it for a specified

period of time to a lessee. This practice is commonly called third-party leasing. Its roots lie in a 1956 Supreme Court ruling that directed IBM to sell, as well as rent, its computers.

Companies should plan and select the correct balance of equipment leasing and ownership to aid in managing their acquisitions. Before making decisions, businesses should evaluate the risks they wish to assume and risks they need to avoid. The lessee can then enjoy increased productivity and profit while the lessor assumes the credit, equipment and legal risks.

Lessors can also help in selection of equipment, warranties, guarantees, delivery and other options specified at the outset of the transaction. The term of the lease and whether the sales tax, delivery and installation charges will be included as part of the package can be negotiated with the lessor. By signing the lease, the lessee then assigns the purchase rights under contract to the lessor. When the property is delivered, meeting all specifications satisfactorily, the lessor pays the dealer or manufacturer for the equipment, and the lease becomes effective.

A lease agreement usually covers a period of 12 to 72 months.

Frequently, taxes, insurance and maintenance are the responsibility of the lessee. These items are not deducted from the monthly payment. The monthly rental payment is net to the lessor.

Every company acquiring equipment

has a unique profile: cash flow, credit situation, financial balance sheet and type of business. Start-up companies, new in a particular market, typically have limited credit lines, little cash on hand and are uncertain about equipment needs—present and future. Established businesses typically need to keep credit lines free, complying with creditors. Also, leasing helps offset tax liabilities with new tax incentives.

Incentives

Leasing continues to grow in popularity because of the Tax Reform Act of 1986. A unique feature of this act is attracting purchasers who may never have considered leasing in the past. The Alternative Minimum Tax (AMT) creates an incentive to lease. Many choose to lease now to reduce their dollar amounts of "preferences," including the accelerated depreciation write-off of owned equipment. Taxpayers subject to AMT are unable to realize the full value of accelerated depreciation. The result is that equipment costs go up. Because of a new rule that limits depreciation deductions, many companies that typically purchase equipment in the fourth quarter are more likely to lease. Also, by leasing from a company

that can use the tax incentives, a savings will result from lower lease payments. The tax act has spurred even more ingenuity in an already competitive field by keeping rates lower and thus encouraging innovation.

Leasing continues to grow in popularity because of the Tax Reform Act of 1986.

Just as businesses are unique, so are the many forms of leasing. A "typical" lease transaction does not exist, because each business needs different equipment at different terms. This difference sets leasing apart from other forms of financing. Lessors are risk-oriented, and commercial lenders are credit- and liabilities-oriented. Lessors are concerned with equipment management and remarketing and are generally more flexible than commercial lenders. Every transaction can be tailor-made to fit the specific needs of the business client. A "full payout" lease implies the lessor will receive from the lessee the cost of financing, overhead and acceptable return on investment in payments structured over a specified

period of time. The term of an "operating lease" is shorter, and before the payments have recovered the equipment cost, the machine is returned to its original owner. Shorter-term leases are advantageous because they help users stay ahead of equipment obsolescence and accelerate depreciation.

The leasing environment ultimately depends upon the need for growing businesses to acquire new equipment. A market for tax-oriented leasing will remain, because companies subject to the AMT will find it less costly to lease from lessors that can take advantage of tax benefits associated with leased assets. This scenario typically applies to larger-ticket leases.

The pro audio industry is experiencing tremendous growth by using leasing as an expansion tool. It seems that once studios have their "A" rooms completely booked, they realize they can use leasing to build "B" rooms and have the generated income pay the lease payments. As you can see, they've expanded their businesses with little or no cash outlay, which should improve their profit margins. Leasing is expected to grow even more rapidly in the future as companies adjust to the new tax changes.

REP

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Circle (33) on Rapid Facts Card

Active Balanced Inputs and Outputs

By Richard Cabot

Don't become unbalanced over hum and noise.

System interfacing problems have existed since the first audio engineer tried to wire two pieces of equipment together. Equipment that works flawlessly alone can fail miserably when wired together. One of the first problems encountered was ground loops.

Early audio workers used transformers to solve differences in ground potentials between equipment or signals picked up in cabling. These transformers helped to usher in the age of balanced and floating interfacing. However, they also added significant cost, weight and distortion to the equipment.

As the performance of the electronic equipment improved, these shortcomings became more noticeable. The large size of the transformers also made them inconvenient for use in the ever-shrinking chassis of transistorized equipment. As solid-state technology became more common, designers began to look for ways to eliminate the use of transformers. The result was electronically balanced inputs and outputs (I/O).

Let's first examine what a *balanced line* is and what problems it solves. Figure 1 shows a basic source and load connection. No grounds are present, and both the source and load float. This is the optimum condition for equipment interconnection.

Either the source or the load may be tied to ground with no problems, provided only one ground connection exists. Unbalanced systems are created when each piece of equipment has one of its connections tied to ground, as shown in Figure 2. An example of this is if the source is an amplifier output whose power supply

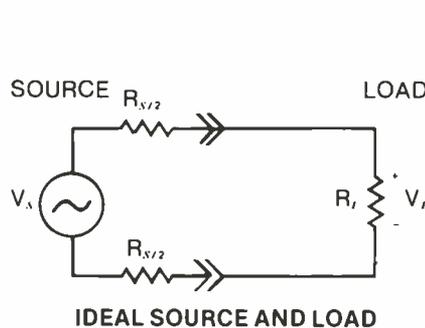
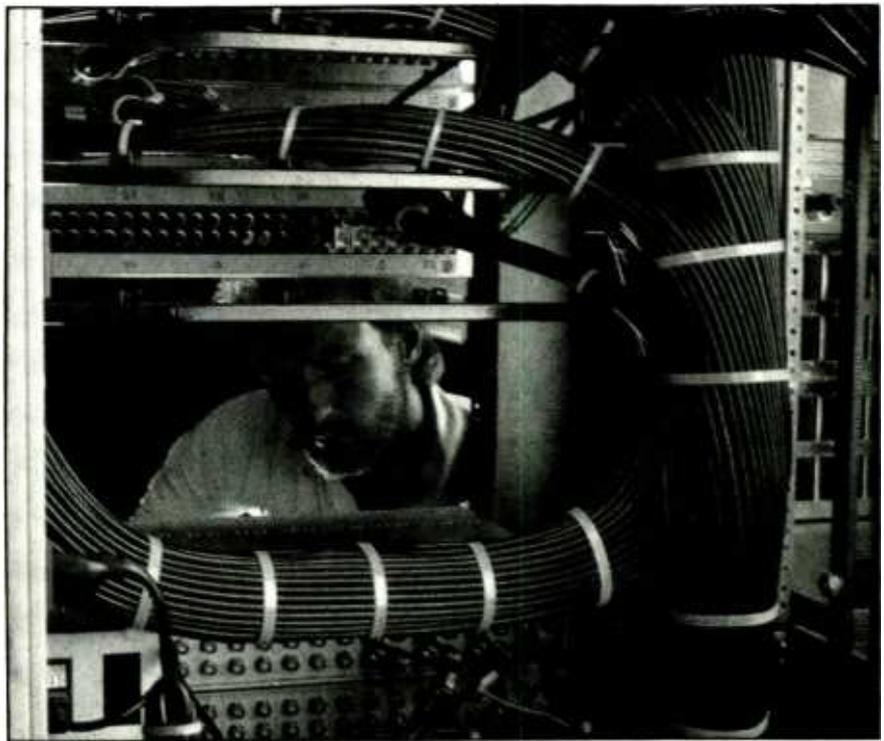


Figure 1. A basic source and load connection. No grounds are indicated, and both source and load float.

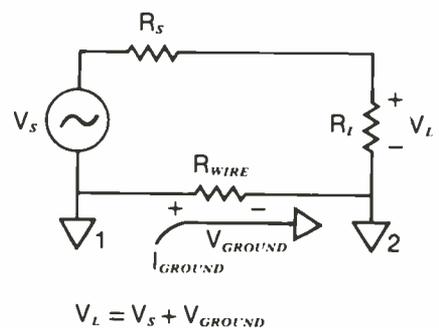


Figure 2. An unbalanced system in which each piece of equipment has one of its connections tied to the ground.

Richard Cabot is vice president and principal engineer with Audio Precision, Beaverton, OR.

is tied to the chassis. The difference in ground potential causes current to flow in the ground wire and develop a voltage across the wire resistance. The ground-noise voltage adds directly to the signal itself. Because this ground current is usually from leakage in power transformers and line filters, the current is 60Hz ac and gives rise to hum. Reducing the wire resistance through heavier ground wire helps, and the hum will be reduced, but it is difficult to get an adequately low resistance.

By amplifying both the high side and the ground side of the source and subtracting the two, it is possible to cancel the ground loop noise (see Figure 3). This is the basis of a *differential input* circuit. Unfortunately, the cancellation runs into trouble when the source impedance of the unbalanced source is taken into account. One side of the line will have a slightly lower amplitude because of the attenuation of the source impedance.

By creating a signal that is out of phase

with the original, you can make the source balanced and eliminate this error, as shown in Figure 4. An added benefit is that for a given maximum output voltage from the source, the signal voltage is doubled over the unbalanced case.

CMRR

The measure of how well an input rejects ground noise is called the *common-mode rejection ratio* (CMRR). (See Figure 5.) If a differential input is used to reject noise, the desired signal is applied between the plus and minus amplifier inputs. The amplifier will have a certain gain for this signal condition, called the *differential gain*. Because the ground-noise voltage appears on both plus and minus inputs simultaneously, it is common to both inputs.

The amplifier subtracts the two inputs, giving only the difference between the voltage at the two terminals at the output. The gain under this condition should be zero, in practice, but it is not. CMRR is the ratio of these two gains in decibels. The larger the number, the better. For example, a 60dB CMRR means that a ground signal common to the two inputs will have 60dB less gain than the differential signal. If the ground noise is 40dB below the desired signal level, the differential input will make it 100dB below (the desired signal level). However, if the noise is already part of the differential signal, the CMRR will do nothing to improve that signal.

Common-mode range is a specification of the largest common-mode signal that can be handled at the input without clipping or other malfunction. Virtually all active input stages are adequate in this parameter, being able to handle several volts of common-mode signal. If the common-mode signal is higher than this, the system grounding is grossly inadequate. However, with 1V of common-mode signal, the CMRR of a typical active input (about 60dB) will not provide adequate signal-to-noise ratio. In this situation, it's time for a transformer.

Active-balanced input circuits

One approach to active-balanced inputs is shown in Figure 6. The positive input is buffered and inverted by an inverting op-amp stage. This signal is then added to the signal from the negative input in a second inverting amplifier stage. Any common-mode signal on the positive input has been inverted and will cancel when it is added to the negative input signal. Both inputs have the same impedance and can be easily protected from overloads because of the large input resistor. The matching of resistors limits the CMRR to about 50dB without adding

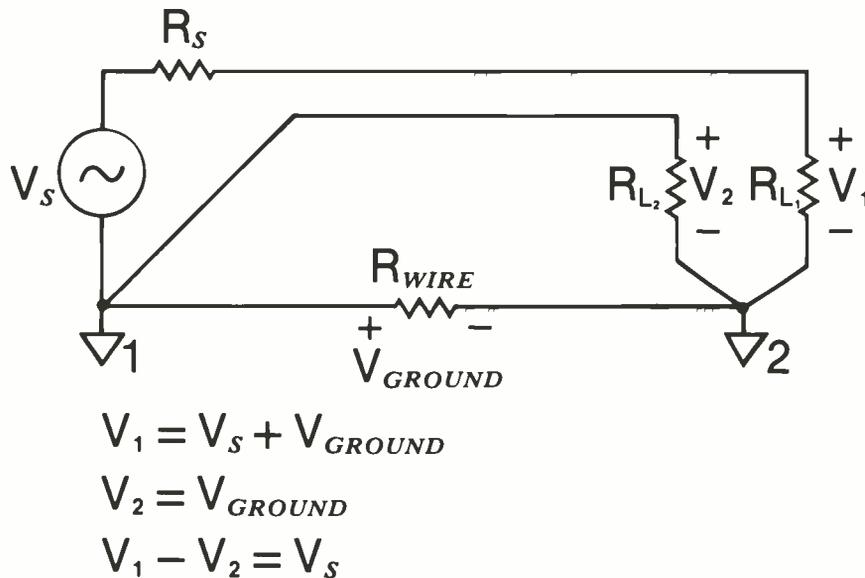


Figure 3. The ground-loop noise can be canceled by amplifying both the high side and ground side of the source and subtracting the two.

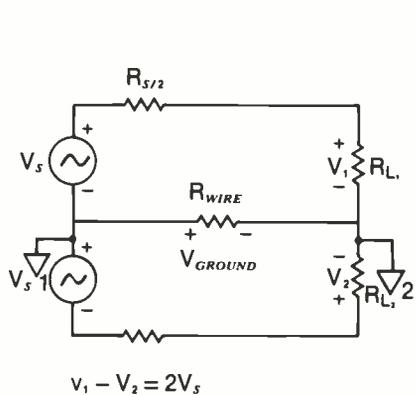


Figure 4. A balanced source where the amplitude error is eliminated.

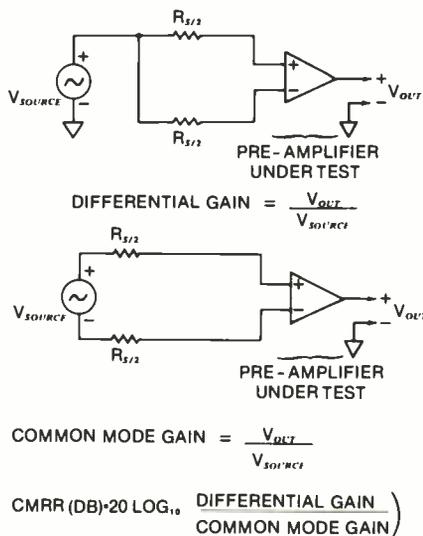


Figure 5. Illustration of common-mode rejection ratio (CMRR).

adjustments. With the addition of an adjustment, it is possible to achieve 80dB CMRR, but component aging will degrade this substantially over time.

The most simple and least expensive active-balanced input is the single op-amp circuit. (See Figure 7.) For a unity gain stage, all of the resistors are made the same value. This circuit presents an input impedance to the line that is different for the two sides. The positive input impedance will be twice that of the negative input. This does not cause a problem except in applications in which many of these inputs are paralleled. The CMRR is dependent on the matching of the four resistors and the balance of the source impedance. The noise performance of this circuit, which is usually limited by the resistors, is a trade-off between low loading of the line and low noise.

Adding a pair of buffer amplifiers before this circuit results in an *instrumentation grade* input, as shown in Figure 8. The input impedance is increased greatly, and source-impedance effects are eliminated. Additional noise is introduced by the two added op-amps, but the resistor noise usually can be reduced by dropping impedances, causing a net improvement in noise. Adding resistors to the input stage makes it possible to add gain while maintaining the low-noise performance. This gain also increases the CMRR by the same amount of the gain in decibels. However, the amount of gain that can be used is limited by the line level and the clipping point of the op-amp.

Active-balanced outputs

Early equipment with active-balanced output stages used the approach shown in Figure 9. The signal was buffered to provide one phase of the balanced output signal. This signal was then inverted with an op-amp inverter to provide the other phase of the output. The outputs were taken through two resistors, which are half of the desired source impedance. Because the load is driven from between the outputs of two op-amps, the maximum output voltage is about 20V, double that of an unbalanced output. This circuit works reasonably well if the load is always balanced, but it suffers from two problems.

The first problem arises when unbalanced loads are being driven or when one side of the signal is inadvertently shorted to ground. If the negative output is shorted to ground by an unbalanced load connection, the first op-amp is likely to distort. This produces a distorted signal at the input to the other op-amp. Even if the circuit is arranged so that the second op-amp is grounded by unbalanced loads, the distorted output current probably will show up in the output from coupling

through grounds or circuit board traces. Equipment that uses this type of balanced output often provides a second set of output jacks, which are wired to only one amplifier for unbalanced applications.

The second problem is that the output *does not float*. If any voltage difference (power-line hum) exists between the local ground and the ground of the device receiving the signal, it will appear as an addition to the signal. The only ground-noise rejection will be from the CMRR of the input stage at the receive end.

A few companies offer unbalanced outputs that sense ground at the load and attempt to reject hum voltage. A schematic of such an arrangement is shown in Figure 10. This type of circuit often encounters problems when driving long lines. However, in short interconnect situations the circuit can work well.

These circuits do not lend themselves to driving several loads in parallel, because there is no longer a single remote ground to sense. They also have a problem patching into other devices because the load must be unbalanced, and the ground must be isolated from the patchbay or system.

Several manufacturers now are offering electronically balanced and floating outputs on their products. The basis of these designs is shown in Figure 11. The circuit consists of two operational amplifiers, which are cross-coupled with positive and negative feedback. The output of each amplifier is dependent on the input signal and the signal present at the output of the other amplifier. These designs may have gain or loss depending on the selection of resistor values. Also, the output impedance may be set via appropriate selection of resistor values. Some resistance is needed from the output to ground to keep the output voltage from floating to one of the power-supply rails.

Because of the added resistors to ground and any output ac coupling, the impedance from the output to ground is not infinite. This reduces, but does not eliminate, the coupling of ground noise. If care is not taken with device compensation, stability problems also may result.

Designing the output stage with a gain of two allows the signal to drive balanced loads to approximately 20V from standard op-amp supplies, as with the previous circuit. However, when unbalanced loads are used, the voltage swing is limited to one-half this value before clipping occurs.

If the circuit is designed for unity gain, it will not clip until the circuits driving it clip, but the output voltage is halved. Aside from these minor problems, the circuit works well and can exhibit bandwidth superior to that of a transformer output stage at significantly lower cost.

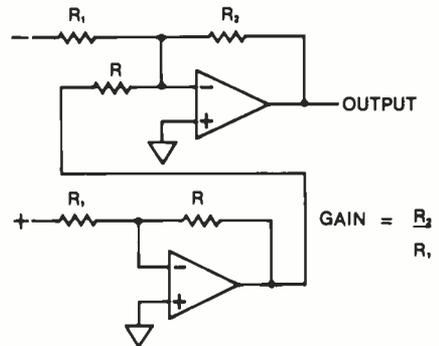


Figure 6. Without adjustments, this circuit will provide about 50dB of CMRR.

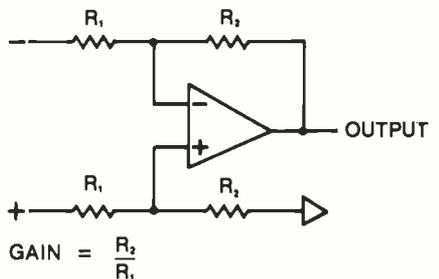


Figure 7. The simplest and least expensive active-balanced single op-amp circuit. Performance depends on resistor-matching and the balance of the source impedance.

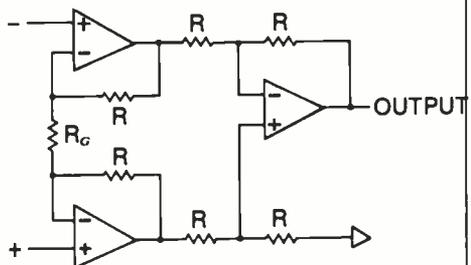


Figure 8. The two additional op-amps create an instrumentation-grade input circuit.

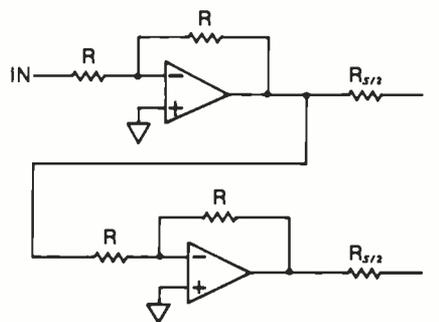


Figure 9. This simple balanced output can produce about 20V, double that of an unbalanced output.

System noise

By Terry Pennington

In electronic circuits, noise is being generated in all conductive elements regardless of whether a purposeful signal is applied to the system. Consider the hiss you hear on an input channel when the gain is raised to a normal operating level. This noise is an amplified version of the random electron activity, thermal noise, taking place in the components at the input of the channel and all successive gain stages.

These components include anything in the dc portion of the circuit such as resistors, transistors, wire, vacuum tubes or whatever else has been included in the paths of inputs. Usually, ac components (capacitors) are not included in the noise equation unless they are operating as something other than a pure capacitance, which sometimes happens with capacitors that are improperly applied.

Once the thermal noise at the input stages of the amplifiers is established, it is amplified (multiplied) by the gain of all succeeding amplification stages. In a normal microphone to line-level situation, as much as 80dB of gain may be present. As a result, the noise developed at the input could increase by 10,000 times.

Terry Pennington is director of technical marketing and product development at Rane Corporation, Everett, WA.

In practice, the input noise of a good amplifier may be as low as $3nV$ to $4nV$ ($10^{-9}V$) times the square root of the bandwidth (usually 20kHz), which yields a noise level of about $0.5\mu V$ ($10^{-6}V$) over the audio range. Multiplying this number by 10,000 (the gain of the pre-amp) yields a noise level of $5mV$ ($10^{-3}V$). This level, by any standard, represents a lot of noise and is unavoidable.

Active noise

The two types of noise, passive and active, are both important in terms of equipment design. For this application, however, the noise that causes the most concern almost always lies in the active components or even in the device's basic design philosophy.

The most common gain block used in modern processing equipment, the operational amplifier, has an equivalent input noise specification that supplies the required information to indicate how much noise will be applied at its input. The amount of noise present is a function of the conductive density of the input stage, just as it is with the density of a passive component.

In resistors, the higher the conductive density, the lower the noise that follows with a lower resistance. High densities create less noise than low densities. An operational amplifier may be designed in such

a way that the physical area of the transistors is larger and operated at relatively high current to reduce the amount of noise generated.

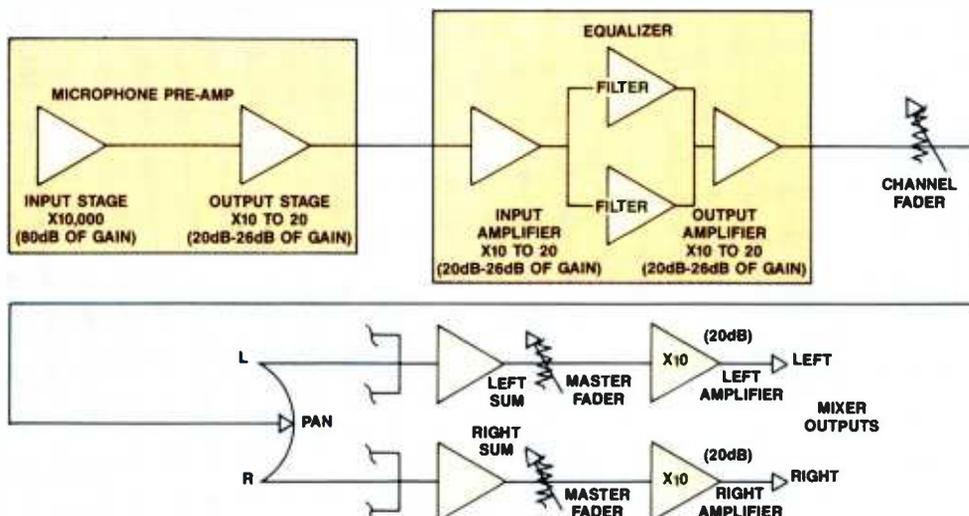
The same sort of trick has been used in discrete amplifiers for special applications, such as instrumentation pre-amps and moving-coil phonograph cartridge amplifiers. In these cases, many small-geometry transistors are connected in parallel to effectively increase the area and, therefore, the density of the active elements. This has the same effect as one large transistor.

Noise in practice

The net impedance (resistance) at the input of a microphone pre-amplifier has a great deal to do with the ultimate noise performance of the amplification process. The microphone's impedance ends up paralleling the input impedance of the pre-amplifier, thereby reducing the overall impedance to a level that is low enough to guarantee satisfactory noise performance.

Let's look at an example microphone pre-amp with an unloaded input impedance of $10k\Omega$. With nothing connected to this input, and the gain raised to its maximum, a great deal of noise probably would be present.

One reason for this is the inherent noise of the elements in the input stage as they rub their electrons



Simplified block diagram of an audio mixer, showing points in the signal flow where noise is amplified.

together. The second noise source is the hum fields and radio signals that enter the system through this unterminated high impedance. This noise is amplified to an audible level by the time it reaches the output. High impedances make great antennas for all manner of undesirable background noises.

To prevent these problems, avoid high impedances at all costs. It is safe, however, to allow high impedances to exist if they will be reduced by external means when the circuit is in actual use.

Most amplifier inputs have fairly high impedance when open, and for a good reason. It is assumed that a low-impedance source will be connected to the input, which will lower the net result and improve the noise performance. If the input impedance were low to begin with, the input would load the source and reduce the level at that point. This is not a good idea because the loss incurred must then be compensated with extra gain, which, in turn, raises the noise.

Example case

Let's consider the example of a 600Ω source driving a 600Ω input. When this occurs, exactly one-half the signal voltage will appear across the source and the other half across the input. Because the signal across the input is all the amplifier can see, half of the signal is effectively lost.

If the input impedance is $10k\Omega$, the vast majority of the signal will appear across the $10k\Omega$, with only 5% of the original signal remaining across the 600Ω source. This equates to a loss of only -0.5dB , as opposed to -6dB with the input impedance set at 600Ω .

All the concepts that can be applied to microphone inputs can be expanded to include line-level inputs and gain structures. The most basic rule is to provide all outputs with as low an impedance as possible and, conversely, to ensure that all inputs are at as high an impedance as possible. Doing so will ensure minimum signal loss (requiring less make-up gain and, therefore, less noise). It also will ensure that the net input impedance of each gain stage will be as low as possible to minimize thermal noise and induced pickup of extraneous signals.

Interfacing problems

Susceptibility to radio-frequency interference (RFI) is a common problem with active-balanced inputs. Strong radio signals often can be rectified by nonlinearities in the input operational amplifiers or transistors. Although wide-band, low-distortion circuits will be less prone to this problem, they are not immune to it. Therefore, any signals that are outside the range of the active circuits must be filtered out before they are inadvertently demodulated. To reduce the problem, most manufacturers add small series resistors and capacitors to ground at the input terminals.

Inductors also may be added, but they are susceptible to external magnetic fields. If package shielding is inadequate, the inductors may pick up as much garbage as they are supposed to filter out.

Toroidal inductors usually will reduce substantially the pickup of external signals. A schematic of a typical input RFI-rejection filter is shown in Figure 12. For the reasons just cited, and because of cost, the inductors often are omitted.

Consider the application

The clipping point of active-balanced inputs and outputs is not always what you might expect. The active-balanced output stages discussed previously can deliver as much as 20V into a high-impedance balanced load. However, when the stages are unbalanced, the maximum output is cut in half. The maximum output specifications of any product you intend to use should include the conditions under which the level must be reduced and the amount of reduction for unbalanced operation.

Output floatability often is unspecified in actively balanced and floating outputs. When a remote power amplifier is being driven, such as in a large studio complex or for a remote broadcast, a large potential difference may exist between the equipment chassis. This problem often is the result of the equipment being on different phases of the power line.

A transformer floating output will have a small capacitance from the center tap of the balanced line to ground and will induce a little 60Hz common mode onto the line. An active-balanced and floating output, on the other hand, has an impedance from each output to ground, typically several tens of thousands of ohms. This induces a common-mode potential between the chassis, which must then be rejected by the CMRR of the balanced differential input.

Mixing balanced and unbalanced outputs and inputs can be done if care is taken in planning where signals go and how wiring is to be performed. It's important to remember that the ground of one

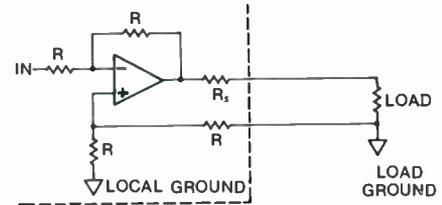
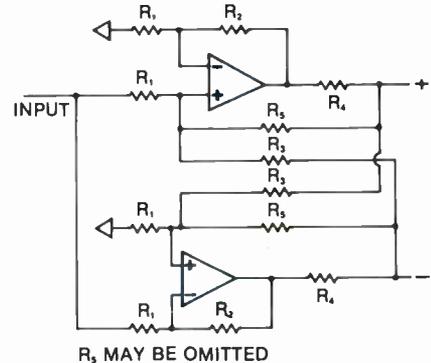


Figure 10. The remote ground-sensing output should not be used when driving long lines or parallel loads.



R_3 MAY BE OMITTED

Figure 11. An electronically balanced and floating output can exhibit bandwidth performance superior to that of a transformer.

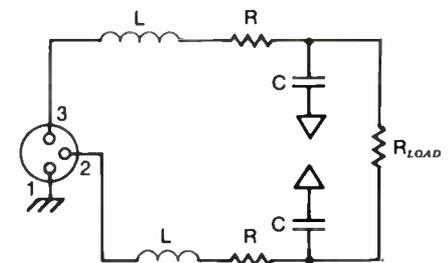


Figure 12. A typical input RFI-rejection filter. Note that the inductors are sometimes eliminated because of cost or susceptibility to noise pickup.

device is not quite the same as the ground of another.

The biggest problems will arise with patchbays because all outputs and inputs can be thrown together in potentially random order. In this situation, it is best if all inputs are of one type and all outputs are of another type. Otherwise, one repatching of an effects device can destroy the S/N of the entire system.

Although active-balanced I/O circuits are not as good as transformers for rejecting ground noise and RFI, they usually are adequate if well-designed. Because of their advantages in cost, weight and low distortion, they are generally the best choice for a circuit in all but the most difficult situations.

REP

Talkback

Moving with Murphy

By Todd Lockwood

I'd like to offer an addendum to Douglas Beard's article, "Shopping for a Used 2-Inch Tape Machine" (July), and to those considering the purchase of a large piece of used equipment as well. The basic message is this: Plan on the worst when it comes to shipping large pieces of gear by common carrier or "electronics movers."

In the past six years, my studio has purchased two new 24-track recorders and a used Neve console. These three purchases happen to be the largest (and heaviest) pieces of gear we've had shipped to us during that period. All three shipments were near-disasters.

Our first 24-track machine was shipped from the distributor via a large air freight carrier. In our excitement to get the machine installed, we failed to notice that the front of the machine housing has been dented, indicating that the shipping crate had fallen over somewhere in transit. Though the machine was insured during shipment, by the time we noticed the damage, we had already made our first shipping blunder: We had thrown away the packing material. We had to sue the air freight company to get any compensation for the damage.

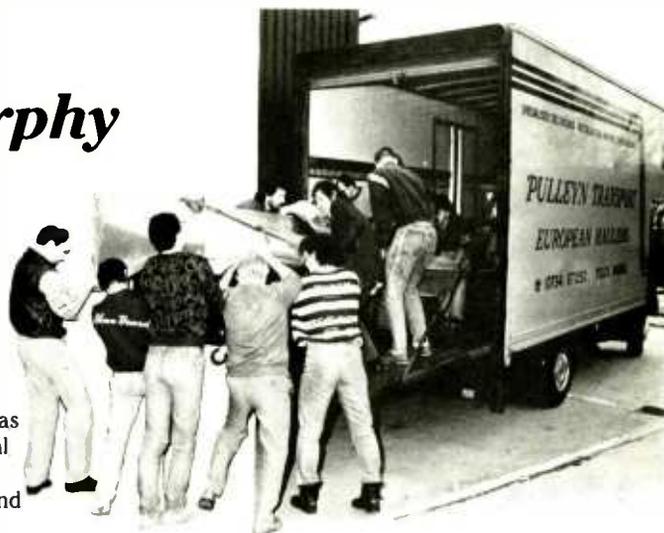
The second fiasco involved a used 8068 recording console that we had purchased from a studio in Alabama. I flew to Alabama to see the console, and it was shipped to us a couple of weeks later via a nationally known "electronics mover." The mover showed up with a crew of two men. They went ahead and tried to move the delicate, 500-pound console frame themselves. They got it about halfway onto the truck when one guy's knee gave way and they dropped it. Thus, I had made shipping blunder No. 2: I was in Vermont, when I should have been in Alabama

supervising the packing.

The third tale has interesting political implications. This involved our second 24-track machine, shipped to us about a year ago from the distributor. Having gone through the first two adventures, I thought I had seen it all. It was a brand-new tape machine, worth about \$60,000. We had it shipped to us via an air freight carrier the distributor has used many times. As it happened, the machine arrived at our airport the night before a visit by Air Force Two, carrying Vice President George Bush. When a local trucking company went to the airport the next morning to pick up the 600-pound crate, they found our \$60,000 machine sitting naked in the middle of a dusty shipping warehouse, packing material strewn about or missing. We were advised by an unnamed warehouse employee that it was not uncommon to have things "inspected" by secret service men if the vice president's plane is going to be parked nearby. The next day, I personally supervised the loading of the machine onto the truck. Every step of the process was documented and photographed.

Insurance, you say? Naturally, you'd have to be crazy not to insure shipments such as these. But the reality of large hardware purchases is that you already own the gear before it's shipped to you. Damage would have to be pretty severe for you to get a complete replacement. And what about something like a classic Neve console that isn't even made anymore? Lucky for us, the damage to our console was repairable. But it is certainly disheartening to find a dent in your brand-new tape machine.

I have a theory that as the weight and size of a shipment increases, so does the likelihood of shipping disaster, particularly when you get over the 300-pound region.



With used gear, you can improve the odds by being on hand during the packing and unpacking. When purchasing a used console, I recommend supplying your own bubble-pack (for modules), and don't let the movers do the console disassembly. Have them drop a few dozen moving boxes by in advance, and do it yourself. Even simple things like wedging padding between the equipment and the inside wall of the truck might require your prodding. Be sure to photograph *everything*, especially the equipment as it sits in the truck after loading, and before unloading. If the gear is being shipped a good distance, the shipment may get transferred to another truck while en route. This does open the window of opportunity again, thus the importance of looking carefully inside the truck when it arrives at your destination.

I imagine most multistudio facilities have had at least one similar experience with shipping. You can take Murphy's Law and square it when it comes to shipping large studio hardware.

REP

During a recent job, have you encountered a problem or unusual request 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 forum for sharing your solutions to difficult production situations other engineers may encounter. In a continuing effort to educate, we believe that 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.

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 to Michael Fay, Editor, RE/P, 8885 Rio San Diego Drive, Suite 107, San Diego, CA 92108.

Todd Lockwood is owner/chief engineer of White Crow Audio, Burlington, VT.

STUDIO UPDATE

Studio News

Northeast

Howard Schwartz Recording (New York) has purchased two Sony PCM 3348 48-track digital machines, the first studio in the world to do so. To be introduced at the AES Convention in Los Angeles, the machines are scheduled to be installed in mid-November. *420 Lexington Ave., Suite 1934, New York, NY 10017; 212-687-4180.*

Skyline Studios (New York) has opened Studio M6, a 24-track MIDI production facility and the facility's third 24-track room. Equipment includes an Amek 2500 Series console, Macintosh and Atari computers, Akai MPC-60, Emulator, Casio and Akai samplers, and Roland, Yamaha and Oberheim synthesizers. Skyline has also added two Mitsubishi X-800 multitracks.

Dreamland Recording Studio (Bearsville, NY) has added a Studer 820 24-track machine and has rewired Studio A with Monster Cable. *Box 383, Bearsville, NY 12409; 914-338-7151.*

Century III (Boston) is constructing a 20,000 square foot facility at the Universal Studios Florida complex in Orlando, FL, and will provide on-site post-production and teleport services. The facility will service episodic television, commercial and corporate projects generated by Universal, as well as projects for national and regional ad agencies, TV stations and corporations. The facility was scheduled to open in mid-October. *651 Beacon St., Kenmore Square, Boston, MA 02215; 617-267-6400.*

Flight Three Recordings (Baltimore) has named James W. Riley as director of sales and marketing. *1130 E. Cold Spring Lane, Baltimore, MD 21239; 301-532-7500.*

Platinum Island (New York) has added an Otari MTR-90II recorder and a Sony ETC-1000 DAT recorder. *676 Broadway, New York, NY 10012; 212-473-9497.*

Sear Sound (New York) has moved to new and larger headquarters at the former site of the Hit Factory. Studio A has a Neve 8038 32x16x24 console, with Studer and Ampex tape machines. Monitors are UREI and JBL, powered by Macintosh and Fut-

terman power amps. Studio B, dubbed Remote Recording, is a 16-track voice-over facility. Studio C is a 35mm 10-track film/mixing room. *353 W. 48th St., 6th Floor, New York, NY 10036; 212-582-5380 (Studio B, 212-581-2730).*

Sanctuary Studios (New York) has named Howard Kessler as general manager of the 6-room facility. He is the former general manager and chief administrator of Eastern Artists Recording Studios, East Orange, NJ. *632 Broadway, New York, NY 10012; 212-353-2000.*

Shelton Leigh Palmer and Company (New York) has named Bruce Breslau as vice president in charge of the broadcast division. *19 W. 36th St., New York, NY 10018; 212-714-1710.*

Kajem/Victory Studios (Gladwyne, PA) has opened a MIDI suite at the Victory West facility. John Averese handles in-house programming, and equipment includes an Apple Macintosh with Performer software, Yamaha DX-7 and RX-5, Roland D50, S550 and Juno 60, Oberheim Matrix 6, Ensoniq Mirage and ESQ-1, Linn LM-2, Roland SBX-80, Human Clock, MiniMoog, Prophet 6000 and a Kurzweil 250X RMX. The system is portable and can also be used at the Victory East facility in Society Hill. *1400 Mill Creek Road, Gladwyne, PA 19035; 215-642-2346.*

Wendell Recording Studio (Wendell, MA) has added several pieces of new equipment, including an Otari MTR-90 Series II recorder, a Trident 24 console, with MegaMix automation, a 3/4-inch video editing suite and Adams Smith synchronization. The studio also began classes in audio engineering in June, and appointed Bruce Kahn as facility director. *Locks Hill Road, Wendell, MA 01379; 617-544-6466.*

Sound on Sound Recording (New York) has added various equipment, including the 12-second update for its Lexicon 480L, two Tubetech ME-1A equalizers, a Teletronix LA-2A compressor/limiter, and Neumann U67s and KM-84s. *322 W. 45th St., New York, NY 10036; 212-757-5300.*

Presence Studios (East Haven, CT) has added two Lydkraft PE 1B tube EQs, a TC Electronic 2290 delay with 11-second sampling, two AKG C414BULS mics and 20 Sony TC-150 cassette decks for its du-

plication room. New to its keyboard room are a Roland D-550 and MKS-70, and a Yamaha CD-80M. *461R Main St., East Haven, CT 06512; 203-467-9038.*

Sleepy Hollow Sound (Dobbs Ferry, NY) has completed a major upgrade. The studio has been rewired and an MCI JH114 24-track recorder has been installed. Other new equipment includes a Roland D-50 synthesizer and TR 626 drum machine, Nakamichi cassette decks, Yamaha SPX-90II, ART Proverb, Valley People Gatex and a dbx 166 compressor/expander. *39 Cedar St., Dobbs Ferry, NY 10522; 914-693-8537.*

Westrax Recording Studio Ltd. (New York) has completed construction on a custom MIDI-based composition studio, featuring a Voyetra Sequencer Plus, Roland S-50 and D-50, an Emulator SP-12 and a Yamaha TX-816. Also new is a Sony PCM 501 and a Beta recorder. *484 W. 43rd St., New York, NY 10036; 212-947-0533.*

White Crow Audio (Burlington, VT) has completed installation of a GML Servo Automation System on its Neve 8068 console. *19 Marble Ave., Burlington, VT 05401; 802-658-6475.*

Creative Audio Recording Services (New York) has appointed Robin Thomas as head of engineering and chief maintenance engineer.

Sanctuary Recording Studios (New York) is a new six-room complex with two 24-track rooms, two editing suites, a Synclavier room and a MIDI keyboard room. Each has an independent lounge, and all rooms are linked together via a central patching station. Studio A, with a 650-square-foot control room, features an API 40x24 console, Studer tape machines and an extensive variety of outboard gear. The studio also has an attached office/bedroom suite with a private bath. Tom Silverman, chairman of Tommy Boy Records and executive director of the New Music Seminar, is the owner. *632 Broadway, New York, NY 10012; 212-353-2000.*

Paul Lehrman Productions (Arlington, MA) has relocated. The MIDI facility, which specializes in original music for visual applications, has moved to 184 Palmer St., Arlington, MA 02174; 617-643-2700.

STUDIO UPDATE

Southeast

Criteria Recording Studios (Miami) has made a number of equipment purchases, including two Drawmer DS201 gates, Adams-Smith Zeta 3 synchronizer, Eventide HD3000 UltraHarmonizer, its fourth pair of Yamaha NS10Ms, a pair of Yamaha SN10M Studios and a pair of self-powered triamp Genelec S-30s. New Sony digital recording equipment includes a 3202 with center-track time code, PCM 2500 DAT recorder, 1630 digital processor, DMR 4000 U-matic recorder and DTA 2000. The facility has also taken delivery of an AMS AudioFile. 1755 NE 149th St., Miami, FL 33181; 305-947-5611.

Strawberry Skys Recording Studio (West Columbia, SC) has appointed Mike Smith as operations manager. Recent equipment purchases include an additional Atari 1040ST computer with Creator/Notator software, a Korg M-1 digital music workstation, Sonus SMPTE interface, three SCS MOSFET power amps, JBL 4406 monitors and an Electro-Voice RE-20 mic. 1706 Platt Springs Road, West Columbia, SC 29169; 803-794-9300.

The Post Group (Orlando, FL) has appointed Gary Migdal as general manager of its Orlando Facility, which is located at Disney-MGM studios. Box 10000, Hollywood Blvd., Lake Buena Vista, FL 32830; 407-934-5781.

Mr. O Audio (Columbus, GA) has added a Tascam MS-16 16-track recorder, a Roland Octapad, an Alesis HR-16 drum machine, Simmons SDS-1000 electronic drums and dbx 1531-P EQs. David Norman has also joined the staff as an engineer/producer. 2035 S. Lumpkin Road, Columbus, GA; 404-687-6221.

Miami Sound Studio (Miami) has installed a full-service MIDI room, including an E-mu Systems Emax, Roland S-50, Yamaha DX-7 and an Apple Macintosh Plus. The room is directly connected to the studio's 24-track system. 697 N.W. Seventh Ave., Miami, FL 33127; 305-635-4890.

Doppler Studios (Atlanta) has promoted Bill Quinn to studio manager, and Joe Neil to chief engineer. Kristi Harding has joined the staff as administrative assistant. 1922 Piedmont Circle, Atlanta, GA 30324; 404-873-6941.

Cedar Crest Studio (Mountain Home, AR) has updated its audio and video capabilities. Audio additions include a 16-track tape recorder, 20-input console, two drum machines, a digital sequencer and four digital reverbs. Video adds include 3/4-inch editing with A/B roll capabilities, digital frame sync still-store, a character generator and duplication capabilities for 8mm, 1/2-inch, Beta, VHS, 3/4-inch and 1-inch formats. Box 28, Mountain Home, AR 72653.

Midwest

Samsonics (Evanston, IL) has relocated to 1007 Sherman Ave., Evanston, IL 60202. The phone number remains 312-328-4810.

Second Floor Recording (Albany, KY) has taken delivery of a Valley International Gatex noise gate. Route 2, Albany, KY 42602.

Creative Professionals (Milwaukee) has added Raymond Fister to its engineering staff. 1245 N. Water St., Milwaukee, WI 53202; 414-291-9666.

The Brookwood Studio (Ann Arbor, MI) has added a Macintosh Plus computer with Professional Composer and Performer software for MIDI sequencing and synchronization to multitrack tape. 839 Brookwood Place, Ann Arbor, MI 48104; 313-994-4992.

Flyte Tyme Studios (Minneapolis) has ordered a Harrison Series Ten and a second MR-4 as part of its expansion.

Future Media Corporation (Okemos, MI) has recently purchased a 16-channel Soundcraft 200B console.

Studio A (Dearborn Heights, MI) has purchased several pieces of equipment. New to its Synclavier/MIDI 24-track room is a JVC 8250 3/4-inch video deck with remote, a Mitsubishi 19-inch color monitor, Roland Octapad 8, Valley 430 Dynamite and a Yamaha TX-802. New to the main mixing room are two Yamaha SPX-90s, a Barcus Berry 822 and Tannoy NMF8 monitors. The studio has also purchased a Sony 2500 R-DAT recorder. 5619 N. Beech Daly, Dearborn Heights, MI 48127; 313-561-7489.

Hedquist Productions (Fairfield, IA) received a Grand Award at the recent International Radio Festival, for its audio work on a commercial for the Stamford (CT) Downtown Shuttle. The commercial also received a gold medal for best use of sound. Hedquist also received three finalist awards in the entertainment and humor categories. 1007 E. Madison Ave.,

Ron Rose Productions (Royal Oak, MI) has opened up its fifth facility at Farmington Hills, MI. The studio features a full 6-channel Dolby Surround system, a 36-input Neotek Elite mixing console with Audio Kinetics disk-based mixing system, a Sony BVH 3100 1-inch video deck and an Otari MTR 1/4-inch deck. The studio is constructed on four levels, with the booth being at the lowest level and subsequent levels for the engineer, producer and observers. 25885 York Road, Royal Oak, MI 48070; 313-545-1696.

Southern California

Video-It (Culver City) has become one of the first West Coast post houses to install the Sony D-2 composite digital videocassette recorder. 5000 Overland Ave., Suite 6, Culver City, CA 90232; 213-280-0505.

Record Plant (Los Angeles) has named Bill Dooley as director of audio engineering. He will supervise all aspects of engineering, including refining the control room and studio acoustics at the new Sycamore Street facility and the scoring rooms on the Paramount lot. 1032 N. Sycamore, Los Angeles, CA 90038; 213-653-0240.

Summa Music Group Studios (West Hollywood), up and running since May, is planning to add more channels to its SSL console by the end of the year. The SL-4000 G Series has Studio Computer and Total Recall automation. Tape machines include Studer A800 and A820 analogs, and a Mitsubishi X-850 digital machine. Acoustic design was by Lakeside Associates; monitors were custom-made by George Augspurger and were tuned by Steve "Coco" Brandon. 8507 Sunset Blvd., West Hollywood, CA 90069; 213-854-6300.

Music Grinder (Los Angeles) has added a Neve 48-channel console with Massenburg Moving Fader Automation. Also new

are Studer 800 MkIII and MCI multitracks. 7460 Metrose Ave., Los Angeles, CA 90046; 213-655-2996.

Studio on Wheels (Glendale) has added a second remote truck, which features a 16'x8' control room with variable lighting. Equipment includes an Amek console, Ampex MM1200 machines, Altec crossovers and Yamaha monitors. 339 W. Windsor Road, #6, Glendale, CA 91204; 818-243-6165.

Sound Advice (Hollywood) has relocated to a new 1700-foot facility, featuring a multitrack room with MIDI capability. Studio control room designs were by F. Alton Everest, with construction management handled by Jan Orgen of JT Construction. Hollywood & Vine Plaza, 1645 N. Vine St., Suite 360, Hollywood, CA 90028; 213-462-7505.

Le Mobile (North Hollywood) has added an Eventide H3000 Ultra-Harmonizer, two TC Electronic 2290 samplers and a Yamaha REV-1 reverb unit. 11131 Wedington, North Hollywood, CA 91601.

Northwest

Spectrum Studios (Portland, OR) recently shared honors with Oregon Public Broadcasting at the National Public Broadcasting Promotion Awards. Two of producer Liz Aktinson's spots claimed finalist status, and one took the grand prize for radio spot production. The studio also shared Silver Microphone Awards with Providence Milwaukee Hospital and ad agency Gard & Leash. Two radio spots won national finalist honors. 1634 S.W. Alder St., Portland, OR 97205; 503-248-0248.

Southwest

Soundworks Recording Studios (Las Vegas, NV) has renovated its 24-track studio and has installed a Neotek Elan 26-input console. 2570 E. Tropicana Ave., Suite 18, Las Vegas, NV 89121; 702-451-6767.

Nevada Recording Studio (Las Vegas, NV) has opened for business. Featuring a Chips Davis design, the studio offers an MCI 56-input automated console, Ciper Digital audio editing for video and a MIDI preproduction room. Additional equip-

ment includes 3M and Ampex machines, Sony and JVC video gear, and JBL and UREI monitors. Market direction is post-production for video and film scoring, commercial production and record projects. 4665 S. Procyon, Suite M, Las Vegas, NV 89103; 702-798-3032.

Canada

Ocean Studios (Vancouver) has added an audio/video interlock facility. 1758 W. Second Ave., Vancouver, British Columbia V6J 1H6 Canada; 604-733-3146.

Philippines

Track Studios (Pasig) has installed a 24-input Soundcraft TS12 console, interfaced with a Soundcraft SCM762III remote. The previous console, a Soundcraft 1600, has been moved to the facility's MIDI room. No. 9 cor. Shaw Blvd. & Pioneer St., San Buena Bldg., Pasig, Metro Manila, Philippines.

England

Eel Pie Studios (Twickenham), owned by Pete Townshend, has ordered a Focusrite 54/64-channel console with Massenberg automation, to be installed in a control room suite of five adjacent interconnected rooms. The facility's Soundtracs Eric console, installed in late 1987, remains in use with its Synclavier. The Boathouse, Ranelagh Drive, Twickenham, TW1 1QZ, England; 01-891-1266 789.

Manufacturer announcements

Soundtracs has delivered an ERIC console to the Societe Francais de Production, Paris.

Soundcraft has installed consoles at Audio Sensations, Arcadia, CA, 24-input 200B; ByVideo, Sunnyvale, CA, 200B; Tapestry Productions, New York, 16-channel 200B; AC Post Audio Group, Agawam, MA, TS12 audio mation system to an existing TS12 console; and Nickel Recording, Hartford, CT, 6000.

Solid State Logic has announced console contracts with the Post Group, Disney/MGM, Orlando, FL, 56124 M Series film post-production system; Mosfilm, Moscow,

two SL 5676 Series film post-production systems; and Paris Studio, Billancourt, France, SL 5684 M Series film post-production system.

Tascam has sold an M600 console to Acme Sound Works, Hollywood.

Audio Kinetics has installed the first Reflex automation system at Soundhouse Studios, England.

Total Audio Concepts has sold SR9000 live sound consoles to SSE Hire, and Town and Country Club, England; EML Hire, Belgium; Hibino Hire Co., Japan; Music Sound, Germany; and TBS Tokyo and TVB Hong Kong.

Sony has sold a PCM-3324 recorder to Soundmirror, Boston.

Trident Audio has delivered the third DI-AN console to the Music Room, London.

Apogee Electronics has delivered filters to Conway Recording Studios, Hollywood, to be installed in its three Mitsubishi multitrack machines.

Neve has delivered V Series consoles to Marathon Recording Studios, New York (plus an additional 8232); Baby 'O, Los Angeles (a second V series); Compass Point, Nassau, Bahamas; Memphis Sound Productions, Memphis, TN; and Devonshire Audio/Video, North Hollywood.

FM Acoustics has delivered four FM 1000 power amps to PWL London.

Audio Kinetics has supplied five Pacer Chase Synchronizers and a Stripper Time Code Generator to Molinaire Audio and Video Communication. The company has also delivered equipment to the following facilities: Telco, 10 Q.Locks; Midwest Communications, one Q.Lock; Seh Woong Industrial Co., Q.Lock; Westlake Audio, Gearbox time code converter; Studer International, 32-channel Mastermix for a 903 console, to be delivered to the Finnish Broadcasting Co.; and Magnatech, 54-channel MasterMix retrofit for Neotek Elite console, to be delivered to TV New Zealand.

Amek has delivered G2520 consoles to Slippery Studios, Los Angeles; Total Access Recording Studios, Redondo Beach, CA; and the Moody Broadcasting Network,

STUDIO UPDATE

Chicago. BCII systems have been delivered to Record Technologies, Camarillo, CA; the Central Intelligence Agency; Sears and Roebuck; and the Financial News Network.

New England Digital has sold two Direct-to-Disk systems to Tape Gallery, London.

Solid State Logic has announced the following console contracts: Wisseloord Studios, Hilversum, Netherlands, SL 4072 G Series; Studio Marcadet, Paris, SL 4056 G Series; Blue Canyon, Los Angeles, SL 4056 G Series; Yamaha Studios, Tokyo, SL 4056 G Series; Le Studio, Montreal, SL 4056 G Series; Master Mix Studio, Rio de Janeiro, Brazil, SL 4056 Series; Video Sun-mall, Tokyo, SL 4048 G Series; Mosfilm, Moscow, two SL 4040 G Series; Nanjing, China, SL 4040 E Series; Bel Studios, Norway, SL 4040 E Series; TVI, France, SL 4040 E Series; River North Studio, Chicago, SL 4040 E Series; and Tokyo Sound Production, Japan, SL 4032 G Series.

TimeLine has delivered Lynx Post Production Systems to three Japanese facilities: LaserDisc Co., Azabu Plaza and Toei Kagaku Akasaka Video Center. Lynx Keyboard Control Units have been delivered to Intersound, Los Angeles, and AME, Burbank, CA.

Sony has delivered an MXP-3000 console to Magno Sound and Video, New York.

Soundcraft has delivered consoles to Logos Productions, Clayton, OH, TS12; Soundscape Recording Studio, Poughkeepsie, NY, 6000; Crescent Sound and Light, New Orleans, 200B; Audio Sensations, Arcadia, CA, 200B; and Journey guitarist Neal Schon, TS12.

Alpha Audio has delivered a BOSS automated audio editor to Digital Magnetics, Hollywood.

Master Blaster has signed a contract to build and outfit a second mobile studio for Eurosound, the Dutch-based company.

Sondor has delivered OMA S recorders and peripheral equipment to the Mosfilm Centre in Moscow.

Audiotechniques, the Sony dealer in New York, has delivered six MXP-3020 consoles to the Bertelsmann Music Group

Studios in New York. MXP 3000s have also been delivered to Howard Schwartz Recording, New York, Night Wing Recording, Shreveport, LA, Secret Sound Recording, New York, and KMA Tunes, Mamaroneck, NY.

Soundracs has announced the following orders for the Eric console: SFP Television, France; Zeus Studio, Tokyo; and Izanelli Studios, Paris.

TimeLine has delivered the first Lynx Keyboard Control Units to Intersound, Los Angeles, and AME, Burbank, CA. Sound One, New York, has added six Lynx SAL Time Code Modules and two Film Modules. Digital Magnetics, Santa Monica, CA, has added three Lync VSI Time Code Modules and one Film Module.

New England Digital has sold a Synclavier and an 8-track Direct-to-Disk system to Ambassador Television, the production facility associated with the Worldwide Church of God in Pasadena, CA, for audio post-production.

Audio Kinetics has announced the following contracts: Canadian Broadcasting Company's Montreal facility, two 16-channel MasterMix systems, to be fitted into Neve 5104 consoles; Universal Studios, 64 channels of AK2 faders on its Mitsubishi Westar console; Soundcraft, three MasterMix systems to be fitted on TS24 consoles; China Motion Picture Studios, Q.Lock 4.10 three-machine Alpha system; and Angell Recording Studios, Q.Lock 4.10-2A.

Editron USA has announced the following installations: a 520 Audio Editing System at Dolby Laboratories, San Francisco, and three 620 telecine/video editing systems and one 520 audio editing system at Creat National Videotape, Hollywood.

Aphex Systems Ltd. has sold four Dominators to Lucasfilm Ltd.

Lexicon has delivered an Opus system to Blue Light/Illusions in Sound, Burbank, CA, the first Opus installation on the West Coast.

Soundcraft has sold an 8000 VCA subgroup console to Sound Southwest.

DDA has sold AMR 24 consoles to PWL Studios, England, and the band Simple

Minds; DCM-232s to Rainbow Studios in Norway, Random Access Studios in London, Studer Revox of Japan and Park Lane Studio in Scotland; and D series consoles to Tel Aviv Museum in Israel, Triple Onda in Spain, and B+H Sound Services and TVS, both in the United Kingdom.

Sony has delivered a PCM-3324 digital multitrack recorder to New Age Sight & Sound, Atlanta.

New England Digital has sold Synclaviers to Clement Associates, Greensboro, NC, and Martin Recording, El Paso, TX.

Soundcraft has installed consoles at The Sound Department, Portland, OR, TS-12; Jefferson Pilot Productions, Charlotte, NC, 200B; John Friedmann Productions, Chicago, 200B; and Cloud Nine Recording, Mt. Dora, FL, Series 600.

API Audio Products has installed its new 4032 console, with 48 inputs, 32-channel monitoring and Audio Kinetics MasterMix automation, at Bias Recording Studios, Richmond, VA.

Harrison Systems has sold a Series Ten console to the Ministry of Culture of France.

Sony has installed an MXP-3000 console in RCA Studios' Studio C in New York.

REP

THE CUTTING EDGE

By Laurel Cash

Neve signs new automation agreement

Neve has signed an agreement with Martinsound Technologies to be the exclusive worldwide distributor for the Flying Faders Automation System. Neve will also be responsible for marketing and technical support.

Martinsound Technologies research and development team, headed by Joe Martinson, includes Dale Manquen and B. Morgan Martin. Together they developed the Flying Faders Automation design, working in close cooperation with Neve.

The Flying Faders system is designed to work with all new Neve consoles and will also be available to retrofit existing Neve and non-Neve consoles. The company will also continue to sell the NECAM 96 system.

The Flying Faders system is reported to have a number of advantages over the existing top-of-the-line moving fader systems. "The architecture is designed to be expandable to 256 faders, but you don't have to invest in a huge computer if you're planning to use 96 faders or fewer. You can start with a smaller unit and, as your needs expand, so can the computer," says Martin. The computer is an AT class for the displays and uses a 68000 series microprocessor for the automation program and a mouse or keyboard as a control device.

The color display uses "windows," or pull-down screens, and can be displayed on any size monitor, including projection. How can this make your life easier? For starters, it saves all your passes in RAM. If you do 15 passes and the producer suddenly wants to go back to pass 7 or 8, you can go back instantly because it is still in RAM. An on-screen RAM indicator shows how much RAM is available. You can save to disk any or all of the passes you did, any time you want.

In addition, on each pass, only the updates are copied to RAM, instead of the entire mix with the changes. This allows for storage of more passes in the two megabyte RAM. Other items allow on-screen label lists and machine control interfacing



with on-screen status.

The faders are Penny & Giles 3000 series that Martinsound motorized themselves. An LED on each fader indicates when that fader is recording a new change. If you have made a new move and want to match that fader to the original play-pass position, there are two LEDs, with arrows, which light to show you the way. Also, each fader has a Match button (flush-mounted at the bottom) that will bring it back automatically to the play-pass position.

In the Auto Match mode, the fader automatically nulls (match) to the original play-pass position. The speed at which the faders reach the null can be changed by choosing the glide rate (this is also applicable when doing a match), so you can do either a slow fade or a lightning-fast one.

Fader modes can be set either locally (on each individual fader) or globally using the Global Master. Plus, there are no restrictive "change mode now" messages, so you can set any fader or mute to any mode at any time. You can even lock selected faders or mutes into a "safe" mode.

It is claimed that the Flying Faders Links allows a fader to be moved independently of, yet remain locked to, the link. In the Link mode, whichever fader you touch first becomes the master of that link. If you decide you want to move fader No. 2 independently of the link, you must first touch and hold another fader (this tells the

computer which fader is the master), then move fader 2 as desired. If you move fader 2 after designating fader No. 1 as master (in a four-fader link), fader 2 will move independently of the link while the master fader (No. 1) still controls faders 3 and 4. When you let go of fader 2, it returns to the link, but starts from wherever you have adjusted the fader. Also, there is a Link Match feature, and you can include your mutes in the link.

Flying Faders also uses traditional groups. In this mode, any fader can be a group or subgroup master. As a result, you have any number of group

masters available, limited only by your total number of groups and faders. This means you don't have to spend the extra dollars on separate group masters. Any group can have any number of subgroups, and any fader can also act as the subgroup master. The mutes are included with the group as they are with the links and can be activated by the master. If you set the mutes first, you can then set them to safe, before proceeding with the rest of the mix.

When you want to store that "perfect mix," you can either save it to the built-in hard disk or transfer to a standard 3.5-inch diskette. Group and link assignments are also stored, and when you load the mix later, the assignments are automatically remade.

Many automation systems have a tendency to drift, causing misalignment. Flying Faders has introduced an automated digital calibrator that can align the entire servo system in less than 10 seconds. The result: "Your moves will always come back the same, never a little off," claims Martin.

Essentially, if the new Flying Faders Automation does all it's reported to do, we may have an important new step forward in the evolution of console automation.

REP

Laurel Cash is RE/P's executive consultant and a free-lance writer based in Los Angeles.

NEW PRODUCTS

Yamaha C1 computer

The MS-DOS-compatible computer is the first fully dedicated professional music computer, according to Yamaha, and offers several specific advantages for music and recording. Connections include MIDI (two In, one Thru and eight Out), SMPTE and RS-232. Other features include ROM-based music fonts, high-resolution display and 1Mb of internal memory. Two versions are available, one with an on-board 20Mb hard disk and one with two 3.5-inch 720k floppy drives. Both include an external CRT jack, an ac connection for an external monitor and a parallel printer port. The computer comes with MS-DOS 3.3 and a MIDI monitor/bulk data manager program.

Circle (100) on Rapid Facts Card



Tascam DA-50 DAT recorder

The recorder contains four ZD (zero distortion) circuits, which add and subtract digital dither in the conversion process to drastically reduce granulation noise, which occurs as sound levels decrease. To prevent electrical interference, external vibrations and resonance from reaching each circuit, the five main sections are shielded in individual, internal boxes. Other features include four direct-drive brushless motors for the drum, capstan, supply and takeup reels, 38-key full-function hardwired remote and a 25-segment/3-color level meter. Suggested retail is \$4,000.

Circle (140) on Rapid Facts Card



NED optical disk effects library

Developed in conjunction with Sound Ideas, New England Digital's sound effects library contains 1,462 captioned and cross-referenced sound effects, totaling two gigabytes of sound data. All source material was recorded digitally and then sampled into the Synclavier at 50kHz or more. Most effects are in true stereo. The library comes with a manual that cross-references effects by title, file name, category, caption and volume, and lists lengths and sizes in seconds, sectors and megabytes.

Circle (101) on Rapid Facts Card

Studiomaster Mixdown series consoles

The Mixdown series is available in 16x4x8x2 and 16x8x16x2 versions. Both are designed with the MIDI studio in mind, allowing users to record MIDI sequencer-controlled instruments direct to 2-track. Features include six aux sends per channel, 3-band EQ with sweep mid and bass, channel mute, 48V phantom power per channel, direct outputs on channels and EQ defeat per channel. The consoles can also be expanded via 4-channel input expanders.

Circle (102) on Rapid Facts Card

Milab D-37 mic

The dynamic cardioid mic is constructed of solid brass. Features include a heavily shock-mounted moving coil element for minimum handling noise, built-in pop protection and a frequency response of 50Hz to 20kHz with a favorable boost in the vocal/presence range.

Circle (103) on Rapid Facts Card

Soundcraft VSA24 serial interface

The VSA24 allows the Soundcraft 200B/VE console to interface to video editors using common serial interfaces, and allows the console to eavesdrop on the existing editor lines. In the standard auto mode, the unit listens for editor commands and follows them using the editor's start point and fade rate information, allowing the audio from the two video machines to crossfade at the same time as the video. Each unit will respond to crossfade information from up to eight video source machines, with additional units allowing a larger mixer to track up to 24 machines. Audio effects can also be controlled in-

dependently from the video editor. Suggested list price is \$1,950.

Circle (104) on Rapid Facts Card

Weetech low-voltage interconnection tester

The model 20 tests cables, cable harnesses, wiring systems and bare PCBs. The unit tests for opens and shorts for up to 128 test points and will store up to 16 test points on each EPROM, using a built-in EPROM programmer. An assembly mode guides an assembler in making correct point-to-point connections during an assembly process.

Circle (105) on Rapid Facts Card

Audio-Technica AT8506 4-channel phantom power

The unit produces 48Vdc to microphones requiring phantom power, and operates from 100Vac to 120Vac, 50Hz-60Hz. Each channel can provide up to 14mA, and a constant voltage source is maintained with no channel interaction. Other features include internally protected regulator IC to prevent overheating or damage, and locking XLR-type connectors with silver-plated beryllium copper contacts.

Circle (107) on Rapid Facts Card



Opcode Systems HyperCard stack

Opcode Demo 2.0 is a collection of HyperCard documents that allow users to learn about Opcode hardware and software. Unattended, the stack will run continuously as a slideshow. When the slideshow stops, users can find out more about the product shown on the screen or can look at related products. The demo is free to dealers or interested MIDI users.

Circle (113) on Rapid Facts Card

NEW PRODUCTS

CEPCO equipment security system

The Thiefbug is installed in the electrical outlet box and detects when a computer or other electrical equipment is disconnected, either by being unplugged or when the power cord is cut. Using the existing ac power wiring, it transmits a coded alarm signal that gives its location throughout the area. Monitoring units indicate where the alarm is sounding, and plug directly into the 115Vac power outlet. Various versions are available, depending on the application.

Circle (106) on Rapid Facts Card

Astatic JT 8 mic

Designed for acoustic instruments, the JT 8 is a cardioid unit that features a smooth frequency response and -55dB output. Impact Noise Reduction is a shock-mounting system that isolates the microphone transducer from the handle, which reduces handling noise. An internal pop filter is included.

Circle (108) on Rapid Facts Card

Jensen output transformer

The JE-11-EMCF is a smaller-sized and lower-cost addition to the M series of output transformers. It handles +18dBm at 20Hz. The 80% nickel bifilar design allows for extremely low distortion and works well with a wide range of source impedances, including consumer audio outputs.

Circle (109) on Rapid Facts Card

Intelligent Music MidiDraw

For use with the Atari ST, MidiDraw combines drawing and music making in one program that allows users to perform on the computer as if it were a musical instrument. Drawing in the Drawing Field controls notes on one axis and dynamics on the other. Additional features include a mainscreen recorder for recording and playing back drawings, a delay control to delay, transpose and reroute drawings as MIDI data and an interpreter function, which automatically creates music by interpreting picture elements. Data can be saved as MIDI Files, allowing them to be used in other applications. System requirements are an Atari 520 ST, 1040ST or Mega ST and one or more MIDI synthesizers. Suggested retail price is \$95.

Circle (111) on Rapid Facts Card

Aphex 612 expander/gate

The 612 uses downward expansion with

a variable ratio of 1:1 to 20:1, allowing for greater dynamics control. Features include variable high- and low-pass filters that may be switched in and out of the circuitry; external key switch; adjustable threshold from -30dBm to +20dBm; and adjustable release time from 40ms to four seconds.

Circle (154) on Rapid Facts Card

Peavey 380N mic

The mic features a diaphragm made of neodymium and titanium, which the company says combines the advantages of neodymium in a more flexible package. As a result, the mic avoids breakup in high SPL situations. List price is \$199.50.

Circle (115) on Rapid Facts Card

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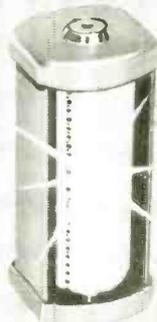
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Circle (35) on Rapid Facts Card

NEW PRODUCTS

Perma Power surge suppressor

The Power Tender line provides 3-way protection against power line surges. They are rated at 1,800W, have a response time of 1ns and has a let-through voltage rating of 400V. Failure indicator lights warn against unprotected operation if the suppressor circuit fails, and a buzzer is available on some models. The group includes four strip-type units; also available is a cordless unit that converts a standard duplex outlet into six outlets.

Circle (112) on Rapid Facts Card

Intersonics Contra-Bass

Contra-Bass utilizes a small cabinet design and has a flat response from 16Hz to 125Hz. It can produce 114dB output at 16Hz with a 250W input. The unit has two active 15-inch drivers run by a servo motor and two passive 18-inch radiators that have less than 1.5-inch excursion capability. Weighing 120 pounds, the unit has a nominal 3-ohm impedance.

Circle (116) on Rapid Facts Card

Lelan Co. Cable Lock

Cable Lock is a cable organizer that includes a nickle ring for easy hanging or storage. One end of the cable is closed between the lock's flap; the cable is then coiled normally. The rest of the flap then wraps around the bundle, with the ring at the top of the bundle. The locks are available in a variety of colors. Retail price is \$1.49 per lock.

Circle (114) on Rapid Facts Card

McKenzie additions to Studio 7 Series

The company has added 15- and 18-inch loudspeakers to the speaker series. Both feature a 4-inch voice coil wound on ventilated high-temperature Kapton to handle and dissipate generated heat. A single-coil suspension system is used to improve stability and control on long excursion. Both have an impedance of 8, 400W power handling and SPL of 99dB.

Circle (120) on Rapid Facts Card

Audioaccess PX-240 CD changer

The programmable changer features a 240-disc capacity and can store up to 10 playlists of 99 songs or discs in each playlist. Maximum access time is 10 seconds. The unit also features 4x oversampling, dual 16-bit D/A converters, audio and digital outputs, RS-232 interface, individual disc load and unload, and a rack-mounting option.

Circle (122) on Rapid Facts Card

Analog Solutions dual-channel analog subsystem

The ZSH202 is an audio dual-channel sample-and-hold amplifier and multiplexer subsystem. When used with a true 16-bit A/D converter, the subsystem provides THD + noise of less than 92dB from 20Hz to 20kHz. Two proprietary sample/hold amplifiers have been combined with a proprietary analog multiplexer and ancillary circuitry needed for simple implementation.

Circle (123) on Rapid Facts Card

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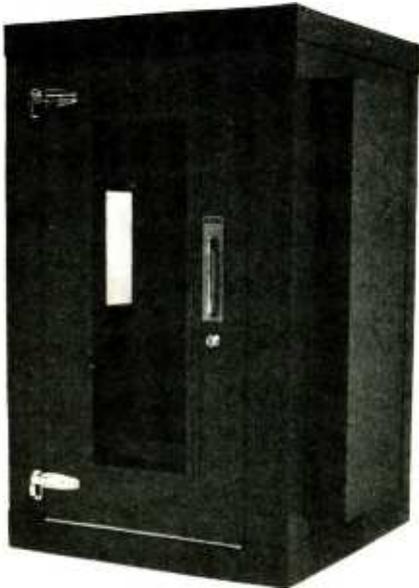


Circle (36) on Rapid Facts Card

Acoustic Systems voice-over booths

The pre-fab booths provide easy-to-order, acoustically engineered booths for voice-over or recording requirements when standard construction is not feasible. They are available in 10 sizes ranging from 3' 8" x 3' 8" to 8' 8" x 8' 8". Standard inside height is seven feet. All models are constructed with 4-inch panels with an STC rating of 45. Options include STC 53 panel construction, wall carpet with stained wood trim, custom colors, additional interior height, canted windows, lighting dimmer controls, track lighting and an "on air" light.

Circle (119) on Rapid Facts Card



Monster Cable M Series

The M100 is a high-performance pre-amp-to-speaker connection that incorporates PowerDrive technology to improve smoothness, dimensionality, clarity and the reproduction of inner detail and harmonic structure of the music. The technology bypasses the amplifier's internal ground connection, which provides the signal with an accurate reference point. By connecting one end of the loudspeaker to the RCA input ground of the amplifier, the resulting loop is tied to a singular reference point, which provides an accurate baseline for the amplification of the input signal. Prices are \$300, 1-meter pair; \$350, 2-meter pair; \$475, 20-foot pair; and \$150 for an upgrade of any M1000 cable.

Circle (117) on Rapid Facts Card

Yamaha PC series power amps

Three models comprise the second generation of the power amp series. The PC2602M and -2602 feature 400W per channel into 4, or 260W per channel into 8 in the stereo configuration, and 800W in BTL-monaural configuration. The PC1602 features 240W per channel into 4, or 160W per channel into 8 in the stereo configuration, and 480 in BTL-monaural configuration. All have several levels of power protection built in, and have a frequency response of 10Hz to 50kHz.

Circle (118) on Rapid Facts Card

Passport Designs NoteWriter

NoteWriter is a Macintosh-based music notation program that features a Quick-Scrawl mode, which allows notes to be sketched onto music staves with the mouse and then be converted into the intended musical symbols. Notation may be entered in any order, or in sequential note entry. Any portion of individual symbols can be adjusted to ensure quality print-outs. Suggested retail price is \$295.

Circle (125) on Rapid Facts Card

Audico tape length verifier

The model 200-9 is an audiocassette timer/rewinder/exerciser that determines the length of tape in a cassette. A sensing device distinguishes between tape and leader, and provides a readout in minutes and seconds of playing time. A C-60 can be timed in less than 20 seconds. The unit has two independently operated stations, both of which can be used as high-speed rewinders. Combined rewind capacity is 400 C-60s an hour. The exerciser tests cassette shells to determine their quality.

Circle (126) on Rapid Facts Card



Akai MX76 master keyboard

The MX76 contains all of the features of the smaller MX73 master keyboard with

several additions. Keyboard length has been increased to 76 keys, and the keys are now weighted. Internal memory contains 64 banks and four groups. External memory storage is available on 16k IC cards. MIDI ports include two independent Outs and one In, which merge incoming data with the data being generated by the unit. Suggested retail price is \$2,699.95.

Circle (129) on Rapid Facts Card



SEA computer utility program

AXE, from System Enhancement Associates, compresses computer files with the .ese or .com extension, saving from 10% to 60% of disk space. Programs operate normally, and compression is achieved using the Dynamic Lempel-Ziv Method, which has no effect on the speed of the program. AXE adds its own code to the compressed program, which decompresses and loads it. System requirements are IBM PCs or compatibles, one floppy drive, 128k of RAM and DOS 2.1 or later. Suggested price is \$50.

Circle (127) on Rapid Facts Card

Winsted rack doors

Made of Plexiglas, the doors are designed for Winstead rack cabinets and allow easy viewing of rack systems. The doors are recessed 1.5 inches to offset knobs and switches, and can be mounted for either right- or left-hand openings. The doors can be locked, and the Plexiglas is smoke-tinted.

Circle (128) on Rapid Facts Card

Audio M MIDI Long-Shot

MIDI Long-Shot is designed to extend the maximum effective distance over which MIDI information can be transmitted. It consists of two identical units, each with a transmitter and a receiver, and uses standard XLR connectors and cables. All MIDI information transfer rates are supported. Crosstalk of MIDI to adjacent channels in snakes or studio runs is two orders of magnitude below that of unconverted MIDI. Price is \$140; the unit is available directly from the manufacturer.

Circle (130) on Rapid Facts Card

NEW PRODUCTS

Hardware, software updates

Trident patchbay console

The Series 24 console now comes in two patchbay versions, both using Mosses & Mitchell all-metal patchbays. A 364-point TT patchbay is available on the 28-input mainframe, with a 468-point TT patchbay available on the 36-input mainframe. The 24 features a 4-band high and low mid-sweepable EQ with a variable highpass filter, balanced mic and line inputs with separate gain controls and phase reverse, and eight aux sends with pre/post switching in pairs. Prices are \$31,900 for the 28-input version, and \$35,900 for the 36-input version.

Circle (133) on Rapid Facts Card

IMS MIDI interface for Dyaxis

Integrated Media Systems' Dyaxis direct-to-disk digital recording system now comes with a MIDI interface, developed in conjunction with Opcode Systems. It combines a time code reader and generator with SMPTE-to-MIDI conversion in a rack-mount package, allowing the system to operate under time code control without third-party hardware. Two MIDI Outs are provided in addition to a switchable MIDI In, which allows MIDI instrument recording when Dyaxis is not in use.

Circle (135) on Rapid Facts Card

Bacchus software for C1 computer

Bacchus Software Systems' TX81Z and TX802/DX7II graphic editing systems are now available for the Yamaha C1 music computer. The systems are used as a voicing tool to create, store and retrieve sounds. Both programs feature mouse-based editing, icons, graphic control panels and multiple overlapping windows. Suggested retail for the TX81Z system is \$199.95; retail for the TX802/DX7II system is \$249.95.

Circle (136) on Rapid Facts Card

Sound Designer software for Atari ST

Digidesign's Sound Designer sample editing software is now available for

the Atari ST computer, providing waveform editing and DSP functions. The program supports more than 20 keyboards, including those from Ensoniq, E-mu, Roland and Akai. System requirements are an Atari 520, 1040 or Mega ST, with a monochrome or color monitor. A hard drive is highly recommended. List price is \$349.

Circle (137) on Rapid Facts Card

Community grille option

Steel grilles can now be installed on five models of Community's CS Series II loudspeakers. On the CS70, -52, -38 and -35, the grilles cover only the LF speakers, leaving the molded horn assemblies exposed. For the CS60 subwoofer, the grille covers the entire cabinet face. The option is designated by an SG suffix added to the regular model number. Grilles must be factory-installed and are not interchangeable with stock grilles because of differences in mounting methods.

Circle (138) on Rapid Facts Card

Additional Sound Blocks for Kurzweil 1000 series

The new line of Sound Blocks increases the number of ROM-based sounds available in the 1000 series, and both old and new sounds are simultaneously accessible. Initially available for the 1000 PX, the blocks will also be available for the SX, HX and GX. Installation can be performed at authorized Kurzweil service centers.

Circle (194) on Rapid Facts Card

Graphic Notes software upgrade

Version 1.1 of the company's Music Publisher program has been released, which allows users to play back the compositions on MIDI devices and also accepts input from MIDI keyboards. Also upgraded are part extraction and transposition functions. For use with Macintosh computers, the upgrade is free to registered users.

Circle (131) on Rapid Facts Card

Organizers software from Integral Systems

Organizers is comprised of two PC and compatible software packages for facilities. Tape Library tracks tapes that need to be referenced. Location is achieved via title, artist, content, dates, keyword index and producer. Order Tracker keeps track of rental time and tape ordering, and is designed to turn billing cycles into hours instead of days. Both programs can be integrated into existing accounting packages, and can run on a stand-alone system or in networks. Pricing is dependent upon features desired.

Circle (132) on Rapid Facts Card

Simpson 560 multimeter

A menu-driven unit, the 560 features fast autoranging, data-logging capability on any selected range with 2,150 measurement memory, 500kHz frequency counter, Centronics printer or RS-232C interface, and continuity and diode check with audible beeper. Functions include dc volts and amps, true rms ac or ac+dc coupled volts, rms ac or ac+dc coupled amps, low- and high-power resistance, and 5Hz to 500kHz frequency.

Circle (110) on Rapid Facts Card

Korg C2M programmable MIDI audio mixer

The C2M uses dbx VCAs, which the company says makes the mixer cleaner than more expansive digital mixers. Each of the unit's eight channels has two effects sends and three bands of EQ with a sweepable midrange. Level, pan, effects and EQ on each channel can be programmed or controlled in real time over MIDI, as can the master volume and effects return levels. A total of 64 programs with programmable crossfades and mutes can be stored.

Circle (121) on Rapid Facts Card

BSS FDS-310 crossover

The unit is a sweepable frequency-dividing system that uses 24dB/octave Linkwitz-Riley filters with four frequency bands. Depending on the frequency configuration, the unit can be used as a two-way stereo or three-way mono crossover. Each frequency band has its own level control, frequency control, polarity reverse switch, mute button, signal present LED and peak signal warning LED.

Circle (124) on Rapid Facts Card

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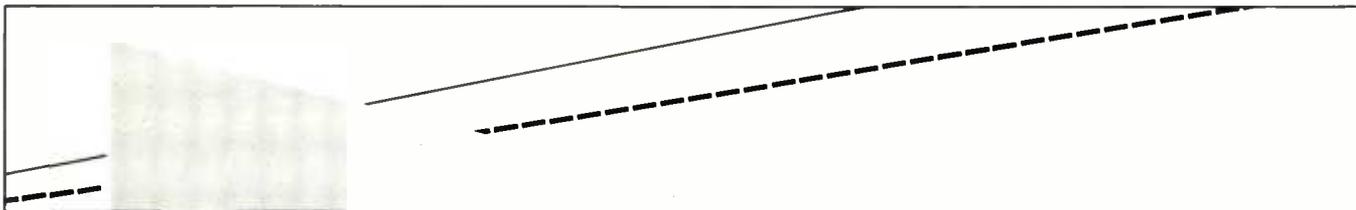
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Passport software for Yamaha C1

Passport Designs has ported two of its music software programs, SCORE and Master Tracks Pro, to the Yamaha C1 music computer. SCORE is a desktop music publishing system that gives control over placement, size, shape, spacing and output resolution of printed music notation. Master Tracks Pro is a sophisticated MIDI recording and editing package. Master Tracks' latest version, 3.0, is available for the C1.

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Akai S1000 sampler

The S1000 16-bit sampler has two sampling rates, 44.1kHz and 22.05kHz, and comes standard with 2Mbytes of memory, yielding 23 seconds of mono sampling at 44.1kHz and 11.5 seconds in stereo. A time-compression feature allows the sample time to be changed without altering pitch. The unit receives on all 16 MIDI channels, and each keygroup in a program can receive on its own MIDI channel. Disks from the S900 are fully compatible.

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Continued from page 12

People

Pro Co Sound has announced two appointments. **Jerry Smelker** has been promoted to vice president of professional products. **Jeff Garstick** has been promoted to vice president of commercial sound products.

Larry Banks has been appointed vice president of marketing for DOD Electronics.

Jeffrey N. White has joined Innovative Electronic Designs as sales applications engineer.

Artec Consultants has named Nicholas Edwards and Robert Essert as associates of the firm.

Ron N. Dow has been named staff director of design engineering at Precision Monolithics.

Mason Jenkins has been named director of sales for Cook Laboratories.

William J. Dorman has been named connector product manager for Neutrik USA.

Publications

ANSI standards catalog

The American National Standards Institute's catalog of American National Standards is a subject index that lists more than 8,000 ANSI-approved standards in such fields as acoustics, construction, electrical and electronics, information technology, measurement and automating control, and image technology. The catalog is free to ANSI members and libraries that serve the public. Non-member price is \$20 plus shipping and handling.

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Nemal selection guide

Nemal Electronics has published a revised edition of its cable and connector selection guide, which covers a wide range of electronic cable, connectors, tooling and cable assemblies.

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A-T product guide

Audio-Technica has released a product guide for audio-for-video production. Included are descriptions of the 40 and 800 series mics, the 4462 field production mixer and accessories such as stereophones, mic mounts and cabling.

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Shure product catalog

Shure Brothers had published a 25-page catalog detailing its line of sound reinforcement, recording and broadcast products. The catalog is free.

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TAB recording handbook

TAB Books' "The Complete Handbook of

Magnetic Recording—3rd Edition," by Finn Jorgensen, is intended as a continuing reference for seasoned professionals and as an introductory work for anyone wanting to learn more about recording. The book covers fundamental and technical magnetization, recording and playback theory, magnetic head and media, drive mechanisms, write/read processes and equalization, detection and coding, and applications. List price is \$44.50.

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Onkyo 18-bit technical paper

Onkyo USA has released a technical white paper on 18-bit technology. The paper explains the difference between pseudo 18-bit conversion systems and linear 18-bit, and is free.

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Full Compass catalog

Full Compass Systems has released an "interim" catalog covering about a third of the products it carries. Because of the world economy, work on the full catalog has been lengthened because of continual price changes. This catalog is a good starting point, according to company, and a toll-free number is available to get additional information on products listed or to inquire if a certain product is available. The full catalog will be published in the near future.

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REP

Corrections

Because of space limitations, credits for the August cover were not able to be listed in the issue. This cover design was conceived and developed with the help of many, and special thanks are in order to the following: Andy Munro of Windmill Munro Design, London, for developing the basic design using Schema CAD software; Hugh Morrow of Paracomp Inc., San Francisco, for his technical assistance in running the Schema software; and Dan Martin and Greg Calvert of Artifax, San Diego, for their assistance in creating the final image.

Hardware and software used included the Macintosh II with 2Mb RAM, 140Mb HD and 256 color palette; and Schema, Color Schema and Adobe Illustrator '88 software.

In the October issue, the photo credit for the article "Producing Music for the Moscow Ballet Theater" was not included. All photos were taken by the author, Ron Streicher.

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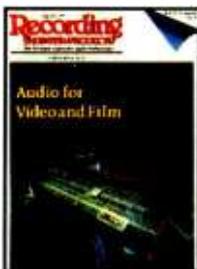
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