SEPTEMBER/OCTOBER 1988
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db

THE SOUND ENGINEERING MAGAZINE

serving: recording, broadcast and sound contracting fields

Guides: Microphones, Studio Accessories

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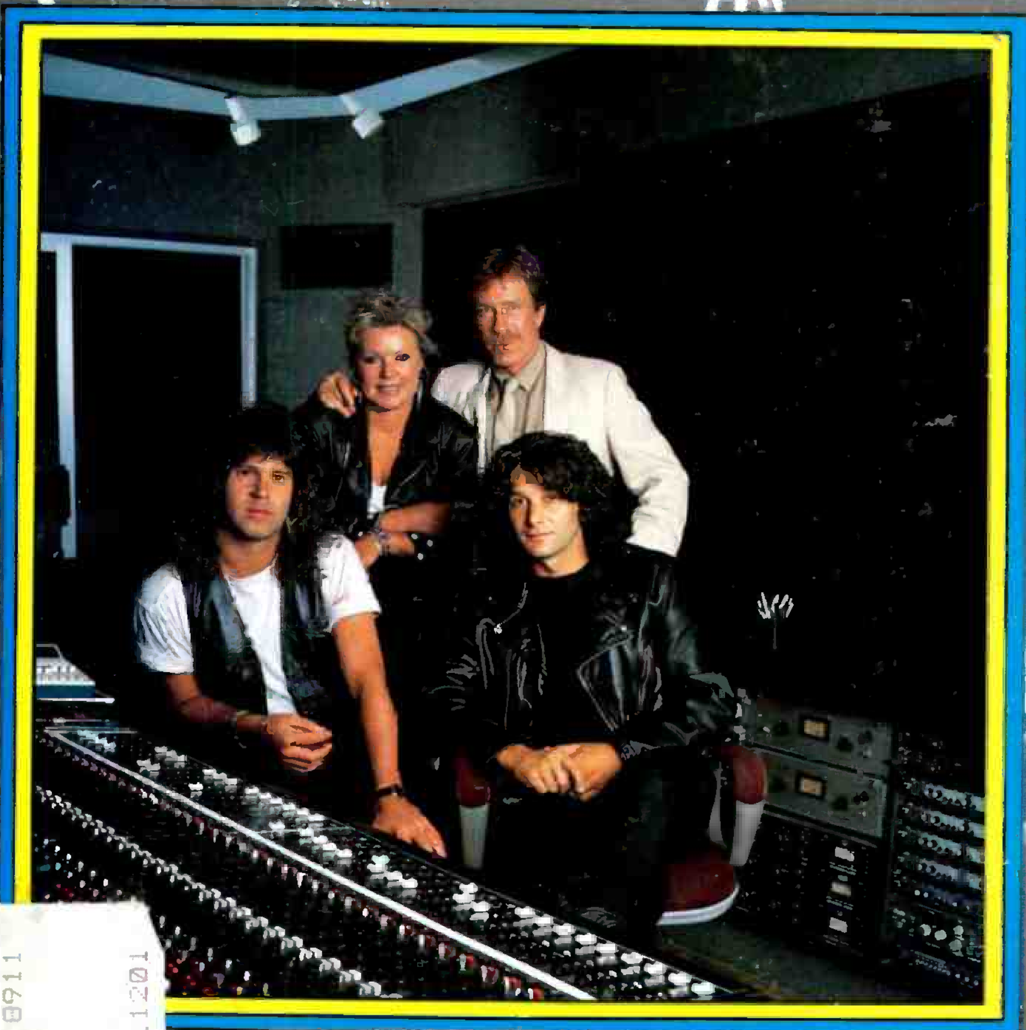
for audio professionals



SEPTEMBER/OCTOBER 1988 VOLUME 22 NO.5

the recording engineer

the broadcast engineer

the sound contracting engineer

THE ELECTRONIC COTTAGE

See page 5

See page 20

About the Cover

• The cover of this issue features a color shot of TMF Communications' studio. Learn all about them and more in *California Creative* beginning on page 5. The black and white film stills in the background of our cover are courtesy of Paramount Pictures Corporation and are from "Sunset Boulevard," and "We're No Angels." "Sunset Boulevard" ™ and © 1950 and 1988, Paramount Pictures Corporation. "We're No Angels" 191 ™ and © 1954 and 1988 Paramount Pictures Corporation. All Rights Reserved.

Editorial

Over the years that **db Magazine** has been published, we have done many stories on visits to studios and manufacturers thus bringing you further insight into what this professional audio industry is really all about. In many ways the future will see even more of these kinds of stories, but with one important difference.

Beginning with the July/August 1988 issue, we have sought to geographically tie the theme of each issue, at least with several of the articles in that issue. Accordingly, we went to Nashville, TN for that issue, and Los Angeles for this issue. In November/December, we will look at New York City. A plan is worked out for all of 1989's issues to report on different areas.

Not all of these themes will be articles about recording studios, the broadcast audio and sound contracting/performing markets will also be included. And of course, while each issue will have a theme, each issue will also have other articles on audio engineering subjects included.

This issue is certainly an example of just that. While the issue starts with the *California Creative* articles which cover two L.A. based post-production houses and a major record producer, it goes on to other articles on recording, sound reinforcement, and broadcast audio.

Nothing around here is ever cast in concrete—possibly excepting our commitment to pro audio. We really do want to hear from you, dear reader, because it is not trite to say that this is your magazine, as much as it is ours.

So, do send those cards and letters.

Please also note our address in this issue; we moved to larger quarters back in May, but a lot of mail is still going to the old address. That slows it down, eventually it will stop it cold. While the postman might like a lighter load, we want every communication to reach us. LZ

db

THE SOUND ENGINEERING MAGAZINE

Editor/Publisher
Larry Zide

Associate Publisher
Elaine Zide

Associate Editor
Carol A. Lamb

Technical Editor
Corey Davidson

Contributing Editors
John Barilla
Bruce Bartlett
Brian Battles
Drew Daniels
Len Feldman
Robyn Gately
Randy Hoffner

Graphics & Layout
Karen Cohn

db, The Sound Engineering Magazine (ISSN 0011-7145) is published Bi-monthly by Sagamore Publishing Company Inc. Entire contents copyright 1988 by Sagamore Publishing Company Inc., 203 Commack Road, Suite 1010, Commack, NY 11725. Telephone: (516)586-6530. db Magazine is published for individuals and firms in professional audio recording, broadcast audio-visual, sound reinforcement-contracting, consultants, video recording, film sound, etc. Application for subscription should be made on the subscription form in the rear of each issue. Subscriptions are $15.00 per year ($28.00 per year outside U.S. Possessions, $16.00 per year in Canada) and payable in U.S funds. Single copies are $2.95 each. Editorial, Publishing, and Sales offices are at 203 Commack Road, Suite 1010, Commack NY 11725. Second Class postage paid at Commack, NY 11725 and an additional mailing office. **Postmaster: Form 3579 should be sent to db Magazine, 203 Commack Road, Suite 1010, Commack, NY 11725.**

Letters

Dear Mr. Daniels,

As your article in a recent issue of **db** quite accurately suggests, 70-volt distribution systems are a good way to traverse long distances with speaker cabling.

Our company has been involved in public address and sound reinforcement for about ten years. We have installed and rented our many 70-volt systems over the years very successfully.

It is our belief that two codicils should be attached to your paper on 70-volt systems:

1. In many areas, 70 -volt wiring used in commercial or industrial installations must be run through conduit, to meet electrical codes.

2. Not all line matching transformers are created equal. That is, transformers which will deliver high fidelity, broadband signals are not cheap. Conversely, many cheap transformers will not deliver high fidelity, broad-band signals.

With all else—we agree heartily.

Rick Shriver
Atmosound

Dear Mr. Shriver,

In fact, 70-volt transformers deserve scrutiny from wary sound system designers. Those innocent looking little devices attached to the speakers usually have poor performance at low frequencies because they have insufficient core material to avoid magnetic saturation at rated power.

Some manufacturers state a frequency range for the power rating of their transformers, and this usually refers to the range of frequencies over which the device can be operated while suffering some minimal amount of saturation-induced distortion, a typical specification might call out less than 1 percent total harmonic distortion.

The impedance of any transformer drops with frequency, strictly due to core size. This is why 400 Hz is used in place of 60 Hz for power aboard aircraft. The small core needed to pass 400 Hz saves weight. You will notice that switching power supplies (e.g. those in computers) often contain walnut-sized power transformers to provide as much as 250 watts, and this is possible because switching power supplies operate their transformers at much higher frequencies, usually above 20 kHz. A typical cheap ceiling speaker transformer may perform fine above say, 50 Hz, but produce obnoxious distortion at 30 Hz. In this case, the transformer is also loading the driving line with a much lower impedance, which can, in turn, cause amplifier problems, and a spiraling degradation of overall system performance.

If a designer desires clean 30 Hz bass from a distributed speaker system, the transformers in the system should be rated for 30 Hz operation, or should be operated conservatively—perhaps at one-quarter power.

In general, the rule of thumb to apply to loudspeaker transformers is that for a doubling of frequency, the voltage that may be applied to the transformer without producing distortion is doubled (power is thus quadrupled). This means that in a system suffering from high core-induced distortion (e.g. a supermarket system where the ceiling speakers produce "fuzz" whenever the music has bass content or the person paging has a low voice), the distortion will normally be ameliorated by inserting a high-pass filter in the audio signal path immediately before the power amplifier. Conversely, for example, if you have a transformer rated to deliver 60 Hz at 5 watts at less than 1 percent t.h.d., you can de-rate the input to somewhere below 1.25 watts or 35 volts for the 70.7-volt device, and operate it as low as 30 Hz.

Drew Daniels

Calendar

• On October 10-14, 1988, a short course on Underwater Acoustics and Signal Processing will be offered at Penn State University. Among the topics to be presented are: An Introduction to Acoustic and Sonar Concepts, Transducers and Arrays, Signal Processing, Active Echo Location, and Turbulent and Cavitation Noise.

For more information contact:

Dr. Alan D. Stuart
Penn State Graduate Program in Acoustics
PO Box 30
State College, PA 16804

• **The 130th SMPTE Technical Conference and Equipment Exhibit** will be held **October 15-19, 1988**, at the Jacob K. Javits Convention Center in New York City. The event annually provides a forum for discussions and demonstrations on advanced motion-picture and television technology. The theme of this conference is **"Innovations in Imaging and Sound."**

• Upcoming seminars for **SYNERGETIC AUDIO CONCEPTS** are:

Chicago—September 22-23
Minneapolis—September 27-28
St. Louis—October 6-7
Anaheim—November 1-2

Upcoming workshops are:

Grounding and Shielding (Los Angeles Area) November 17-19

Concert Sound Reinforcement (Los Angeles Area) January 17-19, 1989

Contact: RR #1, Box 267, Norman, IN 47264, (812) 995-8212.

ANNOUNCING

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***List Price after this offer: $37.50**

This trio of stories set in Los Angeles represents perhaps only the tip of the iceberg of what this great southern California city has to offer in professional audio. Two major post-production houses and a producer make up this present look at Los Angeles.

TMF COMMUNICATIONS

by COREY DAVIDSON

• While in Los Angeles, California, I visited a studio that I believed would be worthy of the *Electronic Cottage* section of this magazine. I had been told that TMF was a private home that had been transformed into a serious studio. What I didn't realize is that TMF Communications is a world-class facility with specialization in sound design. A great effort was made in order to create a technically superb studio and at the same time this exquisite estate, featuring posh and comfortable surroundings complete with household facilities, Jacuzzi and pool, were preserved.

My L.A. stay was dotted with various interviews and intermittent visits to the NAMM Show in Anaheim. TMF was certainly one of the highlights. I think that it is important to know that Ron Bloom is an eloquent man with much talent and savvy for music. My acquaintance with him goes back about 7 years to Atlanta, GA where I was working as a synthesist/programmer with a number of recording bands and artists. Ron Bloom, then and now, is regarded as a 'musical monster' by both the listening public and his peer professionals. It was a pleasure seeing him again in an environment that is as becoming to him as his stage performances and compositions.

TMF is co-owned by Chuck Norris and his wife, Dianne (who maintains a high level of activity in facility management and client relations). Together with Ron, they have established themselves as a major contender in the world of post-production while maintaining a strong recording artist clientele. In this studio profile, **db Magazine** explores the intricacies of this facility's strengths.

MIDI THROUGHOUT

Ron tells me, "The goal of the studio is to host any type of project and have it occur seamlessly with technology. If it is a keyboard related project or if you're in the keyboard stage of a rock and roll

Figure 1. The control room includes a Fairlight System III and Trident console. Note the strategically-placed, centered near-field monitor.

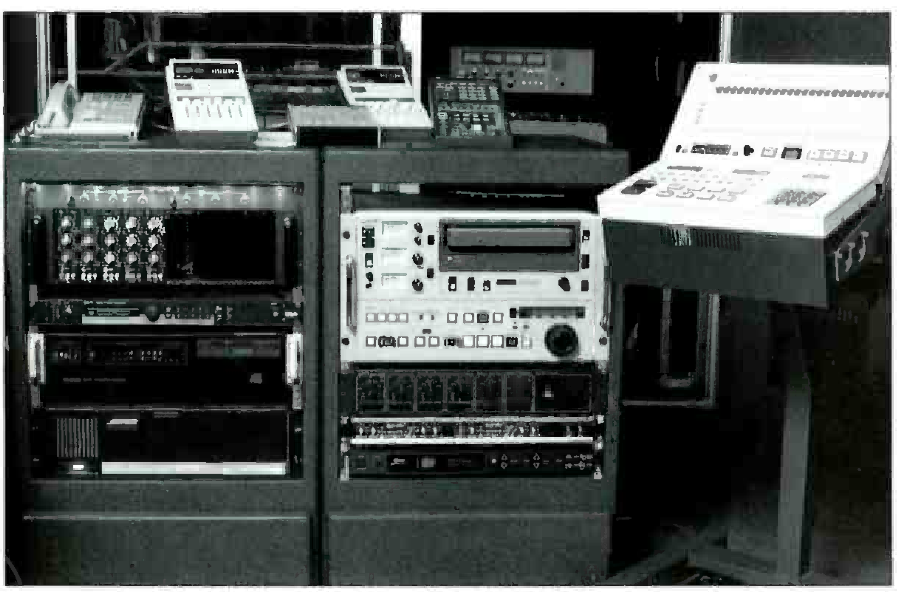
Figure 2. A full complement of signal processors and outboards. Note the alternate near-field monitor pair.

project or if you're doing a film score, the artist need not worry about interfacing hardware, cables, compatibility or any of the problems that typically arise when multiple keyboard generation is needed."

Ron is talking about the elaborate, yet simple to use MIDI system that has been wired throughout the studio. "We designed a MIDI interface system that is separate from our patch bay but works as a patch bay might work. All the direct MIDI processing gear that we have appears at our 16 MIDI channel bay. This MIDI bay is hand wired with Mogami MIDI cabling. When you run a MIDI cable over 20 feet, the quality of the cable becomes critical. Data speed can be maintained if careful attention is paid to cables. People who work with computers come to find out that there is a point at which one might want to use a line amplifier. In MIDI they have discovered that the quality of the cables can increase the length of cable that you can run before there is any noticeable deterioration of MIDI speed."

Are you utilizing the Series III in a tapeless capacity or primarily as a source for sounds?

Figure 3. Critical effects/processing, video editing and auto locators are all within comfortable reach from the console position.



"We use a MIDI computer (Syco-Logic) that has 16 MIDI in and 16 MIDI out and is programmable by patches. One channel can be designated as control and others as slaves in any number of configurations. This basically works out to be 16-factorial combination possibilities. Problems that typically occur, such as sticky note on/off information can be rectified by the Syco-Logic device. The synthesizers' functions and presets can also be altered in conjunction with the Syco-Logic."

"We believe that our sound-generating equipment reflects the best representatives of the various classes of synthesizers that are available in the world today. In the MIDI system we have a rack of the Yamaha TX (8-16) modules, a Fairlight series III which has hard-wired MIDI ins and outs, a Photon guitar converter with the Hyper Speed module, a DPX-1 (the Oberheim sample/disc player), and the Yamaha MEP-4. The MEP-4 enables all sorts of MIDI delay and echo and can actually be used as a time base corrector for MIDI against SMPTE. This is particularly useful because we are a SMPTE interlocked studio (utilizing the Adams Smith 2600 with remote control) and we do film work here. For those who are familiar with the Fairlight and the Sycologic, they know that those pieces are addressed from the back of the units, which slows down the work pace. All of our units have had their access points brought up to the front so that they operate like a traditional TT patch-bay. The advantage to all of this MIDI hyper-jive is that one can bring in virtually any configuration of synths, interface it to this studio real fast and forget about the incompatibilities and data stream problems that are so often encountered elsewhere."

SOUND SOURCE

Are you utilizing the Series III in a tapeless capacity or primarily as a source for sounds? "Fairlight has a digital-to-digital sampler and an analog-to-digital sampler that samples at 48 thousand cycles per second in stereo for many minutes depending upon the frequency response that you would like. This year, several things are being developed by Fairlight that will convert the unit into a tapeless studio. We're not really concerned right now with a tapeless studio. What we use the Series III for is, in the case of the live album that we're producing now, we sample key areas of the audience response,

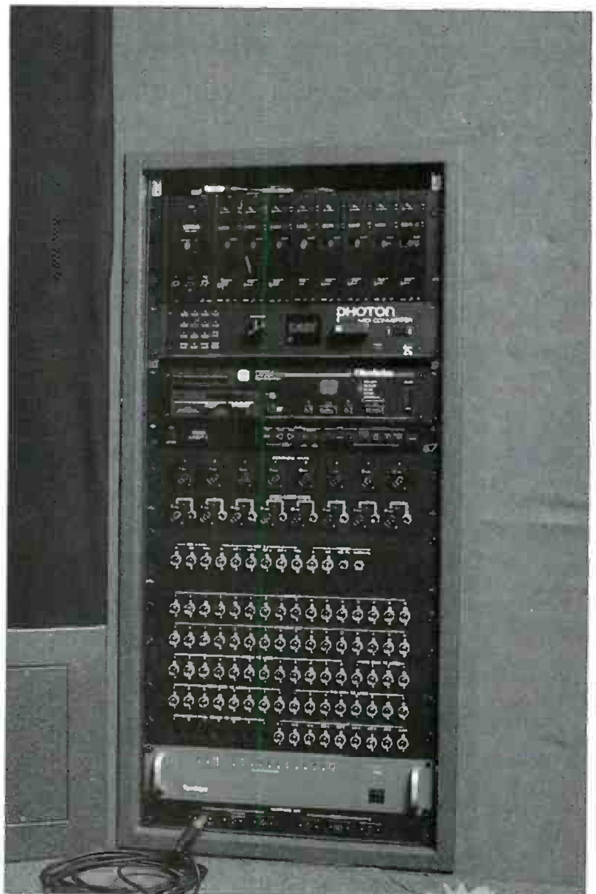


Figure 4. MIDI peripherals. Note the patch-bay and the choice of high or low impedance direct-instrument inputs.

lock the Fairlight up to SMPTE and mix one whole side of an album. When the multi-track machine stops, the SMPTE can trigger the Fairlight and the Fairlight can play a sampled version of the crowd response of that song. We can then stop, back up the machine and do a digital edit and start the next sequence going."

"So the Series III serves us as in fact partly a tapeless studio, part overdubbing machine, part scoring device, part library development of sound effects and part save your butt in case of a technical emergency. We just hosted the soundtrack for the movie *Wall Street* which was done primarily on the Fairlight by Stewart Copeland (drummer from *The Police*) here at our studio. In the editing stage, at the end of the movie, there is a song with vocals that was supposed to be the final theme for the movie. The producer, Oliver Stone, decided that he didn't want lyrics, so they would have had to come back in and recreate a song. There was no time for that. We sampled all the non-vocal parts of the song into the Fairlight, re-organized them, edited them, played it back on a 4-track tape as a master and that became the finished arrangement of the song. So that was a semi-tapeless situation."

Our other two smaller rooms each have small monitors for scoring pre-production.

"We don't like to think of ourselves as a keyboard studio. We are a music studio...if its a rock and roll band that has never in their lives used synthesizers, they might find a use for the Series III in terms of track-repair or sampling the first chorus of a live performance or sampling a great line that the singer did on a bad track. In the last case, instead of mixing it over again and bouncing it down, we can sample it into the Series III and trigger it with time code. A couple of good things about the our Fairlight is that its hardwired into the patch-bay, and the brain, like our tape machines, is in a separate room so that the noise from the Fairlight and the tape machines, not to mention the heat generated by it all, doesn't destroy the creative process in the control room. We ran all Mogami cable wherever there was the need for cable. For the *Wall Street* project we used two Fairlights. We're set up for multiple sampled instrumentation/implementation. We're also set up with the Elco wiring harnesses to bring in two 32-track digital machines and its already wired into the patch-bay."

"Our console has been modified to allow an extra monitoring bus to be used simultaneously with the re-mix bus so that you can monitor 72 channels at once and re-mix. The Trident Series 80 console has been modified to receive full automation (Master Mix) which is a SMPTE based, one channel, floppy disk automation with full muting, busing, assignments and grouping...the same automation found on the finest consoles. The monitoring system consists of Augspurger designed cabinets with TAD components. The power behind the speakers is Perreau. The room is acoustically designed to have a little bit of bounce, however, there are two mid-range traps in the ceiling, and a four foot base trap (tuned) in the back wall. Fortunately there is a minimum of equalization in this room which means that the room is as close to the right kind of listening room that we could possibly hope for. What we have in the back wall are tuned bats. We spent a lot of time cutting, sizing and shaping until every frequency that we thought was a problem was rectified. Several people were involved with the design of this room. Some of the people that we started construction with were not used as the construction progressed. We knew what we wanted and when someone told us that an idea wasn't feasible, we went and got someone else. We simply refused to take *average* for an answer."

"With a floating floor and floating walls and only two support beams in this room, this is a true floating room. No lead except for the doors and yet no RF problems which by the way was a carefully researched affair prior to and after construction. We were fortunate as far as the RF fields are concerned. Because we are a free-standing building in a residential area, we didn't have to do any exterior sound proofing other than limiting *sound-escape*. The filtering system for the swimming pool is much louder than any sound that might escape this control room where we often monitor at 5 million dB." That last statement of Ron's does not reflect an exact measurement!

"We do have a strong film scoring aspect here. The film work is handled on three quarter-inch video which interlocks via the Adams Smith synchronizer. A BVU-800 is the master deck. You just put in the three quarter-inch dub of the film, lock-up whatever you want with SMPTE or the Adams Smith, be it the tape machines and/or the Fairlight, designate any machine as the master, watch the film on the 28-inch monitor and go to scoring heaven. Our other two smaller rooms each have small monitors for scoring pre-production. Each of the three rooms has its own air conditioning system and over-sized ducting eliminates any air noise whatsoever...you can hear a pin drop (on carpeting) before you will hear any noise in the room."

"There was a need to make sure that TMF would be compatible with outside equipment. Although we have so many outboards, this is no reason to limit the possibility that someone might want to bring in his own. The patch-bay interfaces all outboard gear and has lines to accommodate a slew of outboards that one might want to bring from outside of this studio. The goal is not to be a super-public studio but rather to be a private studio where only a handful of producers/engineers work in the room. Rick and I have teamed up to do a lot of rock and roll and we hope to keep the studio busy with our projects about 6 or 8 months out of the year and the remainder film work. The key for us is to allow only people that can appreciate what we are trying to do to work here."

WORLD CLASS ELECTRONIC COTTAGE

How did you arrive at this place for the site for a studio? "We searched and searched. We knew that we were not going for a conventional approach or a conventional space. The entire studio as you see it was a family room with a linoleum floor poured over concrete. The concrete was very firm and stable and was anchored into the ground. We then investigated vibration by having a selected person drive by while an *inside man* observed. The support system for the roof was minimal so the crossbeams did most of the work which means, in simple terms, there weren't a lot of poles in the way. To maintain specific non-parallel surfaces, we wanted to keep the integrity of the view to our backyard. We created a triple paned window, as if it were a control room window to the outside world and made sure that there were no parallels to the glass of the taperoom doors."

"The cue system is extremely accommodating. Headphone cues can be picked up at the console and at the outboard racks. There are separate headphone cue sends and receives so that when you're doing a headphone check you can listen to both cues simultaneously. There are headphone outlets at the MIDI rack for the keyboard player who might want to play in the control room and go through headphones. The client lounge is designed so that it has a pull-through for cable in the event that another live room is needed."

When we do film, we utilize our quad bus modification (on the Trident) that enables us to deal with the surround sound

If you had to describe this place to someone who might not quite comprehend the concept behind your studio, how might you do so in just a few words? "Well...it is not a home studio. It is a professional world-class recording facility with the luxury of being in a high quality home."

Sounds like a vacation from a vacation. "That's the way that we see it. Sometimes we like to tell people that we have a home studio 'cause when they walk through the door they're pretty much devastated by what the studio's capable of accomplishing. As you noticed, when you walked into the house, you really don't get the slightest clue (other than a reception desk nestled into a gorgeous living room) that there is a studio here. Then you walk in the one door to the main room and you begin to enter a different world. The studio is designed to give you a solid 10 to 15 hour work day...without fatigue. Yesterday was a 22 hour day for us. No sweat."

Rick Delena adds, "That was 22 hours of *speed metal* so that actually makes it 44 hours."

Ron continues, "When we do film, we utilize our quad bus modification (on the Trident) that enables us to deal with the surround sound, THX and all those people that bring in their goodies when we have to do a mix with theater/film-compatible sound. We also usually deliver a 4-track time-coded half-inch master."

Where do you begin a dialogue with a new client in order to ensure that the client will indeed get what he or she wants? "When a band, film, or a television show comes in, all we ask them to do is to describe in their words what they need. A part of that communication is for the client to understand that we would like to do *our* job too. If say, we decide that somehow the tempo of a speed metal band has to be improved by using the Fairlight, we can integrate that idea because we're Fairlight experts. If we have a techno-pop project that needs some realism, we don't just have a computer programmer sitting there banging out drum beats...we know how to loosen up a machine generated rhythm so that it sounds *real*."

FILM SOUND PACKAGE

I've been informed that you are currently working on a Chuck Norris film. "We call ourselves a communications company because we do any project that uses music and/or sound to communicate. For a movie such as *Braddock: Missing In Action III* (starring Chuck Norris), we did three key songs for the movie which I wrote and together we produced here at TMF. I did not score that film. We interfaced with another scorer named Jay Chadaway. However, I just completely scored a film called *Another Chance*. Besides myself, we utilize other talent like Tim Truman and a team by the name of Michael and Rand Weatherwax. Together, we offer our clients in the film industry a complete package that includes the design of custom sound effects, true contemporary foley, score, and the writing and production of key songs. We have recently begun to offer this service with the Fairlight as the center piece of our sound effects and foley processing system."

"For those who are familiar with traditional methods of capturing sound for film in a foley stage, know that the frequency response and the control of the Fairlight gives us an opportunity to do some very creative things. We're doing 10 to 15 demos a week for directors and producers who want to buy the whole package. In Chuck's film we did custom sound effects for speed-boat launches, a chase scene and aviation sounds."

Do you intend to increase your post production clientele at the expense of the album oriented clientele? "No! That's why we call ourselves TMF 'Communications,' not TMF Productions, not TMF Studios. We have a commitment to maintaining a grasp of the whole picture. This means that no medium will be neglected. Fifteen years ago music was a backdrop for

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film. Today, music is crucial. As music becomes more crucial, everything that makes a sound competes with the music. The music, in turn, competes with the dialogue. It only goes to follow that the next step in technology for film will be to use the same creativity and knowledge of pitch, a knowledge of tension, a knowledge of awareness that musical people have, combined with a historical skill that foley people have. A marriage of two gigantic worlds. This is what enables one to offer a total package to a film's director or producer with the consummate skill and ability to understand technological/artistic compatibilities."

The technology is apparent to a technical person, but to a non-technical person it is the ease and pleasure of being creative and recording.

EXPERIENCED PERSONNEL

What advice might you give to an up and coming facility? "Bring in as many experts as you can find because no two experts agree on everything. When you're finished with all the experts, find the guys that make the records. Ricky consented to work here, and I say 'consented' because he saw the studio 10 times. Each time he'd come, something else had been designed, added or improved. As the flexibility factors increased, his enthusiasm increased. When there are experienced people that want to work in your facility, then you know that you have done something right. When an engineer who has never done a record, or a producer that has never worked on a score tells me that my room sounds good, that's not good enough. It shouldn't be good enough for anyone else who builds a studio."

Rick tells us, "I was at The Record Plant (NY) for about 12 years as a house engineer. I started as an assistant engineer. I then came out here and worked at Village, A&M, and many other studios. This studio was quite a find. When you find a comfortable environment that can produce the desired results, you usually stick to it. A&M and the Record Plant are like that. But a place such as this is a dream for a guy like me. I've done many records at many other studios, yet I'm very turned on by the comfort factors and technological soundness of TMF. The advantage for me here lies in the finely tuned skills that couple with the finely tuned tools. It is also refreshing to work with someone who has a keen understanding of music and machines. That's Ron."

Ron adds a final philosophical note, "There is a set of criteria that TMF requires: The recording equipment and instruments have to sound great, nothing can break (if so...backed up pronto!), we have to love being in the room, and it has to be easy to use. Lump that all together and you come up with what we call *seamless* recording (as opposed to *tapeless* recording). The technology disappears underneath the creativity. The technology is apparent to a technical person, but to a non-technical person it is the ease and pleasure of being creative and recording. There is a comfort zone here and that zone is created by staff, equipment, seamless technology, superb electronic condition, and the day-to-day maintenance of it all."

WHAT'S AT TMF

STUDIO A

Trident Series 80 console

64 channels-Audio Kinetics Master Mix automation

Otari MTR-90 24-track recorder w/remote

Otari MTR-20 4-track recorder w/4-track .5-inch, 2-track .5-inch, 2-track .25-inch head stacks w/remote.

Various 2-track recorders.

TAD monitoring system housed in George Augsberger design cabinets in conjunction with Perreau and Hafler amplification w/White EQ.

JBL 4311B studio monitor

JBL 4410 studio monitor

JBL 4412R studio monitor

Auratone 5C studio monitors (3 pair)

Adams Smith compact controller

Adams Smith 2600 synchronization system

Roland SBX 80 SMPTE/MIDI synchronization system

Sony U-Matic video recorder

Sony BVU 800 video recorder

Sony BKS remote

Sony Trinitron video monitor

Fairlight Series III music computer system (fully loaded)

Yamaha DX7 digital algorithm synthesizer

Ensoniq ESQ1 digital wave synthesizer

Roland Juno 106 analog synthesizer

Korg Poly 800 analog synthesizer

Oberheim DPX digital sample player

Yamaha CP80 electric grand piano

Kimball 5100 baby grand piano

Alesis HR16 drum machine

Yamaha RX11 drum machine

Roland TR-727 rhythm composer

Kurzweil Midiboard controller keyboard

Syco-Logic MIDI switching system

Yamaha MEP 4 MIDI event processor

Photon MIDI guitar system

Roland MC-500 MIDI sequencer

Yamaha QX1 MIDI sequencer

Macintosh Plus computer and printer w/Performer 1 & 2 MIDI sequencing software and others.

Apple IIE computer w/printer

Lexicon 480L digital reverb w/LARK

API EQ modules (6)

Drawmer dual gate DS201 (4) stereo

Valley People-Gatex (2)

Roland SDE 1000 digital delay

Roland SRV 2000 MIDI digital reverb (3)

Roland SRV 3000 MIDI digital reverb (2)

DR2 digital reverb

Yamaha SPX 90 multi-effects processor

Lexicon PCM 70 digital processor

Lexicon PCM 60 digital processor

Lexicon PCM 42 digital delay

Lexicon PCM 41 digital delay (4)

Lexicon Prime Time digital delay

UREI 1176LN peak limiter (3)

DBX 163 compressor

MXR pitch shift doubler (2)

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PRODUCER TOM WERMAN

Why does a rock and roll band choose a specific producer? Why are the record labels so concerned about the band/producer relationship? Can a producer make or break a band's impact? How crucial is the producer to a record's success?

by COREY DAVIDSON

• These are some of the questions that often arise when discussing that gray area known as production. *Producer* is an impressive title and yet it is the producers that seem to get stuck with the reputation when the sessions go very well or fail. How often have you heard this one from a musician/band member: "If only we had the right producer" or "That producer really screwed up our record" or "We won't do the record unless we get Mr. Z to produce it." Producers are the people that, together with engineers, bring the musicians' ideas to the reality of the recorded mediums. Once the musicians know what they are doing, the onus is on the producers/engineers to see that a product is turned out in behalf of the band for the label.

Now, how does one distinguish the producers from the band? The quickest and easiest way is to read the credits. However, in this story we're hoping that some things are brought to light that can't be read in the credits. Sometimes the producer's work is so good that you can't really hear it...it might sound *natural.* At other times the record might sound like a strange brew conceived by a mad scientist. Still, in other instances a record might sound like something that couldn't have possibly been made on this planet. Whatever the case may be, it is the producer who establishes the links between the band, the engineers, the label, and ultimately the listeners.

I met with producer Tom Werman and his two engineer/assistants, Duane Baron and John Purdell, in the midst of sessions with the band *Poison* at One On One studios in North Hollywood, California. Tom, flanked by his two engineers, offered a three-headed informative and at times surprising first-hand perspective on the mechanics of production.

LOVE OF DIGITAL

Mr. Werman told me that he had read the story on producer Frank Filipetti (db November/December 1987) and was intrigued by the issue at hand. That story was, in essence, an in-depth discussion of digital and analog tape recorders and the comparisons that Frank had made between them.

Tom says, "I'm not a technical guy but I love the digital medium...especially for the kinds of projects that I do. I don't have a particularly extensive musical background other than the rock and roll bands that I played guitar for when I was younger. I went to college and then attended business school, went into advertising for a year and hated it."

Tom Werman at his console.

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"I asked CBS records to give me a job in the A&R department because I still loved music. After more than six years of signing bands, I started to produce...signed and produced bands for six more years, then decided to leave. Upon leaving my position as A&R man/producer, I worked briefly at Electra as head of A&R, chose to leave that position and have been an independent producer for the last five years."

Do they call upon you for a recording project or do you decide which bands you will produce? "They find me. I've asked to do a couple of projects but for the most part my services are solicited by labels and bands. I don't have a manager but I usually get the calls directly from the record companies or band managers. I turn down a lot of projects...most of those bands are new bands and its very tough to do new bands so once in a while I prefer to work with an established band. To date I've produced about fifty records."

Did you pick the studio for Poison? "Yes. We have been working exclusively at Conway for the last year but we were here before going to Conway and we like this studio very much. We came to One On One because it's comfortable, sonically *together*, and it's nice to change the environment once in a while during the course of a project."

What was the chosen medium for this band project? "It's all digital...namely the Mitsubishi X-850 machines. I love what digital offers for rock and roll although I feel that the medium is not all that critical when it comes to this type of rock and roll. In the beginning when digital first came out, we were continually invited to all sorts of demonstrations. The studios, in conjunction with various manufacturers hosted these demonstrations and to this day I am still not sure as to what was technically being done at those demonstrations. Whatever I heard at those demonstrations was usually too clean...too sterile. It just didn't seem to have the *juice* that I feel is required for rock and roll."

What do you think that juice is? Duane answers, "Spirit. That juice is the guts of what the band sounds like in the room. The low-end fatness, stomping drums, and crunching guitars make up that juice. I love digital and the way that it reproduces information. I wasn't crazy about the X80. The converters were not what they are now on the X850s. I'm talking about machines with no front and rear analog filters. The X80s were probably developmental for Mitsubishi, however, I think that their new two-track digital machines are right on the money."

Do you record as hot as you can or do you lighten up on your levels because of the 90 dB of headroom that digital offers? Duane responds, "Engineers are still learning and realizing that there is a different *mindset* for digital recording. I record as hot as I can...just short of the distortion threshold. The reasoning is simply that digital makes its best showing when you fill up the bit space. With analog recording you're looking out for compression and saturation."

"In the beginning engineers were having a hard time relating to the fact that they were no longer watching meters in the same way that they watched meters on analog recorders. One way that you might look at this is to think of the track as a *sample*...the kind of sampling that you might do with an Emulator or the like. When you record (sample) too low, the harmonic information changes. On the other hand when you record too hot, there is total distortion."

WHERE NOT TO COMPENSATE

Duane, what compensations, if any, must be made in the wake of this new digital mindset? "One of the hard things about making the transition from analog to digital is that there is an absence of compensations that one might have gotten used to with analog...such as over-adding your top end or any kind of anticipation of losses throughout the frequency spectrum. When recording in digital, compensation means that you have actually changed the source information. Digital, when used properly, really does give back what you put in. So the hard thing was learning where *not* to compensate. For me, I believe that I have found a medium that is a truer representation of the source."

Tom, if you are not watching the meters and tweaking the gear, then what are you actually doing in the studio?

How does all this pertain to rock and roll projects? Duane replies, "It's very easy to saturate analog tape, especially in rock and roll. There are many peaks, the players hit hard and play heavy...usually at a bone shattering volume. You've got drums that can easily become compressed by analog recordings. By the way, Frank Filipetti said, in your story, that he could hear analog's compression of transients every time he records those types of sounds. A digital recorder will accurately reproduce whatever it sees right on up to the distortion threshold."

Duane, let me ask you this. In a medium that is as unforgiving as digital, engineers and producers typically prefer to work with players that have a very good handle on dynamics. So far the popular consensus in the industry is that digital is best suited to the dynamic music genres such as new-age/experimental, jazz and classical/orchestral. However, in rock and roll there are more peaks, saturations and sometimes less musical experience and background. What do you do to safeguard the digital recording process? "You've got to be careful! That is my responsibility as an engineer. It is also my responsibility to someone like Tom that I ensure his intentions too. In being careful, I have to be extra aware of little nuances that I really didn't have to worry nearly as much about when recording analog. Little pops and clicks can no longer be tucked into the rest of the information."

"Even when recording loud saturated guitars and Marshall amplifiers, levels must be babysat. With analog you could run a track into saturation and it often didn't make much of a difference. Now, with digital, you run the risk of altering the harmonic content if you do not record properly. Another point to make is that digital enables us to record lower frequencies than ever before. It's not unusual (when recording digitally) to record frequencies down to 2 Hz. Recording frequencies that are very low become destructive on analog recorders. The sound seems to break up when the frequency content gets down that low in analog."

Has compression become the tool with which you avoid trouble when making digital rock and roll recordings? "The strange thing is that I have found myself using far less outboard compression when recording digitally. I've always avoided relying upon compressors. Analog machines compress powerful peaks. Digital recorders allow those peaks to reach the top. Again, the engineers job is to protect the medium's integrity."

Tom, if you are not watching the meters and tweaking the gear, then what are you actually doing in the studio? "First of all, I'm hired by the band. It's not my record...I'm here to help them make their record but my role is pretty evenly applied throughout the whole project from the selection of musical material to the end of the mix and final mastering. The main role that I serve is to prepare and adjust material before it gets to the studio. No band goes into the studio with ten equally wonderful hits so you try to make sure that there is a nice variety of material in order to best display the band's strengths. Even if they do one thing better than all the other things, that strength must be capitalized."

"With *Motley Crüe* I was very happy to see the band generate some instrumental diversification such as using slide guitar. I wouldn't be very interested in a project that was based in nothing but one-four-five chord patterns. Beyond the choice of material (which is done at home by asking the band to give me everything they've got on cassette), I will then pick my favorite 14 songs. The band and I then have a meeting to decide upon which ones we will record. As the recording progresses we might ditch the weakest songs along the way before we finish them...we give them a chance." "The most important thing that I do is to supervise and arrange the whole song and the parts. This is where John's input is critical. These days there are many options with sampling and other electronic devices that enable careful observation of kick drum patterns, cymbal placement, vocal composites, etc. I believe that I am able to bring out the best in a musicians performance. That is why I have to be in the studio for the entire duration of the project. If I wasn't there, I might miss one or two things (if not more) that could mean the difference between a good performance and an exceptional performance. John knows how to use the musical tools like a microscope...again, this is the difference between *good* and *exceptional*."

John adds, "There is a big difference working with bands as opposed to working with solo artists. In a band-oriented recording environment, you must deal with each one's strengths and weaknesses. With a solo artist you can hire whoever you need that will be adequate for the job. With a band, you're stuck with the line-up. This can be a wonderful challenge."

Tom interjects, "This is why I love the digital revolution. It has probably helped me, as a producer, more than most of the other guys. When it comes to individual musicians, particularly vocals and guitars, I like to do all the recording myself. At this point in a recording I like to operate the tape and punches. Punching on an X850 is heaven. You can punch-in entire drum kits if you have to. As far as the performance is concerned, you can make seamless punches all day long. So the only limitation is my ability to get in and out whenever I want. A few years ago it became faster for me to get in and out. Many years ago, prior to my ability to operate the machines, I was dependent upon engineers to get me in and out. I realized that I had to learn how to do this myself in order to best serve the music. Now that the machines have become fast and silent, it is too tempting...you can punch anywhere you want. I did a lot of punching on analog machines but the digital machines have enhanced and improved my abilities to edit. With analog recording, you have to be a little more resourceful in hiding your punches."

It is really up to the individual band as to how far they want to employ sampling techniques.

Do you utilize four or eight-track preliminary recordings before entering the studio? "Well, the best answer to that question is: little to none. I might do some tests with drums, bass, and rhythm guitar to make sure that the foundation of the music is solid and driving...that's as far as I'll go with pre-demos. I can't recall ever doing that kind of pre-production for more than four or five days because once you get the basic track together, the fun happens in the studio. No matter how well you plan it out, it always comes out quite different when you get into the studio."

A FRESH APPROACH

It seems to be a popular consensus amongst producers that pre-production is essential...that all the material must be in place. Some go so far as to say that they would not set foot in the studio until the music and parts are rehearsed and letter perfect. Any comments, Tom? "Those producers' approach to pre-production, particularly in rock and roll, will ensure that the band will be stale and tired of the material by the time they get ready to record. If you can't make changes in the studio, you can't produce. You might as well rehearse them to death, and then go in and make either a very sterile album, or make a live album where you use the studio for the control and separation. Under those conditions, all you would have to do is check-in at the end of the day to make sure that things were recorded as they were rehearsed. It never occurred to me to work that way. With repeated listening to the music, I obtain a grasp of the music that is as fresh as the band. The band and me...me and the band...one unit in the studio."

Is John's special kind of input a new endeavor for you? "We're still working on that. In the beginning John was mainly a keyboard player doing synth parts after everything was recorded. Then he became a drum programmer, background singer, and sampling expert. He performs but also co-produces with Duane. It's like having a permanent, floating band member for each project and it helps keep the albums fresh because after a while, I would probably become redundant with my ideas for harmonies. So John comes up with ideas that I don't."

John, how much have keyboards, samplers, MIDI, and other electronic treats infiltrated the rock and roll sector? "There is a lot. I don't know if others are going as far as we have jumped into it. We'll hear a sound, for instance, a drum sound that can be made more potent, then we'll approach the drummer and see if he likes our idea to change the part. We can then sample his drum sound and re-inject a new musical idea. This is nearly impossible to do by having the drummer play the new part. Drums are part of the basics and they typically cannot be changed later. This sampling/injection technique gives us unprecedented flexibility. We have discovered that this same process can work wonders for the bass guitar too."

Is there a sampler that you prefer to work with? John replies, "The Emulator II has been a preferred instrument for us. It is particularly fast. A fast pace of the sampling process helps to keep a session in motion as opposed to a long drawn out affair that can turn a musician cold."

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Tom adds, "It is really up to the individual band as to how far they want to employ sampling techniques. On occasion you might want an incessant bass line that runs throughout the piece (as on various tracks from ZZ Top's Eliminator album). In a case such as this there really is no advantage to having a musician play the part *organically*. The nature of recording has changed and you have to fight fire with fire. A well written rock and roll song that depends upon a locomotive kind of rhythm track simply doesn't sound as good anymore if it doesn't employ machines in some way. We were initially paranoid of these machines when they first reared their ugly heads. They have gotten much better looking since then. If you don't embrace the newest recording technology available, you're being silly. One of John's greatest assets is that he can take each player and walk him through the technical possibilities."

"Some years ago, there was a band that infuriated the public when it became known that they had been using tape in conjunction with their live performance and had not informed the audience. That incident was a milestone for live performances because it wasn't too much later that *The Who* needed to assemble their tour of the **Who's Next** album. There were many synthesizer parts that couldn't be executed by the band and they had to use tape. That tour was a monumental success. I believe that the **Who's Next** album was the first truly successful marriage of synthesizer music and hard rock. Then the use of tapes on stage became acceptable. You no longer have to inform an audience when tape is being used."

ENGINEERING SUPPORT

Tom, what kind of expectations do you have for your engineers? "I never sat down and talked with other producers to find out how much they know, but I do know that a lot of producers out there were engineers. Right off-the-bat they know a hell of a lot more than I do. I wasn't cut out to be technical and I don't read or write music. I bought an eight-track Tascam machine with all the peripherals that would enable me to make great little demos at home and I was really exited about it. I was so bogged-down...I didn't even understand the function of a stereo bus. (We find that hard to believe Tom, but we'll take your word on it.) So in answer to your question, I rely completely on my engineers. I do know enough about the equipment in terms of what it is and what it does, to occasionally suggest something that's not impossible. We learn how to talk to each other and he (the engineer) learns to interpret what I want. The more records we make together, the more heavily I depend on him...which frees me up to concentrate on the music."

If you were to use a great session drummer with great sounds, it might be difficult to tell the difference between the drummer and the machine.

Do you feel thwarted by the recent onslaught of technology and the duplication of so many types of outboards? "When I first started producing, there were three or four ways to go. There was digital delay, tape delay, and plate reverb. You could turn the tape over and get backwards effects. The consoles were very simple. Now there is too much for me to get carried away with. I have readjusted my thinking and no longer feel an iota of guilt for not knowing how and why everything works. Look what happens when you get carried away with machines...you get dance music...especially young female vocalists who make dance music. That's the unfortunate part about recent technological advances. To me, dance music represents retched excess. It means that a producer and an engineer, or an arranger and an engineer, or just an engineer (if he's a musical engineer), can go into the studio, make an entire record and at the last minute call in a vocalist to sing over it and have a number one hit. You could even have a little career that is nothing more than an assembly-line approach to what once was spontaneously creative. On the other end of the spectrum you can look to incorporate technology and tastefully apply one or two specific synthesizers or machines to improve and strengthen your brand of music. Machines can strengthen the material but shouldn't become the material."

John, do you have any thoughts on this techno-music matter? "Some of the best dance music that is currently being made is not all machines. My favorite dance music usually incorporates live sounds with synthetic sounds. The more successful dance artists who have been a part of the scene and are not one shot deals, understand that there must be a natural feel to dance music. You can't just run a sequencer that has everything auto-corrected to the same note resolution and expect it to yield a *musical feel*. There is a great deal of knowledge involved in making these technologies work well with live music. This is what Tom, Duane and I are doing. We use the instrument/machine technology to tighten a rock and roll sound."

Tom tells me, "We have managed to recreate a drummer's kit by our sampling injection techniques without the loss of the original drummer's feel and character."

John says, "We use triggers on the drums as the device by which we get to a MIDI format. That MIDI information is now stored in a computer. Once all of the triggers and dynamics have been stored, we can then assign all events and their respective levels of dynamics to the sampled sounds."

Tom states, "In a fill where the drummer is hitting the snare and each hit is a different intensity, the snare drum itself yields a different sound with each hit. John's expertise in this case, lies in his ability to document these different dynamic events and assign them to different samples that become a faithful recreation of the original snare fill. You need a musician to direct and use the machine properly. If you get a technoid to run this sort of system, you get a drum machine. If you use a good drummer, and you use John, and the system, you get a brilliant/well-played drum part."

John adds, "When you're recording a live kit there are often leakage problems. For instance, the snare mic might pick up the hi-hat sounds...they are very close to each other. The same problem arises with other drums with or without the use of gates. The injection techniques that we use for drums enables us to isolate each drum so that there is absolutely no other sound than the drum that we want with its respective signal processing. Equalization also becomes more effective because there are no longer compromises and compensations that have to be made...perfect isolation."

Tom says, "If you were to use a great session drummer with great sounds, it might be difficult to tell the difference between the drummer and the machine. But if you take a guy who is in a heavy metal band that has never made

a recording and has been playing in rock dives all around town, that's another story. If you take this heavy metal drummer and put him behind a battery of 1988 microphones and record on a digital machine, you're in big trouble without a few options. So we're not trying to pull the wool over anybody's eyes (ears). We simply know what we have to do to optimize the impact of the music."

THE VILLAGE RECORDER

Here is a studio operation that offers complete production and support services in one convenient location

by SCOTT WILKINSON

• Located in an unassuming three-story building just off Santa Monica Blvd. and the San Diego Freeway in Los Angeles, California, the Village Recorder offers state-of-the-art recording facilities in a comfortable, wood-paneled environment. A staff of 32 includes engineers, assistant engineers and traffic managers as well as accounting, secretarial and maintenance personnel. The four main studios are equipped to handle projects ranging from albums, movie and television soundtracks, video editing and jingles to dialogue replacement, sound effects and even books on tape.

The Village Recorder has formed a unique partnership with the tenants to whom they rent office space on the second and third floors of the building. Nick Smerigan, Executive Director and Manager of the Village, explains. "I try to maintain a group of tenants in which everyone does something with the studio. We have a management company (Village Producers), a PR company (Bobbi Marcus Public Relations), a graphic design artist (Todd Pearl), writer/producer Robbie Robertson, arranger Steve Lindsey (who also works with Randy Jackson and Richard Perry), producer P. J. Knowles of Like DAT Music, composer Patrick Williams and producer/arranger Mark Hudson (of the Hudson Brothers). All of the people who rent space from us have a need for the studio and we have a need for them. Everyone provides us with business and we provide them with business, office space and a sanctuary. Our reception area can act like the front end for all of the tenants. The receptionist can take calls for all of them. Anyone wanting to see a tenant must go through our reception area first. You can't get up to their offices unless they approve. We provide a buffer for them."

ONE-STOP SHOPPING

Smerigan is also quick to point out the advantages of this partnership to the clients who wish to use the facilities. "Anytime someone has a project that includes all of the support activities that surround any type of recording, they can do it all right here in this place without ever leaving. They can get their artwork, PR and arranging done here in addition to using the studio and/or video edit bay. If they need it, we can also provide management and production services as well. Whatever they need to do, it's here."

This association poses some unique problems as well. "It's like an extended family. Everybody has to get along because we are all locked in here. It's almost like getting renters to share a house with you. It's not always easy to do. We all have to agree on new tenants. No one person makes the decision on who's coming in. We have 32 people in the studio and when you add another 8 or 10 tenants, that's a lot of people to be jammed into 24,000 square feet, particularly since we are all involved with one another. So it's important that the 32 of us can work comfortably with the 8 or 10 of them."

Village Recorder's Studio D. Note the monitors and the view into the studio.

Scott Wilkinson is a MIDI and Electronic Music Consultant in Los Angeles. He is the author of "Tuning In: Microtonality in Electronic Music,"

Village Recorder's Studio F.

The funny thing about this room is that many of the rock 'n rollers scoff at its size and dead acoustics, but then they come in here to do guitar or vocal overdubs.

In such a cooperative atmosphere, these problems are easily solved. Smerigan is very happy about the way things have turned out for all concerned. "So far, it has worked quite well because we get a lot of subsidiary business from the tenants. They get the benefit of using the studio without leaving the building in which their business is located. In addition, they have the advantage of offering their clients (especially those in the movie and advertisement business) a complete office and production environment without having to run all over town. Their clients like the setup because it is so much safer for them. They have one place to call to get everything done. That's the most important thing for the client, especially in the advertisement business where everything is so fast-paced."

Patrick Williams is the newest tenant at the Village. A prolific composer for television and feature films, Williams' credits include thousands of TV shows and over 65 feature films. He wrote the scores for almost all of the MTM shows in the early years as well as *The Streets of San Francisco*, *The Slap Maxwell Story* and *The Days and Nights of Molly Dodd*. His film credits include *Breaking Away*, *The Best Little Whorehouse in Texas*, *My Turn* and *Violets Are Blue* to name a few.

SPECIAL ATTENTION

Williams finds the facilities at the Village to be top-flight. "I'm very happy here. This month has been particularly busy for me. We brought in a television movie (*Double Standard* to be shown on CBS) and a theatrical feature film (*Fresh Horses* for the Weintraub Group with Molly Ringwald and Andrew McCarthy). We're doing the entire soundtrack for Fresh Horses here in Studio D. We'll be finished by the first part of September. We're using three synthesizers, a couple of guitars and a saxophone to do the tracks and then sweetening with thirty strings."

Even before becoming a tenant, Williams was no stranger to the Village. He recorded part of an album for his Sound Wings label here last year. "The studios are not only comfortable, but very well equipped. That's what I need. They've got a lot of the things I require right here in this facility. I love Studio F. We just did almost all of the soundtrack for Double Standard in F. We did it all with pre-programmed synths and a few live musicians."

Williams recalls an incident that illustrates the satisfaction he feels working at the Village. "I've been working early. I often come here at 5:30 or 6:00 in the morning. One Sunday morning, I was working around 9:00 AM. Not exactly the high point of normal studio operations! I have a DX7 and a small tape deck here in my office that I can use to record the DX7 or monitor the tape. I needed to record and play with the existing tracks at the same time, but couldn't with the setup as it was. I went downstairs and found one of the maintenance guys trying to open his eyes. I explained my problem to him and within 5 minutes, he had a mixer in here with patch cords and everything else I needed. Later, I thought about how I might have done that somewhere else, you know, what I would have been up against. That kind of attention and service is something that distinguishes the Village."

SELF-CONTAINED

The facility itself includes three main rooms (Studios A, B and D) on the first floor and a smaller audio/post room (Studio F) on the second floor. All three major control rooms have recently been refurbished and brought into the 80s in a cooperative effort by Charlie Brewer (head of maintenance), Jeff Harris (chief engineer) and studio designer Vince Van Haaf. Terry Davis implemented their ideas as he performed the actual installation and remodeling. His company, Golden Saw Construction, has installed studios for Lorimar, Paramount and the Record Plant. Studios A, B and D also have a separate lounge complete with kitchen, bathroom, TV & VCR.

"Clients can come in and be comfortable. It's completely self-contained. They never have to leave to find comfortable alternate surroundings."

Studio F includes a Trident T-24 console in a 36x24x24 configuration, Otari MX80 24-track machine, MCI 1/2-inch 4-track and 1/4-inch 2-track machines, Sony PCM701 digital processor and 5800 3/4-inch VCR, two Dynamax cart machines, Lynx synchronizers and controllers, Sony KV2040 20-inch video monitor, JBL 4411, Yamaha NS-10M and Auratone 5C audio monitors as well as a full complement of signal processing equipment and microphones. The booth is quite small, able to accommodate only three or four people at most.

Smerigan notes with a laugh, "The funny thing about this room is that many of the rock 'n rollers scoff at its size and dead acoustics, but then they come in here to do guitar or vocal overdubs. It's working out quite well for us. We're starting to hit our stride with it and getting a little reputation in this particular part of the business. We have a custom-built podium and chairs and a huge desk where people can spread out their scripts. We do vocal overdubs, voiceovers, sound effects and books on tape in this room. Ed Asner was in here last week reading a book for Bantam Books On Tape."

FEATURED EQUIPMENT

Studio D is the largest of the three major rooms at the Village. The control room was completely redecorated and refurbished in January with a new Neve Series 60 board with 60 input channels. However, they left the recording booth alone. Smerigan notes, "It has always been a classic studio. We didn't want to disturb that in any way."

The Neve board is fully computerized with a Necam 96 system. Other equipment in this room includes a Studer A800 24-track machine, two Ampex ATR-102 2-track machines, Lynx synchronizer, custom large audio monitors, a variety of near-field monitors, and a full complement of outboard gear. The booth also houses a Yamaha 9-foot concert grand with MIDI. Studio D is best suited for tracking, overdubs, mixing, video scoring and sweetening. It can provide a wide range of acoustic environments with two large isolation booths in addition to the main room.

Studio B includes a small overdub booth (although somewhat larger than the booth in Studio F) that includes a 7-foot Steinway piano. Smerigan points out, "This is where Robbie Robertson tracked some of his album—bass drum, guitars, keyboards—because he wanted that 'close' feeling you can get in here."

The control room includes a Neve 8108 board with 48 inputs and Necam I automation. The tape decks and other equipment are virtually identical to that found in Studio D. Studio B is used mostly for overdubs, mixdown and sweetening. It also features a hydraulic drum/piano stage and a separate machine room. Studio A includes an SSL 4056E console with 56 inputs and SSL Total Recall automation. Like Studios D and B, this room includes a similar complement of equipment and a 7-foot Yamaha grand piano. It is used mostly for tracking, overdubs, mixing and sweetening. It also includes an isolation booth and separate machine room.

Clients can count on comprehensive service that addresses all of their needs in one convenient location.

Of Studio A, Smerigan says, "Most clients like to mix in this room. We have four perfect size rooms. You can track in D, overdub in B or F, and come in here and mix or overdub as well."

The use of each room is not predetermined. Smerigan believes that this has always been a very subjective art form. "I have people who like to track in D, overdub in A and go back to D or B to mix. I have clients who like to do their drums in here, go into D to record the rest and come back in here to mix. It's all a matter of personal taste. With our facilities, they can do it any way they want. I laugh when people say 'make it a tracking room.' How do you do that? Do I then tell clients 'You must have more than four instruments out there?' Ultimately, it's just like a hotel. We rent you the room and whatever you do in there is your business. Whatever they want, we try to provide for them."

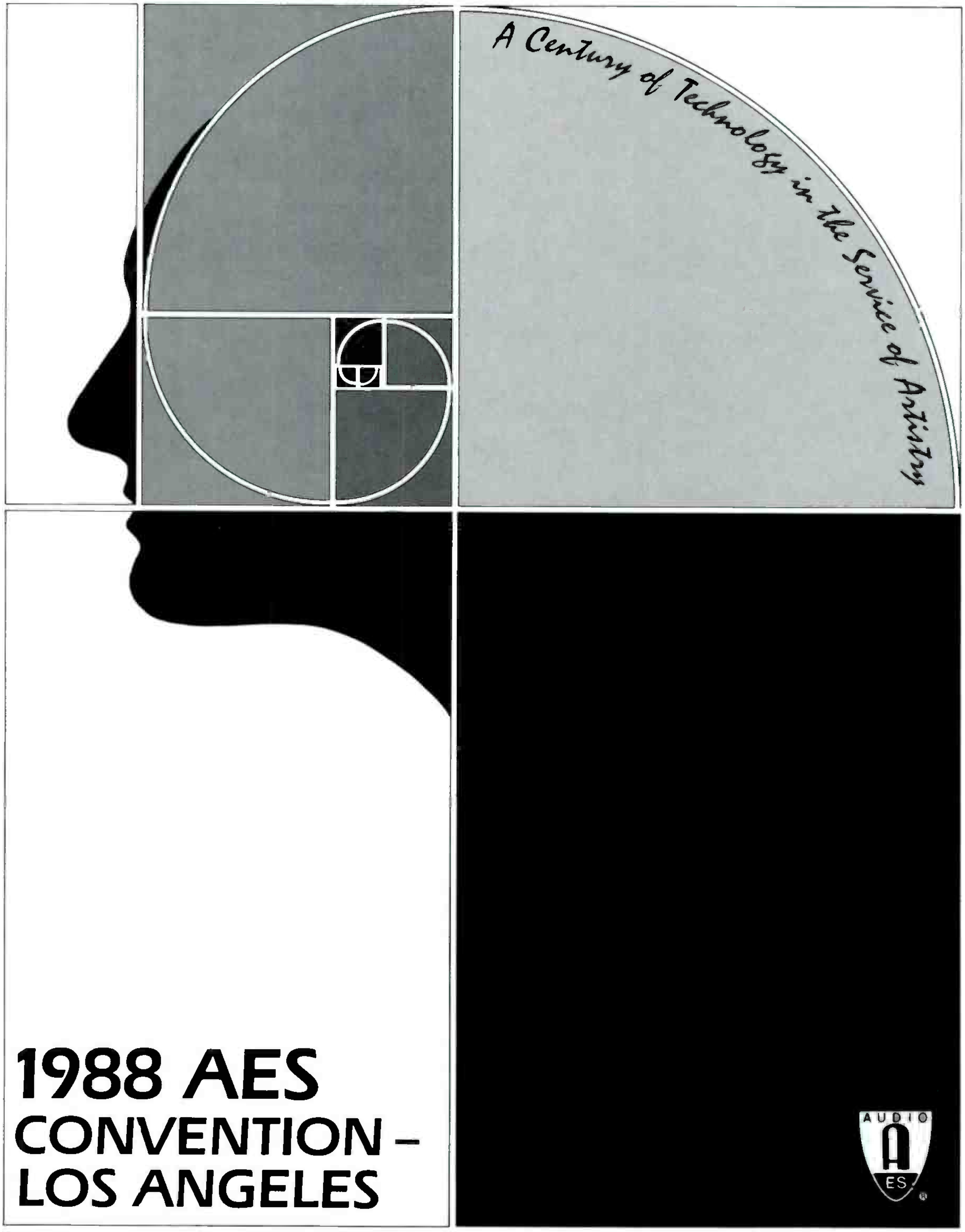
In addition to the standard equipment found in each room, the Village has other equipment available for an extra charge. "We have two Mitsubishi 3324s and an X800 digital machine, a 1610 BVU and X80 for 2-track work and DAT. We also have quite a few MIDI keyboards, drum machines and the like as well as a Fairlight on hand, although clients and players usually bring their own equipment."

Speaking of clients, the Village boasts some of the biggest in the business. The list of recording artists who have worked there is enormous and reflects a variety of musical styles that spans the spectrum from rock to pop to jazz to classical. Among the names that appear there are Phil Collins, Chick Corea, The Crusaders, Spencer Davis, Thomas Dolby, Placido Domingo, Bob Dylan, The Eagles, Keith Emerson, Eurythmics, Donald Fagen, Fleetwood Mac, Jan Hammer, Herbie Hancock, George Harrison, Bruce Hornsby, The Jacksons, Bob James, Al Jarreau, Keith Jarret, Elton John, Quincy Jones, Huey Lewis, Madonna, Manhattan Transfer, Joni Mitchell, Mr. Mister, Robert Palmer, Tom Petty, Pink Floyd, Pointer Sisters, Jean Luc Ponty, Lionel Richie, Lee Ritenour, Robbie Robertson, The Rolling Stones, Linda Ronstadt, Tom Scott, Wayne Shorter, Frank Sinatra, Bruce Springsteen, Steely Dan, Rod Stewart, Talking Heads and Frank Zappa to name just a few. Many of these artists have made gold and platinum records at the Village.

The list of movie and video soundtracks produced at the Village is just as extensive. The soundtracks for feature films such as All The Right Moves, Beetlejuice, The Buddy Holly Story, Dirty Dancing, Ferris Bueller's Day Off, Ghostbusters, The Karate Kid and The Lost Boys were recorded at the Village. Their HBO, Showtime, MTV and cable television credits include specials on Bette Midler, Fleetwood Mac, Sammy Hagar, Stevie Nicks, Richard Mulligan, Tom Petty and the US Festival. Other shows for television recorded at the Village include Faerie Tale Theatre, Jack Smith's You Asked For It, Ripley's Believe It Or Not and The Late Show.

CONTINUED GROWTH

Village Recorder continues to grow and flourish due in large part to the cooperative, accommodating environment that Smerigan and his staff have created there. They and their tenants have established a working village of professionals that interact and share ideas and business for the benefit of all. Clients can count on comprehensive service that addresses all of their needs in one convenient location. State-of-the-art equipment, comfortable surroundings and the expertise of the staff and tenants at all levels assures the Village continued success in the years to come. db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ROBYN GATELY

The Sound Contracting Engineer

LESS CAN BE MORE: THE 3:1 RULE

• As a sound company becomes larger, more and more time must be devoted to the management of the company, as opposed to the performance of the gig. Pursuing new clients requires **time**, a commodity that is in short supply if you are constantly working.

The factors determining which sound company will be hired for a tour or concert are usually: price, equipment, personnel, and quality of monitors. Of these, monitors usually determine if the band will want to use you again.

A quick survey of the top 20 sound companies would probably show that over half make their own monitors. Why bother, when so many manufacturers offer products designed for this purpose? A possible answer might be that most manufactured monitors are designed for what the public asks for, as opposed to what might make for an ideal monitor.

That might be considered a pretty bold statement, especially considering one of the companies I own manufactures monitors. However, from a sound company's point of view, there are many reasons that make building their own monitors the ideal solution.

SMALL MARKET

Even though the esoteric home audio marketplace is small in comparison to the entire consumer base, to a manufacturer of consumer products there is still a fairly large marketing base. However, the esoteric live sound market is small, *very small*. Even if the top 20 sound companies each own 50 monitors, that's still only 1,000 esoteric monitors needed by the entire marketplace.

Even if they are replaced every few years, the market is still only a few hundred every year. Add to this the fact that many companies make their own, and the need for super high quality monitors becomes so limited that it becomes almost worthless for a manufacturer to pursue the market, except for promotional purposes that carry over to their lower-priced, mass-marketed monitors.

Obviously, manufacturers are going to sell where the market is, and it is at the other end of the spectrum from esoteric monitors. Therefore, because the choices in super high quality monitors is so limited, and because your business may live or die by your monitors, it becomes very important to be sure that whatever you have is used correctly.

Due to space constraints, this article will devote itself mainly to the concept that you are not running multiple monitor mixes on stage, but rather only one or two mixes that everyone must share. After all, that is what 95 percent of the industry works with.

A few common mistakes can make a good monitor system work poorly, while some prior thought can often make a mediocre system perform very well. When I visit clubs, I usually see a bunch of monitors scattered around the stage, one in front of each microphone, with no prior thought to psycho-acoustics, phasing, or the 3:1 Rule. The reason most often given for doing this is, "That's the way it's always done," yet it is often the worst possible thing to do, especially on small stages. Let's look at some of the things you need to be aware of to get the performance you want out of your system.

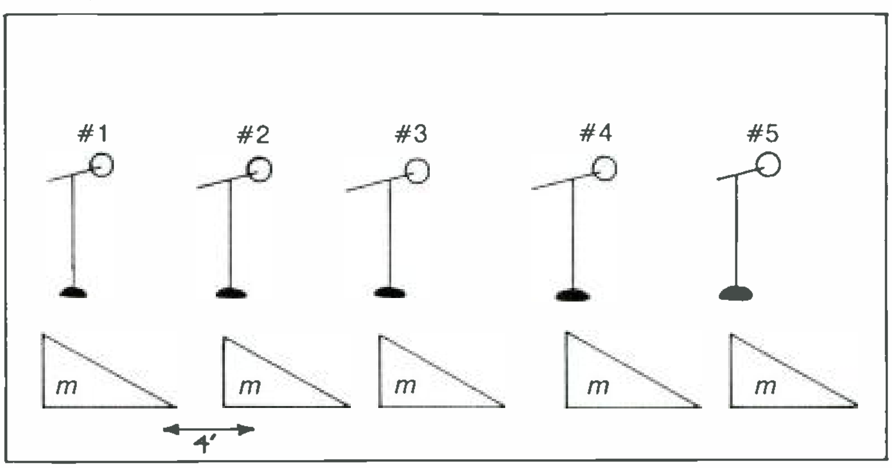
Every soundman should familiarize himself with both acoustics and psycho-acoustics. Acoustics is what happens in a room, psycho-acoustics is what your brain *thinks* is happening in the room. There can be a very large difference between the two.

For instance, you would think that if a monitor is placed to one side, only one ear hears it, so if it is placed directly in front of the mic, both ears would now hear it and therefore it would be more apparent. Although this is logical, in a noisy environment (like onstage) the brain searches for things that stand out as being different. A monitor placed slightly off to one side is going to be heard by one ear more, and therefore will be noticed by the brain more. The ideal position seems to be about 30 degrees off center.

UNUSED MICS

If you ever work with David Bromberg, he will be very specific about this point, but he will be adamant about the subject of phasing caused by unused microphones left on. If there is a mic being used a couple of feet from an on, but unused mic, the unused mic will help create an environment that is the

Figure 1. Front view of a 20-foot wide stage. The distance from mic 1 to monitor 1 is five feet. However, the distance from mic 1 to monitor 2 is less than eight feet. In addition, mics 2, 3, and 4 all have a total of three monitors within eight feet of each mic.



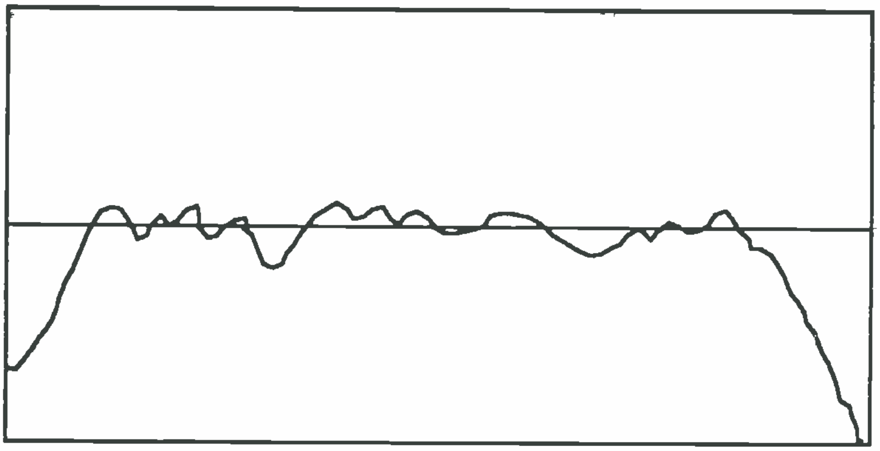


Figure 1A. Frequency response of mic 1 with only one monitor on.

equivalent of setting a digital delay on two milliseconds and sending that back through the monitors.

This results in a monitor system that sounds "washed out" and feeds back much quicker. The sound is lacking in mid-bass and has no clearly defined midrange. The only way to avoid this is by turning off all unused mics in the monitors. (Of course, this also applies to the "house" mix. The more you turn off mics, the cleaner your mix will be. Even turning down the unused mics several dB makes a big difference. So, if a background vocalist isn't used in a song, turn off that mic.)

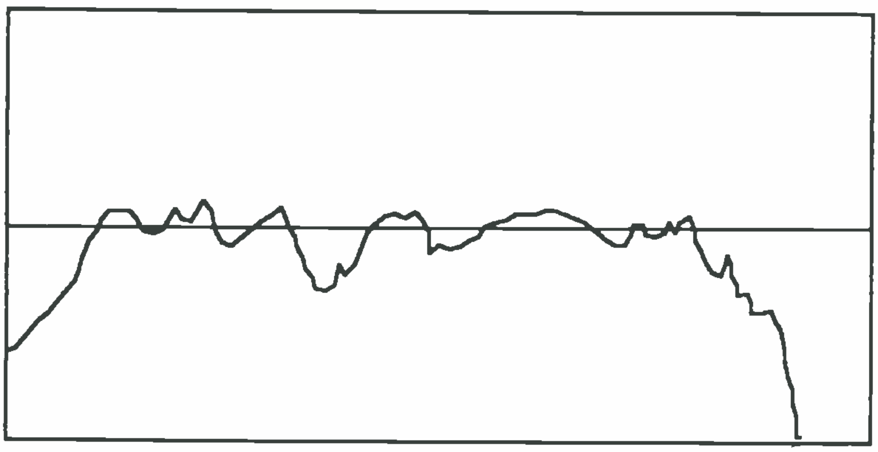
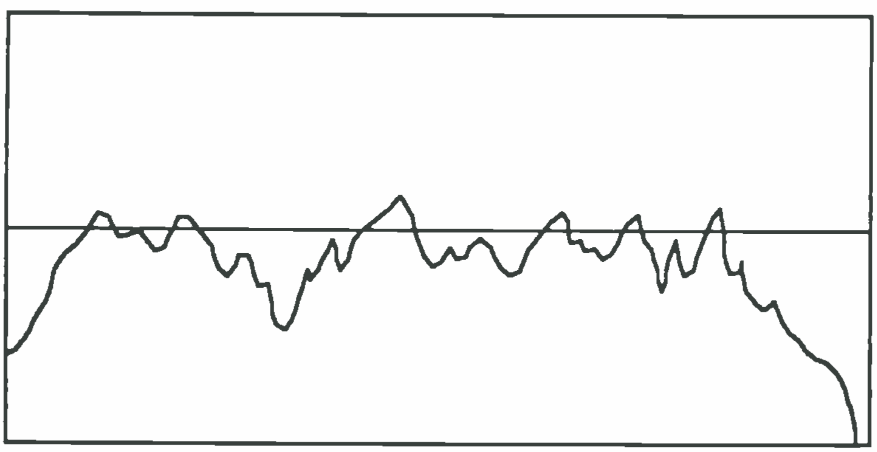


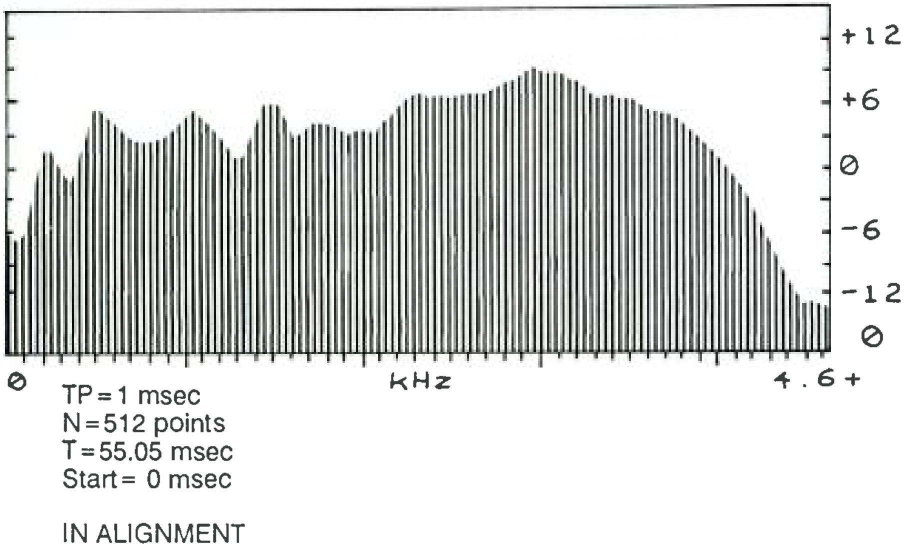
Figure 1B. Frequency response of mic 1 with all monitors on.

Figure 1C. Frequency response of the system with all mics and monitors on.



The concept of two sound sources washing out the mix also applies to speakers. In fact, many people refer to the "3:1 Rule." The concept is that if the monitor is located next to the microphone on the stage (about 5 feet away), the next closest monitor should be located a minimum of 15 feet away from the mic.

A good example of this is a club with a 20 foot wide stage, and a band with five vocalists across the front. The concept of putting a monitor in front of each microphone would result in a situation where each individual monitor is located 5 feet from the microphone, while the next closest monitor is only 7.5 feet away (*Figure 1*). Obviously, this is going to wash out the sound. However, this doesn't happen just once, but is multiplied five times across the stage. In fact, the three center mics each have two monitors within 7.5 feet washing out the sound (*Figure 1A, 1B* and *1C*).



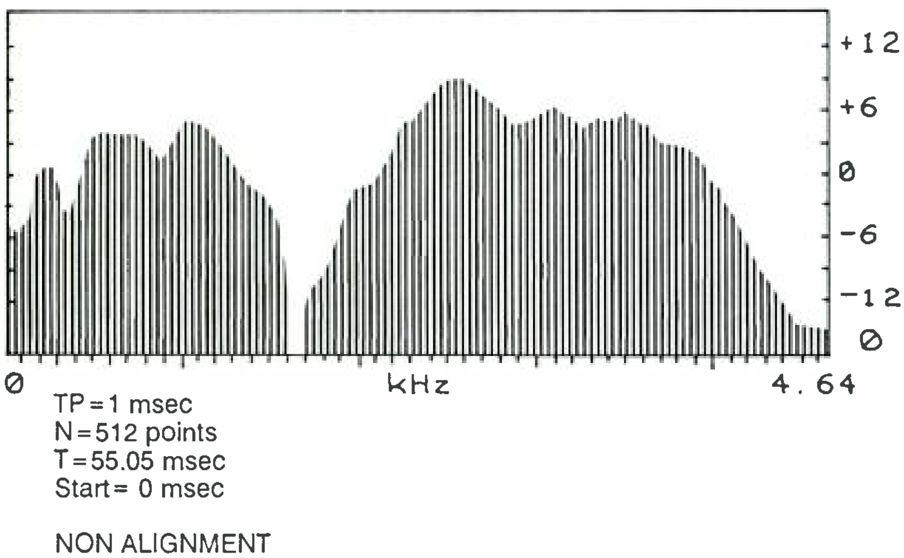


Figure 2. These two graphs demonstrate the advantages of driver alignment. They show the crossover region where the high and low frequency drivers interact. The graph above is the response curve for the Sub-Atomic Pile speaker. Below, the graph is of the same system with the high frequency system slid back into the box five inches. A small change in driver placement can make a profound difference.

Figure 3. Two equalizer settings of aligned monitors.Because of the smoothness of an aligned system, the top eq is set to remove the peaks common to the Shure SM-57 while the bottom eq is set for a SM-58. Notice that the prescence peak had to be removed from the SM-58.

In this situation, two strong sidefill monitors would provide more mics with more volume than five stage monitors. The center mic might benefit from a spot monitor (preferably on its own mix with only that mic in the mix), but the greatest level to everyone is from less, not more, monitors.

PSYCHO-ACOUSTICS

Taking this logic further, you begin to see how two high quality monitors can be more advantageous than five inexpensive ones. If you remember last month's article on artist accommodation, some musicians don't feel comfortable without seeing a monitor in front of them, and feel naked with only a sidefill system. They may complain that the system is not as good as individual monitors. This is part of psycho-acoustics, the musician needing to feel comfortable. Sometimes psycho-acoustics can drive you psycho. (Please realize that I am not disapproving of spot monitors, just don't put a bunch of mics and monitors close together on one mix.)

One of the advantages high-quality monitors often have is that the components (woofers and tweeters) have been aligned. Basically, this means that the system has been set up so that the sound from the woofer reaches your ear at the same time as the sound from the tweeter. There are several ways to accomplish this (and this one subject could fill an article), but the importance is how it affects the sound at the crossover point, especially since the crossover point is usually in the middle of the vocal region.

An aligned system will add the woofer and tweeter together evenly, creating a smooth transition from the lows to the highs. However, if the high end horn is as little as 0.5 millisecond behind the woofer (6 inches), this will create a dip in the frequency response in the crossover region of 15 dB or more! This causes the midrange to essentially disappear for a 0.5 octave or so. Considering there are approximately 6 octaves from the lowest bass singer to the highest soprano, a 0.5 octave disappearance is the equivalent of throwing out 10 percent of the vocal range.

WHEN NOT TO EQ

Some people might assume that this can be equalized back in. **DON'T EVEN TRY IT, YOU WILL BE PLAGUED BY FEEDBACK!** In the live sound field, too many people try to fix acoustic problems by inserting another electronic device into the chain. Learn how to fix acoustic problems acoustically, instead of trying to "patch it" and you will have a much easier and better sounding system.

Notice that I have spent a long time describing how to use monitors, without ever mentioning equalizers. This is because most equalizers end up being overused fixing many of the previous problems, or the system becomes "over-equalized" and the equalizer creates as many problems as it solves. Many soundmen comment about how great a Modular P.A. sounds with the equalizer left "flat." In fact, 95 percent of our shows never use the equalizer at all. This is *not* to say that I don't believe in equalizers, but that most equalizers are misused or overused.

An equalizer is supposed to help you correct for deficiencies in the system. Properly used, an equalizer can improve the gain before feedback by as much as 6 dB.

The usual way to use an equalizer is to make the system feedback, then reduce the frequencies that are too loud, then turn the system up again and repeat the process. This works to a certain extent, but can get you into serious trouble. Although the equalizer is being used to remove unwanted frequencies, once you've pulled down the majority of faders, all you are doing with the equalizer is turning the system down.

A good general rule is: once you've used one-third of the faders on the equalizer, STOP! If an equalizer has ten bands (octaves), use three or four of them; if it has twenty-seven bands (1/3 octaves), don't use more than nine of the faders. Although rules are made to be broken, rarely break this one. If you need to use more than 1/3 of the faders on your equalizer, then you probably have other problems that need to be fixed.

Many soundmen choose their mics according to the sound they want to achieve in their "house" mix. This might make sense from that point of view, but from a monitor point of view it is severely flawed. A built-in peak in a microphone must be removed to get greater output from the monitor system. Choose your microphones carefully, or you'll use all of your equalization getting rid of the "sound" of the microphone.

A good example of this is the Shure SM-58. The SM-58 has a built-in prescence peak that the SM-57 doesn't have. Assuming your monitor system is flat, in order to get the loudest possible monitors you would need to remove the 58s prescence peak with your equalizer. This isn't to say don't use 58s, but rather give some prior thought to how you utilize your mic selection (*Figure 3*).

As you begin to experiment more and more with these concepts, you will begin to understand why many top soundmen truly believe that less can be more. More importantly, you will begin to experiment and think for yourself.

Sound reinforcement is a very young industry, with a great deal still undiscovered. Remember, we're talking about sound, something that isn't seen. It's not like the lights where either it's on or off, green or blue. Whenever you are told something can't be done, think about the problem and decide for yourself. After all, the superstar soundmen of tomorrow are the kids in garages experimenting today. db

ANDREW T. MARTIN

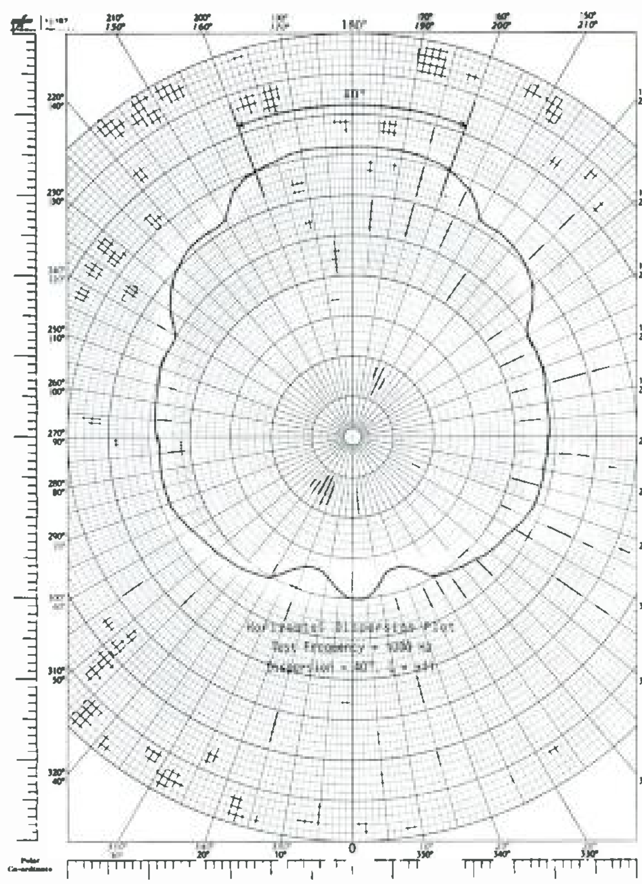
Sound Source Dispersion and Directivity Factor

Some of the most commonly overlooked aspects of sound reinforcement are also the most basic, and, as with the majority of subjects, the most basic are usually of the highest importance. An excellent case in point is the significance of sound source dispersion and directivity factor.

If a sound system is to be designed properly, these two concepts must be known and understood by the sound system designer. Yet, in the past, many designers have not been well versed with the effects that dispersion and directivity factor have on all sound systems. They have been unaware that these two factors can be utilized to extend the overall system's efficiency, increase intelligibility, and diminish unwanted room reflections. They have been unaware that misuse of sound source dispersion and directivity factor will in almost all instances cause a sizeable reduction in the performance of the sound system. It is for this reason this paper is dedicated to introducing the sound system designer to these concepts, and try to communicate the magnitude of these concepts clearly and without a tremendous amount of technical terminology. The following fictional narrative has been constructed to be helpful in understanding the importance of sound source dispersion and directivity factor.

Super Neeto Sound Company was in the process of installing a sound reinforcement system in a medium-sized church with a balcony. They had decided to try to uniformly cover the entire audience with sound from a central loudspeaker cluster mounted from the ceiling of the room directly over the minister's podium. Since the system was meant only to reproduce speech, a frequency response of 250 Hz to 5000 Hz would be totally acceptable. With this in mind, the company carefully chose the proper high frequency compression drivers and low frequency loudspeaker enclosures that would adequately generate the desired frequency response; and then picked out the horns that the compression drivers would be mounted on before being installed in the suspended loudspeaker cluster. Super Neeto Sound had decided to install three of the same 90 degree horizontal dispersion by 60 degree vertical dispersion horns with identical compression drivers on each. The directivity factor of the horns was 6. They mounted the horns in a contiguous vertical array with each horn mouth aligned precisely flush with the next, and with the high frequency compression drivers in synchronization. To complement the horns they decided to install two 15-inch vent loaded woofer cabinets which would be mounted directly underneath the horns. The suspended loudspeaker cluster was then constructed and hoisted into place: the top horn was aimed at the center of the balcony seating section,

Figure 1.



Andrew T. Martin is President of ATM Audio/Sound reinforcement Inc. in Rancho Dominguez, CA.

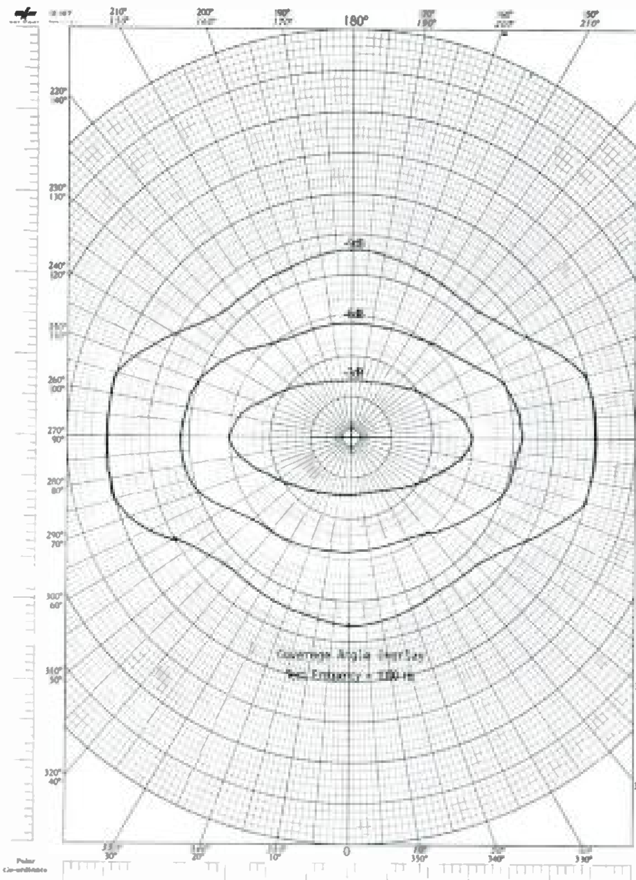


Figure 2.

and the bottom two horns and the woofers were aimed directly at the center of the floor seating section.

After the sound system had been connected with the power amplification and the various signal processing and program mixing equipment, the designer began what is always the final performance test: he began to listen to the system he had worked so hard to manufacture. As he walked around the church, he noticed that in many locations he could not understand what the speaker on the intelligibility test tapes was saying. He also noticed that the sound pressure level in the front and at the center of the seating areas were drastically higher than at the rear and the edges of the audience seating areas. The designer also found that the fidelity of the sound system was appreciably worse in the balcony than on the floor level. In desperation, the designer began to modify the equalization he had previously adjusted so precisely in the hopes of improving the fidelity of the sound system, but it was to no avail, the sound system could not be corrected with equalization.

Unbeknownst to Super Neeto Sound Company's designer was the significance of sound source dispersion and directivity factor. It may be helpful at this time to explain these two concepts in further depth.

Sound source dispersion, also known as coverage angle, is easily understood by viewing either polar charts or 3-D plots of the acoustic output of the sound source at a specified frequency. Usually the 6 dB down points are used to designate the dispersion angles in degrees, as shown in *Figure 1*. This measurement is made for both the horizontal axis and the vertical axis. With this information the designer is able to choose the sound source which will cover the audience maximally, but reduce the reflections of the sound source energy in the room to a minimum; thereby increasing the intelligibility of the program significantly. To further aid the designer, many manufacturers publish coverage angle overlays which outline the dispersion characteristics of the sound source as they would affect the audience area, as seen in *Figure 2*. There are now many price effective computer software programs which will simulate the dispersion of a sound source in a computer-generated replica of the environment the loudspeaker is to be installed in.

Directivity factor, also known as Q, is the ratio of the sound pressure squared, at a specific distance and fixed direction, to the mean squared sound pressure level at the same distance; then averaged over all directions from the sound source. Therefore, the directivity factor of a sound source is not an average measurement, but rather an average of all the individual Q measurements. Hence, an omnidirectional sound source would have a Q = 1, and a hemispherical sound source would have a Q = 2, and so on. What this means in practicality is that a sound source with a Q = 1 would be half as directive as a sound source with a Q = 2, and a sound source with a Q = 4 would be twice as directive as the sound source with a Q = 2. As seen in *Figure 3*, it is apparent that an increasing Q is directly proportionate to increasing sound source power, as long as the input power to the sound source remains the same. This is due to the horn's ability to take the acoustic energy of the sound source and tighten the dispersive pattern, thereby creating a more directive and concentrated output. As a result, utilizing higher Q loudspeakers make more efficient sound systems if the higher Q sound source conforms with the dispersion requirements of the system. But use extreme caution when installing high Q devices, for if they are not installed properly the adversity of the sound system can be as extreme as the profit. Additionally, not all manufacturers include the directivity factor measurement (Q) in their specifications sheet, but most will supply the information if it is requested.

So, applying the two concepts of sound source dispersion and directivity factor to Super Neeto Sound Company's installation, it is clear that some modifications need to take place.

Firstly, the balcony seating section would require a horn with a 70 degree by 40 degree dispersion and a Q of about 9. This sound source would considerably cut down the unwanted room reflections, and also increase the efficiency of the balcony horn and compression driver combination. In addition to changing the horn, its center axis should be aimed toward the seats in the center-rear of the balcony. By doing this, the sound pressure will remain more constant for the entire balcony audience due to the inverse square law; see appendix 1-0.

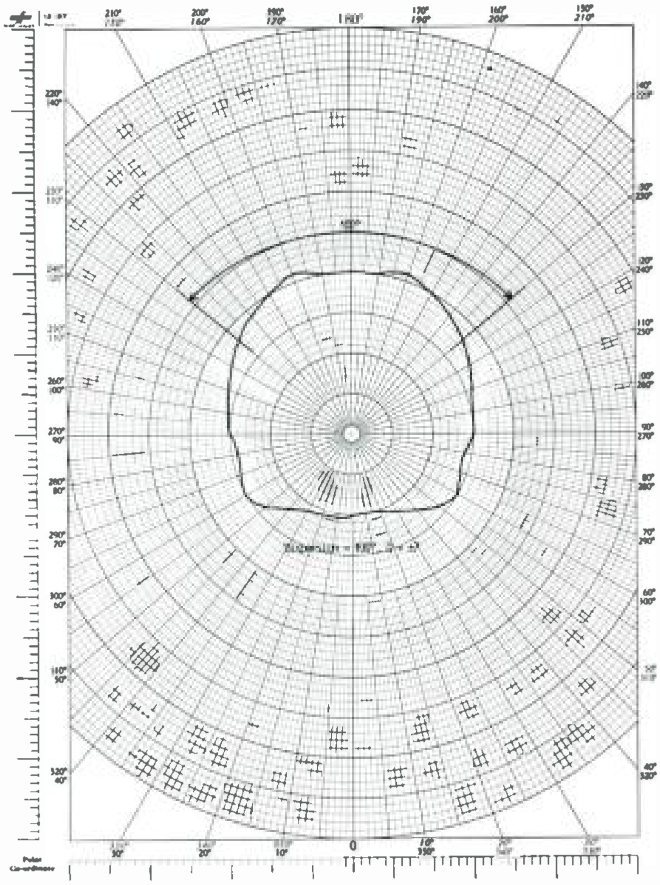


Figure 3A.

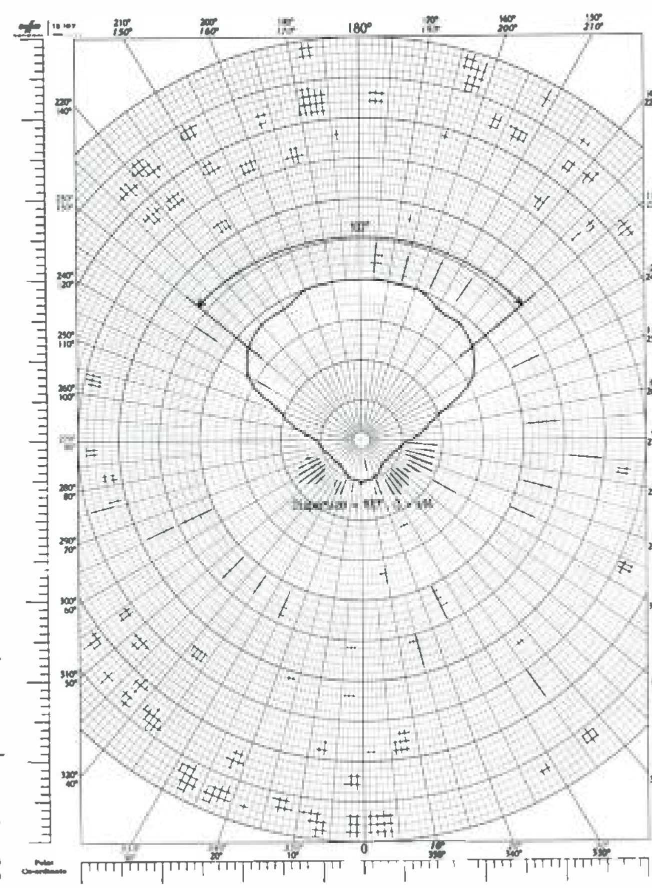


Figure 3B.

Secondly, the floor seating section would require only one of the existing 90 degree by 60 degree, Q=6, horn and compression driver combinations. This would reduce the unwanted room reflections tremendously while maintaining adequate sound pressure level in the room. Again, the center axis of the horn should be aimed toward the center-rear section of the audience, once more taking advantage of the inverse square law.

Lastly, the two 15-inch woofers should be aimed in the direction of the floor seating horn. It is not necessary to aim additional woofers at the balcony seating section because the dispersion of lower frequency devices is inherently very wide and will sufficiently cover both the balcony and floor seating sections; see appendix 1-1.

After reviewing the effects of sound source dispersion and directivity factor it is obvious that they are an imperative part of sound reinforcement system design. When these two concepts are applied correctly by the sound system designer the benefits are immense. For the sound contractor the cost of the sound reinforcement system will be reduced thereby giving the contractor a larger profit margin and a lower bid. For the client, an installation that meets or exceeds all performance specifications is achieved at a reduced cost which will result in satisfaction and future referrals to the contractor. Sound source dispersion and directivity factor are not to be taken lightly.

APPENDIX 1-0

Inverse square law rate of level change.

This law describes the geometric expansion of sound from a sound source. The change in level for a spherical expansion from a point sound source is approximately 6 dB for each doubling of the distance. However, the reverberant field indoors is relatively constant, and therefore must be taken into consideration.

loss in dB-SPL at the measurement point,

where: $$r = 10\log\left[\frac{Q}{4\pi r^2} + \frac{4}{R}\right]$$

r is the distance to the measurement point

Q is the directivity factor of the sound source
R is the room constant

Example:

$$\text{loss } (4') = 10 \log \left[\frac{6}{4\pi(4)^2} \quad \frac{4}{1000} \right] = -14.71 \text{ dB}$$

$$\text{loss } (40') = 10 \log \left[\frac{6}{4\pi(40)^2} \quad \frac{4}{1000} \right] = -23.67 \text{ dB}$$

The difference is 23.67 - 14.71 = 8.96 dB-SPL

Outdoors, the formula for the inverse square law is as follows:

loss in dB-SPL at the measurement point, $r = 20 \log \frac{r}{r_1}$

where:
r is the distance to the measurement point
r_1 is the measured reference distance

APPENDIX 1-1
Low frequency dispersion control.

At lower frequencies sound is dispersed in a very wide pattern. This is because the cone of the transducer is small in comparison to the wavelength of the low frequencies being reproduced. The approximate wavelength of a frequency in feet can be achieved by using this simple formula:

$$\text{wavelength} = \frac{1130'}{\text{frequency}}$$

example:

$$\text{wavelength of } 250 \text{ Hz} = \frac{1130'}{250 \text{ Hz}} = 4.52'$$

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WAYNE MITZEN

Stereo Production Techniques

Unlike other articles written on the subject, this one will not confine itself to different stereo microphone placement techniques.

The most natural sounding stereo effect is, of course, the use of a well-placed pair of microphones of similar model. Many of the manufacturers of high-quality condenser mics have models with two (or more) elements that are designed specifically to capture a stereo image of an instrument or voice in a well-balanced, acoustically-pleasing environment. These microphones can usually be adjusted for various patterns to be selected so that the engineer can use any of the many well-researched techniques developed over the years. The outputs of the two microphones are then recorded onto two tracks of the multi-track tape. Many fine publications are available on the types of two-mic techniques currently in use. The depth of this subject precludes its detailing, and if you are interested in this subject, a trip to the local library can provide useful information.

The above pertains to acoustic instruments or amplified instruments using the character of the amplification system for their overall sound. This is due to the fact that all types of microphone placement techniques tend to characterize the signal with the interaction of the room acoustics and the instrument's (or amplifier's) acoustical properties. If the original instrument is not acoustic, or the remaining amount of tracks do not dictate the use of two tracks for every instrument or vocal, another means of reproducing a stereo image is required.

PANNING

Simple *panning* of instruments or vocals is one way in which to separate each track in the stereo mix. But panning alone can result in some unnatural side effects in the final product. Panning is the procedure in which a mono instrument or vocal track is moved left to right by the use of a stereo balance control. If you watch the console's meters for the stereo output bus, you can see that the result of moving an instrument to one side causes the channel for that side to register higher than the meter on the opposite channel.

What occurs then is the concentration of uneven spectral energies in the left and right channels. The reason that this effect is unnatural is that in nature, both ears receive very close to the same RMS or average volume of a naturally occurring aural phenomena. At large distances, the ears sense almost the same volume of sound at either side of the head. The way in which the brain localizes a sound source is through the time, frequency, and amplitude differences in which the left and right ears receive the information regarding the sound source. This includes the direct sound that travels from the source directly to the listener and the reflected sound waves from the surrounding environment. By deciphering this vast amount of information at an incredible rate, the senses tell the listener where the exact location of an object is. The size, shape, texture and other physical attributes of a sound source can be detected as long as the enclosure (or lack of one) does not adversely affect the received composite sound wave. Along with detailing a source's placement, this also relays to the brain the type of environment that the listener is in, whether it be absorptive, reflective, or any combination of the properties involved in determining an enclosure's character.

TWO-CHANNEL SIMULATION

Realizing that the sensed placement of a sound source is due to time as well as amplitude, recent developments in digital delays have allowed audio engineers a way in which to simulate in two channels, what actually occurs inside of the head due to head-related directional transfer functions. With the advent of digital delays that are capable of passing full-bandwidth signals, a means of actually imaging a single channel of information in a stereo field is available. The delay used must be as frequency linear as possible because binaural perception can be corrupted due to anomalies in the response of a delay in the critical mid-band spectrum of human hearing.

This artificially derived effect is accomplished by performing separate, time-based adjustments to the left and right channel information along with amplitude placement (pan-potting) of the direct or dry signal to one side and then re-directing the returned delay signal or signals to the center and opposite sides of the stereo panorama.

AMOUNT OF DELAY

The amount of delay will depend on the distance and size desired. It usually will be an amount between .1 and 50 milliseconds (a millisecond is equal to one one-thousandth of a second). A very tight image will result, one without the ear-popping side effects that occur with simple panning alone. Multiple-tapped delays can offer even more versatility by presenting many mirror-images of a signal, each offset slightly different in time to each other, and each presenting a very tight image when used together with one another ans each being panned separately.

In order to configure a console during mixdown to be able to take advantage of this technique, one must have enough pannable effect returns for each desired delay return signal or else enough extra input modules to accommodate the returning delay signals. You will also need as many delays as you need *separate* images. Since many instruments will either be centered in mono or will be part of a group of tracks that are to each be placed separately (as in natural or artifi-

TABLE OF STEPS TO SCALE DEGREES

STEPS	UP	DOWN
1	FLAT II	VII
2	II	FLAT VII
3	MIN III	VI
4	III	MIN VI
5	IV	V
6	AUG IV	DIM V
7	V	IV
8	MIN VI	III
9	VI	MIN III
10	FLAT VII	II
11	VII	FLAT II
12	OCTAVE	OCTAVE

Figure 1. Pitch conversion from steps to degrees.

cial doubling – see below) only a certain amount of tracks will require different delays. The input to the delay is then connected to either one of the auxiliary send outputs from the console. But if the delay is imaging only one track, it seems a waste of a mixing bus to use an auxiliary send because there is no mixing of two or more tracks to go to the input of the delay. Since many of the console manufacturers wire a direct send or else make available a post fader insert point, I would suggest that the input for the delay be taken here. It is very important to point out that whichever send is used that it be post-fader so that once the balance between the left and right image-generating signal is set, the overall volume can be adjusted by moving only one fader and not result in the image's left vs. right balance being disturbed. Also, if you are using an insert point's send, see if the console maker designed the console to break the signal path on return insertion only. This perverted-sounding request is to make sure that putting a patch cord in the insert point send will *not* break the normal signal path in the channel's signal flow to the two-mix bus. If it does, *don't* make the mistake of returning the delay's output to the channel insert "return" jack. If you do, you won't be able to pan the delay signal separately to the opposite output channel. Just use a "Y" cord to send back into the channel and allow you to tap a feed, post-fader, to send to the delay unit's input.

CONTROLLING THE IMAGE

After connecting the delay's input and output, each track to be imaged will be handled separately, so mute everything except the channel that is to be imaged. If you want the image to appear to be coming from the left, pan the original, dry signal to the hard left side. Now, look at the meters on the two-mix bus. Set the left channel (since it is the only one registering anything because you should only be sending the particular track that you want to image down the two-mix bus at this time) for a convenient reference point. In other words, set it so as to just show something that will deflect the meter to near its mid scale (–3 to 0 dB) on the left channel. This does not have to be the level that will appear in the final mix, but it instead is to reference the proportional amount of the left undelayed channel that is then compared to the delayed right channel information. Set the delay so that the delay's front panel "MIX" is all the way to the "wet" or "delay only" setting and the regeneration control is all the way off. Adjust the delay to about 25 milliseconds and place the echo return or extra input module's pan pot to the right side. Then adjust the console's echo return volume or the extra input module's fader to match the right channel volume (the delayed signal) to that of the left channel's (the un-delayed or dry signal). If you are listening in stereo, you will notice that the image appears to be fairly close to the left side. The size and apparent angle that the image is coming from can be controlled somewhat crudely by adjusting the pan pots in toward the center slightly. Another way in which to adjust the size is to shorten or lengthen the delay time. If you are using a multi-tapped delay, you should have a returning channel (with a separate pan pot) for each tap used. This will allow you to position each image separately.

A delay of over 50 milliseconds applied to one channel will result in what is known as an artificial doubling effect. This is also the misunderstood term known as artificial double tracking (ADT). ADT was developed by engineers to electronically achieve the same effect as recording two separate performances of the same part of an arrangement. Doubling by either means enhances the stereo image by spreading the energy across the stereo field. Due to the fact that no matter how seemingly exact the performer "doubles" (plays along) with the original performance, certain subtle timing and pitch variations will occur. By panning the two performances to either side of a stereo field, the two distinct images aid in reducing the amount of energy concentrating at the center of the two speakers. The effect results in the energy of the doubled instrument being spread across or to either side of the monaurally-mixed instruments.

LONG DELAYS

Since a delay time of over 50 milliseconds is much longer than would normally occur due to head-related directional transfer functions, a doubling will occur whenever the one channel is delayed longer than 50-100 milliseconds. The configuration for setting up to do this is basically the same, with the dry signal sent to one channel (either the left or right channel of the two-mix bus in mixdown) and the longer *non-regenerated delay* signal being sent to the opposite stereo channel of the two-mix bus. Again the signal is derived from either an auxiliary send or else a direct post fader output from the channel you wish to double. A lot of people think that by just inserting the delay into the channel insert point and setting the delay's front panel control to "double" while mixing the dry vs. delay to 50 percent, that this is doubling. All it actually is then is a slap echo of the original signal. The only real way to accomplish an artificial double is to send the results to separate left and right channel positions. This will spread the acoustic energy across the two speakers slightly out of time so as to (as is with a natural double) relieve some of the concentration of energy in the mono field between the two speakers.

VARYING THE SIGNAL

In addition to distributing the doubled image through the use of time-based differentiation of the artificially generated double, the pitch of the ADT signal can be varied to distinguish it even more from the original track. A common pitch-ratio and delay setting used over the years by engineers with the Eventide series of Harmonizers has been loaded into one of the ROM programs of the Yamaha SPX-90 (a ROM program is a program that is used as a starting point to access the various programs and parameters available with this particular unit). By selecting program number 23, the user is immediately able to generate two pitch and time shifted doubles that are output separately to the left and right output jacks of the SPX-90. It does not require that any parameters be changed with the front panel controls in order to realize this effect, but full facilities are available for each to be adjusted over a wide range of time and frequency based parameters. The values that it defaults to (i.e. the values that it automatically assumes whenever program 23 is recalled) is a very short delay on each channel (.1 milliseconds) and a positive 8 cent pitch change on the left channel in conjunction with a negative 8 cent pitch change on the right channel (one cent equals one one-hundredth of a semi-tone or half-step). The SPX-90 allows the user to move from this subtle a double to a full-fledged harmonization (over a ±1 octave range in twelve steps) with individual delays for each channel adjustable from 0.1 milliseconds to just under 1 second. (A table listing the pitch conversion from steps to degrees of the scale is *Figure 1*.)

While trying any type of stereo recording technique, from live mic'ing to ADT, the mono compatibility of the signal should be evaluated. This is accomplished by using the mono switch located on the control room monitor module. Since anytime a signal is being reproduced by more than one channel (especially after it has been time altered), a certain cancellation will occur at a set of given frequencies related harmonically to the signal and to the amount of difference in time that lies between the two channels while being recombined into a mono summation. By adjusting the delay times to compensate for a given instrument's spectral distribution and timbre, mono compatibility can be assured. Also, a well-equipped studio should have an oscilloscope set up for phase analysis (X-Y mode) or a phase correlation meter connected to either the two-mix bus or the control room line out (so as to allow the solo function access to the phase metering separately from the stereo mix bus).

Another important factor is the monitor system and the acoustics of the control room. In order to be able to distinguish either the perceived placement of an artificially generated stereo instrument/voice or the effectiveness of an artificial or natural double, a control room must not cause interfering reflections to occur at the listening position between the speakers. A well-designed monitor system that is properly interfaced with an acoustically well-controlled listening environment is preferred to accurately represent what is actually happening in the stereo field with each image. db

RANDY HOFFNER

SETTING AUDIO LEVELS FOR INTERNATIONAL PROGRAM EXCHANGE

• In this age of electronic sophistication, it may seem that the setting of audio levels would be a trivial pursuit. Indeed, we have been at this business of sound broadcasting long enough that such a basic operation as level setting should be the least of our worries.

When we enter the word of international audio program exchange, however, we encounter a confusing potpourri of reference levels, terminologies, and metering devices. This confusion often leads to misunderstandings between sending and receiving parties, and thereby, to audio level discrepancies.

THE UBIQUITOUS VU METER

The vu meter has been the standard audio level indicator in this country since 1939, while the peak-program meter (ppm) has occupied that niche in most of Europe. The vu meter's ballistics and scale are well-standardized, but peak-program meters are characterized by several different sets of ballistic specifications, and numerous different scales. The European Broadcasting Union, in EBU-3205, has specified a standard peak-program meter for international program exchange. *Figure 1* contains a schematic representation of various types of ppm scales used in Europe. The IEC Type 1 scale is philosophically similar to the vu meter, being laid out in a logarithmic topology with its operating point labelled "0 dB."

The three test levels are called alignment level (AL), measurement level (ML), and permitted maximum level (PML).

Type IIa, the BBC scale, illustrates another scale philosophy. Like many other ppm scales, it displays 24 dB in 4 dB divisions, and is laid out in decibel-linear topology rather than logarithmic. The scale is marked not in decibels, but with the arbitrary numbers 1 through 7. Type IIb is the EBU scale, which also displays 24 dB in 4 dB divisions. On this scale, however, the mid-scale point is marked "0" and "test," and the operating point is a mark at the point 9 dB above "test." To further muddy the water, some users operate this meter at the "+8" mark.

If this state of affairs does not create sufficient confusion, standard line-up level is variously referred to as "reference level," "normal level," or "nominal level," to name a few, and is expressed in such units as vu, dBm, volts, Nepers and others.

ATTEMPTS AT STANDARDIZATION

In an attempt to clear up the confusion associated with audio level setting and monitoring for international program exchange, the CCIR, in its Recommendation 645 (1986), recommends standard test signals for international sound program connections, and in Recommendation 661 (1986) describes a composite test signal program to be used for identification and alignment of such circuits.

Recommendation 645 describes three test signal levels. The name for each sine wave test level was carefully chosen to describe its purpose while remaining

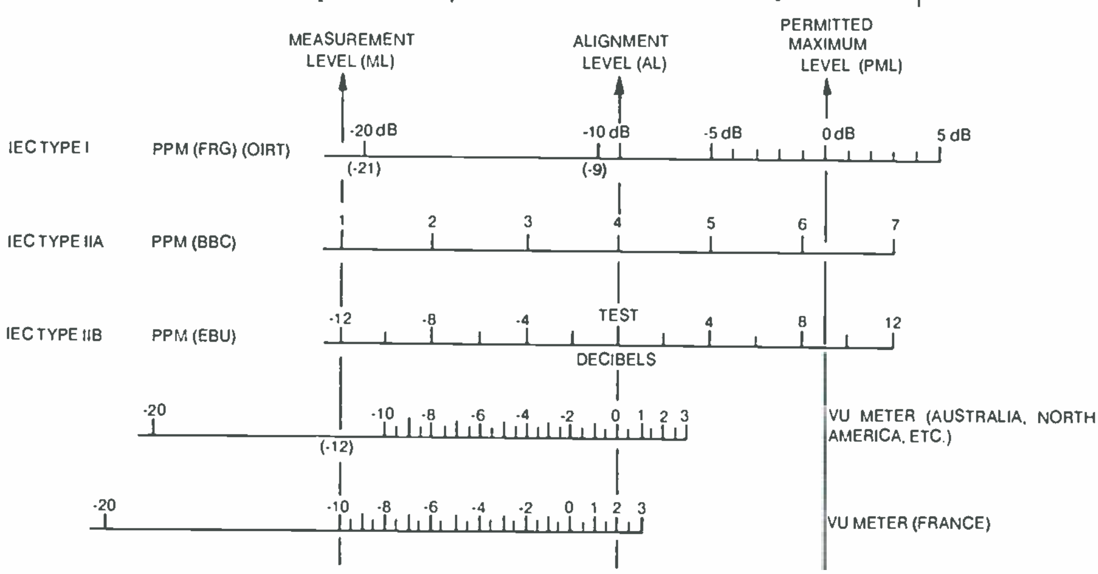


Figure 1. A schematic representation of various types of ppm scales.

free of ambiguity. The names were selected to avoid previous association with other names, and to be unambiguously translatable into various languages.

The three test levels are called alignment level (AL), measurement level (ML), and permitted maximum level (PML). The levels are expressed in terms of "dBmOs," which is the normalized test signal level. The normalizing factor is the "dBrs" value at the measurement point. This factor permits the absolute power level of a signal at any point in the transmission path to be determined by the equation: dBm (power level)- dBmOs + dBrs. For example, 0 dBm or one milliwatt is the actual power level at the measuring point in some European sound circuits. In that case, 0 dBmOs corresponds to an actual power level of 0 dBm and the dBrs value is 0. In North America, the actual power at sound circuit measuring points is +8 dBm. The dBrs factor at these measurement points is +8, and the circuit is described as a +8 dBr system. To give an example of what may be encountered internationally, the United Kingdom uses 0 dBr systems, the Federal Republic of Germany, -3 dBr, and France +6 dBr. The letter "s" identifies a sound program circuit, rather than a telephony circuit. The expression of relative levels in dBmOs provides a means of avoiding complications and the mental arithmetic involved in translating between the various reference and power levels often encountered at opposite ends of an international audio program circuit.

Recommendation 645, "Test Signals to be Used on International Sound-Programme Connections," recommends that only the following test signals be used.

The alignment signal is a 1 kHz sine wave signal used to align the circuit. Its level is 0 dBmOs. The measurement signal is a sine wave signal 12 dB lower in level than the alignment signal or -12 dBmOs. It is to be used for frequency response measurements and for long-term testing, as it is recommended that the alignment signal's duration be kept as short as possible. The permitted maximum signal is a 1 kHz sine wave signal at a level of +9 dBmOs. This signal is to correspond to the permitted maximum program signal level. The program signal's peaks as read on a ppm should only rarely exceed the indication of the permitted maximum signal.

The spoken identification is followed by two seconds of silence, two seconds at measurement level, then eight seconds at alignment level.

Figure 1 illustrates the indication that each of the three test signals will produce on various audio level meters. The 0 dBmOs alignment signal will produce a "test" or mid-scale reading on the EBU peak-program meter, or any other peak-program meter with a scale of a 24 dB in decibel-linear format. It will also produce a reading of "0" on the vu meter used in North America and Australia, while in the +6 dBr French systems, a +2 vu reading will be indicated. Note that on the Type I ppm scale, alignment level does not produce an indication on a scale mark. Measurement level falls on a definite scale mark for the BBC and EBU ppm scales and the French vu meter scale, but does not on the North American vu meter or on the Type I ppm scale. Permitted maximum level corresponds to a scale mark for the Type I and the EBU ppm scales, but falls at an unmarked point on the BBC ppm, and on a vu meter, is so far off-scale that it will result in a pinned meter. It is seen that while an EBU ppm scale, for example, can take advantage of all three test signal levels, the vu meter as used in the United States may be precisely calibrated with only one of the recommended test levels, alignment level.

DEFINITIONS

The three test signals are defined as sine wave signals, the purpose of which is to provide calibration points in a given circuit. Permitted maximum level is a test signal, and not to be confused with the upper limit of program peaks. It is understood that the true peaks of the audio program signal will exceed the permitted maximum signal level, and that an overload margin must be provided above this level. The vu meter's average-reading ballistics produce an indication several dB below the program peaks. The peak-program meter, which would more correctly be called a quasi-peak program meter, also fails to indicate the true peak level of the audio program material, although it comes much closer to doing so than the vu meter. It is interesting and instructive to note that over the past fifty years peak-program meters and vu meters have been used to equal advantage to control audio levels. From a practical point of view, general agreement exists that the upper limit for actual program peaks is about 15 dBmOs in systems using both ppms and vu meters.

So the three-level test program is in fact currently a two-level test program.

THE CCIR RECOMMENDATION

CCIR recommendation 661 describes a composite test program incorporating the three test levels. The first section is a spoken announcement which identifies the source of the signal. The spoken identification is followed by two seconds of silence, two seconds at measurement level, then eight seconds at alignment level. After the alignment level signal, channel identification is provided by two seconds at permitted maximum level in the left channel and silence in the right channel, three seconds of silence, then two seconds at permitted maximum level in the right channel and silence in the left channel. The program repeats cyclically.

It should be noted that because all transmission systems are not at present capable of carrying sinusoidal signals at +9 dBmOs without producing excessive channel loading or crosstalk into other channels, the permitted maximum level portion of the test program is presently replaced in the recommendation with a signal at alignment level. So the three-level test program is in fact currently a two-level test program. Because vu meters are pinned by the PML signal and have no mark at which to set measurement level, only alignment level may be precisely used, reducing the test program to a single-level test program in this case.

Although the three-level test signal cannot be fully used everywhere and at all times, it provides a basis for clarification of some of the confusion involving level measurement for international audio program exchange. db

Recording Techniques

BRUCE BARTLETT

RECORDING LIVE TO 2-TRACK

•To achieve a commercial recorded sound, most bands require a multi-track recording, overdubs, and mix-down. But some musical groups can be recorded on a 2-track recorder in real time as the music is performed. You can record them using either multiple microphones and a mixer, or a stereo pair of microphones. The latter method works well with many groups: quartets, soloists, folk groups, orchestras, pipe organs, or symphonic bands.

PROS AND CONS

The advantages of recording live to 2-track are:

•You record only one generation. By omitting the multi-track recording, you eliminate its noise and distortion. The result is a cleaner recording.

•Recording is faster. When a satisfactory take is done, the recording is done, too. No overdubs or mixdown are needed.

•The musical performance can be more exciting. The musicians know that mistakes can't be fixed in the mix, so they play better. Also, they play as an ensemble—because there are no overdubs—and react emotionally to each other's presence.

•If you use a single stereo microphone pair, the recording can be more realistic and natural. There's no artificial reverberation and no close mic'ing to color the timbre.

The disadvantages are:

•You must make all the decisions about mic placement, balance, etc. at the session—a high-pressure situation.

•If one musician makes a mistake, you must record the entire ensemble again. You can't overdub just the flawed part.

•If the recording engineer makes a mistake in setting the balance among instruments, the recording must be done again. (If the mistake occurs near the end of the song, you might be able to start another take just before the error, and edit the two takes together.)

•If you record with a single stereo pair, you must control the ambience and balance by adjusting the room acoustics and the players' positions. This is more difficult than turning the appropriate knobs on the mixing console.

In this article we'll explore two types of live-to-2-track recording:

1. Recording true stereo with two mics into a 2-track tape deck.

2. Recording with multiple mics and a mixer into a 2-track tape deck.

EQUIPMENT

Good "true stereo" recordings can be made with simple equipment. You need:

•A quality stereo cassette deck or open-reel deck.

•Some higher-cost alternatives to the above are an R-DAT deck, a Beta Hi-Fi or VHS Hi-Fi VCR, or a digital audio adapter with a VCR. All these provide superior sound quality.

•Blank tape—Use the tape recommended by the recorder manufacturer. For open reel, use high-output low-noise tape. For cassettes, use high-bias (chrome or metal) tape, C-60 length or shorter.

•A stereo microphone costing at least $50.00, or two separate high-quality microphones.

•One or two microphone stands and booms, plus a stereo mic stand adapter (for two mics).

•High-quality headphones for monitoring.

THE TAPE DECK

Let's consider the requirements for a quality tape deck. Obtain the published specifications for the deck you want to buy or use, and look for the following:

•Noise Reduction. The Dolby and dbx systems are commonly used in cassette decks. Noise reduction is essential with cassette recorders because the slow tape speed and narrow track width result in audible tape noise.

Dolby C is more effective than Dolby B, and dbx is more effective than either. Still, Dolby is free of the "breathing" sound (modulation noise) that is sometimes heard on dbx'd tracks. Chances are you'll be equally satisfied with Dolby C or dbx.

•Wow & Flutter. Wow is a slow periodic variation in tape speed; flutter is a rapid variation. If excessive, they wobble the pitch of recorded instruments. The lower the wow & flutter spec, the steadier is the reproduced pitch.

Regarding the wow & flutter specification:

0.03 percent RMS weighted (or WRMS) is excellent.

0.04 percent RMS weighted (or WRMS) is very good.

0.1 percent IEC/ANSI peak weighted is very good.

Higher values than the above are not as good, and mean that you may hear the pitch wobble on recorded stringed instruments.

•Signal-to-noise Ratio. This is the ratio, expressed in dB, between the maximum undistorted recorded signal level and the noise level. The higher the figure, the more noise-free is the recording. All the following specs are measured with noise reduction:

90 dB is excellent (typical of dbx).

70 dB is very good (typical of Dolby C).

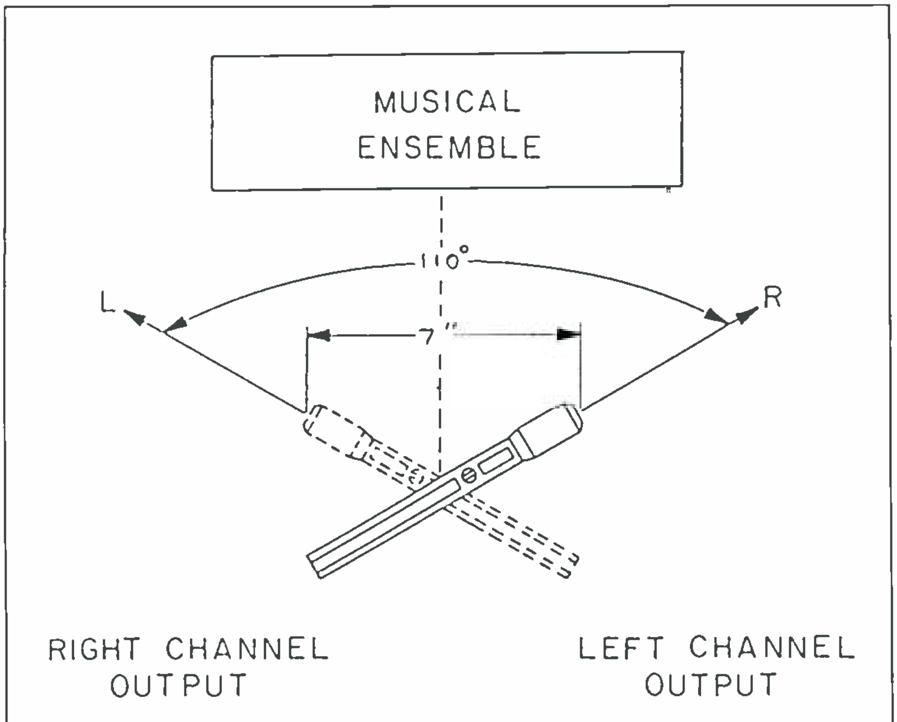


Figure 1. A method of mic'ing stereo.

65 dB is good.

55 dB is fair.

These specs are A-weighted, which means that the measurement was done in a way to correlate with the annoyance value of the noise. When comparing two different decks, be sure that both signal-to-noise specs are A-weighted.

•Record/Play Response. This is the range of frequencies that the recorder will record and play back at an equal level, within a tolerance (such as ± 3 dB). The lower the lower frequency, and the higher the upper frequency, the better the fidelity.

40Hz-12.5kHz ±3dB is fair.

40Hz-14kHz ±3dB is good.

40Hz-18kHz ±3dB is excellent.

MICROPHONES

Now let's consider microphones. If you're recording a singer/guitarist, a classical-music soloist, or a small acoustic group such as a vocal quartet or folk group, get a stereo microphone costing at least $50.00. An alternative is two identical microphones. If these microphones are the cardioid type (preferred), mount them on a stereo bar—a device that holds two mics on a single stand for stereo mic'ing. Angle them apart about 110 degrees (55 degrees either side of center) and space their grilles 7 inches apart horizontally, as in *Figure 1*.

If the two identical microphones are the omnidirectional type, place each one on a mic stand and space them 3 feet apart for a small group or 10 feet apart for a large symphonic ensemble.

For a singer who accompanies oneself on piano, get an omnidirectional microphone for the voice. Unlike a directional microphone, an omnidirectional unit does not get bassy when you place it close to the mouth. Chances are that an electret condenser type will sound best. Some models require a battery to operate, which lasts up to a year. If you already have another type of microphone, it's okay to use, but an omnidirectional condenser mic is recommended.

For grand piano, you might want to use a miniature condenser microphone.

For the piano, get one of the following microphones:

•A miniature omnidirectional condenser microphone

•A cardioid electret-condenser microphone

•A boundary microphone.

Or just use whatever you have if it sounds good to you.

PRE-RECORDING SETUP

Clean the recorder heads before each recording with a cotton swab moistened with isopropyl or denatured alcohol (from a drugstore). Do not use rubbing alcohol. Rub the swab on the head surfaces that contact the tape. Clean the rubber roller, too.

If you're using a cassette recorder, set the tape-type switch to the type of tape that you're using (it's specified on the cassette). This should be "high bias," "CrO^2 (chrome)," or "metal." Switch on the noise reduction, both during recording and playback.

If you plan to send your tape to someone else, find out whether their cassette machine has noise reduction, and what type it is. Set your machine to match. If you don't know what type they have, use Dolby B because it is the most common. Write on the cassette label what kind of noise reduction you used (dbx, Dolby B, or Dolby C).

CHOICE OF RECORDING ROOM

A folk or bluegrass group is best recorded in an acoustically "dead" room that is free of echoes. Such a room probably has carpeting, acoustic-tile ceiling, stuffed furniture, and drapes. If you need to deaden the room acoustics, hang some absorbers such as heavy blankets, sleeping bags, or comforters spaced out from the walls.

A classical-music soloist or ensemble sounds best when recorded in a "live" room that has noticeable reverberation, such as a church or recital hall. The room acoustics enhance the recording for this type of music.

MICROPHONE TECHNIQUES

Screw the microphone stand adapter onto a microphone stand. Place the microphone in its stand adapter.

When mic'ing a soloist or ensemble in stereo, put a stereo microphone (or a pair of identical microphones) close to the musicians, about two-to-five feet away, as in *Figure 2*. Place the mic about one-to-two feet away to pick up a singer playing an acoustic guitar. For a grand piano solo, raise the lid on the long stick.

To record a singer who accompanies oneself on piano, you'll need two separate microphones—one for the voice, one for the piano. You'll also need one or two booms. A boom is an adjustable pipe that mounts on a mic stand for positioning the mic. Put a foam pop filter or windscreen on the vocal micro-

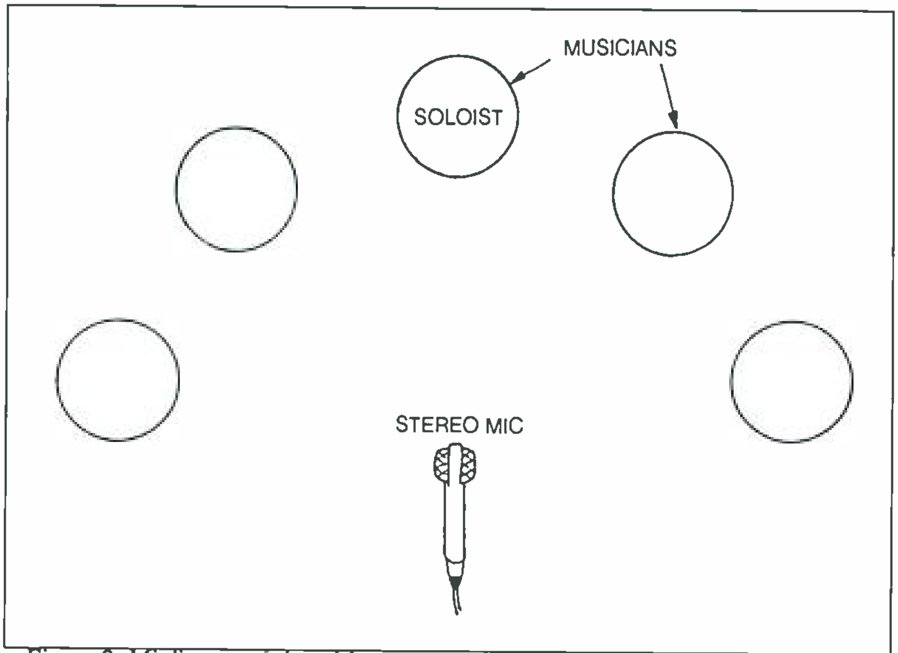


Figure 2. Mic'ing a soloist with a stereo microphone (top view)

phone to reduce breath pops. Place the mic about 1 to 3 inches from the mouth (as shown in *Figure 3*).

For grand piano, you might want to use a miniature condenser microphone. Tape it to the raised lid in the middle, as suggested by the manufacturer. Another useful microphone is a surface-mounted boundary microphone, a flat-plate unit. Alternatively, remove the lid and aim a cardioid electret-condenser mic down over the middle strings, at least 1 foot up, and about 1 foot horizontally from the hammers.

This method produces exaggerated stereo. You may prefer to mic the piano in stereo, and use a mixer to pan the vocal midway between your stereo speakers.

Tips for recording large acoustic ensembles have appeared in earlier issues of **db**.

RECORDING

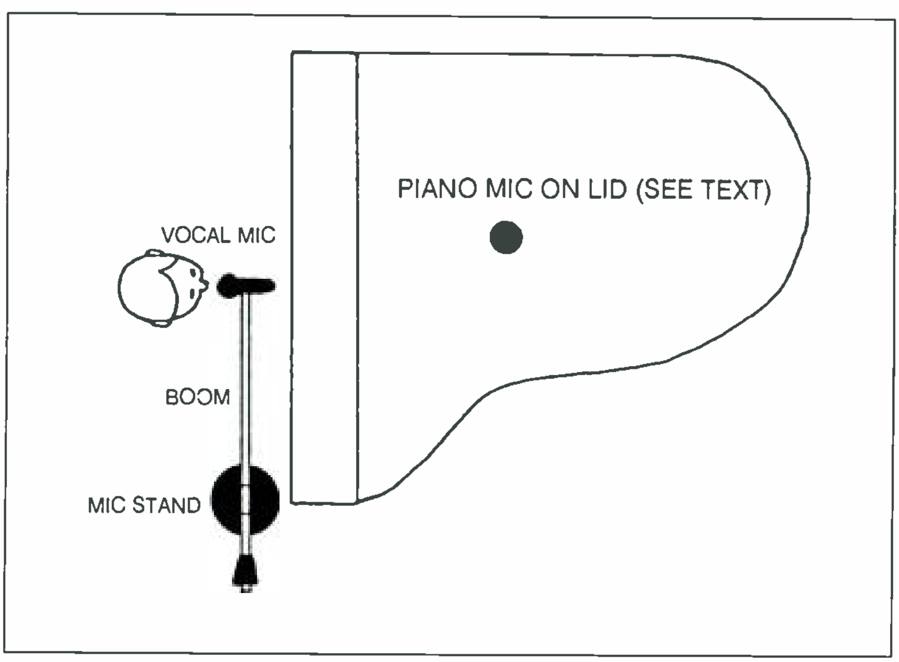
With the microphones carefully placed, you're ready to begin recording. First, plug the mic cables into the left and right mic inputs of your cassette deck. Press the record and pause buttons. Set the recording level to peak around 0 on the loudest part of the song.

Now you're ready to record and make adjustments. Press the pause button again to release it so that the deck starts recording. Don't make any noise before or after performing the song. After the performance is done, rewind the cassette and listen to it.

Suppose you've recorded a singing pianist. If the voice is too loud relative to the piano, turn down the volume control slightly for the vocal microphone and try again. Do the same for the piano-mic volume if the piano is drowning out the vocal.

Suppose you've recorded a small acoustic ensemble. If the sound is too distant or muddy, place the microphone(s) closer to the ensemble and try again. Or add more acoustic absorbers to the room. If any musician is too quiet relative to the others, have him or her move closer to the microphone and try again. If the balance still is poor or the recording has too much room acoustics, try mic'ing voices and instruments up close, and blending them with your mixer. This technique is described later.

Figure 3. One way to mic a singer and piano (top view).



If you want to make copies of your master tape, you can either copy from one deck to another, or use a dubbing cassette deck that holds two cassettes.

Now suppose you've used a stereo microphone to record a singer who plays guitar. Move the mic closer if the sound is too distant, and *vice versa*. Raise the mic on its stand if the voice is too quiet; lower it if the guitar is too quiet.

When you're satisfied with the balance and microphone distance, record other tunes. Leave about 2 to 4 seconds of silence between songs. There's your finished master tape!

RECORDING WITH MULTIPLE MICS AND A MIXER

To record live to 2-track in this manner, you must take extra care to achieve good isolation between microphones. To do this, try to record in a large room, use direct boxes, and mic close with directional microphones.

Set up your mixer as if you were going to do a mixdown, except with the input selector switches set to "MIC." Plug in your microphones, synth cables and direct boxes. Patch in effects and the 2-track deck. Set the master and sub-master faders to design center (about 3/4 up), and do the mix with the input faders. Adjust equalization, panning, and effects. Set recording levels: both the mixer and recorder meters should peak around +3 VU maximum for open-reel recorders, or 0 VU for cassette recorders. When all is ready, hit the record button.

You may need to adjust the mix during the performance. Do a few run-throughs and note on the faders the positions for each change.

TAPE COPIES

If you want to make copies of your master tape, you can either copy from one deck to another, or use a dubbing cassette deck that holds two cassettes.

To copy from one deck to another, simply connect a cable between the play-deck output connectors and the record-deck input connectors. This cable should have RCA phono plugs on each end to match your equipment. Set the recording level carefully to peak around 0 VU maximum on the loudest parts (+3 VU for open-reel decks).

To copy a tape with a dubbing deck, insert your master tape into the "play" section and insert a blank chrome or metal tape into the "record" section. The dubbing deck might work at two speeds; the slower speed usually provides better fidelity. If necessary, set the recording level and press the record button to copy the tape.

Whether using a dubbing deck or two decks for your copy, be sure to set the tape-type switches and noise-reduction switches appropriately. Ideally, the copy will sound nearly as good as the master tape.

CONCLUSION

While multi-track recording is the norm these days, excellent tapes can be made live to 2-track. Although this method is easier and costs less, it has limitations. Once these are overcome, the recording can sound cleaner and more realistic than a multi-track mixdown, and the musical performance can be very exciting. db

LEN FELDMAN

Lab Report

Tascam Model 238 "Syncaset" Mixer/Recorder

GENERAL INFORMATION

The Tascam 238 "Syncaset" is a rack-mountable 8-channel/8-track multi-track recorder. It records on ordinary, readily available standard compact cassette tape, Type II (high bias) and has a built-in dbx noise reduction system that can be turned on or off in groups of four tracks (1-4 and 5-8). The 238's discrete 8-channel format head is arranged so that the head gaps for tracks 1 through 4 are oriented one above the other while the gaps for tracks 5 through 8 are staggered with respect to tracks 1 through 4. When "bouncing" or performing a mini-mixdown of two or more tracks onto a single track, Tascam recommends doing so from one set of tracks (e.g. 1, 2, 3 or 4) onto a track found on the other half of the record/play head (5, 6, 7 or 8.)

The 238 is designed to work as the multi-track tape recorder in a system consisting of five other components: a mixer, input devices, output devices, signal processing and a final mixdown recorder. All of the steps involved in multi-track recording (tracking, overdubbing and mixdown procedures) must involve an external mixer since the 238 has no level control of its own. The basic functions performed by the 238 include insert capability with automatic monitor switching from tape to source, "locate" functions that allow you to quickly wind tape to any one of three specific locations on the tape automatically, "repeat," which will play a desired section of tape over and over again, "rehearsal" and "auto in/out" functions that allow you to set, audibly check and execute a punch in/out operation automatically and "tape sync" (on track #8) that's designed to synchronize the 238 with other external devices.

CONTROL LAYOUT

The left section of the front panel of the 238 contains the power on/off and eject buttons, a three position tape speed selector (fixed, variable, or external control and a continuously variable pitch control knob. Mounted below the cassette compartment are the usual tape transport buttons (Rewind, Fast Forward, Stop, Play, Pause and Record.) Further to the right is a counter display that shows tape run either in real time

Figure 1. Frequency response at 0 dB and –20 dB record levels. High tape speed results in little difference between the two plots.

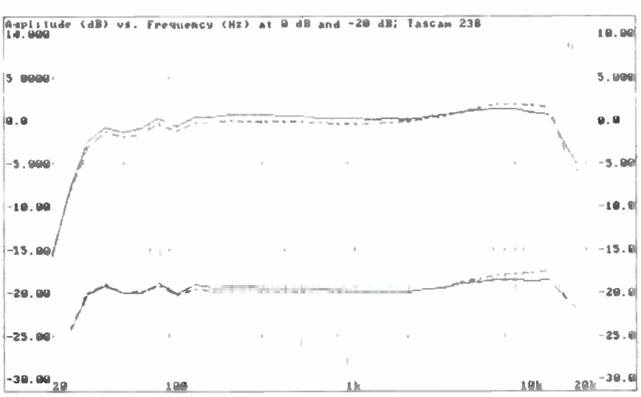
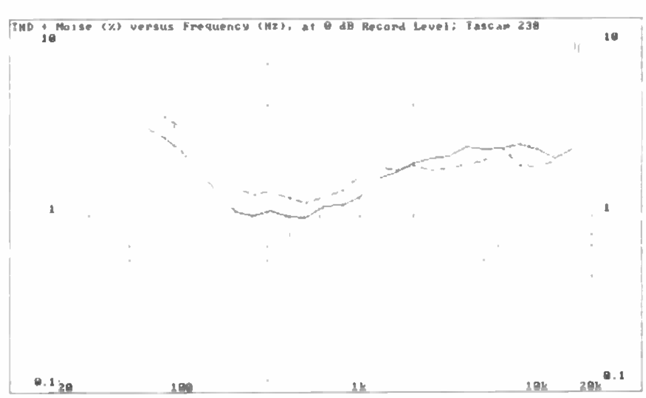


Figure 2. Distortion versus frequency at 0 dB record level.



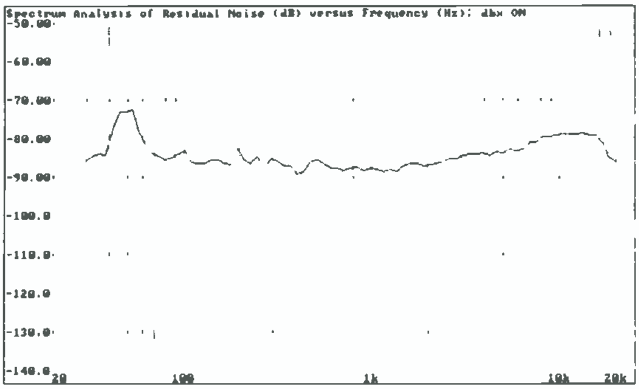


Figure 3A. A spectrum analysis of residual noise, referred to 0 dB record level, dbx on.

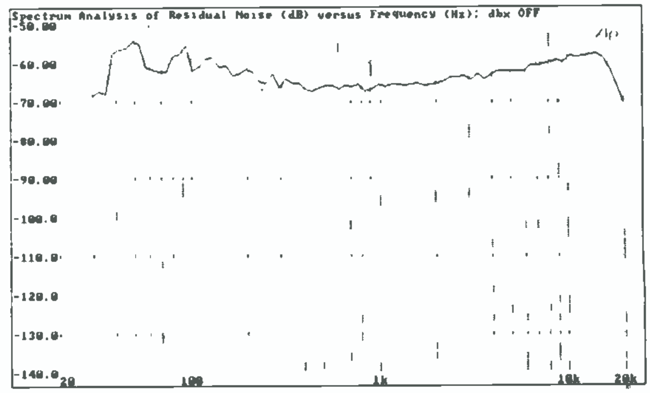


Figure 3B. A spectrum analysis of residual noise, referred to 0 dB record level, dbx off.

or as a four digit tape counter. Just below the display are a reset button and the automatic locator buttons that set location points and the buttons that are used to shuttle the tape to those memorized locations. Banks of four buttons further down are perhaps the most useful ones in terms of the professional user of this machine. They are labeled "Repeat," "Rehearsal," "Auto In/Out," "Clear," and "Insert." A "Tape Sync" LED is also found in this area of the panel, and it lights when a rear panel "Tape Sync" switch is turned on to make track 8 the track used to record and play back an FSK or SMPTE time code signal. "Rehearsal" is the first stage of an automatic punch-in recording. In this stage, the recorder memorizes the pre-roll, punch-in and punch-out counter locations. After you set the tape's pre-roll start point, the punch-in and punch-out points in rehearsal mode, the auto in/out button is pressed to put the recorder in a ready state and then, when the play button is pressed, the action is initiated. A shuttle knob is used to roll tape forward or backward as you monitor playback. Speed of tape roll depends upon how far you rotate the knob clockwise or counterclockwise. LED's indicate which, if any, sets of tracks have been set to record or playback with dbx noise reduction turned on. The right section of the front panel is equipped with 8 identical banks of LED indicators that serve as input or output meters for each of the 8 available tracks of the 238. Pushbuttons below each meter bank are used to put the corresponding track in the "Record Ready" mode, which is then indicated by a flashing LED below that bank. When recording actually begins (using the separate "Record" and "Play" buttons), the LEDs light continuously for those tracks that have been activated.

The rear panel of the "Syncaset" 238 is equipped with eight sets of unbalanced inputs and outputs (RCA type jacks), a pair of dbx switches (for activating dbx for tracks 1-4 and/or 5-8), a Tape Sync switch, a level control for the sync tone (if it is to be recorded on track 8), a remote punch-in/punch-out jack, a serial I/O (RS-232 type) port for linking the 238 to an external computer, a pair of dip switches for setting the bit rate, a filter switch (used when recording an FSK type synchronizing signal) and a remote control jack for connecting Tascam's RC-88 remote control unit. If this remote control is added to the system, all tape motion can be controlled from a distance of up to 15 feet.

LAB MEASUREMENTS

Our lab measurements were concerned primarily with determining the quality of recording that could be made using this 8-track compact cassette recording format. We had been impressed by Tascam's original 4-track/4-channel Syncaset recorders and wondered whether dividing the narrow cross section of a standard cassette into 8-tracks could yield acceptable results. Thanks to the cleverly designed 8-track head developed by TEAC especially for this application and to such electronic features as dbx noise reduction, lab results were surprisingly good.

Figure 1 is a plot of frequency response taken for two adjacent channels at 0 dB recording level and at -20 dB level. Thanks to the higher tape speed of 3-3/4 in./sec., results were virtually the same at both recording levels. Response at maximum recorded level extended from 30 Hz to 16 kHz, for the -3 dB roll-off points. At -20 dB recording level (the level at which consumer cassette decks are traditionally checked), response

Figure 4A. Distortion versus recorded level, measured from –6 dB record level to + 10 dB, with 1 kHz test signal.

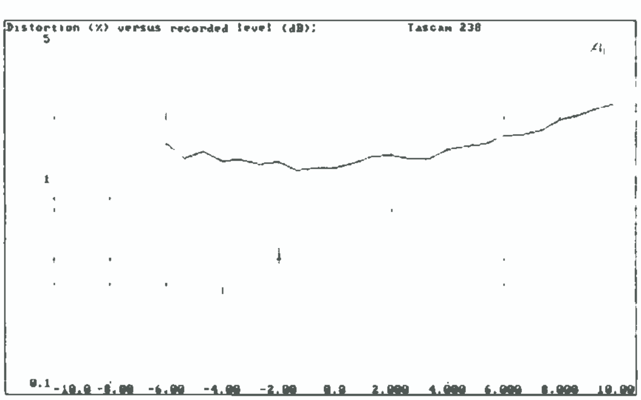
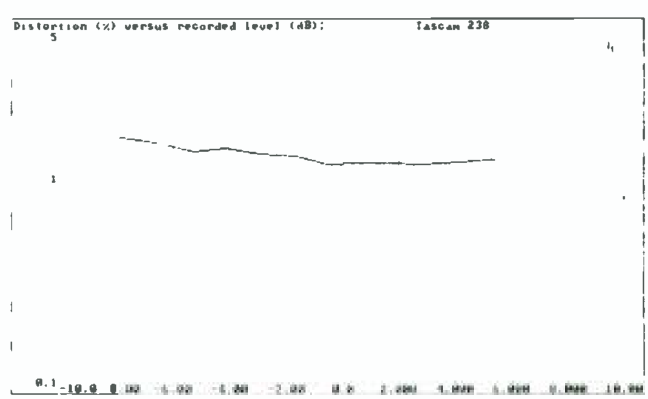


Figure 4B. Distortion versus recorded level, measured from –6 dB record level to + 6 dB, with 10 kHz test signal.



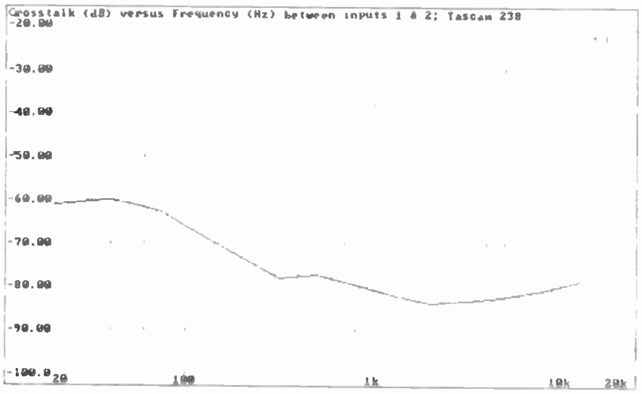


Figure 5A. Crosstalk versus frequency measured between inputs 1 and 2.

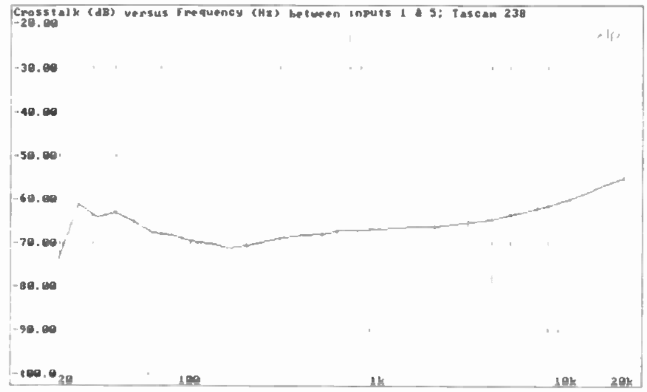


Figure 5B. Crosstalk versus frequency measured between inputs 1 and 5.

was even a bit better, extending down to around 27 Hz and right out to 20 kHz for the -3 dB roll-off point.

Figure 2 is a plot of total harmonic distortion plus noise versus frequency. At 400 Hz, our sample fell very slightly short of meeting Tascam's claimed figure of 0.8 percent. We read 0.9 percent on one channel and 1.2 for adjacent channel 2. It is possible that this slight discrepancy is the result of our measuring distortion plus noise, whereas TASCAM may well have measured only true harmonic distortion. In any event, even the slightly higher figures are perfectly acceptable for an instrument of this type.

Overall signal-to-noise ratio, referred to 3 percent distortion levels, was slightly better than claimed, measuring 94 dB, A-weighted, with dbx turned on and 60 dB without dbx. Figure 3A is a spectrum analysis of residual noise versus frequency. This plot is, of course, made without any weighting curve interposed between the output and the measuring instrument, and the slight rise in noise at 60 Hz is, of course, caused by the power supply of the instrument. Bear in mind, too, that this plot was made relative to 0 dB record level, whereas the published S/N figure is with respect to the 3 percent THD level which, in this case, occurred at a record level of +11 dB!

Figure 3B is similar to *Figure 3A*, except that for this spectrum analysis plot, the dbx noise reduction system was turned off for the channels recorded and measured. The increase in residual noise is obvious, amounting to more than 20 dB at most frequencies.

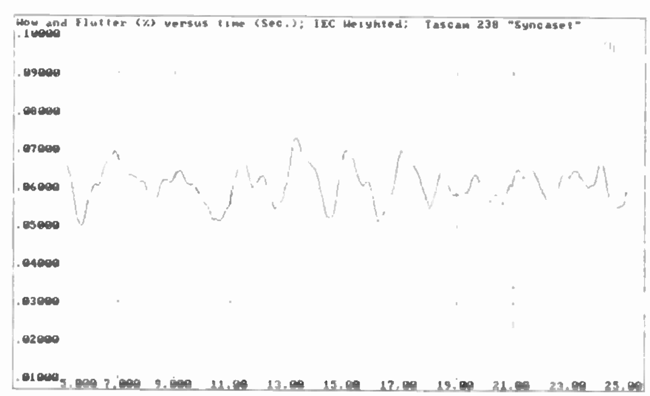
That is as good as we have measured for some open-reel decks in the past!

Figure 4A is a plot of THD plus noise for a 1 kHz recorded signal versus recorded level. The range of recorded level extended from –6 dB to +10 dB. At that maximum recorded level, THD was still under 3 percent. A separate spot check revealed that increasing the level by one more dB (to +11 dB) would result in a 3 percent THD reading, as mentioned earlier in connection with our S/N measurements. Normally, if you try to repeat this type of measurement at 10 kHz, using an ordinary consumer type cassette deck operating at 1-7/8 in./sec., you would find that THD at that high frequency would reach the 3 percent point at 0 dB or even lower recording levels. Not so in the case of the 238, with its higher tape speed and superior head configuration. We ran a test similar to that of *Figure 4A*, using a test frequency of 10 kHz. Results are shown in *Figure 4B*, and even at +6 dB record level, THD was only slightly above 1 percent.

We ran into one strange result when testing the sample 238 machine for crosstalk between channels. The TASCAM 238 owner's manual, you will recall, had suggested that when "bouncing" signals from a couple of tracks onto a single track, that tracks on opposite halves of the specially designed 8-track head be used. We supposed that this was to obtain the least amount of crosstalk. Yet, our own tests of crosstalk seem to prove just the reverse. For the test results plotted in *Figure 5A*, we recorded a frequency sweep test signal on track 1. During playback, we measured the relative output at track 2 (where nothing had been recorded. At 1 kHz, crosstalk measured an extraordinarily high 81 dB. Even at 10 kHz, crosstalk was 80 dB. The same experiment was repeated by using the same recorded signal as before, but making the measurement at the output of channel 5. These results are shown in *Figure 5B*. While the crosstalk was still certainly low enough so as not to create any audible problems, we are at a loss to understand why the readings were poorer than before: 67 dB at 1 kHz and 60 dB at 10 kHz.

There are any number of ways of measuring wow-and-flutter, and in fact, Tascam lists a couple of figures for this important tape recorder specification. We chose to use the IEC weighting method and, rather than to simply provide a single, average number, to plot actual wow-and-flutter over a period of 30 seconds, as shown in *Figure 6*. Peaks of wow during this period reached approximately 0.07 percent (better than the

Figure 6. Wow and flutter measured over a period of 30 seconds using the IEC weighting filter.



0.08 percent claimed by Tascam), while average wow-and-flutter hovered around the 0.06 percent mark. That is as good as we have measured for some open-reel decks in the past! Additional test results and general data are summarized in our table of VITAL STATISTICS at the end of this report. (A block diagram of the Tascam 238, showing signal paths and switching facilities is reproduced in *Figure* 7.)

COMMENTS

In the course of testing and evaluating this unusual little recorder, we used all of the controls and buttons, just to get a feel for how they work and how easy (or hard) it might be to use them. We found that the 29 page owner's manual supplied with this unit was an excellent tutorial in the use of this product. The step-by-step examples of setting monitor levels, recording a first track, overdubbing, punch-in or insert recording, rehearsal procedures, track bouncing (ping-pong), mixdown and the use of tape sync are clear enough so that even a novice could follow them successfully. Based upon our limited use of the 238, we would conclude that it is a reliably built, well-designed professional recording tool that will appeal particularly to the small studio owner/engineer. Of course, the real-use test of a product such as this can only be evaluated by a recording engineer attempting to use the 238 in an actual session. For that purpose, our sample was turned over to Corey Davidson for his hands-on evaluation of the Tascam 238. So, Corey, take it away and "roll tape!" db

VITAL STATISTICS

SPECIFICATION	MFR'S CLAIM	db MEASURED
Tape Type	Type II (High Bias)	Confirmed
Track Format	8-track/8-channel	Confirmed
Head Configuration	1 rec/play, 1-4/5-8	Confirmed
Motors,	3 (FG capstan, dc reel dc Ancillary)	Confirmed
Tape Speed	3-1/2 in.sec.(9.5 cm/sec)	Confirmed
Pitch Control	±12%	+12,–14%
Wow & Flutter (IEC Wtd.)	0.08%	0.07%
(NAB, WRMS)	0.04%	0.04%
Fast Winding Time (C60)	70 sec.	68 sec.
Record/Play Time	15 minutes (C60)	Confirmed
Nominal Input Level	–10 dBV (0.3 V)	0.321V
Nominal Output Level	–10 dBV(0.3 V)	0.28 V
Bias Frequency	85 kHz (±5%)	Confirmed
Frequency Response	30 Hz to 16 kHz	30 Hz to 16 kHz
THD, 400 Hz, 0 dB	0.8%	0.9% (1);1.2% (2)
S/N, dbx ON (re:3% THD)	93 dB, A-wtd.	94 dB, A-wtd.
S/N, dbx OFF (re:3% THD)	58 dB, A-wtd.	60 dB, A-wtd.
Crosstalk, Adj. Chs.	70 dB, 1 kHz	81 dB (See text)
Erasure	70 dB, 1 kHz	N/A
Power Requirements 120 V AC 60 Hz,	47W	Confirmed
Dimensions (WxHxD, in.)	19x5-7/8x13-9/16	Confirmed
Weight	20.94 lbs (9.5 kg)	Confirmed
Price	$2,299.00	

COREY DAVIDSON

Hands On

As you may already know, a hands-on review of equipment differs from a lab report in that the piece of equipment in the hands-on context is actually put through the rigors of usage that approximate virtually any condition and application that might arise in real life. Let's face it...specs don't really tell you everything about the piece in question.

Every person who uses audio and recording equipment will have a different approach to the process. Some individuals are particularly light-handed. Others are abusive. Although I have a musical and technical background, I realize that these abilities really bear little to no weight when it comes to creativity. (Some of the most talented individuals that I have ever met had little to no regard for the "hows and whys" of a unit's operational theories and principles.) It is with this notion in mind that I approached this hands on. I threw my preconceptions and principles to the wind in an attempt to rediscover what its like to learn the simplest recording motions.

ANSWERING QUESTIONS

In multi-track recording there are a couple of things that the average user will often wonder about. Number one: How well does this unit document my sound sources (whatever they might be)? Two: When I run out of tracks and have to bounce, how much deterioration must I tolerate? Three: Is this unit easy enough to use so that my creative inclinations are not infringed upon? You might have more to add to this list, but please be patient because those other questions are quite probably answered by the process of the hands-on experience.

For this review, I chose to work with one of my friends (and industrial colleague) Phillip Antonucci. Phil has been roughing it in the music industry for a good 20 years. As the owner/engineer of his own commercial 8-track studio, Phil has always put all of his gear through special trials. First of all, much of the equipment that he owns, including the multi-track machines, have been moved in and out of his studio repeatedly in order to facilitate his live/location work. I believed he would be especially suited to assist in this hands-on for he has owned and discarded many small-format multi-track machines. He is the owner of 2 Tascam Model Fives, an 80-8 that had its heads relapped once and now has new heads, a 25-2, and a 40-4. It is for this reason that I feel his opinions are so valuable. Presently, he singlehandedly composes and records the music for Madison Square Garden Sports Network at home in his Electronic Cottage...a separate part of Phil's creative life.

I told Phil, "Let's approach this like a real, clock-running, recording session...with one exception...we never used this Tascam 238 before in our lives."

Phil said, "Let's switch off. You play the uptight musician who has no time for down time, and I'll play the engineer who's got to get it together so that I don't lose the client. Then, we'll switch and you can play with the machine." I liked that idea.

> **We now had about 2.5 minutes of drums on tape. I said to Phil, "So do you think the tape sounds real close?"**

After selecting a virgin Maxell UDX-LII cassette (Tascam recommends type II high bias tape), the first thing that we did was calibrate zero on the 238's LED indicators to 0 VU on the console meters. This was done with a 400 Hz test tone with the 238 on input (record ready)...dbx out. The tone was generated from within Phil's Tascam Model V console. We rolled tape, documenting the tone and a playback comparison was made. The level in playback was within less than 1.5 dB of the original. The tone was used again, recorded and played back with dbx on.

The dbx threw us a curve for which we had to adjust. The playback levels were consistently lower than those during input. This is not the 238, but rather the result of the way in which dbx does its thing. After some trial and error in determining what levels rendered the closest to 0 on the 238's LED indicators, we found that by using dbx there was a marked improvement in the sonic clarity and a tremendous reduction in noise. At this point we decided that dbx would be used for the entire session.

IS THERE A DIFFERENCE?

Phil had a piece that he needed to complete at a deadline for Madison Square Garden. We started with basics, drums and bass. First the drums were laid down from a Yamaha RX 11 onto tracks 2 and 3. While in input, the peaks of the transients from the drums topped out at +2 on the 238's LED bar. Upon playback the peaks were down a notch, however, there wasn't the slightest difference in level that either one of us could detect. At this point Phil and I played a guessing game. I turned my back to the monitors and the tape machine and Phil switched between drum source and playback, challenging me to determine which was which. Then we changed positions. Neither one of us was capable of telling the difference. As a matter of fact, the only clue that could possibly reveal a difference was a volume cue but even that didn't work because in some cases we found that the dbx was an indescribable sonic enhancement. By now you might have determined that the dbx is a necessary and useful tool in context with the 238.

We now had about 2.5 minutes of drums on tape. I said to Phil, "So do you think the tape sounds real close?"

He replied, "I've got to hear more! Let's lay down a bass track."

After selecting a MIDI composite bass sound, I was ready to play. I told Phil that he had better have his *punch* talents ready because the tempo of this piece was near 150 BPM. The bass had to keep up and we were not using any sync information yet. I started playing and after about 10 bars I botched my part. The thing to do at this point was to punch-in and finish out the piece. We did just that using manual punch-in. I botched again. "Can't this be easier?" we said.

Here is where the real fun began. It became clear that a *punch-in* would not be sufficient. I needed to punch-in and then punch-out long before the end of the piece. The treat here is the rehearsal of inserts. You can set the points at which you have to punch-in and punch-out and play and hear the whole process without really doing it. Super! But there's more. The punch-in and punch-out points can be committed to one of two memories that can be updated at any time, almost instantly. It's very simple. Once we had selected the track input, the RHSL (rehearsal) switch was pushed. The record function lights were no longer blinking, they were now solidly lit giving the impression that we were going to tape. However Tascam just wouldn't let us unintentionally record. Next the insert switch was depressed, lighting it. Phil started the tape rolling and at the desired punch-in point, hit RECORD and the memory display proudly lit the same number. When the punch-out point came up Phil hit PLAY, and the punch-out point was now committed to memory. The beauty of the RHSL mode is that when the punch-out point is reached, there is a 3-second roll of tape and the machine rewinds to the starting point, setting you up for as many practice runs as your heart desires.

During the RHSL mode the memories that contain the in and out points are displayed adding further to the logic and clarity of operation.

There are four ways to get out of the RHSL mode: Pressing the RHSL button (which maintains all memories of punch points), pressing PLAY (which takes the tape back to the start point and clears the in and out points), pressing CLEAR (which clears in and out points yet does not dictate a tape location), and ejecting the tape.

Phil said, "Anyone can easily learn the functions of this machine. There are two memories that can be kept independently of your rehearsal punches. You can have memory 1, memory 2, and set them in the beginning and leave them there. Then you can do the manual punches wherever you like to fix things and rehearse. It's very accurate too. Really accurate."

COMPARES TO OPEN-REEL

We continued to record successive tracks and were amazed at how good those tracks sounded. We both couldn't help comparing what we heard on the 238 to what we have heard on any of the other open-reel 4 and 8-track machines. That's right...this little cassette machine is that impressive. It shouldn't be so hard to believe in light of the advancements that have taken place in the large scale integration of circuitry combined with the improvements that have been made in already existing operational amplifier slew rates and frequency response.

A stranger to the cassette multi-trackers is the shuttle function. This little one-knobbed character enables the user to slowly rock the tape back and forth in order to find a specific location on the tape. This can be useful when looking for holes in the music that can help to determine punch points. But be careful!

Shuttling is a feature that is typically found on larger open-reel machines. When in a shuttle mode, both reels are under tension in opposite directions. With 0.125-inch tape, remaining in the shuttle mode can cause the tape to stretch. Tascam recommends using the shuttle sparingly. They say that extensive use of the shuttle can cause premature wearing down of the heads. Phil and I discovered that using the shuttle even to a lesser degree caused the tape counter to ever so slightly slip location. With a function that is mostly associated with grease pencils, it was a surprise to see a shuttle function on a machine such as this where you couldn't use a razor blade without much contortion. Nevertheless, Tascam has covered every base and the availability of shuttling is basically another plus on the long list of plusses.

One point of curiosity was what happens in a bounce. With the piece that we were working on, only one bounce was necessary. However after we were finished with our serious work, we tried some unconventional things. We bounced seven tracks down to track 8. The integrity of all those tracks was seemingly unaffected by the bounce. If it can take that many tracks and scrunch them down without noticeable degradation, then we're sure that two or three tracks will bounce superbly.

PLAYBACK ABILITIES

Another experiment that we performed was an exploration of the sync recording and playback abilities of the 238. By using a drum machine, we laid down drum sync on track 8 which Tascam has set aside for a double life. Normally track 8 is an audio track, yet this eighth track has a level and filter control all its own enabling the fine tweaking of sync level that is so often needed when using this ratchet-like function. Many devices that operate in conjunction with sync tones have a very narrow tolerance for deviations in level. In addition, sync has a way of leaking and bleeding onto adjacent tracks. Tascam has provided for this by the use of a filter that allows the passage of the narrow band needed for the intelligibility of sync while reducing the lowest and highest frequency information, thus reducing sync's overbearing tendency to infiltrate other tracks.

Finally, a word about the remote. We used the RC-88 remote for the entire hands-on, an indispensable convenience. The use of the remote also will greatly reduce any need to touch the panel on the 238, helping to keep it dust and grime-free. Fortunately all functions, short of tape load and eject, appear at the remote. This is a highly professional option with an ultimately practical purpose.

It is the transport that is so obviously impressive. Handling a cassette with the speed, accuracy and delicacy associated with much larger and expensive machines is a feat in itself. The FF and RW times are much faster than most other cassette machines. If this transport system stands a test of time, then this little guy will probably go down in the annals of recording as "that little BIG guy."

CONCLUSION

From the very first page of the 238 manual, the user is treated like a professional. The manual is a veritable introduction to recording techniques and is incredibly easy to read. From tracking to overdubbing to mixdown, the 238 manual takes the user on a step-by-step tour that leaves no room for error. Whether your experience is vast or limited, this manual can be appreciated on all levels of understanding. db

TEKTEXT #104 CONTINUED

Dear Editor,

I applaud your decision to publish a "Glossary of Audio Terms" (TEKTEXT #104, Nov/Dec issue). The need for universally understood definitions is clear and particularly pressing in the audio world.

Unfortunately, Part I of the Glossary does more to demonstrate the need for authoritative definitions than it does to provide them. There are mixed up units and errors of fact that support common misconceptions in some cases while in other cases the column has errors I have never before encountered. If we confuse the meaning of the units we use in a measurement, we run the risk of incoherent communication.

Sound intensity cannot be changed by **absorption**—intensity is the power which is transmitted across a unit area; it has nothing to do with absorption. The **absorption coefficient** does not contain frequency information, that information is always separately conveyed. On the perceptual side of things, **ambience** is affected by many factors, not just reverberation, and no room has a *lack* of ambience because ambience is not a scaled quantity. It is particularly irksome to find **axis** defined as a point in a respected journal.

Among the errors of fact that appear in the column, changing the shape of a waveform while maintaining its amplitude will change the power in the signal without changing **amplification**; the **audio frequency** range quoted in the column is larger than current literature supports (12 hertz!?); the function of an **acoustic baffle** is to acoustically separate two regions and not to absorb energy from either; the bus illustrated in the column is an example of a *data bus* and not the stated term; and **baud** is the number of *bits* (not bytes) that is transmitted in one second.It belies a common misconception to say "baud rate" while baud communicates the information to determine transmission speed it is a unit which uses no further modifiers. (We do not say "hertz frequency" or "hertz per second" nor should we say "hertz rate.")

With an eye to the length of this letter, I will constrain my comments to only the more glaring examples from the A's and B's. Readers who seek readable (even humorous), accurately-defined audio terms will find a treasure-trove in *The Audio Dictionary*, by Glenn D. White, University of Washington Press, 1987.

Yours truly,

Doug Solowan, Ph.D.
Seattle Central Community College.

Drew Daniels responds:

Excuuuse Mmmeee!! Among the hundreds of applause I have received on the Glossary, I have "irked" one reader. It appears I may be guilty of rushed rhetoric and sloppy syntax! All I can say is that I wish I had had Dr. Solowan's consultation during the odd minutes between phone calls spent on the composition of the Glossary, but as all writers, filmmakers, musicians and artists know, anything you do, draws critics—to paraphrase an overused line—those who can do, and those who don't, criticize.

In my own defense, let's take each stated objection, one at a time:

ABSORPTION: It is true that absorption does not change intensity. Absorption changes one form or energy to another. Changing sound to heat is one example. The sentence would have been correct if I'd included the word "local" before sound intensity or if I had said "reflected sound" intensity.

ABSORPTION COEFFICIENT: As Dr. Solowan states, no frequency information is implied by the term "absorption coefficient," but in my haste, I forgot that not absolutely every soul on earth remembers that acoustical engineers universally use frequency references, specifically, ISO octave centers, to describe the absorption coefficients of building materials. Most common materials as they are commonly used, exhibit different absorption coefficients at different frequencies, generally, being better absorbers of shorter wavelengths, except where panels vibrate sympathetically, and can cause losses by non-adiabatic energy conversion or mechanical loss conversion (another form of heating). Absorption is not easily understood even by graduate engineers, and I did not want to scare my non-engineer readers by writing a definition that might stupefy them. Bruel & Kjaer publishes a great little booklet on noise control that describes absorption, transmission and reflection in basic engineering terms. I recommend this for those who want to get their feet wet in these subjects.* For an absolutely marvelous tutorial review of basic thermodynamics, I suggest readers view the Cal Tech-produced "Mechanical Universe" programs broadcast on PBS stations, dealing with that subject. The very gentle introduction to thermodynamics provided by Mechanical Universe will suggest how complex the mechanisms of sound absorption can be.

AMBIENCE: We could argue that the perceptual side of things is one way or another. Perception is dangerously subjective. I suggest that it's arrogant to assume everyone's perception should be like yours, 'cause it ain't so. If you were to suggest to an orchestral conductor or an opera singer, for example, that a draped and carpeted movie theater had any sort of ambience, you might be risking bodily injury or at least blistered ears.

AXIS: According to Webster's New World Dictionary, a real or imaginary straight line around which the parts of a thing, system, etc. are symmetrically or evenly arranged or composed, or a straight line through the center of a plane figure or solid, and so on. In attempting to briefly state what is "on-axis" and what is not, I made a slip of the word processor, using point instead of

*Bruel & Kjaer
185 Forest Street
Marlboro, MA 01752

line. Germany and Italy never crossed my mind. Boy is my face red!

The good Doctor claims there exist errors of fact in the Glossary, while some definitions are of common use and others define common applications of subjectivity...

AMPLIFICATION: *Oops!* Dr. Solowan did not read my definition carefully. I said "An increase in signal quantity of *either* amplitude *or* power level. These can, of course, be independent quantities, for example, at the same peak-to-peak voltage, a square wave would represent about twice the power of a sine wave driving a load.

AUDIO FREQUENCY: Current literature includes or should include Dr. Marshall Buck's study of low-frequency pitch perception, published several years ago by the Audio Engineering Society, delivered in person by Dr. Buck at the international AES convention, and widely regarded as a valid study, since not many would go to the expense of recreating it. Dr. Buck placed numerous human subjects in a large, sealed chamber, one wall of which was covered with woofers to produce high sound levels and low distortion. He found that humans can hear down to a few hertz, and can perceive pitch down to about 12 Hz if the source is loud enough. Dr. Solowan's literature needs an update to be current.

BAFFLE, ACOUSTIC: Maybe the function of Dr. Solowan's baffle is as he states, to separate two "regions," but the function of my baffles and those of thousands of other recording engineers are as the Glossary stated. Again, a case of "let's confuse the novices for the sake of accuracy." In this case I am willing to go even further and say that in the interest of recording art, recording engineers are apt to use anything and everything as what they will call a "baffle" (because no other names for the things have yet been invented), to modify, filter, equalize, reflect and even amplify sound to produce some novel or desired sonic effect

BUS: The illustration referred to was a literary one. Novices who have no exposure to a technical term sometimes need a literary illustration to break a conceptual barrier they may have to understanding what the term means in a particular usage. In any case, audio information, be it analog, digital or acoustic is *data*, isn't it? I stand by my definition, and suggest that it's a bit of a narrow view for an academic to hold that literal definitions are the only ones acceptable. I teach a class in loudspeaker and acoustic technology at USC, and I have run into the need for conceptual blockbusters even with graduate Electrical Engineering students in my class.

...There are cases where no universally accepted definitions fit the facts.

BAUD: According to Webster's New Universal Unabridged Dictionary, baud is "a unit of signaling speed in telegraphic code, or the number of bits per second that can transmitted in a given computer system." Telegraphic code such as Morse code, is made up of "dits" and "dahs." Each dit or dah requires two pieces of information (on-off and long-short), and so each individual piece can be considered a "byte." It is this reference I used when inadvertently including the word byte. In my opinion, taking issue with modern word definitions like this – words that haven't even had time to settle firmly into the language – is petty.

An anecdote is in order here. I remember vividly, delivering a technical paper at a New York AES convention a few years ago. The session chairman was D.B. Keele Jr., and my paper was entitled "Thiele-Small nuts and bolts with painless math." It is a tutorial designed to put the established laboratory techniques of loudspeaker engineers and the rather abstruse math of electrical circuit modeling of loudspeakers, into the hands of non-mathematician practioners – an effort I still deem worthwhile. Dr. Stanley Lipshitz, a mathematician and now an AES officer, posed several questions after my talk. He pedantically insisted that I should have used five decimal places in my work where I used only four, and that I should have striven for 0.1 percent accuracy instead of 5 percent. Don Keele rose to defend my methods in view of the fact that even the best real loudspeakers can only be expected to hold to 5 percent tolerances in production in most cases, and that 0.1 percent accuracy in the real world is an absurdity in view of the fact that a few degrees of air temperature shift would negate such accuracy and so on. The point is, why spoil it for practitioners just to maintain an absurd level of accuracy.

Science and Engineering *are* different, and that's why universities have different degree programs for physicists and mathematicians, and for engineers. The audio equipment for sale from manufacturers would be un-buildable and therefore would not even exist if it had to be built to the standards of theoretical perfection. The evidence of this is easy to see, just look at the high cost of military electronic components and equipment, a cost driven up by tens or hundreds over commercial counterparts, just to achieve a slight improvement in performance and some added reliability, but nothing even approaching theoretical limits. Further, consider equipment designed by theoreticians that somehow makes it through the design review process without being looked at by end user types. I have seen an awful lot of good equipment with little or no practicality engineered in. I have seen a sixteen-track tape deck that would not punch in without making huge pops, and did not have any way to listen to pre-punch sync. When the design engineers were questioned about this, they became indignant and replied that the machine would record 8 stereo albums on one roll of tape! Really! I wrote a manual for a recording console which boasted a simple one-button remix, and while working with a prototype unit during the writing, discovered that pushing the remix button with the inputs set to "line" instead of "tape," caused all the channels to feed their outputs to their inputs and send the whole board into hard feedback! Really!

Nit-picking a generalized, popularist and highly simplified Glossary is as much trouble as writing a major article, take it from one who knows. I seldom write *Letters to the Editor* because it's too much trouble. Although accuracy of definition and syntax is highly desirable, the Glossary, as it is, provides something (as opposed to nothing) for those whose knowledge does not include any exposure at all to the terms it includes. Obviously, Dr. Solowan is learned, observant and fastidious. I for one would welcome a **db** article from such a keen mind. An article that speaks to **db** readers without teaching them calculus first, and is also accurate and thoughtful perhaps. We must encourage those seeking to enlighten the audio community, because it *is* such a diverse group made up of people with such differing amounts of technical background.

Drew Daniels

Buyer's Guide—Microphones and Studio Accessories

On the pages that follow you will first find a Guide to microphones in chart form that is immediately followed by paragraph forms for wireless microphones and then studio accessories. Manufacturer's addresses conclude the Guides.

As usual, be aware that we attempt to contact every manufacturer, but not all are cooperative or prompt enough for our necessary deadlines.

Model	Type	Patterns	Frequency Response dB, +-dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
AKG ACOUSTICS, INC.												
C401/ B9	cond	fig 8	10-10k	200			1.5 .3	.3	blk	.125 plug	$140.00	Contact mic which mounts directly on stringed instrument bodies. Includes power supply with volume/balance controls.
C402/ B8	cond	card	2.5k-20k	200			1.5 .3	.3	blk	.125 plug	$150.00	Complements the C401 and installs in the sound hole picking up sound radiated by the strings. Plugs into B9 power supply.
C408/ B9	cond	hyper card	80-20k	200			3 1.4	1.4	blk	.125 plug	$220.00	Tailored for percussion in that it clamps on the rim. Response is optimized for drums. Plugs into B9 power supply.
C409/ B9	cond	hyper card	20-20k	200			5.7 1.4	1.6	blk	.125 plug	$220.00	Designed for wind instruments in that it clamps to the horn giving freedom to move while maintaining optimum placement.
C410/ B9	cond	card	20-20k	200			5.1	4.6	blk	.125 plug	$220.00	Headset mic available with B9 battery power supply eliminating the need for phantom power.
C1000S	cond	card	50-20k	200		137 .5	8.7 1.3	9.2	dark gray	XLR	$325.00	Rugged vocal mic, 9 volt or phantom powered. Features include on/off switch with a response similar to the C535.
C747	cond	hyper card	30-18k	400		133 .5	5.3 .4	1.2	blk	XLR	$400.00	Pen-sized mic with a specifically tuned acoustic tube in front of transducer yielding high sensitivity and smooth response.
C414/ TL	cond	card hyper omni fig 8	20-20k	180		140 .5	5.6 1.4	11	blk metlc	XLR	$1195.00	Transformerless version of the C414B-ULS. Some prefer this version for its powerful low frequency reproduction.
ALTEC LANSING CORPORATION												
C641L	elec cond	omni	80-15k	150	45		.851 .415	.67	blk matte	A3M	$268.00	Lavalier mic for sound reinforcement. Has high quality, natural sound.
C644	elec cond	card	50-18k	150	50		7.5 1.95	12	blk matte	A3M	$208.00	Ideal for speech or singing reinforcement systems where there is a tendency for feedback.
D645	dyn	super card	80-15k	150	56		6.56 1.375	6	beige matte	A3M	$348.00	Offers an outstandingly smooth frequency response with great naturalness and accuracy of inflection.
C649	elec cond	card	40-18k	150	45		6.94 1.06	8	beige matte	A3M	$356.00	Wide range, high input Z without distortion. True cardioid pattern. Designed for voice.
D90P	dyn	omni	50-15k	150	58		7.5 1.625	16	satn chrome		$220.00	15-foot, 3-conductor cable, 80% shielded permanently attached to mic.
D91P	dyn	omni	180-10k	150	60		3.625 2.5	4.5	charc gray		$152.00	2 shielded conductors, 2 unshielded conductors.
D646	dyn	super card	80-15k	150	57		7 1.875	8	silv beige	A3M	$376.00	
654A	dyn	card	50-15k	200	56		7.25 1.875	8	mat satn nickel	A3M	$224.00	
AMS/CALREC												
CM 1050C	cond	card	30-20k	200		130 .5	5.5 .875	4	blk alum	XLR	$215.00	
CM 2001C	cond	omni	20-20k	200		130 .5	6.625 .875	4.2	blk alum	XLR	$295.00	
CM 1051C	cond	card	40-20k	200		130 .5	5.5 .875	4	blk alum	XLR	$215.00	
CM 2003C	cond	omni	20-20k	200		130 .5	6.25 .875	4.2	blk alum	XLR	$280.00	
CM 2051	cond	card	40-20k	200		130 .5	6.25 .875	4.2	blk alum	XLR	$280.00	
CM 2050C	cond	card	30-20k	200		130 .5	6.25 .875	4.2	blk alum	XLR	$280.00	
CM 2056C	cond	bass roll off	40-20k	200		130 .5	7.25 .875	4.5	blk alum	XLR	$330.00	
CM 4050 Soundfield	cond	any	20-20k	100		140 .5	9.5 1.5	18	blk alloy	XLR	$5400.00	Unique stereo mic system capable of variable patterns from omni thru all cardioids to figure 8. 3D ambisonic pick-up.
AUDIO TECHNICA U.S., INC.												
AT4051	cond	card	20-20k	250	35	143 1	6.125 .812	4.2	blk chrome brass	XLR	$550.00	Transformerless. Head capsule interchangeable for omni or hyper. 48 volt phantom powered.
AT4049	cond	omni	20-20k	250	34	142 1	6.125 .812	4.4	blk chrome brass	XLR	$550.00	Transformerless. Removable head capsule is interchangeable for cardioid or hyper. 48 volt phantom powered.
AT4053	cond	hyper card	20-20k	250	35	143 1	6.125 .812	4.2	blk chrome brass	XLR	$550.00	Transformerless. Head capsule interchangeable for cardioid or omni pattern. 48 volt phantom powered.
AT4071	cond	lobar	30-20k	250	25	127 1	15.56 .812	5.8	blk chrome brass	XLR	$900.00	Transformerless. High output, low noise, very light weight. 12-48 volt phantom powered.
AT4073	cond	short line	30-20k	250	56	129 1	9.125 .812	4.2	blk alum	XLR	$750.00	Transformerless. High output, low noise, very light weight. 12-48 volt phantom powered.
ATM33R	elec	card	30-20k	150	45	141 1	7 .812	4.7	matte	XLR	$235.00	
ATM73	elec	card	60-15k	250	57	140 1		1.1	matte	XLR	$235.00	Headworn cardioid. Very light weight, comfortable. Battery or phantom powered.
BARCUS-BERRY, INC.												
ZMC-1	elec	spec	20-20k				2 1.5	1.8	blk	.25	$995.00	Mic system for cymbals. Power mixer with five cymbal mics and one hi-hat mic. Mixer has volume and pan for each mic.
1574	elec	spec	20-20k				1.5 .375	1.4	blk	.25	$109.50	For brasswind instruments. Attaches to bell. Quick installation or removal. Includes power supply.
1376	elec	spec	20-20k				1.5 .375	.2	blk	spec	$109.50	For single reed instruments of the flute family. No tools required for installation or removal. Includes power supply.
1372	elec	spec	20-20k				1.5 .375	1.6	blk	.25	$109.50	For instruments of the flute family. No tools required for installation or removal. Includes power supply.
1525	elec	spec	20-20k				1.5 .375	.6	blk	spec	$109.50	For use with stringed instruments. Clips to strings behind bridge. For violin or viola; as additional mic for cello/bass.

Model	Type	Patterns	Frequency Response dB, +-dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
1580	elec	spec	20-20k				1.5 .375	1.4	blk	spec	$109.50	For use with all types of harmonicas. Can be quickly attached to or removed from comb of instruments without tools.
1485	elec	spec	20-20k				1.5 .375	1.4	blk	.25	$109.50	For use with accordion. Requires custom installation inside instrument. Power supply furnished.

BEYER DYNAMIC, INC.

Model	Type	Patterns	Frequency Response dB, +-dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
MC734	cond	card	20-20k 3	150	46	138	8 3	12	blk anod brass	XLR	$785.00	Ultra low noise/low distortion. Frequency tailoring switches, excellent for critical vocal, sampling, instrument recording.
MC740	cond	omni hyper bi-dir	40-20k 3	150	40	144	9 4.5	14	blk anod alum	XLR	$1325.00	Large diaphragm, multi-pattern, smooth response, ultra-low noise, low distortion. excellent for vocal, sampling, instrument recording. Frequency tailoring and attenuator switches.
MCE10	cond	hyper card	40-20k 3	600	43	116	1 .5	11	blk anod alum	XLR	$395.00	Miniature, high sensitivity. Excellent for close-micing brass woodwinds, stringed instrument recording and reinforcement.
MCE81	cond	super card	40-20k 3	150	50	138 1	7.5 3	9	blk anod brass	XLR	$299.00	Excellent weight balanced feel, slim design. Great gain be fore feedback. Internal shockmount reduces noise. For vocals.
M422	dyn	super card	100-12 3	200	57	140	4 1.5	3.5	blk anod brass	XLR	$135.00	Small size, rugged design, fast transient response. Excellent for snare and hi-hat.
M420	dyn	card	100-12 3	200	57	140	5 1.5	4.5	blk anod brass	XLR	$199.00	Small size, rugged design, boosted midrange response. Excellent for snare, rack and floor toms.
M201	dyn	card	40-18k 3	200	57	140	7 1.5	7	blk anod brass	XLR	$260.00	Slim design, rugged, high SPL capability. Extended low-end response. Excellent for rack and floor toms.
M380	dyn	card bi-dir		200	46	140	11 4	14	blk/ bronze	XLR	$280.00	Internally shock mounted, large diaphragm, high SPL without overload. Frequency response down to 15 Hz. For bass drums.

CARVIN CORPORATION —See our ad on page 15

Model	Type	Patterns	Frequency Response dB, +-dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
CM90E	cond	card	30-20k	250		132 3			dark gray alum	XLR	$139.00	Super low distortion studio mic.
CM67	dyn	card	40-15k	250		130 3			dark gray alum	XLR	$99.00	Instrument mic for live or studio.
CM68	dyn	card	40-15k	250		130 3			dark gray alum	XLR	$99.00	Excellent mic for live vocal work.

COUNTRYMAN ASSOCIATES, INC.

Model	Type	Patterns	Frequency Response dB, +-dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
Isomax 2-HHG50	elec card	hyper 3	40-20k	200	37 1	130	.625 .312	.1	blk matte	XLR	$220.37	Excellent choir mic. Has ruler-flat response. Excellent off-axis rejection and extremely small size.
Isomax 4	elec	hyper card	70-15k 3	270	47	145 3	.437 diam	4	blk matte	XLR	$341.00	Exceptional on podiums. This gooseneck mic has internal electronic vibration isolation...no need for shock mounting.
Isomax Headset	elec card	hyper 3	40-20k	200	57 1	150	.5 .25	.05	blk matte	XLR	$242.00	Yields recording quality vocals. Innovative headset design frees your hands. Flat response enhances voice.
Isomax TVH	elec	hyper card	70-15k 3	270	47	145 3	.875 .437	.5	blk matte	XLR	$319.56	Available for wireless or hardwired operation. Combines outstanding off-axis rejection with low handling noise.

CROWN INTERNATIONAL

Model	Type	Patterns	Frequency Response dB, +-dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
PZM-30R	elec	hemi	20-15k 6,3	240	65	150 3	5 6	8.5	blk/ gold alum	XLR	$349.00	Pressure zone microphone. Smaller models available.
PZM-30FS	elec	hemi	20-15k 3	240	67	150 3	6 5	6.5	silv alum	XLR	$349.00	Pressure zone microphone. Smaller models available.
GLM-100	elec	omni	20-20k 3	240	71.5	150 3	.755 .31	1	blk PVC	XLR	$199.00	Miniature clip-on mic for voice and instruments. Model GLM-100E for wireless appl. Model GLM-100D for dual lavalier use.
GLM-200	elec	hyper card	60-20k 2.5,6	100	69	150 3	.755 .310	1	blk PVC	XLR	$229.00	Miniature clip-on mic for voice and instruments.
PCC-160	elec	half super card	50-18k 3,6	150	53	120 3	6.7 3.2	11	blk steel	XLR	$275.00	For stage-floor pickup of drama, musicals, opera. Also for lecterns and news desks. Bass-tilt switch.
LM-200	elec	super card	80-15k 2.5,6	100	68	150 3	16 1.1	10	blk steel	XLR	$289.00	Lectern mic with swivel mount for noise-free adjustment. Phantom or 12 volt powering. Pop filter, low-cut filter.
CM-200	elec	card	80-15k 3,6	200	73	151 3	7.53 1.8	7	blk alum/ steel	XLR	$259.00	Smooth, articulate sound for handheld stage vocals and instruments. Wood handles available.
CM-310	elec	card	60-17k 4,6	200	77	151 3	7.33 2.04	7	blk alum/ steel steel	XLR	$309.00	Differential cardioid has outstanding gain before feedback. For handheld stage vocals. Wood handles available.

C-T AUDIO MARKETING, INC.

Model	Type	Patterns	Frequency Response dB, +-dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
C-ducer Gigster	cont		42-22k 0,3	10k		160 .02	8 .75		brwn vinyl	.25	$145.00	Contact condenser mic. Suitable for strings, acoustic keyboards and drums. Battery powered.
C-ducer CX series	cont		42-22k 0,3	600 10k		160 .02	8 .75		brwn vinyl	XLR	$266.00	Phantom powered or AC powered (via adapter) and has both 600 ohm balanced outputs and unbalanced outputs.
C-ducer Lost Cord	cont		42-22k 0,3	600 10k		150 .1	3 .6		silv	.25	$129.00	Condenser bridge mic for guitar. It fits under the guitar saddle and comes complete with preamplifier.
C-ducer Drum Wizard												Is an 8-channel drum microphone and mixer system that also provides MIDI and trigger outputs.

Model	Type	Patterns	Frequency Response dB, ±dB	Impedance, ohms	Sensitivity, 1 kHz,dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
ELECTRO-VOICE, INC.												
N/D408	dyn	super card	60-18k 3	150	50		4.55 2.85	6.7	blk	XLR	$234.80	Pivoting instrument mic with special element for wide frequency response and high output.
N/D457	dyn	hyper card	55-18k 3	150	50		7.12 2.05	7	blk	XLR	$228.60	Hand-held vocal mic with hypercardioid pattern for very high gain before feedback.
N/D757	dyn	super card	50-18k 3	150	50		7.12 2.05	7.7	blk	XLR	$306.00	Hand-held vocal mic with extended frequency response, switchable low-frequency roll-off filter and special element.
BK-1	cond	card	50-18k 3	150	50		7.5 1.97	12	blk	XLR	$185.00	Hand-held electret condenser mic offering condenser sound and high performance.
RE45 N/D	dyn	card	150-12 3	600	50	135 1	11.5 1.87	7.5	blk	XLR	$375.00	Short shotgun mic designed for hand-held field applications requiring ruggedness and reliability.
RE20	dyn	card	45-18k 3	150	57		8.5 2.3	26	fawn beige	XLR	$545.00	Wide response studio mic for demanding recording, reinforcement and broadcast applications.
PL10	dyn	super card	75-15k 3	150	55.8		5.3 1.7	11	gray	XLR	$368.00	Variable instrument mic for recording and reinforcement applications.
PL80	dyn	super card	60-17k 3	150	56		7.5 2	12	gray	XLR	$219.00	Hand held vocal mic designed for high gain before feedback, low handling noise, and smooth frequency response.
FOSTEX CORPORATION OF AMERICA —See our ad on page 11												
M20RP	dyn	m/s stereo	40-18k	600	55		7.75 6.75	11	blk alum	5pin	$695.00	Can be used for studio or broadcast.
M11RP	dyn	card	40-18k	600	55		9.65 2.76		blk alum	XLR	$595.00	Studio and/or broadcast mic.
M88RP	dyn	bi dir	40-18k	250	56		5.35 2.05		blk alum	XLR	$650.00	Studio vocal and/or instrument mic.
M85RP	dyn	super hyper card	50-12k	250	60		6.42 2.05		gray alum	XLR	$395.00	Noise cancelling, near field mic.
GOTHAM AUDIO CORPORATION (NEUMANN)												
SM-69 FET	cond	2X var	40-16k	200		123 .5	10.2 1.9	20	dark matte nickel	spec	$3840.00	Concert hall standard. m-s/x-y stereo mic.
KM-84	cond	card	40-20k	200		120	4.3 .83	3	blk matte nickel	XLR	$435.00	Uniform off-axis response ± 135 degrees. Same quality as the SM69.
KMS-84	cond	card	40-18k	150		138	7 1.6	9	blk matte nickel	XLR	$1140.00	Neumann's only "live" performance mic.
TLM170	cond	five	40-18k	150		140 .5	6 2.4	22	dark matte nickel	XLR	$1750.00	Most advanced Neumann studio mic. Transformerless.
KMR-82	cond	lobe X1	40-20k	150		128 .5	15.5 .82	8.8	blk matte nickel	XLR	$1080.00	Shotgun type with low off-axis coloration.
RSM190	cond	hyper card	40-18k	50		134 .5	8.4 1.2	10.5	dark matte	XLR	$2445.00	Transformerless. m-s/x-y stereo short shotgun with active matrix.
U-89	cond	five	40-18k	150		122 .5	7.3 1.8	14	dark matte/or nickel	XLR	$1700.00	For on-air broadcasting, narration, voice over and film scoring.
U-87	cond	omni card	40-16k	200		117 .5	7.87 2.2	17.7	dark matte/or nickel	XLR	$1875.00	Improved studio standard mic. 10 dB greater output than previous U-87.
HM ELECTRONICS, INC.												
RM77	elec	card	150-15 2	200	72		7.4 2	4	silv alum	XLR	$144.00	Built-in adjustable reverb, mic mute switch.
HM58	dyn	card	80-14k 2	200	75		6.5 2	6	non glare	XLR	$164.00	Mic mute switch, perfectly balanced for live use.
EM43	elec	omni	20-20k	2.2k	63		8 .3	1	blk	4pin	$70.00	Designed to work in radio frequency environments on wireless microphone transmitters.
PASO SOUND PRODUCTS												
M-501	dyn	card	50-15k	250	20		6.25 1.75	15	gray charc	XLR	$90.00	Sharp carrying cardioid pattern. All models feature an electroplated, non-reflective finish.
M-601	dyn	card	50-15k	250	20		6.5 1.75	9	gray charc	XLR	$104.00	Unique dual shock mount system practically eliminates all handling noise. Pop filter, carrying case and 18-foot cable.
M-701	dyn	card	40-16k	250	20		6.5 2	11	gray charc	XLR	$134.00	
M-800	dyn	card	40-18k	250	20		6.5 2	11	gray charc	XLR	$160.00	
PEAVEY ELECTRONICS												
PVM48	elec	card	60-20k	250	57		5.75 1.437	7.5	gray	XLR	$219.00	Brazed steel mesh screen. Improved transient response. Reduced handling noise. Increased gain before feedback.
PVM-380N	neo dyn	card	50-16k	300	52		5.75 1.437	14	blk	XLR	$199.00	Titanium and neodynium element. High sensitivity.
PVM-38	dyn	card	50-16k	300	56		5.75 1.437	7	gray	XLR	$199.00	Hum compensation. Internal pop filter to minimize wind noise and close-up vocal effects.
PVM-45	dyn	hyper card	40-16k	300	56		5.75 1.437	7	gray	XLR	$199.00	Tight polar pattern to maximize off-axis signal rejection.
PV	dyn	card	50-14k	500	59		6.5 2.25	8	gray	XLR	$99.50	On/off switch. Low impedance. Rugged aluminum housing.

Model	Type	Patterns	Frequency Response dB, +-dB	Impedance, ohms	Sensitivity, 1 kHz, dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
RAMSA/PANASONIC —See our ad on page 9												
WM-S1	elec	card	50-18k	600	42	148	1.312 .5	.57	blk	XLR	$199.00	High SPL miniature mic for close mic'ing applications. Designed for unique mounting systems such as for perc. etc.
WMS2	elec	card	120- 15k	250	56	138	1.312 .5	.57	blk	XLR	$160.00	High SPL miniature mic for phantom or 3-volt battery capability. For use with horn-bell mounting and percussion.
WMS5	elec	card	70-16k	600	52	158	1.312 .5	.57	blk	XLR	$270.00	High SPL miniature mic. For tom-toms, snare, percussion. Its transient response is great for triggering digital samples.
WMS10	elec	card	120- 15k	250	56	138	1.25 .5	3.2	blk	XLR	$210.00	Same as WMS2 in headset version for vocal applications.
SENNHEISER ELECTRONIC CORP.												
MD518	dyn	card	50-15k	200	62				blk nickel	XLR	$219.00	Lightweight, high SPL handheld. For stage vocals.
MD409	dyn	card	50-15k	200	62			6.5	blk nickel	XLR	$269.00	High proximity, high SPL drum microphone.
MD421	dyn	card	30-17k	200	60			14	blk plas	XLR	$399.00	All around flexibility, high SPL microphone.
MD431	dyn	super card	40-16k	200	61			8.8	blk enamel	XLR	$429.00	Constant polar pattern with frequency range to insure against premature feedback.
SHURE BROTHERS INC. —See our ad on page 19												
SM91	cond	hemi card	20-20k	150	45	144 .1	5.093 3.75	9.3	blk matte steel	XLR	$300.00	Optimal for piano, kick drum, stage productions. External preamp has 12 dB/octave rolloff sw. Battery or phantom power.
SM94LC	cond	card	40-16k	150	48	141 1	7.5 1.093	8.8	gray steel & brass	XLR	$250.00	Accepts virtually any phantom power source up to 52 volts dc or operates from 1.5 volt AA battery. Ideal for sampling applications and instrument mic'ing.
SM98	cond	card	40-20k	150	54	153 .1	1.25 .468	.4	blk matte brass	XLR	$250.00	Full-range response in miniature sized unit. Many optional accessories include: drum mount kit, Keen clamps for bell mounting on horns, overhead hanging adapt., pop filter etc.
SM99	cond	super card	80-20k	150	48.5	130	1.25 .437	5.8	blk matte brass	XLR	$240.00	Miniature gooseneck mic with lo-noise preamp built into base. Full RFI protection, 102 dB dynamic range, excellent gain before feedback. Locking mounting flange, windscreen supplied.
SM7	dyn	card	40-16k	150	57		5.843 7.531 3.781	27	dark gray alum	XLR	$542.00	Independently switchable bass rolloff and presence boost switches, internal air suspension shock isolation, on-board pop/blast filter. Heavy-gauge storage/carrying case included.
SM81LC	cond	card	20-20k	150	40.5	146 1	8.343 .937	8	stl champ	XLR	$367.00	Lockable 10 dB attenuator, 3-pos. bass rolloff switch. Low noise, high output clipping level. Pop filter, swivel adapter
SM84	cond	super card	80-20k	150	46	129 1	1.031 .437	1.6	blk matte brass	XLR	$300.00	Lavalier design, fully field serviceable. Chest resonance dip filter gives natural response. Windscreen, multiple mounting options supplied.
SM87	cond	super card	50-18k	150	49	142 1	7.562 1.937	6.3	gray alum	XLR	$329.00	Provides excellent isolation and gain before feedback. Integral wind/pop filter, shock mounted cartridge, phantom power.
SONY COMMUNICATIONS PRODUCTS COMPANY —See our ad on page 13												
F720/ 730	dyn	uni dir	50-11k	300	60		6.3 .25	9.2	blk alum	XLR	$124.95	Features a shock mount which protects the capsule from external noise and vibration. Excellent mid and high freq.
ECM672	cond	uni dir	50-16k	250	42	114	11.97 .94	8.1	blk cortex	XLR	$475.00	Short shotgun designed for use on compact video cameras, such as Sony's DXC series. Lightweight made of polyester film.
C74	cond	uni dir	40-16k	250	38	126	16.81 .98	12.6	dark metlc gray	XLR	$860.00	Shotgun-type mic, designed for the theater and performing arts as well as for engineering. Wide response and clean, crisp transient response.
C48	cond	omni card fig. 8	30-16k	150	41	128	9.02 2.13	20	satin color coating	XLR	$995.00	An ideal mic for critical recording applications. Directivity options are switch selectable, allows optimum sound pick-up in diverse settings and with varied renditions.
C535P/ 536P	cond	uni dir	30-16k	200	41	138	6.06 .83	5.1	satn nickel	XLR	$487.00	Designed for multi-microphone music recording. Incorporates high performance transformer for exceptional sound quality.
ECM77S /77B	cond	omni dir	40-10k	150	52	120	.49 .22	1	satn nickel/ blk	XLR	$320.00	Ideal for television broadcasting and stage applications. Ceramic back plate assures audibly superior sound quality.
ECM33F	cond	uni dir	20-20k	250	54	130	6.93 1.06	6.6	satn nickel	XLR	$241.00	Designed for multiple applications, features 2-way power and pad and low-cut switches. Neoprene suspension isolates mic.
C76	cond	uni	40-16k	250	38	126	26.69 .98	1	dark metlc gray	XLR	$995.00	Short shotgun-type designed for the theater and performng arts as well for engineering. Features a RF tuned circuit which helps to assure low noise.
TASCAM/TEAC PRO DIVISION —See our ad on Cover III												
PE50	elec cond	card	20-20k	200	67	127	7.6			1/4	$75.00	
PE80	elec cond	card	20-20k	200	67	127	7.6			XLR	$125.00	
PE125	elec cond	card	20-20k	200	67	127	7.6			XLR	$150.00	Utilizes phantom power.
PE150	elec cond	card	20-20k	200	76	127	7.6			XLR	$175.00	
PE250	dyn	movg coil	20-20k	250	73	150	7			XLR	$275.00	
MC701G	dyn	uni	20-20k	600	70	127	12			XLR	$125.00	
TELEX COMMUNICATIONS, INC. —See our ad on Cover II												
TE10	cond	card	30-20k	150- 200	75	140		7.4	blk matte	XLR	$183.00	Bright natural sound. Element is suspended by flexible fingers which isolate it from shock and vibration.
TD11	dyn	card	50-16k	100- 250	77			9.2	blk matte	XLR	$163.00	Low distortion mic with tight pattern. Die cast case with reinforced steel mesh wind screen and multi-stage pop filters.
LM100	cond	20-20k	150	74		.75	1 .437		blk matte	XLR	$220.00	Lapel mic system includes 3 foot cable and in-line power supply. Supplied with 3 styles of mounting clips.
LM300	cond	card	100-15k	150	82	.687	1 .312		blk matte	XLR	$220.00	Lapel mic system includes 3 foot cable and in-line power supply. Supplied with 3 styles of mounting clips.

Model	Type	Patterns	Frequency Response dB, ± dB	Impedance, ohms	Sensitivity, 1 kHz, dBm	Sound Pressure Level, % Distortion	Dimensions, L,D,W	Weight, oz.	Finish	Connector	Price, $	Features
YAMAHA MUSIC CORP., USA (PROFESSIONAL AUDIO DIVISION)												
MH100	elec	card	10-10k	1.6k	70					1/4	$49.00	A headphone set and mic are combined into one compact unit. Headphones have lightweight pads that are easy on the ears.
MZ101	dyn	card	40-17k	250	76		6.5 1.875		brwn metlc	XLR	$135.00	Suited for vocal use. Well defined midrange and high end. Excellent feedback rejection and 3-point suspension system.
MZ102 BE	dyn	card	40-18k	250	76		6.5 1.875		brwn metlc	XLR	$190.00	Suited for vocal use. Very wide range with deep lower mid-range quality. Beryllium diaphragm, gold plated connectors.
MZ103 BE	dyn	card	40-18k	250	76		6.062 1.875		gray metlc	XLR	$235.00	Suited for vocal use. Resists off-axis sound for feedback protection. 3-point push-pull suspension, pouch included.
MZ104	dyn	card	30-17k	250	77		7 1.4		brwn metlc	XLR	$145.00	Suited for instruments. Excellent low end response with lowered sensitivity which avoids high SPL overload.
MZ105 BE	dyn	card	40-18k	250	77		6.062 1.44		brwn metlc	XLR	$200.00	Suited for instruments. Designed to avoid unwanted bass boost with close-mic'ed instruments. Excellent feedback rejection.
MZ106S	dyn	card	40-18k	250	77		6.75 1.875		dark gray	XLR	$140.00	Ideal for vocal use. On/off switch for easy mute. Gold-plated connectors, cable and soft pouch included.
MZ204	dyn	card	20-18k	200	76		7.25 2		dark gray	XLR	$315.00	Suited for vocal use. excellent high frequency response. excellent feedback rejection. Foam lined hard case included.
MZ204	dyn	card	20-18k	250	77		5.5 1.4		dark gray	XLR	$295.00	Designed for bass drums and floor toms. Tight transient response, high resistance to feedback. Includes foam lined case.
MZ205 BE	dyn	card	40-18k	250	77		4.33 1.33		dark gray	XLR	$295.00	Compact. Well suited for snare drum and toms. Tight transient response, excellent feedback rejection. Includes hard case.

WIRELESS MICROPHONES

AKG ACOUSTICS, INC.

WMS 185 modular system provides a flexible group of products which can be configured to address diverse applications. Five AKG microphone heads can be combined with a standard transmitter body (T185N). In addition, an adapter (A85) is available to interface virtually any audio source to the transmitter body; this includes microphones or instruments. The transmitter and receivers operate in the 174 to 216 MHz frequency band, with either true diversity (SR-185.11) or non-diversity (SR-185.10) receivers available.

Price:

$3000.00 to $4000.00 depending on system

HM ELECTRONICS, INC.

System 50: 150-174 MHz or 50TV: 174-216 MHz is designed expressly for the church, theater, professional entertainment and broadcast markets. Features include NRX-II noise reduction circuitry, dual-frequency selection on the body-pac transmitter, state-of-the-art switching diversity receiver, TX550 or TX550TV transmitter, RX520 receiver, TA4F connector, a.c. adapter, detachable belt clip and antennas.

Price:

$1095.00

System 55 and 55TV hand held systems (150-174 MHz and 174-216 MHz respectively) feature NRX-II noise reduction for crisp, clear audio. A new RF link greatly improves the capture ratio for dropout-free performance. The specially designed hand held has a mic-mute switch with lockout, auto-lock for the power switch and low battery LED indicator. System 55 comes standard with HME's state-of-the-art switching diversity receiver and choice of 4 different mic elements. Also included is a.c. adapter, MC15 mic clamp, and vinyl bag for transmitter and antennas.

Price:

$1110.00

System 515 body-pac includes RX522 receiver, TX550 body-pac, transmitter, reversible belt clip, a.c. adapter with locking clip, antennas and mic connector (TA4F).

Price:

$770.00

System 525 hand held includes RX522 receiver, TX555 hand held transmitter with HM58 capsule, a.c. adapter with locking clip, antennas, mic clamp and vinyl bag.

Price:

$770.00

NADY SYSTEMS, INC.

101 wireless system operates on a choice of five channels in the VHF high band between 170 and 216 MHz. Companding circuitry gives this system a dynamic range of 120 dB. Audio frequency response is 25-20 kHz ±3dB, with an operating range of up to 1500 feet. The system is available with (LT) lavalier, (HT) hand held and (GT) instrument transmitters. The lavalier system includes transmitter bodypack with attached lavalier, and 101 VHF wireless receiver with power supply. Hand held systems include hand held transmitter and 101 VHF wireless receiver with power supply. Instrument (GT) systems include bodypack transmitter and 101 VHF receiver with power supply.

Price:

$279.95 (lavalier)

$299.95 (hand held)

$249.95 (instrument)

201 system has two complete front ends for true diversity reception and operates on a choice of five channels in the VHF high band between 170 and 216 MHz. Companding circuitry gives a dynamic range of 120 dB. Audio frequency response is 25-20 kHz ±3dB, with an operating range of up to 1500 feet. The system is available with (LT) lavalier, (HT) hand held and (GT) instrument transmitters. The lavalier system includes transmitter bodypack with attached lavalier microphone, and 201 VHF wireless receiver with power supply. The (HT) hand held systems include hand held wireless receiver microphone/transmitter and 201 VHF wireless receiver with power supply. The (GT) instrument system includes wireless bodypack transmitter, VHF receiver with power supply.

Price:

$389.95 (lavalier)

$409.95 (hand held)

$349.95 (instrument)

650 system has two complete front ends for true diversity reception and operates on a choice of ten channels on the VHF high band between 151 and 216 MHz and has a dynamic range of 210 dB. Audio frequency response is 25-20 kHz ±3dB with an operating range of up to 1500 feet. The system is available with lavalier, hand held and instrument transmitter. The lavalier system includes transmitter bodypack with mini XLR connector and 650 wireless receiver with power supply. The hand held systems include hand held microphone/transmitter and 650 wireless receiver. The instrument systems include wireless bodypack transmitter and 650 VHF receiver.

Price:

$659.00 (lavalier)

$659.00 (hand held)

$639.00 (instrument)

1200 system has complete front ends for true diversity reception for drop-out free performance and operates on a choice of ten channels on the VHF high band between 151 MHz to 216 MHz. This system has a dynamic range of 120 dB. Audio frequency response is 25-20 kHz ±3dB with an operating range of up to 1500 feet. The system is available with lavalier, hand held and instrument transmitters. The lavalier system includes transmitter bodypack with mini XLR connector, and 1200 VHF wireless receiver. The hand held systems include hand held microphone/transmitter and 1200 VHF wireless receiver. The instrument systems include wireless bodypack transmitter and 1200 VHF receiver.

Price:

$1,599.00 (lavalier)

$1,699.00 (hand held)

$1,599.00 (instrument)

PASO SOUND PRODUCTS, INC.

MA-25/R8 is a single channel hand held true diversity VHF system. The MA-25 mic/transmitter features a dynamic cardioid mic element, adjustable gain control, separate audio and RF on/off switches, low cut filter, integral pop filter, and durable alloy case. There are 10 operating frequencies available from 174 MHz to 200 MHz. Frequency response is 50-17 kHz ±1 dB. R/8 true diversity VHF receiver has 2 independent VHF receivers. The microprocessor comparator circuit selects signal from both receivers at an audio level with the widest strength to noise ratio and is forwarded to the system expander circuit then to the pre-amp.

Price:

$1478.00

MA-23/R8 single channel true diversity VHF system with combination hand held/lapel microphone transmitter pack. The mic/transmitter features adjustable gain, separate audio and RF on/off switches, belt clip, electret microphone mic connector: 3 pin pro. Miniature. There are 10 operating frequencies available from 174 MHz to 200 MHz. Audio frequency response is 50-15 kHz ±1 dB. R/8 receiver features are same as above.

Price:

$1380.00

MA-25/R7 is a single channel hand held VHF system. Transmitter features and specifications are same as above. R/7 VHF receiver features LED audio and RF signal strength meters, adjustable squelch, line level output, balanced low impedance output, rack mount kit and road case are included. a.c. or 12 volt d.c operable.

Price:

$998.00

MA-23/R7 is a single channel VHF system with combination hand held tie clasp/lavalier mic transmitter. Specs and features for both units are listed above.

Price:

$900.00

PEAVEY ELECTRONICS

Wireless Performer has diversity expandable capability. Features include neo-dynamic high output capsule, high band long range operation, an 80-15 kHz frequency response, THD of less than 0.5 percent, a dynamic range of better than 80 dB (companded), and 6 operational frequencies.

Price:

$699.50

SAMSON TECHNOLOGIES CORPORATION

Stage II series is a low cost, non-diversity wireless system which features dbx noise reduction. A variety of lavalier and hand-held microphones are available.

Price:

from $289.00

Stage 22 series is a low cost, true diversity wireless system featuring dbx noise reduction. SR-22 receiver has optional 19-inch rack mount kit.
Price:
from $399.00
Concert TD series is a professional true diversity wireless system featuring dbx noise reduction. CR-2 receiver is rack mount type with balanced XLR output, remote antennas and LED ladders for both RF signal strength and audio.
Price:
from $950.00
Broadcast STD series is a professional true diversity wireless system featuring a 10-channel synthesized receiver and a 10-channel synthesized transmitter. Receiver has the ability to scan frequencies for the clearest channel.
Price:
from $1695.00

SENNHEISER ELECTRONIC CORPORATION
VHF 1B consists of SK 20R body pac transmitter and EK 20R body pac receiver. All metal construction, tuned antennas, unique d.c-d.c. power supply for constant range performance. Includes MKE 2-omnidirectional lavalier.
Price:
$3515.00
VHF 2B consists of SK 20R body pac transmitter and EM 2003 diversity receiver. All metal transmitter, tuned antennas, unique d.c-d.c. power supply for constant range performance. Includes MKE 2 omnidirectional lavalier, two 25-foot antenna cables and receiver antennas.
Price:
$3905.00
VHF 2H consists of SKM 4031 hand held transmitter and EM 2003 diversity receiver. Acoustic characteristics of the MD 431 stage microphone, unique d.c-d.c. converter for constant range performance. Includes tuned antennas, two 25-foot antenna cables.
Price:
$3330.00
UHF 2H is same as VHF 2H, but operating on UHF carrier frequencies.
Price:
$6430.00

SHURE BROTHERS INC. —See our ad on page 19
W1020S wireless body-pack system consists of one each W10BT body-pack transmitter, W20R receiver, and WL83 (omnidirectional) or WL84 (super-cardioid) lavalier mic. System features full-range frequency response. Receiver features 9-pole linear-phase filters for high selectivity, low distortion. Lightweight, compact W10BT transmitter accommodates virtually any input source, from low-level, low impedance mics to high-level, high impedance electric guitars. Transmitter also features separate power on-off switches to permit non-pop muting, and will operate up to 8 hours on a 9 volt alkaline battery.
Price:
$1200.00
W1025S Diversiphase wireless body-pack system consists of W10BT body-pack transmitter, WL83 or WL84 lavalier mic and W25DR diversiphase receiver. The W25DR monitors signals from two antennas, locks them in-phase to prevent multi-path cancellation, then adds the signals together to provide maximum gain. Typical dynamic range is 98 dB. The transmitter accepts either mic or instrument input levels, while the receiver is switchable for either mic or line-level output. Computer selected frequencies allow 12 or more W1025S systems to operate at one location without interference.
Price:
$1700.00
W1520 hand held wireless system consists of one wireless mic and one receiver. Both transmitter models deliver the total performance and reliability of the cabled versions of these vocal mics: W15HT/58 features the SM58 cardioid dynamic mic, while the W15HT/87 has the SM87 super-cardioid condenser. Mic heads are interchangeable. High efficiency circuitry provides up to 12 hours of peak performance with a 9 volt alkaline battery. Other features include audio mute and power switches, battery indicator.
Price:
$1450.00 (W1520/58)
$1600.00 (W1520/87)

W1525 Diversiphase hand held wireless system is a full-diversity system consisting of one hand held mic and one Diversiphase receiver. The W25DR receiver maximizes RF gain by locking dual-antenna signals in-phase and adding them together. Both mics feature 138 dB maximum SPL, power and audio switches, and a fully shielded quartz-locked transmitter. Mirror-image companding circuitry. User has a choice of 15 computer-selected frequencies. Travelling frequencies and custom frequencies are both available.

Price:

$1950.00 (W1525/58)

$2100.00 (W1525/87)

SONY COMMUNICATIONS PRODUCTS COMPANY —See our ad on page 13

WRT-410 VHF synthesized unit incorporates a newly developed uni-directional dynamic element for vocal recording. The mic features a frequency response ranging form 70 to 15 kHz, a signal-to-noise ratio of greater than 60 dB (at ±1.9 kHz deviation) and maximum sound pressure input level of 140 dB SPL (with 1 percent distortion).

Price:

$1100.00

WRT-420 VHF synthesized transmitter is a concealable body pack transmitter. Its input level control provides optimum performance with any mic, while the mic on/off switch can be used effectively for eliminating undesirable noise. Features include a frequency response of 100 to 15 kHz, signal-to-noise ratio of more than 60 dB and total harmonic distortion of less than 1 percent.

Price:

$1100.00

WRR-410 VHF synthesized tuner enables tuning to the full range of 168 channels from 174.6 to 215.4 MHz in the VHF TV band. The tuner features a muting level control and LEDs for indicating the RF level, AF level and battery condition.

Price:

$1250.00

WRR-420 VHF synthesized diversity tuner comes complete with two antennas for space diversity reception, which is most effective as a solution to multipath reflection drop-outs. Two LEDs indicate the RF level to each tuner.

Price:

$2100.00

WRT-27A UHF transmitter is a compact, lightweight system designed for operation in the 900 MHz UHF band. Its advanced circuitry comprises a crystal-controlled oscillator and double-tuned coaxial filter for stable, powerful signal transmission. The frequency response is 100 to 15 kHz and the signal-to-noise ratio is better than 50 dB (±2.4 kHz deviation, at 1 kHz).

Price:

$1445.00

WRR-27 UHF portable tuner features an integral whip antenna capable of yielding adequate reception in any situation. Other features include a selectivity of better than 65 dB, output impedance of 600 ohms and antenna input impedance of 50 ohms.

Price:

$2700.00

WRT-67 UHF system features a uni-directional dynamic capsule especially suitable for vocal recording. Frequency response is 70-15 kHz and the inherent noise is less than 44 dB.

Price:

$1445.00

WRR-37 UHF diversity tuner incorporates two double-superheterodyne tuner circuits plus a diversity circuit to eliminate the problem of signal drop-outs. The specially developed C-MOS analog switch in the diversity circuit assures silent, high-speed switching. Frequency response is 40-15 kHz and output impedance is 600 ohms.

Price:

$4875.00

TELEX COMMUNICATIONS, INC. —See our ad on Cover II

HT400 is a two channel system with integral transmitter and antenna. Includes on/off switches for both audio and mic and a low-battery warning LED. The HT-100 series includes models with a Telex TE-10 condenser, a Shure SM-87 condenser and a Shure SM-58 dynamic head configuration. All of the above heads are interchangeable and can be attached in just seconds.

Price:

$885.00 to $1095.00 (depending on head)

HT-100 is a single channel system with integral transmitter and antenna. Includes on/off switches for both audio and mic. The HT-100 series includes models with Telex TE-10 condenser, Telex TD-11 dynamic, Shure SM-58 dynamic and Shure SM-87 condenser heads.

Price:

$370.00 to $620.00

VEGA (A MARK IV COMPANY)

R-42/T-87 Pro Plus hand held system consists of the R-42 Pro Plus true-dual-diversity receiver and the T-87 Pro Plus transmitter. The system features 108 dB typical signal-to-noise ratio, ultra low noise and wide RF dynamic range. Highest adjacent-channel rejection is achieved with 16 poles of IF filtering. The T-87 features a Shure SM87 condenser element and a contoured black case with internal dipole antenna.

Price:

$3,933.00

R-33/T-86 Pro Plus portable system consists of the R-33 Pro Plus miniature portable receiver and the T-86 Pro Plus portable transmitter. The R-33 receiver is ideal for mounting on the side of a camera. Its preamplifier transistor provides improved sensitivity for excellent system range and intermodulation performance. Other features include a high signal-to-noise ratio and a Shure SM90 condenser element with omnidirectional pattern.

Price:

$2,297.00

67B/77DII Pro Plus portable system consists of the 67B Pro Plus portable true-dual-diversity receiver and the 77DII Pro Plus bodypack transmitter. The 67B operates on four internal 9-volt alkaline batteries, and has external power capability for field and portable use from a 12-volt camera belt pack or other +10.5 to 18-volt d.c. source. The system features a high signal-to-noise ratio and wide dynamic range in excess of 100 dB. Interference-free audio is achieved with IF selectivity and adjacent-channel rejection via a precision LC network and 11 poles of IF filtering. Other features include battery status and peak-deviation metering.

Price:

$2381.00

Traveler portable system consists of the 66B portable receiver and either the T-37 Pro bodypack transmitter or the T-36 Pro hand held transmitter (with Electro-Voice BK-1 Black Knight element). Designed for "on location" and portable use, the 66B is extremely sensitive, highly selective and very stable. System ranges up to 1200 feet.

Price:

$1599.00 (1-B bodypack)

$1879.00 (1-HE hand held)

$1969.00 (1-HS hand held)

Pro 2-HE hand held diversity system consists of the R-32 Pro true-dual-diversity receiver and the T-36 Pro hand held transmitter. The system operates on any crystal-controlled frequency from 150 MHz to 216 MHz, at a range of up to 1200 feet. Two LED bargraph displays on the front panel of the R-32 show selected RF signal level and audio output level. Eleven poles of IF filtering provide adjacent-channel rejection. The T-36 uses the Electro-Voice BK-1 Black Knight condenser element.

Price:

$1998.00

Pro 1-B bodypack system consists of the R-31B Pro non-diversity receiver and the T-37 Pro bodypack transmitter. The R-31B receiver has two LED bargraph displays (RF and audio). The T-37 bodypack transmitter accepts virtually all electret lavalier mics, which plug into the transmitter's miniature four-pin XLR mic connector. Recessed control switches can easily be operated by feel. An LED indicator assures optimum microphone gain setting. The system has an excellent signal-to-noise ratio, wide dynamic range and most natural sound.

Price:

$1439.00

Ranger 2 true diversity systems have a high signal-to-noise ratio, wide dynamic range, and clean, natural sound. The systems operate on VHF high-band frequencies for clear, reliable audio. Eight bodypack and hand held system configurations are available, including Ranger 1 systems for applications not requiring diversity.

Price:

$925.00 to $1315.00

Reporter portable systems are ideal for all engineering and on-location applications, but can also double as a capable studio system. Available as either a bodypack or a hand held system, the Reporter features CVX audio processing for clean, crisp sound, high signal-to-noise ratio, and wide dynamic range. The equipment operates on any crystal-controlled frequency between 150 MHz to 216 MHz, with a range of up to 1000 feet. The Reporter 1-B consists of the T-23 bodypack transmitter and the R-26 portable receiver.

Price:

$1299.00

STUDIO ACCESSORIES

CANARE CABLE INC.

V-Series/Multi-Channel 75 ohm video coax cable is available in 3, 4, and 5 channel versions (R,G,B). The wire is flexible, has low loss/high definition, PVC jacket, 96 percent copper braid, and is used for computer graphics, HDTV, digital VTR and projection TV.

BCJ-JRV connector has recessed 75 ohm BNC panel mount, VSWR .1 to 2 GHz, and beryllium copper inner contacts.

CONNECTRONICS CORPORATION —See our ad on page 40

Musiflex is microphone cable with a high flexibility PVC jacket available in 11 different colors. The outside diameter is approximately 1/4-inch. Features include two 22 gauge 30 strand plain copper wire conductors insulated with LD polyethylene, 100 percent shield manufactured from carbon loaded conductive thermoplastic terminated by means of a further 22 gauge 30 strand plain copper wire. It is ideal for most forms of microphone cabling including stage, broadcast, studio and contracting applications.

Starquad is microphone cable jacketed in high flexibility black PVC with four 24 gauge 19 strand conductors insulated with LD polyethylene in four different colors. The outside diameter is approximately 1/4-inch. Shield is a rusen type double spiral stranded copper shield, each of the two shields 98 percent nominal coverage. Suitable design for microphone cabling in high noise areas.

Studiflex is cable jacketed in colored PVC. Conductors are 24 gauge 7 strand tinned copper wire with a PVC insulation. Shielding is 100 percent using carbon loaded conductive thermoplastic which is terminated by a further 24 gauge drain wire. The wire is available in 1 pair format in 8 different colors, and also available in 4, 8, 16 and 24 pair formats each pair individually shielded and jacketed in a color coded jacket. Used extensively for studio and installation wiring.

Multipair is foil shielded cable with PVC jacket enclosing foil shielded pairs. Each conductor is 24 gauge and insulated using conventional color coded pairs. Shield is from aluminum foil available in 1, 3, 6, 12, 16, 20, 28 and 32 pair formats with the outside jacket color coded. Used for snake and installation applications.

Phonoflex is PVC jacketed phono cable with 24 gauge conductor insulated with foam polyethylene. Shielding is from carbon loaded thermoplastic terminated with 24 gauge drain wires. The wire is available in 4 different colors and is ideal for use with RCA phono connections or where an unbalanced patch cord or termination must be made.

Speakerflex is PVC jacketed and the conductors are color coded with PVC insulation made from plain copper wire with a high strand count to maintain high flexibility. Jacket is "filled" thereby ensuring high durability during use. Available in four versions: Speakerflex 150 (2x15 gauge 30 strand conductors), 250 (2x13 gauge 50 strand conductors), 250/4 (4x13 gauge 50 strand conductors), and 250/6 (6x13 gauge 50 strand conductors).

J Bay patch bay is a PCB card based unit based on 1/4-inch sockets either 2 or 3 pole. It can be normalized, semi-normalized or brake point at each connector. Rear panel is 1/4-inch jack or RCA phono. Cards available for "insert" purposes for use to bring sleeve insert points on mixing consoles. Two racks spaced high and standard 19-inch rack wide the units feature tie bars and facilities for smooth and easy cable exit. Finished in gun metal gray with plain and easily replaceable legend strip attachments as standard. Standard is 22 pairs of connectors.

X Bay is a patch bay system which is designed to take a wide variety of connections as options from XLR, BNC, Bantom, Professional Din, Jack etc. Selections can be made at time of purchase or the patch bay can be changes later. Available to a maximum of 32 connectors.

Utilux high power audio connector is designed as loudspeaker termination for 2 conductor loudspeaker cable. High currents are carried successfully and are rated at 30 amps at 100 volts. Easy termination to self wiping contacts, the UX connectors are hermaphroditic which means that there are no male or female parts (each connector will fit into any other connector). The system is locking and comprised of two pieces, the UX 100 (line conductor) and UX 500 (chassis mount connector).

FURMAN SOUND INC.

PB-40 series patch bays have forty point modular patch bay mounts in a single rack space. Available in three connector options, 1/4-inch phone, 1/4-inch tip-ring-sleeve, or RCA front or rear, or any custom combination to meet special needs. Each vertical pair (four connectors, two front and two rear) is on a separate PC board supplied with removeable normalling jumper(s).

Price:

$145.00 to $160.00 depending on connectors

Patch cords are highly flexible, with synthetic rubber jacketed patch cords which are 75 cm in length. Available in 1/4-inch phone, 1/4-inch tip-ring-sleeve, and RCA.

Price:

$15.00, $20.00 and $15.00 respectively per set of ten

GOTHAM AUDIO CORPORATION

Multicore cable offers a conductor resistance of =<95 ohms/km, a shielding resistance of =< 22 ohms/km, a conductor-to-conductor capacitance (at 800 Hz) of = 100 nF/km, a conductor-to-shield capacitance of =<180 nF/km, and an insulating resistance of => 1000 Mohms at 20 degrees C. Attenuation is =<1.1 dB/km, 5.3 dB/km, 6.8 dB/km at 1000, 10 k and 20 kHz respectively. The cross-talk attenuation is = 90 dB from 1 kHz to 20 kHz. This cable is available in 100 and 200 meter bulk reels and comes in a distinctive and easily identifiable array of colors.

Also offered is 2 and 3-wire microphone cable. The 3-wire cable offers a maximum conductor resistance of 93 ohms/km, a maximum screening resistance of 20 ohms/km, and a minimum insulation resistance of 18 Mohms/km. Transmission loss is <8 and 9 dB at 10k and 20 kHz respectively. Impedance is 434 and 150 ohms at 1k and 10 kHz respectively. The 2-wire mic cable specifications are available from manufacturer.

An extensive line of connectors and electrical accessories designed exclusively for Gotham's microphones (Neumann) include: XLR3F, XLR3M, XLR5F, XLR5M, DIN12F, DIN12M, DIN7F, DIN7M, LEMO2F/2M and swivel versions of the above.

Available cable material is 3-pole, 7-pole or 11-pole, all with double cage screen, triaxial cable.

MARSHALL ELECTRONICS (MOGAMI)

Mogami snake cables, 2-48 pairs available, are individually jacketed, numbered and color coded for easy installation. This is part of the Hi-Definition Series designed for sonic transparency for the highest quality digital and analog recording. Each individually-shielded pair utilizes Cross-Link polyethylene insulation which prevents shrink back during soldering. The cable also includes a drain wire within the shield of each pair for crimping operations.

Price:

range from $0.63 to $9.03 per foot (2-48 pairs)

2799 quad rack and console cable is for audio and recording systems for mic and line level. The design offers an economical method for reducing electromagnetic induction and crosstalk problems that can cause possible reduction in transparency of the sound quality. Extremely pure oxygen free copper is used for the conductors. Cross link polyethylene insulation is used for the inner conductors and flexible PVC for the outer jacket which is 0.126-inch.

Price:

$0.33 per foot in 1000 ft. rolls

Pro speaker cables are 13 gauge and come in 4, 6, and 8 pairs. Designed for remote and touring show applications, these unplated oxygen free copper cables are for bi-amp and tri-amp applications. A tough black matte outer jacket is designed to reduce light reflection during live performances.

Price:

range of $2.16 to $3.16 per foot

2435 is high definition oxygen free quad cable with excellent sonic characteristics. The quad design offers the best economical method for reducing electromagnetic induction and cross talk problems that can interfere with the transparency of the recorded sound. Cross link polyethylene insulation is used for the inner conductors which prevents shrink back during soldering. Overall diameter is 0.236-inch. Each cable has 4 conductors plus overall shield, and comes in a variety of colors.

Price:

$0.41 per foot

Bantam patch cords are designed specifically for the recording and broadcasting industry. The cable includes a number of features to enhance the sound quality and reliability. Features include quad wiring, oxygen free copper, high stranding, and high-quality nickel plating which eliminates the need for polishing or burnishing and eliminates uneven wear which causes intermittent connections. The cable comes in lengths of 1, 1.5, 2, 3, 4, 5 and 6 feet. Each cable comes with a color code identification kit. Cords are also available in colors which include black, red, green and blue.

Puroflex series of patch cords come in either gold or nickel plating and are available in various combinations of RCA, 3.5mm and 1/4-inch. Lengths are available from 1-20 feet.

Price:

range from $5.00 to $25.00

MONSTER CABLE

Prolink Series 1 microphone cable uses "MicroFiber" dielectric insulation precisely wrapped around the large conductors to greatly reduce signal distortion for vastly improved clarity, better transient response, lower "inter-transient noise" and wider dynamic range. Using a combination of heavy duty foil plus a 95 percent high density braid ensures maximum rejection of RFI and EMI interference. The cable uses "Duraflex" jacket, and is UV stabilized and resistant to abrasion and temperature extremes.

Price:

range from $68 to $300 including connectors.

Prolink Series 2 microphone cable comes from "Bandwidth Balanced" technology, using a sophisticated construction employing one large center conductor to carry the bass frequencies, surrounded by a network of six sub-groups of ultra-fine strands for the midrange and high frequencies. This cable configuration compensates for cable time domain problems and carries all the frequencies in accurate phase and time alignment. The cable also uses "MicroFiber" dielectric insulation precision wrapped around the bass conductor to greatly reduce signal cable losses for impressive response at bass frequencies: greater clarity, better transient response and wider dynamic range across the full audio spectrum. It is double-shielded with a combination of full coverage foil and a high density braid for maximum rejection of RFI and EMI.

Price:

range from $40 to $72 including connectors.

Prolink Series 3 is a high-resolution "Phase-Maintained" microphone cable featuring "Bandwidth Balanced" conductors with "MicroFiber." Each cable is double-shielded with heavy-duty foil and a 95 percent high-density braid to ensure maximum rejection of RFI and EMI interference and annoying hum and noise. The "Duraflex" jacket is rugged for tough conditions, yet is flexible and easy to wind after use. The precision XLR connectors are equipped with high-conductivity contact points and Collet-type strain relief to keep connections secure.

Price:

range from $28 to $48 including connectors.

Prolink patchbay cable has "Bandwidth Balanced" technology which corrects for the phase and amplitude distortions found in conventional microphone wire. These distortions cause "time smear" and prevent the full recreation of music harmonies, instrument tonal balance, ambience, dynamic range and frequency response. Prolink Series 2 solves these problems by using a special wire construction comprised of one large center conductor to carry the bass frequencies, surrounded by 6 networks that carry the midrange and highs in accurate "phase and time alignment." It also utilizes "MicroFiber" dielectric insulator that is precision wrapped around the bass conductors to greatly reduce signal losses for greater clarity and better transient response. Terminated with 1/4-inch TRS and TT connectors, equipment can be patched together without sonic degradation.

WIREWORKS CORPORATION

Microphone multicable is available in 3, 6, 9, 11, 15, 19, 27, 36, and 50 channel capacities. The stranding is comprised of typically 7/30 tinned copper. The insulation material is polypropylene. The jacket material is polyvinylchloride and the shielding is aluminum/polyester tape (with foil facing inward for 100 percent coverage).

Professional microphone cables appear as the C, CM, CH, CN and CP series. These series include overall diameter dimensions that range from .19-inches to .263-inches and a wide range of materials.

Multicable is also sold as complete assemblies from 3 to 50 channels that range from 10 to 250 feet in length. Prices range from $160.00 to $4,600.00 complete.

Patch bays are available and offer a choice of kits or pre assembled configurations.

Connectors include XLR (Neutrik) precision, Swiss-crafted audio connectors that are available in nickel finish with silver plated contacts, and black finish with silver or gold plated contacts.

Wireworks multipin standard is based primarily on AMP products. AMP connectors are a rectangular connector design that completely eliminates out-of-round and cross threading problems that often occur in circular connector styles. Contacts are gold-flashed to provide excellent conductivity.

ADDRESSES

AKG Acoustics, Inc
77 Selleck St
Stamford, CT 06902

Altec Lansing Corporation
10500 W. Reno
Oklahoma City, OK 73128

AMS/Calrec
PO Box 31864
Seattle, WA 98103

Audio-Technica US, Inc
1221 Commerce Dr
Stow, OH 44224

Barcus-Berry Inc
5381 Production Dr
Huntington Beach, CA 92649

Beyer Dynamic, Inc
5-05 Burns Ave
Hicksville, NY 11801

Canare Cable
832 N. Victory Blvd
Burbank, CA 91502

Carvin Corporation
1155 Industrial Ave
Escondido, CA 92025

Connectronics Corporation
652 Glenbrook Rd
Stamford, CT 06906

Countryman Associates Inc
417 Stamford
Redwood City, CA 94063

Crown International
1718 W. Mishawaka Rd
Elkhart, IN 46517

C-T Audio Marketing, Inc
3050 SW 14 Pl Suite 3
Boynton Beach, FL 33434

Electro-Voice Inc
600 Cecil St
Buchanan, MI 49107

Fostex Corporation of America
15431 Blackburn Ave
Norwalk, CA 90650

Furman Sound
30 Rich St
Greenbrae, CA 94904

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

HM Electronics, Inc
6675 Mesa Ridge Rd
San Diego, CA 92121

Marshall Electronics
PO Box 2027
Culver City, CA 90230

Monster Cable
101 Townsend Street
San Francisco, CA 94107

Nady Systems, Inc
1145 65th St
Oakland, CA 94608

Paso Sound Products
14 First St
Pelham, NY 10803

Peavey Electronics
711 A St
Meridian, MS 39301

Ramsa/Panasonic
6550 Katella Ave
Cypress, CA 90630

Samson Technologies Corporation
485-19 S Broadway
Hicksville, NY 11801

Sennheiser
6 Vista Dr
PO Box 987
Old Lyme, CT 06371

Shure Brothers Inc
222 Hartrey
Evanston, IL 60204

Sony Communication Products Company
1600 Queen Anne Rd
Teaneck, NJ 07666

Tascam/Teac Pro Division
7733 Telegraph Rd
Montebello, CA 90640

Telex Communication Inc
9600 Aldrich Ave S
Minneapolis, MN 55420

Vega (a Mark IV Company)
9900 Baldwin Pl
El Monte, CA 91731-2204

Wireworks Corporation
380 Hillside Ave
Hillside, NJ 07205

Yamaha Music Corporation
PO Box 6600
Buena Park, CA 90622

HOT LINE. . . . . to the experts.

• I have been told by some that I don't need much more that the rate amount of power for my near-field monitors. And, I've also been told by others that I should have as much as I can afford to ensure the best headroom. How do I determine exactly how much power I really need for my NFMs?

Brian Waxwirth
Waxwirth Studios
Evanston, IL

Would that an answer could be as simple as "as much as you can afford." But that answer is wrong. If you had a pair of Auratones sitting on the meter bridge of your console, you would not need a thousand watts per channel. (But you could use that much.) The rated power of a speaker system represents some maximum capability of the speaker, (usually elevated distortion, not the final destruction point.). Generally, using an amplifier that gives even five times the speaker's rated power is all you need. That would likely be a few hundred watts per channel.

What you should never have is an amplifier with no more power than the rated power of the speakers. Then you would be running the amplifier near its distortion curve-up, hence no headroom. And if you did use the kilowatt-per channel amp, you would only need to use care that you never can apply that full power to the speaker. Getting hit in the face by two speaker cones that have ripped through their moorings is not a necessary part of an audio experience. (Ed.)

• An experienced engineer friend told me that the wider the tape is, the better the sound quality. I certainly can't afford a big multi-track tape recorder and I want eight tracks. How do I decide on the tape format to buy for 8-track recording that will give me the most for my money?

L. Tarping
Denver, CO

You can get eight track decks that use one-inch tape, half-inch tape, quarter-inch tape, and even cassette tape. But there is a simple rule that will answer your question. Every reduction of track width by half is a 3-dB reduction in s/n.

It's also true that every reduction of tape speed by half is also a 3-dB reduction. A typical tape deck using the widest tracks at 15 in./sec. might have a 75 dB s/n (before the use of any noise reduction systems, of course). At the other end of the scale, the eight-track cassette deck running at 3.75 in./sec. would have a basic s/n in the 55 dB range.

Obviously you can use noise reduction systems to improve any tape deck by 20-30 dB. But think what that means to a cassette deck and also think what it means to a one-inch deck. Then start also realizing that noise reduction systems always add something to sound, so ideally, it would be nice not to need them.

Perhaps, these are not really answers to your question. But you now realize that any tape deck represents a compromise with the absolute laws. (Ed.)

Write to HOTLINE!, db Magazine, 203 Commack Road, Suite 1010, Commack, NY 11725. All letters become the property of db Magazine.

JOHN BARILLA

THE ELECTRONIC COTTAGE

DESIGN CONSIDERATIONS

PART II

●In this issue of **db**, we will complete the speculative odyssey that was begun in the July/August edition. Our goal is to continue exploring some of the most useful and cost-efficient methods of constructing (or upgrading) an electronic cottage: that such a facility might exhibit a great ease of operation, and be capable of turning out unquestionably professional product. Having divided this quest into four areas of concern, we previously dealt with the foundational considerations—*room layout* and *electricity*—in the last edition. Now let's put it all together by discussing the areas of *interconnections* and *monitoring* in the electronic cottage.

INTERCONNECTIONS

Let the reader beware from the outset: getting highest quality audio from the electronic cottage can sometimes be an arduous journey, but it is indeed possible. First, let's remind ourselves that we are working within the limitations of "unbalanced" audio, which is justly considered to be a less-than-professional standard. This stigma has almost nothing to say about the inherent quality of the device itself, but says much about what happens when we connect several devices together.

Whatever degradation in sound quality that occurs in an unbalanced system (beyond the practical S/N ratio of the mixer, outboard gear, etc.) will be primarily induced in the wiring that hooks them all together. If spurious audio (EMI or RFI) enters the signal path at any point, the unbalanced system is helpless to eliminate it. The balanced audio system, however, can factor out much of this trash even after it has invaded the system, rendering clean audio under circumstances where unbalanced systems would positively freak out. But I reiterate: it is possible to get high quality, trash-free audio from an unbalanced system. We simply need to work harder in order to get it!

If any noise does makes it beyond the shield, there is a second line of defense—the differential input.

The key features of the balanced system that gives it such a high degree of immunity to interference are:

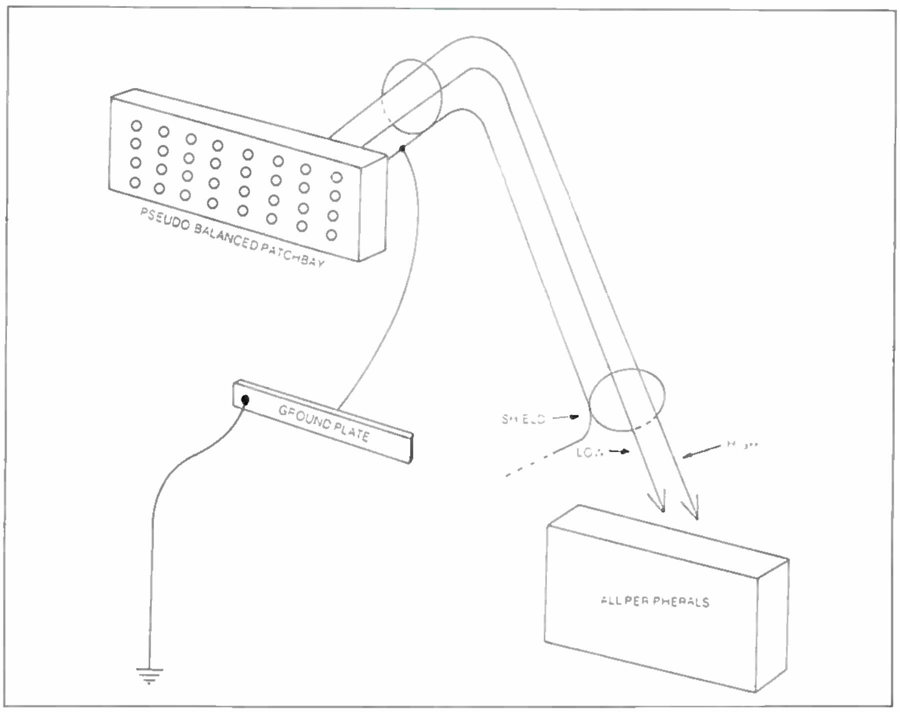
1) the differential input

2) the use of three-wire cabling (i.e., two conductors plus a shield)

3) higher nominal operating levels.

The idea here is that audio information is conducted from one device to the next down two conductors, neither of which is referenced to ground. The third lead, which is a shield (and attached to ground only at one end of the cable) is then able to intercept a good deal of extraneous noise from the environment and shunt it to ground. If any noise does makes it beyond the shield, there is a second line of defense—the differential input. It discriminates against any signal that does not bear the usual phase relationship (one side high [+], one side low [–]) at

Figure 1. The pseudo-balanced patchbay concept.



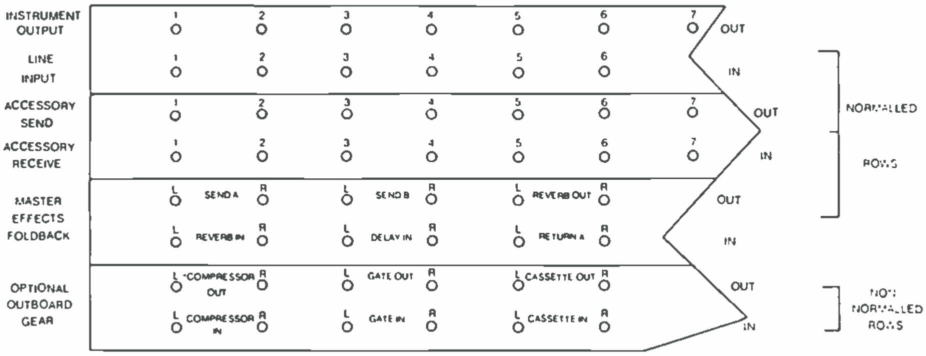


Figure 2. A sample patch bay concept

a given point in time. Recognizing only pairs of opposites, it effectively cancels any invasive signals, since they have hit both conductors at virtually the same time. This is definitely a neat trick, but for various reasons (increased cost, for one) it has been reserved for strictly pro equipment.

The higher operating level also helps out here in achieving a clean signal. Instead of using a nominal −10 dBv as most unbalanced gear does, balanced equipment sports the elevated level of +4 dBm. If extraneous sound does find it's way beyond the first two lines of defense, the elevated audio signal can make the noise seem rather unimportant in proportion to the signal, essentially masking the noise by brute force.

Now the purpose of this little side tour was not simply to extol the virtues of balanced equipment. Most electronic cottageers do not use balanced equipment. (Even if, for example, your recording console offers a +4 balanced master output, and you are recording into an unbalanced input on your mixdown machine, well good for you! You have an elevated output, but the other characteristics of a balanced system are not operating.) The point of this is to see if any of the very obvious virtues of a balanced system can be incorporated into the unbalanced studio. While it is plainly obvious that short of buying an interface amplifier for every piece of gear in the house, an unbalanced system will never function as a balanced system. But two of the three previously mentioned characteristics can actually be utilized in an unbalanced studio to improve the rejection of extraneous noise. The key to achieving this maxxed-out performance is found in a method of wiring up your studio patchbay.

Certain advantages can be gained from using a balanced type (three-wire) patch-bay in an unbalanced system. While it takes a little more forethought to integrate this into the elegant simplicity of the unbalanced (two-wire) system, in the end, it will prove a very worthwhile investment of your time.

One thing that needs to be determined before you fire up your soldering iron is the layout of your patchbay.

The theory goes something like this: Normally, in the unbalanced configuration, two-wires serve three purposes. One wire is designated as high signal carrier, the other serves as both the low signal carrier *and* shield. While this does idiot-proof the task of interconnecting equipment (neatly avoiding all ground-looping or phase problems), it also greatly compromises the unique purpose of the shielding: to intercept and divert spurious electrical currents. So if we can separate the signal carriers from the shield, it is possible to afford an extra measure of protection against invasive electricity. (Admittedly, this is not identical with the balanced system where none of the signal carriers are referenced to ground, but having a *dedicated* shield hooked into a solid ground can really make a significant improvement. Many electronic cottageers in urban areas—where RFI is extremely high—have found benefit from this technique.)

The underlying concept is very similar to the one we discussed in the last issue of db regarding electrical grounding: In order to provide the most effective drain (and avoid loops), audio shields should have a single, efficient path to mother earth. The pseudo-balanced patchbay provides a central point at which all shields can meet and be integrated into the unified system ground.

The basic procedure is as follows: Considering the patch bay as the central point and *all* devices (console, signal processors, etc.) as peripherals, wire the plugs connecting all peripherals to the patchbay, using only the red and black wires. Leave the drain wire (the shield) *detached at the device*, but make sure all drain wires are *connected at the patchbay*. All of these drain wires are then bussed together at the patchbay and solidly attached by a fat copper wire to the central grounding plate (see the previous issue of **db**).

In most cases, this will work without a hitch, but occasionally there will be a complication. For example, having attached the insertion points (accessory in/out) on my AKAI 12-12 to the patchbay in this manner, I found that when I pushed the "EQ-in" switch a ground loop was formed. It was however, easy enough to iron out this wrinkle by reattaching the shield at one side of the insertion point. (The "in" or the "out," it made no difference.) Fortunately, I was prepared for this by an experienced studio maintenance engineer, who advised that when wiring the plugs, the drain wires should be *detached*, but not *clipped*. Instead they should be insulated in shrink-wrapping and left intact within the shell of the plug, just in case some final jockeying was necessary. Any honest studio technician will tell you that due to differences in design philosophy between various manufacturers, anomalous conditions do crop up in wiring a studio that require a more empirical approach. We always need to be prepared for this eventuality.

PATCHBAY LAYOUT.

One thing that needs to be determined before you fire up your soldering iron is the layout of your patchbay. Naturally, for convenience and efficiency you will probably want to bring every piece of gear, every insertion point, up on patch. You want to be able to patch anything to anything else with just a short patch-cord. Truly, there are as many ways to achieve this goal as there are studio owners. Still there are some rules of thumb, some tried and proven conventions that make this task less confusing.

For example, the concept of *normalling*. Which devices or configurations of equipment will be most consistently used in a standardized way? Only you can answer that. And if your studio is a new one, your answer will probably be speculative and provisional. Nevertheless, you must define the scope of your studio (at least theoretically), out front, or else you will find yourself constantly re-patching really obvious signal paths. If your main thrust is synthesized sound, then it would be well to normal the outputs of the sound modules *through the patchbay* to the appropriate input modules on your recording console.

For this, you will need a patch bay that contains normalled-through patch points. Sometimes, you can assemble them yourselves choosing which pairs of points (upper and lower) will be normalled through and which will be non-normalling (where upper and lower points are isolated from each other). This non-normalling type is preferred for use as tie-lines, where optional equipment is to be patched in. Usually, a combination of these two basic types is best, but some people (of necessity or ignorance) use the common normalling patch bay for everything. In the latter case, if you get feedback from any device being normalled through to itself, you can just stick a "dummy plug" into either the upper or lower point, and the normal will be broken.

The following conventions are also useful for laying out a patchbay that can be comprehended quickly by the user. Your entire patchbay may consist of several rack mountable patch panels, each with two rows of points (of various numbers, commonly 2 pairs of 24). It has been found most efficient to designate the upper row for outputs and the lower row for inputs.

OUTPUT OVER INPUT.

If this convention is consistently followed throughout each panel, your patchbay will be a joy to use. (It will also be a lot harder to do dumb things like patching two outputs together).

Another worthy convention is to somehow approximate the signal flow in vertical organization of the patch bay. Usually, this would mean having "earliest" points in the signal flow of your recording console appear at the top panel and subsequent points (like the various insertion points) on lower rungs of the patch-bay. If this is not feasible in every case, then let it be a left to right movement first, then moving to the next lower rung. (See example) This of course, is harmonious with the Western mind. In any case, try to roughly map the signal flow of the console in the initial several rows. The next

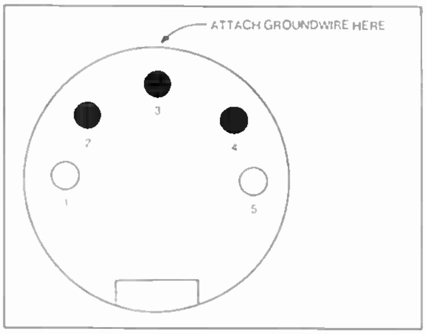


Figure 3. A do-it-yourself MIDI cable using studio hookup wire.

rows are best reserved for auxiliary send/ receive circuits and their normal destinations, (such as Send A Out normalled through to your "Usual Ambience Device"). Finally, every piece of non-normalled outboard gear, auxiliary recorders, etc. should appear on patch. It is also good to have a few blank patch points wired in, terminating in a box with 1/4-inch jacks, so that the inevitable piece of borrowed or rented gear can be used to full advantage.

PRACTICAL WIRING TIPS.

There are many lessons to be learned about studio wiring, and the only way to really learn them is by doing them. Still, there are a few sensible pointers that will save many a costly mistake. So without any pretense to comprehensiveness, I offer these few aphorisms, anecdotes and personal recommendations, based on my experience putting together an electronic cottage.

As to wiring, never underestimate how much cable you will need.

Start with a good soldering iron. A rinky-dink iron will really hold you back, because it takes so long to heat up, and temperature sometimes varies with usage. All this impedes the rhythm of flowing on solder in a productive fashion. Never underestimate how many connections you will need to make. (In my case: a twelve-track studio with a dozen or so signal processors and 144 points up on patch, required over 1500 solder joints. Happily, there was not even one cold solder joint to be found amongst them!) A good iron is therefore, a worthy investment.

Also, take some time to keep the iron clean and well tinned. A corroded tip will never heat up quite right, so when you shut down the iron, make sure there is a good fresh gob of solder on it to protect it until the next time you fire it up. If the tip eventually gets funky anyway, just dip the (hot) tip in some flux and rub it on some fine steel wool to clean it.

As to wiring, never underestimate how much cable you will need. (In my case: the above-mentioned twelve-track studio residing in a 12-ft.x 12-ft. room, required nearly 2000 feet of wire.) Understandably then, unless you are an audiophile with a rich uncle, you will have to find a cost-effective, reasonably low-capacitance cable. Now, a few years ago I had sojourned on a wiring crew for a short period of time. The crew chief used West Penn 291 exclusively at a 24-track installation, because of the need to come in on a tight budget. Some of the staff engineers spoke derisively about this, saying it was better used for wiring telephones than pro-audio facilities. (That's really a dumb statement, because the entire legacy of pro-audio comes directly from Ma Bell.) Today, West Penn 291 is considered quite respectable stuff, showing up in the wiring troughs of many professional facilities. And at current retail prices of about $70 for a 1000 ft. reel, it's a secret that's worth sharing.

Another good investment for the would-be wirer is the shrink-rap gun. It's really nothing more than glorified hair dryer that's used to apply very hot air to slightly over-sized plastic tubing, thereby shrinking it to a snug fit. It's a tool of many uses, all of them tending to help render a more stable installation. For example, drain wires on studio hookup cable are uninsulated. After all, they are required to be in constant contact with the aluminum foil shield. At termination points (like plug ends and patch bay connections, bare wire, if accidentally twisted out of position, could hit a signal bearing lead and short it out. To ensure against that kind of trouble, all drain wires should be shrink-wrapped with the appropriate size plastic tubing. Another important use for larger shrink-rap is a permanent lamination for any cable markers.

These markers, usually found in the form of a sticky-backed tape should be utilized as a numbering system for the purposes of studio documentation. Each cable should be marked at both ends with an identifying number, and then sealed up with shrink-tubing. Such scrupulous documentation will prove of immense value if any part of your system should go down or need to be updated.

It is worthy to note, that MIDI cables can also be made from standard studio hook-up cable. While MIDI cables utilize a five-pin DIN connector, only three of the leads carry any information, at this time. So for the price of some good hardware and just a few pennies for wire, you can build your own MIDI cables, cut to exact size for your installation. To do this, you simply connect the ground (drain wire) to the center pin, and the red and black to the pins on either side of center pin. Remember *not* to attach the ground to the casing as you would a standard DIN connector, but to the center pin only.

A word about cable length. We know that there are discrete limits to how far you can push a signal in an unbalanced system without noticeably degrading it. While it varies a bit with the quality of the cable you use, practically speaking, anything beyond 25 feet will result in a perceptible loss of high frequencies. So the wise person will endeavor to keep cable lengths as short as possible. But don't get too neurotic about this. You'll need to leave about 18 inches of slack at the rear of each device, so that you can pull a piece away from the rack (to test it or do maintenance, without dismantling your whole studio. If you keep your equipment centrally located, your cable runs will naturally fall in the very acceptable 8- to 10-foot range. Finally, I mentioned that there might be a way of utilizing higher operating levels as a way of keeping your system quiet. This can be approached in two ways. On an extra-ordinary level, one can have modifications done to critical pieces of gear. The master output on my AKAI 12-12 was modified to put out a 6db hotter level, simply by changing the feedback resistor on the master output amplifier. Of course, this is a possibility for other gear, as well. From the more ordinary perspective though, running all "send" circuits hot at possible, and "returns" low as possible can contribute to a better signal-to-noise ratio. Additionally, some signals processors, such as Yamaha's SPX 90, have input/output level switches giving options of –20 dB and +4dB. These sensitivity controls can often be creatively manipulated to improve the gain structure of the effects circuit and hence, the whole system benefits. Once you start thinking in this manner, noticeable improvements can be achieved.

...if you go this far, remember not to restrict the current, by sticking any resistive obstacles in its way.

MONITORING

Monitoring in the electronic cottage bears many things in common with the professional recording studio. There is the need for a clean, flat, power amplifier, accurate speakers, and solid low-resistance interconnections. But beyond this, the similarity ceases. Given that the electronic cottage work space is generally not the realization of an acoustical consultant, but the simple, mundane reality of available space, we can say that many issues the pro-studio owner will fret over, are irrelevancies to us. Not because they are unimportant, but because they are unattainable.

One area of great concern in the acoustically designed room, is accuracy in the low end. State-of-the-art studios go to great expense to achieve this goal, with varying degrees of success. But such noble hair-splitting would be a endless *megilah* in the electronic cottage. So instead, I believe, most of us would opt for establishing electronic accuracy to the point of the speakers (an easy task), and then spending whatever time it takes to establish a genuine intimacy with our environment. After all, we are the "rugged individualists" of the audio industry, aren't we?

As mentioned, accuracy out to the speakers is not hard to establish. We don't need high power or elaborate crossover schemes. A clean stereo power amp of at least 50 watts should be all you need. Remember that mixing in the EC is best done at moderate to low levels in order to keep room resonances to a minimum. The best way to work with an untuned room, is not to work with it at all. Learn to utilize near-field monitors, exclusively. It is a skill that needs to be developed if consistency in mixing is to be achieved.

The issue of monitor speakers in the electronic cottage boils down to a matter of prudent personal preference. My choice for main monitor speakers are TOA 280-MEs. They are three-way sealed enclosure type speakers, with an 8-inch woofers, tweeter, and super-tweeter. They are virtually flat from 20k down to about 90 Hz. (TOA makes a slightly larger monitor the 312-ME which is flat down to about 60 Hz.) I also installed TOA 22-MEs (very similar to the Auratone cubes) as secondaries, and as tertiary speakers, I bought Roland SRS-80s, because they were cheap and provided additional insight from the perspective of your "average" two-way speaker. How do I know what's happening on the low-end when my best monitor is flat only to 90 Hz? Well, it may sound esoteric, but by cross referencing information gleaned from all three speakers, one learns to quite accurately deduce what is going on in the low-end; even in regions below the flatness curve of the monitors. The task is initially spending a lot of time in the experimental mode.

Most of what I said thus far, is purely discretionary. But there are a few hard-core facts I would like to mention. Don't hook up your amp to speakers with anything like "normal" speaker wire. If there was one place in your system where you should go for broke, here it is. Audiophile speaker cable (like Monster and other similar brands) is really quite expensive, but from amp to speakers should be a relatively short run, and it will be worth the expense. It has been shown that power, frequency response, and phase relationships are disturbed by inferior cabling—so go the extra mile. (If finances are really tight, some fat zip cord, 12 gauge or larger, will work decently.)

And if you go this far, remember not to restrict the current, by sticking any resistive obstacles in its way. I am speaking, of course, about things like switch boxes—which are necessary contrivances if you would use multiple speakers with a single amplifier. If you pull one apart, you may find it's built with wire so thin, you could use it to darn your socks! Instead of living with this electrical bottleneck, re-wire the beast with the same wire you chose for your speakers, (or at least the fattest wire that can be soldered onto the lugs in the switch box). The difference this makes will be quite discernable, so it's worth another of evening of fun with your soldering iron.

If you are currently planning or revamping your EC, we hope this series has been helpful to you. Stay tuned for more insights about life in the Electronic Cottage. db

Hearing the Future in Dallas

From small acorn beginnings do large oaks grow. Future Audio is such an oak.

In 1982, Tony Rodriguez realized a long-standing dream; after years of attending recording industry seminars such as Full Sail Recording Workshop and SYN-AUD-CON, and operating a home studio, he built his first full-scale studio, Sierra Recording, in Fort Worth, Texas. Five years later, in 1987, Rodriguez found himself with no room for expansion and a new landlord whose attitude toward the basement studio left much to be desired. Since the Rodriguez family owns several radio stations and other businesses, a search was begun for a new building large enough to house the entire operation.

Construction began in April of 1987 on Mr. Rodriguez' new dream studio, *Future Audio*. The new facility occupies most of the lower floor of the 20,000 square foot building, with two radio station control rooms and offices on the second floor.

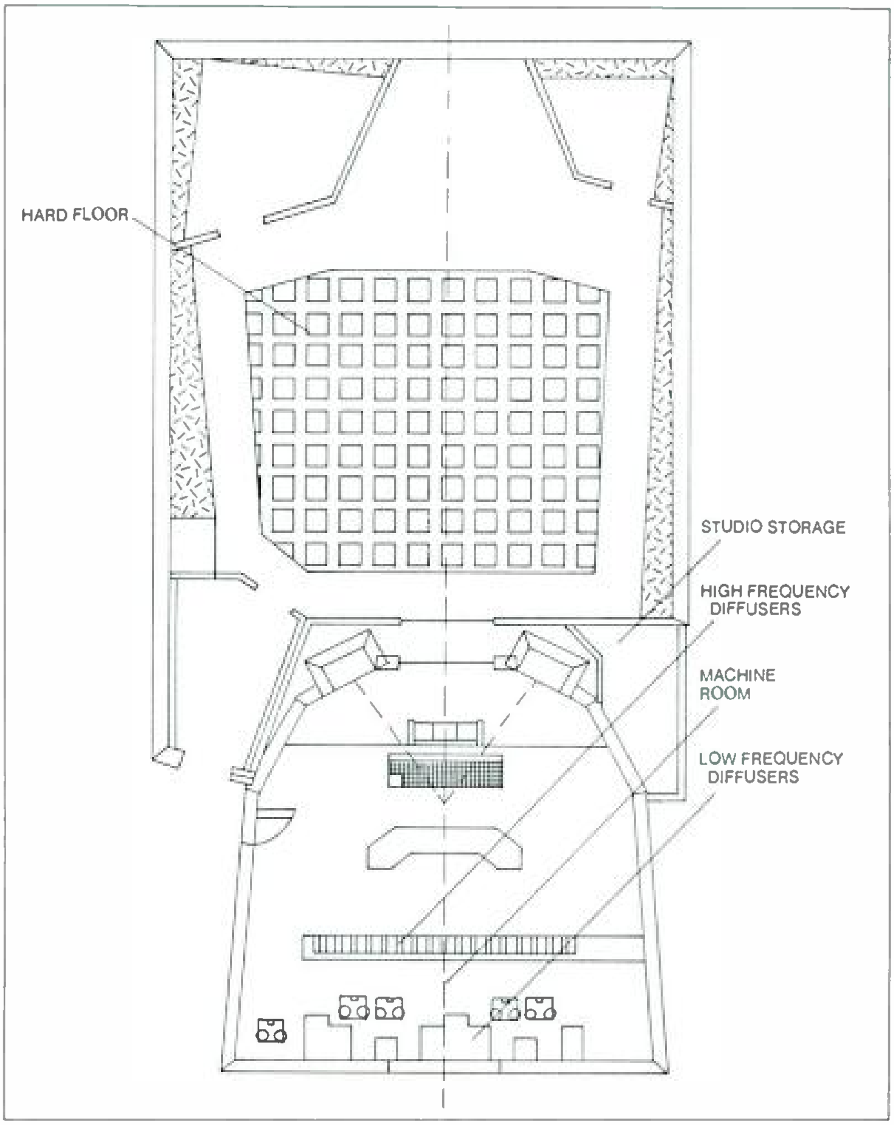
"This building presented several isolation problems, both in dealing with environmental noise from outside, such as airplanes and traffic, and in stopping the transmission of sound from other areas of the building through the structure to the studio," said Mr. Rodriguez. "We were very happy with the LEDE design in our previous facility, but we had to make the new room larger. We worried about losing the clean, accurate listening environment that we had before, but this room actually sounds even better."

TAKING RISKS

The equipment list of the new facility is impressive, with a custom-modified Neotek Series IIIc console, with Mega-Mix MIDI/SMPTE automation, Otari MTR 90 MkII and Stephens 821B multi-tracks, UREI and Tannoy monitors, and a wide assortment of signal processing gear, microphones and musical instruments.

Randy Adams is the studio manager/chief engineer at Future Audio.

Figure 1. Floor plan of the studio.



Even with the consistent track record established in 5 years of business with the other facility, the move to build a huge new production facility was a risky one, especially with the current economic climate and the proliferation of small home studios with impressive technological capabilities. Although a certain amount of in-house work for the related radio stations would continue, and the new location would be close enough geographically to keep all of the regular clients, the new facility would have to attract new clients to survive. Fortunately, the new location is much closer to most of the major ad agencies and production companies located in Dallas.

In constructing his former studio, Mr. Rodriguez relied on the services of a now defunct design firm. This time, armed with lessons learned from that sometimes disillusioning experience, and over 5 years of research into electro-acoustics and studio design, he decided to design the facility himself. Although he did consult a well-known studio designer with specific questions, the cost of a turnkey operation was deemed out of reach. Mr. Rodriguez enlisted the services of Doug Phelps, a builder with residential and commercial construction experience, to serve as general contractor. Mr. Phelps also happens to have extensive experience as an audio engineer, having operated his own home studio for several years, which he has since moved into the Rodriguez building.

"The biggest difference between this type of construction and the usual residential or commercial job is the much smaller margin for error," said Mr. Phelps. "Although I was used to working with tolerances of less than a 1/4-inch, it was sometimes difficult to convey the importance of that standard of quality to the subcontractors and laborers."

Figure 2. The control room.

CONSTRUCTION BEGINS

The existing building in which the new studio was to be located was formerly the headquarters for a large insurance company, and was configured with a large, open secretarial area in the center of each floor. To reduce the cost of demolishing the interior structure, and to maximize acoustical isolation from the outside, the decision was made to place the control room and studio in the center of the structure, with offices, shop, a smaller MIDI room, and other rooms to be constructed around the main studio. The control room wall, made of concrete block filled with mortar, went up while isolation springs were hung from the concrete structure of the floor above, to be used in suspending the ceiling. The concrete wall was surrounded by a small air space, a metal stud wall filled with acoustic attenuation blanket, and three layers of 5/8-inch sheet rock. A single layer of 5/8-inch sheet rock was plastered on to the inside of the cement block wall and covered with various surface materials. The ceiling was isolated from these walls with neoprene rubber and acoustical caulk. The studio wall construction consisted of two metal stud walls filled with acoustic attenuation blanket, separated by a small air space and covered with 3 layers of 5/8-inch sheet rock on the inside and 2 layers on the outside. The ceilings, which consisted of 2 layers of 5/8-inch sheet rock in the studio and 3 layers in the control room, were suspended from the floor above by hundreds of isolation springs, supplied by Mason Industries of Dallas. Larger springs were used to isolate 2 massive air handlers and the chiller unit for the building air conditioning system. Electrical power is supplied by a completely separate shielded single-phase transformer. Grounding was established using isolated ground receptacles and a "star" grounding configuration, using 1/0 wire to augment the code ground terminating under the console as the central node. From there, ground travels down 1/0 wire to a 6-foot copper stake in the ground outside the building.

The actual design and testing of the control room and studio was accomplished with the help of a Crown Techron TEF computer, purchased a few years ago for use in designing sound reinforcement installations. The TEF was used in conjunction with an IBM PC with AutoCad software to generate the drawings from which the facility was constructed. Unfortunately, during the process of developing the final detailed drawings, the unthinkable happened; the hard drive crashed, taking with it all of the drawings. After so many time-consuming hours spent designing and re-designing the facility on computer, Mr. Rodriguez could not face the prospect of having to start over. Fortunately, the basic design had been printed out previously, so he traced the available prints and finished the project with hand-drawn plans.

The geometry of the front wall of the control room was designed to eliminate all early reflections

The completed control room measures 35x32 feet, containing over 1100 square feet, with a machine room at the rear which is separated from the rest of the room by the diffuser wall. The layout of the new room was intended to approximate, in expanded form, the excellent listening environment of their previous control room at Sierra Recording. The geometry of the front wall of the control room was designed to eliminate all early reflections. A simple mirror test can confirm that early reflections which might have colored the sound at the listening position have

Figure 3. View into the studio.

been eliminated. By placing a mirror on the surfaces at the front wall of the control room, an individual seated at the listening position can predict early reflections, which will occur at any point where the speakers can be seen in the mirror. Therefore, those surfaces should be covered with absorptive material or, better yet, eliminated completely by altering the geometry of the walls.

ACOUSTIC MODIFICATIONS

After completion, testing with the TEF pointed out the need for some acoustical modifications which must still be made. One unexpected turn of events involved the fabric which covers the front ceiling in the control room. Due to fire department regulations, the fabric had to be fire treated, and apparently the fire retardent chemical changes the texture of the surface enough to render it somewhat reflective at shallow angles. The only apparent remedy is replacing the fabric with a different type which already meets fire code, and has the proper acoustical rating. The computer was also used to design custom quadratic residue diffusers, which were constructed by local carpenters at a fraction of the cost of those commercially available. Through consultation with experts and the study of articles about Dr. Schroeder's mathematical formulas on which these diffusers are based, Mr. Rodriguez translated the formulas into a computer program with which he figured the exact width, depth, and sequence of the individual segments of the diffusers. The design, properly implemented, would yield a diffuser which operates at a broader bandwidth than any of those commercially available. A prototype was built and tested with the TEF, and found to exceed expectations.

The full diffuser array measures 25x5 feet, and covers most of the rear wall of the control room. The main array of 7 horizontal and 6 vertical diffusers operates between 13560 Hz and 849 Hz. These are mounted on a structure which simply serves to support their weight, thereby forming a "false wall," through which low frequencies pass almost completely unaffected. These low frequency waveforms are broken up and diffused by low frequency diffusers which are constructed on the back wall if the control room. These are basically large, blown-up versions of the smaller diffusers, and operate from 275 Hz to 58 Hz. The action of these low frequency diffusers is intended to produce a quick and even decay without actually absorbing or removing any low frequency content from the room.

The studio measures 45x38 feet, with three isolation booths measuring 15x14, 12x9 and 10x9. The latter doubles as an airlock/entry, and has a ramped floor to facilitate the move from the level of the existing slab outside to the raised floor in the studio. The floor in the studio consists of a newly poured concrete slab, which is floating on a bed of sand to isolate it from the slab below. In addition, each iso booth has its own separate slab, isolated from the rest of the room. The new floor in the studio represents one of the more nightmarish episodes lingering from the construction.

Because of a complicated trade arrangement involving the installation of a sound system in exchange for concrete, drywall, and masonry work, Mr. Rodriguez found himself twice removed from the actual concrete workers, who insisted on carrying on all negotiations first through the contractor with whom the trade arrangement was made, who then insisted on dealing with Mr. Phelps. This unfortunate circumstance, combined with what must have been copious amounts of alcohol consumption, produced a concrete slab which undulated unpredictably like the surf at Malibu.

The new floor in the studio represents one of the more nightmarish episodes lingering from the construction.

The original concrete workers refused to fix the slab, which had to be completely broken up and re-poured, this time under closer supervision and with a different crew. The only really lasting negative impact of this episode, aside from a large dent in Mr. Rodriguez' wallet, was a decidedly mixed feeling toward the advisability of trading for labor. The construction of the diffusers was the only good trade experience, costing less than $1000.00, including materials. The carpenters, all members of a local band, have a new album as a result, and Future Audio has an array of diffusers which actually cost less than 1/10th of the going price.

The floor in the control room could not be poured in the same manner because the load-bearing capacity of the original slab would have been exceeded, so a floating floor of steel beams was used. A solid concrete pad was poured directly underneath the listening positions at the console and producer's desk, which eliminated any vibrations felt through the floor. An unseen benefit of this arrangement was discovered when Mr. Phelps suggested using the hollow part of the floor for air conditioning returns, which are channeled through the hollow low frequency diffusers on the back wall and back to the unit. This has served to make the control room air conditioning system exceptionally efficient, with very even coverage throughout the room.

The control room monitors are UREI 813 A's which were reconditioned by Altec after the move from the old location, and have been encased in a box made of 3/4-inch plywood, which measures 2 inches larger around than the original speaker cabinets. The 2-inch space was then filled with concrete to reduce low end loss from vibration through the cabinets. In fact, one of the most difficult situations encountered in the entire construction project occurred when these concrete-reinforced monsters had to be hoisted into place. A special steel beam was permanently fixed in the wall above the speakers, to be used in mounting a hoist.

The speaker cabinets are mounted on steel platforms supported by steel posts filled with concrete. The angle of the speakers is adjusted by 3 large bolts upon which the speakers rest. This massive mounting system serves to eliminate structural sound transmission, which occurs when walls, floors or ceilings are vibrated by low frequency energy. Although a 13 degree vertical angle is thought to be ideal for stereo imaging, an 8 degree angle was used here because of limited ceiling clearance. The distance between the speakers was calculated to be the same as the previous control room, expanded to account for the larger overall size of the room. The resultant stereo image is considered to be quite good, according to the subjective opinions of engineers who have used the facilities.

The control room contains several panels with a total of 32 mic/line inputs, allowing the recording of direct inputs and even vocals to take place in the control room, if desired. Also, MIDI patching allows samplers and sound sources located in the control room to be played from almost any location in the studio. There are also tie lines which connect the control room with a separate MIDI production/voiceover room.

FINAL NOTES

Most mic lines and tie lines were wired with Mogami Mini-Quad, while a few mic lines and the sends and returns for the main two track machines use Monster Cable Series 1. Sends to and from the 24-track machines use individually-jacketed Mogami 48 pair cable. Gold-plated DL connectors are used for all multi-pair connections.

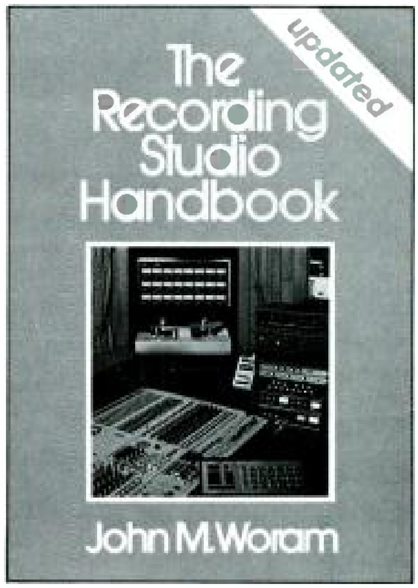
Now that the studio is up and running, with a healthy blend of demos, record projects, jingles and commercials, Mr. Rodriguez has had time to reflect on his decision to design and build the facility himself.

"We really had no choice at the time, because a turnkey job was so much more expensive than what we thought we could build it for," said Mr. Rodriguez. "However, not only did we end up going significantly over budget, but I sweated rocks with all of the problems. The whole thing basically ruined my disposition for about six months. There were many times during the construction when I wished I had just called Russ Berger or someone and had them do it. Of course, now that it's finished I feel better about it, and I think we've ended up with a great facility, but I don't know if I would ever try it again myself."

Although construction is finished, the growth process for Future Audio continues. Planned equipment updates and acquisitions include expanding the automation system to control equalization, auxiliary sends, and effects, more signal processing gear and a digital mixdown machine. There are also plans underway to lease the rest of the space surrounding the studio to production companies and others who would use the studio as their base of operations. db

For Your Audio Library

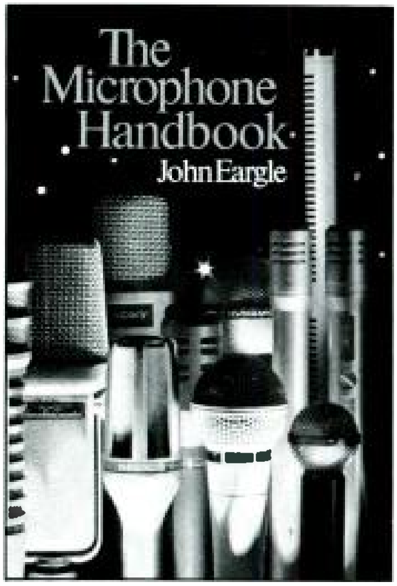
The Recording Studio Handbook



by John Woram

$39.50

The Recording Studio Handbook is an indispensable guide with something in it for everybody. It covers the basics beautifully. It provides in-depth insight into common situations and problems encountered by the professional engineer. It offers clear, practical explanations on a proliferation of new devices. In this updated edition, among the items covered are: Transducers, signal processing, noise reduction, recording techniques and more . . . In addition, it has been expanded to feature three all-new chapters . . . chapters on the in-line recording studio console, digital audio and time code implementation.



The Microphone Handbook

by John Eargle

$31.95

Completely up to date, this vital new handbook is a must for any professional whose work involves microphones. Among the topics covered are: Directional characteristics—the basic patterns, using patterns effectively, microphone sensitivity ratings, remote powering of condensers and proximity and distance effects. Other topics include: Multi-microphone interference problems, stereo microphone techniques, speech and music reinforcement, studio microphone techniques and microphone accessories. You'll find yourself reaching for it every time a new or unusual problem crops up.

Please send the books indicated below:

_______ copy(s) of The Microphone Handbook at $31.95 plus $2.00 for postage and handling per book.

_______ copy(s) of The Recording Studio Handbook at $39.50 plus $2.00 for postage and handling per book.

_______ Both books at $67.50 plus $3.00 for postage and handling. (Over 5% savings!)

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New Products

ANALOG RECORDER

● Otari Corporation announced the MTR-100A, which is a digitally controlled, pinchrollerless analog 24-track mastering recorder that features fully automated alignment of all record and reproduce parameters including level, bias, HF, MF and LF record equalization, phase compensation, HF and LF repro equalization and repro level. Other features include a tilt-down front panel with alpha-numeric keypad, and a backlit LCD display for use in entering transport and audio alignment parameters. The VU meter display can be remotely configured, and swings up for easy viewing during manual alignment, and to provide access to the optional Dolby SR card cage.

Mfr.- Otari Corporation

Price- under $60,000.00

Circle 60 on Reader Service Card

POWER SUPPLY

● Audio-Technica U.S., Inc. has developed a device for supplying phantom power to as many as four microphones simultaneously. The AT8506 is a four-channel, line-operated unit, producing 48 V d.c. to microphones requiring phantom power, but are operating in systems that lack this facility. Operating from 100-120 V a.c., 50-60 Hz, this unit is a highly regulated power supply. Even with heavily loaded or shorted inputs, a constant voltage source is maintained with no channel interaction. Each channel can provide up to 14 mA. A die-cast case withstands tough road conditions and provides RF and electrostatic shielding. Other standard features include internally protected regulator IC to prevent overheating or damage, and deluxe locking XLR-type connectors with silver-plated beryllium copper contacts for high reliability. Weighing less than 2-1/2 lbs., the unit has a 6-foot line cord with grounded plug, power switch and LED power indicator.

Mfr.- Audio-Technica U.S., Inc.

Price- $150.00

Circle 61 on Reader Service Card

WIRELESS MIC

●HM Electronics, Inc. has introduced a new generation of wireless microphones. The 50 Series represents a new concept in wireless microphones, with its two-channel body pac, newly designed handheld and two channel switching diversity receiver. A new RF link greatly improves the capture ratio for drop-out free performance. HME's new NRX-II noise reduction system, designed for wireless microphones, guarantees crisp, clear audio. Other features include: Mic-mute and power switch lock-outs on handheld, and operator selectable RF frequency selection on body pac system.

Mfr.- HM Electronics, Inc.

Price- $1095.00 (lavalier)

$1110.00 to $1365.00 (hand held)

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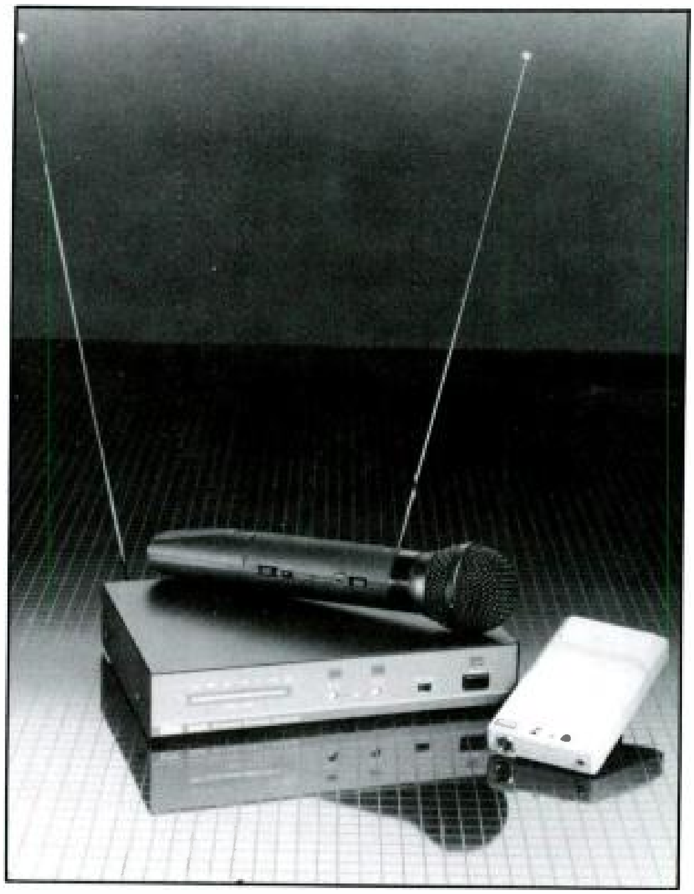


WIRELESS SYSTEMS CASE

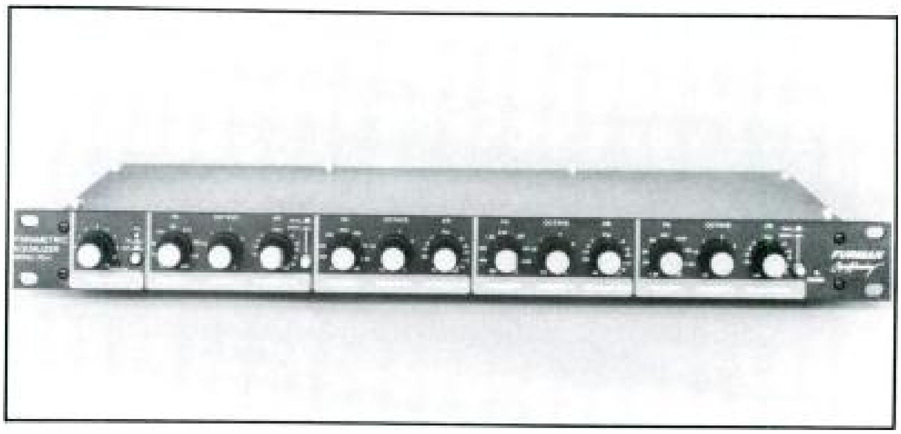
●Nady Systems offers a roadcase for the company's 101 and 201 VHF wireless systems. The case is constructed of molded polypropylene for light weight and durability. Other case features include high-density S-5000 foam, custom die-cut to house either a 101 or 201 receiver, and two transmitters, as well as transformer, battery, cords, and corrugated foam inside the top lid. The 101/201 roadcase is available in black or gray.

Mfr.- Nady Systems, Inc.

Price- $29.95

Circle 63 on Reader Service Card

PARAMETRIC EQUALIZER



● Furman Sound introduces Model PQ-4, a parametric equalizer. Its equalization range is unusually large, going from a full 20 dB of boost to an infinitely deep cut. Its more musically useful constant-Q curves allow a bandwidth spread from extremely narrow notches to boosts as wide as four octaves. The unit is a full-function, four-band parametric equalizer, but its top and bottom bands offer Peak/Shelf switches that allow either the usual peaking equalization, or shelving equalization where all frequencies above (for the top band) or below (for the bottom band) the selected frequency are boosted or cut. The equalizer may be used as an instrument preamp. A footswitch jack is provided for remotely controlling the equalizer. Other features include isolated connectors, a ground lift switch, and a low-noise design.

Mfr.- Furman Sound

Price- $359.00, $20.00 additional for optional balanced input/output

Circle 64 on Reader Service Card

SIGNAL PROCESSING SYSTEM

• Industrial Research Products announces the System 41, a complete signal processing system contained in a single 10 1/2-inch high chassis. Modules include line and mic mixers, equalizers, notch filters, cross overs, line drivers, distribution amplifiers and remote control capability. Balanced inputs and outputs on all modules eliminate hum pickups, crosstalk, and noise pickup. Shop fabrication and field installation time are minimized.

Mfr.- Industrial Research Products, Inc.

Price- $820.00 (chassis), and range from $240.00 to $450.00 per module

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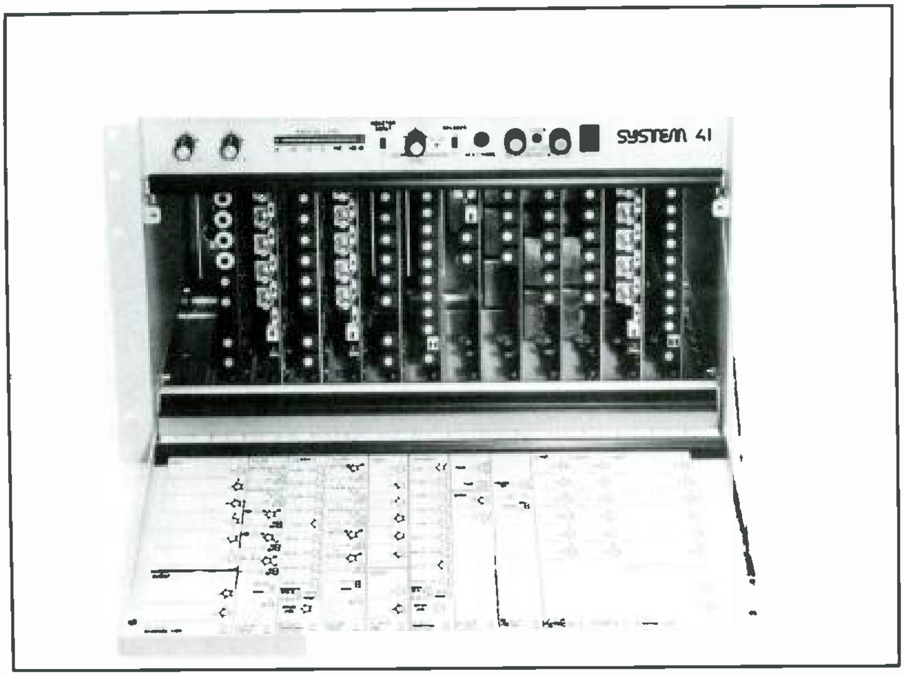


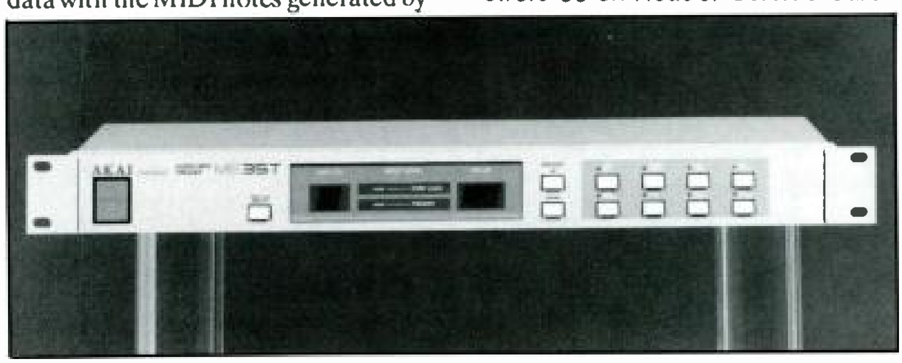
TRIGGER-TO-MIDI INTERFACE

•Akai Professional recently introduced the ME35T audio trigger-to-MIDI interface. The unit will accept eight trigger inputs (via 0.25-inch phone jacks) and convert them to user-defined MIDI notes, for driving samplers, synthesizers, sequencers, and so on. The trigger signals may be supplied by electronic drum pads, microphones, GPI triggers, drum sounds on tape, etc. For each of the eight inputs the user can adjust the following parameters: MIDI channel (1-16), sensitivity, capture time (delays the note on time 0-20 ms), on time (sets the duration of the sound, up to one second), MIDI note (0-127), trigger threshold, recovery time (to eliminate double trigger attacks), and velocity curve (eight different curves). The MIDI ports include MIDI IN, OUT, and THRU connections, and the MIDI IN can merge incoming MIDI data with the MIDI notes generated by the triggers. The ME35T is a convenient one rack space size.

Mfr.- Akai Professional

Price- $499.95

Circle 66 on Reader Service Card



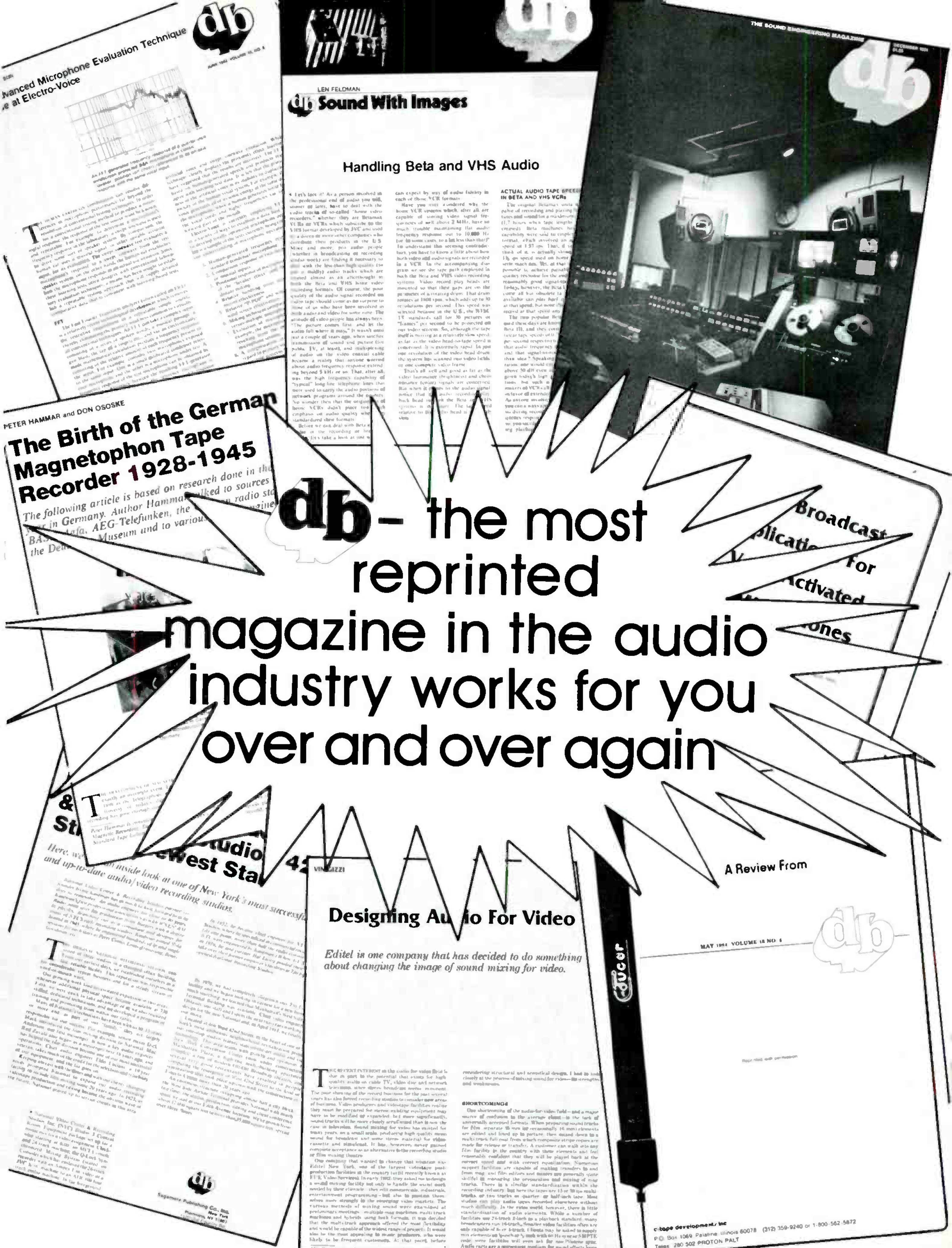
POWER AMP

•Crest Audio has introduced the model 7001 power amplifier which is a 2-rack space version of the model 8001. Rated at 550 watts/ch. at 8 ohms and 810 watts/ch. at 4 ohms, it is fully modular and weighs only 49 lbs. Capable of swinging 83 volts RMS per channel, tremendous peak power of 3,200 watts per channel is available at 2 ohms. Proprietary circuits include instantaneous gain modulation (IGM) impedance sensing, auto-ramp signal control and an RMS clip limiter.

Mfr.- Crest Audio

Price- $2589.00

Circle 67 on Reader Service Card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A Review From
MAY 1984 VOLUME 18 NO. 5
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VINTAGE TAPE RECORDERS: 2 Ampex 401A F/T, 1 portable, 1 console, $50.-each; 1 Presto 800 Series, F/T, rack $75. Also, 1 Infonics RCD-2 cassette duplicator (4 cassettes in 2 min.) $250. All manuals. Audio tape, reels/boxes 7"/5". **(201) 836-0194.**

WANTED PULTEC EQ'S. We will pay $1000.00 for almost any Pultec EQP equalizer. Dan Alexander Audio buys, sells and trades all kinds of second hand professional audio equipment. **DAN ALEXANDER AUDIO, 2944 San Pablo Ave., Berkeley, CA 94702. (415) 644-2363. Tlx: 650 291 8567 MCI.**

MIXING BOARDS

Factory direct, models for home recording and live sound. Quality boards priced to move. Quantities are limited, call while supplies last. **For info and prices call (201) 423-2176. Ask for Dave Fox.**

EMPLOYMENT

Audio Services Assistant Director. Oberlin College invites applications for the position of Assistant Director of Audio Services. The person in this position is responsible for professional recording, concert sound, administrative work and assistance in maintenance of equipment. Desired qualifications include Baccalaureata Degree, recording studio environment work experience, oral and written communication skills, familiarity with musical notation or instrumental technique and knowledge of classical music. Thorough knowledge of electronics relating to audio required. To ensure consideration, apply by October 30, 1988 to **Thomas Bethel, Director of Audio Services, Oberlin College, Oberlin, OH 44074.** Include three letters of reference with your letter of application. Salary commensurate with experience; range 16k to 22k. Affirmative Action, Equal Opportunity Employer.

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Send copy to:

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

People, Places... & Happenings

•Radio production company Sound Advice has relocated to a new, custom-built production facility in the Hollywood and Vine Plaza. Encompassing approximately 1700 square feet, the new facility includes a modern, multi-track control room and studio with full production capabilities, conference and meeting rooms, ans ample storage space. Sound Advice partners Lou **Schwing** and **Matt Wright** built their studio specifically for radio production. The result is a studio that is extremely flexible. Built with full MIDI capability, the studio meets the needs of many types of production in addition to radio, and is adaptable for other uses as the production climate changes. It is also equipped with the technology for direct to hard disk digital recording. The studio and control room were designed by acoustic engineer F. Alton Everest. The facility was designed and constructed according to the principles of live end/dead end theory, and uses the RPG diffuser system.

•**The Museum of Broadcasting** has received a contribution of equipment worth $3.5 million from **Sony Corporation of America Foundation, Inc.** The donation by Sony will help to equip the Museum of Broadcasting's new building with state-of-the-art electronics and includes several different types of tape machines, new tape formats for video and audio recording, television and radio systems, as well as individual projectors for the theaters. The equipment will be used in all of the new building's facilities, including the television console room that will have 95 viewing consoles, a 200-seat principal theater, an 80-seat theater, a 45-seat screening room, a 45-seat education room, and a radio listening room that will be the first Museum space created expressly for exhibiting radio programming. The donated tape machines will ensure that the copies of radio and television broadcasts kept in the Museum's archive, and those made available for public viewing, will be of the highest quality obtainable. Masters of radio programs will be transferred to and stored on digital audio tape. The donation also includes the PCM-2500 professional digital audio recorder.

•**Bruce Borgerson**, after a year's absence, is back doing publicity services for **Le Mobile, Guy Charbonneau's** mobile audio production facility. Le Mobile has been busy with recording projects, and Guy Charbonneau was in Montreal to engineer a live recording of the Montreal Jazz Festival. Recent additions to Le Mobile's equipment roster include an Eventide H3000 Ultra-Harmonizer, two TC Electronic 2290 samplers, and a Yamaha REV 1 reverberation unit.

•**Allen & Heath** has announced the appointment of **John Ball** as Chief Executive. He has a long connection with the sound business as well as other related, technically-based industries. He came to Allen & Heath from the Robert Luff organization.

•**Shure Brothers, Inc.** has announced the appointment of **Lottie Morgan** to the position of Vice President, Sales. In her new role, she will be responsible for the supervision of all domestic distributor sales. She has been with the company for 26 years.

•Also announced was the appointment of **Alan Hershner** to the position of Director of Sales, Domestic Distributor Products. In this role, he will supervise the activities of domestic sales representatives. His previous position was Western Regional Sales Manager. He has been with the company since 1984.

•**SMPTE** has announced the session topics for their 1988 Technical Conference and Equipment Exhibit to be held in New York City on October 15-19. The theme for the 130th conference is "Innovations in Imaging and Sound." Session topics include TV Post Production I and II, Sound Technology, Post Production (Film/TV), TV Audio, and New Technology in Imaging and Display I and II.

•**Brystonvermont Limited** has announced the appointment of **Martin Bartelstone** as Vice President of Sales for their professional division. Martin has thirteen years of experience in the audio industry. He will be responsible for all professional audio and Special Product sales in the U.S. He is looking to strengthen the national dealer structure.

•**Carillon Technology, Inc.** has acquired the business operations of **dbx/ADC** and **BSR (Japan) Ltd.**, division of BSR. Carillon Technology, Inc. is a privately owned company based in San Bruno, CA. A restructuring of dbx/ADC has been announced by **Jacques Robinson**, President of Carillon Technology. **Michael L. Kelly** has been named dbx President. Mr. Kelly is former Executive Vice President for Research, Product Development and Manufacturing for Analog and Digital Systems, Inc. (ADS). dbx production and manufacturing facilities will be moved to the West Coast and other locations. dbx operations, including marketing, sales and engineering, will remain in the Boston area. ADC has been organized as a separate company, Audio Dynamics Corporation. Its operations, including marketing and sales, are being moved to San Bruno, CA.

•**Garry Margolis**, Vice President, Marketing, **JBL International**, announced an agreement with the **People's Republic of China** to install JBL loudspeakers and electronics in the Great Hall of the People in Beijing. In addition to government offices, the Great Hall of the People includes three major areas to be JBL-equipped. The main hall is used for musical performances, conferences, and meetings of the national congress, with seating capacity of 10,000. The Banquet Hall is a circular, domed area used for government banquets and performances, with a total seating of 7,000. The third key area is a 700 seat Concert Hall. Product installation is scheduled to begin in Fall, 1988, with equipment being supplied through **Advanced Communications Electronics** (ACE), Hong Kong, JBL's distributor for China.

RON TUDOR

Update from Down Under

When we did a story back in March/April 1987 on what was then AAV Studios in Australia, we soon learned that the studio had a name change and it was still growing from those days of our report on its doing the audio for the original version of the film "Crocodile Dundee."

Every construction undertaking brings along more than its share of inherent difficulties. However, when that construction concerns a new recording/mixing studio, where the end result must be absolutely perfect to achieve international status, then there can be no room for second chances. It must be right the first time!

Imagine the relief for the heads of Metropolis Audio in Melbourne, Australia, when their "million dollar plus" Studio Three was pronounced in perfect condition on the very first day it was completed.

AVOIDING PROBLEMS

From an acoustic viewpoint, a lot of the possible problems were eliminated with exhaustive acoustic tests during the latter part of studio construction and equipment installation, wiring, etc. These tests were carried out by Graham Thirkell and Lindy Deck-her, acting in tandem with construction and acoustic designer, Peter Brown and Associates, together with the audio craftsmen of Metropolis Audio.

The very first project in Studio Three was of monumental importance because it was the follow-up album for John Farnham to his smash "Whispering Jack" which is now the biggest selling Australian album of all time. Farnham's record producer, Ross Fraser, was to some extent a little nervous about being the first to undertake a top flight final mix in the new facility.

Fraser said, "I must admit to having some reservations to being involved in the first project undertaken in Studio Three especially since we had previously spent some months in Studio One and the decision to change studios midway into mixing the Farnham album was, in hindsight, taken at some risk. But it worked brilliantly.

Figure 1. The console mounted on a turntable so it can do multiple duty.

The monitoring felt balanced from day one. It felt like a room that had been operating for years. It's a credit to Ernie and Mack (Ern Rose and Ian McKenzie, chief engineers at Metropolis) that the room sounded that good from the first day. I can't wait to get back in again."

It is worth noting that within three days of release, the new John Farnham album "Age of Reason" was Number One across Australia – a position it still holds now. The "Whispering Jack" album has returned to the Top Ten after some 90 weeks on the chart which is prepared and issued by the Australian Record Industry Association (ARIA).

VARYING POSITIONS

A revolutionary idea in monitoring control and variety has been successfully achieved in Studio Three at Metropolis Audio. They mounted a Solid State Logic 6000E Series console on a huge turntable, thereby allowing the whole thing to be turned to face a different monitoring system at either end of the room. In one position, it is perfect for near field stereo rock mixes. In another position, it is exactly right for film/television mixing, with left-center-right monitoring. The other two speakers can then be utilized to achieve "surround" monitoring. At the midway position, it gives engineers/producers direct eye to eye contact with talent in the overdub booth.

Every stage of development in the design and construction of Studio Three was carefully thought out to create a totally integrated electronic music and video environment. The floor area of 48 square meters (516 sq. ft.) more than adequately accommodates a large array of electronic keyboards and sequencers. Acoustic design of the studio includes the use of Helmholtz resonators, membrane LF absorption and Schroder diffusers.

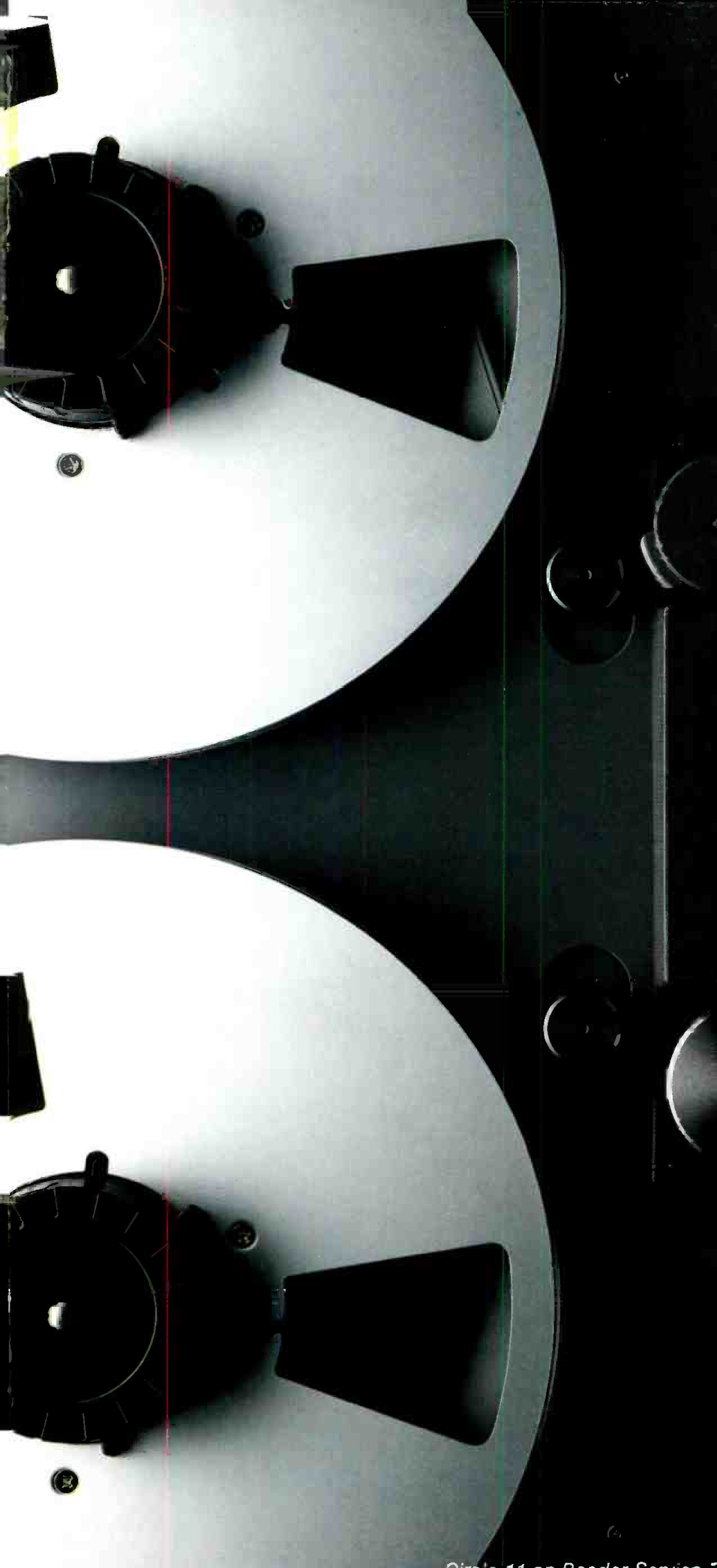
Special emphasis has been placed on providing a comprehensive array of outboard equipment, which is included in the studio rate.

This control room is a unique audio environment with an impressive array of ancillary equipment, put together by the creative team at Metropolis Audio. db

EQUIPMENT LIST

Lexicon 480L
Audio Design Limiters
Window recorder
2 x AMS reverbs
digital delays
Yamaha SFX 90s
CML mic pre-amps and equalizers
Yamaha Rev 7 reverb
Drawmer gates
AMS Harmonizer

Ron Tudor is with Metropolis Audio in downtown South Melbourne, Victoria, Australia.



INTRODUCING THE NEW MX-55

Before you take us to task for trying to improve the BII, a design that has become the "workhorse" standard for two-channel audio machines, consider what the new MX-55 offers:

An integral autolocator; a voice editing mode that allows 2X speed playback at normal pitch; a built-in cue speaker; GSPIPO (gapless, seamless, punch-in, punch-out),* and because you told us you wanted it, *all* adjustments are available through the front panel! These, and many more improvements will help make your job easier, and the results more creative.

And for those of you who still won't forgive us, we're keeping the BII in the line. So either way, you can get exactly what you need from Otari; Technology You Can Trust. Call Otari at (415) 341-5900 for information about the new MX-55.

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