May 1989

Recording Engineer/Producer

The Applications Magazine for Audio Professionals

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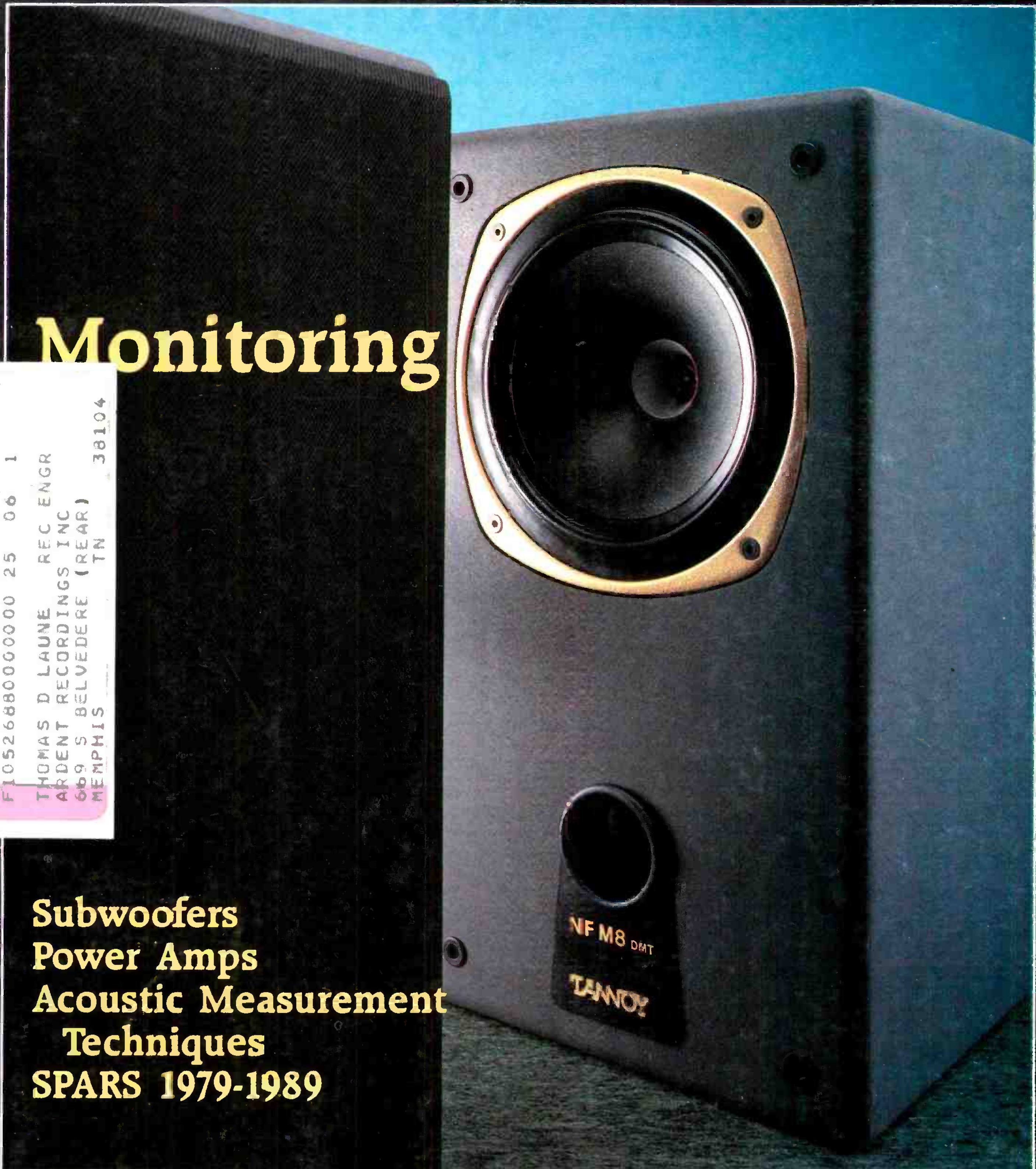

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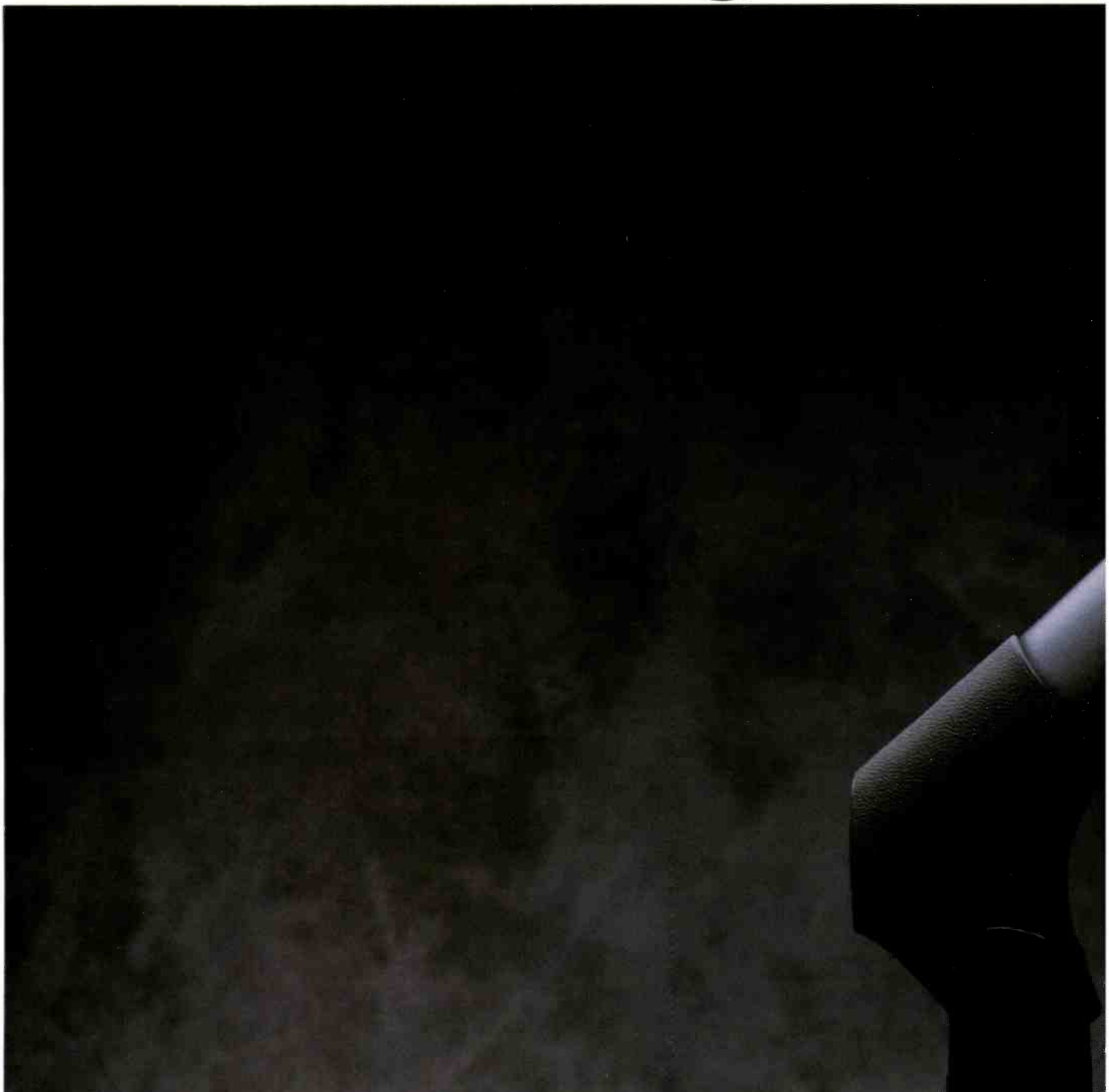

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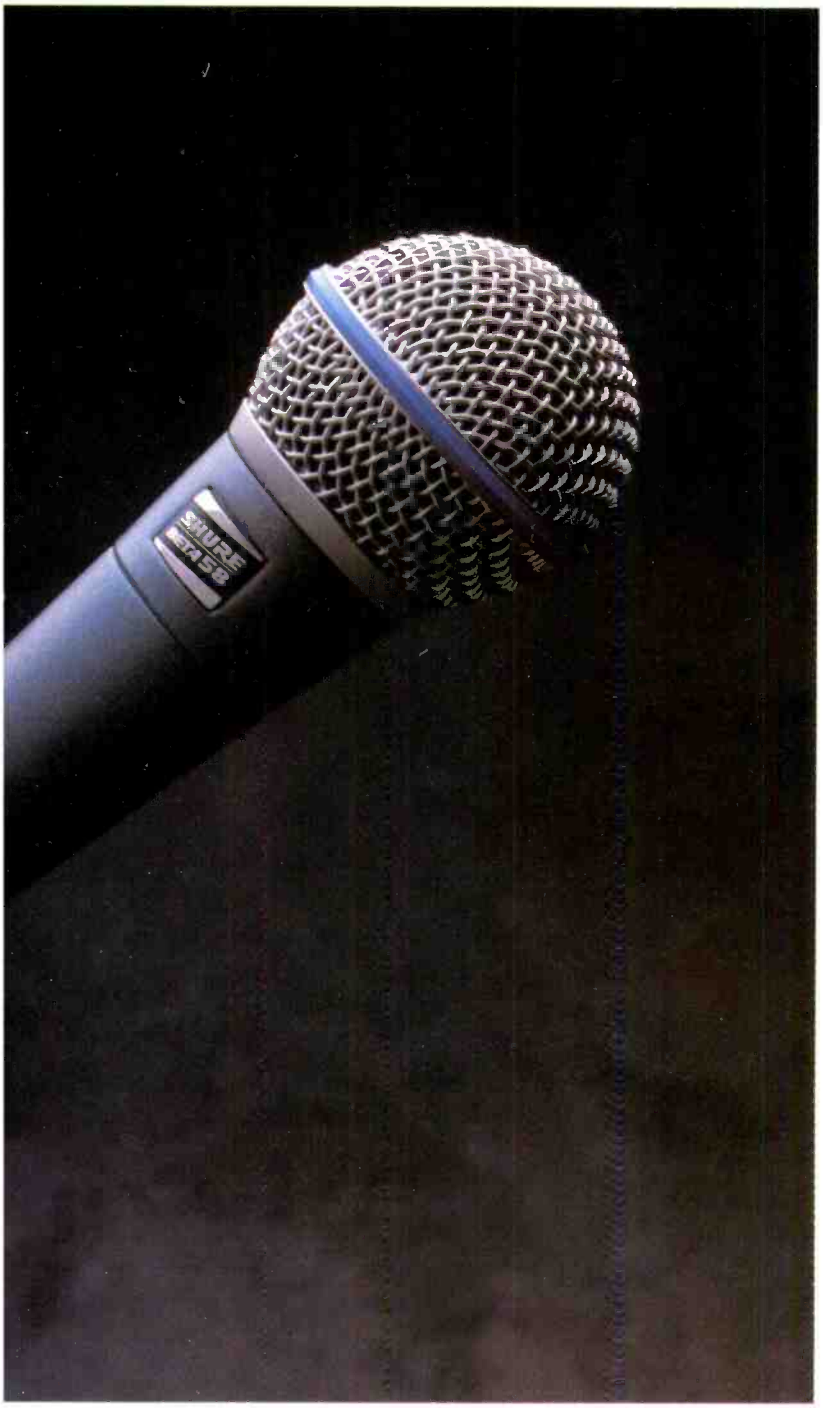

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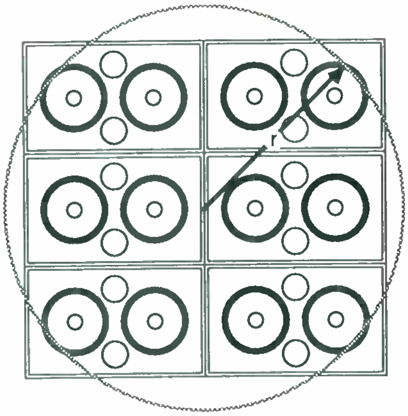

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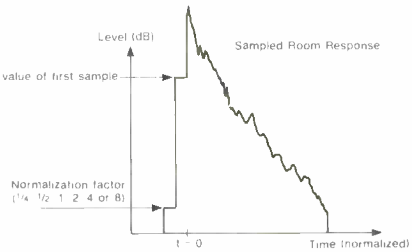

Other Features

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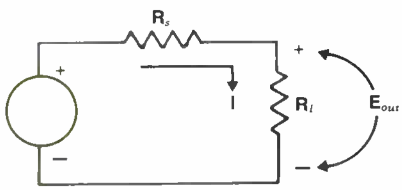

Departments

Editor's note:
Due to the overwhelming response to the "Engineer/Producer Index" and "Tracks," RE/P has had to reformat both departments to make room for the volume of information supplied by our active readers. All current information will appear in next month's redesigned, comprehensive sections.

On the Cover

Tannoy's NFM-8 close-field monitor with 8-inch dual concentric drive unit.

Volume 20, No. 5

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But there's a down side to digital, too. For one thing, there's no friendly tape noise to cover up mistakes, or to add that mysterious "something" to the mix. And the initial cost for a digital machine can be *scary.*

So what's the final mix, or the bottom line, if you prefer? The cost is high, and even though the Otari DTR-900B is a powerful client draw, it's important to consider your return on investment.

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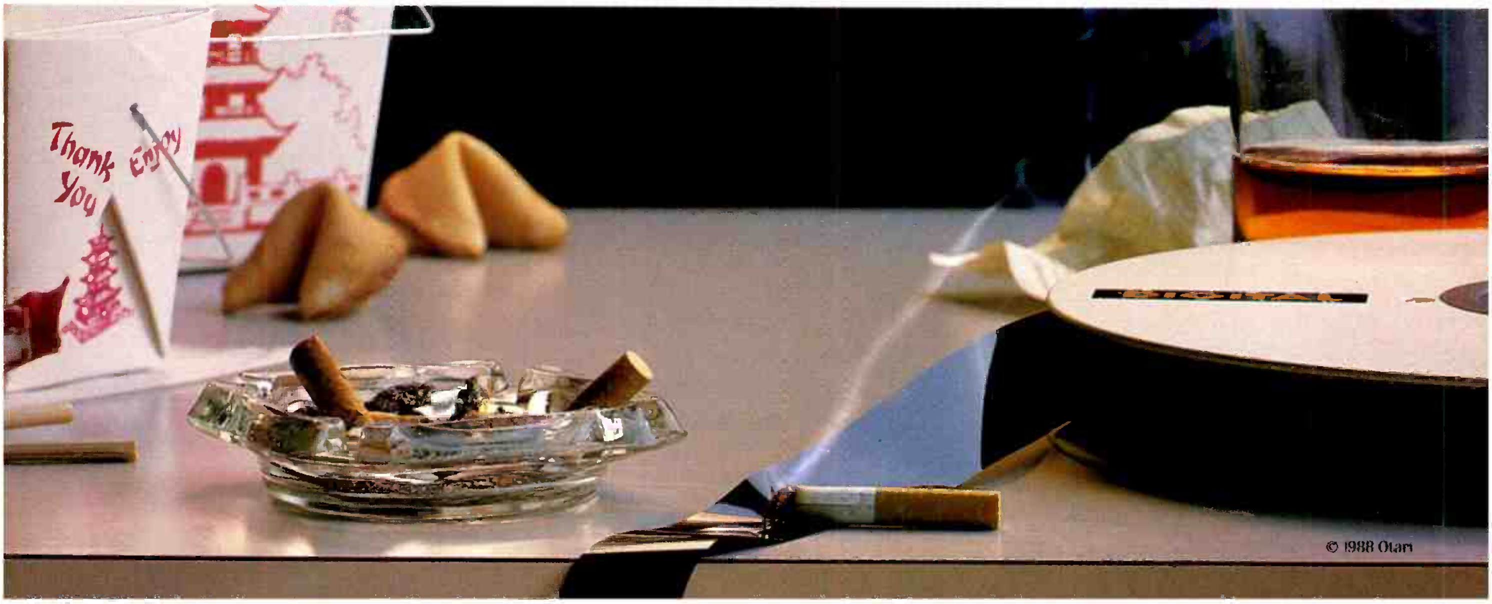

GUEST EDITORIAL

By Mark E. Engebretson

More Comes to MOR

While attending a recent MOR concert, I was struck by the fact that several members of the audience arrived with packages of surgical cotton and were distributing small quantities to their friends and people nearby. Before the lights went down, the recipients wadded the cotton into small balls and packed their ears. Observing that the person at the mixing console chose not to partake in this ritual, I mused about the frequency attenuation properties of cotton wads.

I decided that this home-spun approach probably didn't afford any real protection from damaging sound levels, but was likely to attenuate the upper frequencies, masking details that I wanted to hear. In spite of this, I spent the first several minutes of the show contemplating leaving my seat to locate a cotton supplier. For reasons unrelated to my auditory discomfort, however, I didn't go. The remaining two hours were divided between trying to appreciate a wonderful performance and analyzing why the sound bothered me so.

The sound system consisted of quality equipment. The guy at the console was the band's studio engineer, with a proven track record of very good-sounding CDs. From my vantage point in the auditorium, I could see that the power amps had gobs of headroom, yet there was almost no dynamic range and lots of distortion that I could not immediately rationalize. I felt a tingling sensation in my ears, mixed with periodic pressure discomfort, similar to rapid altitude changes when flying with blocked sinuses. Mine were clear, however.

At one point, one of the musicians made an announcement. Although I was in the direct field of the loudspeaker arrays, the words were almost unintelligible and the distortion persisted in spite of the much lower sound level of the announcement. As I struggled to understand the words, I studied the loudspeaker arrays and tried to imagine the frequency-directivity interaction of the dozen or so horns that were more or less pointed in my direction.

I'd heard similar distortion in arena concerts before, but had dismissed it as a product of garbage acoustics and had given the subject no further thought. But here I was in a fairly good environment and close enough to the loudspeakers to be in the direct field. The fact that the distortion was independent of the sound level convinced me that I was hearing temporal distortion and that the source wasn't the loudspeakers, but the manner in which their energy outputs combined in the space.

When the music started again, my first reaction was to cover my ears with my hands. When my ears were covered, the highs vanished, along with the distortion. But I also observed what seemed to be a restoration of middle-frequency detail. I could distinguish individual conga hits that before had been only a smear. I repeated the experiment several times, achieving the same results. The *receiver* was overloading.

Good God, the problem is *with me!* What if my auditory system overloads 10dB earlier than everyone else's? Maybe there's some truth to those stickers on the road cases that say, "If it's too loud, *you're too old*." I contemplated suicide.

Since that evening, I've thought a lot about musical communication and human auditory factors with regard to sound reinforcement. Certain sound reinforcement practices cannot be reconciled. We know that sensation of loudness depends upon amplitude and frequency. At high SPLs, the lowest frequencies require roughly 100 times the energy needed to sound as loud as mid-high frequencies. Even so, most sound reinforcement systems continue to be designed with capabilities that are almost exactly opposite.

The sound reinforcement community invests huge sums of money to support a major act on the road. If the artist is a new client or a superstar, only the latest and best equipment will do. And the sound system needs to be bigger and more powerful than the last tour, thank you very much. While most artists usually know what they want to communicate, the vast majority are completely unfamiliar with the frequency/energy range of human hearing. So they cannot articulate their desires in terms that engineers understand.

The human auditory system is remarkable. It can process a range of frequencies in excess of seven octaves and intensity ratios of 10^{12}:1. It also incorporates built-in neuro-physiological systems that protect the various auditory components and the brain from damage when pressures exceed the normal range.

The effects generally begin with desensitization (threshold shifting), followed by the neurological equivalent of filtering or limiting (depending on the sound), then interruption and, finally, dysfunction. The nature of the sound, the loudness level and duration all affect whether the damage will be temporary or permanent. The auditory effects vary from person to person and may also be accompanied by other physiological sensations. Of immediate concern are those that affect music.

For entertainment systems, 120dB SPL at 30Hz is reasonable. But at 120dBA, or even 110dBA, there is no "high fidelity" for humans with normal hearing. Our auditory system won't allow it.

Unfortunately, our auditory protection mechanisms aren't subtle. They didn't evolve to protect humans from wide-range amplified entertainment. Even mild threshold shifts introduce gross hearing non-linearities. When the auditory system goes into the limit-protect mode, it becomes a brick wall, with crummy transient characteristics and a compression ratio of infinity.

As close as I have been able to determine, my threshold—typical for undamaged equipment of my age—is between 105dBA and 108dBA for wide-bandwidth signals, with crest factors of 10dB. Even if the acoustic energy had been flawless at 120dBA—which it wasn't—it wouldn't have made any difference. My own auditory system would have turned it into garbage.

Artists want powerful musical communication with their audiences. To achieve it, they need more powerful, very low-frequency energy, but neither they nor their sound reinforcement people seem to have caught on. The people that pay the bills don't know what to ask for, except more. They expect to see more and are likely to walk if they don't get what they think they want. No sound company wants to risk a major account, especially when dozens of competitors are ready and willing to step in and give their version of what the account thinks is needed.

More and more is the order of the day. Today's concert sound is unfit for human consumption. It is a very sorry state of affairs. RE/P

Mark Engebretson is principle consultant for Electroacoustic Technology and co-owner of Summit Laboratories in Warner Springs, CA.

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NEWS

Supreme Court hears sound reinforcement case

The Supreme Court is expected to rule early this summer on whether a governmental body, in an attempt to control noise levels, can regulate the use of an outdoor sound system and the people who work on them at outdoor concerts.

The case, Ward vs. Rock Against Racism, was argued before the court on Feb. 27. At issue is a City of New York noise control regulation requiring musical performers at Central Park's Naumburg Bandshell to use a city-supplied sound system and sound technician.

Rock Against Racism, which sponsors an annual concert at the bandshell, argued that the regulation is too broad and infringes upon the First Amendment's provision of freedom of expression.

The city argued that it has a legitimate interest to regulate noise, that the regulations are content-neutral and that there are no feasible alternatives to using a city-supplied system and technician.

After the suit was filed in April 1985, the district court found that the city has a legitimate interest in controlling excessive noise levels and that the regulations were constitutional. The regulations allowed RAR to control the mix, it said, but not the volume. RAR appealed the ruling, and the appeals court reversed the lower court's ruling that the guidelines were constitutional, citing less intrusive means to control noise. However, it upheld the right of the city to limit the volume of music performed at the bandshell.

The court is expected to rule on two issues: what First Amendment rights are given to live musical performance, and whether content-neutral regulations of speech are subject to a concept called "least restrictive analysis," meaning that the least drastic measure has been taken.

Boston AES presents MIDI lecture

The Boston chapter of the Audio Engineering Society will present an introductory lecture/demonstration on MIDI for audio professionals. The speaker will be Paul D. Lehrman, *RE/P's* electronic music consulting editor.

The lecture will be held on Tuesday, May 23, at the headquarters of Bolt, Beranek & Newman, in Cambridge, MA. For more information, call Kristen Beard, vice-chairperson, at 617-891-6790, extension 128.

Iowa holds audio seminars

The University of Iowa will hold its 10th annual seminar in audio recording from June 12-23 in Iowa City. The principle instructors will be Jerry Bruck, president of Posthorn Recordings in New York; Bob Ludwig, vice president and chief engineer of Masterdisk in New York; and Lowell Cross, professor of music and director of recording studios at the university. Russell Hamm, president of Gotham Audio, will be the special guest.

Topics will include stereophonic and amwww.americanradiohistory.com

NEWS

bisonic mic techniques, mic comparisons, A/D and D/A comparisons, noise reduction systems and preparing tapes for CD and LP mastering. Tuition is $154 for undergraduates and $242 for graduate students.

For more information, contact Cross at Recording Studios, School of Music, the University of Iowa, Iowa City, IA 52242; 319-335-1664; fax 319-335-2777.

SPARS announces exam schedule

SPARS will administer its National Studio Exam this month in various cities across the country. The exam consists of 200 multiple-choice questions covering equipment maintenance/operation and session planning/setup.

The exam will be administered May 13 in Seattle, Miami, Los Angeles, San Francisco, Atlanta, Chicago, Boston, New York and Dallas. On May 20, it will be administered in Denver.

For more information, contact SPARS at 4300 10th Ave. N., Suite 2, Lake Worth, FL 33461; 407-641-6648.

News notes

DOD Electronics has been purchased by John Johnson, the company's president and CEO, and three investors: Charles Chewning, Tom Henderson and Robert Henderson. The company was previously owned by Johnson and David DiFrancesco, who left the company on Jan. 1.

Steve Durr, of the design firm **Steven E. Durr & Associates,** has been contracted to design studios for Windmark Recording, Virginia Beach, VA; Concept Recording Studio, Birmingham, AL; and Perfect Pitch Recording and Production, Hickory, NC.

New York Technical Support has formed a separate installation company, New York Tech, offering site planning, computer-aided services, ergonomic optimization, panel design, installation and wiring design, and AutoCAD operation.

George Massenburg Labs has relocated to a larger facility at 8721 Burnet Ave., Van Nuys, CA 91405; 818-781-1022; fax 818-781-3828.

People

Solid State Logic has announced three appointments. **Piers Plaskitt** has been named chief executive officer. **Chris David** has been relocated from Oxford headquarters to the Los Angeles office. **David Collie** has been named product development manager.

WaveFrame has announced several appointments. **Gus Skinas** has been named senior product manager for professional audio. **Dave Frederick** has been named product manager for music applications. As part of an expansion of its domestic and international service, WaveFrame has appointed **Mike Buffington** as director of field engineering. **Arthur "Midget" Sloatman** has been named systems engineer. **Craig Damon,** formerly customer service manager, has been named European support manager and will relocate to a newly opened office in Copenhagen, Denmark.

Cary Fischer has joined George Massenburg Labs as vice president of marketing and sales.

Alesis has announced two promotions. **Ralph Goldheim** has been named vice president of sales and marketing. **Allen Wald** has been named vice president of advertising and promotion.

Tore B. Nordahl has been named president of Studer Revox America.

RE/P

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

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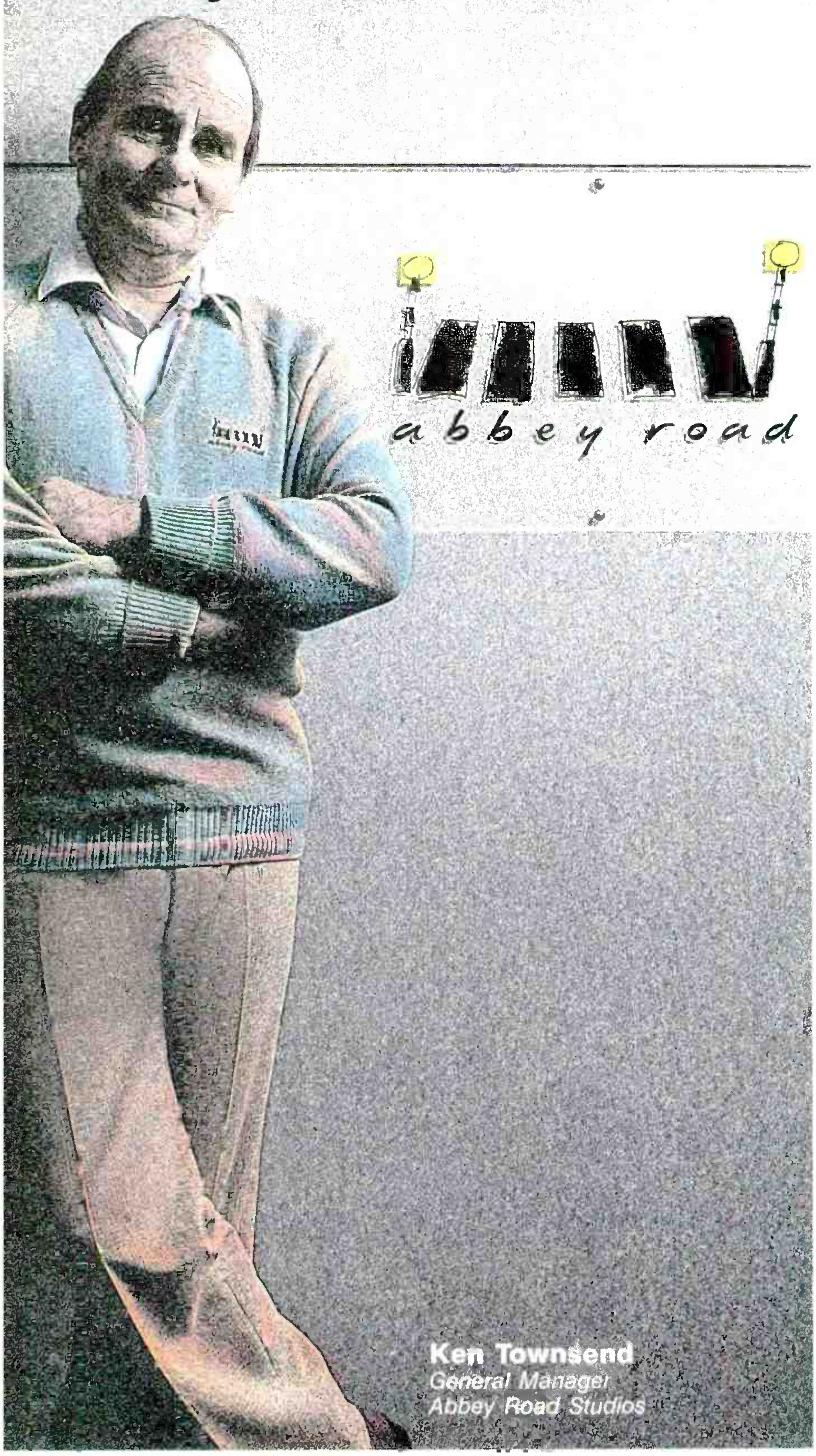

Abbey Road Studios—the name evokes a time, an era—the Beatles.

Ken Townsend, Abbey Road general manager, has heard it all—from the record that put the street on the map to Pink Floyd's "Dark Side of the Moon".

Today, in a very modern state-of-the-art facility, Ken hears it on AGFA PEM 469. It's bias compatible and..."It simply sounds better," he says.

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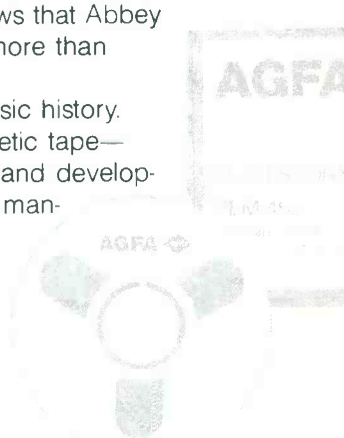

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MANAGING MIDI

By Paul D. Lehrman

Automation is Alive!

In the February issue of this magazine, two articles appeared that raised some important points about the future of MIDI as a tool for automation. The articles were extremely valuable, but, unfortunately, contained some misinformation, and our esteemed editor has graciously allowed me to try to set the record straight in my column. Besides noting these corrections, I think this is also a good opportunity to continue the discussion by offering an opposing point of view to one of the articles—but first things first.

Peter Jostins's article, "MIDI and the Mixing Console," is an excellent overview of the various methods used to achieve console automation with MIDI, but it appears he was given inaccurate data by some manufacturers. This has led to a serious misunderstanding of the roles of various types of MIDI data.

In the Yamaha DMP11, the manufacturer has not "abandoned continuous controllers [to represent automation data] in favor of MIDI note information." In fact, the company did just the opposite—unlike its older brother the DMP7, which responds to both notes and controllers, the DMP11 responds *only* to controllers. I haven't spoken to all of the other manufacturers mentioned in the article, but people at Allen and Heath tell me they have always used MIDI controllers to operate the mutes in their consoles, not notes.

The advantage of using controllers over notes for this task is subtle. Both are equally efficient: Each is a 3-byte command, and under certain circumstances (which we'll get to in a minute) can be two bytes. Although there is a common misconception that controller commands must always be sent in huge globs that choke MIDI lines, this is completely untrue—any decent sequencer that features list or graphic controller editing can record and/or generate a single controller event.

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

The difference has to do with the way sequencer designers have implemented "chasing." Many more sequencers today allow controller chasing—a feature that, when you start a sequence in the middle, executes the most recent controller settings *previous* to that point before starting the sequence—rather than chasing of "note-on" commands. One major reason for this is that MIDI controllers (especially number 7, volume) were *supposed* to be used for this sort of task, while notes were designed more for dealing with discrete events.

Bandwidth limitations shouldn't close the door on MIDI automation.

One more bit of confusion generated by Jostins' article needs to be addressed: the idea of "double-speed" MIDI. The MIDI specification is quite specific about MIDI's data rate: It is 31.25Kbaud, ±1%. A form of MIDI running at 61.5Kbaud was implemented, once upon a time, by a sampler manufacturer that wanted to speed up sample transfers over a MIDI line. It wasn't a bad idea, but because it was incompatible with existing MIDI systems, it was immediately shot down by the various MIDI standards committees, and has pretty much vanished (and that manufacturer is no longer in business).

One popular software sequencer has a switch that allows double-speed transmission, but according to a spokesman for the manufacturer, that switch is only there for doing dumps to that particular sampler, and is *not* used for sequencing. That's good—if the program did generate MIDI note data at double speed, there is not a single device in the world that would respond to it. Devising a 61.5Kbaud transmission protocol for console automation is certainly possible, but it won't be MIDI, and you can't call it that.

The article by Jim Cooper, one of the foremost exponents of MIDI automation, painted what was, for me, a surprisingly dark picture of the medium's future. Part of his pessimism may be traceable to his underestimating the capabilities of today's sequencers. He states that sequencers "almost certainly will not" chase controllers, when, in fact, I can name at least half a dozen popular ones that will, and their designers put this feature in expressly for the purpose of automation. He also says that, when it comes to being able to erase and replace data from a single controller, "only a very sophisticated computer-based sequencer *might* have this capability" (his emphasis), but, actually, there are quite a few sequencers that can do this, and anyone seriously trying to do MIDI automation would certainly be working with one of them.

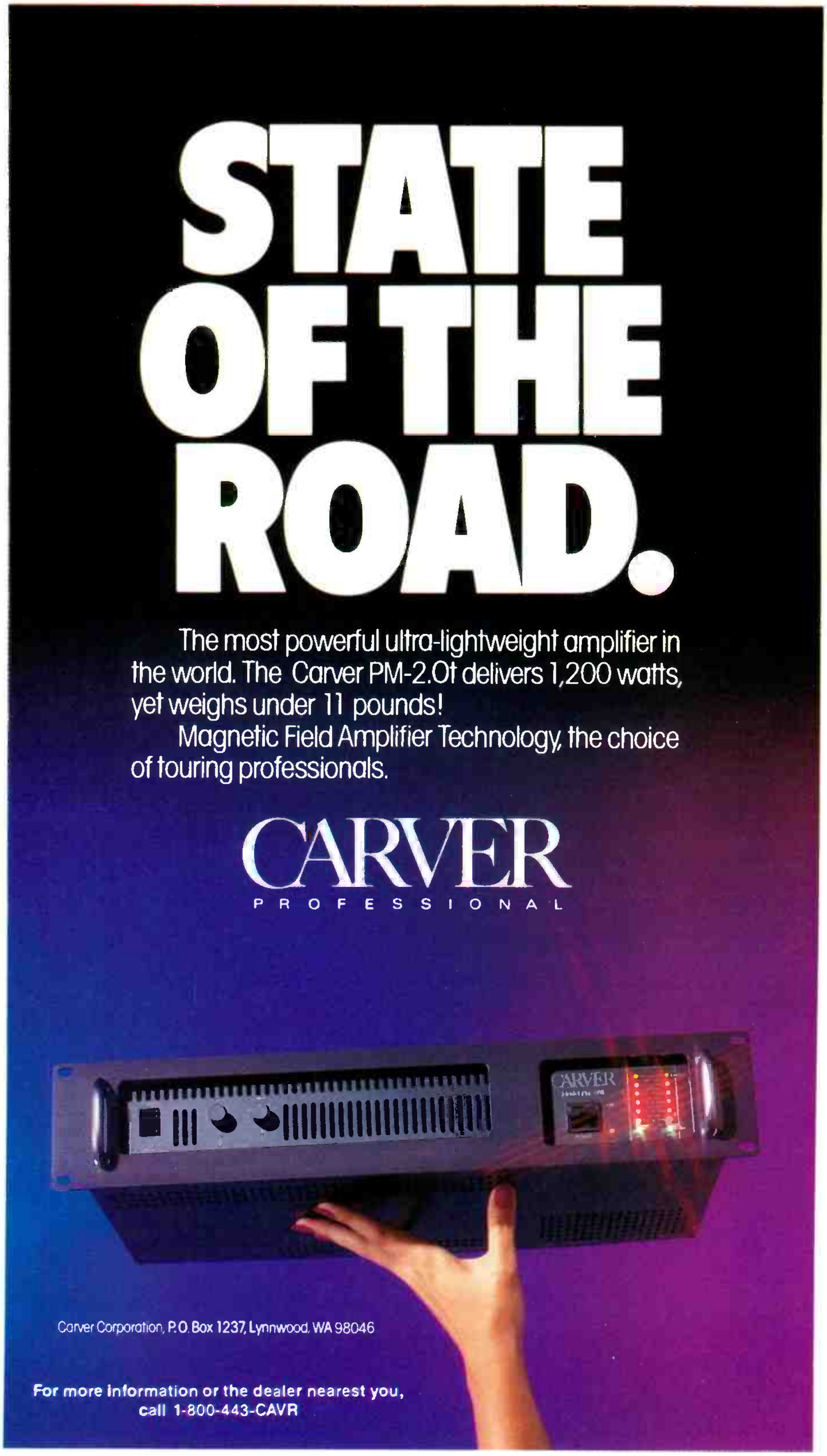
Nevertheless, Jim's understanding of the problems of MIDI automation is quite sound. But that doesn't mean I agree with his assessment of the future, that "there is a real limit to what we can expect from MIDI automation."

It's true that in the strictest parochial sense, a single MIDI cable is not nearly fast enough to carry real-time movement of 32 faders and three times as many EQ knobs. Because of this, Jim believes that systems that integrate console automation with music sequencing capabilities will not succeed, but, instead, SMPTE-synchronizable automation systems contained within the consoles themselves will have to do the job.

To me, this is a step backward. MIDI sequencers have had a revolutionary effect on music production because they can combine all of the functions of composition and performance into one user-friendly interface. The next logical step is to include the mixing process in that interface—for mixing to be forever condemned to be an entity separate from composition doesn't make any more sense than for mastering engineers, long after they've switched over to CDs, to continue to use the techniques they developed for working with vinyl.

There *are* ways to make MIDI behave. The most obvious is to use multiple MIDI lines. Today's 80386 and 68030 computers are fast enough to handle a dozen or more simultaneous MIDI cables. With as few as eight MIDI lines, a system could dedicate two lines to music, two more for system-exclusive or sample dumps, and one for MIDI Time Code, and still have three left over for automation—say, two for faders and one for mutes and EQ changes.

A corollary to this is to take advantage of Running Status, which the MIDI specification provides to allow faster transmission of redundant information. A MIDI line that is *only* handling fader



MANAGING MIDI

automation with controller commands can use Running Status all the time, which means that each command is only two bytes, not three, immediately giving the system 50% better resolution.

But automation-system designers can go even further to make their products work with MIDI. Perhaps, as Jostins begins to suggest, a hybrid scheme of real-time and non-real-time control could be developed, combining controllers, bulk dumps, MIDI Time Code cue lists, and program changes.

Although there is a common misconception that controller commands must always be sent in huge globs that choke MIDI lines, this is completely untrue.

A system like this might have intelligence that would thin the data out at moments when timing is crucial and send data more densely during less time-sensitive moments.

Again following Jostins' lead, perhaps it's not necessary for each and every position of a fader in motion to be recorded—instead, the data could be stored and transmitted in "slope/intercept" form, reflecting the distance traveled and the time taken to get there. This would require only two commands, instead of hundreds.

These are not far-fetched ideas—remember, it wasn't so long ago that everyone knew that AM radio couldn't handle stereo, or that there was no way to get high-fidelity sound on VHS videotape. It just takes the right combination of cleverness, motivation, and ambition.

MIDI's greatest asset is that it is a standard, which everybody agrees how to use. It should not be abandoned lightly; if you think that kind of acceptance is easy to achieve, talk to someone who has implemented the AES/EBU digital "standard" in a piece of equipment, and find out how few other manufacturers' devices it will actually work with! Bandwidth limitations shouldn't close the door on MIDI automation—instead, they should be taken as a challenge for designers to open new doors. RE/P

It's Not Just A Phase We're Going Through.

The tremendous success of the Tannoy PBM series of reference monitors is by no means coincidental. Since the introduction of the world renowned NFM-8 nearfield monitor, much time and effort has been spent on discerning the needs of the mixing engineer and the applied requirements of "playback monitors". The PBM Line exemplifies this commitment to excellence in reference studio monitoring. These compact loudspeakers sport robust poly cone mid-bass transducers utilizing efficient long-throw, high power voice coils. The low frequencies are carefully controlled by optimumly tuned ports located on the rear of the loudspeakers. Hi frequencies are provided by Hi Power ferro fluid cooled polyamide dome tweeters which extend H.F. bandwidth beyond 20KHZ. The driver accompaniment is knitted together by means of a precision hardwired crossover unit, utilizing robust low loss components, and heavy-duty input terminals which will accept standard 3/4" spaced banana plugs and the majority of high quality, specialist audio cables. Transducers and crossover assemblies are neatly housed in a stylish, high density, partical wrap cabinet, specially designed to minimize unwanted cabinet resonance, and high frequency reflection. In summarizing, we have left the best feature of all for last "price versus performance."

TANNOY
Professional Products

Tannoy North America, Inc.
300 Gage Ave., Unit 1
Kitchener, Ont.,
Canada, N2M 2C8
(519) 745-1158
Telex 069-55328
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AWARD WINNER
TRANSDUCER TECHNOLOGY

On-line Database Services, Part 1

In March, we talked about going on-line with subscription database services such as The Source, CompuServe, PAN and IMC. Because IMC's Esi service is specifically geared to the music industry (it's owned by *Billboard)*, we decided actually to take you on-line and show you what a computer equipped with a modem can access, besides exchanging files directly with co-workers and friends.

Be aware that these on-line systems provide services that you pay for, typically with a 1-time subscription fee, a monthly minimum and prorated charges for your actual time on-line. Often there are prime-time rates and evening rates.

Generic services of interest to the public-at-large are typically less expensive than services that specialize in a given professional area. Esi falls in the latter category, and the inherent costs are substantial enough to keep amateurs away, yet quite reasonable for music industry pros.

The fundamental costs of using Esi include a 1-time $200 subscription charge, $15 monthly service charge, U.S. connect time of $9 an hour and a communications fee of $6 an hour (or 10 cents per 1,000 words).

The chances of your being in the same metro area as the "Big Computer in the Sky" are roughly akin to those of winning $1 million in Las Vegas—starting with $5 in your pocket. So as not to incur long-distance charges, services such as Telenet and Tymenet provide a local phone number in major metro areas that acts as a data gateway to these large remote systems. The user is typically not charged for this access.

The sign-on process involves calling the local access telephone number, telling the access system what type of computer or terminal you have, specifying the service you wish to access and giving it your ID and private password. At this point, Esi presents us with a main menu. (The system prompts are typically terminated with a colon. At these points, you have to type a response to navigate through the system.)

For this exercise, the computer screens shown may not correspond with what you would see on-line, as we've taken some editorial liberties to save space. To the right of the screens, running commentary provides an explanation of what's going on.

OK, we're on-line!

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

```
[MAIN] System Menu
COMM    DIRECT    SERV    IPS    UTIL    ASSIST    GAMES    OFF
INFO)rmation, or <CR> for full menu

<COMMAND LEVEL>: {RETURN or ENTER}

[MAIN] SYSTEM MENU
COMM......Communications Services        DIRECT....Directory Services
SERV......Music Business Services        IPS.......Information Providers
UTIL......Account Utilities              ASSIST....Assistance
GAMES.....Fun & Games                    INFO......System Information
OFF.......Log-Off the System

<CL>: {DIRECT}
```

Because CR stands for "Carriage Return," we hit the return key here for a full menu. Some computer keyboards use "Enter" in place of "Return." Also, the words "Command Level" are usually abbreviated as CL.

"Direct" can be used to find a list of studios out of the more than 2,200 subscribers. We could further qualify our search by specifying all studios in New York, for example.

```
Enter your search term, F)ull, H)elp or Q)uit: {studio}
Searching for STUDIO, please wait...

Approximately 78 matches found.

Press <ENTER> to Display, or type in an additional search term,
N)ew search, H)elp or Q)uit: {NY}

Searching for NY, please wait...

Press <ENTER> to Display, or type in an additional search term,
N)ew search, H)elp or Q)uit: {ENTER}

ATLANTIC-US    IMC991     S.BRAMBERG  ATLANTIC STUDIOS/ MGR-BOOKING-SPARS/NY
CIANI-US       IMC2795    SUZANNE     CIANI MUSICA/MUSIC PRODUCTION-STUDIO/NY
SKYLINE-US     IMC988     OFFICE      SKYLINE STUDIOS/RECORDING STUDIOS/NY
SCI-US         IMC3031    DOUG        STUDIO CONSULTANTS INC/PRO-AUDIO/NYC
KAY-US         IMC3463    STEPHEN     TECHNISOUND/FAIRLIGHT/STUDIO/NY
```

"Atlantic-US" and "Kay-US" are "handles" for use when referring to other subscribers. This information is used to communicate with various subscribers via electronic mail. Now, let's return to the main menu.

```
Enter your search term, F)ull, H)elp or Q)uit: {QUIT}
[MAIN] System Menu
COMM   DIRECT   SERV   IPS   UTIL   ASSIST   GAMES   OFF
INFO)rmation, or <CR> for full menu

<CL>: {COMM}
[COMM] Communications Services Menu
MAIL      DIR      XMAIL  MAILGRAM  BBS   CHAT   ONLINE
NET-TALK  REGISTER  BIN   AA        UK    OFF    MAIN
INFO)rmation, or <CR> for full menu

COMM>: {MAIL}
Send, Read or Scan: {READ}

To: BURGER-US (IMC9006)
From: REP-US (IMC822) Delivered: Thu 23-Feb-89 18:50 EST
Subject: April Computer Column

--More-- {RETURN}
```

Now we'll check our electronic mailbox.

The "More" prompt gives us the option of continuing with the subject or escaping.

```
Jeff... deadline is approaching. How's April's "Understanding Computer" column looking? - Mike

Disposition: {RETURN}

Send, Read or Scan: {SEND}

To: {REP-US}
Subject: {Keep Your Pants On!}
Text: {Mike... Finishing the column now. Will send it tomorrow to your Esi electronic mailbox. - Jeff
.S}
REP-US -- Sent
Send, Read or Scan: {QUIT}

[COMM] Communications Services Menu
MAIL    DIR        XMAIL     MAILGRAM   BBS    CHAT    ONLINE
        NET-TALK   REGISTER  BIN        AA     UK      OFF
        MAIN
INFO)rmation, or <CR> for full menu
```

"Disposition" allows us either to save or delete the message. To save, type Q for "Quit." To delete, type D.

".S" is used to signify the end of the message so that the system can upload it. An important concept here is that most communications software allows you both to capture the content of an on-line session into a text file, as well as send a text file. "Text capture" was used to record the session you are reading, which was subsequently edited in a word processor. File transmission is what I use to place a file, such as this story, in RE/P's *mailbox. Once the file has been uploaded, the "sent" message appears, as shown above. To exit the mail service, respond to the next prompt with "Quit."*

```
<COMM>: {BBS}

ESI Bulletin Board System
S)end     R)ead     SC)an    C)ategories     Q)uit     H)elp

Command: {SC}

Available Categories:
A> ADS: EQUIPMENT      B> ADS: SERVICES       C> ADS: FOR SALE
D> ADS: WANTED         E> ADS: JOBS OFFRD     F> ADS: JOBS WANTED
G> FORUM: AUDIO        H> FILM & VIDEO        I> ADS: FACILITIES

Selected areas: {A}

ESI Bulletin Board System
S)end     R)ead     SC)an    C)ategories     Q)uit     H)elp

Command: {SC}

F)wd, B)ack, N)ew or H)elp: {F}
```

If you are in the market to buy a used console, the bulletin boards offer services such as classified ads.

```
19  MACDONALD-US (IMC *ADS: EQUIPMENT* NEVE EQUALISERS
38  SEA STAGE-US (IMC *ADS: EQUIPMENT* STAGES FOR SALE
66  MATTHEWS-US (IMC5 *ADS: EQUIPMENT* MIDAS CONSOLE FOR SALE
88  CUNNINGHAM-US (IM *ADS: EQUIPMENT* TASCAM 85-16B 16 TRK FOR SALE

R)ead or SC)an: {R 66}

Msg#:     66  *ADS: EQUIPMENT* (Read 36 times, 0 replies)
From:     MATTHEWS-US (IMC590) Posted: THU, JUL 28 1988 15:16
Subject:  MIDAS CONSOLE FOR SALE

--More-- {RETURN}
                              FOR SALE
                       MIDAS 36/8/2 HOUSE CONSOLE
<<At this point you would see a detailed description of the hardware that is for sale. When you are finished
reading the ad, respond to the next prompt.>>

R)eply, N)ext, S)top, or Msg #: {S}
```

At this point, you would see a detailed description of the hardware that is for sale. When you are finished reading the ad, respond to the next prompt.

Next month, we'll continue by exploring value-added services.

RE/P

THE IEQ WITH Smartcurve™

Not Just Another Programmable Equalizer With MIDI™

The IEQ with Smartcurve™ is a programmable, high performance graphic equalizer that includes a video output. For those who wish to enjoy the video output of the IEQ, ART makes the IEQ Video Monitor.* Smartcurve™, proprietary software developed by ART gives you instant actual frequency response as easy as the push of a button.

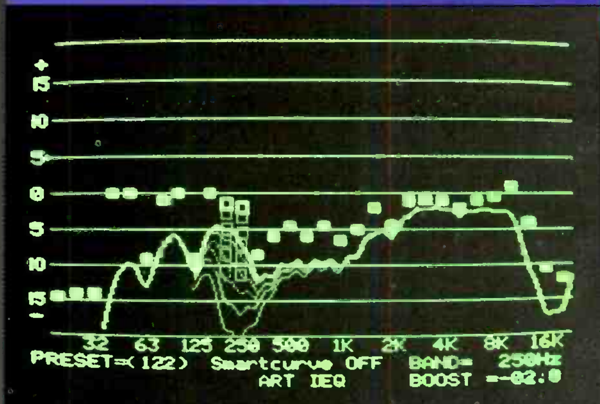
The IEQ Family consists of both the 2/3 octave and the 1/3 octave graphic equalizers. Both types come in two varieties, Controllers and Satellites. A Controller is a self-contained programmable intelligent graphic equalizer capable of controlling 15 satellites at once. IEQ Satellites are exactly the same unit except the front panel controls are eliminated.

IEQ Model Specifications:
Controller & Satellite

- 128 battery backed presets
- MIDI
- Frequency Response 20Hz-20kHz ± 0.5dB
- THD - ≤ .009% @ 1kHz, 0dBM typical
- Dynamic Range - ≥ 100dB typical
- Balanced inputs and outputs

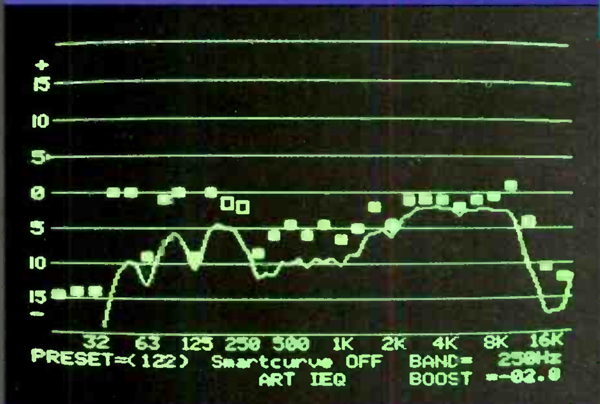
***IEQ Video Monitor Features**

- 19" rack mountable
- NTSC compatible monochrome monitor
- 4 Selectable inputs
- Standard RCA jacks for easy connections



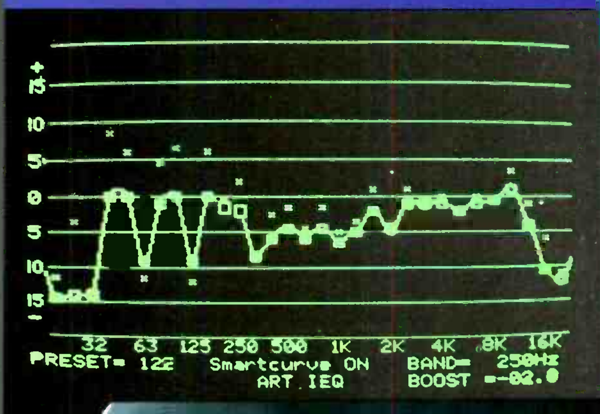
1 See the Sound

This is a video output of the IEQ as the unit is being adjusted. The sliders can be moved ± 15dB in 1/2dB steps to get the exact response you need. With the simple push of a button, complex equalization can be done in seconds with incredible accuracy.



2 Hear the Sound

The power of the IEQ readily becomes apparent as the video display plots the frequency response due to the slider settings. The IEQ offers high quality constant "Q" equalization. The video graphic display shows the correlation between the sliders and the frequency response.



3 Perfect Sound

Turn Smartcurve™ on and perfect equalization is at your fingertips. Note the difference between the second frame, (Smartcurve™ OFF) and this frame. The position of the sliders represent the actual frequency response of the EQ. Interaction between bands is virtually eliminated. Incredible! Just think now when you adjust the EQ you get exactly what you need. The "perfect" EQ? Let your eyes and ears decide.

It Has To Be A Work Of . . .

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Subwoofers and Sound Systems

By Mark E. Engebretson

The advent of the electronic keyboard and its immediate acceptance have revolutionized very low-frequency acoustic output requirements for sound systems.

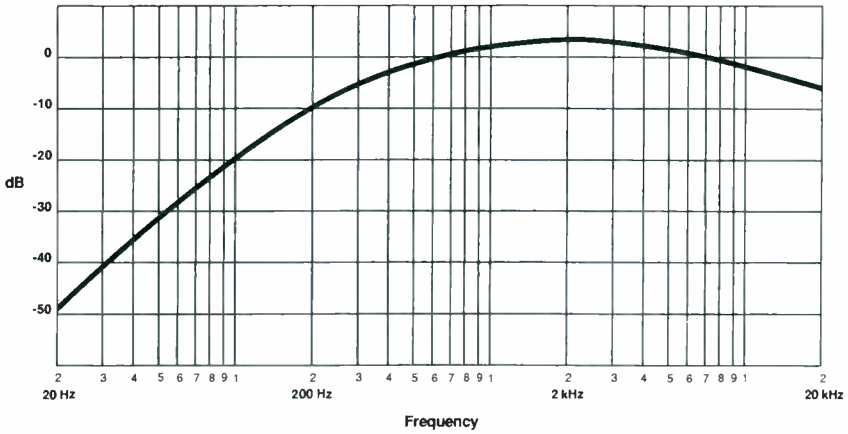

Figure 1. *A-weighted response. The sensation of loudness depends upon the frequency and magnitude of the sound. Most hearing-damage-risk criteria is presented in dBA, which is a weighting scale that rolls off the low frequencies.*

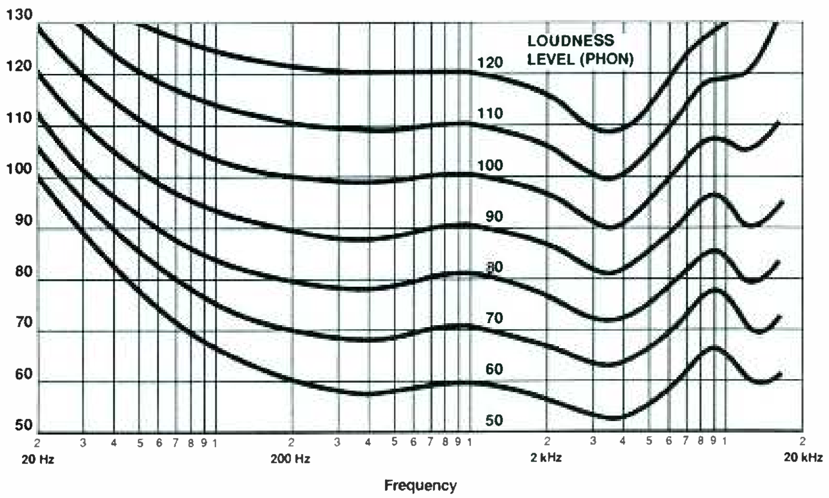

Figure 2. *ISO equal loudness contours.*

More than any other factor, keyboard instruments have been responsible for the rapid growth in concert sound reinforcement systems and have precipitated development of dozens of products intended for subwoofer applications. In the shadow of these developments, however, all is not well.

The concert sound reinforcement industry is caught between mounting client demands for more powerful musical communications on one side, and public outrage and impending regulatory intervention over high sound levels on the other. Seemingly insatiable demands for very low frequencies emerging in musical performance have not been widely understood or satisfied.

Sound system practices

Commonly accepted sound reinforcement practices indicate a general lack of knowledge of and concern for human auditory factors by parties on both sides of the microphone. Demands for ever more powerful musical communications have been widely misunderstood to mean more SPL, often resulting in too many of the wrong types of loudspeakers, yet not enough energy at very low frequencies, where it is needed.

A common complaint is that the sound is too loud, and it certainly is in many cases. Enormous demands for low-frequency energy imposed by new instrument technology and musicianship have

Mark Engebretson is principle consultant for Electroacoustic Technology and co-owner of Summit Laboratories in Warner Springs, CA.

In The Past We Had A Big Advantage Over The Competition.

Now We've Got A Small One.

Until UREI's 813 Time Align® Monitor entered the studio, speaker systems had become a "smear" on the industry. A "time smear," in which high and low frequencies subtly assaulted the ear because they arrived out of sync. The results were general listener fatigue and unrealistic sound, particularly on lead instruments and vocals.

The UREI 813 solved the "time smear" problem with Time Alignment™, unifying sound into a single point source. This dramatic breakthrough, along with other major technical advances, soon established the 813 as the industry standard.

Now UREI introduces less of a good thing: the 809 Time Align® Studio Monitor. The 809 delivers all the engineering depth of its big brother, but at a compact size and price that's ideal for small control rooms and near-field applications.

UREI's 809 features a remarkable, all-new 300mm (12") coaxial driver that achieves a true one-point sound source, superior stereo imaging, and tight bass. It incorporates a unique titanium diaphragm compression driver that unleashes unequalled high frequency response.

The 809 has exceptional power handling capabilities, high sound sensitivity, and low distortion. It accomplishes precise acoustic impedance matching and smooth out-of-band response with UREI's patented high-frequency horn with diffraction buffer. And its ferrite magnet structures assure the system's high sensitivity drivers will not degrade with time and use.

UREI's Model 809 Time Align® Studio Monitor. Smaller package. Smaller price. Same impeccable "813" sound quality. See how it fits into your studio today.

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UREI

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driven system designers to use loudspeaker arrays that are too large to radiate coherent information, except at low frequencies.

The human auditory system cannot reconstruct complex sounds that have been distorted in amplitude and frequency-dissected spatially over time. The sound from multiple directional sources separated by finite delays is smeared, inconcise and distorted from normal structure and harmonic content.

Distorted sound is always too loud, but when the distortion is caused by time-smear, the sound level is never loud enough to be clear. Most sound mixers attempt to cut through the confusion by "mixing up." This explains why humans, otherwise highly sensitive to musical balance and detail, seem so heavy-handed at the controls. It isn't our fault. Increasing the sound level when sonic details cannot be discerned is a natural response, but it will never unmask the unintelligible. Time-smear distortion can be reduced only by eliminating sound sources.

This is not to suggest that everyone will live happily ever after if three-fourths of the speaker systems are left on the loading docks and the house-levels are turned down. High SPLs at very low frequencies are absolutely essential in today's music, yet the majority of sound systems in use have far more mid-high information than needed, but not enough *real* bottom. The lowest frequencies demand the greatest acoustic output from the sound system, and high acoustic levels at very low frequencies have traditionally required large transducer and cabinet systems.

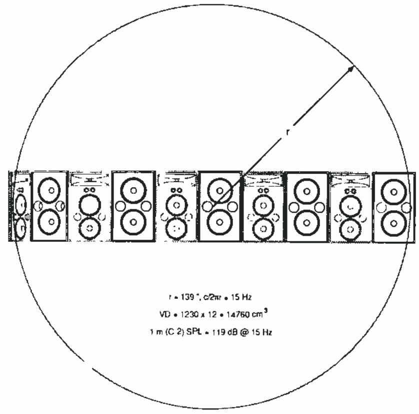
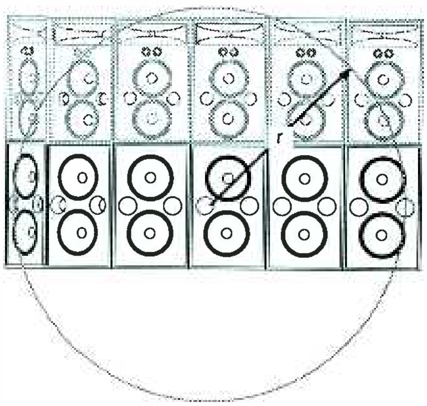


Figure 3A. *An array that alternates subwoofers with full-range boxes.*



r = 76", c/2πr = 28 Hz

VD = 1230 x 12 = 14760 cm^3

1 m (C-2) SPL = 130 dB @ 28 Hz

Figure 3B. *The components of Figure 3A in a better array.*

Designing for loudness

Sound reinforcement systems should be capable of radiating acoustic information that is a faithful energy, frequency and time replication of the electrical signal being amplified. The modern sound system must also be a musical instrument with the potential to produce a nearly infinite range of sounds and musical textures. Electronic instruments impose special new requirements, as their acoustic performances are *created* by sound systems.

Loudness discomfort thresholds are frequency-dependent. Because our hearing is least sensitive at the lowest frequencies, we can comfortably tolerate greater sound pressure levels at low frequencies than at higher frequencies. Most hearing-damage-risk criteria are presented in dBA, which is a weighting scale that rolls off the low frequencies, as shown in Figure 1. (Readers: We encourage you to listen to a wide-range program through an A-weighting filter to determine your own discomfort thresholds in dBA, and to observe that comfortable listening conditions return when the filter is removed, without changing A-scale sound pressure levels.)

Figure 2 depicts the upper range of equal loudness contours for pure tones in a free field, as compiled by Robinson and Dadson in 1956 and adopted as ISO R-226 in 1961. In the absence of more definitive data, this information is a useful indicator of the range of sound pressure levels at various frequencies desirable for the creation of musical communications.

Assume that our desired maximum A-scale sound pressure level is 100dBA, which roughly corresponds to 100dB SPL at 1kHz. The contours show that the 30Hz frequency region requires approximately 120dB of sound pressure level to be perceived as equal in loudness to 100dB at 1kHz. It is safe to assume that we are at least as tolerant of sound pressure levels for similar loudness at different frequencies as we appear to be sensitive. Our experience supports this thesis. To most people, 120dBA is painful, but 120dB SPL at 30Hz, seldom achieved with sound systems, can be very pleasant.

Toward the upper range of sound pressure levels, the sound system should be capable of 20dB greater acoustic output at 30Hz than at 1kHz. The need for expanded capabilities begins just below 200Hz and increases approximately 6dB per octave. The distinction between sound system *capabilities* and *characteristics* in this context is important. Sound systems require relatively flat frequency/amplitude characteristics for accurate reproduction.

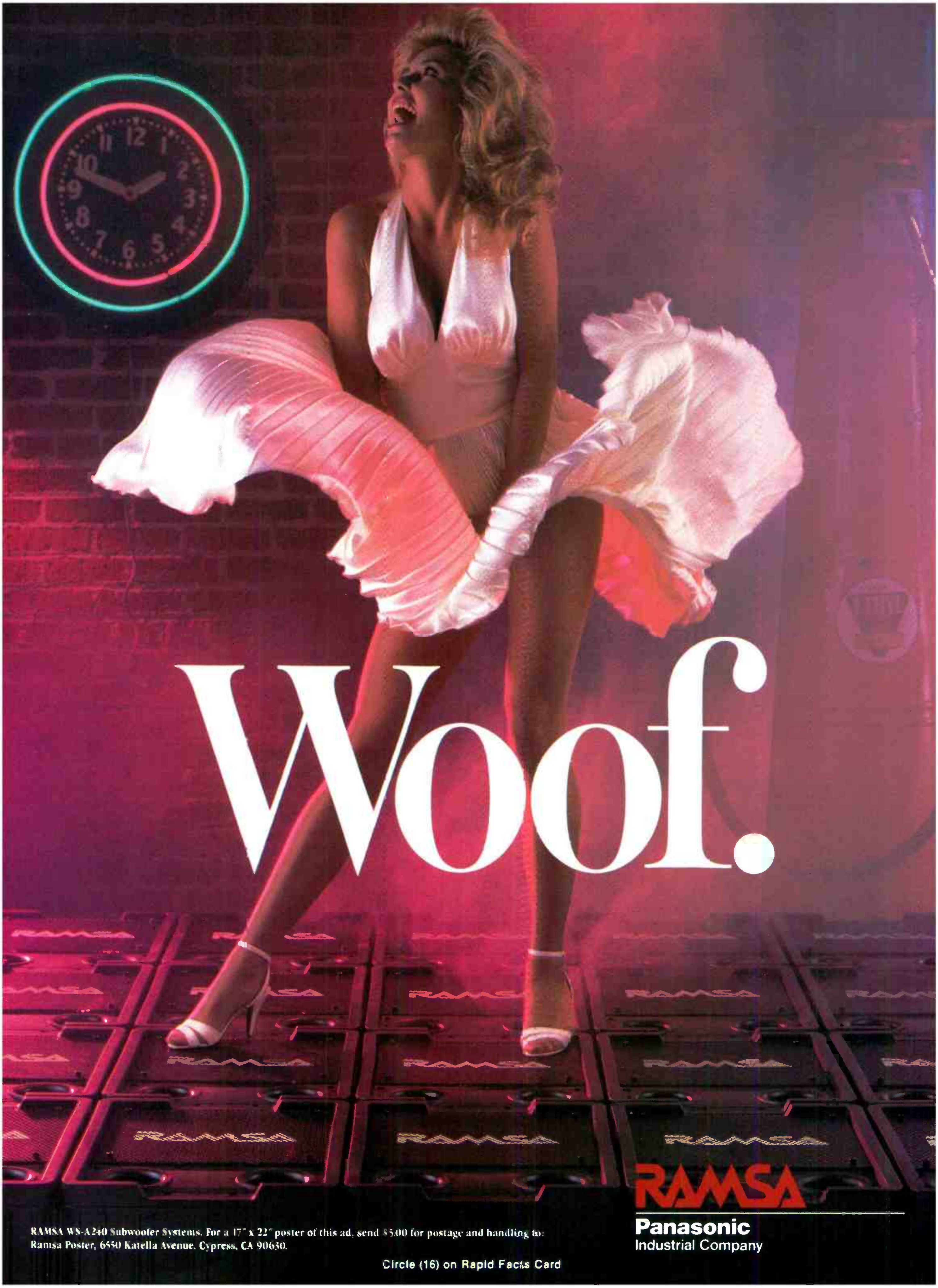
Because mixing and balancing are based on human loudness perception, when very low-frequency program is present, the sound system must have the *capability* of delivering 20dB greater sound levels at 30Hz than at mid-frequencies. 20dB represents a 100:1 power ratio. Clearly, the performance and physical requirements for subwoofer systems differ radically from those of upper-range transducers.

Determining subwoofer requirements

In spite of the wide acceptance of subwoofers, very few sound systems make optimal use of them. There is no magic to the design and use of these systems, however. When proper design and application criteria are used, direct radiators are very predictable and capable of high performance. The objective is the greatest possible linear acoustic output from the lowest frequency of interest to the highest attainable—while minimizing the overall size of the subwoofer array. First, we must understand the transducer criteria necessary to produce desirable and predictable results. (Bear with me, this won't take too long.)

In sinusoidally vibrating direct radiator systems under half-space conditions, SPL is proportional to the volume displacement and the square of the frequency. Volume displacement (V_D), the product of projected-diaphragm surface area and the 1-way displacement, is the factor that limits output capabilities at very low frequencies—independent of sensitivity, power handling and frequency response factors. The greater the volume displacement, the more energy radiated.

For a given diameter, the 1-way displacement (X_{max}) specification becomes a figure of merit, ensuring that other performance characteristics are comparable. (Readers should confirm that specified X_{max} values are approximately one-half the difference between a speaker's voice coil height and top-plate thickness.)



Woof.
RAMSA
Panasonic
Industrial Company
RAMSA WS-A240 Subwoofer Systems. For a 17" x 22" poster of this ad, send $5.00 for postage and handling to: Ramsa Poster, 6550 Katella Avenue, Cypress, CA 90630.
Circle (16) on Rapid Facts Card

Table 1. Sub-woofer performance in typical installations.

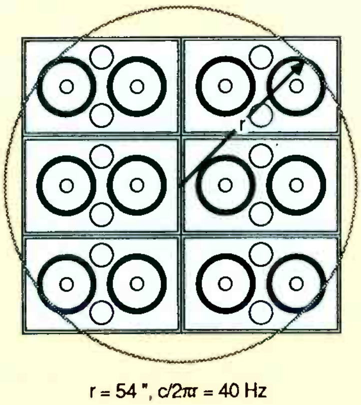
Description	R (room constant in m²)	SPL rev
Medium Auditorium	743	124dB
Performing Arts Center	1,755	120dB
Large Sports Arena	4,950	116dB
Domed Stadium	23,175	109dB

Packaged Subwoofer Systems

A great number of packaged subwoofer systems are available. Most are direct-radiating 15- or 18-inch transducers in large, vented plywood cabinets. One alternative to the genre is the Ramsa model WS-A240 subwoofer system and the companion subwoofer processor, model WS-SP-2.

The WS-A240 is a 12-inch transducer in a small, vented, high-impact resin enclosure. According to the manufacturer, the WS-A240 was designed *backward* by starting with the desired cabinet size and performance, and developing a family of Thiele-Small parameters for the transducer to fit. Because a 12-inch speaker has approximately one-half the projected diaphragm surface area of an 18-inch speaker, comparable volume displacement could be realized from a single 18-inch or two 12-inch devices, given similar cone displacements. Theory says the system efficiencies should also be similar, given comparable materials and construction techniques. Figure 4 shows an array of 12 high-quality 18-inch transducers and the array capabilities, while the comparable group of 24 12-inch speakers is shown in Figure 5.

Exercising design options is a matter of balancing trade-offs to achieve a set of desired results. This Ramsa device is said to deliver approximately one-half the volume displacement of comparably built 18-inch units, but needs roughly one-fourth the cabinet volume to do so. Dissipating the power across two voice coils instead of one should provide a greater margin of thermal safety with less power compression at high output levels.

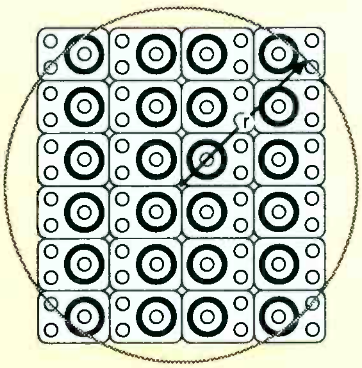


r = 54", c/2πr = 40 Hz

VD = 1230 x 12 = 14,760 cm³

1 m (C-2) SPL = 136 dB @ 40 Hz

***Figure 4.** Boxes clustered in a tight array for a 10Hz increase in bandwidth.*



r = 54", c/2πr = 40 Hz

VD = 600 x 24 = 14,400 cm³

1 m (C-2) SPL = 136 dB @ 40 Hz

***Figure 5.** An array of 24 WS-A240s.*

Solving for equivalent sound pressure level at 1 meter, we have:

sound pressure level (in decibels) =

$$20 \log \left[\frac{f^2 V_D}{3.7}\right]$$

where: V_D = transducer volume displacement, in cm³

For constant SPL, volume displacement quadruples for a 1-octave frequency reduction. Similarly, limited volume displacement yields a slope of 12dB per octave.

Doubling the volume displacement quadruples the sound pressure level. The relationship assumes half-space conditions, provided to 30Hz by a baffle with a perimeter of 38 to 40 feet. However, smaller systems require one nearby boundary to yield half-space conditions at very low frequencies.

The upper frequency limits for flat power are dictated by the effective circumference of the transducer. At higher frequencies, where wavelengths become shorter than the circumference of the radiator, pressure changes near the surface of the diaphragm occur faster than they propagate clear of the device.

Because the diaphragm surface is perpendicular to the direction of normal radiation, cancellation begins at 90° to the radiation axis, narrowing the directivity and reducing the power output at a rate of 6dB per octave-increase in frequency. The frequency where these cancellations begin is termed "normalized frequency."

Arrays

A single direct radiator can be described as a piston of finite dimensions, consisting of an infinite number of infinitesimally small sources, all vibrating in phase and with identical velocity amplitudes. If this is so, multiple direct-radiator systems may be described as a single piston, provided the array is of finite dimensions and all of the component parts vibrate in phase and with identical velocity amplitudes. This simple convolution takes the phenomenon known as "mutual coupling" out of the Twilight Zone. We need to define conditions under which this proposition is valid, however.

It doesn't matter how many transducers are used to achieve the volume displacement, doubling volume displacement will yield 6dB more sound pressure level and acoustic power, assuming that the frequency is low enough for coherent radiation from all of the diaphragms. Doubling the volume displacement in a single device can only be achieved by quadrupling the electric power (assuming the volume displacement capacity exists).

However, if a second identical device is

The Bose® Acoustimass® Professional Powered Speaker System

You're looking inside the new technological flagship of the Bose Professional Products family—the Acoustimass Professional powered speaker system. It incorporates almost everything Bose has learned about professional sound. We submit that this system, created by the fusion of multiple technologies, will establish a new standard of performance, convenience, and reliability. Most important, here's what this system means to *you:*

Just hang it, connect it, and aim it.
The design of the Acoustimass Professional system takes this important fact into consideration: your time is money. So, all system electronics, including the amplifier and loudspeaker equalization (1), are right where they belong: with the speaker. Forget about spending your valuable time building heavy electronics racks. Everything you need for sound amplification is in one compact, self-contained package that's easy to transport and install. The enclosure's integral hang points make hanging or flying the 80 pound cabinet a simple task.* Setting up the Acoustimass Professional system is as easy as plugging in AC power, connecting line-level from the signal source, and aiming the system.

Clear, natural sound for a variety of applications.
Besides convenience features, the Acoustimass Professional system contains two new proprietary Bose audio technologies: *Acoustimass speaker design and the Two-State Modulation amplifier.* Together, they deliver performance far beyond what you'd expect from such a compact, manageable package. Rather than relying on vibrating cones to directly radiate sound like conventional speakers, the patented Acoustimass speaker uses two masses of air to launch sound into the listening environment (2), dramatically reducing distortion. The unique circuitry in the system's patented Two-State Modulation amplifier delivers enough clean power to meet the needs of almost any sound reinforcement situation. The result: *Purer sound and increased bass output* at nearly any volume level—all from a speaker that's a fraction of the size and weight of conventional designs.

Use it almost anywhere—you can *count* on it.
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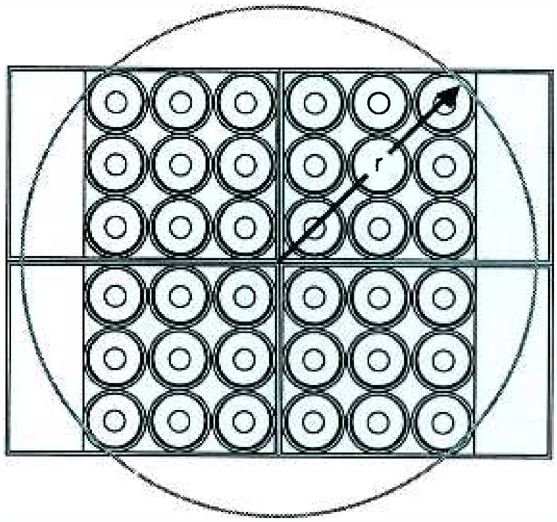

Figure 6. Hypothetical VLF array.

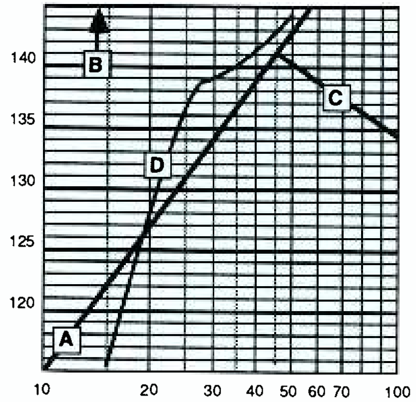

Figure 7. Plotting performance limits.

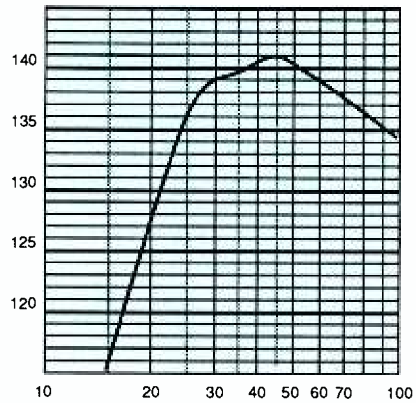

Figure 8. Summarized results.

added (a 3dB increase in electrical power), the volume displacement will be doubled and a 6dB increase in level will be realized. This is like winning the lottery. Half of the increase is due to the additional electrical power, while the remaining 3dB is a gain in efficiency. Like the lottery, however, there is a price to pay for winning—the laws of physics don't give anything without taking something in return. *As the array gets larger, its upper-frequency limit decreases.*

Low-frequency arrays quickly reach a point of diminishing return when the surface area required for transducers brings the highest frequency of coherent power summation into the operating frequency range of the array. As with a single transducer, an array suffers from transverse cancellation above the normalized frequency, but the frequency is much lower. The bigger the array, the lower the normalized frequency. The normalized frequency for an array has a wavelength that encircles all of the component radiators. Figure 3 shows three different subwoofer arrays and their normalized frequencies. The imaginary circles surrounding the transducers contain the most distant radiating elements. Their circumference represents the wavelength of the normalized frequency for each array.

The array shown in 3A alternates subwoofers with full-range boxes. The most distant transducers are 24 feet apart, which brings the highest frequency for coherent summation from all of the radiators down to 15Hz. Figure 3B shows the same components in a better array, and Figure 4 clusters the boxes in a tight array for a 10Hz increase in bandwidth.

If each component in the array is to be a self-contained ported-box system, the overall size of the array and the spacing between its most widely spaced components obviously increases, decreasing the upper-frequency limits of coherent power summation. This suggests that subwoofer array components should be as *small* as practical, while attempting to satisfy the volume displacement needs of the application. This is not a simple task. It involves much more than just stacking boxes and adding watts.

Factors affecting subwoofer performance

Subwoofer transducers are specialized devices, with unique features unsuitable for higher-frequency operation. They must produce high volume displacements to achieve the desired acoustic output at the lowest frequencies. The voice coil, magnetic system and suspension components must be designed to allow large diaphragm movements, while maintaining a linear relationship between input voltage and displacement. Large displacements require a diaphragm of high internal stiffness to prevent spurious radiation; wider-than-normal voice coil gaps to prevent rubs; and robust voice coils for continuous high-power, high-temperature operation. These factors all increase moving mass, which results in lower efficiency than for other transducers of comparable diameters. High efficiency and sensitivity are definitely *not* desirable attributes in a subwoofer transducer.

Sensitivity and power-handling data are essential to determine certain subwoofer systems' operating parameters and for planning amplifier requirements, but they should not be used in the traditional manner to determine performance capabilities. Sensitivity is generally specified at frequencies well above the subwoofer range, while efficiency specifications (which should be dimensionless) can spring from almost anywhere.

Designers should confirm that numbers derived from manufacturer's data represent characteristics below the normalized frequency for the device. Specification data for 1W, 1-meter sensitivity is usually reliable in the 200Hz region. This should be converted to 1W, 1-meter volume displacement (V_D) value, which can be readily transposed for other frequencies and power levels. Volume displacements can then be added directly to calculate array performance capabilities below the normalized frequency.

Improvements in adhesive and voice coil technology have given transducers thermal power capabilities well beyond their limits of linear operation. Different rating methods and raging numbers battles make calculation of thermally limited maximum SPL a matter of experience and interpretation. Manufacturer's power capacity data should only be used to indicate the electric power needed to break the device when driven by a signal similar

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AMS Industries plc

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New Technologies for Improved Low-Frequency Reproduction

By Thomas Danley

In the 60-plus years since the invention of the voice coil loudspeaker, loudspeaker manufacturers have often justified the cut-off frequencies of their low-frequency drivers by saying, "Below that frequency, there is little program information anyway."

To a certain degree, because of low-frequency limits in the recording chain, that has been true. But too often these arbitrary cut-off frequencies were chosen based on listening tests shaped by the displacement capacities of the drivers that were under test.

Recently, Dolby Labs took a look at the question of adequate low-frequency performance from a different angle, asking "What can we hear?" And because they are concerned with movie soundtracks, they included environmental sounds as well as musical program material. (See AES preprint #2537 from the 83rd AES convention, New York, 1987.)

Among other things, their work revealed that the ear's low-frequency response limit depends on a signal's loudness, which explains why it is pointless to have extended low-frequency response if a driver cannot move enough air to be audible at those frequencies. The tests showed that even 3Hz is audible if produced at 120dB, but at the same time, the ear's sensitivity to harmonic distortion tends to increase as the low-frequency limit decreases.

Producing sufficient quantities of low-frequency sound with low distortion is difficult. The amount of air that must be moved to maintain a fixed SPL quadruples for each 1-octave decrease in frequency. Also, the ear's sensitivity to these frequencies decreases at 12dB to 18dB per octave, so to maintain a constant *apparent* loudness requires vast increases in displacement.

Thomas Danley is an acoustic engineer with Intersonics in Northbrook, IL.

The commutated voice coil

The commutated voice coil is a new approach in which power does not flow through the entire voice coil winding. Instead, the power flows through electrical contacts, touching the coil just above and just below the magnetic gap. Only the portion of the coil that does the work gets the amplifier power. This is said to reduce "loss" resistance and increase coupling between the amplifier and the electromechanical conversion process.

In pursuing this technology, we realized that a low-inertia dc servomotor—like those used in computer tape transport capstans—was the rotary equivalent of a commutated coil. The motor can rotate in either direction indefinitely, and once the rotation is converted to linear motion, the servomotor becomes, in effect, an infinitely long commutated voice coil.

Servodriven loudspeakers

The Intersonics SDL-5 is an example of a servomotor-driven loudspeaker configured for low-frequency horn loading. In the SDL-5 loudspeakers, a typical motor consists of four parallel sets of current paths located in four magnetic fields. The current paths are actually sets of coils attached to the output shaft. As the motor rotates and one of the coils begins to leave the magnetic field, carbon brushes switch the current in that branch to the adjacent coil, just entering the magnetic field.

The audio signal applied to the motor causes the output shaft to oscillate, with the rotational velocity proportional to voltage, and torque proportional to current. "In and out" cone motion is achieved by a rotary-to-linear converter that drives a pair of radiators. And because the radiators are in opposition, the motor is only needed to supply torque, relieving the motor shaft of side forces.

As the pipe organ builders of the past centuries knew, when produced at sufficient intensity, the psychoacoustic effects of very low frequencies are not subtle. On the other hand, until you've experienced them at adequate intensity and accuracy, you're not likely to know what your missing. Imagine hearing a portable radio and then listening to a real hi-fi system. What had satisfied you before just won't do anymore.

---

to that used for the rating. In subwoofer applications, the continuous thermal power capacity can be as little as one-half that of shaped-noise or "program" ratings.

Power ratings cannot be converted into dBW and added to sensitivities to indicate maximum SPL capability. Because voice coils increase in dc resistance when heated at high power, the acoustic output will be compressed. Although there will be some unit-to-unit variation at 100W input, typical subwoofer transducers (rated at 400W to 600W) show approximately 1dB power compression after only a few seconds operation. Designers should provide at least the manufacturer's rated power for each transducer. But they should expect 3dB to 5dB power compression when operating at continuous power levels of 200W or more into each transducer.

Figure 6 shows a hypothetical array consisting of 36 12-inch transducers, clustered frame-to-frame within four modules of nine units. Each transducer is capable of 600cm³ volume displacement, for a total of 21,600cm³. In a half-space at 1 meter, this equates to 139dB SPL at 40Hz.

In Figure 7, the 12dB/octave line (A) drawn through 40Hz at 139dB defines the maximum, displacement-limited, 1-meter equivalent SPL for closed-box (C-2) alignments. At 200Hz, if each transducer has a sensitivity of 91dB, 1W, 1 meter, this equals a volume displacement of 3.28cm³. Assuming 3dB of power compression at a power rating of 400W, we can expect to produce a volume displacement of 46.4cm³ from each device.

With luck, all 36 transducers will move in the same direction at the same time, resulting in 36 x 46.4cm³ or 1,670cm³ of volume displacement. Solving for SPL yields 145dB at 1 meter, which has been plotted (B). This is the maximum thermally limited 1-meter equivalent SPL for the array.

The circumference of the imaginary circle containing all of the diaphragms is the wavelength of the normalized frequency, calculated to be 45Hz. Above this frequen-

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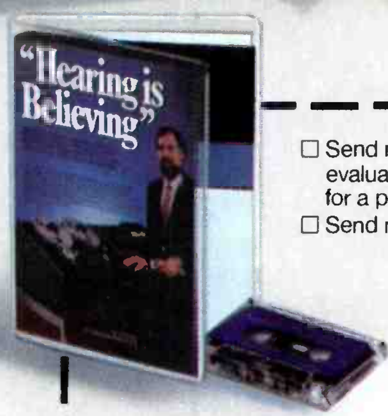

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cy, the power response *characteristic* will roll off at 6dB per octave. The 6dB-per-octave slope (C) begins at 45Hz, and is plotted where the displacement-limited or thermally limited SPL, *whichever is lowest*, intercepts the normalized frequency.

The cone displacement for flat energy in ported systems is similar to closed boxes—an octave above the region or port tuning. Beginning at approximately 1.4 times the tuning frequency, the cone displacement for flat *energy* falls off sharply as the tuning frequency is approached. Because electronic music has the capacity to produce energy below the port frequency, a high-pass filter with a slope of 24dB per octave is required. In our example, we've decided on a flat, B-4 (4th-order Butterworth) alignment with a tuning frequency of 28Hz. The resulting displacement-limited SPL has been plotted (D).

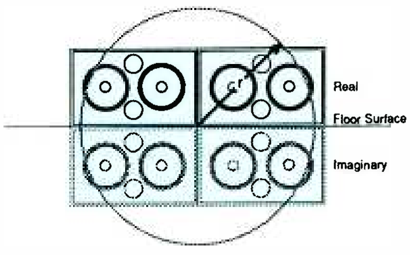
The results are summarized in Figure 8. The contour represents the maximum 1-meter (equivalent) capabilities for the array. Note that the thermal SPL limit is well removed from the displacement limits, indicating that the array will be virtually free of power compression. The 6dB-per-octave low-pass is a power response *characteristic*, and is independent of level. We will examine how these capabilities may be used.

Performance in a room

Having predicted the 1-meter capabilities for our hypothetical array, we need to know how much attenuation to expect at subwoofer frequencies to predict the performance in typical installations. Two major factors affect the level of a reverberant field: the sound power put into the space, and that which is absorbed in the space. Because our subwoofer array is a coherent source with uniform radiation at 30Hz, the acoustic power radiated may be calculated directly from the predicted half-space SPL.

While there will always be exceptions, most large music performance environments will be semi-rigid spaces with at least three-fourths of the boundaries constructed of reinforced concrete. (That fuzzy covering for "acoustics" doesn't do much at 30Hz.) Allowing for transmission loss through the roof and audience absorption, an estimated absorption coefficient of 0.2 seems reasonable. Calculations have been performed for four sample environments, with the results shown in Table 1.

Our goal is to deliver 120dB SPL at 30Hz. The data indicates satisfactory performance for the medium auditorium and the arts hall, marginal performance in the sports arena and not enough for the stadium. If a new array can be designed to double the volume displacement coherently (a 6dB increase in power), the sports arena becomes very nice; but the domed stadium will need 11dB more power, which will not be easily delivered.



$r = 45"$, $c/2\pi r = 48$ Hz

$VD = 1230 \times 8 = 9840$ cm^3

1 m (C-2) SPL = 133 dB @ 40 Hz

Figure 9. *Ground plane array.*

Application notes

Ported-box systems for high-level applications should employ a 4th-order high-pass filter at the port tuning frequency to avoid possible damage caused by overexcursion. This becomes especially important in large live reinforcement systems as the available amplifier power below the port frequency invariably exceeds the displacement-limited power capacity of the system. Accidental transients could prove hazardous to unprotected transducers.

The choice of a low-pass frequency will depend upon the specific application. Experience has shown that subwoofer systems that are low-passed between 40Hz and 50Hz can be used independently of the main system over a very wide range of levels. This enables the creation of powerful visceral effects and musical underpinnings that do not interfere with more-delicate upper-frequency textures. Assign these systems an independent mix, as a great number of sounds should be isolated from the subwoofers.

However, if the system is low-passed at too high a frequency, the subwoofers cannot be operated as independently without bringing up too much 80Hz information and sounding boomy. The system then loses some of its potential communicating power and dramatic impact.

If the console has channel equalizers with good frequency selectivity at very low frequencies (to keep from mucking up the balance in the main system), by all means, twist the knobs until the desired sound leaps out. Each instrument needs different subwoofer equalization as well as different mix levels. Bass and kick drums need very subtle subwoofer balancing to keep them from overpowering.

Subwoofers take lots of current. Provide at least the manufacturer's power rating for each transducer to be assured of a reasonably stiff power supply. Be mindful of power compression, however. Today's transducers can handle power levels 3dB to 6dB beyond their linear operating range, depending upon make and model. A good rule of thumb is to plan for 1dB compression at 100W, 3dB at 200W and nothing but heat above 200W.

Array placement

As explained, subwoofer systems perform best when configured as coherent arrays. For a stage setting, the options are limited. The floor in front of the stage, behind the security barriers (where used), is one option. In this case, the subwoofers are in quarter-space conditions. The real loudspeakers are above the floor, while a second imaginary set is located below the floor. Figure 9 shows the real and imaginary components. The system will behave as if it were a single array, except that the thermal power capacity will remain that of the real sources only.

A second option is a central overhead array, but this location may be needed for the vocal cluster or may interfere with lighting needs. Another option is behind the performers—they'd like that, if it didn't kill sight lines. You can always fall back on the split cluster, with floor placement preferred for reasons previously discussed. In this case, the direct energy from the arrays creates interference as a function of frequency and relative off-axis distance, but reverberation might disguise the problem in most of the house. For outdoor systems and wide-angle seating, don't attempt split subwoofer clusters.

Very low-frequency energy in music, once the exclusive domain of theologians and royalty, can evoke powerful emotions. The computer/digital age has brought the lowest octaves within reach of nearly every performing musician, bringing changes in the requirements for amplified entertainment. The development of powerful very low-frequency acoustical energy follows the physical laws of nature—there is no magic. Once you understand the fundamental relationships, you accurately can predict subwoofer performance.

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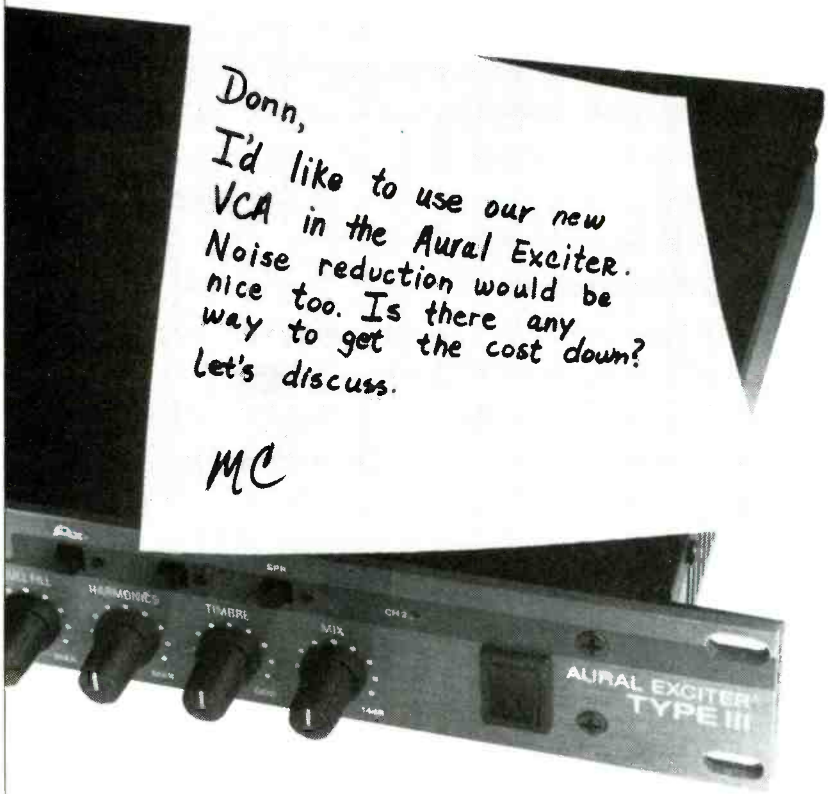

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May 1989
Recording Engineer/Producer

A Message From the President

Some years ago, while attending a staff meeting of a large arts organization, I listened to the director discuss long-term goals and directions.

"You can tell where we're going by looking at where we've come from," he said. Fortunately for the organization, he had significantly greater vision than his remarks implied, but the idea has always stuck with me. Why would anyone allow their progress in years past to determine their future direction? It seemed like piloting a boat through a mine field by keeping an eye on the boat's wake.

In truth, we most often struggle through life head-on, confronting or ducking, as the case may be. But periodically, we crest a hill and are offered a vantage point from which to view our progress, as well as the challenges yet to be met.

The 10th anniversary of SPARS is one such vantage point. As we wrestle with the evolution of digital technology, studio economics, evolving technical and business practices, and so on, we would be well-advised from this 10-year point to take a look over our shoulders and see how far we have come. To consider the wisdom of our founders in addressing the issues of our industry. To remember some tried-and-true methods of efficiently solving problems. To take inspiration from the achievements of our predecessors. To understand better how to forge on into our future.

Those who belong to and participate in SPARS owe much to the leaders and workers who brought SPARS to this point. Indeed, it would be difficult to measure how much the entire audio industry has benefited from the contributions of those people through the last decade. Now, from our vantage point, we must look at where we've come from, turn about and determine where we must go, with confidence and inspiration.

All of us at SPARS are grateful to *Recording Engineer/Producer* for offering, through the pages that follow, a thoughtful perspective on this past decade of SPARS efforts. We hope you that you will be inspired to join us and share our commitment to excellence through innovation, education and communication.

Bruce M. Merley
President, SPARS 1988-89
President, Clinton Recording Studios, New York

A project between *Recording Engineer/Producer* and the Society of Professional Audio Recording Services.

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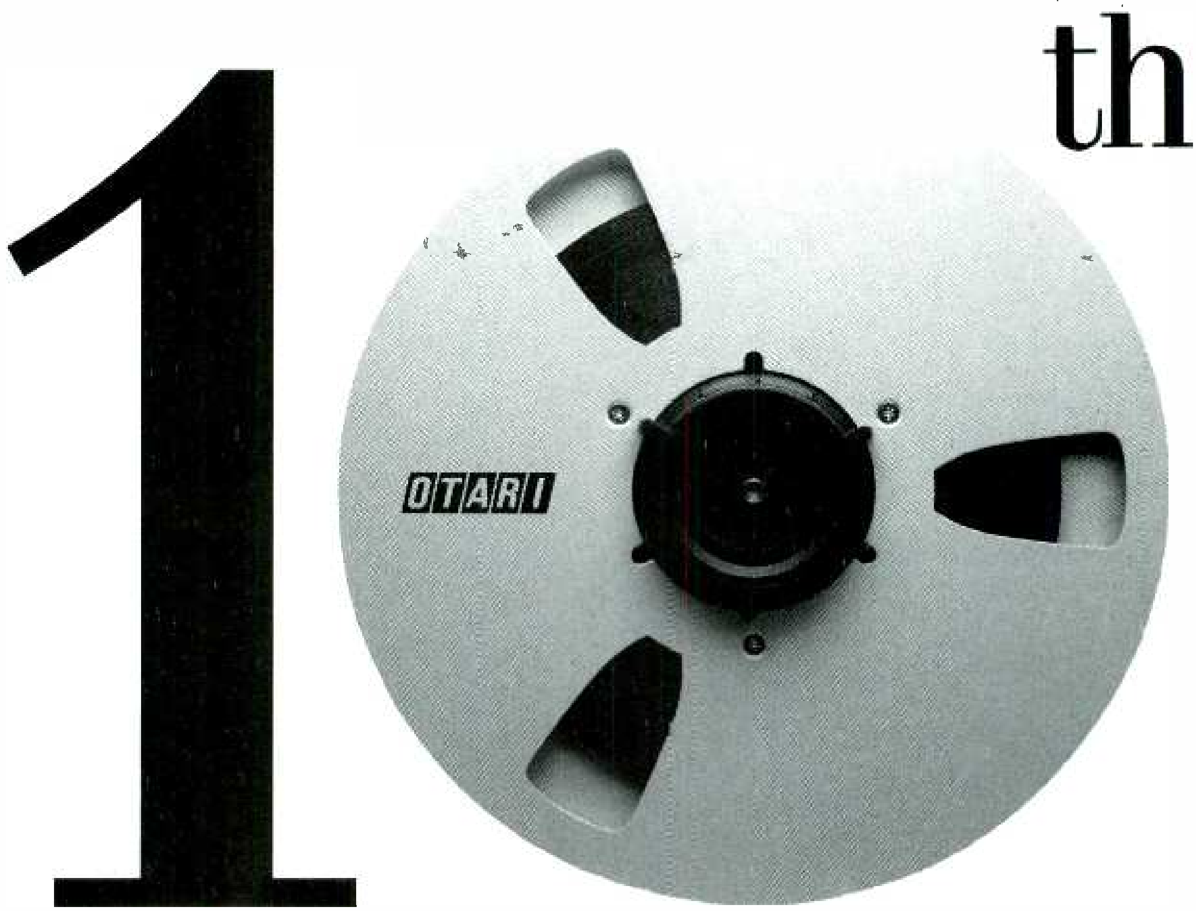

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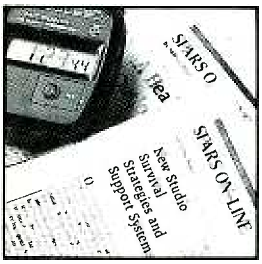
1989 Board

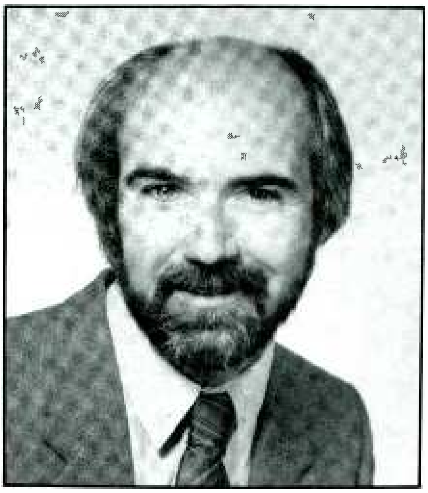
Guy Costa
Chairman of the Board

While serving in the Army Security Agency, Guy Costa was introduced to professional recording while moonlighting at Coast Recorders in San Francisco. Since then, he has held many positions in the field, including recording and disk mastering engineer, studio manager, sound reinforcement engineer, and equipment and facilities designer. He joined Motown Record Corporation in 1970 and held various positions there. He currently is CEO/owner of Quadim Corporation, Westlake Village, CA.

Bruce Merley
President

As a teenager, Bruce Merley was fascinated with electronics. This led to electrical engineering studies and a stint with General Motors. After completing a bachelor's in music, he accepted an administrative appointment at the Yale School of Music. Later, he was an administrator with the Fine Arts Museums of San Francisco. Returning east in 1980, he started KMC Records, a custom-products record label. In 1983, he founded Clinton Recording Studios, New York, with Ed Rak.



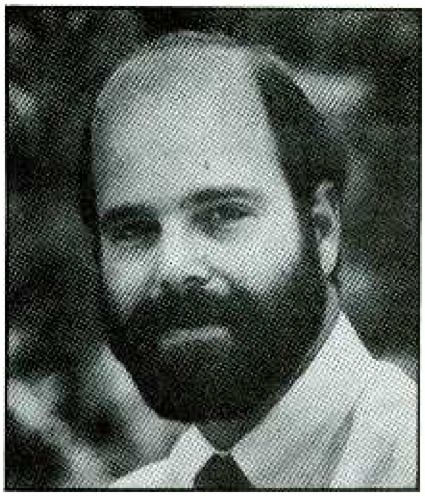


David Porter
First Vice President

The founder of Music Annex, David Porter has bachelor's and master's degrees in music. His master's program included constructing a recording studio as part of the thesis project. What started out as a home studio has evolved into two 24-track music studios, an 8-track production studio, a cassette mastering suite, a video soundstage and a large-scale cassette duplication plant located in Melno Park, CA. In 1986, Music Annex expanded into San Francisco by building two audio post rooms.

Richard Trump
Regional Vice President/Treasurer

Richard Trump is president and general manager of Triad Productions in Des Moines, IA. He majored in electrical engineering, music and telecommunicative arts at Iowa State University. After managing a high-end stereo store and an 8-track studio, he co-founded Triad in 1974. In addition to operating three studios with four, eight and 24 tracks and high-speed duplicating, Triad also operates a custom audio/computer control division.



Charles Benanty
Regional Vice President

The president of Digital Music/Pictures, Charles Benanty is comfortable in both the studio world and the academic world. His Soundworks Studios, New York, was the first digital studio in town and the first music recording studio in the country to incorporate video production in its services. In the 10 years it has been in business, Soundworks has been awarded 25 gold/platinum albums, six Grammy nominations, two Grammies and one Academy Award nomination.

Dwight Cook
Regional Vice President/Secretary

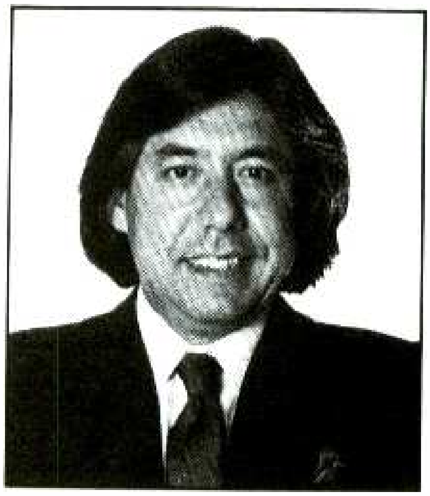
The president of Cook Sound Productions, Houston, Dwight Cook is comfortable on both sides of the glass. An active voice talent/actor, he also engineers when the time allows. He has held a variety of radio positions in sales and talent in Tampa, Miami, Chicago and Houston. He has served as president of the Motion Picture Council of Houston and as a board member of the Houston Academy of Motion Pictures.

Pete Caldwell
Regional Vice President

Pete Caldwell has pursued a varied and sporadic musical and technical career. After graduating from Davidson College with an economics degree, he spent the next four years involved in dubious musical endeavors, the least musical of which being the U.S. Army. In 1969, he co-founded Doppler Studios in Atlanta, producing music for ads, television and other commercial endeavors. Today, the Doppler facility includes five 24-track rooms with an emphasis on audio-for-video post-production.

Charles Comelli
Regional Vice President

As the U.S. director of recording studios for Capitol Records, Charles Comelli is responsible for all studio and production operations for Capitol and EMI-Manhattan studios and video production. Born in Japan, Charles began his career as a producer. He later co-established a production company that promoted Japanese concert appearances for such artists as Buddy Rich and the Ventures. He moved to the United States in 1969 and joined Capitol the following year.



Tom Kobayashi
Regional Vice President

As vice president and general manager of Sprocket Systems, San Rafael, CA, the post-production facility of Lucasfilm, Tom Kobayashi directs and controls the division and has overall responsibility for planning, budgeting, marketing, negotiating post-production services and directing the development and production use of new technologies. Tom joined Sprocket Systems in 1986; before that, he spent 21 years with Glen Glenn Sound in Los Angeles.

John Fry
Regional Vice President

A Memphis native, John Fry is president of Ardent Recording. Since its founding in 1966, Ardent has expanded to include three 32-track studios, film and videotape production and editing, TV commercial production, corporate and industrial TV production, record production and music production for radio and TV commercials. In addition to his SPARS duties, John has been active in the Memphis chapters of NARAS; the Memphis and Shelby County Film, Tape & Music Commission; and the State of Tennessee Film, Entertainment and Music Commission.

John Rosen
Regional Vice President

John Rosen is president of Fanta Professional Services, Nashville, which specializes in mobile recording for the entertainment industry. Since its founding in 1973, Fanta has completed more than 5,500 projects, including live albums, films, radio programs and TV shows. John has lectured at several universities and is active in the AES, NARAS, NATAS and the Nashville Entertainment Association. He is also a licensed tractor-trailer truck driver.

Howard Schwartz
Regional Vice President

After spinning discs for Armed Forces Radio during his Army service, Howard Schwartz moved to Hollywood and became a recording engineer working for Mel Blanc and writing, producing and recording comedy radio commercials. He then went to work for Heider Filmways Studio and engineered numerous platinum and gold albums. In 1975, he started Howard M. Schwartz Recording with one room, one engineer (himself) and one receptionist. Since then, one room has grown to seven.

Shirley Kaye
SPARS's executive director brings a varied background to the organization. Aside from her present position, Shirley Kaye was on the SPARS board of directors from 1982 to 1987 and was a regional vice president/treasurer. She owned and operated Coconuts Recording Co. in North Miami Beach, FL, for 10 years; before that, she was a choreographer, dancer and a dance instructor. She is currently writing a book about her experiences during a recent year-long sabbatical with her family.

David Teig
Currently the Northeast coordinator for SPARS, David Teig has been associated with the organization from the beginning. A former disc jockey and a studio veteran of more than 30 years, David attended the founders meeting and then accepted a position on the first board of directors. He remained on the board for three years, and then assumed his present position. As Northeast coordinator, David schedules monthly luncheons at New York restaurants, which allow New York-area members to meet in a social atmosphere and to observe product demonstrations from SPARS manufacturing members.

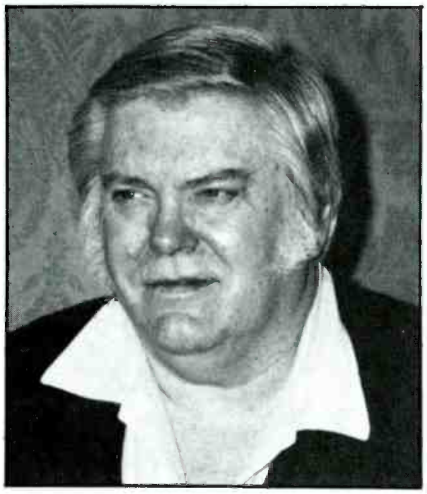
Past Presidents

Joseph D. Tarsia
1979-1980
The chairman emeritus of SPARS, Joseph Tarsia served as the association's first president. He is the founder and president of Sigma Sound Studios, with facilities in Philadelphia and New York. He entered the recording field in 1958 from an electronics background. In 1962, Joe joined the staff of Cameo Parkway Records, where he served as chief engineer for five years. In 1968, he opened Sigma

Chris Stone
1981-1982
Before starting the Record Plant in Los Angeles, Chris Stone was a national sales manager for Revlon Cosmetics and Mattel Toys. He holds a master's degree in marketing from UCLA. In his 20 years with the Record Plant, the company has built 37 rooms, both for the Record Plant and for outside facilities. Chris is also the founder of Livingstone Audio, Audio Intervisual Design and Filmsonix.

Murray R. Allen
1980-1981
The president of Universal Recording, Chicago, Murray Allen has had an extensive career as a musician and an engineer. He has performed with Stevie Wonder, Frankie Lane and the Platters; as an engineer, he has recorded Duke Ellington, Stan Kenton and Ramsey Lewis. In addition, Murray has lectured at IIT, Dartmouth and Vanderbilt, is chairman of the NARAS Education Committee and is the sound designer for the Grammy Awards broadcast.

Mack Emerman
1982-1983
A hobby of recording and collecting jazz records led Mack Emerman to establish one of the industry's premier recording complexes. After moving to Florida in 1953, he started Criteria Recording in 1956. One move and four additions later, Criteria totals more than 24,000 square feet and is celebrating its 33rd anniversary. Mack has been an active recording engineer since his hobby days, and he is still doing sessions.

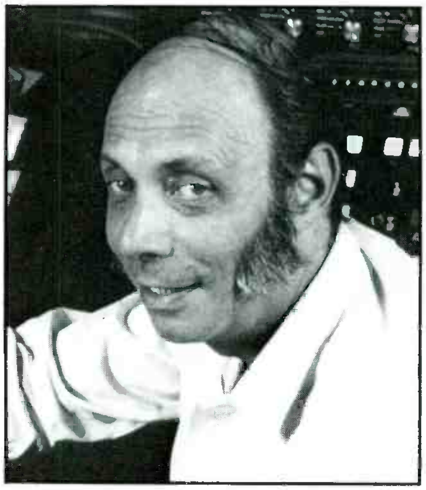


Bob Liftin
1984-1985

A SPARS founding member, Bob Liftin began his career at CBS as an audio engineer for radio soap operas and live television. In the late 1950s, he launched Regent Sound Studios in New York. As a consultant and engineer, he worked on many events and TV shows, including the 1986 Liberty Weekend, the Tony Awards, Live Aid from Philadelphia and "Saturday Night Live." He died on Jan. 8, 1988.

Jerry Barnes
1983-1984

As the the owner of Resmiranda Records in Manchester Village, VT, Jerry Barnes is carrying on the tradition of Jackie Lomax—field recording services for clients who could not afford to record in a studio or hire a remote truck. He has held a variety of executive positions for several facilities, including vice president/recording at United Recording Corp., general manager of United/Western Recording Studios, and general manager of Larrabee Recording Studios.



Len Pearlman
1985-1986

Len Pearlman is the president of Pearlman & Associates, Highland Park, IL, a management consulting company that specializes in working with companies engaged in TV production and post-production services. Before that, he spent 14 years at Editel-Chicago, forging working relationships with many advertising agencies, corporations, production and post-production firms that comprise the industry. He was directly responsible for the design and construction of the current Editel complex.

Nick Colleran
1986-1987

As president of Alpha Audio, Richmond, VA, a studio and manufacturer/distributor, Nick Colleran's background is both as a musician and a businessman. While studying to become an accountant at the University of Virginia, he also performed as a guitarist in a local band. He started Alpha Audio in 1971, which now operates as a 4-room complex. Alpha also distributes Sonex acoustic foam and manufactures the BOSS editing system.

How It All Began

By Wilber W. Caldwell

Criteria's Mack Emerman talks about how SPARS evolved from an informal dinner at the Pacific Dining Car to its formation at Jeep Harned's MCI plant.

The following interview with Mack Emerman of Criteria Studios is more than a discussion of the beginnings of SPARS. It is a story. Actually, it is three different stories.

The first is about heroes, or at least, my heroes. To those of us who began making recordings back in the Sixties, names like Mack Emerman, Joe Tarsia, Bob Liftin and Kent Duncan still bring a sigh. These men were visionaries and dreamers. Many of them still are.

The second story is about SPARS, although it is not really about the birth of SPARS or about its conception. It is about a gleam in a father's eye, a dream. It is the best kind of story.

The third story is about a time gone by. SPARS began in 1979. The recording world was very different then. It was a world of mega-studios and mega-deals. It was a world dominated by giants. Those of us not in New York or Los Angeles were operating more or less in a vacuum. The SPARS of those days was very different, too. It was an expensive and exclusive organization.

Times have changed, and SPARS has changed. The recording world of today is specialized, diverse and competitive. The SPARS of today represents a broad spectrum of recording industry interests all across the country. It is open to all for very reasonable dues. Every industry facility and service should belong to SPARS.

This interview celebrates the dreamers, the heroes and the industry of 1979, for without all of these, the SPARS of today would not exist.

***WC:** Well, Mack, I'll just set the stage, and then you take it from there. What I'm most interested in is why SPARS was started, how it came about and what your remembrances are of Jeep Harned's New River boat trip back in June 1979.*

Wilber W. Caldwell is president of Doppler Studios in Atlanta and a regional vice president of SPARS.

ME: Well, it was very obvious to me early on that there was a need for some kind of studio organization. I was especially sensitive to it, being in Miami, because then there were no recording schools, nothing much to read, nobody here to talk to.

I had started the studio as a kind of hobby, and so I got really excited when I heard about the studio organization that Phil Ramone and Mr. Levine from A&M Records were trying to put together. They had one meeting, I guess in New York. I was at the second meeting. It was in New York at A&M, and most of the people didn't show up, and it really just died.

***WC:** What year was this?*

ME: I don't know, several years before SPARS.

***WC:** Was that, to your knowledge, the first effort to put together any kind of national organization for recording studios?*

ME: Yes.

***WC:** Basically, nobody came, is that it?*

ME: Yes, it just didn't happen. But then it all sort of came together a few years later. One night during the L.A. AES, at the Pacific Dining Car on Wilshire Boulevard, Kent Duncan of Kendun Recorders hosted a bunch of people, a lot of studio owners, and people involved in the industry, and everybody was having a whopping good time.

***WC:** A moment of truth?*

ME: Well, the discussions got into, "Mr. Manufacturer or Mr. Studio Equipment Supplier, none of you are giving us anything that we want. You don't listen to us." And it got sort of heavy.

***WC:** Do you remember who was there, not only studio owners but manufacturers as well?*

ME: Jeep Harned of MCI was there. He was the only manufacturer I remember.

It was mostly studio owners from the L.A. area. But Jeep got riled up. I can't say that he got angry, but he got up and said, "Hey, wait a minute! We'll listen to you. We always listen to you! Why don't you guys come on down to Fort Lauderdale, we'll have a big confab about this thing and we'll see what it is that you really want."

So that was it. Ten years ago, Jeep invited us all down there to Fort Lauderdale, and, of course, he took the opportunity to show us all around the plant.

WC: Had he moved into Andy Granitelli's old STP building at that time?
ME: Yes, and he showed us all around. There were still race cars all over the place. He showed us all of that stuff.

WC: Who all came?
ME: Well, there was Don Frey from A&R Studios in New York. Dave Teig and Jimmy Douglas came from Atlantic Studios in New York.

WC: So this was well after Teig left Bell?
ME: Oh, yes. Dave was the chief technician at Atlantic and Jimmy Douglas was one of their hot engineers. From Criteria, there were myself and Karl Richardson and a couple of other people. From Filmways/Heider in Hollywood, there was a guy by the name of Norman Schwartz. Wally [Heider] did not come. Group IV Recording in Hollywood had Angel Balestier, and House of Music in New Jersey had Charley Conrad. Howard Schwartz of Howard Schwartz Recording in New York came, and Kent Duncan brought three people from his organization. Larabee Sound had Bob Stone, and Media Sound in New York had Fred Porter, and, of course, Chris Stone and Penn Stevens came from the Record Plant.

WC: What about Liftin? Was Liftin there?
ME: Yes, Bob Liftin from Regent Sound and Joe Tarsia from Sigma. There was an independent, a guy who was an engineer that joined in, named John Stronach.

WC: So it basically was New York and the West Coast and Miami.
ME: That was it.

WC: No hinterland at all?
ME: No, no hinterland.

WC: That was really at the height of all the Miami activity for you, wasn't it? The Bee Gees and Stills and all of that other stuff?
ME: Right. Of course, the big questions that was on everybody's mind was that the digital studio was three, five or even seven years in the future. It seemed that the equipment manufacturers were not offering anything to fill the gap in between, not with the current equipment. So with digital on the far horizon, we had a lot of questions. We talked about a "super console"—that's basically what we had in mind.

WC: A super console?
ME: Well, a console that was analog but digitally manipulated.

WC: Programmable, that sort of thing?
ME: Yes. So we had our meeting, and on that night Jeep played host to us by taking us on a boat trip, on a 98-foot boat called Pilgrim II. And everybody really got to know each other that evening. Drinks were served, and food and hors d'oeuvres. It was a very, very beautiful evening, and after the first hour everybody felt like they were old friends.

Of course, the conversations went very quickly into all of the common problems that we were suffering at the time. After a time, it became very clear how wonderful it was to be able to communicate with other people in the industry so openly.

I don't know who actually got the idea, but I think it was Kent Duncan. He said, "You guys! You know what we ought to do? We ought to have an organization. That meeting that Jeep has set up about the console...why don't we ask if he will give us the morning by ourselves?" That was Saturday morning, as I recall.

WC: So even though SPARS began as an industry conduit to manufacturers, Jeep Harned was the only manufacturer there?
ME: Yep, he was the only manufacturer there, and when we put the pressure on him, he said, "Sure, why don't you guys hole yourselves up in there and do whatever you want to do?"

WC: So it was clear to him that there was a chance for something larger than just feedback for him personally, but for the industry as a whole.
ME: Exactly. Before the whole thing was over, we had a board of directors. Joe Tarsia, of course, was the first chairman, and we had...

WC: The name?
ME: Yes, we gave it the name of the Society of Professional Audio Recording Studios. It was a closed meeting...by ourselves.

WC: How long did that first meeting last?
ME: I would guess about three or four hours.

WC: You guys were a lot more efficient back in those days, weren't you? You started the whole thing in three or four hours. Hell, it takes us five hours to decide what kind of sandwiches we're going to have.
ME: That's for sure.

WC: Who was on that first board?
ME: As I said, Joe Tarsia became the chairman of the board, and the regional members of the board were Bob Liftin of New York, Chris Stone of Los Angeles, myself and Glen Snoddy from Nashville. The acting treasurer was Dave Teig and the acting secretary was Kent Duncan.

WC: Glen Snoddy from Woodland Sound? He wasn't there, wasn't he? Did you guys elect him in absentia?
ME: Yes, because we felt very strongly that we needed a representative from the Nashville area.

WC: So what then? There was paperwork to be done...and scheduling a meeting?
ME: Right. You know, Joe Tarsia really took the ball and ran with it. Nobody really realizes how much he did that first year and how bold he was.

That was just a real fun time. Such a great feeling was created among all of us, we just left there with the warmest feelings for each other.

WC: That is one thing that I have found in being a member of SPARS. To a certain extent it's like therapy. You go there, and suddenly you realize that you are not out there in the world by yourself, and that there are a whole bunch of people who have identical problems.
ME: That is so true. If only the other people in this world would understand what we all get out of it.

Past, Present and Future

The past and present executive directors of SPARS talk about what the organization has been and where it is headed.

No organization exists in a vacuum. After all, organizations are made of people, and the same events that affect people also affect organizations.

In its 10 years, SPARS has reacted and adapted, along with everyone else, to the changes that buffeted the industry. What started as an exclusive organization has evolved into a broad-based one, open to almost anyone in the industry. It has gone back to its roots, as when it re-established the manufacturer interfaces, and is expanding its horizons with its intention to stage more regional events.

To gauge where the organization has been and where it is going, *RE/P* talked with the immediate past executive director, Gary Helmers, and the current executive director, Shirley Kaye.

Gary Helmers: A need for business information

As executive director, Gary Helmers helped SPARS provide business information to studio owners on such issues as taxes, insurance and financing. More than anything, he says, technology forced owners to run the facilities more efficiently. After leaving SPARS, Helmers became the publishing editor of *Music Technology, Home & Studio Recording* and *Rhythm* magazines. He now teaches and writes from his home in Garden Grove, CA.

***RE/P:** When you were executive director, it seemed that there were a great deal of changes in the industry and in the organization. Is that a fair statement?*

GH: Definitely. There were changes in technology, changes in how people were doing business, in diversification and specialization, in education—how people were trained in the industry.

***RE/P:** Was the biggest change in the technology?*

GH: Actually, the biggest was how the technology changed the way people needed to do business. People needed to be more aware of basic business principles. It seemed that in the 1970s, a lot of people were very successful with recording studios without being too concerned about business issues like insurance, taxes, financing, marketing, sales and employee relations. In the 1980s, all of those things became more important.

Perhaps it was just that the whole generation of businesses that had been spawned during the 1970s was now growing up and entering new stages, so they needed to place more concern on managing their businesses.

The need for SPARS arose because the recording studio industry was a cottage industry—a group of highly independent, highly individual owner/operators. The need for SPARS and the difficulty in recruiting both came from that same fact.

***RE/P:** You were trying to get people who didn't join organizations to join an organization.*

GH: Exactly. And if there's been any change in the industry in the last five or six years, it's become more of a cottage industry. The high technology at lower prices has created more studios, smaller studios, and more independent and more individual owner/operators. So the need for the organization is greater than it ever has been.

***RE/P:** The Mothership Scenario, which you outlined in the earliest SPARS columns, ties right into this.*

GH: Although when we developed the Mothership Scenario, I don't think that we foresaw the great amount of technological power that would be available at a low cost in a small package. I think that the development of a mothership—a large studio complex where people would bring

their product in to work on—has not come to fruition as much as I expected. People have the power to do a very great deal of what they need to do in their bedroom or garage.

***RE/P:** What other issues did you deal with during your tenure?*

GH: I think the most critical was establishing the business conferences, to provide up-to-date information. The second most important thing was the interface with state and federal agencies to clarify issues and to effect change, specifically concerning sales taxes, labor laws, employee relations, that kind of thing.

That covers the communication part of SPARS. The innovation part of SPARS is a little harder to see. But a very important function is members regularly meeting with manufacturers to assist them in creating and marketing products to the industry. It's a small industry, and it can't afford the manufacturers making too many false steps that raise the price of equipment across the board.

During my tenure, we re-established those manufacturer interfaces, and I think that's a very important part of what SPARS is about.

Last, but not least, is education. I think the most enjoyable part of my time was working on developing the SPARS National Studio Exam. It was a very lengthy procedure, but it was very satisfying and revealing to canvass the country, finding out what people thought were important pieces of knowledge for the future employees to have. I think that it's still SPARS's most important mission, to work with educational institutions to make sure that the future employees truly have the skills and the knowledge that they need.

> ***"If somebody asked me how to find out about how to run a studio, I can't think of another place I would send them."***

***RE/P:** Having the test independently developed would seem to give it more credibility.*

GH: We were very intent on making sure that the development of the test was done right so there wouldn't be any quibbling with the outcome. It resulted in something that was truly useful to studios.

***RE/P:** What's facing the industry now?*

GH: Looking toward the future, I'm reminded of an article in *Inc.* magazine written by Tom Peters, the guy who wrote "In Search of Excellence." He talks about how he has now changed his logo to a gazelle, that we are now in the "age of the gazelle." The gazelle is an animal that can change direction and speed very, very quickly. Now more than ever, studios need to become gazelles.

Things are changing so fast, that someone who is in the industry has to change quickly, to fill needs.

***RE/P:** Is the best way to keep ahead through SPARS?*

GH: It's not only the best way, but it's possibly the only way. If somebody called me up and asked me how to find out about how to run a studio or build a business, I can't think of another place I would send them.

Shirley Kaye: A unified voice for the industry

In her two years as executive director, Shirley Kaye has consolidated the gains made under Gary Helmers and is planning an aggressive program for future growth. In the future, she says, SPARS will expand its role as an information clearinghouse, initiate more regional events and expand membership.

***RE/P:** What are going to be the important issues facing SPARS in the next few years?*

SK: I think the bonding of the industry, with a single voice, is important. People need the support of one another, for growth, education and development. If you work through an organization, you're richer because of the interdisciplinary experience.

I think the studio exam will become even more important. We'd like it to become a standard through use, but we are not permitted to do so. We want quality people working in diversified facilities. Our task is to encourage educators to work with us to make them more familiar with what is needed in the real world.

Membership expansion is the keynote to growth, and significant input requires numbers.

***RE/P:** Do you have a target number that you would like to reach?*

SK: No. The potential is there. We have a mailing list of more than 5,000 studios. The problems are common, even if people think they're mutually exclusive. If you have 10 small-studio owners who get together, they can talk, utilize our expertise, solve some of their problems and understand others in a multidimensional way.

***RE/P:** Do you sense that people in the industry are more willing to join an organization?*

SK: In the past, some potential members were reluctant, because they felt that SPARS was an organization that catered only to the industry leaders. That's no longer true. We really want to be of service to everyone in the industry, and our board of directors reflects this viewpoint.

As a member of SPARS, you can call other members. They will interact with each other and they'll discuss things that you wouldn't openly discuss in public. It's the people-touching-people thing that works.

Recently, a SPARS member who needed someone to testify at a hearing with the IRS called me and said, "I don't know what to do." I referred him to a SPARS member who has experience with the IRS. He's been through a comparable tax situation. Our member actually went to the hearing and testified.

***RE/P:** What plans or projects do you have in the works?*

SK: Currently, we are working with educational institutions to help develop curricula, and coordinating our internship program with the schools. We also want to focus on business practices. This is terribly important. A committee has been formed on Recommended Practices and Procedures, which will develop new ideas and formats. They have a broad mandate to cover anything related to the industry.

SPARS is going to push regional participation, so people don't have to go so far. We will come to prospective members, as it is more efficient for them and we can appreciate local problems more rapidly. We will dedicate ourselves to going where the action is. We're going to be in Nashville in May, Minneapolis in August, Chicago in the fall. We were in Austin in February.

***RE/P:** One of obvious consequences of technology is that a viable, first-class studio can be built almost anywhere.*

SK: Absolutely true. The industry is going to be altered. It's going to be changed by the technology that is being developed. Ten years from now, everything will be different.

For example, not long ago, we thought overnight mail was wonderful. Now a fax machine sends documents in seconds, and some of us think it could be faster. What will be next?

Industry adapts to change. Years ago, I remember someone saying, "The 78 record will never go out." How wrong they were. Now we're into compact discs. What will come after compact discs? The industry will reorient itself to whatever the new technology is, and in fact serves as a stimulus for the development of additional innovative technical design.

***RE/P:** Given this climate, what's the best thing that SPARS can offer?*

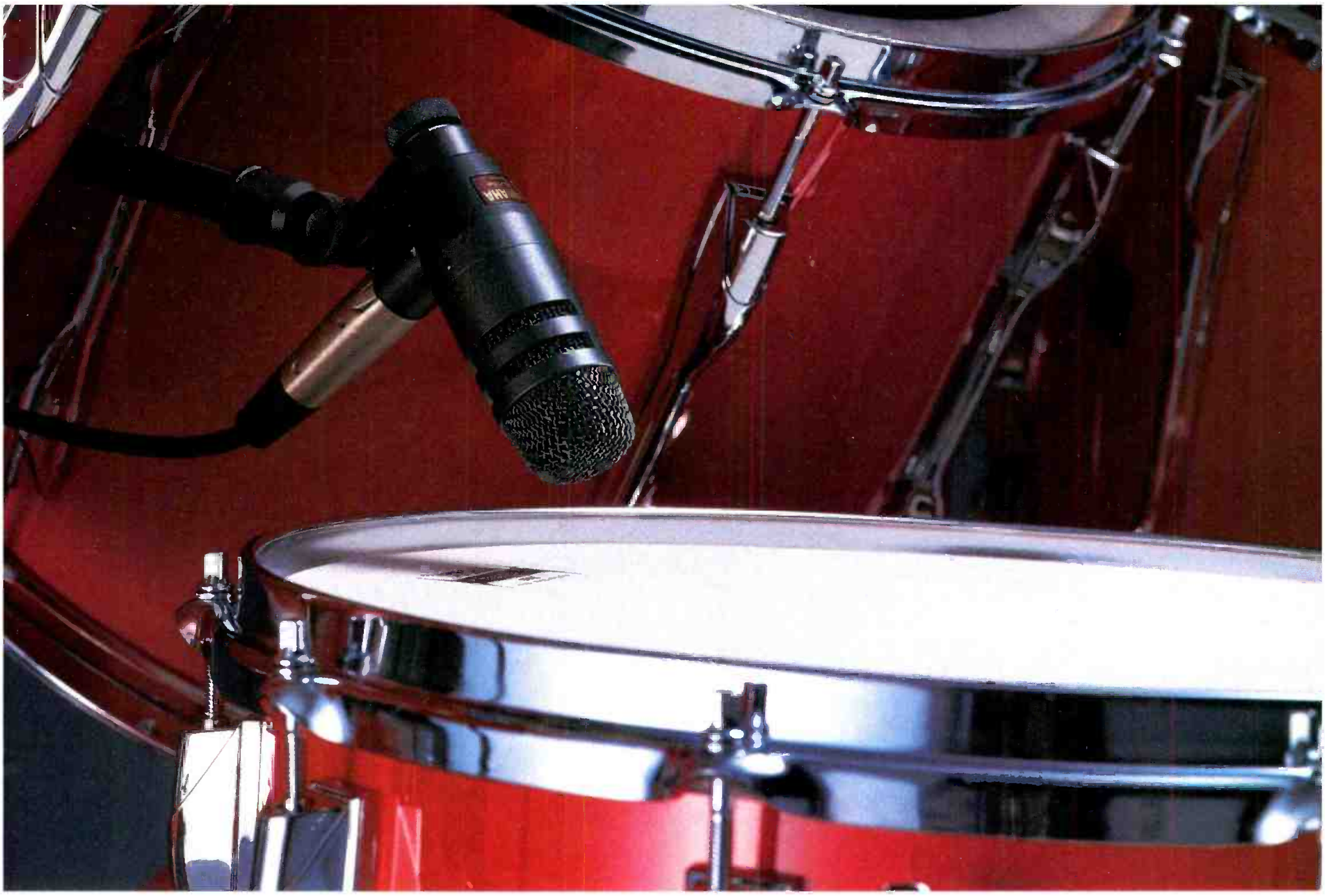
SK: SPARS will keep abreast of the latest technology for its members, who don't necessarily have the time to keep up. Therefore, they can come to us for assistance. That's part of what we're about. But it's not a 1-way street. Technology is great, but it's people who make it work. Members have to participate by making their wants and needs known, which we convey to the advisory members, the equipment manufacturers. Together, we can solve problems, from the simplest to the most complex.

For example, SPARS's computer hard disk drive went bad. The computer repairman, from a large company, came in to service the machine. After evaluating the problem, he had no choice but to install a new disk drive. With the equipment he had available, he couldn't recover any of the information on the hard disk or from the backup disks.

We really needed that material. After speaking with three SPARS board members, each said, "Send me the old disk drive. I think I may be able to recover some data. I have special equipment available."

The repairman was skeptical because this type of equipment can be costly. I said, "Wait and see." SPARS members have technology and knowledge extraordinaire. They are creative and they enjoy the challenge of figuring out answers to problems. And they did. We are now in the process of entering the data recovered from the old disk drive. That's the spirit that keeps SPARS going. People helping people with the use of technology, and SPARS advisory members providing the technology to make the helping possible.

RE/P



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Circle (41) on Rapid Facts Card

Using Power Amplifiers in the Real World

By John Monforte

Power amplifiers are too often considered accessories in audio systems when, in fact, their operation can have unanticipated sonic implications.

In the real world, power amplifiers do not always live up to their specification sheets. After being installed in sound reinforcement systems or studios, even high-quality power amplification units, with honest specs, can fall short of the mark. There are many reasons.

Making an amplifier with high power output is a relatively simple affair. The basic circuit topology is similar to that used in operational amplifiers, although because of the higher voltages and currents involved, the components have to be more durable.

Figure 1 shows an ideal amplifier as an op-amp. It has an infinite input impedance, so it takes no current from its source. The output is a voltage source with a 0Ω output impedance. That means the amp delivers an exact amount of voltage to its load, regardless of the current involved. The amount of voltage required is related to the amplifier's gain, which is a ratio of output voltage to input voltage.

Power source

The power supply needs to be strong because it is the sole source of the energy delivered to the load. Generally, it isn't regulated because high-power regulators are almost as expensive and complicated as the amplifiers themselves. Therefore, it is wise to provide a circuit capable of delivering high currents because good, clean ac power is essential for optimum performance.

Wire Gauge	Wire Diameter (inches)	Incremental Resistance (Ohms per foot at 25°C)
20	0.0320	0.0103
18	0.0403	0.0066
16	0.0508	0.0041
14	0.0641	0.0026
12	0.0808	0.0016
10	0.1019	0.0013

***Table 1.** Wire gauge vs. resistance.*

Solid power connections are a must. Use a 10A circuit for every three good-sized amplifiers (5A, if you're working with 220V). Most extension cords use no larger than 16-gauge wire. But if the power runs are long, do not hesitate to step up to 12-gauge wire. Most people don't think twice of doing this for the loudspeaker runs, but they often disregard the fact that the real source of current is actually the utility company feed. (Admittedly, the amplifier's filter capacitors will help average the instantaneous current surges normally found in music programs. This greatly reduces demands on the supply voltage during music peaks and only slightly increases them on average. Even so, all wires will drop some voltage when the currents get high.)

Table 1 shows the resistance of several standard wire gauges. To find the total resistance in the installation, multiply the run length by the incremental resistance and then multiply again by two (to account for each conductor in the circuit). When no current is drawn, there is no voltage drop in the wiring. When the amp is attempting to put out full power, the loss in voltage will reduce the amp's output potential. Also, if another amp is sharing the feed and working with the same program signal, the drop will be doubled.

Most amplifiers are designed to coast along at low current consumption when they are not delivering loud musical passages to the loudspeakers. The dynamics of music being what they are, the peak demands are infrequent. Most of the time the amplifier is delivering only a few percent of what the peaks will require.

When banks of amplifiers are carrying the same signal, very large currents may

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

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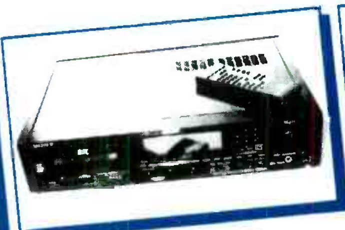

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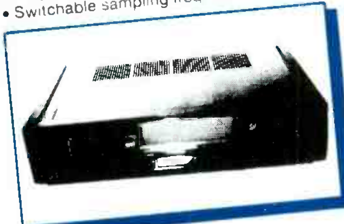

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be drawn instantaneously. When a system is properly fused for these surge current levels, you may find the protection inadequate if one of the amplifiers fails at a time when the musical demand is light.

This is because the effect of one amplifier (out of a multi-amp system) blowing up is similar to the current surge the fused system must handle when delivering a full-power peak signal. The overall current in the amp rack may still be reasonable while the distressed amplifier is outdoing the smoke machine. Each amplifier must be individually protected from current excesses—another compelling reason for giving each amplifier a separate feed.

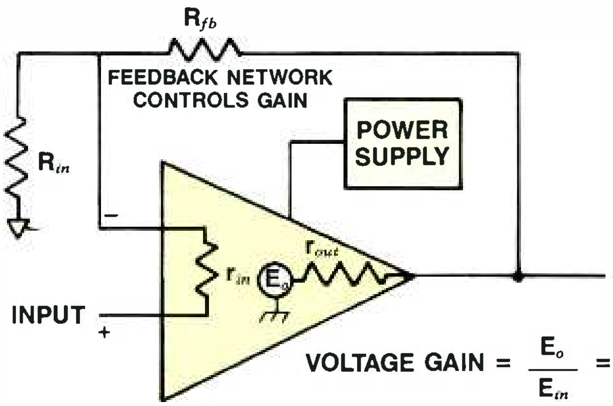

Figure 1. *Power amp as operational amp.*

Input considerations

As mentioned earlier, the ideal amplifier has an infinite input impedance. This allows for, among other things: the paralleling of many amps from a single source; a reduction of radiated magnetic fields from input lines; and relative freedom from signal loss due to input cable resistance. It is not at all hard to come up with a circuit that can approach this ideal. In fact, most amplifiers have a parallel resistor on their input to lower the impedance. This prevents the amp from buzzing or picking up RF when the input is disconnected. It also reduces the "thumping" that occurs when the input is either plugged in or unplugged.

There is no reason for hums and buzzes to be anywhere in a studio, no matter how low-level they appear (with a possible exception of what is being picked up by an analog tape head). The rules of good grounding, when rigidly applied, should prevent the propagation of all hums and buzzes.

To benefit from the use of balanced lines, balanced inputs are needed. Even when there aren't any long input runs, this

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今や私には空気のような存在である。——時には、音の発想の手足となりあるときは、複雑な要求にも応えてくれ、いや、もしかしたら求めた以上のものを与えてくれるかもしれない可能性を秘めている。今や私の仕事空間には必要不可欠なコンソールである。

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Ohm's Law Revisited

Ohm's law is an elegantly simple relationship that relates voltage, current and resistance to each other.* Simply stated, a voltage across a device is proportional to the current through it and its resistance. (See Figure A.)

We can express this mathematically with the equation: $E = I \times R$, where E is voltage in volts, I is current in amps and R is resistance in ohms.

If we know any two of them we can solve for the third:

$$I = E \div R \quad \text{and} \quad R = E \div I$$

This shows us a few things. First, if there is no current in a device (our loudspeaker wire for example) there will be no voltage dropped (lost) across it no matter what the wire resistance may be. Loudspeakers, however, can have a very low resistance (around 8Ω in the audio range) and, therefore, will draw a great deal of current for any given voltage. This is the same current that flows in the wire, so with the loudspeaker load, the wire will drop the voltage level. If a speaker draws 5A, a 1Ω wire will lose 5V of signal. The amount of voltage lost can be calculated from the voltage divider relationship, which is the basis for attenuator design in general. The voltage divider relationship can be described as follows: The ratio of output and input voltage is proportional to the ratio of the resistances. (See Figure B.)

$$\frac{E_{out}}{E_{in}} = \frac{R_l}{R^l + R^s}$$

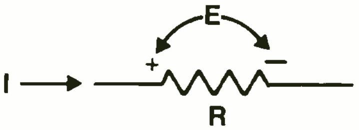

Figure A. A resistor and its electrical parameters.

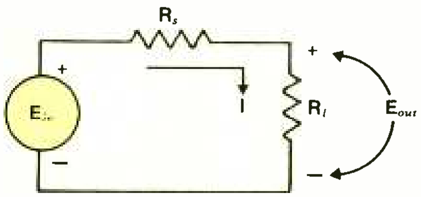

Figure B. A voltage divider circuit.

which can also be written as:

$$E_{out} = \frac{R_l \times E_{in}}{R_l + R_s}$$

where R_l is the loudspeaker (or any other) load and R_s is the source wiring resistance.

For example, Ohm's law says a 10A load on a 110V circuit represents a resistance of 11Ω. Let's say 1Ω of wiring connects the load, with the load resistance effectively being the other 10Ω. The wire will drop the voltage (attenuate) by a factor of 0.91, resulting in a load (output) voltage of 100V. The voltage hasn't disappeared, it is the voltage across our resistive wire. A check of Ohm's law across the wire shows that 10V is indeed the drop of 10A in a 1Ω wire. The wire is giving off heat and generating a magnetic field with this power.

Impedance is the term used to describe a resistance that varies with frequency or, conversely, resistance is the impedance a device has at dc. There also happens to be a phase relationship involved with impedance, but for the purposes of calculating voltage drops and power, it doesn't need to be considered.

We can use the divider relationship to calculate an input attenuator for an amplifier. Let's say that an amp is specified to deliver full power with 1V on the input, and the console driving it does not clip until 15V. If the amp has a rated input impedance of 10kΩ we can solve for the source resistance by rearranging our divider equation to solve for the source resistance:

$$R_s = \frac{R_l \times E_{in}}{E_{out}} - R_l$$

Solving this, we get a resistance of 140kΩ, which can be wired in series with the input connector of the amp.

* The simplicity was not expected. As Georg S. Ohm was preparing to publish his studies on voltage and current relationships, his equation contained exponential terms and temperature coefficients. He discovered at the last moment that those terms were an artifact of the chemical batteries he was using in his experiments. Electronics would have been a lot more complicated if the extra terms were really needed.

can still help reduce hum.

Usually, consumer power amps have their signal ground potential connected internally to the chassis, which is sensible in a home system but usually violates the grounding scheme in a large facility. Most professional units allow this connection as an option on a rear panel terminal strip, but many studios use consumer amps or "professionalized" versions of consumer designs with grounding schemes that aren't switchable.

A related problem is hiss. Quite a few amplifiers are capable of delivering full rated output with only one or two volts input. This represents a signal level of around 0dBV. This arrangement offers about 10dB of headroom in a home system that operates at around −10dBV, but professional systems have a +4dBV operating level. By the time the signal reaches operating level in the professional mixer, the consumer amp begins clipping.

For sane volume levels in the control room, the console's monitor section is often being operated at a very low level. Consequently, the signal picks up hiss in the monitor module. The solution is to use an attenuator on the amp's input, but many studios leave their amplifier level controls full up. (See sidebar on attenuator design.)

The same considerations apply to live sound. In fact, there is even less justification for hiss because there is no recording device in the chain. Still, I've seen boards in operation with output signals that are barely moving the meters. Sound engineers often blame it on the mixer. They are half-correct. The noise does indeed come from there, but mixer design noise specs are limited by the laws of physics. Correct gain structure dictates that the amps should be adjusted so that the volume in the room is adequate when the signals are reading 0VU.

As mentioned earlier, it is relatively easy to design for high power amplification. Relatively, because there is the real challenge of delivering the power safely. An amplifier is often required to deliver a lot of power over a short period of time, and a good design will not thrash around several joules (a unit of energy, 1 joule = 10,000,000 ergs.) irresponsibly. The fate of the amplifier and a system's loudspeakers hang in the balance.

The simplest and most effective means of protection are fuses. Circuit breakers may not be fast enough. One can be used to protect the power supply, a couple more for the output devices and one each for the outputs to protect the loudspeakers. The loudspeaker fuses are best located at the amplifier end of the line to protect the amp from a short in the line, as well.

But amps used for live sound cannot depend on fuses. An overload at the amplifier's input, lasting only an instant, may result in a blown fuse. This would result in an unacceptable signal loss. A circuit is needed that senses fault conditions and restores operation as soon as it is safe to do so, which is not easy to design. It is very difficult for a circuit to know just when to protect. Unplugging a mic with the fader up is like a cleanly delivered timbal hit, for example.

Two amplifiers of similar specifications perform very differently if one has an overzealous protection circuit. Also, because these circuits are in the signal path, they may add sonic artifacts even when not activated. When triggering, they are often obvious. I believe these circuits account for more of an amplifier's sonic quality than is commonly suspected, with audibility varying from make to make.

Test conditions used for amplifier specification tend not to reveal details of the effects of protection circuits, because under test conditions it is unlikely that they will be triggered into activity. I recall an incident when evaluating a new control room monitor that sounded unreasonably distorted. I tried another amp of similar specification (identical power) and heard a marked improvement. The loudspeakers, as it turned out, were only 2Ω. Even though both amps were rated for 2Ω operation, the first amp showed visible distortion of the music waveform when viewed on an oscilloscope.

Generally, the protection circuit is there for the amplifier's benefit. Some speakers may not be ready to accept what the amp can deliver. An amp with comprehensive output current limiting may still require an output fuse. Loudspeaker protection circuits that would reduce power consumption when signals threaten the drivers, and restore full power the moment the threat has passed, could be very useful. Unfortunately, few speaker systems employ this kind of protection circuit.

Thermal considerations

Bipolar transistors used in amplifiers have an interesting property known as a negative temperature coefficient—meaning that as a device gets warm, its resistance drops, allowing it to draw more current. This, in turn, tends to heat it even more, causing a condition called thermal runaway. The transistors will heat themselves to oblivion.

Although an amplifier may be thermally stable on the test bench, external heat sources, such as other amplifiers or sunlight, may entice the transistors into thermal runaway. To combat this, a resistor can be put in series with a transistor's output. Resistors have a positive temperature coefficient and will keep the amplifier thermally stable if they are large, compared to the transistor's effective resistance. Unfortunately, this extra resistance adds to an amplifier's output impedance, making it less ideal as a voltage source.

Field effect transistors (FETs) have a positive temperature coefficient—hot devices use less current. These amps may be free of thermal runaway, but no amplifying devices, even vacuum tubes, thrive on excessive temperatures.

Even though most amplifiers have thermal circuit breakers that prevent the amp from self-immolation, even a momentary loss of signal can be disastrous in a professional situation. Proper ventilation is of paramount importance. Just because the amp is equipped with an internal fan, you should not assume it will be adequate when ganging several amps in a closed space. A rack should have its own ventilation. When forced-air cooling is used, it should be designed to work in concert with natural convection. Trying to force cool air into the top of a rack may even be counterproductive, as it prevents hot air from escaping.

Several years ago I worked in an electronics repair shop where I saw several "broken" amplifiers that performed flawlessly on the test bench. Some customers complained that the amp sounded fine right up until the point it failed. Others said their amps quit intermittently. These are often symptoms of thermal protection. Few amps have indicator lamps to tell the user that an overheating condition exists, although adding one would do little to increase cost.

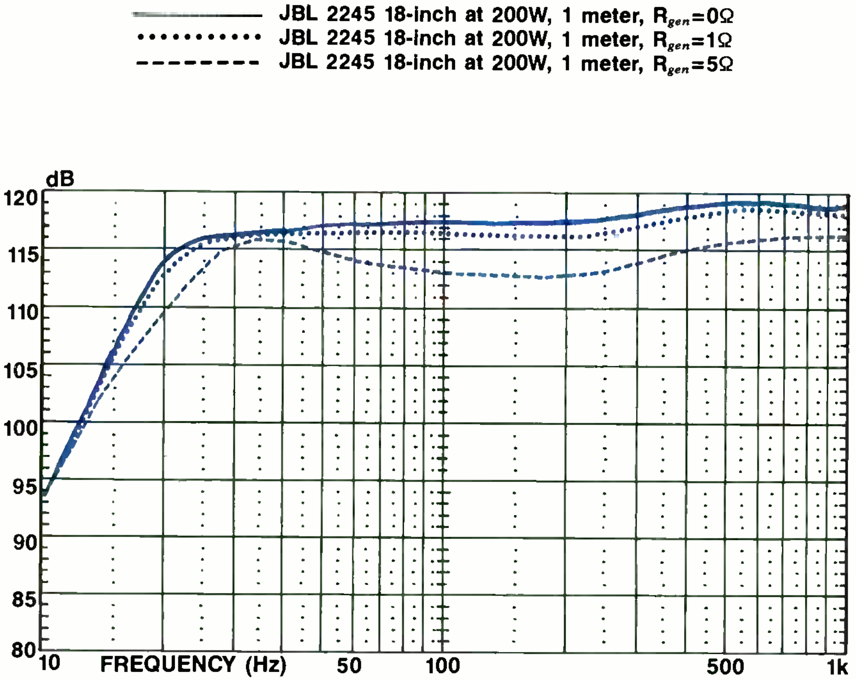

Figure 2. Acoustic on axis SPL.

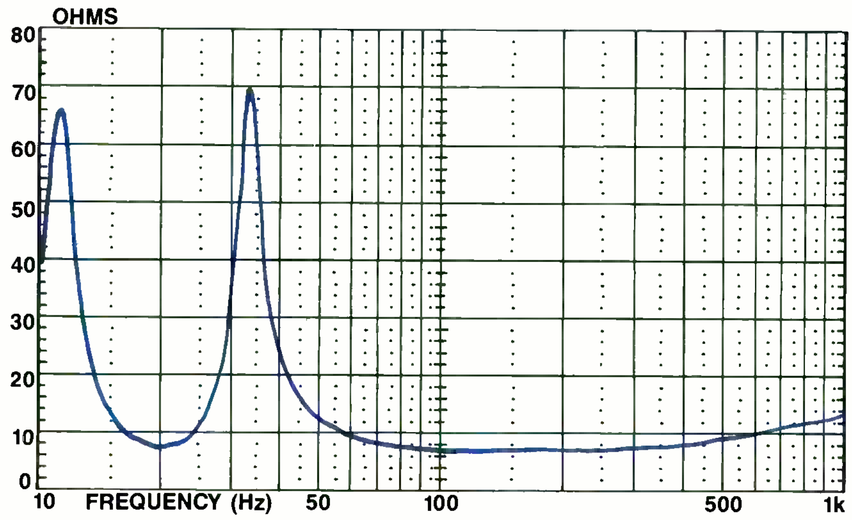

Figure 3. Speaker impedance.

Damping factor

A poorly understood element in amplifier interfacing is the damping factor. Often given in amplifier specifications, damping factor has little to do with how the amplifier will perform in real-world situations.

In general, damping factor is the ability of an amplifier to control the position of the loudspeaker driver. If low, the loudspeaker will tend to move as a result of mechanical factors more than as a response to electrical commands from the amplifier. Because of inertia, the cone's mass may continue to move despite a signal to stop. Inertia works both ways—when stopped, the cone may be slow to accelerate. Also, unless the damping factor is high, the cone can overshoot, causing distortion. The need for good damping is most important near the loudspeaker's resonant frequency. Usually the woofer and cabinet combination form a strong resonance near the lowest frequency of usable response. Poor damping can result in a loss of "tightness" in the bass.

The damping factor is the ratio of the output load impedance to the source impedance. When damping factor is measured for specification, it is figured as 8Ω (the nominal load impedance), divided by the amplifier's output impedance. For example:

damping factor	output impedance (ohms)
10	0.8
100	0.08
1,000	0.008
10,000	0.0008

The real world tends to differ from these specifications in many respects. Real load and source impedances vary with frequency. The load may not be 8Ω at all. In a 4Ω situation, damping is halved. More important, the wiring to the load is not included as a portion of the source impedance. The wire impedance may be significantly greater than the output impedance, to the point that the installation's damping factor can neglect the effects of the amplifier altogether.

Resistance attenuation

When the output impedance of an amplifier is non-zero, either through internal circuitry or though the connecting wiring, some of the output voltage will be lost. When the load impedance is only a few ohms, as in a loudspeaker, it doesn't take much resistance to start losing power.

The source and load impedances act as a voltage divider. Whenever the two impedances are equal, the power is cut in half. A 100-foot run of 16-gauge speaker cable can waste 10% of the amplifier's power—double that figure at 4Ω. This is one of the primary reasons for using very heavy-gauge speaker wire and installing the amplifiers as close to the loudspeakers as possible.

Adding sense wires is another way to minimize problems from wire losses. Sense wires are a staple item in regulated high-power supplies. (A regulated power supply is essentially a power amp for dc.) Two small-gauge wires are run from the load terminals back to the input of the amplifier. Since the input impedance is high, almost no voltage is dropped.

These wires present the feedback circuits with a signal that represents what is occurring at the load itself and not just what is feeding the lines. Effectively, the output cabling has been included in the feedback loop and, therefore, benefits from the distortion reducing advantages of negative feedback. I know of no commercial amplifiers that use this method, but it would not be difficult to implement. The modification to an amplifier circuit would be minimal, and the cost would be far less than attaching what amounts to welding cable to the outputs.

Loudspeaker behavior

Many people do not realize it, but the finite output impedance from the amplifier and cabling also affect the frequency response of a loudspeaker. Figure 2 shows the frequency curves for a woofer cabinet with a single driver and no crossover, driven by different source impedances.

There is some overall attenuation, but virtually no loss at the low-frequency end of the curve. This means the attenuated versions appear underdamped and heavy on the bass.

The reason for this is best described by the impedance curve of the woofer and cabinet shown in Figure 3. The system has an effective resistance that varies with frequency. At the point of resonance, the impedance rises dramatically to many tens of ohms. At these frequencies, the contributions of the small source resistances are negligible. But in most of the upper ranges, the impedance is only a few ohms. Here, a small resistance from an amplifier will contribute to form a power drop at these points. More-involved loudspeaker systems have several drivers and many crossover components that add even more complexities to the resulting load-impedance curve. Depending on exactly how the speaker system is designed, it may be very sensitive to amplifiers and wiring.

Most specification sheets tell only how an amplifier will respond in an ideal environment. General parameters—such as maximum power output, bass cleanliness, noise levels and perceived frequency response—are ultimately controlled by installation conditions. Application of a few important techniques can noticeably improve the sound with little added cost and no elaborate test gear required. RE/P

Acoustic Measurement Techniques

By Adrian Weidmann

The sound of a control room is dependent upon the monitors' interaction with all aspects of the room.

The modern recording studio has become a sophisticated array of high-tech recording and processing equipment, professional recording engineers, demanding clients and musicians—all functioning in a sensitive acoustic environment. Along with this level of professional equipment and technology comes an equally high level of competition and expense. A "good-sounding" control room and studio can determine where an influential client works. To remain competitive, the acoustics that define a good-sounding room should be evaluated and maintained.

The control room monitors are often the focus of what constitutes a good-sounding room. This, however, is an oversimplification. The control room is, in fact, a complex fusion of subjective and objective acoustic design factors. To begin to understand the degree of this complexity, envision a control room that is entirely filled with an excited, undulating mass of your favorite flavor Jell-O! In other words, room acoustics cannot be fully understood because of the complexity of sound-wave motion taking place 3-dimensionally, over time.

The sound of a control room is dependent upon the monitors' interaction with all aspects of the room. Since physical changes, such as the addition or removal of equipment, are quite common, it follows that the originally designed acoustic "fingerprint" of the room also changes.

By measuring loudspeaker balance, frequency response, reverberation time, background noise, transmission loss, studio isolation and spatial uniformity of the sound field in the room can be documented and maintained over time.

Adrian Weidmann is international manager of Brüel & Kjær's pro audio group. All art courtesy of Brüel & Kjær.

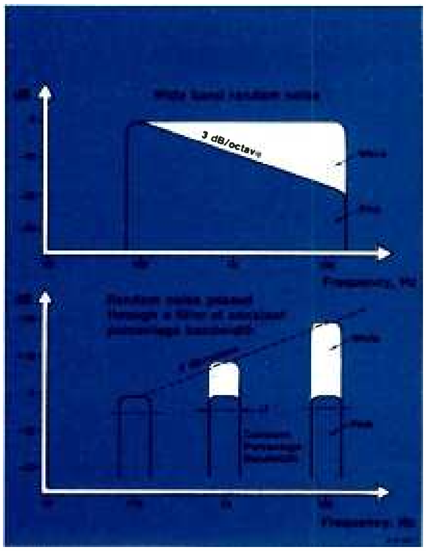

Figure 1. *The difference between white and pink noise is graphically shown on a logarithmic scale. The lower graph shows the result of using filtering with either white or pink noise.*

Background noise

All equipment—whether tape machine, computer, typewriter, copier or air handling system—generates noise. Together they combine to form the overall ambient noise level of the studio complex. High background noise levels cannot be tolerated and the ambient noise level should be kept below 25dBA to 40dBA to comply with typical, professional recording studio design criteria. Any additional equipment, or expansion of the complex, can affect background noise. Because of daily exposure and acclimation, small incremental changes in the ambient noise level often go unnoticed. By tracking the changes in ambient and background noise, it is possible to isolate problematic noise generation. To measure the ambient levels accurately, an integrating sound level meter should be used. In this way, a time-average of the background noise is obtained.

Spatial uniformity

A good-sounding control room produces an even distribution of sound from the monitors throughout the room. To determine the room's sound distribution characteristics, begin by emitting either white or pink noise through the monitors.

White noise is a broadband random signal containing all the frequencies of the spectrum with a random amplitude distribution and constant signal level for every frequency over the entire spectrum. Pink noise is similar, but decreases by 3dB for each doubling of frequency. When pink noise is used as the excitation signal and octave or ⅓-octave filters are used, the measured SPL in each frequency band will be approximately the same. (See Figure 1.)

With the monitors emitting white or pink noise, the sound pressure level at various positions in the control room can be measured. In this way, the distribution of the sound field can be mapped. Distribution for different frequency bands can be obtained if octave or ⅓-octave filters are used with the sound level meter. Due to the nature of sound propagation,

the higher frequencies are more directional and are more likely to be unevenly distributed throughout the room.

Reverberation time

In the control room, if the monitors are switched off, the sound does not immediately stop but, instead, decays over time. The amount of time it takes for the sound pressure level in a room to decay by 60dB is the room's reverberation time (RT_{60}). In a control room, too long a reverberation time produces high background noise levels, while making speech less intelligible and music more cacophonous. Conversely, a short reverberation time deadens background noise but muffles speech and makes music sound thin and staccato.

The two main parameters that affect the reverberation time of a room are the volume and the varied amounts of sound-absorbing material used. Since the volume of a control room rarely changes, one can concentrate on the effects of sound-absorbing material. Simply stated, the more sound-absorbent material in the room, the shorter the reverberation time. In general, all materials absorb sound to some degree. Typically, the harder and

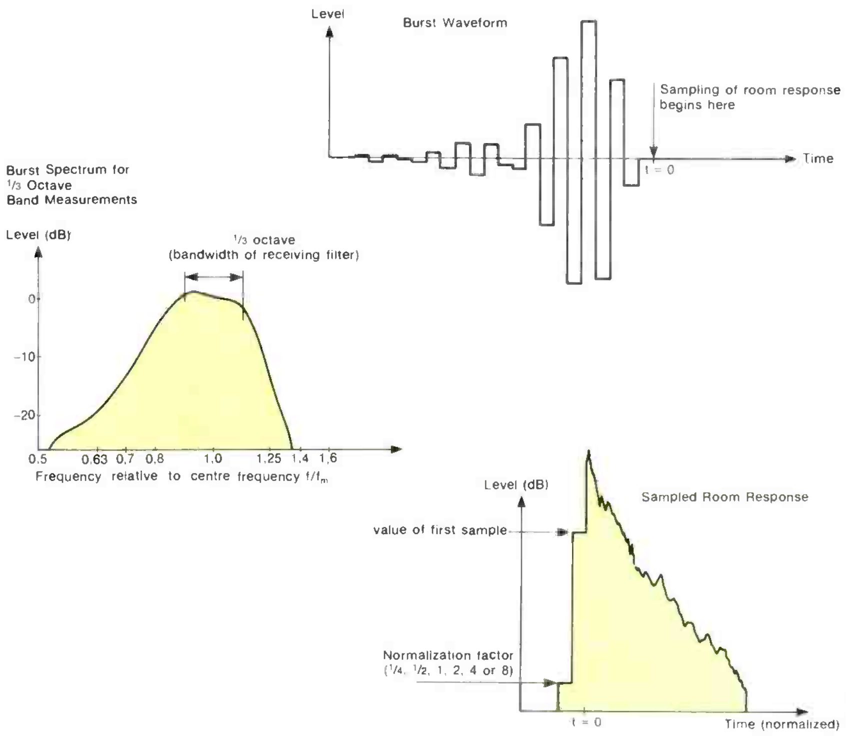

Figure 2. *A burst waveform and its spectrum for ⅓-octave measurements and a sample impulse response recorded on a level recorder.*

smoother a surface is, the less effectively it absorbs sound.

The absorption properties of different materials vary widely with the frequency of the incident sound. As such, the reverberation time of a room is frequency-dependent. Because lower frequencies are less effectively absorbed than higher frequencies, generally the lower the frequency, the longer the reverberation time.

To measure the reverberation time, one needs a sound source to generate sound within the room and a sound level meter to monitor the decay of the sound pressure level after the sound source is shut off. The sound source in a control room is, of course, the monitors; however, the excitation signal should be well-defined. Again, white or pink noise is typically used, and the SPL should be high enough to produce a substantial decay. The test noise can be transmitted as a steady sound that is abruptly shut off or as a series of bursts. (See Figure 2.)

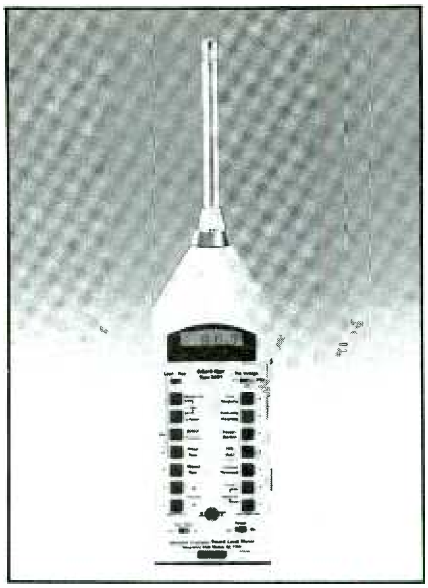

Photo 1. The Brüel & Kjær Type 2231 modular precision sound level meter.

A sound level meter is used to measure the decaying SPL of the sound field in the room. To measure the reverberation time at different frequencies, the sound level meter can be augmented by octave or ⅓-octave filters. A level recorder can be used to graph the sound level in the room as a function of time. From this graph, you can calculate the length of time it would take the sound field to decrease by 60dB. This corresponds to a decrease in sound *pressure* by a factor of 1,000; in sound *power* by a factor of 1 million.

When tone bursts are used as the excitation signal, the graphic results represent the impulse response of the room. Although the reverberation time cannot be directly obtained from this result, it can be derived using a series of calculations. An advantage of the burst (or Schroeder) method is that accurate and repeatable results are obtained faster than with the steady noise/cut-off method, and with an improved signal-to-noise ratio.

Isolation

Not only can the noise transmitted from adjoining rooms or hallways be distracting, but, more importantly, it can affect the quality and sound of the control room. The effect of a small hole in the wall between the control room and studio or machine room (to run a new cable for example) can severely limit the wall's ability to isolate acoustically—as specified in the original design. A wall's transmission loss (TL) can be measured and used as an indication of its effectiveness to deter noise transmission.

To measure the TL, begin by emitting a loud broadband sound source in one room. SPL measurements are then made

PAMS

To facilitate a studio's ability to measure and document its rooms on an ongoing basis, Brüel & Kjær has assembled a portable acoustic measurement system (PAMS). Acoustic and electroacoustic measurements including frequency response, background and ambient noise, spatial uniformity, reverberation time (RT), transmission loss (TL) and noise reduction (NR) can be done via a single system based on a precision sound level meter (SLM).

The heart of this system is the type 2231 modular sound level meter. Various interchangeable application modules are used to adapt the unit for a particular measurement. Once a software module has been inserted, it defines the push-key functions on the front panel. A corresponding front-plate overlay is supplied with each module. Augmented by clip-on, battery-operated filter sets, hard-copy recorders and computers, the unit can function as a central unit. The kit also includes written and video instructions.

The 2231 SLM fulfills IEC and ANSI Type 1 requirements for simple and integrating sound level meters. A parallel detection feature allows the display of both RMS and peak values of the same signal. A, C, Lin. (10Hz to 20kHz), and all-pass (2Hz to 70kHz) frequency weightings are available. The clip-on octave/⅓-octave filter set, type 1625, allows more detailed frequency analysis.

Frequency Analysis: The BZ 7103 is an application module that, when combined with the SLM and filter set, enables serial frequency analysis. This automated measurement method gives a single, accurate and meaningful value for each frequency band. Additionally, the SLM will calculate the average, maximum and minimum levels for each band selected in the analysis.

Reverberation Time: The BZ 7104 reverberation processor allows measurement of reverberation time and early decay time, and records reverberation decays of the impulse responses.

The SLM's memory can store up to nine files, each containing the results from a complete measurement sequence. The BZ 7104 module also provides a facility for digital output of numerical results and reverberation-time bar graphs. Analog output of the impulse response and synthesized reverberation decays is also possible.

Documentation

To document the results, a small graphics printer, type 2318, is available. A simple command provides document printouts that can be inserted into an overall measurement report. Frequency analyses and/or reverberation time measurements can be presented graphically or in tabular form, along with superimposed curves, if desired. For future reference, full documentation of the measurement settings is included in the results. (See Figures A and B.)

Brüel & Kjær and GE Rental, a subsidiary of the General Electric, have joined forces to provide PAMS to the professional audio industry as a rental item.

GE Rental has offices in New York, Atlanta, Chicago and Los Angeles.

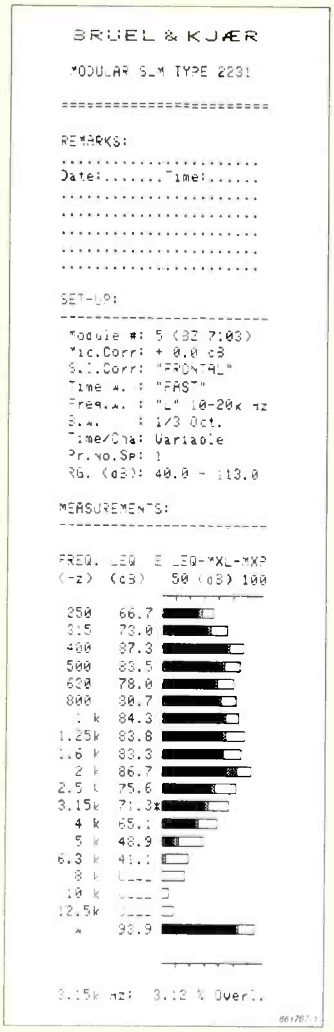

Figure A. *Graphic representation of a frequency analysis done with the 2231 + BZ 7103.*

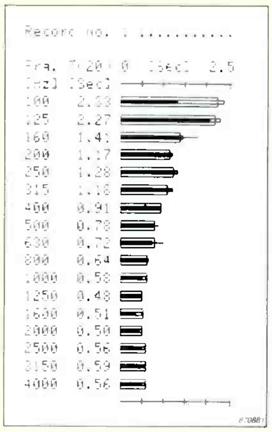

Figure B. *Graphic representation of reverberation times and early decay times (EDT) made with the 2231 + BZ 7104.*

in both the source and receiving rooms. Spatial and temporal averaging, as well as octave or ⅓-octave filtering, are again recommended. The reverberation time (RT) in the receiving room must also be measured. Finally, the surface area of the partition and volume of the receiving room must be determined. Upon obtaining all of these values, the transmission loss of the wall in question can be readily calculated.

Numerous levels and degrees of objective acoustic and electroacoustic measurement techniques are based upon hardware and professional expertise. There are also myriads of subjective opinions and evaluations, none of which is necessarily more correct than any other.

Recent trends in recording techniques are using the acoustics to achieve a desired effect or an overall "sound" on a recording. The acoustic design and construction of control rooms and studios represent a large investment, and a few days out of each year is a small price to pay to protect it and your competitive edge. RE/P

THE CUTTING EDGE

By Laurel Cash, Fred Jones and Ike Benoun

Two Industry Greats Lost: Keith Worsley and Wally Heider

Keith Worsley was a friend to everyone—from engineers, producers and artists to store salesmen, reps, manufacturers and magazine folk. Each of us who had the chance to know him loved him. On January 26, he died of cancer. He was 51.

Before he died, Keith redefined the term "trouper" for the audio biz. It used to mean just keeping your eyes open on the last day of a trade show or at the end of a 20-hour session. Right up to the end, he continued to work. At the last AES, he did Klark-Teknik booth demonstrations from his wheelchair. Now that's a trouper.

Keith was best known for putting the name Lexicon on the industry map. Prior to that, he worked for MCI Canada and, more recently, Klark-Teknik/DDA.

If someone asked us to describe Keith in one sentence, our reply would be that he was a heck of salesman with a wonderful sense of fun. Many were the times that we called him for information, product rush or just about anything. He always responded immediately, from wherever he was. One time, in particular, we called him and left a message. When he returned the call, he was snowed in at O'Hare International after having just been in a traffic accident while taking a cab to the airport. Even so, he was ready with all the answers.

We remember a time in New York when we had dinner with Keith and Bob Taylor from JBL. It was when the AES was still at the Waldorf Astoria, and we were dining in the hotel's Persian (Iranian) restaurant. The next morning, preparing for our breakfast meetings, we called each other in wonder, for the news had just come out that Americans had been taken hostage in Iran. Two days later, the Waldorf closed the restaurant, never to re-open it. To this day, whenever we think of the Iranian hostage crisis, we think of Keith and that evening.

There are no words that will ever describe this incredible, crazy, fun, one-of-a-kind man. If you knew him, he's already touched you. If you didn't, then maybe some of what we've said here will describe what a great loss we've all suffered. One of the last times we saw him was at the bar in the L.A. Hilton, surround-

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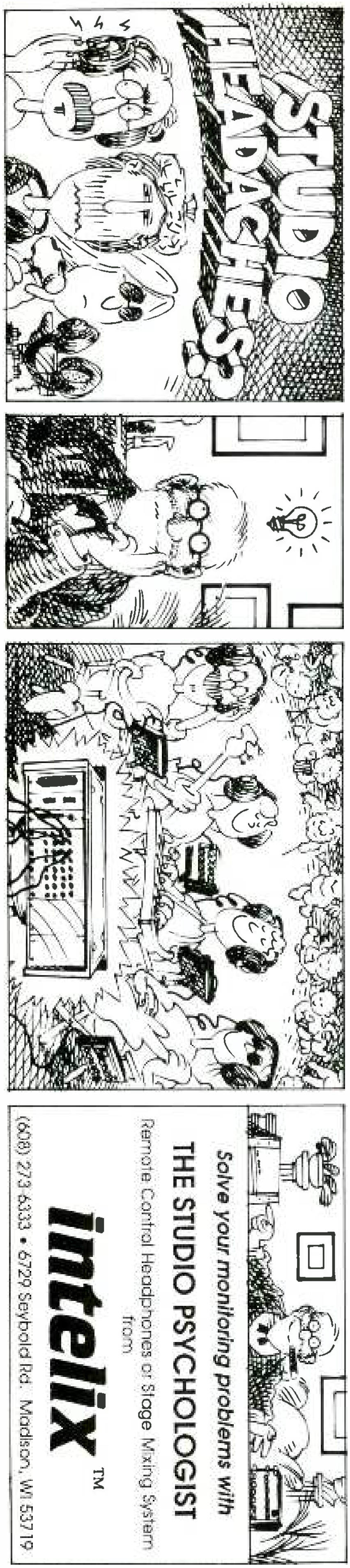
Laurel Cash is *RE/P*'s executive consultant and a free-lance writer based in Los Angeles. Ike Benoun is manager of Audio L.A., a professional audio retailer in Los Angeles. Fred Jones is owner of Fred Jones Recording Services, Hollywood, CA.



ed by his friends and talking shop. We're sure this is how many of us will remember him.

The last night of Keith's life, his family was gathered in his hospital room. He asked to listen to a song that his son Clive had written for his wife. When someone complained that the music was too loud, Keith replied, "Turn it up!" Those were his last words.

On February 19, a celebration of Keith's life took place at the Sequoia Lodge in Oakland, CA. In attendance were his close friends and family.

Wally Heider, the man who invented remote recording as we know it, died March 22 of cancer. He was 66 years old.

Famous for building the industry's first recording faculty on wheels, Heider began his career as a saxophonist, playing big band jazz in his native Oregon. But his real interest was arranging, and while mastering the craft as a music student at the University of Oregon, he formed a 14-piece band.

Heider's first recording studio was a room in Hollywood, where he could mix his remote recordings. He soon branched out, taking over Shelley's Mann Hole, the jazz club next door. Later, he opened a facility in San Francisco, which, in conjunction with his remote trucks, made Wally Heider Recording the premier all-around recording complex on the West Coast.

After selling his recording complex to Filmways, Heider formed Hindsight Records, a label devoted to the restoration of old big band recordings. Later, he founded Swing Time Video. He had obtained the rights to the 5- and 10-minute films of the big bands made in the 1940s and '50s as "shorts" to be shown in movie houses between double-features. They were edited and reissued on videotape.

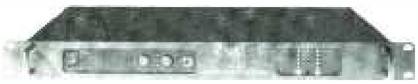
Private services were held in Sheridan, OR, following his death. An industry wake was held in Hollywood on May 5. RE/P

NEW PRODUCTS

Carver PM-100 amplifier

The PM-100 is a single rack-space Magnetic Field power amp designed for a variety of applications. Features include a high-efficiency linear tracking, fully complimentary output stage, the company's exclusive clipping eliminator circuitry, series/parallel mono capability, barrier strip inputs, front-panel metering and a stereo headphone jack. A 3-year warranty is included. Suggested retail price is $629.

Circle (146) on Rapid Facts Card



SSL Screensound recording system

Solid State Logic's digital audio editing, mixing and recording system is designed for off-line video and film post-production, and audio-for-video editing applications. The system interfaces with VTRs/VCRs and film reproducers with full machine control. Using a VDU monitor display, control tablet and pen to edit and mix, the system offers comprehensive splicing, crossfading, time-offsetting and programmable gain-profiling. A WORM optical disc system, for creating and storing library sound clips and effects, and a digital tape streamer system are available for archive storing.

Circle (105) on Rapid Facts Card

Neumann KM 100 mic

The modular condenser microphone system contains a small-diameter capsule that connects directly to the output stage. Four interchangeable capsules are available: the AK 30 omni, AK 40 cardioid, AK 45 cardioid with LF rolloff and AK 50 hypercardioid. A variety of accessories, including stands, extension tubes, goosenecks and cable hangers for mounting, are also available.

Circle (106) on Rapid Facts Card

ARTI video editing system

Advanced Remote Technologies Inc. has introduced a "desktop video" system, similar to a desktop publishing system, that uses IBM PCs/compatibles or an Apple Macintosh to control video post functions. The Executive V.P. (video publishing) Network uses a central processor and a variety of Advanced Remote Modules to provide specific functions. Software programs are window-based or menu-driven for ease of use. According to the company, a basic system can be assembled for less than $5,000; additional remote modules are priced from $895 to $1,295.

Circle (107) on Rapid Facts Card

Ampex 467 digital audiotape

Ampex's 467 DAT cassette uses an advanced formulation of metal particle tape and features the company's DATpak mastering storage system for packaging and labeling. Available lengths are 45, 60, 90 and 120 minutes. The tape shell contains a small tape-view window that maximizes the labeling area.

Circle (108) on Rapid Facts Card

360 Systems Audio Matrix 16

The unit is a programmable, MIDI-controllable patchbay for studio or live use. The 2U device will route signals from instruments, effects sends, channel inserts or other audio sources into effects or signal processing equipment. It controls the routing of up to 16 separate audio inputs or outputs, allowing any input to drive one or multiple outputs. Up to 100 different setups can be stored in internal non-volatile memory. I/O for channels 15 and 16 are duplicated on the front panel, which allows additional equipment to be connected without accessing the rear panel.

Circle (116) on Rapid Facts Card

Time Offset Correction Studio Monitor 2

Available from Professional Audio Systems, the monitors are comprised of a 2-inch throat coaxial loudspeaker with a beryllium diaphragm. Two 15-inch LF transducers with large linear excursion are utilized in the low bass. The High Resolution crossover filters provide high-quality phase response and depth in reverb and stereo imaging. The monitors will achieve levels in excess of 126dB.

Circle (122) on Rapid Facts Card

Neve VR console

The console is the latest version of the V Series. At any point during a session, a complete configuration of console controls can be stored under computer control. In addition, individual channels or sections, such as EQ or dynamics, can be stored at any time. A integral hard disk contains 40Mbytes of memory; an additional 1.4Mbyte 3.5-inch floppy drive can store approximately 140 complete console stores of a 60-channel desk on a single disk. Additional features include a wider center section for increased monitoring facilities, an enhanced aux section and a modified overdub facility.

Circle (144) on Rapid Facts Card

Sound mixing newsletter from Audio Services

"The Production Sound Report," available from Audio Services Corporation, is edited for all persons interested in production sound techniques for motion pictures and video. The newsletter is edited by Fred Ginsburg and is free.

Circle (130) on Rapid Facts Card

Alesis 1622 mixer

The 1622 contains a 2-bus console featuring 16 line inputs on ¼-inch jacks, eight XLR mic inputs and eight direct outs on Channels 1-8. High- and low-shelving filters provide ±12dB of boost/cut at 100Hz and 10kHz. Also included are inserts on every channel, as well as on the

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Hardware and software updates

Nakamichi power amps

The PA-7 and PA-5 power amps have been redesigned to increase their continuous power ratings and output current capability. The PA-7AII has a continuous power rating of 225W per channel and can supply peak currents of 90A per channel, more than double that of its predecessor. The PA-5AII delivers 150W per channel continuously, a 50% increase with a peak current of 70A per channel. Prices are $2,195 for the PA-7AII and $1,317 for the PA-5AII.

Circle (133) on Rapid Facts Card

Adams-Smith 2600 A/V updates

The audio-for-video editor has several new features, including automatic track selection, logging and track sheet printout using the 2600 TMS Track Management System, a motion control knob for use with VTRs, bi-phase dubbers and C:Sound, and enhanced edit list formats and management.

Circle (134) on Rapid Facts Card

Roland FC-100 Mark II foot controller

The Mark II is an updated version of the FC-100, and can be used to control any MIDI instrument when connected to the Roland RMC-1 RRC-to-MIDI converter. MIDI implementation has been expanded and a second expression pedal has been added. The unit transmits and receives MIDI messages, which ensures that patches on both the master unit and remote units will change when patch change commands are performed.

Circle (135) on Rapid Facts Card

E-mu price reductions

E-mu Systems has reduced the price for the entire Emulator III family, the result of reduced manufacturing costs and an attempt to broaden the systems' market. Suggested list prices are: E-III 4Mbyte version (rack or keyboard), $9,995; E-III 8Mbyte version, $13,995; E-III HD300, including 10 sound banks, $3,995; and the E-III RM45, $1,995.

Circle (136) on Rapid Facts Card

Soundmaster editing system update

Soundmaster International has added random access digital audio to the Integrated Audio Editing System. Available in 2-track modules with various hard disk configurations, the update provides users with the ability to integrate traditional and emerging technologies and to use the most appropriate medium.

Circle (137) on Rapid Facts Card

Alesis HR-16:B drum machine

With an operating system identical to the HR-16, the HR-16:B features all-new samples and sample composites. The HR-16:B also contains a software update that allows users to link two machines via MIDI to create a comprehensive electronic percussion system. List price is $499.

Circle (140) on Rapid Facts Card

Upgrade for Passport NoteWriter software

The Macintosh-based music notation program now features an extremely fast, versatile and easy-to-learn set of tools for entering and editing musical symbols. The QuickScrawl mode allows notes to be sketched onto music staves with the Mac's mouse and automatically converts the notes into the intended musical symbols. For example, a zig-zag becomes an eighth note after releasing the mouse button.

Circle (177) on Rapid Facts Card

TAC Matchless update

The 24-bus in-line console has been updated. As well as optional VU meters on the left, right and PFL buses, an all-VU version is now available, with 27 VU meters as standard. Additional features on the standard console include an input reverse switch, which, coupled with the provision of both line and tape monitor inputs on the jackfield, enhances the routing system. Users wanting to use the console with a 32-track tape machine may order the console with eight additional meters, with an additional tape/input return on the patchbay. JMS C Mix, a PC-based fader and mute automation system, is now available.

Circle (178) on Rapid Facts Card

NEW PRODUCTS

subs and masters, six aux sends and eight returns configurable as four stereo pairs. List price is $799.

Circle (141) on Rapid Facts Card

Alesis MEG-230

The ⅓-octave room equalization system is a dual 30-band, ⅓-octave system in a single rack space. Interface is provided through ¼-inch and RCA jacks on the rear panel. Center frequencies range from 25Hz to 20kHz, with each band providing ±12dB cut/boost. Center frequencies are set to ANSI/ISO standards. List price is $199.

Circle (142) on Rapid Facts Card

Tannoy NFM-8 close field monitor

The NFM-8 contains a crossed brace from the rear of the drive unit to the rear of the cabinet and uses an 8-inch dual concentric drive unit. Features include an improved roll-surround design, to ensure linearity and robustness, and a new HF unit that uses the company's research into Differential Material Technology, which provides the rigidity of titanium without the HF breakup.

Circle (124) on Rapid Facts Card

Juice Goose R-Series

The R-Series is a rack-mountable line conditioner that provides constant 120Vac power. The unit provides constant power even when voltage varies between 90V and 130V, and the regulated voltage range can be as wide as 50V to 150V at lower current load levels. Available in 150W, 300W and 600W models, the unit has six ac outlets on the back panel. RE/P

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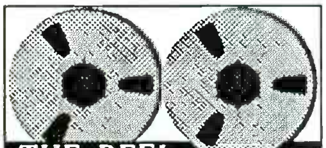

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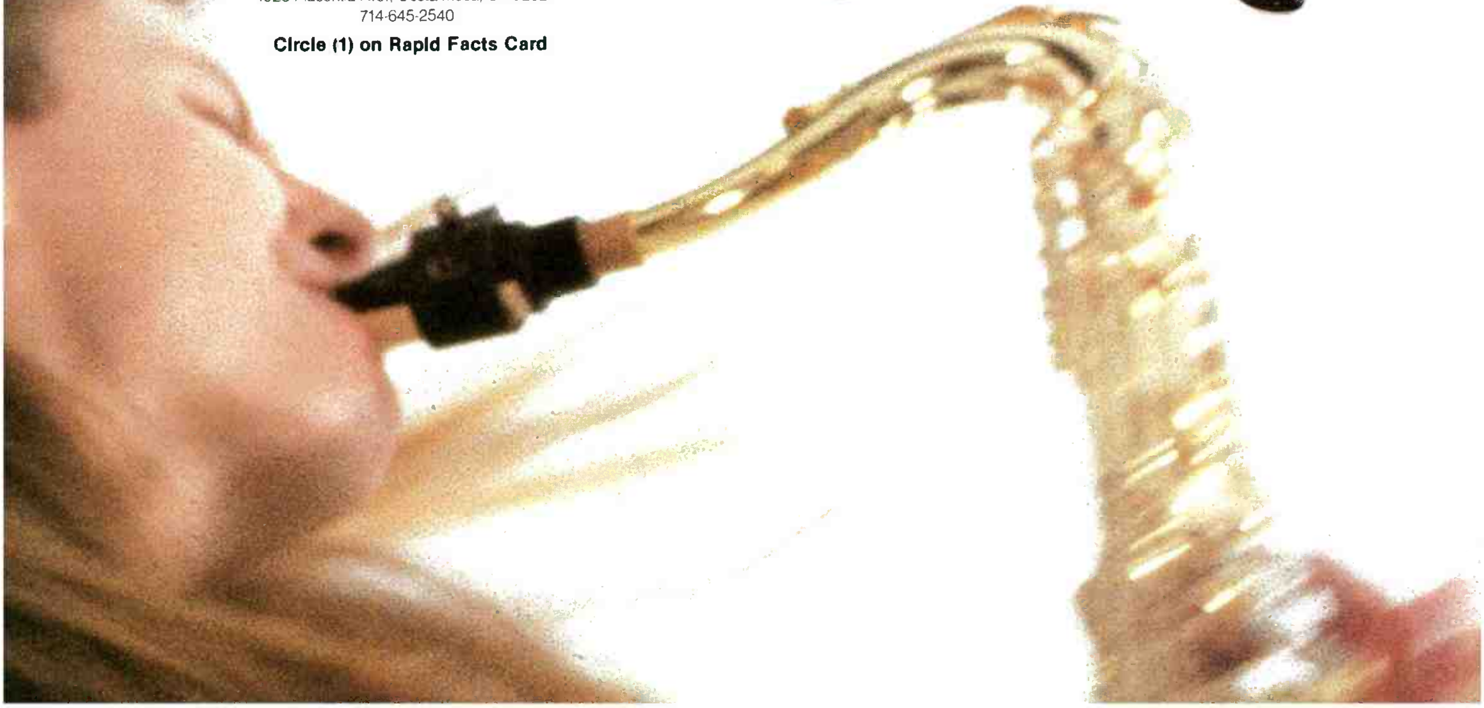

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TRUTH: A lot of monitors "color" their sound. They don't deliver truly flat response. Their technology is full of compromises. Their components are from a variety of sources, and not designed to precisely integrate with each other.
CONSEQUENCES: Bad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

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

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

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

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

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

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

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