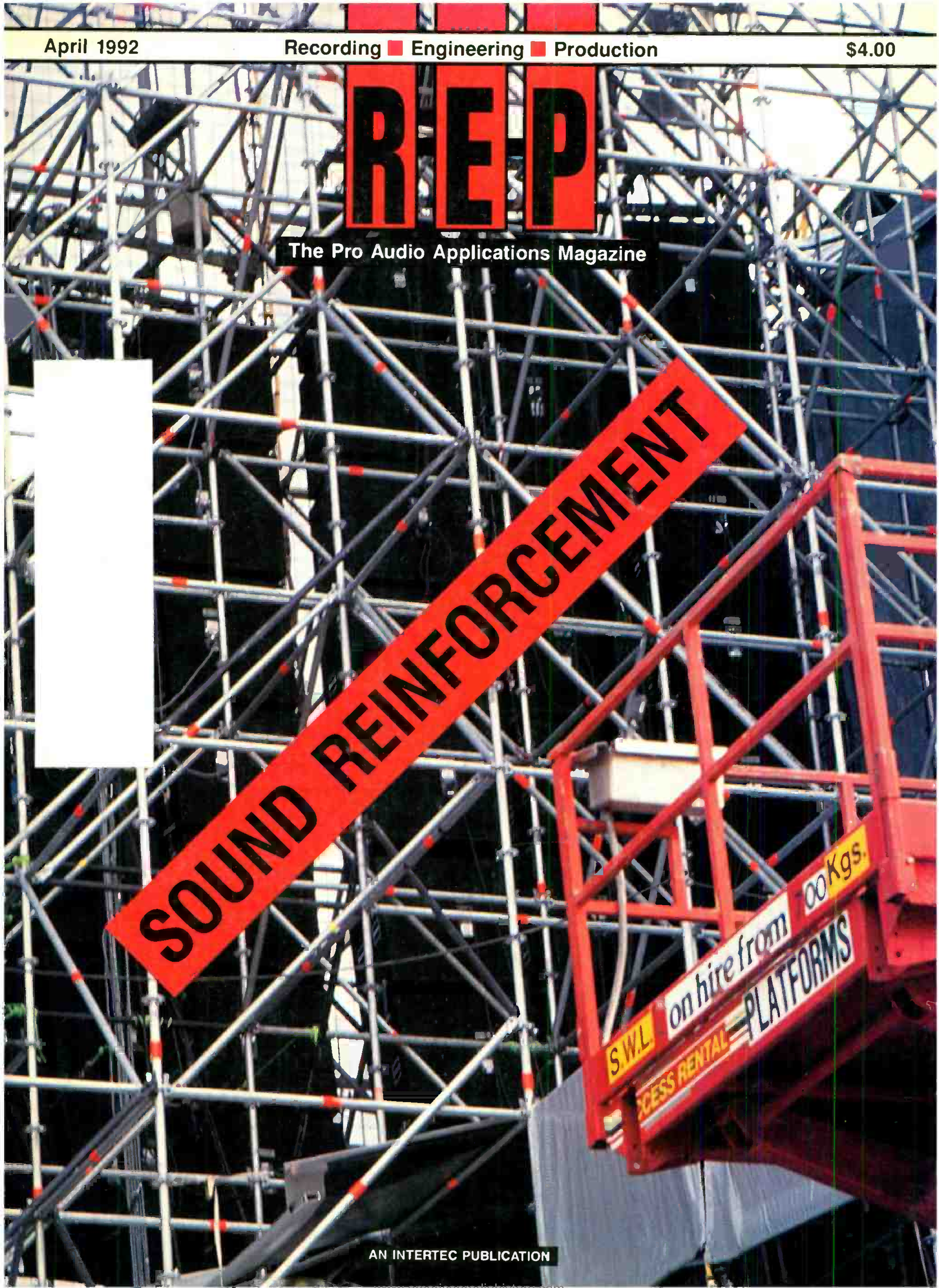
April 1992 Recording ■ Engineering ■ Production $4.00

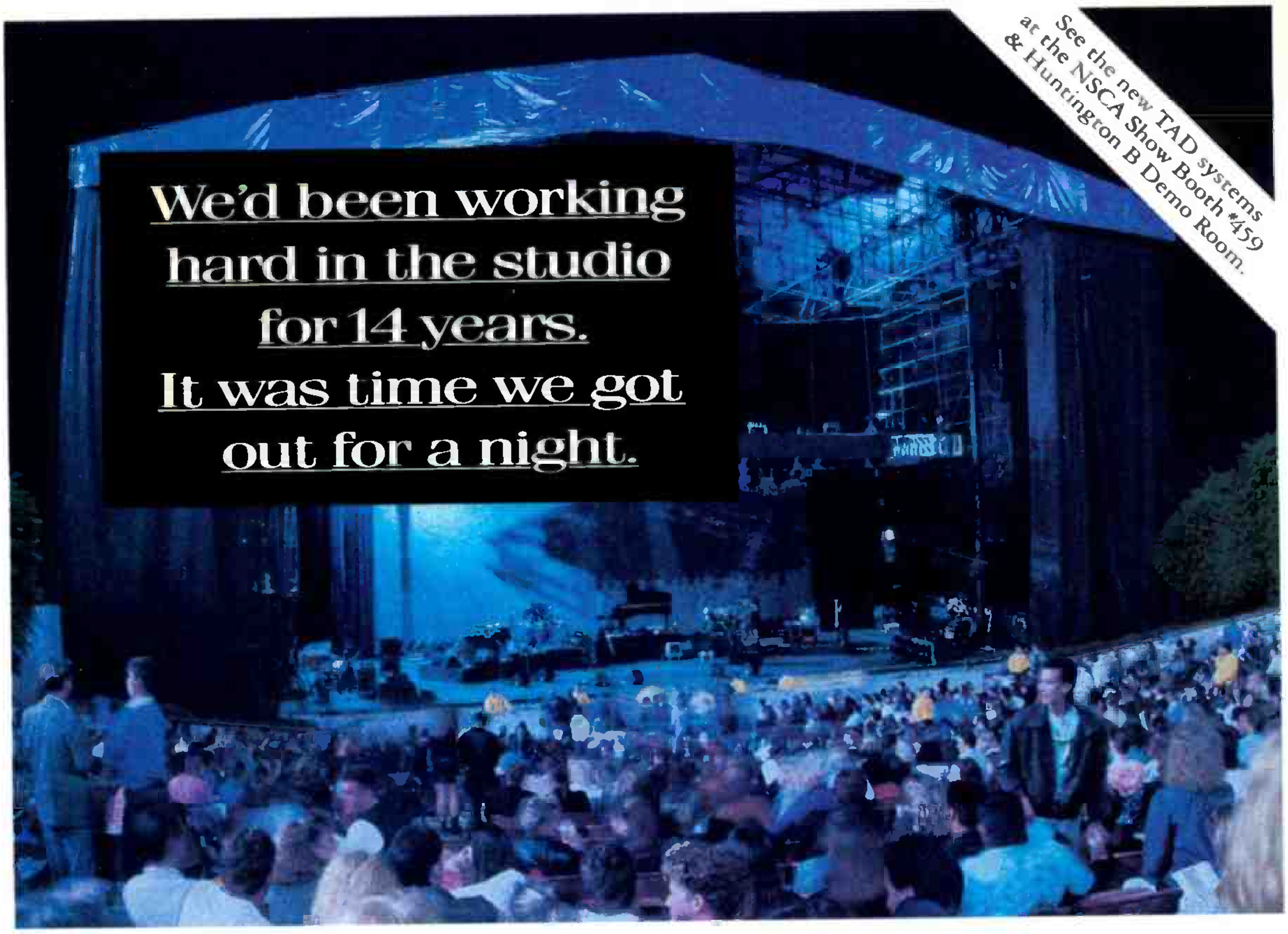
REP

The Pro Audio Applications Magazine



SOUND REINFORCEMENT

AN INTERTEC PUBLICATION



Spending years on end cooped up in small, dark rooms with a bunch of engineers takes certain special qualities. Durability, for one. We've always been known for that. Of course, clear, uncolored sound quality doesn't hurt, either. Or hand-assembled components, with gap precision to plus or minus one-millionth of an inch.

These features got TAD speakers into studios like Record Plant, NOMIS and Masterfonics. And the same features are now getting us out of them.

See, we had this funny idea that if TAD could make music sound terrific in a small room, we could make music sound terrific in a huge arena. And every outing we've had with Maryland Sound has proved us right.

Not that we won't still work our woofers off in studios from L.A. to London all day. But, at night, we'd like to get out and jam more often.

TAD Technical Audio Devices®
Pioneer Professional Products Division

Circle (1) on Rapid Facts Card

THESE CONSOLES SO MANY FEATUR COULDN'T FIT THE ALL ON THIS PAGE

SOLO. A new breed of console packed with more features per square inch than anything in its class. And the pure, transparent sound that has made Soundtracs so popular in studios and on stages around the world. At prices that make sense for today's cost-conscious professionals.

SOLO LIVE. Available in 16, 24 and 32 input frame sizes. Four independent sub-groups, right/left master and mono sum output. Four band EQ with two swept mids. Six auxiliary sends. Balanced inputs and outputs. Four stereo effect returns. 48V phantom powering for all mic inputs. Raised meter bridge.

SOLO MIDI RECORDING. Available in 16, 24 and 32 input frame sizes. Automated MIDI Muting on all channel inputs, monitor inputs, group outputs, stereo effect returns and auxiliary masters. Four band EQ with two swept Mids, assignable to monitor inputs. Six auxiliary sends—four assignable to monitor inputs. Four stereo effect returns with two band EQ, balance and level controls. Raised meter bridge.

We wanted to list *all* of the features on SOLO consoles but we ran out of space. If you want to find out more about ev

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Circle (3) on Rapid Facts Card

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On the Cover

Photo courtesy of Rudy Arias.

R•E•P and **Live Sound! and Touring Technology** now have a presence on the **CompuServe Information Service** network! Submit article ideas, create Letters-to-the-Editor, download articles from our On-Line library or have a "live" conversation with our staff and contributors on the **R•E•P: On-Line Forum.** To receive a free **CompuServe** membership and sign-up kit, courtesy of **R•E•P**, dial **1-800-524-3388** in the **U.S.** or **Canada** and ask for representative **#232.** Once on the network, type **GO REPMAG** at any prompt to navigate directly to the **MIDI VENDER FORUM, SECTION 14.** Or reach the Editor's Mailbox at **75300,3141** by typing **GO MAIL** at any prompt. See you On-Line!

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This is real life. The frog won't turn into a prince and his voice won't sound like Diana Ross — at Brüel & Kjaer capturing reality is what we're all about.
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The B&K 4000 series offers 6 different condenser mics, featuring true omni or cardioid pickup which can be passively and accurately altered by our exclusive *acoustic pressure equalizer* (APE) adaptor. They're available in both 48 Volt Phantom or 130 Volt powered versions offering unmatched 168dB dynamic range for the most demanding snare drum crack to the subtlest details of natures acoustical ensembles.
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R•E•P is an applications-based publication targeted at professional individuals and companies active in the commercial business of studio and field recording, audio for video, live sound production and related fields. Editorial content includes descriptions and demonstrations of audio production techniques, new products, equipment application, maintenance and audio environment design.

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From the Top

Conflicts of Purpose

Whenever I speak with someone outside our industry, somebody old enough to articulate their interests and beliefs in a coherent manner yet young enough to still *enjoy* going to arena and stadium concerts, the subject inevitably comes around to this: Why do they have to play so damn loud?

The people I speak with may not realize it, of course, but what they are asking is: Why does the crew mixing the show feel obligated to hit the SPL ceiling? A valid question. The commentor usually doesn't know the details — that a mixer may feel the audience is creating so much noise itself that the system should be run up just to keep the revelers in check (in effect saying that it needs to be that loud to get over the top of the screams); or that the band is generating 105dBa of ROAR! at 100 feet *all by itself* right off the stage, and maybe the mixer is just trying to punch the vocals out over it.

Are these really valid reasons? Maybe, or sometimes, or often, qualified by the details of tackling horrible stadium acoustics designed to reinforce the perception of hugeness and excitement at sporting events, not control slapback and room reverb decay. Is the excuse of a "situationally difficult working environment" a reason to cruise for several hours at literally painful volume levels? Probably not.

At the risk of sounding like a broken vinyl disc, the SPL level situation in far too many concert environments is getting out of hand. We won't do the soapbox thing here today — there are too many of you out there who carry the flag higher and shout louder on that subject than we ever could; but we will point out several important and noteworthy considerations for those of you who do mix live sound. And from our reader surveys, we know that a very large percentage of production environment audio mixers also mix live.

First, whereas in the old days (pre-'80s), you needed a mega-ton battleship of a system to hit 120dBa cruise at 100 feet all night long. Today, the price/performance devlopments have allowed supreme SPL levels to be attainable by anyone who can buy a stake truck's worth of off-the-shelf trap boxes and half a dozen multi-kilowatt, double rack- space amps. Hell, put them on the AMEX card for the show next Sunday. This is generally a good thing, in that along with the acoustic power has come superior fidelity, but on the down side, lack of judicious and tempered operation can break eardrums. And unfortunately, hearing damage being cumulative, what starts out as a pleasant and "powerful" experience early in the show becomes a serious problem at 1 a.m. when the headliner crashes into its third encore.

Second, an ever larger percentage of major touring groups represent musical styles that hit their popularity peaks sometime behind us — be it the '70s or the '80s. They attract, by definition, a slightly older crowd, maybe averaging 25 to 40, instead of the 16 to 25 crowd who, really, truly, amazingly and sincerely loves meltdown acoustic levels.

What does this mean, exactly? We know hearing changes with age. The functionality of the mechanism becomes less resilient. Natural losses occur. The ability of the muscles around the eardrum to clamp down and naturally "limit" excursions decreases (look it up if you want the details — they're fascinating!). The sensation of pain onset approaches more rapidly, at lower levels, more quickly, with age. Like an older car, push it too hard and it lets you know that the limits are closer at hand.

So here we are with faster, louder gear, often playing to a large segment of an audience that really is more comfortable with reasonable levels, advanced clarity, and higher overall fidelity. Why the latter? Well, the past half-dozen years have seen the massive adoption of digital CD fidelity in the consumer world. *Everyone* has a great-sounding stereo. Even TV has wonderful-sounding audio, which a large number of our ersatz concert goers pipe through their stereos. Get the point? Even a $50 portable stereo walk-thing with $10 headphones sounds killer and gets loud (enough) with superior fidelity to give the most audio technology-illiterate music consumer a relative high standard with which to compare the sound they hear coming out of the flying black boxes. In short, people today have a strong sense of "this sounds good/bad."

What to do? It's our job to figure it out, collectively. We've gotta do it. It involves the obvious things: flatter, smoother-bandwidth radiating devices; control of the acoustic environment itself, meaning modification of the architecture (or at least acoustic treatment) in so many venues; the tighter control of SPLs coming off the stage; better, more effectively "personalized" monitor systems; superior speaker system array design and measurement, allowing more effective control of component directivity and working critical distance (ie: keep the room reverb buildup and slap echoes to a minimum). Clearly, there's much more.

The act of controlling levels has to be as much an attitude shift among system drivers as anything. Let's just keep saying to ourselves: Louder ain't necessarily better. We've already mastered level, bass response and punch. Let's go for transparent depth, clarity and acoustic control. Let's give people the excitement of the visuals, support it with sound quality they're used to, all at a level they can live with. ■

Mike Joseph
Editor

Letters

Kudos, Part III

From: Steven Durr, SDA, Nashville.

I read the review of the Meyer Sound HD-1s in your November 1991 issue, just to see how a trade magazine would rate a pair of $5,000 near filed monitors.

Much to my surprise, not only did you write what I feel to be the truth, but the boxed note that described the monitors as "not musical" and "not fun to listen to" took me by total surprise.

As a studio designer with over 20 years experience designing control rooms and sometimes spending 20 days a month listening to different control room monitors, I would like to thank you for your honesty. I have always felt it was a disgrace to read an article or review about a new speaker, console, or a new sound system, which will change the way the world revolves, only to read the fine print under the author's name: so and so is marketing director for the manufacturer.

I have a sincere respect for Mr. Meyer's sound reinforcement products, but as to Meyer's ensuing comments (Letters, January 1992) that "not musical" and "not fun to listen to" are subjective and therefore not relevent, I feel, and my clients will agree, that the two most important characteristics a control room monitor can have are *fun to listen to* and *musical*. The fact that a speaker measures +/-1dB under specific circumstances is not on my list of requirements.

We have been designing our own monitor system based on TAD components for years, because very few manufacturers understand these two incredibly important "subjective terms."

There are many young people who are entering our business each year and we have an obligation to them to "Tell it like it is," (incredibly musical and fun to listen to). I would be happy to pay for a magazine that offers a realistic picture of our business.

Your editorial in the January issue was right on course. Please keep moving ahead. Thanks! Looking forward to your next issue!

From: John R. McGriff, Nashville.

I want to express my support for more of the type of balanced, honest and in-depth equipment reviews that you did on the Meyer HD-1 monitor. I thought this review was one of the finer ones I have read anywhere. It did a good job of providing the information that is relevant to professional users of equipment; we often operate equipment at (or even beyond) its limits, and need to know both its strengths and limitations in order to extract the most performance from it. For instance, in the case of the HD-1s, I learned that probably the best way to set them up is to lay them on their sides, facing straight out rather than angled in toward the listening position.

Reviews that simply provide the "Rah-Rah" factor for marketing departments do not provide any true service to the reader. Complete (read "minimally edited") comment by professional users who have experienced both the good points and the "warts" of the equipment is also an important part of the reviews. When choosing which magazines to spend my time reading, complete, accurate and honest information is very high on my list of priorities. In order for readers to trust the magazine, the magazine must display the same sort of faith and trust in its readers to understand and apply the information presented to their work. R•E•P, after all, is a professional magazine solely for professional readers. I often wish for *even more* detailed and technical articles.

Please continue to feed your readership reviews that contain honest evaluation, detailed data, a balance of technical vs. listening and use tests and comparisons with other equipment.

From: Geoff Gray, Far & Away Studios, Chester, NY.

I just finished reading your January 1992 "From The Top" editorial, and felt agitated enough to write. Of course *real* reviews are welcomed by the pro audio community. Any of those companies who react negatively by pulling their ads will be off-the-block when the word gets around about a poor product. Your accounts receivable will be chasing them soon anyway; No Loss!

I always admired the late Hugh Ford's honesty in *Studio Sound's* reviews. No product is perfect and it's that much more believable when a flaw is criticized and honestly challenged.

Wear It Out

From: Jeffrey Silberman, earWear-C, Baltimore, MD.

Several years ago, R•E•P ran an article entitled, "dB's can be Hazardous to your Health." I quite agree. My hearing, unfortunately, has been damaged due to excessive sound levels. I suffer from a condition known as "tinnitus," a high-pitched ringing, whistling or buzzing in the ears. As you may know, hearing loss generally is a slow process caused by years of listening to sounds above 90dB. One may only do a small amount of damage in one instance, but each successive exposure can cause a little more damage until one has lost a good portion of one's hearing. Thus, the ear may be damaged without the damage being perceived. Most people do not realize the loss of high frequency sounds — the first to go — until the loss begins interfering with communication. By then, of course, it is too late.

Given the insidiousness of hearing loss, informing your readers of this risk is vital. Moreover, it is imperative to inform them of the availability and necessity of hearing protection. Conventional earplugs, unfortunately, designed to withstand extremely high sound levels, ie: firearms and jackhammers, attenuate more than necessary in moderately high sound level environments such as rock concerts, nightclubs and mixing rooms.

Furthermore, conventional earplugs attenuate high frequency sound up to 20dB more than low frequency sound, resulting in a change in the frequency balance — the music sounds muffled. Ideally, an earplug should mirror the natural frequency response of the open ear, but at a reduced sound level. Because the music industry relies upon hearing, I suspect your magazine would share this goal.

Happily, ear protection has been designed with music in mind, namely, the Hi-Fi Earplug. The Hi-Fi Earplug gives wearers the best of both worlds: adequate protection and uncompromising fidelity. I have enclosed a fact sheet on this product:

Hi-Fi Earplug for those who refuse to wear conventional earplugs because of a need to hear clearly. Contact earWear-C, 4001 Old Court Rd, Suite 517, Baltimore, Maryland, 21208.

As I hope to begin offering this product in the very near future, I would appreciate a quick response to this letter. I would be happy to provide a pair of Hi-Fi Earplugs for your magazine to demonstrate. Thank you very much. Hear today, hear tomorrow ...

Is Anyone Listening?

From: Dan Alexander, Dan Alexander Audio, Berkeley, CA.

I noticed that we were not listed in your recent article on Neve modules. We are the largest supplier of old Neve modules (indeed, we are major suppliers to two of the companies you listed!) in the U.S. I was disappointed.

The proliferation of the "portable preamp" syndrome that has come into vogue as of late, is an interesting comment on the current mixing consoles on the market. Every hip engineer has a box of mic pre's that he carries from room to room, bypassing the studio's $400,000 console! Whether one carries Neve modules, or Trident A range modules, or Langevin tube preamps, the message would seem to be clear. Are any manufacturers listening? ■

Send letters to R•E•P, Box 12901, Overland Park, KS 66282; or fax 913-541-6697. Letters must be signed and may be edited for length and clarity.

Who said a workhorse can't be a thoroughbred?

The Sony PCM-7010 DAT Recorder

It wasn't Sony. Because the PCM-7010 was built from the ground up as a professional DAT recorder that can handle everything from music recording and on-air radio and television broadcasting to audio-for-video production and corporate multimedia systems.

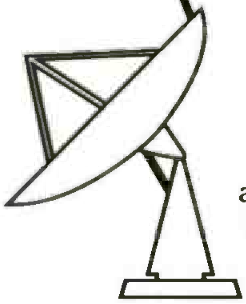
The PCM-7010 features high-speed search, variable-speed playback, punch-in/out with cross-fade and confidence monitoring. And, with its advanced options, you can record, playback and display SMPTE time code and store digital audio in memory for instant-start playback. If you want a workhorse DAT recorder that can do it all, today and tomorrow, you want the Sony PCM-7010. For more information, call the Sony Professional Audio Group at 1-800-635-SONY, ext. 7010.

SONY

Random Access

BIG GIG IN THE SKY

Three digital cable audio services have found their way into the U.S. marketplace. Digital Music Express (DMX), Digital Cable Radio (DCR) and Digital Planet are available by satellite downlink to your cable operator or your satellite dish, ready for in-home listening pleasure – assuming you have paid for the proper converter. The result is a life-and-death struggle to load up international cable systems as fast as possible.



While Digital Planet does provide "compatible announcing" the prevailing rationale is that no jocks and no commercials makes for a better service. But questions remain. For many, music-only programming lacks radio's pulse of human personality and the quirkiness/immediacy. And one wonders how many people will pay for the same music they get free from traditional broadcasting just to ditch the advertising. But the biggest issue is, "Will programming/playlists be improved, expanded and liberated from drive-time constraints? Or is cable more of the same old song and dance without the jocks and ads?"

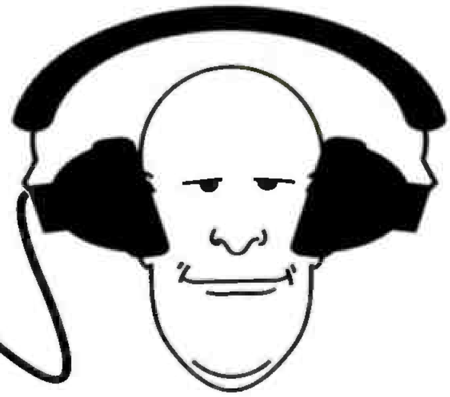
At first blush this cable audio programming seems to be breaking little, if any, ground. These digital audio providers appear to be the aural equivalent of Wal-Mart, K-Mart and Sears. The real trend may be consolidation of vendors and homogenization of services. Admittedly, the existing services have begun to include classical and even some Latin music programming. But a lowest common denominator grab for market share seems at work here.

One ballyhooed side issue is the salesman's pitch that end-to-end digital program material is desirable due to technical superiority. But can J.Q. Public discriminate between FM broadcast of CDs and digital delivery of the same CD? Doubtful. In reality, the potential ancillary markets now served by Musak and the other foreground/background music systems seem best suited for digital cable radio. But the guys driving these systems have much bigger appetites than dentist offices and mall retailers. They won't be satisified until they are in your home.

For artists, digital cable radio re-ignites the whole copyright bag of worms. With on-line advance programming print-outs (they can do this), this service is a 24-hour digital music subscription source for anyone capable of programming a recording device. Over the span of a month, your new release could be captured by millions of listeners for a minimal subscriber fee.

Hoping to capitalize on, rather than being consumed by, downlink audio, Tower Records has conspired with DCR to offer a toll free phone number for direct-programming-response sales. But if cable audio succeeds, the eventual impact of this arrangement could slowly erode all record store traffic and reduce chance encounters (re: sales) with other (re: non-superstar/fringe) artists.

Assuming people go for on-line music-only programming, right-here, right-now music could evaporate everywhere except in



the car and portable radio markets. And in a worst case scenario, radio might be abandoned to only the extreme versions of tabloid news/talk services that now dominate the AM band.

So producers and broadcasters beware. Seize the moment to create and program with more ingenuity and less conformity, or suffer the consequences. A wealth of music will soon flow from project studios worldwide. It may save your hide from obsolescence.

– Anthony McLean

The implied capacity of digital cable radio should lead to narrowcasting and two of the main competitors, DMX and DCR, hint of eventually providing as many as 250 channels. But the Japanese already enjoy incredibly diverse programming from Osaka Usen Broadcasting Corporation. Included in the existing 440 channels are such favorites as the all-Elvis channel and the bleating sheep channel (to be used as a sleeping aid). Our favorite, a trio of all-alibi channels, is presumably used to provide SFX backup for phone calls to the boss or spouse. News, sports and traditional Japanese Enka are provided in addition to environmental ambience and more than 440 international music channels.

Using digital compression, the service plans to deliver 1,000 channels by the end of 1993.

PEOPLE

Shure Brothers has appointed **Alan B. Shirley** to its newly created position of manager, technical markets and strategic planning ... **Michael Pappas** has been named to the newly created position of professional audio liaison for Crown ... At Lexicon, **Lisa Allen** was promoted to sales and marketing administration manager; **Will Eggleston** assumes duties as product development manager; and **Jeff Largent** has been promoted to East Coast and Midwest OPUS regional sales manager ... Crest's new mixing console division, called CrestMix, is headed by **Chuck Augustowski** as division manager/sales and product manager. Joining him is another Allen-Heath alumnus and engineer **John Petrucelli** ... Eastern Acoustic Works has named **Beverly Brignolo-Seidler** to the position of director of sales operations ... **Jack Kelly**, founder and former president of Klark-Teknik will serve in the same capacity for the newly formed Group One, Ltd. Also onboard is **Chris Fichera**, a 2-time Grammy Award winner, who will focus his energies on direct customer contact nationwide, with primary responsibility for the West Coast ... HM Electronics (HME), Communications Systems Division, has appointed **Mr. C. (Cees) J. Weij** to the position of European sales manager ... The Sony Professional Tape Division has announced the promotion of **Kenneth F. Wiedeman** to vice president of sales and marketing, and the naming of **Joseph E. Tibensky** as director of marketing ... Eighth Day Sound Systems has announced the addition of **Mr. Joel Solloway** to "head up" the new division of corporate staging services and special event management ... **Dan Price** has left Bert and Barz Company to launch Oink Ink Radio, a radio creative services company with offices in midtown Manhattan and downtown Philadelphia ... **Karen Manning** has been named West Coast artist relations manager for the Westwood One Radio Networks ... **Jack Sonni** has been appointed to the position of applications specialist for Rivera Research and Development, a division of JBL Professional ... Intersound announced the appointment of two new employees: **Fred Diether** as chief technical engineer and **Serge Perron** as chief remix engineer ... **Paul Basson** has been named vice president of European operations for Avid Technology and will be based in Monaco ... **Mike Smyth** has been named U.S. operations manager for Audio Processing Technology ... **L. John (Jack) Spring** has joined Allied Film & Video in the new position of director of sales and marketing ... Ferrofluidics announced the appointment of **Jan R. Kirk** to the position of vice president and chief financial officer ... **Donna Bisbiglia**, a Bay Area video production veteran, has joined Tam Productions as a line producer ... **Kathy Fry** is Music Annex's new sales representative based in Southern CA.

It's About Time

"In sound systems it is often necessary to connect pieces of equipment from different manufacturers. This standard provides a common scheme for wiring the connectors used – particularly to avoid the inversion of absolute polarity among the items in a signal chain. "... from the AES Journal Jan./Feb. 1992 which decreed that **Pin 2 on 3-pin connectors is Positive Polarity**. Case Closed.

Pushing the Envelope

The high frontier of non-traditional music is expanding. According to *Mondo 2000* magazine, MIT's experimental music facility inside the Media Lab's Music & Cognition Center is the vortex of "... some of the grandest and most creative exploration of 21st century entertainment technologies." Especially interesting to anyone who has ever performed or recorded music is the concept of "hypermusic." In the "hypermusic" model, which is integrated in the MIT concept, sounds that humans do not cognitively perceive (below/above 20Hz-20kHz), are still intentionally processed as program material and assumed to be perceived in ways other than the traditional ear-brain connection. The anticipated outcome of this approach will blur the traditionally distinct "boundaries between music sound and vibration."
Great concept, especially if you can dance to it!

Good Seats Still Available!

Anyone with $8, able to get out of the house long enough, was promised admission to Woodstock. In reality, gate/fence crashing made tickets useless, so most tickets went unsold – until now. One baby boom entrepreneur has begun marketing framed 3-day festival admissions on a 14" x 18" plaque and a smaller single day ticket version for $300 and $100 respectively. For more information call Barnes Ltd. at 800-677-1969 in KS 913-422-7992.

"We're not playing music, we're playing data."

— Jerry Rubinstein, founder of International Cablecasting Technologies. *Forbes*, March 30, 1992.

Random Access

STUDIO UPDATE

Facility/Location	Details
NORTHEAST	
Multivision/Needham, MA	Purchased two Yamaha DMC1000 digital mixing consoles to be used in two new all-digital post-pduction suites.
Taylor Made Productions/Caldwell, NJ	Has added a 40-channel Optifile system to their existing Harrison Raven console.
Studio 4/Philadelphia	Recently installed a Sony 3024 digital 24-track machine in their B room.
East Hill Studios/New York	Completed installation of a 64-input SSL G Series console with Ultimation in Studio A.
SOUTHEAST	
A Cut Above/Nashville	Has purchased a Studer A827 24-track recorder.
Recording Arts Studio/Nashville	Finished a 64-channel Optifile retrofit on their Soundcraft 3200 console.
Treasure Isle/Nashville	Installed a Tascam M700 console and ATR-80 multitrack recorder in their newly remodelled mix room.
GRC Studios and Tonemaster Studio, Baltimore	Completed studio rewiring and installation of Sony APR-24 recorder and a custom MCI JH00 Series console.
MIDWEST	
Editel/Chicago	Installed an AMS Logic 2 digital mixing console and two AMS 16-output AudioFile PLUS systems.
Soundtrek/Kansas City, MO	Completed studio expansion including a Lexicon OPUS with D2 video editing.
Chapman Recording Studios/Kansas City, MO	Recently brought a Sony 3324 digital 24-track recorder online.
SOUTHERN CALIFORNIA	
Soundelux/Hollywood	Added three WaveFrame 400 systems.
HAWAII	
KHET-TV-PBS/Honolulu	Finished a new video post production suite with hardware including a D2, Betacam SP and a Neve 8232. An Adams Smith AV2600 system controls two Sony PCM 7050s, two Sony CDP 2700s, and Otari 16-, 8-, and 2-track machines.
Sea-West/Pahoa	Expanded their MIDI facilities with an Atari Mega-4 computer using C-Lab creator and two Panasonic SV-3700 DAT recorders.
FINLAND	
Oulu Music Conservatory	Has taken delivery of a Soundtracs Quartz 32 and Megas Mix console.
SOUTH KOREA	
Yae Sung Records	Has added a 48-input, split-frame Neotek Elite console with Uptown Automation.
DESIGNERS	
Russ Berger Design Group/Dallas	Has begun modernization design of existing stages, the addition of video suites, rehearsal studios, audio mixing control rooms and relocation of Sony's entire 52nd Street New York City recording studios into the recently acquired 20th Century Fox 54th Street film stages.

NEWS NOTES

Studiomaster announced that Studiomaster U.K. has been purchased by the companys' management. The new ownership took over operations in November 1991 after legal and financial problems of the previous ownership caused a temporary cessation of company operations.

Crest Audio Inc. has formed a new division for the design and manufacture of a full range of low- and medium-priced professional mixer products. Development will be directed initially toward live sound and recording applications, with the first series of products slated for a summer 1992 introduction.

Solid State Logic announced changes to its printed circuit board manufacturing process which will avoid toxic environmental releases. Previously the company had used traditional solvent baths to clean its PCBs.

DDA/K-T appointed the newly formed Group One to act as the national marketing specialists for their recording console group. Group One will also represent Uptown Automation in the retrofit market, and will serve as the exclusive importer and distributor for DynaudioAccoustic monitor loudspeakers and Milabb microphones. Group One will be headquartered at 100 Sea Lane, Farmingdale, NY 11735, with an additional office in Marina Del Rey, CA.

A new production music library of small ensembles and solo instruments has been released by **Signature Music Library**. Performed on real instruments, this "Light & Lively" CD joins Signature's catalog of 16 volumes.

ENSONIQ and C-LAB software announced a new distribution agreement, with ENSONIQ being appointed the authorized U.S. distributor of all C-LAB products.

BASF announced the restructuring of its North American magnetic media business. BASF's research and development headquarters and professional audiotape production at the Bedford, MA, plant will be discontinued by the end of April. Production for that product line will be concentrated in BASF's European sites.

Museatex Audio of Calgary, Alberta has purchased technology rights and certain assets of the Shure consumer Home Theater Sound (HTS) business.

CORRECTION

Emil Handke, of **Otari Corporation**, has been promoted to sales operations manager, not national sales manager as misstated in January R•E•P. In his new position, Handke will work closely with Otari's national sales manager, James Goodman. ■

ENGINEERED TO EXCEED all expectations, Ampex 499 Grand Master® Gold delivers low print through the lowest noise, lowest distortion, and highest output of any high-performance analog mastering tape. Not to mention superior mechanical characteristics to ensure problem-free tape handling on today's high-end transports.

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Fresh Tracks

The Carpenters: "From The Top"

Label: A&M
Produced by: Richard Carpenter, Joe Osborn, Jack Daugherty, Phil Ramone (additional tracks produced by Karen Carpenter, Peter Knight, Rick Jarrard)
Engineered by: Ray Gerhardt, Joe Osborn, Charles Lowery, Dick Bogert, Eric Tomlinson, Jim Boyer, Glenn Berger, James Guthrie
Mixed by: Roger Young (Remixes)
Mastered by: Arnie Acosta
SPARS Code:ADD

Comments: Do you remember classic rock production from the late '60s and '70s? Songs with arrangements that actually built up slowly, culminating in a big payoff chorus? Arrangement is an often overlooked aspect of production. The Carpenters were among the best at this, because Richard Carpenter is an arranging genius, particularly on tracks from their first three albums.

At their best, the background vocal parts are lush, working very well with the lead vocals in classic call and response style. Carpenter masterfully uses non-rock instruments for shape and texture — the oboe intro in "For All We Know" and "Superstar," the harmonica in "Rainy Days and Mondays" — weaving them into beautiful orchestral arrangements. He uses rhythm and timing to build tension: The verse of "Superstar" opens with just bass, drums and vocals with a hint of Rhodes. The second verse adds subtle strings and a bit more Rhodes, until the chorus sweeps in with horns and full string section. In "Let Me Be The One," the horns hit the chorus with a clever syncopation, in the first measure starting on the downbeat but one measure later delaying expectation by an eighth note. In our opinion, Carpenter is as imaginative and skillful as the best arrangers of our era, Billy May, Nelson Riddle and Quincy Jones. It is a great loss that he isn't still working as an arranger.

Of special interest: What tone! Karen had one of the great voices in popular music. She fills up the spare parts of an arrangment effortlessly, and is able to consistently support full arrangments without getting buried or lost. This 4-CD set contains 20 unreleased titles (including demos — you've got to hear these!) and 40 remixes. Though "Sing" is a track we could do without (and the Spanish language version of it is equally uninteresting to us), the rest of the Carpenters' repertoire has aged well. Richard recently re-recorded some of his original keyboard parts to good, though subtle, effect. The remixes by Young add new life to the tracks. Sonically, this set is better than any of the previous compilations and hits packages. ■

Camerata Köln/Telemann: "Concertos for Woodwind Instruments"

Label: Deutsche Harmonia Mundi/BMG Classics
Produced by: Wolf Werth, DLF
Engineered by: Francoise Eckert

Comments: The DHM label is committed to recording classial repertoire using historical, or period, instruments. Though Telemann himself wrote that he was "never completely happy" with the concerti, the works performed here have experienced no lack of contemporary interest. A fine collection of works.

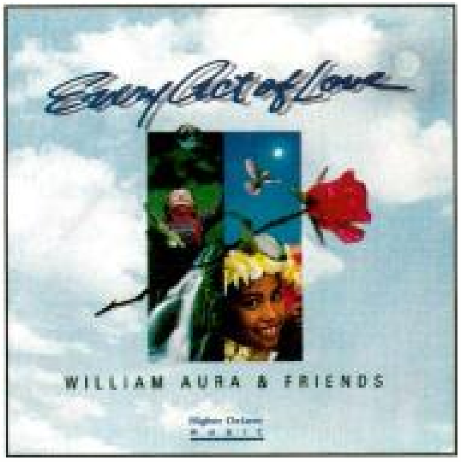
Of special interest: The largeness and power of the low end on this recording is impressive; the contrabass and low harpsichord notes fill up the bottom entirely. The tone of the bassoon in the closing track, "Concerto in D Major for bassoon, 2 violins, strings and basso continuo," is captured magnificently, a richly wooden sound with delicate tufts of air passing through.

Panning and placement in the stereo field in classical recordings is tricky: one wants to separate the instruments the right amount for clarity and spaciousness without sacrificing blend. This recording accomplishes just that. Through the use of reverb, individual instruments take up moderately wide spaces, evenly spread across the stereo field. ■

William Aura and Friends: "Every Act of Love"

Label: Higher Octave Music
Produced by: William Aura
Engineered by: Jeff Mahoney and William Aura
Mastered by: Joseph L. Steiner, III (F.D.S Labs)
SPARS Code: ADD

Comments: Aura plays guitar synthesizer, keyboards, percussion, zither harp and sings, with guest musicians providing sax, flute and percussion. The recording is rich in high frequencies but not at all piercing or thin, and the annoying synthesizer top end "whine" we often hear on synth recordings is miraculously absent here. Spacious ambiences are used effectively to create an expansive, larger-than-life fantasy soundfield with nice movement.



Of special interest: The compositions are built on solid grooves with excellent musicianship throughout. Aura succeeds where many new age or environmental artists do not: he establishes depth and mood in his compositions. Other recent Higher Octave releases include equally good production and engineering for artists such as Ottmar Liebert, Stephen Longfellow Fiske and Richard Buxton. Ottmar Liebert's Neuveau Flamenco is especially recommended. ■

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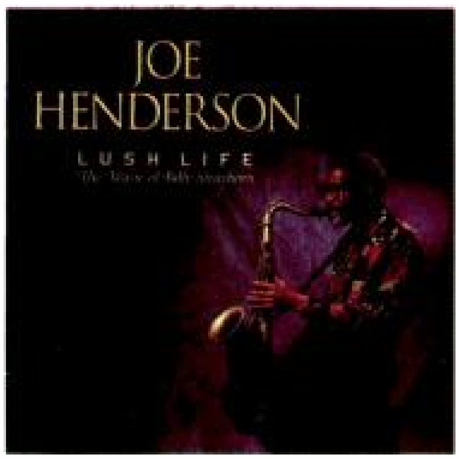
Fresh Tracks

Joe Henderson:
"Lush Life, The Music of
Billy Strayhorn"

Label: Verve
Produced by: Richard Seidel and Don Sickler
Engineered by: Rudy Van Gelder
Mixed by: Rudy Van Gelder
Recorded at: Van Gelder Studios
Mastered by: Rudy Van Gelder



Comments: One of the greatest living tenor players combines efforts with Wynton Marsalis (trumpet), Stephen Scott (piano), Christian McBride (bass) and Gregory Hutchinson (drums). Different tracks combine different combinations of the musicians. Henderson explains in the liner notes, "What I like about each one of them is that I could play with them in any formation and it would come off. I could play with one of them or two of them or three of them or all of them with Wynton and something musical was within the city limits. Nobody ended up somewhere without a map and no awareness of the language." From the sensitive bass/sax duet on "Isfahan" to the uptempo quintet on "Johnny Come Lately," the performances are stellar: it is an instant jazz classic, to be listened to and cited for years to come.

Of special interest: This production achieves a musical harmony not often found, even in musicians who have played together for many years. The album has the feel and inspiration of some of the best jazz albums of the late '50s and early '60s, a gripping artistry of rhythmic and melodic designs, interwoven into the simultaneous multiple improvizations of a small combo. Van Gelder's recording is faithful to the sound of his earliest recordings of jazz greats 35 years ago, but is ultimately modern in the clarity and warmth of the sound. The normally taciturn Van Gelder discusses this project on this page in his first ever published interview! ■

FOCUS:

RUDY VAN GELDER, Engineer, "Lush Life, The Music of Billy Strayhorn"

R•E•P: In your engineering, are you trying to create the sound a person in the room would hear or an artistic impression of the sound?
RVG: There's no one answer to that. It depends on the nature of the project. Certain projects call for just an attempt to duplicate the sound — my studio is oriented toward an ambience of its own, and sometimes I try very hard to capture that. It's not a place where you have to manufacture the sound of the finished product, it just happens on its own. It's not an average studio. It's hard for me to talk about it if you've never been here. When you walk in here it sounds like my records. There are situations where the producer comes in and wants to make a different sound, and then we do that in the way that we do it.

I'm in the middle of a project now where there's been weeks of overdubs. You're talking about what a person would actually hear — you can't get much farther from a real [listening] experience than overdubs. There was a review of Joe Henderson in *Rolling Stone* — they liked it because it was not heavily produced — they think. Little do they know.

R•E•P: How important is it to capture a mood in the studio?
RVG: Everything I do is related to creating the perfect mood for the musician — the location, the room, the sound of the room, the way I approach him on an individual basis, the results that I think the producer wants. It depends on the producer, who he is, and what he's trying to say. Everything is related to mood. For example, at one point on this project, I turned out all the lights in the studio — I mean all of them, in the control room and in the studio. All anybody could hear was Joe playing, I believe it was on "Lush Life." That's about as mood-oriented as you can get.

R•E•P: When you hear a player for the first time do you get in your head the kind of mic you're going to want to use, or do you instead set up several and listen to them, choosing the one that works best?
RVG: I'm not comfortable talking about that — it's too close to what I do for a living. Have you read Bruce Swedien's comment about recording jazz? He said "It's like taking dictation." Did you ever see that?

R•E•P: Yes, I did that interview (R•E•P November, 1990). How many records have you made?
RVG: Thousands.

R•E•P: All your records have a strong mood. Did they always have that?
RVG: I always tried for that — that's what I do. Going way back, the equipment was so limited, I couldn't get there. I often think if I had then what I have now, technically, it really would have been a more rewarding experience for all of us. When I hear those records now, I hear the things I could do better.

R•E•P: What records in particular?
RVG: Well, the Miles records, for instance, Lee Morgan ...

R•E•P: You must develop a personal relationship with some of the artists.
RVG: Some of them.

R•E•P: Does being the engineer mean you can't develop more personal relationships?
RVG: That brings up some things I'm sad about. The last time I recorded Art Blakey I felt I wasn't close enough to him because there was a session and they were paying me to record. It was a session — I had a job to do, and I would rather have spent some personal time with him. Remember, I see these people only while they're recording, I don't have contact with them other than that. ■

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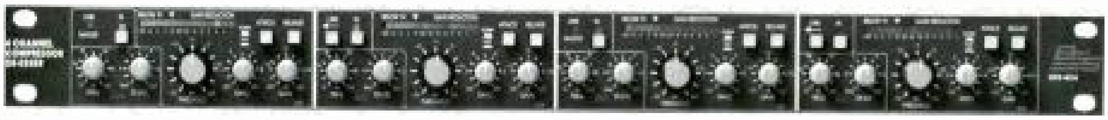

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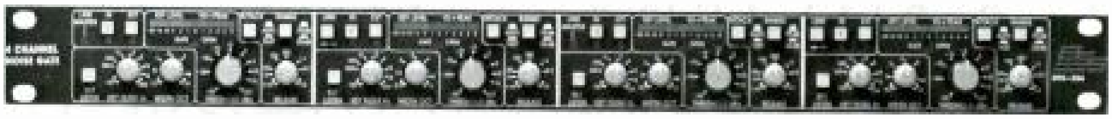

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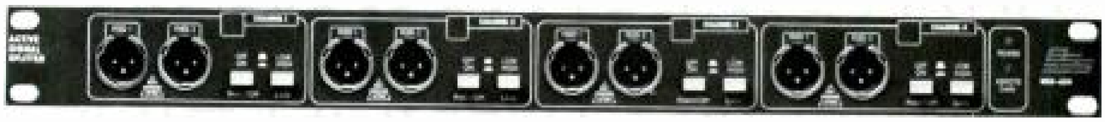

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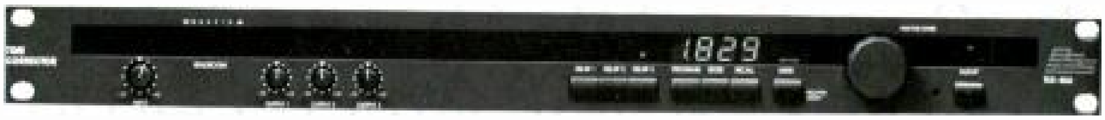

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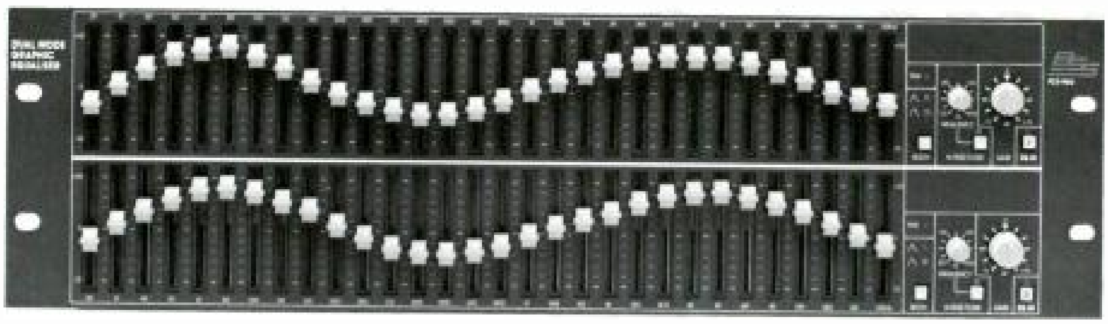

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Circle (9) on Rapid Facts Card

Fresh Tracks

Gino Vannelli: "Live in Montreal"
Label: Vie/BMG
Produced by: Gino Vannelli, Joe Vannelli
Engineered by: Harvey Robitaille
Mixed by: Joe Vannelli, Gino Vannelli
Recorded at: Blue Moon Studio
(Los Angeles)
Mastered by: Bernie Grundman

Comments: In two years of Fresh Tracks reviews, we have never reviewed a live rock album for one simple reason: most of them pale by comparison to their studio counterparts, and come off as messy and out of control. However, there is something about the sound of this particular record that intrigues us: it has that shimmery reverberant sound that one actually hears in a live performance, as the sound bounces around hard surfaced walls and ceilings; but the mixes are perfect. The band was heavily rehearsed for this recorded performance (see accompanying focus interview) and the results are here to hear.

There is tons of reverb — all the instruments are tremendously wet — but it's hard to tell where the hall ambiences end and the outboard mix effects begin. The balances between the top end instruments are just right. Personally, we could use some more stuff happening on the bottom end — the bottom on the kick drum and bass are nearly absent from the mix by today's standards — it sounds as if the recording cuts off at 100Hz, peculiar for an artist who has had so many dance-oriented hits. Still, this is one of the most interesting sounding live recordings in recent memory. The 13 songs here cover most of Vannelli's 10-album career, from 1973-1991, and include "Brother to Brother" and "I Just Wanna Stop." ■

FOCUS:

JOE VANNELLI, Co-Producer/Mixing Engineer, "Live in Montreal"

R•E•P: Did you record multitrack?
JV: Yeah, it was recorded on the truck from Le Studio Mobile — they use a Studer A80 multitrack and a Soundcraft 1624 console; some dbx 165s, 160s, LA4s, 1176s and some Klark-Teknik gates. They used Dolby A on the first 16 channels, and the rest of the channels were non-Dolby.

Our live set up has a lot of MIDI-stuff going on live. I've got two 24-space racks of MIDI-gear fed through two DMP11 digital mixers, so they [the mobile truck] basically have a stereo feed of all the keyboards.

R•E•P: If you had wanted to change the balance between keyboards at mixdown, would you have synced them up again?
JV: We would have been able to if we wanted to, but we didn't need to. The whole show was pre-produced in our studio here in LA and we tried to get the balances as close as possible.

R•E•P: Were the electric guitars recorded direct or miked?
JV: Both. The drums were miked acoustically. We used SM81s as overheads, AKG 460s on the hat and bottom of the snare, a 57 on the top of the snare, some Sennheiser 409s on the toms and an RE20 on the bass drum. The drummer, Enzo Todesco, was triggering some MIDI-samples from my Akai MPC60 for some low tom sounds. We had a chessy mic on one of the toms and fed it through the Akai ME35T Audio-to-MIDI converter.

R•E•P: What kind of vocal mics did you use?
JV: Maxayne Lewis, the backing vocalist, and Gino both used Sure Beta 58s. I also did some backing vocs — I had a Yamaha headset — and the drummer and I were playing to a click. I think people will find that amazing because he's taking some pretty wicked solos, but he's doing that in time to the click.

R•E•P: How did you get such a live sound when you mixed, while still keeping so much control?
JV: The keys were premixed so they had that presence. The guitars and drums were all well recorded. We had two audience mics that had all of the ambience, and we had to sort of ride those things during the mixing. And in order to maintain the ambient sound, we added more reverb and delay and stuff. I have a 480L, two LXP15s, two SPX90IIs, a REV7 and a Quadraverb. On the 480 we used sort of a room sound and we added the bright gate sound on the LXP15 to the drums. Actually, that's what we were using live at the gig, but we didn't record that at that time — we added it back again in the mixing process. We tried to use as many of the effects at mix time that we had used live. We hadn't recorded the effects because all of the mics were split at the source. What we're really happy about with the mobile stuidio is that they got everything on tape really clean, minus distortion.

R•E•P: What do you have in your mixing studio?
JV: We have an Amek 36-input Angela. I don't have an analog multitrack anymore, so I rented an Otari 24-track with the Dolbys, and then dumped it to the 24-track Akai ADAM system. Originally this wasn't supposed to be a live album, it was just going to be the audio for a video we did live up in Canada. After we started hearing it, we realized we had a good thing. When we started going through all the tracks, we realized everything was all pretty much there.

We did some additional songs mixed on Otari 1/2-inch SR 4-track with time code. I transferred the whole thing to digital using the dbx700 system and I used the old VCR editing method and pieced it all together. So the album wound up on the dbx700 and then we mastered it at Bernie Grundman's. Gino and I have worked with Bernie for years, way back when we were both at A&M. ■

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THE R•E•P INTERVIEW:

Out front insight from live rock's high end.

Robert Scovill

MAJOR

MAINS MAN

By Anthony McLean

The list of engineers employed for major act FOH arena mixing is very short. Robert Scovill is on that short list. Regularly employed by world-class artists, Scovill has piloted the mega-systems from Electrotec, Showco, Clair Brothers and more.

Over the past decade he has parlayed experience as a musician, studio and live engineer into a high profile, globe trotting career. Scovill comes across like a renaissance man. His success seems to spring from his ability to fuse organization and management with musical aesthetics. At one moment it seems as if he could be working for NASA, the next he is racing along at full throttle rock 'n' roll.

In a break during the 1992 Rush tour, we queried Scovill about the technology and artistic components of his work. From his California base, called Musicanvas, the articulate and engaging Scovill recounted the beginning-to-end details of mixing in the major leagues.

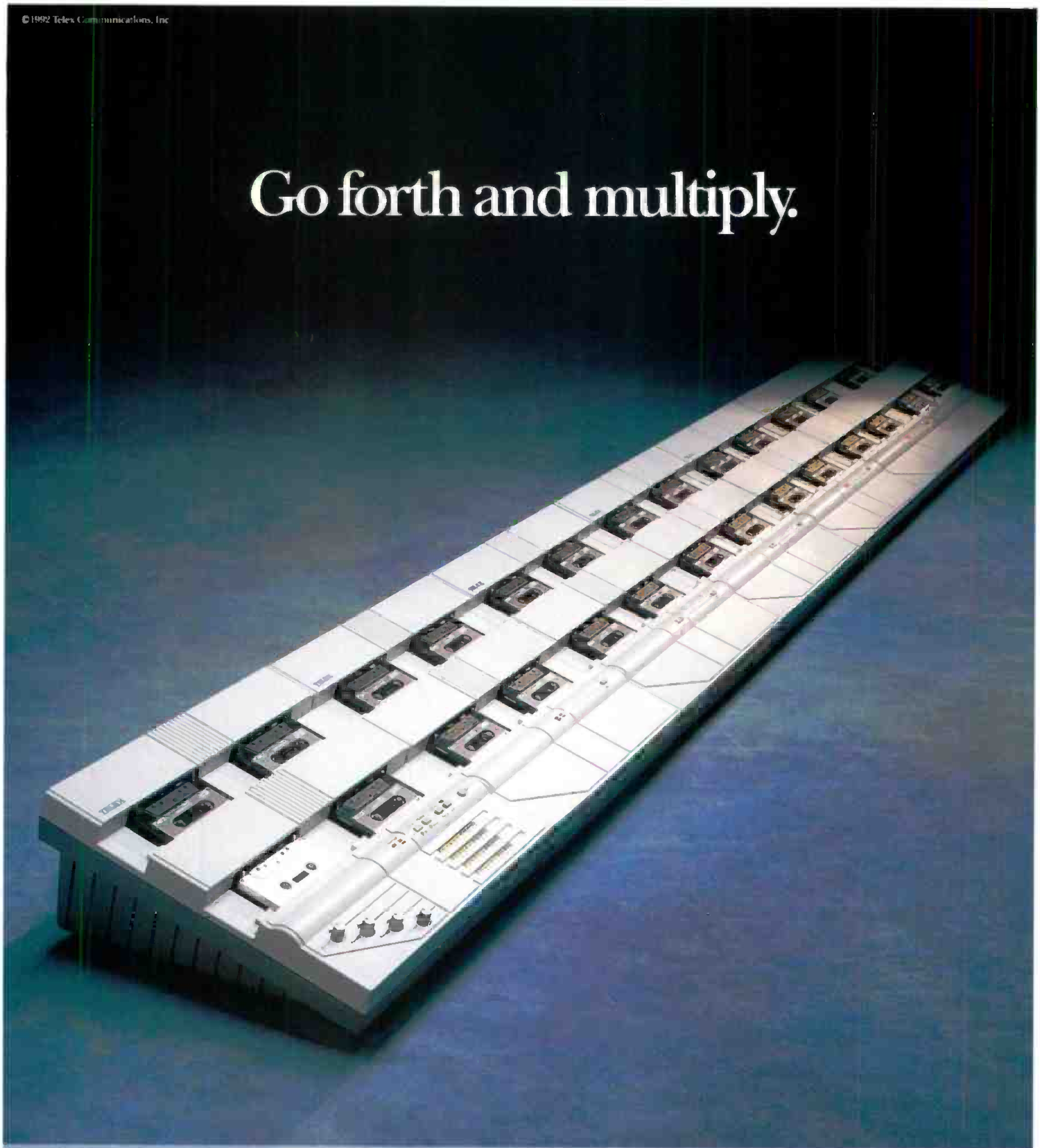
R•E•P: What's your background? How did you get to drive the big rigs?

RS: I remember hearing an interview with Alan Parsons on the radio in St. Louis talking about the roll of a producer and an engineer in the studio. I was hooked from that moment.

After playing in multiple bands, I enrolled in college at the age of 17, shooting for a bachelors in Electronic Engineering. Around 1980 I was working every angle around Kansas City; in a studio recording friends' bands, at a local sound company doing concerts and helping in their service department. Because I continued working as a player, I developed a pretty balanced set of skills.

While working in the studio, a band named Shooting Star cut some demos and said when they got ready to go on the road they would need an engineer. I held them to their word. In the following months they were the opening act on tours such as ZZ Top, Cheap Trick,

Anthony McLean is features editor of R•E•P.



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Jefferson Starship, etc., so consequently, I got to work with a lot of sound companies and made the contacts.

R•E•P: Describe your pre-tour and pre-show preparation.
RS: I spend an enormous amount of time becoming familiar with the music of any artist that I'm going to work with. I don't mean that I memorize names of the songs and the band members and what year they recorded what. I dissect their records to understand their approach to sounds and mixing, because what the audience wants to hear is an accurate electro-acoustic interpretation of that music. As mixers, live or in the studio, we become a buffer between the artist and the listener.

When possible I get involved constructing mixes at the rehearsal stage. At this point I won't use the PA, I'll set up in an isolated room and use a pair of monitors that I'm very familiar with and approach it as if I was mixing a live 2-track. That gives me time to work with the players, getting their sounds together, experimenting with different mics, etc. Then it's a matter of rehearsing all the preset moves that I need to do on the console. The final step is making sure that all the sounds and the mixes themselves translate accurately through the PA and throughout the room. I believe pre-show preparation, on a day-to-day basis, is the most challenging and important part of the live sound business.

I developed my philosophies on pre-tour/pre-show preparations through trial and error. I use what I believe to be applied common sense. Learning how to handle these situations is how you develop an identity as an engineer. You could talk to 10 engineers and get 15 different answers how things should be handled. Whatever works, ya' know?

R•E•P: In terms of bringing the whole house up to a unity gain level, what is your standard operating procedure for balancing and equalizing various speaker system components, such as sidefills, downfills, etc.
RS: First we make sure that the array is laid out as properly as we can for the given venue. If you are on an arena-only tour, this is a bit easier because there are fewer day-to-day changes. Next we pink noise and EQ the different fills, front, side, etc. We then integrate and pink noise the entire PA in a number of different locations to make sure we are getting the desired response, trying to remain aware of the ambience that each one of those components create. Some of those components may even be shut off for sound check to eliminate any destructive ambience that will eventually be absorbed by the audience. We have to stay on top of these electro-acoustic variables every-day.

R•E•P: Which measuring devices do you use for referencing the electro-acoustic environment?
RS: Right now we use a Sound Technologies RTA 4000 which gives us pink noise analysis and comparison measurements between the output of the desk and the output of the loudspeaker system. It also allows us to store our daily analyses, which we then integrate with arena seating charts and analyzer mic positions.

I am very excited about recent technologies, such as computer measuring tools and integrating control platforms like the NEXUS amplifier control system, to assist us in adjusting the system array volumes. I have a particular interest in t.c. electronic's EQ talk.

R•E•P: It is sound check time, everything is on-line.

What rock-crazy teenager wants to sit at home with Mom & Dad watching what some artsy director wants you to watch?

Lead us through the process of building a Robert Scovill mix. We're especially interested in how you approach sub groups, the whole VCA issue and dynamic compression differently.
RS: I'll describe two scenarios here. One is that you are soundchecking under a full production situation in which you had never worked with the PA or mixed the band before. The second scenario is soundchecking a band that you have been in production rehearsals with for some time and it is your first day with the mix running through the PA.

These two techniques strive to achieve the same result. I will try to explain the differences as they apply to my approaches. The first scenario uses the age old analogy of building a house. You need a good foundation for the rest of it to be able to support its own weight. I immediately become as familiar as possible with all the PA components and establish what the PA is capable of doing before listening to instruments through it. I also attempt to listen to as many of the instruments as possible before they ever hit a microphone. Only when the artist and I are satisfied that the instruments are providing the sound that we want, is it time to mic up.

To establish a constant, especially if I am not familiar with the PA's response, I often perform an entire line check — including EQ, blending and effecting — in my personal headphones. I know how things should sound in there. What's left is to concentrate on translating those sounds accurately via the PA into the venue's acoustic environment.

My goal is to make the console mix constant and avoid constantly changing input EQ. I make the same mixing moves every night just as if I was a mix computer, mixing tracks from a tape machine. That way the only variable to be dealt with on a given day is to make the PA sound the same as it did yesterday. That is a lot easier and a lot more fundamentally sound than re-EQing 50 or 60 inputs every day and leaving the PA EQ flat. What's better: 50 or 60 variables or 1?

This approach applies to the second scenario, but at rehearsals, not on show day. At rehearsals I do all my mixes on NS-10s because they are a familiar source. That way all I have to concentrate on is making sure those mixes translate accurately through the PA.

As far as soundcheck on a day well into the tour, for me it involves making sure all input sounds are consistent day-to-day and tuning the PA to make it sound just like it did yesterday.

As far as grouping is concerned, there is no rocket science here; vocals, backing vocals, bass, guitars, drums, keys — whatever applies to the act. As far as VCAs are concerned I do enjoy mixing with them. They are a definite asset, but they are not resident to my current console of choice.

R•E•P:: Talk to us about consoles ...
RS: For me a console has to do three fundamental things well. It must have a great-sounding input section, an accurate, musical EQ and an accurate high resolution metering system. A console with those three basic characteristics will always earn my loyalty over any console with all the "features." A console's sound should be its "feature."

My philosophy is simple. I treat the PA as an infinitely dynamic source. If dynamics control is necessary I do it on the input side. With mix compression I do it prior to the master fader. And at the "wish list" level, I would like to see live console manufacturers head toward some type of "standard" total recall system for live consoles, or at the very least implement some MIDI control for programmable aux bus levels and mutes.

R•E•P: The difference in trap set topology between Def Lepp and Rush seem to be extreme. Could you discuss the contrast, the desired effect, and your management of both? For example, does (Rush drummer) Neil Pert's sonic intensity drastically impact your miking, gating, etc.?
RS: This difference is actually based in the differences between the recording techniques of these two acts translated to live performance. Def Leppards' drum sounds are built around current sampling technology which, once sorted out, can be a fantastic constant in a live or recorded mix. However, it's the hardest I've ever had to work, actually creating drum sounds, because the virtually limitless choice of sounds is not necessarily an asset. Also, all the Lepp sounds you've heard on the past two records have been created at the album mix-

ing stage, therefore they are not available for sampling at a later date (i.e. they are not located on a multitrack).

Conversely, Neil's live drum sounds are synonymous with recorded Rush drum sounds, because virtually the same techniques used to reproduce the kit live are used to record the kit. The difference is that live, the mics are selected for a very controlled pickup pattern to limit the ambience into the mics, because it is generally destructive, due to bleed from other instruments, etc. In the studio, the environment surrounding the kit has become as much a part of the sound as the kit itself. Live, I attempt to recreate those environments with various effects units.

As far as the sonic impact goes, there definitely is a difference in the feel of the show. With Neil you get a sense of the performance happening right now. His sound is also much more subjective to the showtime environment.

R•E•P: Speaking of showtime environment, what about arena acoustics?

RS: In general I am not bothered by arena acoustics. I quite enjoy mixing in a very reverberant space, more so than in an overly dry space or relatively small room. Album production over the last 10-12 years has lent itself very well to mixing in big arenas. Those productions have been very ambient so the shows generally fit in well in that environment.

Arena acoustics for the most part have not changed since they started building arenas. I recall walking into a brand new building within the last two years. I had to list it in my top three worst-sounding places on the planet. Let's face it, for the most part arenas are built with a sports-minded philosophy: "Let's make 1,000 people sound like 10,000," home field advantage and all that stuff. The way you do that is to make the place very reverberant.

The primary thing that has changed is our approach to how we handle places with reverb times longer than most of us can hold our breath. People like Don and Carolyn Davis and their Synergetic Audio Concepts seminars have defined many useful techniques concerning such environments. Electro-acoustic goals such as getting the ratio of direct-to-reflected sound as high as possible for as many people as possible is critical in getting the arrays properly sorted out. And the advent of the Techron TEF analyzer has produced quantum leaps in cabinet and array design. After the PA companies and performers came to understand what was required to cover arenas properly — and we're not there yet, folks! — the next step was to convince everyone that, "Yes we really do need to array 50,000 pounds of speakers."

But even this is not the total solution. As far as venue acoustics are concerned, you could say we are putting the cart before the horse by trying to sort out the systems before we sort out the environments they will be used in. But I think it is the prudent choice, because if we wait on the venues to sort out their own acoustics to accommodate the large electro-acoustical systems, we would have a very long wait.

R•E•P: You've used the word "large" a lot. One body of thought exists that believes the touring industry must downscale to remain viable? How about it?

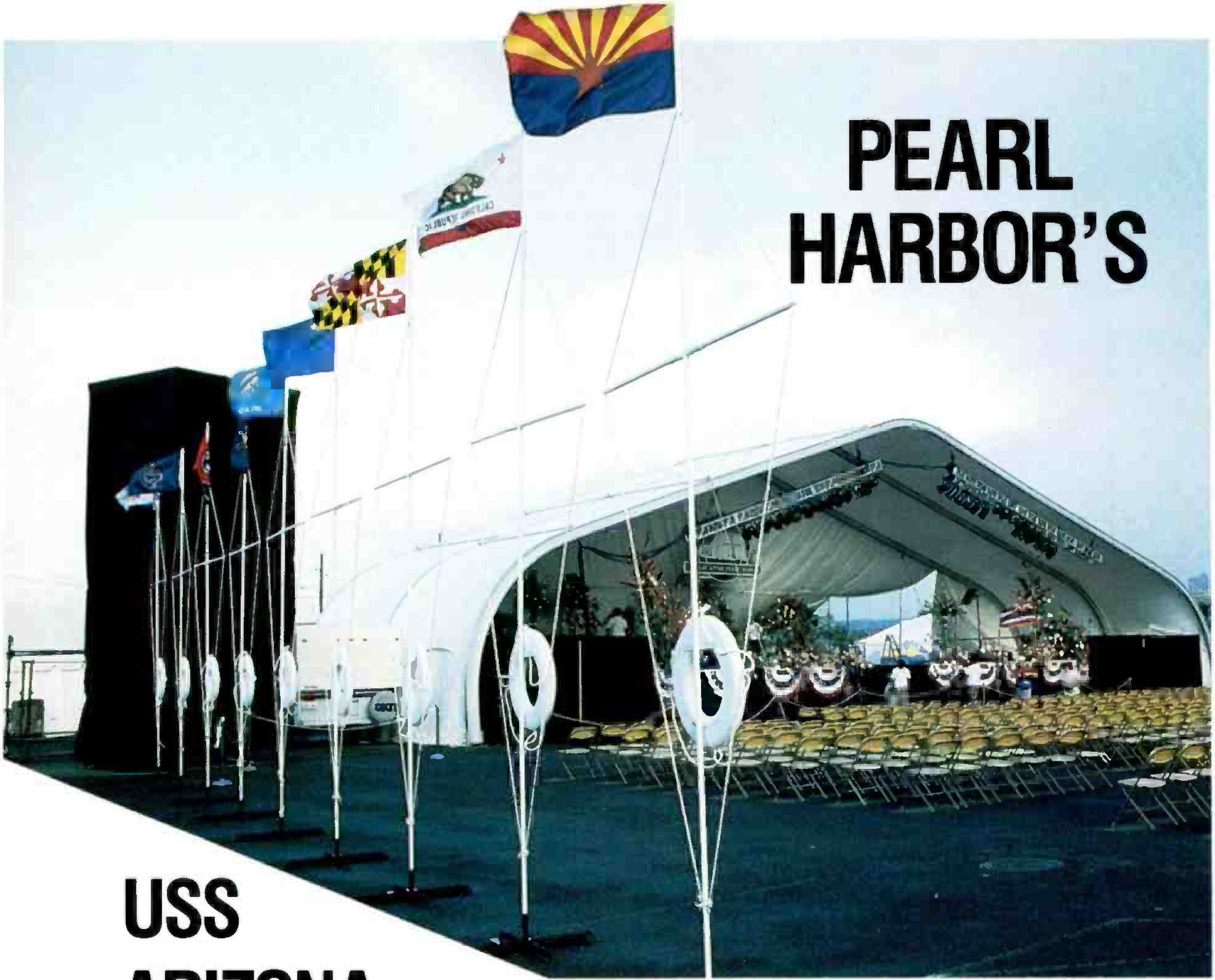
RS: I feel the whole idea is a bit cyclical in nature. We Americans always tend to want what we do not have. So, if big is in, then it is time to go small. Another Americanism is, "Anything worth doing is worth overdoing."

R•E•P: Let's look into the future even more. Will the big name, large scale concerts eventually play only NY/LA and the shed circuit, with the rest of North America serviced by pay-per-view?

RS: I hope not! Pay-per-view is something that I think will surely exist, but I do not think it will totally replace concerts for a long long time, if ever. What rock-crazy teenager wants to sit at home with Mom and Dad, watching what some artsy director wants you to watch? What's next, "stay tuned for the home shopping network concert merchandising show?"

Concerts are an escape for people, and if the only option is to go to the local night club to watch a pay-per-view concert of music that touches your soul, this industry will cease to exist, because there will be no interaction between the audience and the artists, and consequently neither will want to do it.

Who wants to show their appreciation of someone's art by shaking their fist and shouting "Right on! Crank it dude!" to an image on a big screen? It could hardly be considered being a part of the event. If you are at a concert today, it happens, sometimes good, sometimes bad, but you are there. You and 17,999 other people were the only ones to experience it as it just went down. It has its own set of finger prints and will never be exactly the same on any other night of the tour. ■



PEARL HARBOR'S USS ARIZONA MEMORIAL CELEBRATION

By David Scheirman

On December 7, 1941 the Naval forces of Imperial Japan attacked the United States Naval base at Pearl Harbor, home of America's Pacific Fleet. This event ushered the U.S. into the second World War. One-half century later, the National Park Service and the U.S. Navy collaborated to stage a special memorial event to commemorate that incident, and to honor servicemen and civilians killed and injured in the action.

This special event, staged at several different locations on the Pearl Harbor naval base, Dec. 4-7, 1991, featured everything from battleship crew survivors, hula dancers, notable Hawaii residents such as James Michener, and the Pacific Fleet Band, to President George Bush, 13 different United States governors, and the Honolulu Symphony. Sound reinforcement requirements were both technically and logistically challenging. This is an overview of the live sound aspects of that event.

David Scheirman is R•E•P's live sound consulting editor and president of Concert Sound Consultants, Julian, CA.

The National Park Service chose Baus Engineering of Honolulu to handle the technical production of the program, which was held at the USS Arizona Memorial. It was set up on a temporary stage positioned in an existing naval parking lot. Company owner Randy Bauske was responsible for the sound system design along with a myriad of details including crew contracting, scaffolding and stage rentals, stage lighting and site communications.

"This event was an interesting mix of budget considerations, cooperation between different government agencies and careful sound system design," said Bauske, whose company supplied both equipment and crew for the event's sound reinforcement needs. "We had to weigh many different factors and consider different perspectives when deciding how to approach the project. For example, who had to authorize the placement of a delay tower in a different part of the parking lot? Who would authorize media credentials for site and access to the audio program? It was a challenging job, and everyone involved felt that the production aspects turned out extremely well, despite weather, changing programs, and a tight schedule."

SOUND SYSTEM FORMAT

The sound system was required to handle speech, live music programs of assorted types, and music playback for an outdoor audience of nearly 4,000, many of whom would be retirees. A major component of the program was the reunion of Naval officers and enlisted men, some of whom had not seen their shipmates since the bombing of Pearl Harbor in 1941. Many of these men were 70-80-years-old. Another important consideration was media access to the event. Portions of the program would be carried by major networks and cable channels in the U.S. and in other nations.

The primary audience area was laid out in a rectangular format that offered maximum seating provisions for the available space, and that could be given full-bandwidth coverage with a minimum number of speaker arrays (See Figure 1).

A SPECIAL 50TH ANNIVERSARY

Four scaffold towers were erected to house the loudspeaker systems. Each tower suspended an array of four Meyer MSL-3 enclosures on a deck 12 feet above ground level, with three Meyer USW-1 subwoofers positioned on the lower level. Amplifier racks were located at each tower (including Mark Levinson ML-3 and Bryston 7-B power amps). The upper cabinets were grouped in a tight array and given a 10° downtilt. The entire tower was then encased with black theatrical cloth, and a heavy-duty tarp was placed atop the MSL-3 enclosures to act as a rainguard and sunshield.

The main left/right speaker towers were placed nearly 120 feet apart, in order to cover the widespread audience area. To fill in the center forward area, which included V.I.P. seating, a complement of four Meyer UPM compact enclosures was attached to the front edge of the staging platform and integrated into the "flags and flowers" stage design motif. These small speaker systems were signal-delayed to be "in sync" with the full-range sound as heard from the main left and right speaker towers.

The heart of the sound system control area was a 32-input, 12-output console. Manufactured by ATL (Acoustic Technical Laboratories) in Japan, the console offered full parametric equalization on all input channels and was an ideal choice for handling both FOH program signal distribution and stage monitor mixing in one package. A typical foldback mix signal path comprised a discrete mix out from the ATL console, a bus-inserted 1/3-octave T.E.Q. graphic equalizer, a Meyer M-1A control electronics unit, a Yamaha M-85 dual-channel power amplifier (channel A lows, channel B highs), and a Meyer UM-1 wedge-type monitor speaker system (See Figure 2).

A hand-picked group of signal-processing devices, including Aphex and dbx limiters and noisegates, Lexicon, t.c. Electronic and Yamaha digital reverbs and delays, and EQ units from IRP, SAE and Meyer were available at the mix position.

The FOH equipment racks contained not only all digital delay and equalization gear needed for the distribution of sound to main arrays, delay towers, centerfill system, press feeds and stage monitors, but also a control rack for a special contingency "overflow site" sound system, set up at the rear of the audience area and focused on the hilltop across the street from the event site. This system, comprising an available group of 10 UPA-1A speaker enclosures suspended in line-array format, was positioned nearly 200 feet behind the house mix riser with Genie-type hoists (See Figure 3).

An HME-PAL wireless audio link was used to carry signal to the overflow system. A Yamaha D-1500 digital delay and Industrial Research TEQ equalizer fed the HME transmitter located at the house mix position, and the receiver was fixed to the 24-feet-high remote loudspeaker tower. Meyer M-1A control electronics units and Yamaha M-85 dual-channel power amplifiers were used to drive the system.

PROGRAM MATERIAL

The 4-day Pearl Harbor event featured a diverse list of public speakers, military bands and choruses. A highlight of the program was the world premiere of a work specially commissioned by the National Park Service in com-

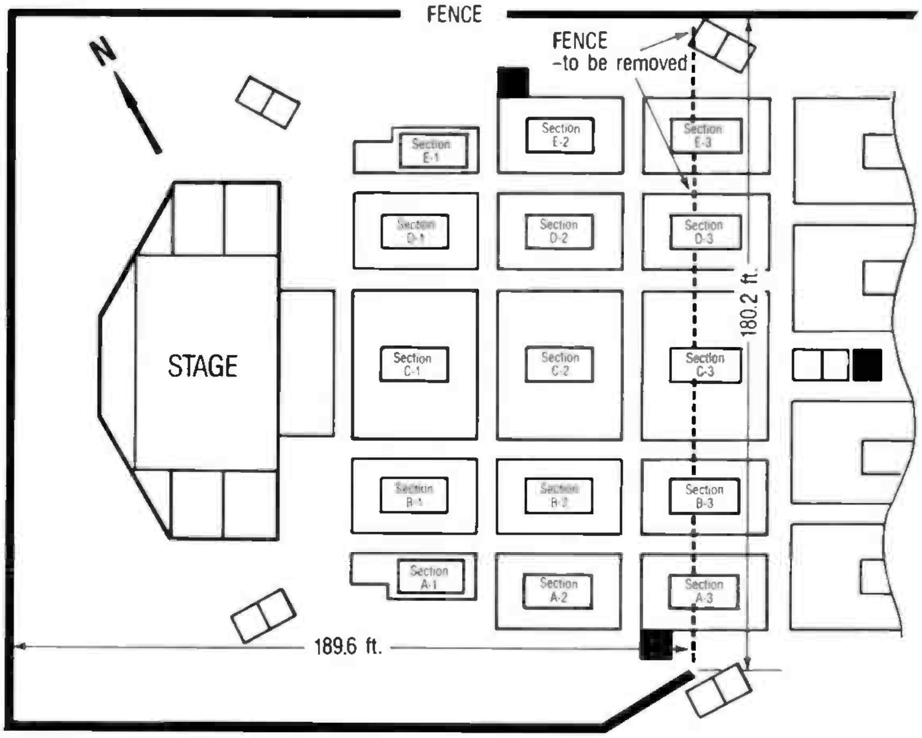


Figure 1. Audience area indicating full-bandwidth speaker arrays.

A closeup of the house left main loudspeaker tower.

A major component of the program was the reunion of Naval officers and enlisted men, some of whom had not seen their shipmates since the bombing of Pearl Harbor in 1941.

memoration of the 50th Anniversary of the Pearl Harbor attack. "Time for Remembrance" was conceived by the American composer Paul Duffy, and was performed by the Honolulu Symphony Orchestra under the direction of Donald Johanos. Senator Daniel Inouye of Hawaii recited a dramatic narrative.

The symphony was mixed by Randy Bauske on a separate console (a 32-input Harrison HM-5) and then routed into the ATL event board for which the author was responsible. This allowed a full pre-show soundcheck for the Honolulu Symphony prior to the beginning of the day's program, while enabling the carefully adjusted site sound reinforcement distribution (including foldback mixes, delay towers, overflow system, and press feeds) to remain constant by using the various outputs of the ATL desk.

Since much of the musical program material featured large ensembles (including both vocal choruses and brass bands), an area-mic strategy was adopted. Bruel & Kjaer 4000 series condenser mics with robust windscreens were used, mounted on tall boom stands weighted with sandbags.

"Microphone choice and positioning can be critical when you have to pick up a 75-voice choir or a 30-chair military band," said Baus Engineering technician Sonny Ah Puck, in charge of stage microphone placement and line patching for the Pearl Harbor event. "We use a complement of precision mics, including devices from B & K, Sennheiser, AKG and other quality-conscious manufacturers. We generally try to choose the right mic for the situation, make sure that the placement is correct and that the unit is not going to fall over or pick up wind noise. And then I let the house engineer call for adjustments as needed."

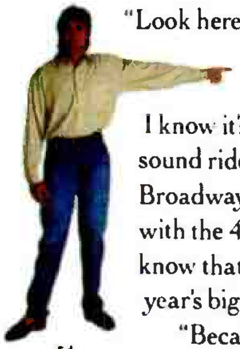
A shock-mounted pair of B & K model 4011 condenser mics with large windscreens was used for the speakers' podium. While perhaps not the favorite choice of TV directors or video producers due to its bulk, this system gave flawless audio performance with plenty of gain-before-feedback achieved through the use of channel-inserted narrow-band Meyer CP-10 parametric equalizers. One microphone could always be on line; a secondary unit was a spare. Either the upper or lower mic could be used based on the speaking person's height.

For 'problem' talkers who drifted away from the mics, the activation of the second mic with reversed line polarity at the mixer and the quick push-button channel insertion of an additional parametric EQ unit gave a significant gain advantage, while preserving the required audio signal.

ENVIRONMENTAL CONSIDERATIONS

Hawaii is known for its sunshine and its ocean-borne storms. In a single day, a sound system that is set up at an outdoor site such as Pearl Harbor can be exposed to intense, direct sunlight and to gale-force wind and rain. The Baus Engineering crew keeps a 'protection' case for outdoor sound jobs, and it includes quantities of military-spec tarps and shock-cords to hold them in place. These were used

Compact Meyer UPM speaker systems were used as center-fills across the front of the 60-foot-wide stage.



Ed
Sound Engineer

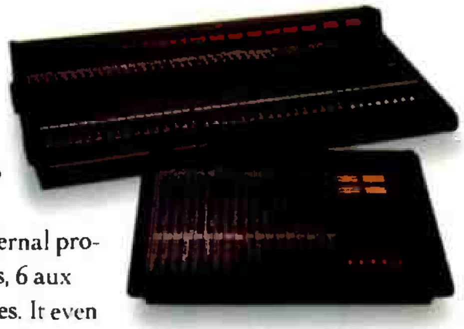
"Look here, I know the PM3000.

"I know it's at the top of the list of the best live sound-reinforcement consoles. I know it's written into all those big concert tour sound riders. I know it's in the major theaters on Broadway. I know it's in the 5,000-seat churches with the 400-seat choirs. And I also happen to know that it's in all those T.V. trucks producing this year's biggest sporting events. And I know why.

"Because the PM3000 is flexible. Because it's logically put together. Because it performs. Because it's a pleasure to use. Because everyone likes working with it.

"But, here's the news.

"There are two more PM series consoles. And they start at a mere $5,500 MSRP. So obviously, they're for those situations where you want the best console available. But you don't have the space or the budget to get the 3000.



"The PM1800A was just updated. So it has an improved signal-to-noise ratio (6 dB better). And 0dB insert points for easy gain matching with external processors. It's got 8 groups, 6 aux sends and 4 mix matrices. It even has the same mute grouping feature you find on the 3000. But that's not the end of it.

"The PM1200 has the same roots. But in a more compact format. It's got 4 groups plus stereo, 4 aux buses, and 4 mute groups. You can get 16, 24, or 32 input channels and you still get two additional full-function stereo input channels.

"Obviously, they're both ripoffs of the Yamaha PM3000."

YAMAHA®

"Obviously, they're both ripoffs of the Yamaha PM3000."

Circle (13) on Rapid Facts Card

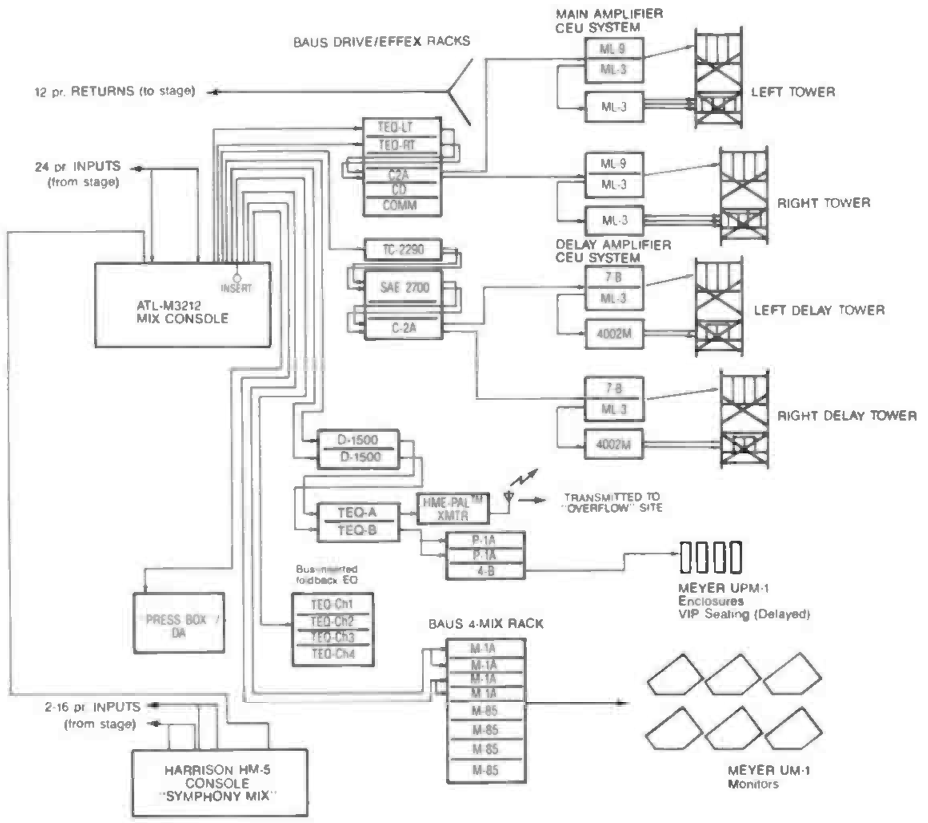


Figure 2. Diagram showing sound system signal flow.

to shield amp racks, speaker locations and the system control position.

A stretched-skin, aluminum frame tented structure was erected for covering the primary stage area, and its graceful lines were used to frame the event space. Lighting instruments were suspended from its support girders, and a 40′ ×60′ sectional stage was provided. The tent structure was over-sized to provide a sheltered space on each side of the stage for microphone stand assembly, cable and monitor storage.

Much of the Baus equipment inventory consists of custom-engineered items. Electronics rack packages, cabling systems and power distribution are all carefully configured to offer ease of setup, compact size and weight for inter-island travel, and rugged design for long-term use in the field.

As an example, the heart of any typical three-phase power distribution system is its circuit-breaker and feeder inter-connect panel. Baus Engineering distributes compact 'spiders' (satellite power centers with circuit-breakers and twist-lock connecters) and road-cased panels wherever the power is actually used, such as near amp racks and console positions. At the mix position, a hinged road case, only 12 inches deep, opens to reveal shock-mounted meters, voltage regulators, and heavy twist-lock and edison-type power cable connecters. Rubber castors keep the panel vertically oriented and up out of potential water puddles. A clean, rugged ac power center is created that takes up minimum space and offers environmental protection.

For the Pearl Harbor event, Baus Engineering coordinated a 10′ ×10′ tent and sunshade for the press audio pool. Aphex distribution amps were rack-mounted and fed with a common input from the primary mix position. A total of 24 gain-adjustable feeds were available, on a first-come, first-served basis.

These feeds ended up going to everything from the Pacific Fleet Band's archival Nagra recorder to European and Japanese film crew SMPTE-locked 16mm-linked DAT units, to globally-distributed broadcast and cable TV networks. As a courtesy to broadcasters, network directors and program producers, we signal-delayed this program feed so that it was in 'time-sync' with the live event sound being captured by roving media crews with their own mics in front of the stage. The signal was pre-equalized with high-pass and low-pass filters to prevent the pass-through of subsonics and ultrasonics.

SHOWTIME

When it was time for the proceedings to open after two days of system installation and prep work, the United States Navy Pacific Fleet Band started out with renditions of music from the 1940s, and selections from big bands such as the Tommy Dorsey Orchestra and the Glenn Miller Orchestra were played ... some of the same songs featured on December 6,1941 during the Navy Band Competition, a few short hours before the attack began on Pearl Harbor. Mixing "In The Mood" over a sophisticated, full-bandwidth system to an audience of

The FOH mix position featured a 32×12 ATL console, used to handle program bus assignments for both stage monitors and main systems. The (4) Industrial Research Products T.E.Q. graphic equalizers in the foreground were assigned to onstage foldback mixes.

Sound company owner Randy Bauske at his Harrison HM-5 mixing console during soundcheck for the Honolulu Symphony.

veterans who had danced to the same song 50 years before was quite an experience.

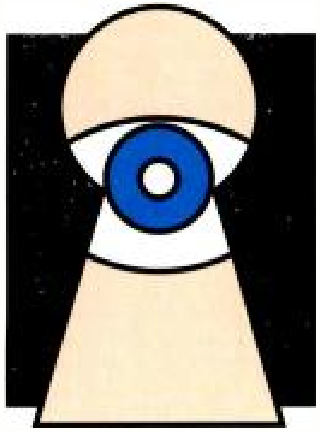
For the next four days, audiences of all ages were treated to Hawaiian culture, symphonic music, memorial speeches by public figures, military honor salutes and a broad range of program material. Despite occasional adverse tropical weather, last-minute program changes and different types of audiences, the Baus Engineering system performed up to its design specifications, in my opinion. The rig was easy to use and the implementations of certain required technologies (central-point console for all-event mix distribution, signal delay for distributed systems, high-gain podium mic system, musical group area mic systems and RF-linked remote delay system) were a pleasure to deal with. The 50th Anniversary program, "Pearl Harbor Remembered," enjoyed a straightforward example of proper, temporary sound reinforcement system design and implementation for a special event. ■

PHOTO BY RANDY BAUSKE

A program scene: The podium is in use by a dignitary and a children's chorus stands by to perform. Bruel & Kjaer 4000 series professional microphones are in use for both applications.

Author's note: The USS Arizona Memorial at Pearl Harbor, site of the sunken battleship by that name, is the final resting place for 945 servicemen (as estimated by the USS Arizona Reunion Association) who went down with their ship and are entombed there. This 50th Anniversary event, sponsored by the National Park Service, which administers the site, was the first time that many of the survivors of that crew had returned to the scene of the attack since WWII.

Note: The mention of specific brands of manufactured audio equipment in this article is not to be taken as an endorsement by the author, R•E•P magazine or Intertec Publishing Corp.



Part One: Avoiding RFI in Multi-Channel Wireless Microphone Systems.

SECRET KNOWLEDGE OF THE ETHER

By Bruce Jones

"Rikko the Dynamo" walks out on stage at a concert in Los Angeles. He scans a serious look across the audience and says, "It's really goozzz ffft sssshhht to beeeezzz ffft." The audience thinks, of course, that they just heard some sort of special effect. Rikko now scans a serious look at the sound crew. The sound crew begins to think about writing resumes to land jobs in a studio somewhere. The tour manager fumbles to find the phone number for his agent, and the agent scurries to locate his lawyer.**

What just happened to Rikko and his crew may well have been a dreaded "interference" problem with the wireless mic system. No matter how long you've been in the sound business, RFI (radio frequency interference) can sneak up and bite you at any time. In light of the fact that the sources of RFI are many and hard to pinpoint, RF interference is an unpredictable phenomenon, at best. As a matter of priorities, it's less important what the audio processing in the wireless sounds like if the RF system is failing and you have a bunch of noise!

With respect to wireless mic systems, interference is generally defined as an undesired RF signal that causes noise or distortion, or a momentary loss of the audio signal. It can also cause limited operating range and/or drop outs. Interference can result from external sources such as television station broadcasts, business radio services, or it can be generated by IM (intermodulation) and crosstalk within a multi-channel wireless system itself. The fact is that RFI can also result from some combination of all of these sources, making it difficult to predict. The primary focus of this article is to define the most common sources of interference in a multi-channel wireless mic system and offer practical suggestions of how to minimize its ill-effects. A comprehensive guide to the design and implementation of wireless microphone systems is available from Lectrosonics, Inc., authored by the engineering and marketing departments. An additional reference booklet discussing wireless microphone systems and sources of interference from external devices is available from Vega.

Bruce Jones is vice-president of marketing for Lectrosonics, Inc., a manufacturer and supplier to the broadcast and motion picture production industries.

Multi-channel systems always require higher performance components than simple 1- or 2-channel systems. Here are the reasons:

1. Interference from external sources is a problem for any wireless mic system, whether a single or a multi-channel configuration. In a multiple receiver system there are just that many more possibilities for external RFI.

2. In addition to external RFI problems there are "in-system" RFI problems that are generated by the multiple receivers and transmitters themselves. These in-system RFI problems can easily be more numerous and harder to cure than the external RFI problems.

3. In addition, external sources can combine with the normal RF signals in the systems to create even more problems.

It is possible to side-step a number of problems by spacing the selected wireless frequencies very far apart, however, this also restricts the number of systems that are usable in any one location. If the user wants a large number of channels in one location, then some of the channels are going to be placed relatively close together. This will very quickly "separate the sheep from the goats."

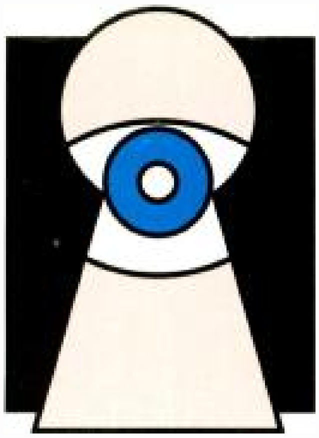
First, let's quickly discuss external RFI sources since they affect all wireless systems and then we'll spend more time on the in-system RFI problems such as crosstalk and intermodulation.

EXTERNAL SOURCES OF RFI

Wireless mic systems operate within specific frequency bands allocated by the Federal Communications Commission. Everybody and their brother's dog wants more spectrum space to operate all kinds of RF devices at whatever power levels they think they need to make their particular gizmos work. The applications include wireless mics, remote controls, communications, video signals, digital data transmissions and so on. The simple fact is that there is not as much frequency spectrum space available as there are demands on using it. So, what we are left with is "shared spectrum space," where the wireless mic systems use the same bands as other "more important" users.

There is little hope that your battery-powered transmitter has much of a chance of overcoming the enormous power of the local TV station signal (which may be powered by Hoover Dam).

Wireless mic systems generally operate in several bands, from 169MHz to 216MHz, which include the VHF TV channels 7 through 13, or, in the 470MHz to 806MHz UHF band, TV channels 14 through 69. Above the TV band is another part of the UHF spectrum, from 902MHz to 928 MHz. This upper UHF



band is a general purpose band used by a multitude of different applications, ranging from garage door openers and amateur radio to telephone company links. Generally speaking, the 902MHz to 928MHz band is not a good choice for wireless microphone systems, especially for traveling applications. There are so many users in this band that interference is almost guaranteed.

Because multi-channel wireless mic systems often use inactive TV channels, one of your first considerations in operating a multi-channel system in a particular area usually involves identifying the local TV stations. If you try to operate a wireless mic system on the same frequency as the local TV transmission, there is little hope that your battery-powered transmitter has much of a chance of overcoming the enormous power of the local TV station signal (which may be powered by Hoover Dam).

There are numerous business radio services that share VHF spectrum space fairly close to some of the wireless microphone frequencies. Interference from these external sources is usually rare, involving some sort of intermodulation problem, rather than from a direct signal on one of the wireless frequencies. Other sources of external direct signals can come from 2-way radios, leaky cabling from CCTV systems, temporarily installed (rental) wireless systems, wireless intercom systems and numerous other radio devices. In addition to direct signals from external radio devices, there are also numerous sources of RFI possible from what is called "man made noise." This is generally broad band RF noise generated by a number of different types of devices, including switching power supplies, computers, computer terminals, computerized telephone systems, digital signal processing equipment and a broad assortment of electrical power equipment. Locating the sources of RFI from these types of sources is usually a matter of turning things off one at a time and locating the culprit through a process of elimination.

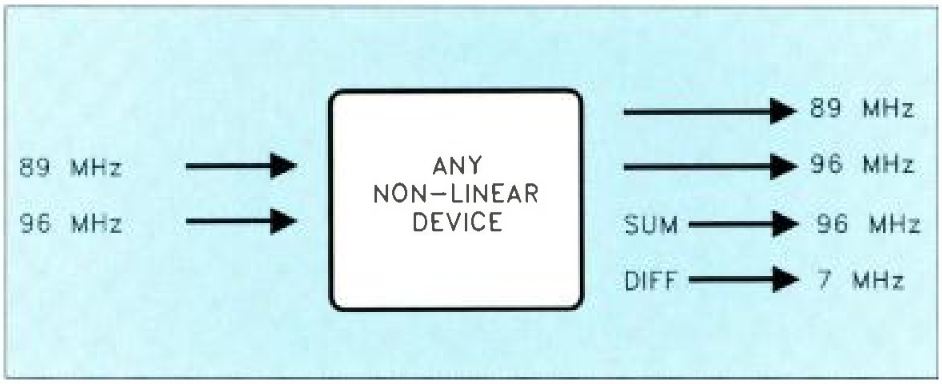


Figure 1. Second order intermodulation.

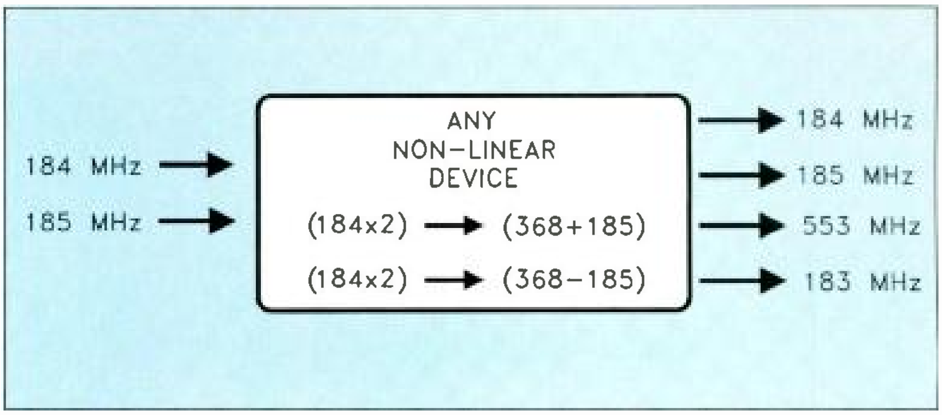


Figure 2. Third order intermodulation.

FINDING PROBLEMS

Doing a "sound check" for the wireless system is just as necessary as checking out the sound system itself. TV stations normally operate with continuous carriers 24 hours a day, so if RFI is going to be a problem generated by the local TV transmissions, it will usually be constant. Business radio services, however, usually only operate within normal business hours of 8 p.m. to 5 p.m., so you are fairly safe in the evening. The other radio signals (and there are lots of them) in the area may operate at any time, so you simply cannot predict when they might generate an interfering RF signal.

Your best bet is to select clear channels, complete a frequency coordination plan for the systems and use only very high selectivity receivers that also provide high IM rejection. If you are not clear on how to rate the selectivity or IM suppression capability of a particular receiver, call the manufacturer. If they can't give you a clear explanation, you should look elsewhere, because this is one of the most fundamental aspects of any radio receiver. Marketing "hype" in advertising is one thing, but reliable RF performance is another.

INTERMODULATION

When two or more signals are present at any level in a non-linear electronic device (all active devices, such as transistors, are non-linear a phenomenon called "intermodulation" occurs. In an audio amplifier this would be called "intermodulation distortion," or "IM distortion." For example, if two signals are present at the same point, a sum and a difference signal will be produced. This is called second order IM (second order intermodulation; also referred to as a two frequency beat), which refers to the sum of the order of the harmonics that produce the distortion. The first harmonic of a frequency is the frequency itself.

In Figure 1 the frequencies (89MHz and 96MHz) fall within the commercial FM radio band. 89MHz and 96MHz, however, are so far below the frequency of a wireless receiver, that the receiver front-end and IF (intermediate frequency) filters can easily reject them. Second order IM from two external sources rarely creates a problem in the receiver except in two unusual situations, which are discussed in next month's section entitled "RFI in Systems."

Third order IM can cause real problems that can't always be prevented by highly selective receiver front-ends. In the case of third order IM, it is possible for the interfering signals to be simultaneously close together and close to the receiver's operating frequency. In this case, the interfering frequencies will go right through the front-end filters and cause problems in the first mixer in the receiver.

Third order IM can occur from the mixing of three signals, or from the mixing of a signal and a second harmonic of another signal. This primarily occurs in two places in wireless systems; at the first mixer in the receiver and between the transmitters themselves. If two transmitters are within several feet of each other, the transmitter output stages can mix the two signals together with very interesting results.

In Figure 2 the second harmonic of 184MHz (368MHz) mixes with 185MHz, producing a signal at exactly 183MHz. Obviously this is going to create a problem with a 183MHz system, since the 183MHz receiver would respond to this IM signal just as well as it would to its own transmitter.

The fact is that radio signals will combine to produce IM signals through second, third, fourth, fifth, sixth and even seventh order combinations. Multi-channel wireless systems work reliably, however, after much time is spent engineering the systems, and a careful installation is performed. The reliability factor improves dramatically if you use only high-quality receivers designed for multi-channel environments. The performance specs on a receiver can be a bit nebulous, but among the most important

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specs for multi-channel capability are selectivity and IM immunity (third order intercept).

Selectivity is a specification that indicates the bandwidth of the receiver front-end filters and the IF filter stage. An excellent receiver front-end will exhibit well over 70dB of suppression of RF signals at ±4MHz away from the carrier. IF filter performance is generally rated as a specified bandwidth between the half-power (-3dB) points either side of the IF frequency. The highest selectivity receivers available on VHF frequencies have IF bandwidths of less than 50kHz and 60dB of rejection at a bandwidth of 90kHz.

IM rejection in a receiver is best indicated by a performance spec called "third order intercept." This spec refers to the input level of interfering signals needed to produce distortion (IM) of the same amplitude as the interfering signals. A good receiver should have a third order intercept spec of at least -15dBm. The best receivers available will have a spec of +5dBm or higher.

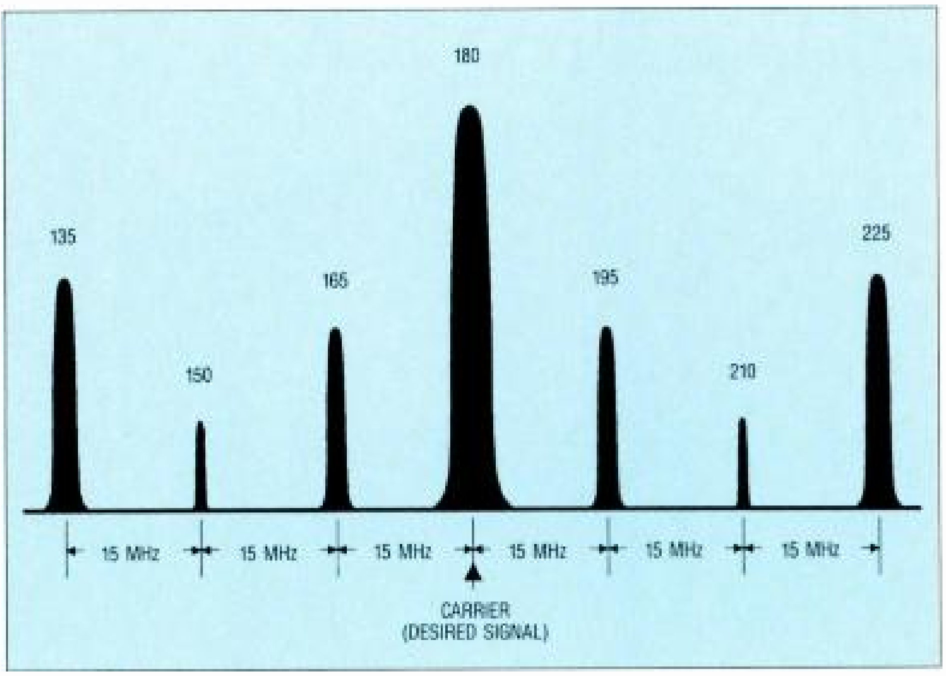


Figure 3. Transmitter spurious.

RFI IN TRANSMITTERS

Obviously, if two wireless systems are too close together in frequency (less than 400kHz) they will generate audible interference in the receivers, or there will be a large reduction in the operating range of the systems. However, even transmitters that are widely separated in frequency can produce interference. Transmitters produce not only the desired carrier but also produce a number of "spurs" (spurious emissions) that are lower in power than the carrier. For a transmitter at 180MHz there will typically be spurs at 15MHz intervals on both sides of the carrier, namely at 135, 150, 165, 195, and 210MHz. You will have interference if you have a receiver on or close to one of these spurious frequencies.

You can test for transmitter spurs by urning all the receivers on and then turning on one transmitter at a time. If two receivers come on at the same time, turn off the receiver that matches the transmitter and see if the other receiver remains on. If it does, you've probably got a spur from that transmitter. If the problem goes away when the transmitter is moved farther away, and the transmitter will always be used at that distance (or farther), you will probably be OK. High-quality transmitters have output filtering that will reduce spurs, but it is difficult to eliminate them entirely. Proper frequency coordination is the best solution.

Transmitters can produce interference if two transmitters are within several feet of each other. The RF signals can combine in several ways that can cause you real problems. Third order IM is one common problem. The other problem is overload of the output stage in one or both transmitters.

The symptom of third order IM is interference in a receiver that is not on either of the two transmitter frequencies. For instance, transmitters on 183MHz and 184MHz can generate interference in a receiver on 182MHz, if the transmitters are within a few feet of each other. Because third order IM is discussed in more detail in the section dealing with receiver problems, just remember that it can happen between transmitters as well as in receivers. Proper frequency coordination is always the best solution.

Another problem related to transmitters is caused by RF energy from one transmitter antenna coupling into the other transmitter antenna and causing output stage instability or overload. The symptom here would be one or both matching receivers squelching or producing very nasty noises.

If either of these symptoms occur between transmitters, your solution is to keep the transmitters farther apart or change frequencies on one (or both) of the transmitters. Moving the transmitter to the other side of a person's body may solve the problem. Just remember to check this out or a passionate embrace on-stage between two performers may sound like "Robbie the Robot meets R2D2."

Everybody and their brother's dog wants more spectrum space to operate at whatever power levels they think they need to make their particular gizmos work.

RFI IN RECEIVERS

All wireless mic receivers operate through a process called "super-heterodyning." A local oscillator inside the receiver generates a fairly strong signal that is mixed (heterodyned) with the incoming RF signal in the mixer stage of the receiver. The result is a "sum" and a "difference" signal. The difference signal (the Intermediate Frequency or "IF" signal) is then heavily filtered and ultimately converted into an audio signal. This process is used to lower the radio signal frequency to make filtering and demodulation much easier. For instance, a carrier at 194.7MHz would be mixed with a local oscillator (LO) at 184.000MHz to produce a standard IF frequency of 10.7MHz.

The oscillator in a superhet receiver can radiate energy outside of the receiver housing, usually through the antenna port. This radiated energy can easily enter another receiver located next to it, injecting the signal into the adjacent receiver. When this happens, the adjacent receiver can respond to signals from the first receiver. In other words, one receiver can generate interference for another receiver sitting next to it, even though neither transmitter is turned on. A 184MHz receiver mounted in the same rack with the 194.7MHz receiver will pick up the 184MHz local oscillator as well as it will its own transmitter.

Careful design regulates how much LO radiation is allowed, but the tolerances are far above the low levels that can create problems in multi-channel wireless systems. A well-designed front-end in a receiver is instrumental in minimizing LO radiation from the antenna port. A simple test of placing the receivers next to each other and observing the squelch indicators (normally labeled "RF") will usually tell you if there is a problem with LO crosstalk.

The local oscillator in the receiver can also generate other spurious RF signals that are not as obvious as the above example. Most manufacturers of wireless receivers have chosen receiver operating frequencies that are compatible for multiple installations. However, using receivers made by several manufacturers in the same installation may lead to some unpleasant surprises and lots of finger pointing. You can test for

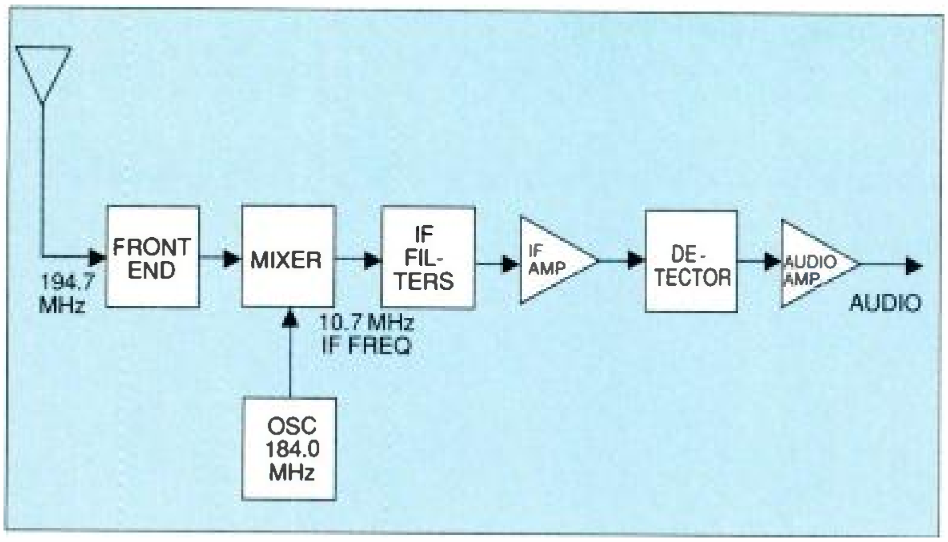
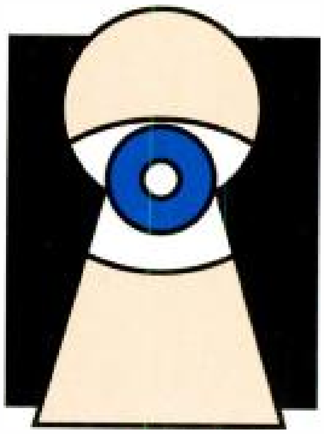


Figure 4. Single conversion superhet FM receiver.



this problem by hooking up the receivers exactly as they will be used (rack, audio cables, antennas, grounds, etc.), and turn all the receivers on with all the transmitters off. If one or more of the receivers is indicating that it is receiving a signal, turn off all the other receivers. If the signal disappears, you probably have crosstalk. Next, try turning on the other receivers one at a time to locate the culprit. The simplest solution is to change the frequency of either the offending or the offended receiver or relocate one of them. You will then have to try the same test again, of course.

In next month's installment of **"Secret Knowledge of the Ether, Part Two,"** we'll address the operation of transmitters and receivers together in large systems, attack specific solutions to RFI, provide practical system compatibility tests you can perform before the show and inform you where to turn for help when you have a problem. We'll also give you some important information for evaluating wireless systems before you spec, rent or purchase. ■

__Note:__ The information in this article is available as a part of a larger publication covering the design concepts and implementation of wireless microphone systems for professional and commercial users. Please call 800-821-1121 for a copy of this publication. Special credit belongs to Larry Fisher, vice-president of engineering at Lectrosonics, for his technical expertise and generous assistance.

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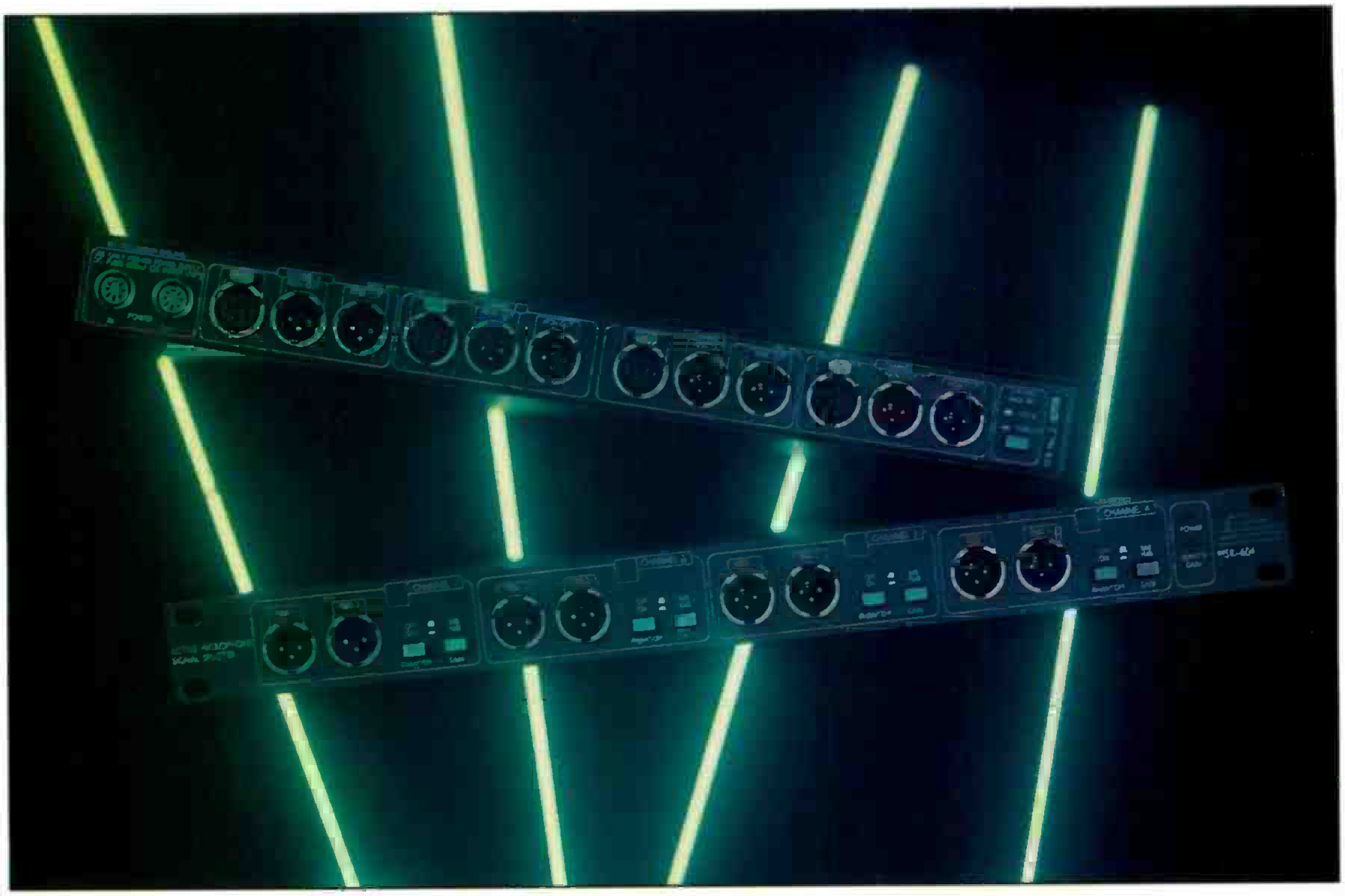

ENHANCING STAGE MIC PERFORMANCE

Evaluating a better way to split and distribute microphone signals on stage.

By Ben Duncan

Concert situations, whether sound reinforcement or live recording, can provide problems galore. Aside from the practical logistics of splitting mic signals from two to four ways, there are always recurring questions regarding ground quality and electrical safety. And then there's fidelity. It has often been observed that in concert PA systems, familiar mics don't give their best performance.

This article explains how mics can be made to sound more like they do in a studio, by using active buffering on stage, while overcoming hum-loop and safety problems that regularly concern mobile recording engineers, audio-for-video engineers and live sound system operators.

Any article discussing the details of mic splitting needs to start with the technical issues. As a rule of thumb, most professionals commonly assume that the impedance of dynamic microphones is about 200Ω to 250Ω. In practice though, impedances are highly variable, ranging as high as 2,500Ω at resonant points, and down to 50Ω, across the full frequency range and between different manufacturer's models. These variations are enough to affect the mic's response when connected to a conventional mixing console with the industry nominal input impedance of about 1,200Ω.

The mic's response is most likely defined by the manufacturer using a reference measurement across a high impedance test receiver. It is with this kind of loading that the performance for almost any professional mic is established.

Ben Duncan is a free-lance pro audio equipment and systems design consultant based in the U.K.

SPLIT SIGNALS

In a concert situation, without a splitter, trouble compounds as soon as we add a second 1,200Ω mixer load to the mic, such as for a stage monitoring system or split feed, and maybe another split or two for a live recording or broadcast hookup.

Using Ohm's law in its simplest application, what we now have is a varying source impedance of up to 2,500Ω connected across 600Ω, 400Ω or 300Ω of mixer load.

But there's more, above 5kHz, the load impedance tends to fall further with the reactance of hundreds of yards (or meters if you prefer) of cable, the console's RF filtering, generally comprised of shunt capacitors, and maybe the occasional choke or two.

Even without making any formal measurements, and based on simple calculation, the outcome is going to be a mid-band attenuation of 3dB to 5dB or more, creating some changes in the mic's frequency response. The problem of multiple mixers or mic amplifiers connected to a mic isn't unique to just touring sound, it can occur in complex installations as well. But the effects of loading microphones are often audibly more apparent with live music than with systems handling speech or reproduced sound.

Engineers responsible for recording the concert mostly have to live with what the PA rental company offers them. Getting, and keeping, the desired sound is made harder for everyone, because if any input on any console is switched during the performance, between the mic (1,200Ω) and line (10kΩ bridging) setting, the abrupt change in loading will affect the level received by the other two, and with some mics, the change in frequency response will be enough to set off a "howlround," requiring some frenzied EQ and level adjustments.

X-FORMERS

In modern set-ups, PA rental companies regularly offer 1:1 × N (eg: 1:1:1 gives two outputs) transformers to isolate recording and broadcast feeds, and perhaps the monitoring system. Aren't these supposed to help with impedance matching? Assuming the transformers are of high quality and 1:1 ratio, and noting the cardinal law of the conservation of energy, 1:1 × N transformers can be seen as a black box with unity voltage gain (0dB) that apportions (through reflection) the impedance seen at each output equally between all the outputs. This happens because the transformer is passive, so it can't increase the mic's power, and the voltage is fixed. So the current must be the quantity that's divided equally into the 'N' outputs, and the reflected source impedance appearing at each output must be multiplied by the same proportion.

Overall, a perfect splitter transformer (the best ones are quite near) doesn't affect microphone loading for better or worse, while a bad one adds frequency response and transient aberrations of its own. What the splitter transformer does achieve is isolation, preventing hum-loops and enhancing safety and system reliability, in the high pressure of the hostile back-stage environment.

BSS Audio (a division of AKG Acoustics) has a long experience of producing equipment for PA, recording and broadcast companies. Having a rapport with many live sound mixing engineers worldwide, they were alerted to the way in which microphones needed extra gain whenever added feeds (slant and fill monitors, in-the-ear monitors, broadcast, recording, etc.) were connected to the stage splitter box at one or more of the shows in a tour.

In 1988, the company released their MSR-604 mic splitter rack, which provides four channels of active, balanced, low-noise, up-front buffering for every mic (and any other transducer) on a stage. Figure 1 shows the contents of one channel of the MSR-604. In effect, the mic amp interface, normally located at the mix console(s), is being brought onto the stage and placed upstream of any splitting. A high current output stage then drives two transformer balanced-and-isolated outputs, and two direct-coupled, actively buffered and balanced outputs. The latter are primarily intended for the FOH and stage monitoring mixers, and assumes they're operating from the same mains supply. In this case, the galvanic isolation provided by transformers shouldn't be needed.

The 100% isolated transformer outputs are then available for hooking-up to recording and

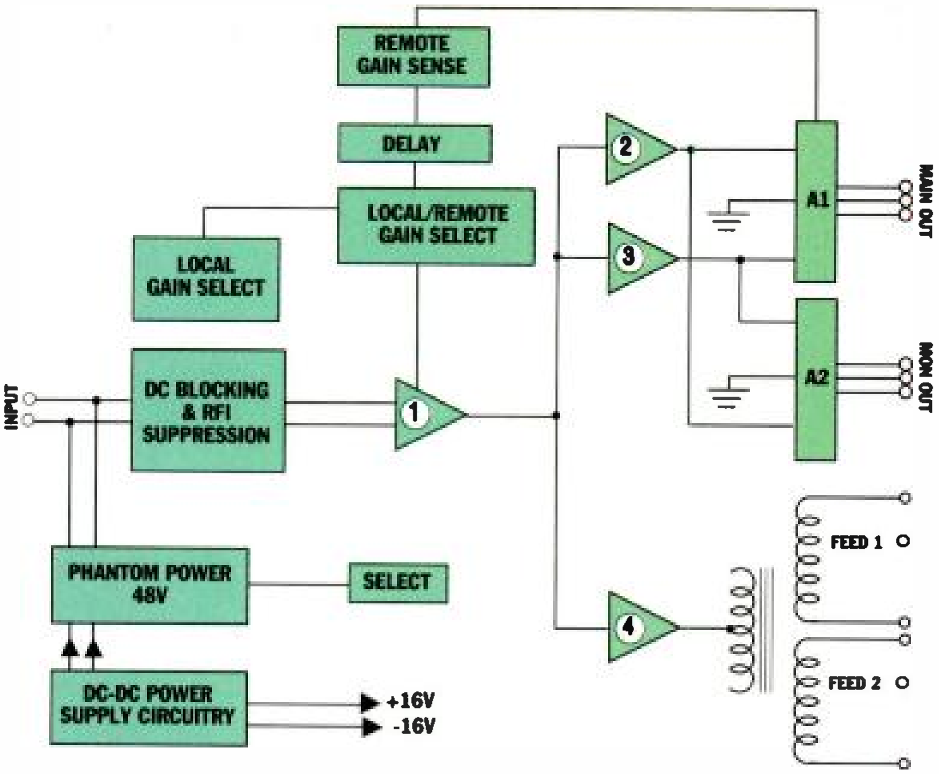

Figure 1. Circuit block diagram of one channel of the BSS MSR604 mic splitter.

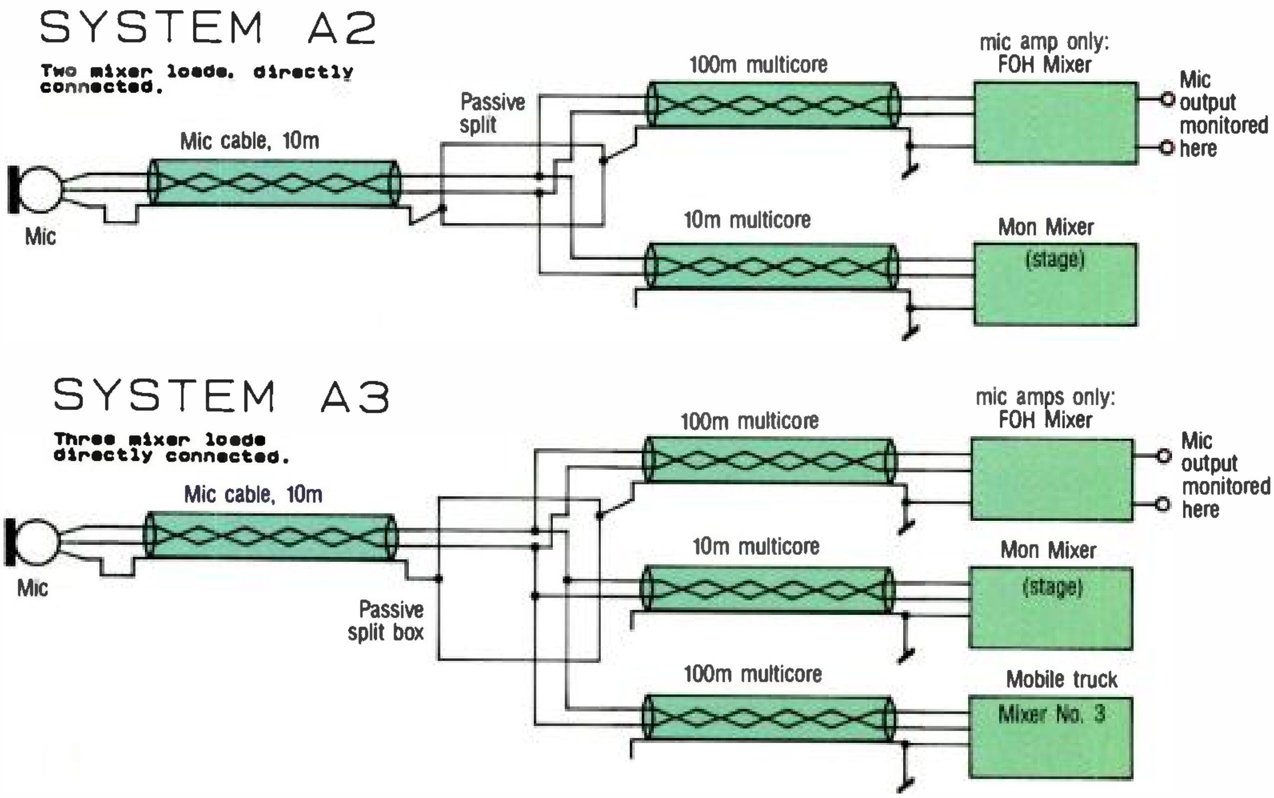

Figure 2. Diagram of directly connected microphone splits. Top: Two mixers as load (A2). Bottom: Three mixers as load (A3).

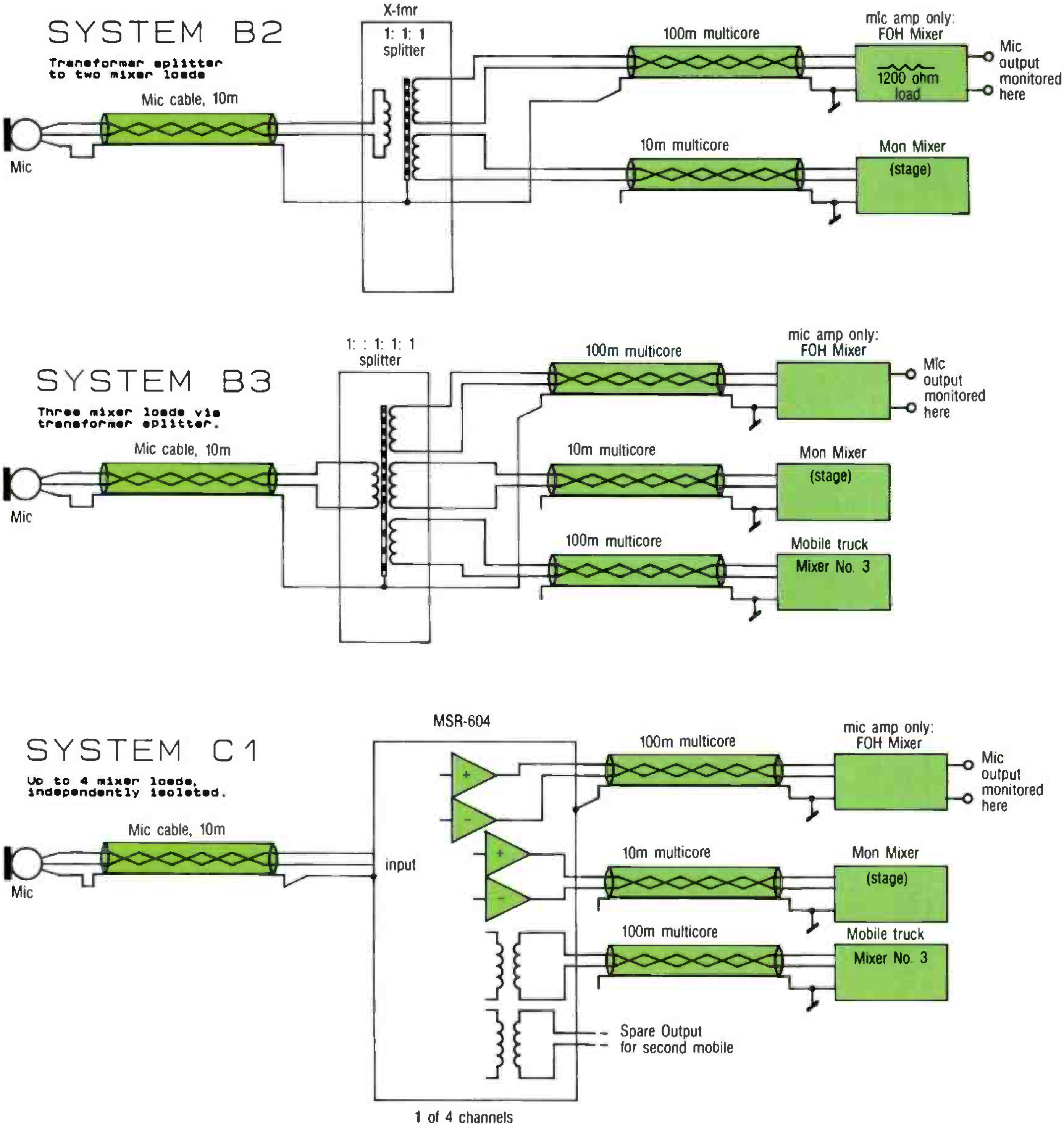

Figure 3. Diagram of transformer-isolated microphone splits. Top: Two mixers as load (B2). Middle: Three mixers as load (B3). Bottom: Active/passive splitter. Up to three additional mixers can be connected without interaction.

broadcast trucks, allowing them to run from different or their own ac mains supplies. As all the splitter rack's outputs are driven from a very low ($<1\Omega$) source impedance, a very high level of isolation is assured, making each console effectively immune from sonic changes caused by the loading presented by the others.

Aside from reinstating the sonic quality of mics on-stage, BSS's MSR-604s have also been used to isolate line-level feeds where there are U.S. and U.K. amplifier racks operating on their native mains voltages (eg: the recent "Monsters of Rock" tour of Europe). By all accounts, the system has been successful, however, being engineers at heart, the manufacturer didn't forget that the performance gained by applying the system had never been formally qualified. And previously published literature on the topic of iterations between mics, cables and mixers is scarce — if any exists at all.

OBJECTIVE ASSESSMENT

When the author was asked to look at qualifying the effect of using the active/passive splitter in typical stage PA set-ups, and compare it with traditional passive-only splitting, the real-world approach of physical measurement seemed the best way to find out what's going on in such a complex situation. But simply measuring a friendly rental company's PA set-up during rehearsals would be too intrusive (for the crew) and too specific (for the measurer). The equipment line-up could be broadened, but ultimately, a physical measurement of a complete system (whether in the field or a lab) was rejected, because of the logistics and costs of hiring and/or borrowing six different models of popular mixing consoles, four snakes (preferably in different lengths), the different, commonly used splitter transformers and dozens of different types of microphones. Even if all the equipment arrived in one place on the same day, and it all worked out of the box, and even using the world's fastest audio test set, methodically measuring a useful sample of the permutations could take weeks. Could there be a cheaper, less pressured approach?

Yes. Computer software for the simulation of electronic networks has reached a stage where a realistic analysis of real audio systems can be carried out at arm's length. With sufficient experience at driving analog simulators, and a knowledge of the attributes of real components, accurate models of the components in

BSS MSR604 microphone signal distribution system with multi-pin interconnects in stage rack.

the chain can be constructed either by measuring samples of the equipment, and/or by derivation from the essential data. As there are countless variables, there is no need to worry about extreme accuracy.

Compared to real measurements, the benefits of simulation are that there's no noise to obscure results, and no time wasted by incidental problems such as faulty connecters or verifying the physical connectivity of diverse multicore connecters. Meanwhile, rental hire, measurement and return costs are reduced to a subset of those parts for which suitable data is unavailable.

MODELLING SYSTEM COMPONENTS

Using Micro-CAP III software for simulation, modelling the mixer front-end was relatively straightforward: A simplified version of a generic mixer front-end is drawn directly onto the screen. By appending reasonable percentage tolerances to the parts that define impedance and frequency response, the envelope of possible responses was broadened to include the majority of actual measured console mic inputs. The model's input impedance and frequency response were then plotted. From this, a condensed circuit was evolved, which still closely modeled the mixer front end while keeping down the number of nodes (as circuit complexity builds up, too many nodes slow simulation). This network was used for the one mixer output being monitored. A slightly simplified model was used for the other one or two mixers, which for analysis purposes, solely present their loading.

The universal PA system's cabling comprises mic leads to the stage box, and multicore cable thereafter. Data was gathered on the resistance, inductance and capacitance of three regular types of mic lead and snake known to be used by a number of U.S. and U.K. PA companies. The cable modeling was kept simple. To figure in any effects the cable might have on impedance interactions and frequency response, a simple network of lumped series resistance, inductance and shunt capacitance between the three cores sufficed. The system models described here used the model of common, well-known mic and multicore cables. The multicore had parameters of 11Ω dc resistance and 30μH inductance per core per 100m (105 yards). Capacitance was 13nF core-to-shield, and 7nF core-to-core, per 100m (130pF and 70pF per meter).

CHOOSING SOURCES

A number of sound engineers helped us identify some of the microphones (out of the myriad models around) that were particularly sensitive to loading. Using impedance vs. frequency plots supplied by the manufacturers, passive networks were constructed that made the simulator's ac analysis driving-source look like a mic. Amoung those evaluated were AKG C1000s, D190s, D112s, and D321s; E-V 757-IIs; Sennheiser MD-521s and 531s; and Shure Beta 58s. From these, we selected the E-V N-Dym and the Sennheiser MD-531 as representative examples. Some of the remaining microphones are less extreme in their impedance fluctuations, but most have impedance patterns which follow one or the other of the chosen units.

Three models of splitter transformer were selected. The first manufacturer we approached refused to supply any detailed data but gave enough information in the data sheet for a model to be constructed. For comparison, Au-

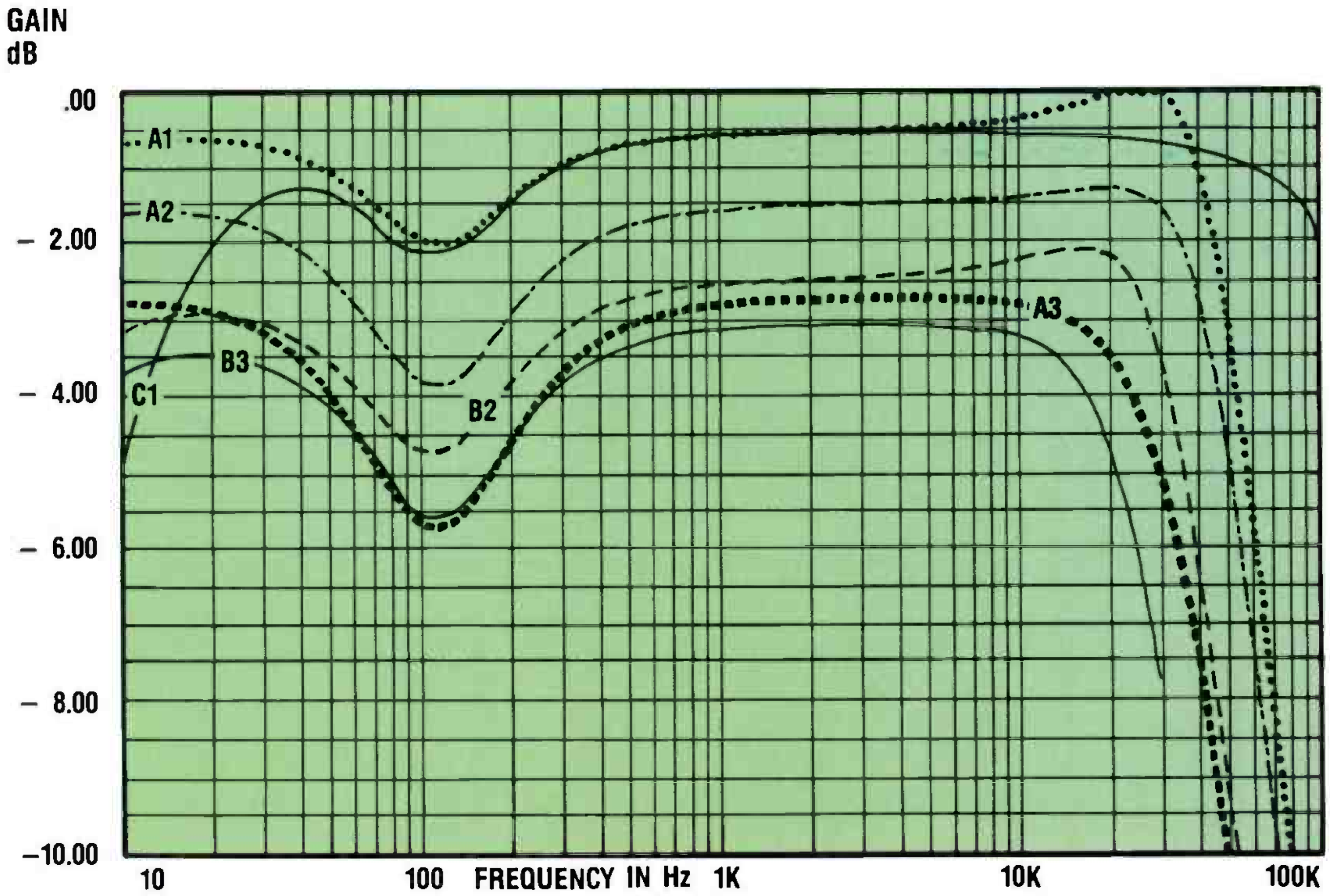

Figure 4. Sennheiser MD-521 frequency vs. amplitude modeling, showing various loading configurations. A1: Direct to one console. A2: 2-way direct split. A3: 3-way direct split. B2: 2-way transformer split. B3: 3-way transformer split. C1: 4-way active/passive split.

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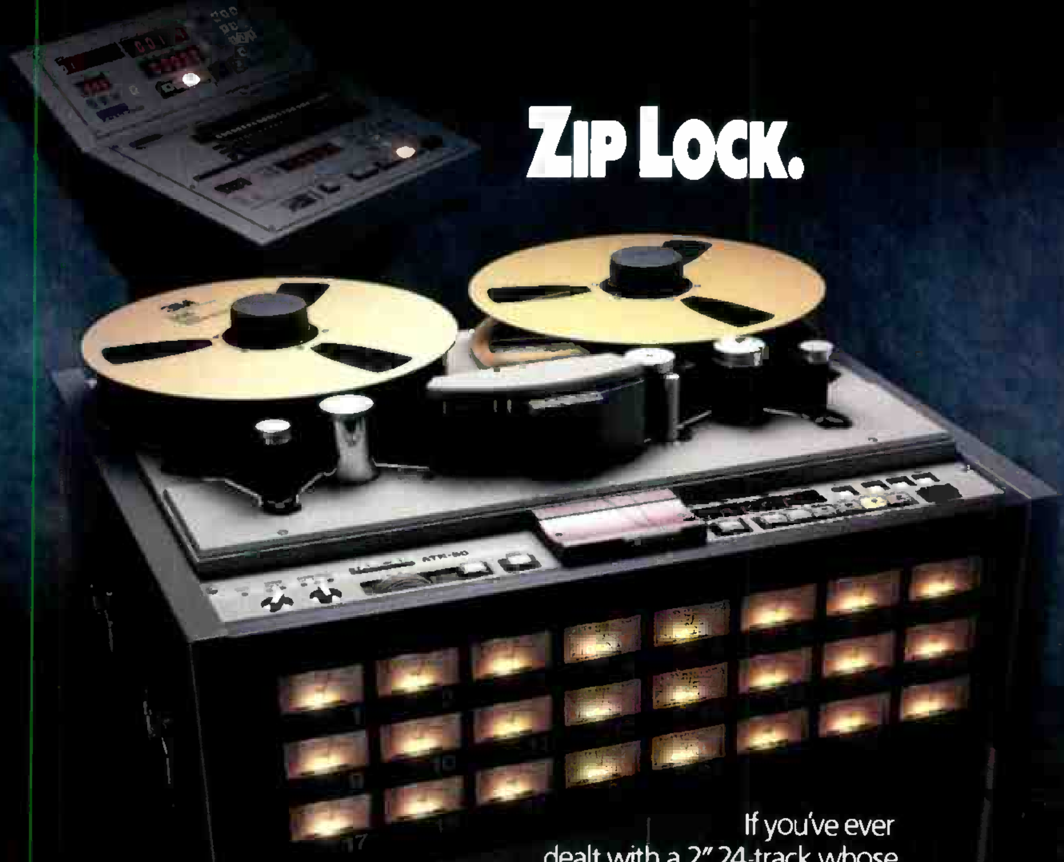

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TASCAM II

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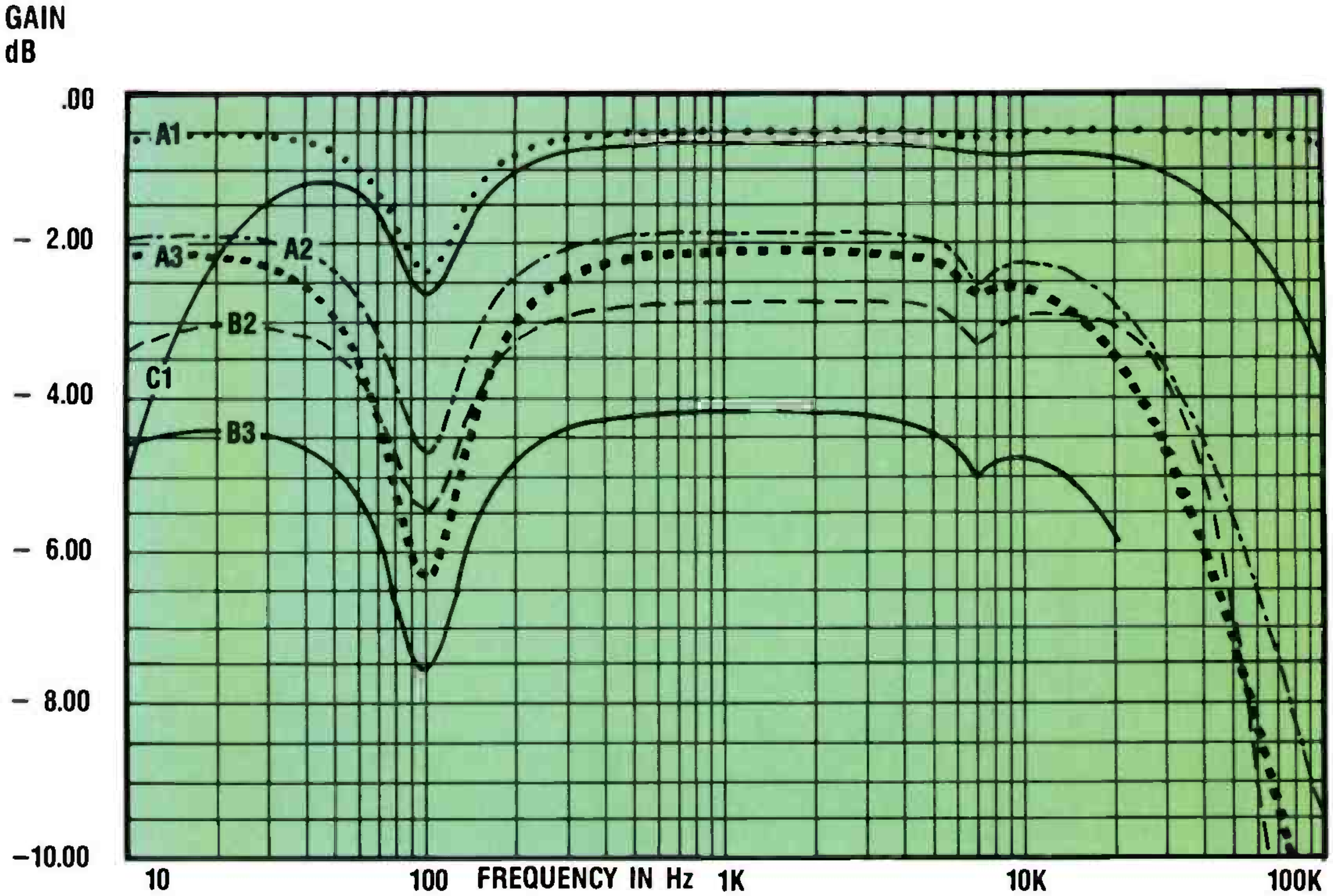

Figure 5. E-V 757-II N-Dym frequency vs. amplitude modeling, showing various loading configurations. A1: Direct to one console. A2: 2-way direct split. A3: 3-way direct split. B2: 2-way transformer split. B3: 3-way transformer split. C1: 4-way active/passive split.

dio Precision (AP) plots of the frequency, phase and impedance response of a sample transformer were made under defined test conditions. The computer model of the splitter transformer was then surrounded by a model of the test circuit and adjusted until the results best fitted the two sources of data. The other two makes weren't so well documented, so they were modeled directly from AP measurement data.

DEFINING THE SYSTEM

Five typical system set-ups (a subset of the three groups, each having four possibilities) were chosen for comparison with the chosen mics. In each case, 10m of mic cable was assumed to exist between the mic and the split, followed by 100m/330' of multicore terminated in a 1,200Ω mic input for the FOH board and (where applicable) mobile feeds, with 10m of multicore for the monitor mixer's feed. System A1 is as above only, and not depicted graphically.

In system A2 (See Figure 2, Top), two mixers are directly connected across the mic. These are usually the FOH and monitoring consoles. In system A3, three mixers (one broadcast or recording) are connected across the mic (See Figure 2, Bottom). In systems B2 and B3, two and three mixers are respectively connected via a splitter transformer (See Figure 3).

In system C1, one mixer is connected to one of the outputs of BSS's MSR-604 splitter rack. Up to three additional mixers can be connected without interaction. This is implicit and, while not shown, was tested and verified by simulation, as well as by measurement.

Figure 4 exhibits the nominal response of the Sennheiser MD-521 mic into the different system schemes. The MD-521 alone has an impedance peak, hence the amplitude drop at around 110Hz and a slight rise in amplitude above 15kHz. The loading effect of the microphone inserted into system configurations A2 and A3 is shown. Note how the 110Hz dip is exacerbated. With the mic connected to systems A3 and B3, the transformer split helps reduce the deepening of the 110Hz dip by about 1/2dB. There is also a faster rolloff at the band edges (<30Hz and >15kHz).

With the MD-521 connected to system C1 with the direct and transformer splits, the output is much smoother above 10kHz, and closely approaches the mic maker's intended response at 110Hz. The visual effect of the dip is changed because the active/passive has an earlier LF rolloff than the transformer or direct split, although it's no more than many other line level processors, being –3dB at 15Hz.

Figure 5 displays the various nominal responses of the E-V 757-II N-Dym mic, which with a single load has an impedance peak (hence amplitude dips) at around 100Hz and a small 'blip' dip at 7kHz. The microphone, when feeding configurations A2 and A3, shows an increase in the depth of the 100Hz dip, from 1dB to 2dB and 4dB respectively. Note also how the 7kHz blip is magnified, enough to have an audible effect on vocals and percussion. The contrast to system configurations B2 and B3 shows that the transformer split clearly ameliorates the magnification of the 100Hz dip. However, when loaded with the B3 configuration, the 7kHz blip is still being magnified and the hf rolloff is beginning prematurely at 10kHz; it's no better than the passive split in A3. This plot was stopped at 20kHz, as the B3 system involved simulation with over 450 components, pushing the computer's expanded memory to its limits.

C1 shows what happens when the E-V 757-II is connected to the MSR-604. The 7kHz blip dip remains insignificant, the hf response is smooth to beyond 20kHz, and the mic follows its intended response without loss, except for the MSR's LF rolloff below 30Hz. Not shown is how the MSR-604 benefits other transducers. Two DI boxes with transformer outputs were modeled and evaluated, and similar improvements were noted.

CONCLUSIONS

A number of commonly used dynamic mics have more widely varying impedance curves than is commonly assumed. This makes their response quite sensitive to loading. Without an active split, the level off the mic can be reduced by between 3dB and 6dB.

High quality 1:1 × N splitter transformers help a little, especially in ground isolation, but cannot be expected to overcome large variations in the driving (source) impedance. Less perfect splitter transformers can add resonaces of their own, particulary above 10kHz, as well as exaggerating existing impedance/response dips. Active buffering within 10m to 15m (33 feet to 50 feet) of the mic (or transducer) and before any splitting effectively prevents microphone impedance variations interacting with the remainder of a complex system, enabling the microphone manufacturer's intended response to be closely and consistently experienced. ■

Note: Micro-CAP II is produced by Spectrum Software, 1121 S. Wolfe Road, Sunnyvale, CA. The author would like to acknowledge the following companies for their help in supplying data: BSS, AKG, Belden, Beyer Dynamic, Brittania Row Productions (U.K.), Electro-Voice, Klotz, Sennheiser and Shure.

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COILS AND WINDINGS PART TWO

In defense of transformers — the continuing saga.

By Richard Guy

This month we continue with Part Two of our series on the Audio Transformer — Relic or Ally. As promised, we'll look at some of the inside secrets about transformers, including some very interesting improvements in the state-of-the-art and cover techniques for using transformers to their best advantage.

Today, audio transformers are referred to as relics in most advertising, with the suggestion that being "transformerless" is a benefit. It's far more likely that expense or cost-competitiveness have eliminated them from professional products rather than any touted performance improvements. The result of all the marketing "bad-press" on transformers is that most users can list all the potential "problems" with transformers, but the same people can rarely think of a single advantage for having them (refer to R•E•P March 1992, "Coils and Windings: Part One").

Bad press not withstanding, the audio transformer still remains the most effective, practical way to solve many common audio problems, such as:

1. Interference. SCR dimmers are ubiquitous and their spikes are unfilterable. The only common mode noise rejection system that works every time is good transformers with static shields. ICs have a limited common mode rejection ratio (CMRR) range, usually much less than is required to block SCR dimmer hash.

2. Ground currents flowing on audio lines. Connecting any two audio systems together with unbalanced signal lines (and even balanced, transformerless lines) causes the ac noises in ground currents to become impressed into the audio signal. The magnetic coupling of audio transformers eliminates this degradation of signal-to-noise ratio.

3. Branching grounds, where one signal enters and multiple signals leave a component. If these signal paths are not transformer isolated, a classic group loop is created: multiple paths to ground. One of the most dangerous places where this commonly occurs is with electronic crossovers. Following ground loops with power amplifiers guarantees poor signal-to-noise ratios, RF and SCR interference, high-frequency instability and unexplained speaker failures.

But then who needs transformers when some expert you know gets away without them, you may be asking?

Richard Guy is president of EXCEL Audio Systems, in Placentia, CA.

THE MAGIC BULLET

One of the amusing aspects of the pro-audio field, concerning both equipment designers and systems engineers, is their quest for, or fixation on, the "magic bullet" ... finding the ultimate problem-solving device/object/component. At best, this device- or object-oriented approach produces only mediocre results.

The manner in which a device/object/component is employed is usually at least as important as the device itself. In other words, the best device combined with poor system engineering rarely delivers acceptable results. Conventional transformers or good transformers used badly produce mediocre results, while conventional or good transformers used well produce outstanding performance.

The following illustration of this "how a device is employed" concept demonstrates its validity. Twenty years ago in the early '70s, Spectra Sonics shipped audio products and mixing consoles using off-the-shelf Triad audio output transformers. These coils were rated by Triad at 1.0% Total Harmonic Distortion at +20dBm, but Spectra Sonics products using them delivered measured distortion of less than 0.01% THD at +18dBm, 20Hz to 20kHz. That's two full orders of magnitude better THD performance than Triad themselves spec'd for the devices.

Today, audio transformers are referred to as relics in most advertising, with the suggestion that being "transformerless" is a benefit.

While the audio industry was busy debating this "outrageous specsmanship," Bill Dilley of Spectra Sonics again demonstrated the importance of correctly employing devices by upping the ante four times, or another 6dB. Spectra Sonics began delivering consoles, using the same Triad transformer, with measured performance of less than 0.01% THD at +24dBm, 30Hz to 20kHz, even though the Triad transformer case was imprinted: "Power Level +20dBm Max!"

This article isn't about publishing the specific Spectra Sonics circuitry, but rather discussing the design concepts that make audio transformers transparent. An understanding of these simple system concepts will make it possible to utilize transformers effectively in your systems and with significant performance improvements.

TRANSFORMERS AND IMPEDANCE

Transformers are impedance-sensitive devices, which is to say, some attention must be given to their source and load impedances to achieve satisfactory results. It isn't necessary to always "match" input and output impedances, except when using vacuum tube equipment, older design passive equalizers or true transmission lines. For most of us, the necessity of impedance matching is long past, along with the 600Ω terminating resistors.

The output amplifiers used in modern solid-state audio equipment have generally low output impedances (30 to 100Ω), while audio inputs are usually mid to high input impedances (10 to 20kΩ and up). This all works to our advantage when using transformers.

An examination of the performance of a series of transformers, from conventional designs to the newer high-tech types reveals some very interesting data.

First, let's look at measurements made on an conventional steel core audio output transformer, the Triad HS-66 (See Figure 1). This coil, when connected using the 600 to 600Ω windings exhibits substantially better performance when driven from lower source impedances. Note the flatter and more extended frequency response, lower harmonic distortion and low-

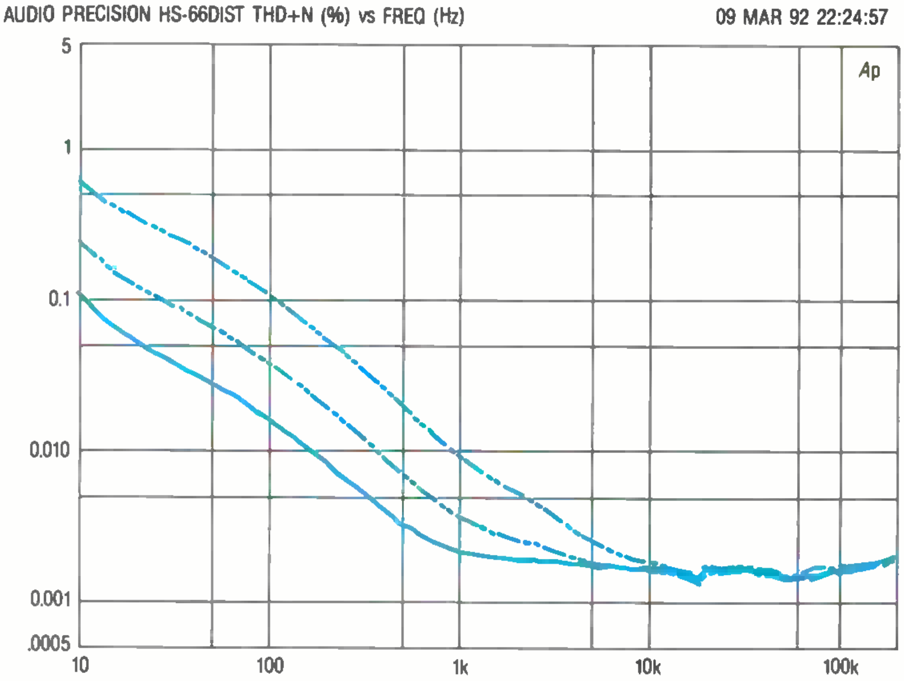


Figure 1. THD + N vs. frequency response of conventional steel-core transformer with optimum secondary load at +4dBm. Top trace 600Ω source impedance; middle 150Ω; bottom 25Ω source impedance.

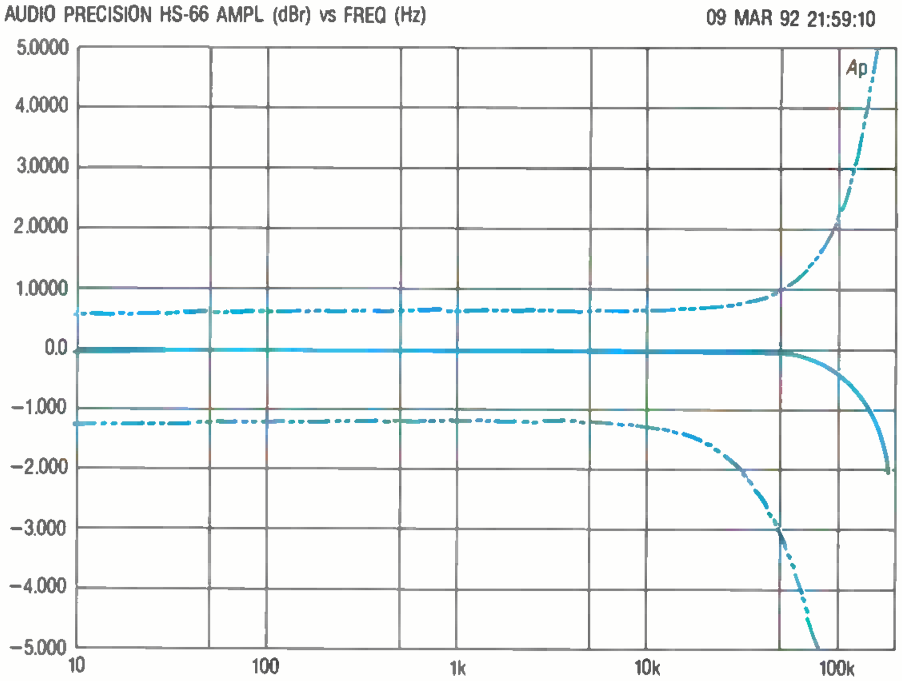


Figure 2. Amplitude vs. frequency response of conventional steel-core transformer with 25Ω source impedance at +4dBm. Top trace 100kΩ (virtually unloaded) secondary; middle optimum termination (see text); bottom 600Ω load.

er insertion loss.

As shown in the graph, the 600Ω source impedance nets performance as spec'd by the manufacturer. A 150Ω source has less than half the distortion and nets significantly improved frequency response. The 25Ω source cuts the distortion to about 12% of the 600Ω source, and further improves frequency response. Measurements were taken at +4dBm.

TECHNICAL DETAILS

A few words about source impedance: Very few contemporary audio products have a 600Ω output impedance. Most are closer to 125 to 150Ω. Many are close to a 25Ω output impedance, especially if the "short-circuit protection" output series resistors were removed (an easy modification to make on many components). The improved damping from the lower impedance source is the real secret here in lowering harmonic distortion, and in extending frequency response by reducing capacitative winding losses.

The audio transformer still remains the most effective, practical way to solve many audio problems.

Figure 2 illustrates this same Triad transformer with 600Ω, 1.8kΩ or optimum termination and 100kΩ (effectively unterminated) loads. The frequency response isn't very spectacular with the 600Ω termination, as the transformer is actually over-damped. Operation with 100kΩ, effectively equivalent to no termination, results in high frequency peaking. This condition, also known as transformer "ringing," is one that leads to system instability or high frequency oscillations.

The optimum termination extends frequency response and minimizes distortion. Because most transformer manufacturers do not publish this optimum value, it's left to the user to determine by measurement. High frequency roll-off response curves will give a good indication, as will 1kHz to 2kHz square wave measurements. Optimum loading/performance is achieved with flattest response, or best square wave performance, with no ringing or rounding of the leading edge of the waveform. The optimum value can easily be determined by using a variable potentiometer as a load and then measuring the pot value.

When this value is determined, a further improvement can be realized with split winding or center-tapped coils. To accomplish this, select two low noise resistors equal to one-half the optimum value. Connect a resistor from each output side of the coil to the center tap. This connection further reduces even-order distortion products. Care needs to be taken to use identical value resistors. Use 1% RN60D types, or match resistors by bridging close values together and paralleling with high value resistors to achieve a precision match.

The point here is that even with conventional audio output transformers such as the Triad, optimum source and loading allows an order of magnitude improvement in overall performance.

HIGH-TECH OUTPUT TRANSFORMERS

Unfortunately, in many audio applications, the source/load impedances aren't always optimum. In fact, we often have no idea what they are! Enter the new generation of high-tech audio transformers. The engineering legacy of Ed Reichenbach, of Reichenbach Engineering, has been the development and introduction to the audio industry of new advanced-winding techniques and improved lamination alloys, accomplishments that have dramatically improved audio transformer performance.

By using special coil winding techniques called bi-filar or quad-filar winding, input and output coil wires are wound together, side by side. As a result, any of the variations resulting from impedance loading on output transformers disappear. Note that in Figure 3, the variations between 100kΩ, 2kΩ and 600Ω loading results only in small differences in insertion loss. Frequency response remains essentially constant.

The transformer performance illustrated in Figure 3 is from a Reichenbach RE-11B, a bi-filar wound steel core design. Comparison with

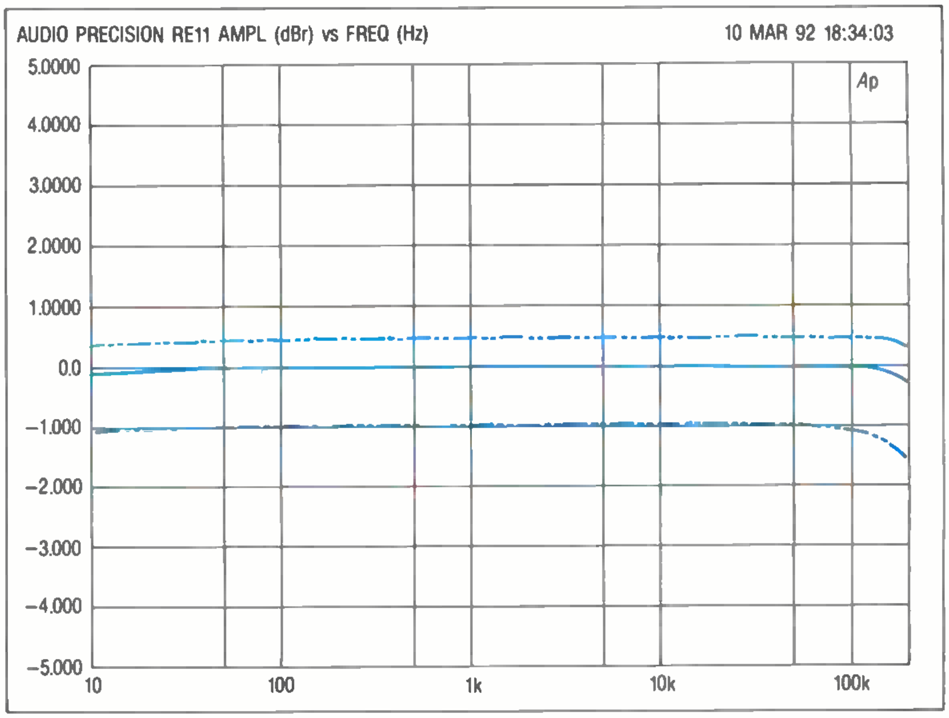


Figure 3. Amplitude vs. frequency response of Reichenbach RE-11B bi-filar wound steel-core transformer with 25Ω source at +4dBm. Top trace 100kΩ load; middle 2kΩ; bottom 600Ω load.

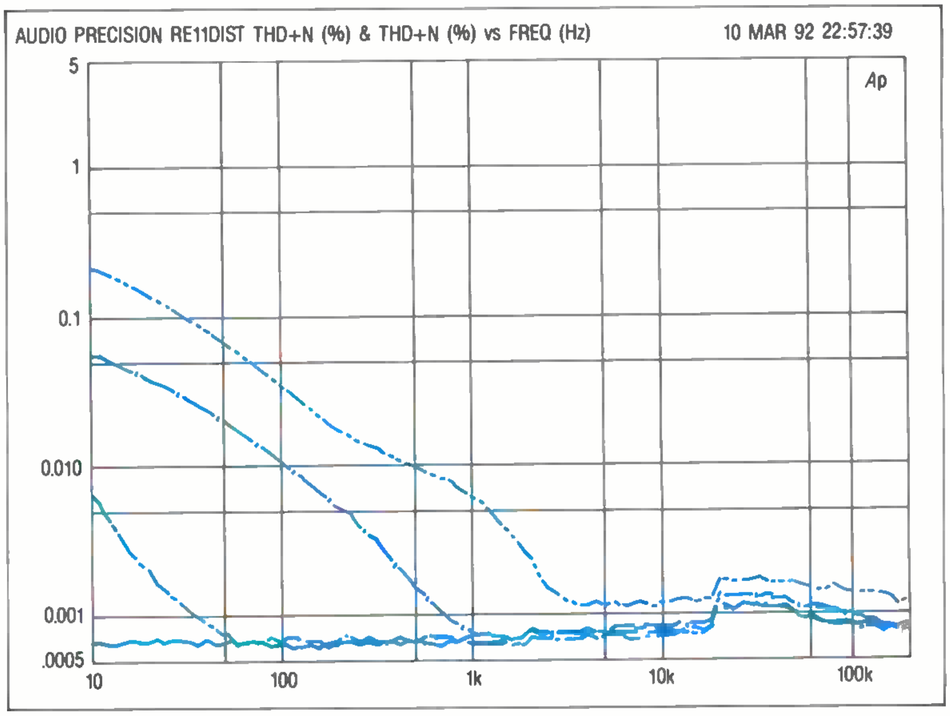


Figure 4. THD + N vs. response of three RE-11 transformers. From top down: steel core design; 50% nickel core; 80% nickel core; test instrument residual distortion level.

the Triad HS-66 conventional design, which uses separated input/output coils, shows a dramatic improvement in high frequency performance.

One of the most liberating aspects of Reichenbach's work has been the development of output transformers virtually immune to gross variations in output impedance loading. By introducing core materials with nickel content, the THD was also greatly reduced.

Figure 4 illustrates three identical Reichenbach transformers, with only the core material changed. The upper trace is a steel core design; the middle trace uses 50% nickel content core; and the lower trace is of an 80% nickel core design. The bottom trace indicated is the test instrument's residual noise.

All the transformers were terminated with 600Ω, usually a worst-case load. Only minor differences were apparent with no load. The 80% nickel content core virtually eliminates distortion products, while delivering a frequency response within .5dB from 10Hz to 200kHz. This would be a tall order for the best audio IC system.

MODERN INPUT TRANSFORMERS

So far, the emphasis has been on audio output transformers. Now we'll turn our attention to the new high-tech line input transformers. Unfortunately, within the limited scope of this paper, we're going to have to pass over microphone input transformers, a subject that could easily fill several articles.

The selection of an input transformer for line-input components rewards the user with an improved signal-to-noise ratio and guarantees virtual immunity from noise interference. Reichenbach, for example, offers a series of coils with medium to high impedances which permits the connected device to see a non-terminating or bridging input. The RE-11P1 is a good example, and a very flexible unit.

This small transformer, about 1-inch round by 3/4-inch high, is wound as a 15kΩ to 15kΩ input transformer. It is equipped with a Faraday or static shield and grounded core, which when combined with its excellent CMRR, makes it virtually immune to SCR hash. Impedance loading with this type of device is important. In fact, the distributor, Bauer Communications, suggests optimum resistor-capacitor loading networks for a range of common input impedances from 10k to 100k.

The performance of this transformer is excellent, as can be seen in Figure 5, with the −3dB point at 85kHz. The high output impedance does require location within a reasonably close distance (a few feet with typical low-cap shielded cable), to the input of the following audio stage.

Another high-tech transformer that fills a wide range of applications is the Reichenbach RE-11SSP-6M. This transformer is designed as a repeat coil and can be used on inputs or outputs. It is a 600 to 600Ω design, with split windings. With its 80% nickel core, mu-metal shielded case and static shield, you can use it almost anywhere. Because this is an input transformer-type design with a static shield, it uses separated coil windings. As a result, you must watch secondary impedance loading (See Figure 6). With performance of −3dB at 200kHz, and distortion typically under .02% THD at operating levels, it's a very handy tool to have around.

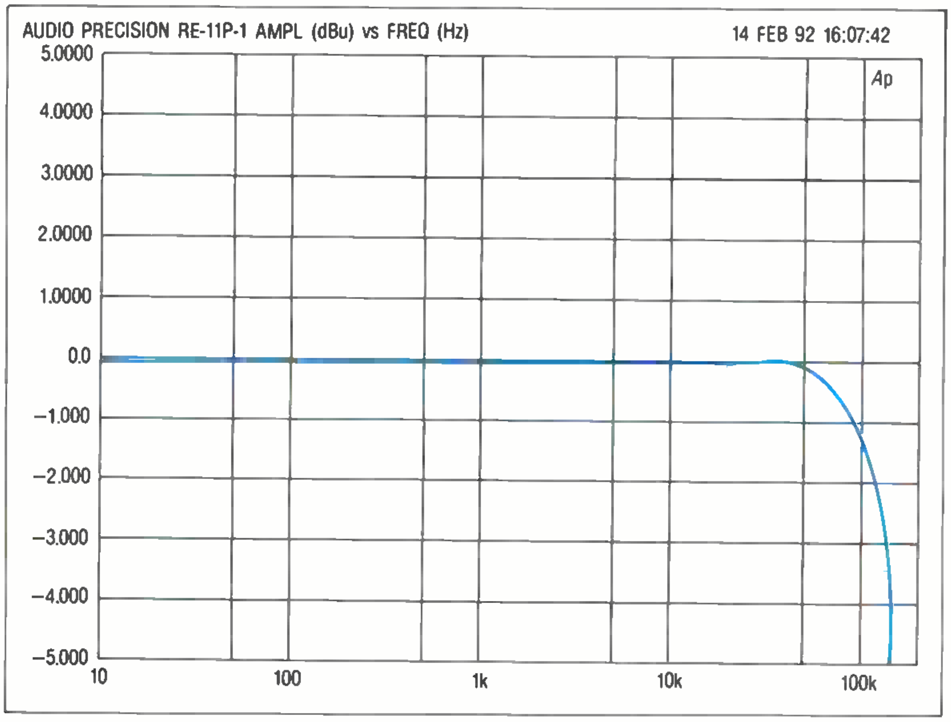


Figure 5. Amplitude vs. frequency response of RE-11P1, 15kΩ to 15kΩ medium and high impedance matching transformer.

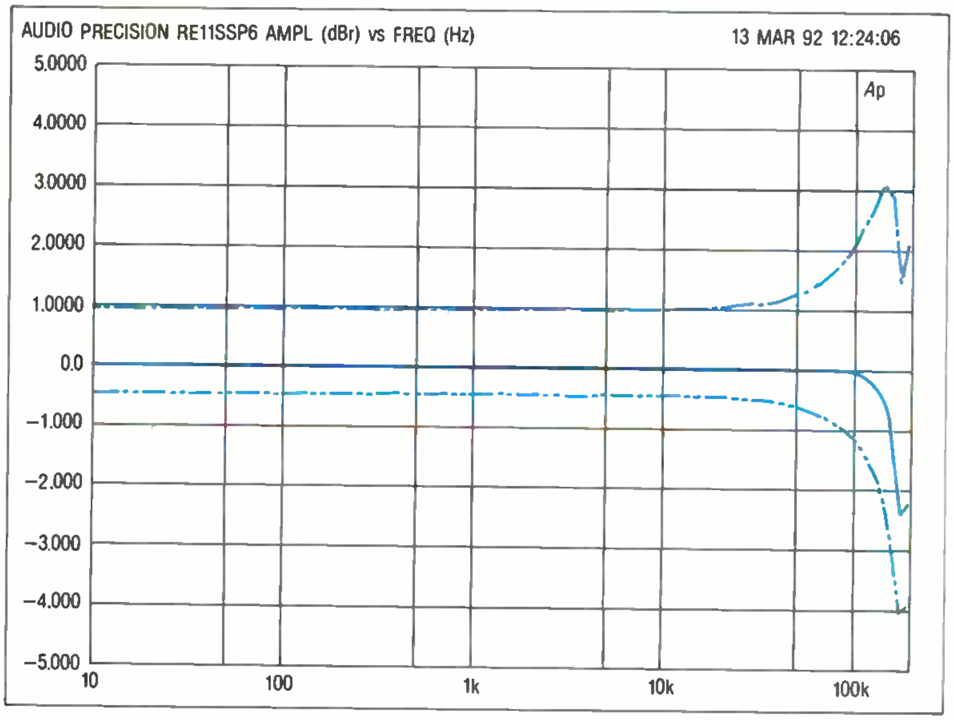


Figure 6. Amplitude vs. frequency response of RE-11SSP6 line-to-line coil with 25Ω source impedance at +4dBm. Top trace 2kΩ load; middle 800Ω load; bottom 600Ω load.

SOME FINAL THOUGHTS

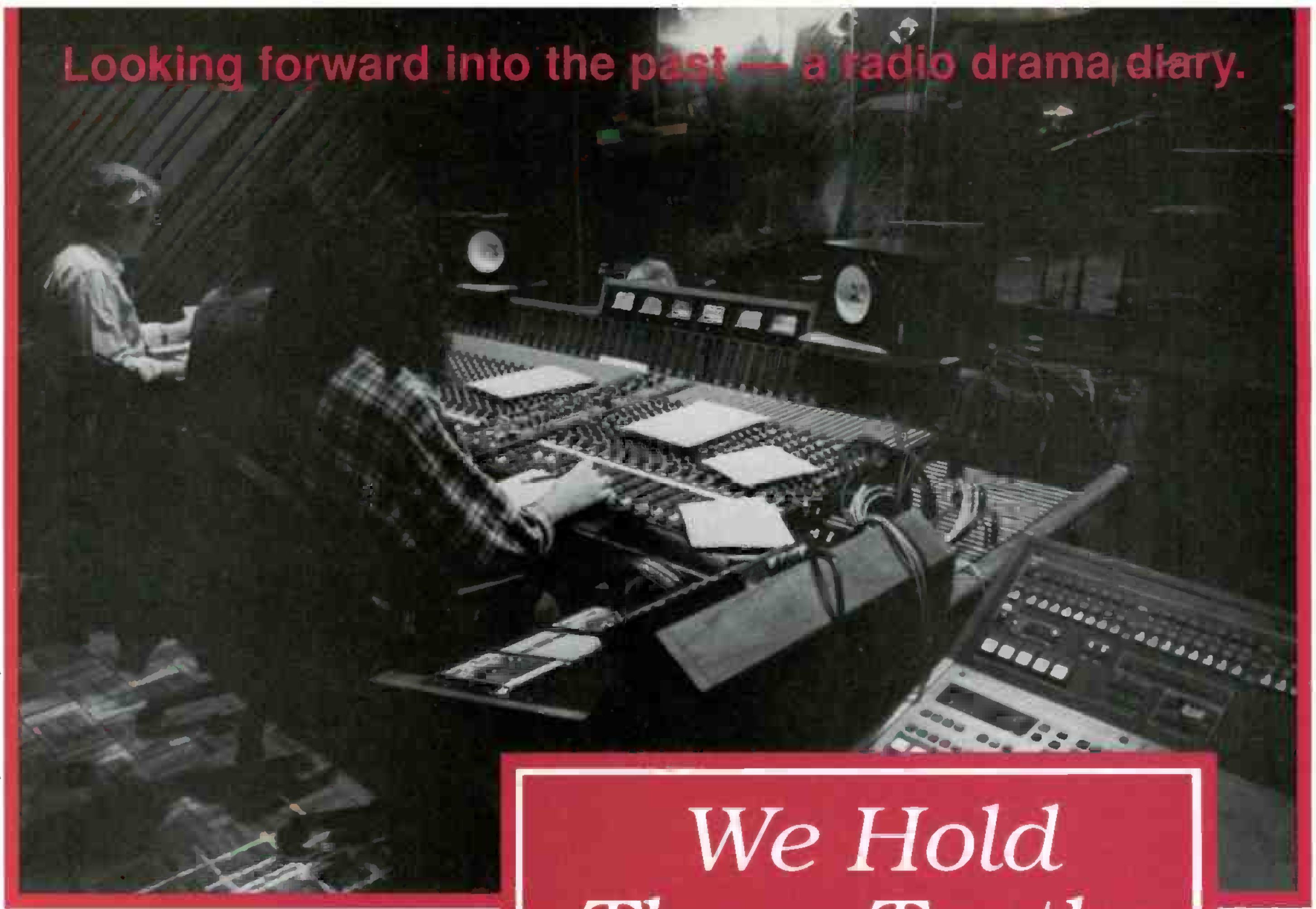
Perhaps this look at transformers has suggested some new applications for you. The next time the audio gremlins are driving you crazy, and the grounding or interference is impossible, try reaching for an old ally, the audio transformer. The new, high-tech designs, no matter which of the several well-respected companies supply them, give you new freedom, without intruding into the character of the audio itself.

Finally, one of the most interesting sidelights to all this talk about transformers and the marketing-implanted fears about what they do to performance:

The Audio Precision System One, the $12,000.00 state-of-the-art measuring system for ICs, component-level devices and electronic circuits and systems, uses an audio output transformer on its ultra-low distortion tone generator — because it's the best way to get transparent results when interconnecting to another system! ■

Note: Thanks to Don Petty of DP Engineering, Mike Hogue with Audio Precision and Jim Bauer of Bauer Communications for their comments and suggestions. Information on Reichenbach Transformers is available from Bauer Communications, 6887 Farmdale Avenue, Bldg. 10, North Hollywood, CA 91605.

Looking forward into the past — a radio drama diary.



Photos courtesy of John Mayer.

From left to right at the console: director David Ossman of Otherworld Media, engineer Fred Jones and assistant engineer Cary Butler during a voice track recording.

We Hold These Truths

By Laurel Cash-Jones

It had all the makings of a truly historical event: I had been invited to attend the recording sessions for a new version of the 1941 radio show by Radio Hall of Fame author and broadcast personality, Norman Corwin. The purpose was to commemorate the upcoming 200th anniversary of the Bill of Rights. Entitled "We Hold These Truths," it was to be broadcast via American Public Radio, CBS, NBC, ABC, Unistar and The Mutual Radio Networks.

The original broadcast of "We Hold These Truths" was performed live on December 15, 1941. The new version was to be broadcast on the same date, 50 years later. This was the only major event (including TV, radio, etc.) that occurred to commemorate this important occasion in U.S. history. The enormity of the production was amazing. The all-star cast included Edward Asner, Rene Auberjonois, Bill Bixby, Steven Bochco, Tom Bosley, Ray Bradbury, Lloyd Bridges, Chief Justice Warren Burger, Pat Carroll, Melanie Chartoff, Richard Dysart, Jill Eikenberry, John Ireland, James Earl Jones, Norman Lear, Rod McKuen, Fess Parker, Melinda Peterson, Philip Proctor, John Randolph, Joe Spano, Studs Terkel, Brenda Vaccaro, Ben Vereen, Jesse White and many others!

Norman Corwin, a famous CBS writer and performer, had updated his original 1941 script. Judith Walcutt was set to produce, and David Ossman (of Firesign Theatre fame) was enlisted to direct the show for their company, Otherworld Media, in conjunction with WETA, Washington. Original music was composed by Libby Larsen. Fred Jones was the engineer.

The show's lofty goals demanded that it be an updated and re-written version of the program originally aired as a live radio show on CBS by Corwin. This time it would have the all-star cast (more than 50 Hollywood television and film personalities lent their voices), and be recorded in a studio with original music and sound effects, using a far more modern approach than the original broadcast.

After the airing date, cassette copies were to be distributed to all the members of Congress and the Senate, along with the President. Add to this the fact that there was an extremely short production schedule (nine days) before the drop-dead satellite date of Friday the 13th (!!!), and you had the blueprint for a pressure cooker.

December 2nd, 1991, 10 a.m.

Today's schedule calls for over 25 pieces of original music to be recorded, a total of 20 minutes of performance from 14 musicians in just under nine hours. The various pieces, ranging from five seconds in length to two minutes, would be added after the actors had layed down their voice tracks. At the Evergreen Studio, in addition to the production staff, the control room was filled with reporters from Entertainment Tonight, *The Wall Street Journal*, *The Los Angeles Times* and a documentary crew preserving this moment in Norman Corwin's life.

Out in the studio room, engineer Fred Jones was setting up the session. His biggest problem was what to do with the people in the control room. "Not only do I have a composer, producer, music director, and several others directly involved in the production at any given time (a table was set up behind the console to accommodate the extra production team), I have to allow for the photographers and

Laurel Cash-Jones is a Los Angeles-based freelance writer.

camera crews to get what they need, while not having any of the noises they are creating creep onto the tape. It should be easy after Norman gets out of the studio and into the lobby, because they don't seem to be that interested in the musicians."

The session is recorded non-Dolby, 30ips on a Studer A 827, using a modified Harrison console. Everything goes fine. They finish about 6 p.m.

December 3rd through 7th

On the 3rd, the mixdown of the previous day's session goes well. The music mix is simplified due to the careful miking techniques of our Fred Jones, who believes that, "If you get it right at the mic, the mix almost happens by itself."

The next two days are spent in a little room editing and sequencing the tracks so that they can be used for the post production session. The production team, headed by Judith Walcutt, finalizes the actors schedules, deals with their agents, and arranges for the last of the funding monies to arrive in time for the Saturday session. It is amazing how many details there are to arrange!

On the 7th, a production meeting is held with David, Judith and Fred to decide the best method of recording and handling the traffic of actors getting in and out at the right time, dealing with the various entourages, and of course, coping with the camera crews. Out of necessity, everything will be recorded out of sequence in order to accommodate the actors schedules. In order to avoid massive confusion in the studio, arriving actors and others will be met and taken to an upstairs office to let them go over their scripts and get prepared. They will be ushered into the studio on an "as-needed" basis. It is decided that the press will be allowed in the studio during mic setup and rehearsals only. They are not happy, but they agree to it if they can film in the control room from time to time.

One of the first decisions David and Fred make is that in order to preserve the continuity of the sound of the recording, there will be a U 47 set up in the corner of the isolation booth at all times to record the narration tracks, because they will be recorded out of sequence and over two days. This is due to the fact that Richard Dysart will appear in many of the other scenes and will have a different acoustical sound from the narration portion.

The script calls for Richard to play "The Citizen," a kind of everyman, who has gone back in time to discover the "truth" in the struggle to ratify the Bill of Rights. However, it does not rely on a traditional style of "going back in time." The character moves around from place to place, scene to scene, with a modern-day attitude, and talks to historical characters with a sense that he is aware of who they are and what they have or will accomplish.

It is decided that if it is possible, the stereo perspective and all elements of the acoustic space (other than any Foley effects) will be done with the microphones and the room itself. Reverb or other means of electronic replication of room sounds will be applied only when the studio doesn't sound right.

Fred Jones says that, "One of the things I prefer to do is to use the microphones and acoustics of the room I am recording to get the sound I want. If I am given the opportunity, I try to actually place the actors and mics in an appropriate acoustical space, not putting them in the middle of a dead recording studio and trying to recreate the atmosphere using a reverb or whatever later. The joy of working with David Ossman is that he gives me the freedom to make the decisions to change the acoustics of any given scene, and to alter the actual staging of the scene in the studio."

Engineer Fred Jones and director David Ossman from Otherworld Media going over production details at Evergreen Studios while being filmed by the documentary crew.

Each scene will be broken down and staged in the studio, and recorded on either an Ampex ATR 102 2-track or an ATR 104 4-track at 15ips (both with Dolby A), depending on the need for separation in post production later. A DAT machine will also roll as a backup.

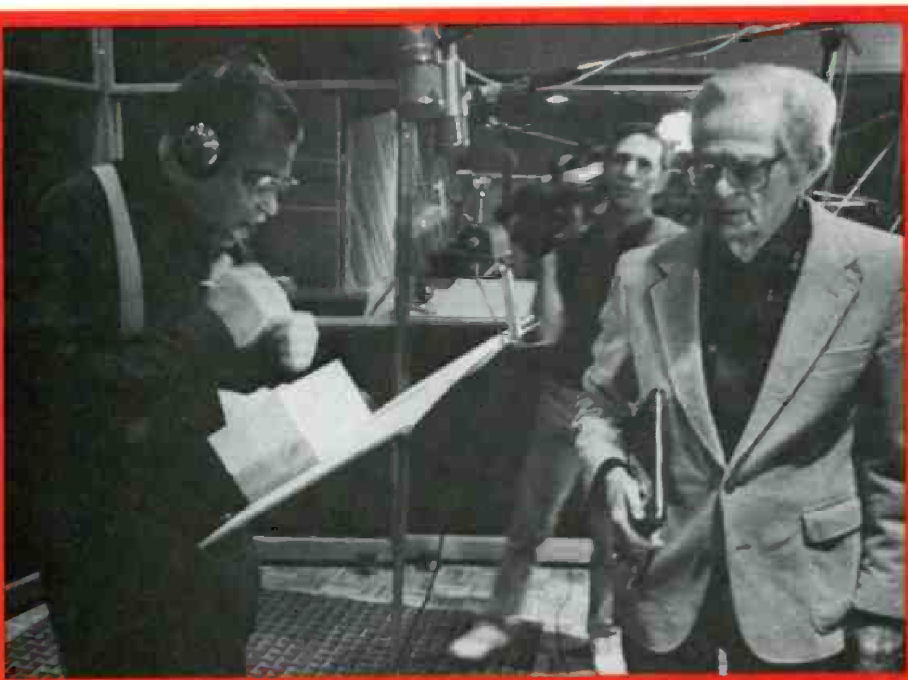
At the appropriate time on the 7th, all in attendance acknowledge the 50th anniversary of Pearl Harbor. The first scene to be recorded is with Edward Asner, Brenda Vaccaro and Richard Dysart. It is set in the small office of a newspaper publisher in 1778. At least a dozen times, the actors are moved into groups to simulate various acoustical environments. In the "newspaper office" Fred moves in a desk and chairs, and then surrounds the actors with reflective glass baffles to get the sound that would have occurred in such a small space. An opening is left that simulates a door so that the actors can enter the space when the script calls for it. Microphones used are two Neumann U47s and a Neumann TLM 170 at the "door." The camera crews hate it.

December 8th, 9 a.m.

The day begins with a very big and important scene starring James Earl Jones. David tells Fred, "I want Mr. Jones to sound as if he is on a stage in a small theater giving a speech to the audience. With him will be several other voices that need to sound as if they are ghosts." "No problem" is Fred's reply.

The studio is transformed into a small theater with a stage made up of baffles to form the backdrop and project the sound forward. Two U47s are positioned in stereo a few feet in front of James Earl Jones so that he may move between them, giving the feeling of stage movement. To get the sound of the theater, a Sony C48 is placed up near the ceiling, about 10 feet up and 10 feet back from Mr. Jones. The other actors are positioned in an isolation booth with two TLM 170s, also set up in stereo. The scene is recorded on a 4-track Ampex ATR 104 with Dolby A. The stereo image for the ghosts are on tracks 3 and 4, while tracks 1 and 2 have the pair of 47s and a P48, combined to form a stereo image of a small theater. No reverb is used on the theater sound, and the result is as if one were listening from the fifth row center seat.

This is the last day at Evergreen. "We chose Evergreen because of the large and varied spaces we could create acoustically. I like the room for recording very much," says Fred. "However, for the type of post we will need to do on this show, I need to work in a room more geared to radio work. You know, one that has several sources available, such as multiple 2-track decks, a turntable, CD player, a DAT machine, and any number of devices already set up in the control room, all patched in and ready to go, all within fingertip reach." They choose Hollywood Recording Services.



Actor James Earl Jones rehearsing a take with author Norman Corwin.

December 9th, 10 a.m.

First day at Hollywood. The actors are gone, as is the press. A production meeting is held. Due to budget constraints, it is decided that since the music is already mixed down, the four reels of 2-inch 24-track from the sessions at Evergreen will be degaussed and used for the post production. "Some people would call it a risky decision, but the budget is very tight and we just can't handle the additional cost," says Judith.

After the airing date, cassette copies were to be distributed to all the members of Congress and the Senate, and the President.

This facility fits the bill of what Fred is looking for. With seven studios to choose from, and lots of editing to do before he can start on building the tracks, production begins in three different studios simultaneously with the selection of the final takes. While the assistant editors cut the tracks in the various rooms, David, Judith and Fred realize that in order to finish on time, they must begin building the show on the 24-track before the editing is complete. This is accomplished by having a 4-track and a 2-track dedicated to the select voice tracks, and an additional 2-track with the music beds in the control room of Studio 1, so that the select voice tracks can be transferred to the 24-track without changing reels, or machines, thus speeding up the process.

As a scene is edited, it is rushed into Studio 1, where the tracks are transferred to an Otari MTR-90 at 30ips (non-Dolby), the music is inserted, and sound effects are added or are created by Foley.

"One of the joys of this kind of production," says Fred, "is that you must create many of the effects. How many recordings of an 18th century blacksmith shop do you think exist in a sound effects library?" David joins in, "This is where being a good scavenger comes in handy. We dug around in the alley behind the building and found this big metal stake that, when you hit it with a metal hammer, sounds just like the anvil we need!" Foley is so much fun!

December 10th through 12th

These three days blur together. Judith is doing a great job at keeping the pandemonium from encroaching into the control room, while Fred keeps everything flowing smoothly from reel to reel. "If I let the material determine the production style, it will all just fall together. For example, if I put down the voice tracks in a scene driven by the music, it won't work as well. But if I put the music in first and pace the voice tracks around it, it will sound just great, and will be much easier to mix. Sound effects can do the same thing. Sometimes you just have to build it a piece at a time to get the timing right. Even if it puts you through major changes while you are doing it, it comes out much better that way, instead of just forcing it by putting the effects in later."

December 13th

Spare time no longer exists. The mix begins at 3:30 a.m., with the production staff having worked solid since 10 a.m. the day before. By 8 a.m. the mix is not yet finished. Although it is scheduled to uplink at KUSC in downtown L.A., someone would have to leave with the finished tape at 9 a.m. to get it on the bird by 11 a.m. West Coast time. It is decided that if the schedule is changed to do the uplink directly from Hollywood Recording's facility instead, this will buy some much-needed time. The mix is completed at 10:15 a.m. Final editing begins to "nip and tuck" it into its final form. The satellite feed is at 11 a.m. The last reel is being edited while the first reel is on the bird. Talk about a photo finish!

December 14th

Everyone sleeps.

December 15th, 8 p.m.

Phil Proctor's house is the site of the listening party. Many of the actors, as well as Judith Walcutt (producer), David Ossman (director), Fred Jones (engineer) and, of course, Norman Corwin (writer) are present. As the program comes over the air, we all gather around the radio to hear the final product as it is broadcast. Looking around, one gets an inkling of what it must have been like in the days before TV. Radio drama truly is theater of the mind. ■

Actor James Earl Jones discussing interpretation with author Norman Corwin prior to recording.

Acknowledgements

Without Judith Walcutt, this project would never have been completed. She went into production $30K shy, expecting to cover that amount out of her own pocket if need be. It was in the middle of the production that she managed to come up with the balance of the financing in the form of a last minute grant from the National Endowment of the Arts. This is truly an example of an American citizen who passionately believes in the Bill of Rights!

Original funding was provided by the PEW Charitable Trusts, The Commission on the Bicentennial of the United States Constitution, The Ahmanson Foundation, and the American Booksellers Foundation for Free Expression, along with the above mentioned grant from the National Endowment of the Arts.

Author and voice talent Ray Bradbury preparing his portion of the show while engineer Fred Jones makes last minute microphone adjustments.

The People Who Made It Happen

Norman Corwin (author) age 81, is recognized as the Golden Age of Radio's finest writer. A member of the American Academy of Letters, Corwin was the first writer to be inducted into the Radio Hall of Fame. In addition to scripts for more than 24 radio series and specials produced from 1938 to 1955, Corwin has also written extensively for film and television. His screenplay for "Lust For Life" earned him an Academy Award nomination, and "We Hold These Truths" won Corwin a 1942 Peabody award.

Judith Walcutt (producer) is the founder and executive producer of Otherworld Media, a leading producer of radio drama in America today. She was the executive producer of the Grammy-nominated "The War Of The Worlds 50th Anniversary Production," and produced and directed the international award-winning "The Door In The Wall." Most recently, she wrote, directed, and produced the American Public Radio special "Which Way Witch," starring June Foray.

David Ossman (director), the other half of the Otherworld Media team, has spent more than 30 years writing, directing and producing for the audio medium. As a highly acclaimed radio theater writer and producer, he has been associated with the American Repertory Theatre and The Acting Company. Ossman adapted and directed "The War Of The Worlds 50th Anniversary Production," which received a Grammy nomination. As a founding member of the legendary Firesign Theatre from 1966 to 1981, Ossman co-created 15 comedy albums and performed on stage, radio and television. His 1973 solo album from Columbia Records was nominated for science-fiction's Nebula Award.

Fred Jones (engineer), began his career as a radio personality and production director in 1971. Since then, he has engineered many live radio broadcasts, with artists such as Loggins and Messina, and has recorded such well known artists as the Manhattan Transfer and Stan Freberg. He has also served as the engineer and producer on many of the Firesign Theatre records, one of which received a Grammy nomination. Among his many credits are sound recordings for thousands of commercials and television shows, receiving essentially every major advertising award (some more than once).

Closing Ceremonies at the 1992 XVI Olympic Winter Games

By David Scheirman

Audiences for "mega-events" such as the Olympics come with a certain sense of expectation, an unspoken desire to see and hear things that have never been experienced before, to be wowed by sight and sound, to be not only entertained but drawn into the group experience of participating in a spectacle. While millions watch at home on their television sets (and listening to broadcast audio), a privileged group numbering in the tens of thousands actually experiences the event live. It is to reach these listening ears, at events such as the large-venue ceremonies at the Winter Olympics, that some of the most critical situations in sound reinforcement exist.

Whether it's sound for the Olympics, a centennial celebration of a major city, a large-audience address by a political leader or religious figure, or another type of special event, it is at functions such as these that the envelope of existing audio technology gets stretched. Sound designers and system suppliers realize that the whole world is watching (and listening!). It is at this type of special event that the fine line is walked between absolute, proven reliability of known technology and cutting-edge, never-been-done-before risk taking with new technologies.

For the opening and closing ceremonies in the 34,000-seat open air stadium in Albertville, France, in February 1992, a unique combination of new audio system input and output technologies came together. Programs that featured dancers, skaters, bicyclists, aerial artists and other entertainers were underscored by a custom-developed playback soundtrack; narrators suspended high above the audience in moving ski-lift chairs used wireless mic systems to make their points in both French and English. A monolithic, 360° coverage mast-mounted array in the center of the stadium stage, supplemented by cable-suspended overhead delay arrays, was used to bring full-bandwidth sound to the packed venue. Bose S.A.R.L. (Bose France) was contracted to supply sound reinforcement services to Telema, a European events production management firm that was in charge of all technical production aspects of the ceremonies.

David Scheirman is R•E•P's live sound consulting editor and president of Concert Sound Systems, Julian, CA.

Of primary importance to these ceremonies was the use of digital sampling, storage and editing technologies for the creation of audio soundtracks to accompany the multitude of visual entertainment spectacles during the programs. Whether the sound of wind, birds and insects, jet planes, accordians, ticking clocks or bagpipes was required, sound designer Elain Francais made use of a sophisticated audio editing and playback system to create and present an auditory feast. Used today for everything from television scores to film soundtracks, the Akai DD1000 gave Mr. Francais the tool he needed to assemble complicated, time-cued music and sound effects segments.

Three of the DD1000 optical disk recorders were linked via a Macintosh computer and Akai's DD-QMAC editing software, and their digital outputs were fed into one of Yamaha's new D-1000MC digital mixing consoles. Each DD1000 can record and store up to 90 minutes of material with instant random access. Up to a 44.1kHz sampling rate is offered on the removable magneto-optical disk, each of which holds 650Mbytes of digital audio information.

> Sound designer Elain Francais made use of a sophisticated audio editing and playback system to create and present an auditory feast.

Watching and listening to the rehearsals for the closing ceremonies in the booth with Francais, I could not help but be astonished at the speed and precision with which subtle timing, sound and cue changes could be made in response to needs from the show producer. The ice skaters needed the tail of the violin note to extend just another second as they made their exit from the ice. No problem. The pause between the accordian chords needed to tighten up a bit for the 100 clowns on bicycles as they made their turn in unison. Easy.

The output of the Yamaha digital console, that functioned as the main musical program control desk for the program, fed a Yamaha PM-3000-40, used to distribute sound to the many different speaker zones and to handle traditional microphone inputs, including 22 of AKG's new battery-powered RF mic systems.

The AKG wireless systems, operating in the 800-900MHz band, were picked up by a mast-mount antenna located high on the stadium rim; a single 70MHz line carried 12 diversity channels in to the receivers, located in the sound control booth. "Setting up the system this way, we can go several hundred meters from the main antenna on coax cable with no signal loss," said Hans Radda, whom I met onsite as I examined the systems. Radda, in charge of export sales and marketing for AKG Acoustics in Vienna, Austria, advised that currently, other similar systems can carry no more than six diversity channels in such a narrow bandwidth. The new AKG system claims to double that capacity.

The wireless mic systems found numerous uses. Miniature C410 mic elements were attached to the blades of ice skaters, with transmitters hidden in their costumes; the icy-crisp sound of blades meeting ice was then panned around the large outdoor stadium, following the skaters around the ring via placement in one of nearly two dozen signal-delayed overhead speaker arrays.

Main speaker systems for this round open-air facility, temporarily erected on a soccer field, were supplied by U.S. Sound (New Jersey). The company has been quietly at work for the past five or so years developing high-tech, horn-loaded integrated loudspeaker systems for use in large-scale sound reinforcement. Designer Cliff Henricksen (formerly with Electro-Voice and Community Sound) has put large-format compression drivers to a good advantage. Combining new, patented driver technologies with advanced enclosure-building materials, U.S. Sound has come up with a lightweight, powerful system that is exceptionally articulate.

"A primary criteria here was a low weight factor," said Henricksen. "The center mast that the 360° system hung from supported a very complex lighting system, along with being the central anchor point for the Kevlar suspension cables used for hanging production rigs, actors and even our own delay arrays. Traditional sound system packaging would have been just too heavy. Our entire 64-box composite-panel speaker enclosure complement weighs only about 5 tons, and gave full-bandwidth sound for 34,000 people in this application."

U.S. Sound has already designed and installed new systems in large American facilities such as Madison Square Garden and the Omni in Atlanta. The lightweight, highly-directive arrayable enclosures are an interesting and effective design trend for large-scale sound reinforcement. The company is now working on roadable versions of its products.

While it may be a while before the next show I mix uses magneto-optical disk recorders for giving me the sampled sounds of an accordian, and it may be a while before you strap RF mics onto the blades of ice skates, we can see that special events such as the Winter Olympics do help to bring out the best in new technology. When the best economic and intellectual resources are brought to bear on audio system design problems for "mega-events" such as this one, exciting new tools and technologies are often the result. ■

CUT ALONG DOTTED LINE

R•E•P Studio Handbook

Tape-path failures show themselves as problems with frequency response, signal-to-noise ratio or high-frequency instability. The physical balance of a tape-path is delicate, so avoid altering components or alignments unless you have first isolated a problem as mechanical. In other words, your first step in troubleshooting a suspected bad head or bad transport alignment should be to check the audio path.

ISOLATING MECHANICAL PROBLEMS

The best way to verify that a machine's playback electronics are performing up to snuff, and therefore not the source of trouble, is to employ the flux-loop method described last month. On the record side, you can measure bias/erase frequency and distortion, and obtain some comparative indication of amplitudes, using a flux-loop as a pick-up, directly off the record or erase head, in record mode with no signal input.

Amplitudes should approximate those from known-good units of the same model ATR. Because the pick-up point is the head itself, these tests are sensitive to all failures between the bias oscillator up to and including the head windings. Check record electronics with an oscilloscope at the output of the record amplifier. A low-pass filter to block bias allows measurement of audio distortion and frequency-response. If you encounter no problems performing these tests, yet the sound quality is poor, your machine has a tape-path failure.

THREE FAILURE SITES

Troubleshooting tape-path problems naturally subdivides three ways: tape guidance and tensioning; head alignment; and head surface condition. Proper tape guidance and head alignment ensure that the tape and head-gaps bear correct relative positioning. Head surface condition and tape tensioning are critical since it is the appropriate micro-geometry of the tape-head interface that makes accurate magnetic recording possible.

Because heads are finely adjustable along or around multiple axes, while tape paths are relatively inflexible, logical troubleshooting or alignment always progresses from guidance to heads.

M. Raymond Jason is an electronic engineer at National Public Radio in Washington, DC.

TAPE GUIDANCE

The plane of the tape should be consistently perpendicular to the deck plate: there should be no warping, in which the top edge of the tape is not directly above the bottom edge (given a horizontal deck-plate). Nor should there be tilting, in which the direction of tape travel diverges from perfectly parallel to the deck plate. Scraping sounds or oxide shedding indicate that the tape is being forced to tilt or warp. Careful observation of reflections off tape from a flashlight are useful: you should see no ripples or curls.

Troubleshooting tape-path problems naturally subdivides three ways: tape guidance and tensioning; head alignment; and head surface condition.

Most 2-track ATR transports use non-shimmed fixed guides at the entrance and exit of the head stack. Heights of all adjustable elements in the path should be set relative to these guides, starting with the supply reel turntable. Your machine's service manual should provide details. If you alter the tape path vertically, you may need to replace or rotate fixed guides possessing wear grooves, and you may further need to relap or replace the heads.

Finally, ensure that the play tensions are correct as described by your service manual before moving on to head alignment.

HEAD ALIGNMENT

A line drawn through the gaps on a head should be parallel with the plane of the tape (correct zenith), perpendicular to the direction of tape travel (correct azimuth), and centered within the area of maximum pressure (correct wrap). Further, the gaps should be equidistant from their respective tape edges (correct height) and the tape's bend angle around the head should be optimized (correct penetration).

The time to perform full head alignment is upon purchase of a new ATR or upon head replacement. After a day or two of use, many heads already contain wear patterns that make further alignment, particularly height and zenith alignment, difficult or impossible. Check zenith visually, using a flashlight, relative to the fixed guides or capstan and from head-to-head. Check record-head height by recording 100Hz at standard level, then "developing" the tape using à magnetic developer (liquid-suspended iron powder) and a microscope: the tracks should be equidistant from their respective edges. Correct the height if necessary. Then check play-head height by recording 5kHz, flipping the tape (exchanging reels) and checking for equal level on the outside channels, or both channels on a 2-track.

Check rough wrap by "painting" the heads with a non-permanent marker, then playing several minutes of tape. Using a microscope, if necessary, verify that the gaps are centered within the area worn clean by the tape. Set fine wrap for the play head by recording 20kHz at 15ips while gently adjusting play-head wrap for maximum output level.

Penetration should be the minimum required for good high-frequency stability: less than ±0.1dB of wobble on an analog meter. Azimuth is set, of course, with a standard alignment tape. On most headstacks these alignments are to some degree interactive, so you may need to iterate the steps in this section several times.

HEAD SURFACE CONDITION

The third critical factor in time-independent tape path performance is head-surface condition. There are two failure modes here: outright damage, such as a scratch or a dent; and wear. Normal wear, that accruing from use given correct tape-path and head alignment, results in much better long-term performance than you'll get from incorrect wear.

Normal wear usually entails the development of a rectangular flat spot centered over the head gaps. The gradual performance degradation is what you would expect given the compromised interface geometry: instability and overall reduced output at short wavelengths.

Problematical wear results from poor wrap, zenith, height, or penetration. In each case, the wear pattern differs from normal: either the pattern is not rectangular, is not centered over the gaps, or both.

Here are two quick checks of degraded head performance. Compare standard alignment-tape playback alignment to flux-loop alignment. If they differ by a decibel at 10kHz, relap or replace both the play and the record head. Additionally, observe the peak-to-peak instability, on an analog meter, of a 20kHz tone at 15ips recorded and then played back. If the wobble is greater than ±1/10 or ±2/10dB, your heads need relapping or replacement.

Damage causes immediate and obvious degradation of performance. Almost all damage is visually apparent without magnification, and generally requires immediate head replacement. Remember that when replacing record or erase heads you must re-align the oscillator circuit's frequency and amplitude, and re-align the bias traps.

Next month we'll examine wow and flutter problems. ■

Beam Me Up

By Rick Schwartz

With so many studios moving into audio-post and jingle work, it's surprising how few have phone patch or satellite capabilities. Maybe it's because so little has been written about the subject. If your studio is ready to boldly go where few studios have gone before (and find a new source of income), read on.

Phone patch or satellite capabilities serve a number of important functions, not the least of which is allowing a producer to direct talent at a faraway location. Live in Omaha, yet need a certain name brand Los Angeles talent for voice-over work? Get him to a nearby studio and dial him up. Hit the talkback button and it's as if he's in the studio.

You don't have to be Ted Turner to afford remote-recording technology. Many manufacturers sell single-line phone patches for under $500. A phone patch simply connects the talkback circuit in your console to a normal phone line. Although it's technically possible to build your own, there are several problems that phone patch manufacturers have already dealt with, including feedback, frequency response and the FCC.

NULL AND VOID

Let's start with feedback. As you know, the phone company transmits both party's conversations on a single pair of wires. To achieve individual level control of each end's signal, the phone patch does a 2- to 4-wire conversion. The host studio removes its own signal from the caller's signal for the board input using a hybrid circuit and the principle of phase cancellation. Carefully trimmed nulling circuitry will prevent feedback under normal conditions, providing you have connected the patch correctly.

In addition to preventing feedback, a good phone patch also includes send and receive equalization, send limiting, receive compression and receive expansion (to reduce line noise). Equalization is particularly important. Normally, frequencies below 400Hz are cut to reduce muddiness. A slight boost is sometimes added at 2.5kHz to improve intelligibility. Most patches include a band-pass filter to attenuate signals outside of the 300-3,000Hz frequency range. The Feds have their own EQ requirements. The FCC requires 18dB of high-frequency attenuation to prevent interference with telephone company equipment (crosstalk).

Rick Schwartz is a contributing editor to R•E•P and director of post-production at Music Animals, Los Angeles.

Finally, transformer isolation is absolutely essential for good common-mode rejection and protection from dc, present on the phone line.

PATCHING A PATCH

Connection is easy. Just split the line from the wall between the telephone and your patch using a standard low-cost modular adapter. The caller output on the back of the device is connected to an input on your console, to be bused to the talent's headphones and control room speakers. It's important to feed a mix-minus signal consisting of everything you want the caller to hear except his or her own voice, to greatly eliminate the possibility of echo or acoustic feedback.

THE SATELLITE EDGE

A "landline" phone connection works well when a producer needs to coach voice-over talent from a remote location. However, if your client needs to record or monitor a wide-range program source from a remote studio, you should consider supplementing your phone patch with satellite services. If you live in a secondary market, satellite services will give you an edge over other facilities, allowing you to provide your clients with big name talent

The single biggest reason to provide satellite link-up services is the time savings it offers your clients.

that would be prohibitively expensive if you had to transport them to your facility. The single biggest reason to provide satellite link-up services is the time savings it offers your clients. Next-day mail is too slow for projects that require several revisions. If your client has a tight deadline for an important presentation, satellite communications will allow them to make revisions on-the-fly to save time. As the saying goes, time is money. Most clients are willing to pay a little extra if they're up against a deadline.

SPEAKING WITH A BROKER

There are a couple of things you need to be able to transmit audio via satellite, including a phone patch and a high-grade dial-up phone line. Although it is possible to install your own dish, it's not feasible for most sites, because of the cost. Most of the time the dish is located off-site and connected to the studio facility using 15kHz audio lines installed by the phone company. These high-grade audio tie-lines sound pretty good due to carefu and calibration. From the recordi tie-lines are run to a satellite Landers of Landco Labs prefers to t self as a networker, because his com ally makes the connection to the re dio for you. Your broker will also inf about availability and transmission con Certain times of the year are busier than Although same-day bookings are generally sible, it's a good idea to book in advance to sure satellite availability. As satellite transm sion conditions vary, it's important to know t several times of the year that the sun interfere with reception for five to 10 minutes a day.

TIPS AND TRAPS

About the only thing a satellite broker doesn't do is call the remote studio and make arrangements with them. Finding a remote studio is easy. The Landco satellite network has studios all over the country, across the border in Canada and even Europe. Your broker will provide you with a directory of studios and contacts. You must schedule the session and pay the remote studio for all of their time and materials. Before you book a session, make sure you've been quoted a firm hourly rate from the remote studio. Next, ask if there are any extra charges or minimums. Since most large cities have more than one studio with satellite capabilities, you can shop around. Be prepared to learn some new terms. Some brokers use the term "plus-a-half," which is a reservation for additional time you think you might use. To avoid being charged for this time you must "goodnight" your session, by calling the broker when you are finished.

Monthly access fees are fairly reasonable and can be covered by the income from one or two sessions. Markups vary, but a 100% profit margin for the studio is common. Unless you request otherwise, it is assumed you will only need a 1-way mono feed. A 2-way feed is twice as expensive and unnecessary most of the time, because it's less expensive to talkback to the remote studio via regular voice-grade phone lines via a standard phone patch.

VOICE LABELS

Voice labels are a 2.5-second digitally recorded ID that identify a source studio with an announcer, followed by a 400Hz tone at +4dBm. They "sound" in the absence of audio to fill the void and identify the link. Most satellite lines carry this ID (developed by Landco) so that when you're listening to your incoming line and hear the ID of the studio or satellite channel you're supposed to be down-linking, you're ready to go.

Obviously to cover a topic as complicated as satellite communications in a column of this size is very difficult at best. If you would like to learn more about this topic, a good introductory book is "Digital Satellite Communications," by Try Tha. ■

Note: Landco Labs can be reached at 619-756-5056.

Open For Business

By Tim Sadler

This month marks the official launch of our new R•E•P information service. In addition to uploading all your favorite articles, columns, features and magazine departments, past and present, we'll present timely new product press release information (months before you see it on *anybody's* pages), technical notes and service updates on equipment from manufacturers (with their blessings!), job information, and equipment wanted or for sale, as well as comments from editors and contributors that didn't make it into the book. And this material will be available early, before the regular issue hits your mailbox.

R•E•P: On-Line lives on the CompuServe Information Service network. And since CompuServe is a 2-way interactive proposition, you'll have the chance to talk to the editors and regular contributors on-line, asking questions, suggesting topics for articles or offering to write one, uploading letters to the editor for publication in the monthly R•E•P magazine, etc. In fact, you'll be able to roll up your sleeves and take on the magazine, the industry, or any other pro-audio related subject at all. It's a great opportunity to provide editorial input to the folks at R•E•P directly. It's even a great way to send and receive info or technical data to and from other studios, manufacturers or friends in the biz. And each month, I'll report the activities, questions and concerns that evolve online on CompuServe, in this column.

And how do you get involved in R•E•P: On-Line? First, you'll need a membership to CompuServe and a modem (assuming you don't already have either). At the end of this article I'll tell you how you can get a free introductory membership kit from CompuServe, compliments of R•E•P. The offer for CompuServe membership allows you to log on and sign up the moment you hook up the modem. If you like instant gratification, you have got to try this!

In addition to the membership, you'll need the modem to translate your computer's digital output to the analog signal necessary for transmission over conventional phone lines. While modems are almost generic hardware, the software, and often the cable that comes with them, are not. You'll want to look for one that is specific to the type of computer you're going to be using it with. Mail order is a viable source for purchase, and a 2,400 baud modem with cable, software and extras should be $100 or less.

OK, what's a baud? It is simply the number of bits per second that a modem can "talk" to another computer's modem. 2,400 is the most ubiquitous standard available, and it's probably a data rate speed that will serve your needs very well and help control your costs, because most 9,600 baud (high speed) connections cost much more per hour. There are some other modem features that will increase the cost accordingly: Send FAX capability, Send and Receive FAX capability, as well as various data compression capabilities. All of these may have value to you, and we will be discussing them in future columns and on-line, but you don't need any of this to get up and running.

> It is a dialogue with the readers, and, most importantly, a dialogue among audio professionals around the world.

Assuming you already have a modem, have access to one, or want to run right out and buy one because of all the wonderful things R•E•P is offering to its technologically connected readers (and the boatload of other pro audio-interested CompuServe network subscribers), you can log on, and when you see a "!" prompt, just type GO MIDIVENDOR or GO REPMAG. In moments you'll arrive at the front page of the MIDIVENDOR Forum (our host location). Or type GO MAIL at the prompt, and leave a message for the editor, Mike Joseph, at the R•E•P editorial address 75300,3141 or me at 75300,3142.

Before we tell how to get the free membership kit, I'm going to ask Forum manager, Jim Maki to introduce you to CompuServe's Forum facilities:

"The CompuServe Information Service is pleased to officially welcome R•E•P magazine and its pro-audio readers to our family of over 930,000 active members. On the Forums, you will find many new friends from all corners of the world. Whether you are interested in how to get more information about something in the music business, want specific technical support on the latest studio gear, or want to try demos of the latest software, you will find fast help on-line. Our members range from hobbyists to working professionals, and you will find representatives from most of the top companies involved in the music industry on our forums daily.

Our libraries are filled with thousands of truly useful info, such as MIDI files, synth patches, samples, commercial software demos, updates, shareware, freeware and handy utilities. In addition, we have hundreds of helpful text files, including startup suggestions, hints and tips, product reviews, upgrade notices, new product announcements and trade show reports.

Our message boards allow you to post and read messages 24 hours a day, seven days a week. Questions asked of other subscribers or CompuServe staff are usually answered in a matter of hours! Help, suggestions and "conversations" with new found friends are only keystrokes away at all times, night or day.

Our "live" conference areas are for real-time communications with other members while online. Special guest speakers and industry representatives are regularly scheduled for conferences.

In addition to the software that comes packaged with the modems, as Tim mentioned, we have available special communications software designed for CompuServe, which can help minimize connect time and make your experience with CompuServe an enjoyable one. They include CompuServe Information Manager for DOS or Mac, Navigator for Mac, and TAPCIS for DOS.

As a CompuServe member, you can automatically download files, or grab messages at up to 9,600 baud rates, and read and compose answers off-line! No longer do you have to wander around the system looking for information while the clock is ticking. To learn more about these programs, simply post a message to any of our staff system operators (Sysops) and we will gladly point you to the right program for your computer.

Looking forward to seeing you all on-line," Jim Maki, manager, CompuServe MIDI Forums.

As time goes on we will archive message "threads" (continuing discussions) that address topics of interest so that those who join us later can go back and see where it all came from.

But perhaps the best thing about R•E•P: On-Line is that the magazine is no longer a monologue. It is a dialogue with the readers, and, most importantly, a dialogue among audio professionals around the world.

Sound interesting? Here is how to get a free introductory membership kit from CompuServe: In the U.S. or Canada, dial 1-800-524-3388 and ask for representative #232. In the U.K. call 0800-289-378, Germany 0130-37-32, or the rest of Europe 44-272-255-111. Outside the U.S., Canada and Europe call 614-457-0802. Be sure to ask for representative #232. We'll be looking for you On-Line! ■

Tim Sadler is president of IntraMedia, a media software consultancy and Sysop of R•E•P: On-Line. His CompuServe address is 75300,3142.

To receive your free CompuServe introductory membership kit courtesy of R•E•P, call 1-800-524-3388 and ask for representative #232. Tell them you're an R•E•P subscriber. R•E•P's editorial CompuServe address is 75300,3141.

Keep on Truckin' That Mobile

By John Rosen

Okay, I'm dreaming about a location recording job. It's a big concert and acoustically, the venue is tops. The band's gear never buzzes, breaks or gets out of tune, and the stage monitors are so good that the musicians keep asking them to be turned down. The house PA sounds better than a CD and the entire crew is knowledgeable, multi-lingual, hard-working and cooperative. Sounds pretty good so far, doesn't it?

The orgasmic performances of every song, coupled with an audience that makes pagan idol worshipers look like Sunday school teachers, motivates the marketing rep for Bozo Records into an animated frenzy. He feels that a live recording of this show would make Garth Brooks' sales minuscule by comparison. Armed with an imaginary budget and more enthusiasm than a born-again biker, he convinces the "suits" at the record label that this project is perfect. It's all so simple — what could possibly go wrong?

Those of us involved in location recording realize that any one of thousands of things could go wrong, usually do, and yet we persevere. The technical aspects are extremely sophisticated. The electronic and electrical interface is complicated. The amount of physical effort is staggering, the hours are exhausting, and the working conditions stink. But, all the torture is worth it when everything falls into place and the audio has the quality of a studio recording boosted with the energy and the dynamics of a live performance.

Is there any clue to what makes for a smooth mobile recording project? Well, good communication is the one and only pathway to great location recording. For starters, pre-production communication between the mobile recording company and the many artistic, business and technical teams is essential. You must realize that the mobile recording truck is a small link in an elaborate production, and all facets of the production team must work equally well. A good techno-artistic environment means the talent has a decent chance for a good performance. Let's face it — if it isn't happening on stage, it doesn't matter if it's recorded anyway.

John Rosen is president of Fanta Professional Services, Nashville.

It's a fact of life that the members of the regular concert production crew rarely look forward to the appearance of a mobile truck. Usually, they don't even get extra pay for the additional time and work required by the interface to their normal audio equipment. But, if the regular production crew is prepared for the additional work, they can be of immeasurable help. If they don't help, you might as well take the mobile truck back to the hotel.

The production manager and the entire crew greet you with open arms. They cheer, they hoist you onto their shoulders and carry you off to the post-show parties.

CALLING THE SHOTS

There is no typical sequence of calls and faxes when a location audio truck is involved, but we'll go over some of the possibilities that might occur in a TV shoot that is broadcast live, with multitrack recording for a post-produced show.

First, there is a call from the customer that sets up the mobile truck for the project. Who, what, where, when and how are the basics of the call. The customer may know how to contact some of the production team players, but he may need some help in finding out about the audio road crew associated with the talent.

Next, you will deal with the line producer of the project, the TV director, the TV technical director, and the TV mobile truck production contact. Analyzing all the newly acquired knowledge sets you up for the next communication salvo. This is a very sensitive series of calls and the producer needs to know in advance that you're going to take care of business. For some of the participants, this may be the first clue that the recording truck will be involved with their regular concert production.

Honesty and diplomacy usually get things off to a good start. A slip-up during the first communications could result in a very bad first impression. Almost all the people in this group have previously had to move projects along to the next level. The random sequence of questions and the ability to catch on and modify the interrogation will form the first impression with the production crew. That first impression will get you to the loading dock. If they sense that you're going to be a pain in the fanny (meaning they don't think you know what you're doing), you better believe that you will become a pain in the fanny. Bear in mind that the time available to learn the proper jargon and style of each road crew could be described as ranging from none to instantaneous.

To get a better handle on the situation, following is a list of a few of the participants you may be dealing with:

- Talent management company
  Boss and assistant
- Road manager
- Stage/production manager
- Sound company
  Boss and assistant
  House mixer
  Monitor mixer
  Stage tech and manager
  Electronic tech
- Recording engineer
- Audio project producer
- Remix studio
  Remix engineer
  Tech contact
  Logistical contact
- Location or venue
  Main contact
  Union technical contact and/or electrician
  Load-in and load-out union personnel
- Promoter
  Boss and assistant

This list represents the basic parties involved in the communication network required for standard medium-to-large venue touring acts. The coordination of all the contacts is a lot of work and there will be about 20 follow-up calls. The key players will require at least five recommunication calls. These many phone calls are absolutely necessary to avoid confusion and increase efficiency. For inspiration, you should keep in mind that each call is a sales opportunity for your truck, and many may provide valuable technical and logistical tips. Phone calls are cheap, while disasters are expensive.

Let's forget about the potential nightmares and return to that dream job I described earlier. After the concert, the very same marketing rep that got this project started is there standing next to the remote truck. He takes you aside and walks you over to the production office. The production manager and the entire crew greet you with open arms. They cheer, they hoist you onto their shoulders and carry you off to the post-show parties. You're a hero once again! Dream on ... ■

The Society of Professional Audio Recording Services is the industry's best source of business information. For details on activities or membership, contact SPARS at 4300 10th Ave. N., Lake Worth, FL 33461; 407-641-6648; fax 407-642-8263.

THE YAMAHA YPDR601

HANDS ON:

CD RECORDER

By Fred Jones

First of all, in this magazine, I am primarily a writer. And as such, I must report to you, the reader, all of the cold specs, hard facts, and emotionless figures about the product I am discussing.

However, in my other identity as a recording engineer and a producer, I am just as intrigued as you are to tear into a new piece of equipment to see what it can do for me in the studio, and to hell with what the specs say.

Please forgive any tendencies I might show in this article toward being schizophrenic.

Fred Jones is a free-lance engineer, producer and writer, best known as the engineer/producer for the legendary comedy group, The Firesign Theatre. Among his many credits are an uncountable number of commercials and TV shows. Fred has won almost every advertising award, and several of his recordings have been nominated for Grammys.

WHAT IT IS

The Yamaha YPDR601 and RC601 is an "Orange-Book" compatible CD recorder, which means you can use it to produce audio only "Red-Book" compatible CDs that can be played on any CD player, once a TOC (Table of Contents) has been placed on the disc.

A brief explanation about the various "color" book standards: About 10 years ago, when the CD audio format was established, the brilliant folks who thought it up decided to write down the exact specifications required in the coding, recording protocol, manufacture and production of CDs, and publish them as a guide for all of us to follow. It was called the "Red-Book Standard."

As time went on, various other versions of the CD format, such as CD-ROM, CD-I, CD-V, etc., were proposed, and these were given names such as "Green-Book" and "Orange-Book." The "Orange-Book Standard" covers the "WORM" or "Write Once Read Many" type of CD recorder discussed here. However, I don't recommend you reading these books for pleasure, as they are very dry. I would wait for the movie version.

Basically, the "Orange-Book" states that the CD recorder must produce a disc that can be read by any standard CD player that conforms to the "Red-Book Standard." Does this mean that they are trying to mix apples and oranges?

Although I do not consider it to be a drawback, you should be aware that the YPDR records audio-only CDs and as such it cannot record CD-ROM, CD-I or CD-G discs. Maximum recording time is limited to the disc size. Only 63 and 74 minute discs are available at this time. One RC601 remote can control up to seven YPRD 601s, so you could outfit a small dupli-

SPECS AND DESCRIPTION

Distributor:	Yamaha Corp. of America 6600 Orangethorpe Ave. Buena Park, CA 90620 714-522-9011 FAX: 714-739-2680
Model:	YPDR601 Professional Disc Recorder
List Price:	$13,980 (with RC601 Remote Controller)
Features:	Records Red-Book-compatible CDs on optical WORM discs Records up to 63 minutes of stereo program Unique recording mode permits playback of unfinished discs on any player Balanced XLR analog I/O, +4dB/– 9dB Digital I/O via AES/EBU, SDIF-2 Automatic transfers from DAT and time code sources RC601 Remote Controller
Max I/O Levels (analog):	+24dBm/+18dBm
Dimensions:	17″×5.8″×15.75″
Weight:	30.9 pounds ■

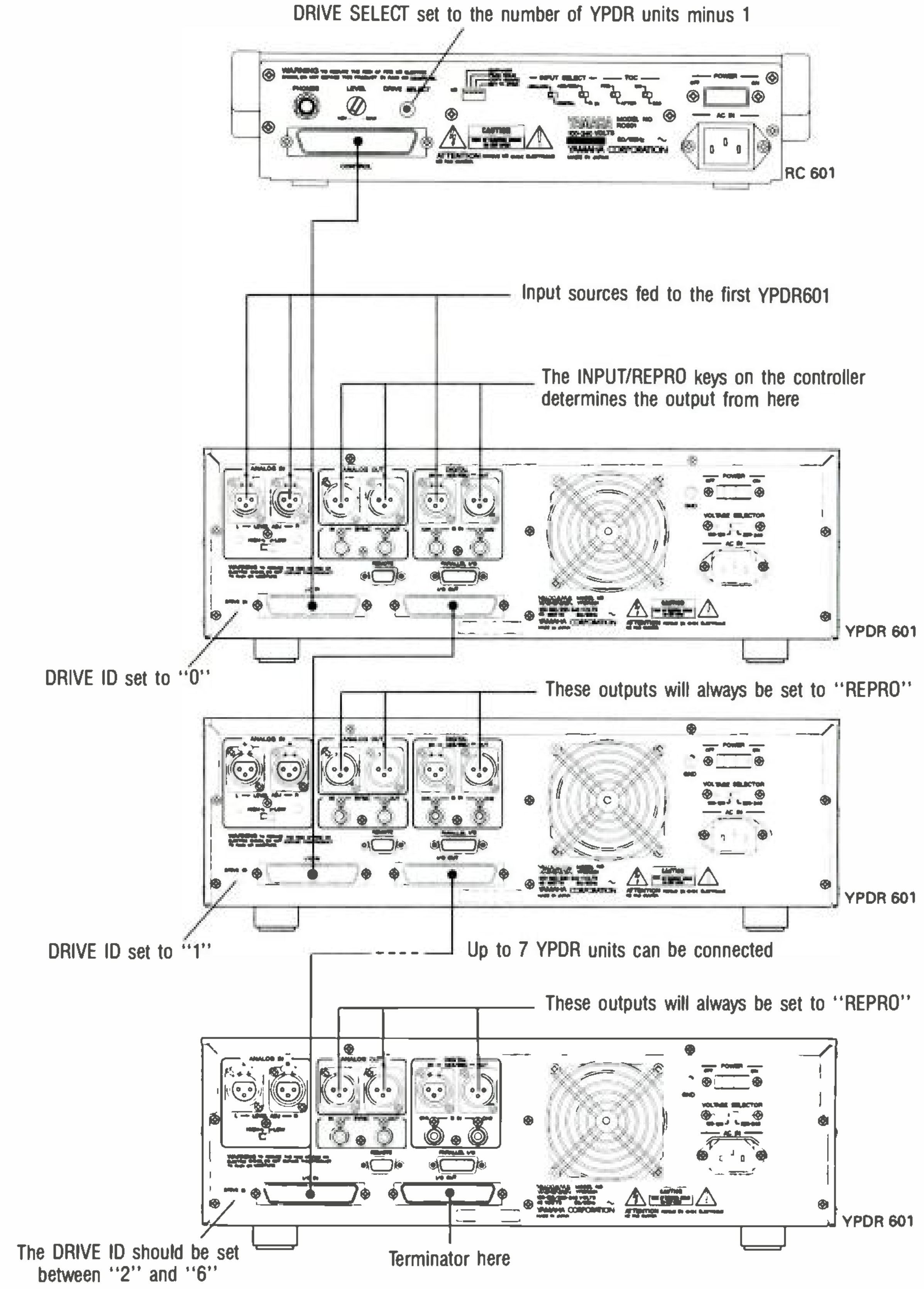


Figure 1. 50-way cable carries digital audio as well as information. Up to seven recorder units may be "daisy-chained" together.

cation facility quite easily (See Figure 1).

THE FRONT PANEL

On the YPDR601, the only control is for the disc drawer opening and closing, so you can rack-mount this unit out of sight, with full peace of mind that you do not have to keep looking over at it to see what is happening. The RC601 remote has all of the metering and alarm status indicators on it, and it will let you know what is happening to each of the units you have connected in the system.

All pertinent switches to select and control the unit's input functions, copy prohibit recording and emphasis on or off can be found on the rear of the RC601 remote, as shown in Figure 1. The various indicators on the front panel tell you if the input selected is analog or digital, the status of the copy prohibit flags and whether or not you have selected emphasis.

The display operates much the same as any standard CD player, in that it shows you absolute time, program time and remaining time. This display is also used to notify you of any problems using a series of codes, such as "01," which would mean "word clock has been lost." These codes are all explained in the manual, so I won't describe them here.

You can increment the track or index and select the input or reproduce modes from keys on the front panel. Meters are 24-segment bargraph-type, with a peak hold function that can be selected via a switch on the rear panel. The rehearsal button tests the cue point of the source machine (when it is connected) and will trigger the source to let you see if it is located to the proper place to allow for the two second pre-roll.

I would like to compliment Yamaha on the well thought-out input, output and external control connection scheme. The YPDR can accept +4dB and -10dB analog inputs, or input sources in AES/EBU or SDIF-2 standard digital formats. The signal can be monitored from either the analog or AES/EBU connecters or via the headphone jack on the rear of the RC601 remote unit. These connections can monitor from either the input source, or from the reproduce confidence output (more regarding this later).

External control is via an industry standard 9-pin serial interface provided for remote start of source machines. A parallel I/O interface also provides external control of the transport functions, including track and index points. Word sync is also available. All of these connections allow you to easily compile a disc from many different sources while you operate the YPDR on a "stop-start" basis, for example, the first track from an analog source, then one from SDIF-2, etc.

SPECIAL FEATURES

One of the best features of the Yamaha YPDR-type of system is the ability to allow you to partially record on a disc and play it on another CD player, then at a later date go back and add more things to the disc. But there are strong limitations governing this flexibility. As you are probably aware, the TOC contains all of the locations of tracks, index points, total time and time of each track. Until the TOC is recorded, a disc cannot be played on any CD player other than an "Orange-Book" compati-

PERSONAL OBSERVATIONS

I have been wanting to play with a "Red-Book" compatible CD recorder ever since I learned that they were about to appear on these shores. Most engineers and studio owners want them so they can make CD "refs" to take home, like they used to do with acetates. Not me. I had the rather strange idea to use them as a primary recording and source machine for post production. My experience as an engineer/producer does not come primarily from music recording, but from what most people would now call "media recording" or "post production." It was only after doing this type of work for many years that I did any music recording.

As a production director in radio, I was called on to produce many types of things. I learned quite early in my engineering career how to "spin" things in on the fly and to do production using turntables, cart machines, and 2-tracks, at the same time reading the copy and mixing it all "live."

One of the draw backs of analog tape (which we are all painfully aware of) is the time it takes to search for something. I know I have wasted many hours running through reels counting the beeps, listening to slates to find an elusive sound effect or voice track, only to find that I need to change reels and keep looking. Thank goodness for sound effect and music libraries on CD!

Many years ago, I dreamed that there might be a way to record tracks in such a manner that would allow instant random access to the tracks. I bet you must be thinking that I am in love with the hard disc workstation systems that claim to make this a reality. But I am not. If this surprises you, then just do a session on one and imagine how I hate seeing "this may take awhile" appear before me while the computer decides what to do with the numbers. I will admit that these devices have their place. But let me explain my original idea a bit further.

When DAT became available, I found that it was close to what I wanted in terms of being able to put an index point at the beginning of a take, and, when the client wanted to hear take #23, I just punched-in #23 and it played. However, as anyone who has ever tried it, DAT is not very easy to "spin" things in with and still have any accuracy. However useful, this was not my "dream solution."

When I learned about CD-R, It gave me the wild idea of doing a session direct to the CD. The advantages would be instant random access to the recorded material, and a virtually non-destructive storage media. Although editing would have to be done outside of the CD-R (as it would be for DAT), I thought "this might be it."

Continued on page 60

I would like to compliment Yamaha on the well thought-out input, output and external control connection scheme.

ble machine. Only CD-R machines currently qualify. In other words, without a TOC, a disc won't play on your standard CD player.

But the Yamaha thoughtfully employs two methods of recording the TOC, one to allow you to play tracks before a final TOC is cut, and the other for final cutting, where the TOC details are known before you start recording. These are called the "pre" and "after" modes.

Let's take an example: let's say you work in a radio station and want to compile a bunch of spots to be used on one disc, and this disc has to be done right now so it can be used on the air, but all of the spots are not yet finished. Yamaha addresses this by offering a partial recording feature. Before you record any audio on the disc, you would "pre" record the TOC with 99 fixed-length tracks (either 10 or 30 seconds), sort of like "pre-grooving" a vinyl LP (what's vinyl?). Audio is then recorded onto these tracks.

If the material is shorter than the track, silence is automatically recorded from the end of the material to the end of the track. If the program material is longer than the prerecorded track, the audio continues past the fixed boundary, without interruption, into the next numbered track.

The advantage of the "pre" mode is you can take the disc out of the YPDR and play the partially recorded disc on any CD player at any time in the recording process. The drawbacks are that you limit the recording time of the disc to a maximum of 49 minutes and 30 seconds (99 tracks × 30 seconds), and there might be times that the tracks displayed would be confusing to you when they change in the middle of the program.

The other type of TOC recording is the "after" mode. This method of using the machine is applicable when you are assembling your recorded material, such as after a date in the studio, when you are making a "ref" to take home. Just as it implies, you record all of the audio first, while inserting the track and index information manually, putting the finalized TOC on the disc last. However, you cannot play it back on just any CD player until you record the TOC, an act which prevents you from adding anything further to the disc.

One drawback of all current CD recorders (including this one) is that unlike DAT, you cannot record or adjust the position of the tracks and index points later on the disc, separate from the process of recording the audio. Unlike DAT, CDs can't erase what's already there.

A creative solution to this problem might be for Yamaha to offer an output on their DAT machine to transfer the index point information to the YPDR, so that an engineer could make all the adjustments to the position of the tracks on the DAT format and a disc could be "automatically" produced with the correct location of this information.

There is one feature that manufacturers other than Yamaha seem to have left out on their CD recorders: the ability to record not only the 99 track IDs, but 99 index points for each of these tracks as well. This is critical to anyone involved with sound effects or post production work, and I am very pleased Yamaha decided not to omit this important feature.

OPERATION

The Yamaha YPDR operates as you would expect. It's sort of a hybrid between a disc cutting lathe and a tape recorder. If you are familiar with the process of disc mastering and the placement of "bands," you will have an easy time getting used to the YPDR. When you start to record and hit the track index button, you see the display count down the two second preroll. The trick is to try to be frame accurate in the relationship of the audio start and the track start; for disc cutting guys, it will remind you of trying to be dead-on at trying to match the start of audio and the end of the spiral. This will probably throw you the first time you use the YPDR. You will, however, get the hang of it quickly. If you consider this to be a potential problem, it can be solved using the remote triggering functions of the YPDR.

Surprising to me was the fact that the unit allows you to monitor just like a standard analog tape recorder with "source" and "repro" capabilities, but unlike analog, there is almost no delay of the signal when you switch between monitoring sources. Even more astonishing, the nice folks at Yamaha tell me that they use only one laser pickup to accomplish this, using reflection of the laser for the playback or "repro" function.

While I did not have the equipment available for measurement, in my personal opinion, switching between input source and disc repro was virtually indistinguishable in time and sound quality.

TINY LITTLE NIT PICKS

I like this unit very much. I (like most any opinionated person) have a few small items to point out that could use some improvement.

Why is there a power plug for the main unit *and* a power plug on the remote? Hey guys! Why couldn't you power the remote from the master unit? It makes placing the remote on the console a lot more tricky.

Another warning to the equipment squeamish is when you are recording the TOC in the after mode, the display goes absolutely crazy, flashing numbers like some sort of electronic pinball machine. There's no need to worry, as everything turns out all right in the end. But I might suggest to Yamaha that it would be nice to put some sort of notice in the manual to warn users of this strange effect, or, even better, mute or freeze the display during this mode.

The operation of this unit is quite honestly very simple and easy to learn. At times you may feel you are driving a truck, as the transport is slow, especially when it is searching for tracks in the playback mode. This is probably due to the fact that it was designed primarily to be a record machine. But seriously, how many of you would be buying this thing to use it as a playback deck anyway?

Overall, this thing is cool, and it sounds great. The A/D and D/A converters are very transparent to my ear. Read the sidebar starting on page 59 to find out what I *really* think of it! ■

Continued from page 59

THE SOLUTION TO ALL MY PROBLEMS

The opportunity to do this article just happened to coincide with a project that I was about to start, one that seemed tailor-made to try my idea out on. This project consisted of 20 or so hours of $^{1}/_{4}$-inch reels of voice tracks, sound effects and music tracks that needed to be edited down to a $1^{1}/_{2}$-hour finished master. Everything was of course recorded out of sequence, because scheduling conflicts.

It seemed that it would take literally days to go through and sequence this stuff just to get it into some sort of usable working order, so that I could start building it onto a 24-track.

Enter the Yamaha YPDR. It saved my butt. Since I had a CD player with a buffer memory in it that allowed me to be frame accurate in where I could cue up to the audio on the disc, I thought I might have the ideal way to handle this production nightmare. I decided that I would transfer all of the select takes directly from the various source reels directly to disc using the YPDR. Each voice track, SFX or music track would have its own separate ID track or index location on a disc. If I needed to do any editing within a take, I would do it before I transferred it to disc, but I did no leadering or other cleaning-up of tracks, things that I normally would do before I transferred them to a multitrack.

We decided to set up the sessions very much like a video edit date, with A and B rolls of voice tracks, separate SFX and music discs. I used a separate CD player for each disc to save even more time. When used, this method took *days* off the production schedule. All I had to do was call up the appropriate track from the disc, cue it up to the start of audio, and transfer it over to the multitrack. Sometimes I even flew in several tracks simultaneously using the multiple CD players.

This technique worked so well, I am not sure I will *ever* go back to doing sessions with the tried and true method you are all familiar with.

Again, some of you are saying, "Why not put it into a digital workstation?" In answer to this, the Yamaha YPDR saved me many days in the studio, and it never said "this will take a while." I admit it was not as glamorous as having a big computer system with a fancy name on it doing all the work. And granted, this method of production will not be appropriate for all types of work, but it was very fast and about as random access as you can get on a budget. Besides, do you use the same microphone or other piece of gear the same way for every project? As for workstations, I would love to have a system with enough memory to handle a project of this size, but it would cost well above the price of the YPDR. Sadly, the real world sometimes does not allow us to spend all the money on a project or equipment we would like.

However, in my opinion, when used in a method as described above, the Yamaha YPDR system at $13,980 is a cost effective solution to an age old problem. ■

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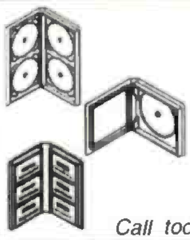

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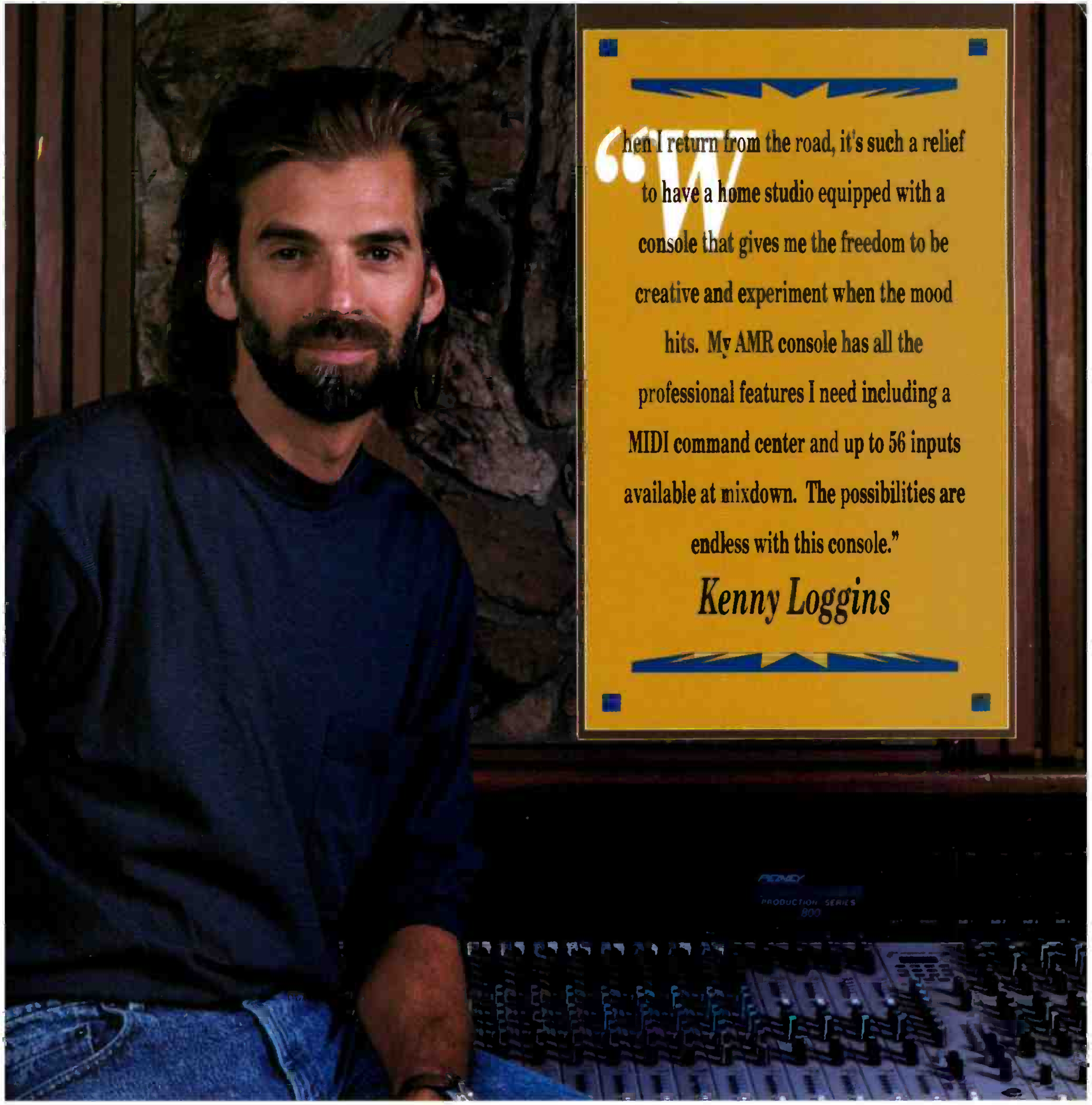

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