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Recording Engineer/Producer

The Applications Magazine for Audio Professionals

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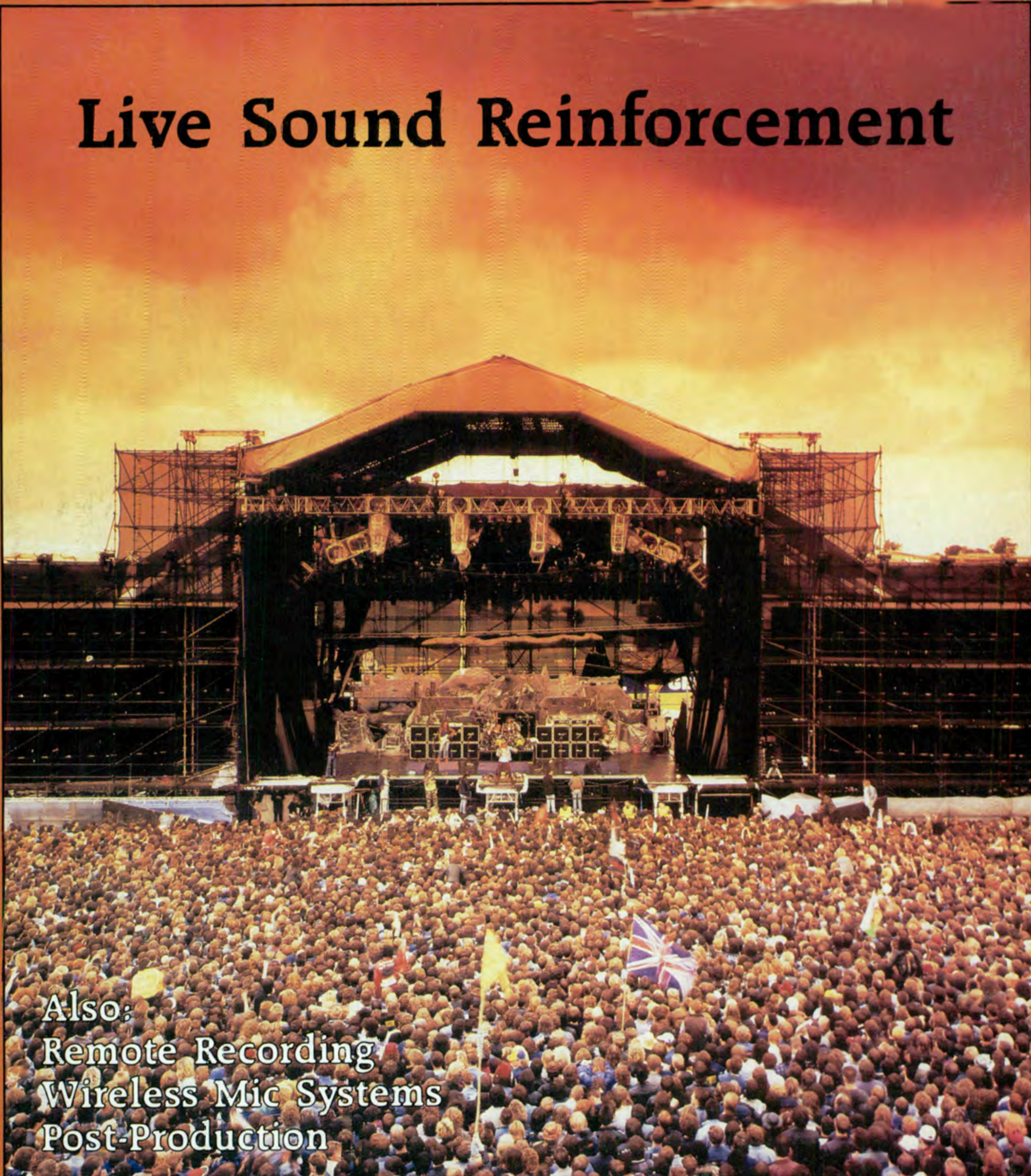

Live Sound Reinforcement

Also:
Remote Recording
Wireless Mic Systems
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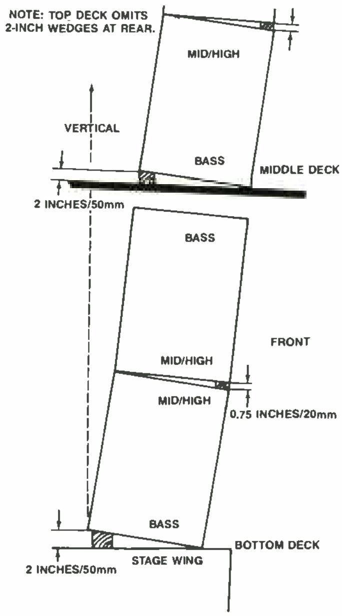
Live Sound Reinforcement



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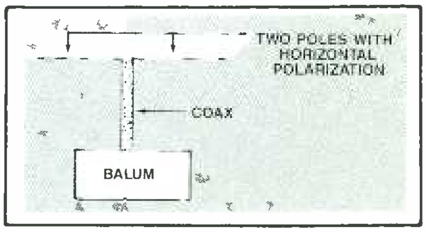

On the Cover

Iron Maiden headlines a heavy metal festival, featuring a Turbosound-based system, said to be the biggest single-source PA ever constructed.

Volume 20, No. 1

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Artists' expectations, engineering limitations, and other myths of digital recording.

Digital. The word itself conjures up visions of a totally perfect recording process where anything is possible.

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And how about *sound;* that nebulous, very subjective quality that is, for each one of us, the raison d'être? After all, even though we build what we believe to be the world's finest digital machine, the new 32-track DTR-900B, some audio engineers would stack our analog multi-track machines up against it in terms of sound quality any day.

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

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EDITORIAL

We Are Too Loud!

For the last 30 years or so, sound systems have steadily increased their capacity to deliver higher sound pressure levels. This evolution has continued to the point that today SPLs are reaching 130dB at the stage, 110dB in the audience and similar levels in control rooms. These are not peak levels, they are continuous program levels. It is, no doubt, an impressive display of power and technology that allows this much sound pressure to be delivered in a reasonably clean and undistorted manner.

At issue here is not the system's capacity to reproduce crushing levels, but one of responsibility. Who is in charge of the awesome program levels that accompany many live concerts? Is it the house-sound engineer, the artists, the promoters or maybe even the artist's management? I suggest that *ultimately* the house mixers have the control—it's at their finger tips. Therefore, the health and well being of every person in the venue become an added responsibility to consider.

We've all got a thrill from turning up the volume and *immersing* ourselves in the sound of our favorite act. But, we have reached the point that we can now do serious, permanent, physical damage to ourselves and others. [See *AES Journal*, September 1988, Volume 36, No. 9, pp. 686-691.] Some of the larger touring systems are capable of *sustained* levels at or near the threshold of discomfort (120dB+).

There are a few well-established guidelines for evaluating loudness. The Fletcher-Munson or equal loudness curves are familiar to most. They show that average human hearing is flattest at 80dB to 90dB SPL. This is useful because it tells us that, at these levels, the sound system requires less EQ to deliver equal power across a given bandwith. Another familiar table show us that the threshold of pain is 140dB SPL, and that the range between 120dB and 140dB represents the sensation of feeling—the level at which sound waves physically move our bodies. In this region, which many systems seem to be reaching and sustaining, the potential exists for short- and long-term damage.

To my knowledge, no long-term scientific studies have been done to examine the effects of exposure to the high SPL levels that are prevalent in the professional audio industry. Experts in acoustics and medicine can only speculate that continuous exposure to high-power sound systems will indeed cause detrimental effects. Certainly, in heavy construction and with airport ground crews, the conclusive evidence is that hearing impairment does occur after exposure to these industrial noises. Just because music is usually less random than industrial noise does not mean that it is any less dangerous.

Over the years we have conditioned audiences to expect, demand or maybe even crave more level. Is this the "bigger is better" syndrome all over again? Where does it all stop? If, indeed, we have slowly conditioned audiences to want louder concerts, is it not possible slowly to condition audiences, and ourselves, to want a little less painful—more listenable—levels in the future?

Another factor that works into this entire equation is that a very low-distortion program signal of 110dB SPL can be tolerated for longer periods, with less ear fatigue, than a highly distorted one. A distorted 110dB program signal may contain distortion peaks well into the threshold of pain. 130dB of pristine sound is not necessarily any less damaging, it is just more listenable. The acoustic-concussion lows and the dental-drill highs still exist.

I wrote this editorial to draw attention to the fact that something has to done to temper the enthusiasm of house engineers who operate systems that can actually harm us. If our industry cannot recognize this as an emerging issue, then we may be faced with some unacceptable alternatives.

One solution, which no one wants to think about, is government intervention. It's already happening in the U.K. The U.S. government is imposing restrictions on the levels that can be generated—at outdoor concerts in particular. Individual states, cities and municipalities have considered or are considering restrictions on noise pollution that fall under OSHA's umbrella of health and safety standards. Using this scenario, a regulatory agent will be looking over your shoulder, monitoring the sound levels and personally controlling the output.

Another way the government could get involved would be through manufacturing restrictions. That is, manufacturers may be forced to limit future developments in power handling. It would be pretty cumbersome to configure a 50kW system with 250W amps. You're probably thinking, "They would never limit us to only 250W amps." Oh, wouldn't they? Take a look at what the FCC did when they allocated a new group of frequencies for the wireless mic guys! [See "Coping with Wireless Microphone Systems" on page 52.]

What, then, is the solution? The answer is self-regulation. Stop and think—there is a real chance that OSHA or some other regulatory agency will step in and place restraints on the concert sound industry to control the levels.

Self-regulation is really quite simple. It starts with an acceptance and understanding of the problem. Of course, if you don't think a problem (or potential problem) exists, maybe a hearing test is in order. Next, get an SPL meter and check the levels at the stage, at the mix position and at various locations within the intended coverage area. Don't stop yet. Now go beyond the intended seating area 100 yards, a quarter mile, even a mile in all directions. Plot the energy dispersion coming from the PA. From this information, not only will you be able to monitor the levels at the mix position continuously, but you'll also be able to do a reasonably good job of predicting the levels elsewhere.

Redesigned coverage patterns can also help to concentrate higher levels throughout the intended listening area without having *all* the SPL coming from the stage. Because of the inverse square law (attenuation of SPL vs. distance from loudspeaker or sound source), levels at the stage are often pushed to the threshold of pain. This is necessary to deliver 100dB to 110dB levels only a few meters away. Maybe it makes sense to find ways of distributing the sound more evenly throughout the audience in order to maintain a lower, but more consistent, SPL?

If you have specific examples or experiences of government interference of this type, please write and share them with us. There's nothing like first-hand experience to convince the skeptics that this is a real threat to the live sound industry today—and possibly the studio tomorrow.

RE/P

Michael Fay
Editor

LETTERS

Phase shift experience

From: J. Russell Lemon, Carlsbad, CA.

Last week, I received my first copy of RE/P, and the letters on phase shift brought back many memories of the studies I did on that subject in 1974 and 1975. The interesting thing is that there is some truth in both sides of the argument.

In the early 1970s, I had access to a digital audio system that was developed for the purpose of testing real-time, digital signal processing techniques (software). Input was a digitally recorded source, and output could be recorded or monitored through a speaker. A high-speed computer that could create any amplitude and phase characteristic desired (within the limitations of the FFT used for the complex convolution), was used for the tests.

The results of those tests indicated that the ear was relatively insensitive to phase, and that hundreds of degrees of phase shift in mid-band (500Hz to 2,000Hz) were required to be heard. Similar results were reported 12 years later by J.A. Deer and P.J. Bloom in AES preprint 2197, called "New Results for Perception of Phase Distortion," which was presented in March 1985 at the 77th AES Convention in Hamburg, West Germany.

Many years later, I made the comment that the ear was relatively insensitive to phase, and Don Davis told me that I didn't know what I was talking about and to reverse my speaker terminals and listen to the difference. I did and found that with certain types of music, at a sufficient playback level, that there was a difference.

To further experiment, I connected my headphones to my computer and programmed several waveforms. As expected, symmetrical waveforms and their inverse sound the same. But a waveform with a large positive pulse is brighter than its inverse. Similarly, music from a solo instrument with a large unidirectional impulse is also the most sensitive to 180-degree phase shift. That is, human voice, brass and, of course, percussion sound brighter when the leading impulse is positive (speaker moves out) as it is originally created.

With a negative impulse, the edge is gone and the sound becomes dull. The effect is level-dependent and obvious at levels above 90dBA. The effect is masked if any component in the audio chain scrambles the phase. If the monitor system is phase linear, a microphone at the listening position will show a large positive impulse when human voice, brass or percussion are reproduced and the microphone output is displayed on an oscilloscope. If headphones are used, the oscilloscope will also show the same large positive peak at the phone jack.

An interesting test is to use two sine wave audio generators and set one to 1,000Hz and one to 2,001Hz. If you put one frequency into one ear and the other frequency in to the other ear, no beats are heard. But listening to both at the same time produces beats. If the ear was not sensitive to the phase between a fundamental and its "second harmonic," you would not hear beats.

After these experiments, I realized that my first tests lost their validity because of the 8th order elliptic filters used on input and output. These filters had a rolloff of more than 500dB per octave and a large resulting relative phase shift. In other words, the test differential phase shift did not significantly change the peak to RMS ratio of the already scrambled signal. Before digital, commercial devices to add second harmonics to a mix were popular, to add the edge back into the music when recorded on systems that lost phase information.

Perhaps the best way to summarize the above observations is to say that the ear may be unable to sense phase, but it is sure sensitive to the phase relationships between tones and that this sensitivity is level-dependent. But then, this can be said of any other non-linear device.

RE/P

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

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NEWS

RE/P to debut new departments

Beginning in the March issue, *RE/P* will debut two new departments: the "Engineer/Producer Index" and "Tracks."

The "Engineer/Producer Index" is a monthly listing of engineers/producers and the commercially released projects that they have worked on. It is open to engineers and producers involved in all types of audio production. Listings will contain the engineer's name, address and phone number, the five most recent projects performed and codes classifying the projects.

"Tracks" spotlights the studios and facilities dealing in commercial production and post-production, the two fastest-growing business categories in the industry. Listings will contain the facility's name, the five most recent projects that occurred at the facility, and the engineers who worked on the projects.

Beginning in this issue, postage-paid reply cards for the "Engineer/Producer Index" and "Tracks" will appear in the back of the issue. To be included in the departments, respondents should fill out the cards and return them to *RE/P.*

EV expands engineering facility

Electro-Voice has relocated its engineering department to a new 28,000-square-foot building, providing working facilities for more than 50 EV engineers, technical assistants and support staff. A large prototype fabrication, assembly and testing area is located at the facility, which includes a metal shop a wood shop, and a listening room. The building is about four blocks from EV's parent facility in Buchanan, MI.

Alan Watson, EV's director of engineering, said the facility would allow the company to expand its facilities and develop a greater number of products.

Ampex opens test facility

Ampex's Magnetic Tape Division has opened its new test facility in Opelika, AL, near the corporation's processing plant. The $2 million, 9,600-square-foot test center has eight separate labs for 1-inch video, cassette video, audio, instrumentation, plastics, physicals and microscopy. State-of-the-art recorders and signal generating/processing equipment will be used to analyze tape performance for a wide variety of formats and user conditions.

The lab will be used to evaluate competitive products as part of Ampex's Audit Program to obtain information about every format and brand of audio, video and instrumentation tape.

Passport to support NeXT system

Passport Designs has announced plans to support Steve Job's NeXT computer system, which was unveiled in early October and is said to be the first computer capable of acting as an all-in-one music workstation.

In addition to porting over the software titles, the company will design new applications that take advantage of the NeXT's multitasking and digital signal processing capabilities.

The NeXT computer is capable of generating a stereo digital audio signal at the standard 44.1kHz playback rate. Also included is a built-in coprocessor, a digital signal processor and a Motorola 6830 microprocessor running at 25MHz.

Aurora Systems files petition

Aurora Systems has filed a voluntary petition in the U.S. Court in San Francisco, and a plan of reorganization that provided interim financing, settlement of all outstanding claims, warrants to purchase shares in Chyron and acquisition of 100% of the new Aurora capital stock by Chyron.

These events were the results of several months of negotiations with Chyron, in which Aurora was unable to secure shareholder approval of offers by Chyron to purchase a majority equity position in Aurora. This incorporation of Aurora into Chyron will provide the former with the necessary financial, technical and marketing opportunities to profitably exploit the opportunities open in the TV industry, as well as to market their newly developed products.

News notes

ROR Audio Research is a new company that manufactures close-field monitors. Recording engineer Shimon Ron is the company president; the company's address is 161-14 Union Turnpike, Flushing, NY 11366; 718-969-3660.

Audio Animation has relocated to 210 W. Magnolia Ave., Knoxville, TN 37917; 615-544-0458.

J.L. Cooper Electronics has moved its warehouse and corporate offices to 13478 Beach Ave., Marina Del Ray, CA 90292; 213-306-4131; fax 213-822-2252.

Digital Audio Research has moved its offices from the San Francisco area to Hollywood. Lee Bartolomei has been promoted to regional sales manager. The company's new address is 6363 Sunset Blvd., Suite 802, Hollywood, CA 90028; 213-466-9151; fax 213-466-8793.

New England Digital has formed a new dedicated sales and marketing group within Harmon International, the European NED distributor. The group is intended to enhance the company's European market position as the continent moves to a unified marketplace in 1992. Mark Terry, NED's director of marketing, will lead the group.

MCA Records has purchased two Sonic Systems with NoNOISE software from **Sonic Solutions,** for use in mastering vintage recordings for CD reissue.

Imrex Financial Services, a financial services company geared toward the film, video, broadcast and audio industries, has opened a California office located at 5777 W. Century Blvd., Suite 265, Los Angeles, CA 90045; 213-670-1009.

At the AES convention, **BASF** honored two companies with its annual Inventors Award, given to companies deemed most instrumental in enhancing the quality, distribution and sale of music cassettes. Receiving the award were Concept Design, Burlington, NC, and WEA Manufacturing, Olyphant, WA.

Neve has signed a contract with Full Sail Center of the Recording Arts to teach applications on the V Series Console and the Flying Faders automation system.

Soundmaster USA has opened a New York technical support office, located at 120 W. 88th St., New York, NY 10024; 212-787-5832; fax 212-787-8888. On the West Coast, the company has installed a second Integrated Audio Editing System at ABC-TV in Hollywood.

Manny's Music has acquired Audiotechniques, a pro audio dealer based in New York.

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NEWS

People
The Society of Professional Audio Recording Services (SPARS) has elected new officers and board directors for the 1988-1989. The officers are **Bruce Merley,** Clinton Sound, New York, president; **David Porter,** Music Annex, San Francisco, first vice president; **Dick Trump,** Triad Productions, regional vice president/treasurer; and **Dwight Cook,** Cook Sound and Picture Works, Houston, regional vice president/secretary. Other regional vice presidents are **John Rosen,** Fanta Professional Services, Nashville; **Charles Benanty,** Sound Works, New York; Howard Schwartz, **Howard Schwartz Recording,** New York; **Charles Comelli,** Capitol Records, Los Angeles; **Pete Caldwell,** Doppler Recording, Atlanta; **Tom Kobayashi,** Sprocket Systems Division of Lucasfilm, San Rafael, CA; and **John Fry,** Ardent Recording, Memphis, TN.

Russell Palmer has been named president of Alesis Studio Electronics.

Phillip A. Phelon has been named executive vice president of Telex Communications.

Edward Allen "Al" Rogers has been awarded Ampex's Excellence in Manufacturing Award. He is the supervisor of instrumentation product engineering at the magnetic tape division's center in Opelika, AL.

Kelly Hannig has been named product line specialist at Gentner Electronics.

John St. John has been named president of 27th Dimension. **Fran Bell** has been promoted to vice president/secretary-treasurer.

Terence O'Kelly has been named director of national sales of audio/video professional products at BASF.

Joseph Kempler has been named technical director at Sunkyong.

As part of Lexicon's expansion of the Opus service network, **Russell Bivens** has been named field service engineer manager. **Chip Gould** has been named field service engineer, operating out of Lexicon's Los Angeles office.

Osamu Tamura has been named vice president of Sony's professional audio division.

Linda Frank has been named general manager for Martin America, the U.S. distributor of Martin Audio.

Will Lewis has been named national sales manager for Symetrix.

Aphex Systems has announced two appointments. **Arnie Christensen** has been named sales manager. **Michael Stewart** has joined the newly formed Systems Design Group, which will develop a variety of MIDI-oriented control devices.

RE/P

MANAGING MIDI

By Paul D. Lehrman

Gotta Resolution

I received an interesting and somewhat distressing letter from a reader recently about a problem he was wrestling with that seemed to have dire consequences for all MIDI users. He had approached various figures in the MIDI world with his problem and had yet to get a satisfactory response, so he tried me.

It seems that he was having difficulty getting his hardware sequencer, when running from an external sync source, to resolve events any closer together than 1/24th of a quarter note. Although the sequencer was capable of much higher resolution when it was using its internal clock, when it was clocked externally, any notes that were supposed to be spread out over 1/24th of a beat sounded simultaneously—in other words, fast runs became groups of chords.

His letter said he had experimented with various sequencers, both hardware and software, and had come up with the same results, and was therefore convinced that this was an inalterable fact of MIDI life—that because MIDI clock pulses are 1/24th of a quarter-note apart, no sequencer, no matter what its internal resolution, could hope to do better than that when it was being synced externally.

This came as something of a shock to me, as I've been using external sync for years, and never noticed my arpeggios clumping up or my carefully "lagged" snare drum hits catching up with the beat. If this reader's hypothesis were true, MIDI music would be even more mechanical-sounding than it already is, and whole portions of the music industry—like electronic film scoring—would probably not exist.

It is true that MIDI clocks are sent at the rate of 24 per quarter note, and they are the only timing reference an externally clocked sequencer has to work with. At a tempo of 120 beats per minute, this results in a resolution of about 21ms, which is adequate for some music, but not nearly all. Some sequencers developed in the early days of MIDI did not resolve externally synced music any closer than this because nobody really thought about it (and the hardware sequencer on which the reader first encountered the problem dated from this period).

By the time the first professional software sequencers came out, however, many developers realized that this could be a serious limitation, and so they incorporated interpolation into their programs. Today, most sequencers use interpolation—including one of the programs my reader said he had tried the experiment with.

Interpolation means simply that the sequencer maintains its high internal resolution, and uses incoming timing pulses merely as a *reference* for its own timing circuits. So even though the incoming MIDI clocks are only running at 24ppq, the sequencer's (or computer's) internal timing circuits are still working at full speed.

Say, for example, a sequencer has a resolution of 240 "ticks" per quarter note (or 10 ticks per clock), and an event is supposed to take place seven ticks after a beat. The sequencer will measure the time interval between the *previous* MIDI clocks (call it t), and then output the event at a time equal to beat + $t \times 7/10$.

"Ah," you say, "what happens when the tempo is changing?" Well, yes, there could be a problem then, but it would be minimal. If the tempo of a sequence were to speed up drastically, and there were high-resolution events immediately following the tempo change, those events might well come out late. But the sequence itself would not slow down (any events on or after the next clock pulse would be right on the money), and so the effect would be a "bunching up" of events that would last only for 1/24th of a beat or so (depending on how tight the interpolation algorithm is).

When using MIDI Time Code, the approach is approximately the same, although it provides much finer resolution: There are 120 quarter-frame messages per second, which translates into about 8ms per message. (For standard MIDI clocks to achieve this resolution, the tempo would have to be 300bpm.) Despite the increased resolution, since MIDI Time Code timing messages don't normally fall on musically significant times, interpolation is that much *more* important: If a sequencer couldn't interpolate, the music would sound very strange indeed, locked up to quarter-frame messages that bear no relationship to the actual tempo.

So, to my concerned reader, this message: relax. The walls of MIDI are not going to come tumbling down. Replace your old hardware sequencer with a newer one, and you'll find your arpeggios intact. If you already have a newer sequencer, it may have selectable resolution—make sure it's set as high as you need it.

Another item that arrived recently in the mail is a little book that I can recommend to anyone involved in computers and music, and that is "The Electronic Musician's Dictionary," by Craig Anderton, published by Music Sales Corporation. Craig is one of the best-known writers in the field of electronic musical instruments, and is editor-in-chief of *Electronic Musician.*

The dictionary is a handy volume containing brief, but informative, definitions of just about any term you are likely to encounter as you wade through the morass of jargon that is electronic music. In fact, one particularly apt definition is of the word "jargon," and includes the sentence, "Jargon is sometimes used to intimidate beginners." This is not an electronics dictionary, and you won't find definitions of Butterworth filters or half-wave rectifiers, but you will learn that "boot" (as in "boot the computer") is short for "bootstrap loading", and that "k" as an abbreviation usually means 1,000, while "K" means 1,024!

You'll also find precise and very useful definitions of synthesis terms like "hard sync" and "FM," as well as the origin of acronyms like DIP, DRAM and DIN. (But is "kluge" really German for "rat's nest"? It ain't in my German dictionary.) Best of all, Craig has described the book as one he intends to work on "the rest of my life," with additions and upgrades as the field develops and expands, and he invites readers to contribute their ideas. It's a great start for a great idea.

RE/P

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

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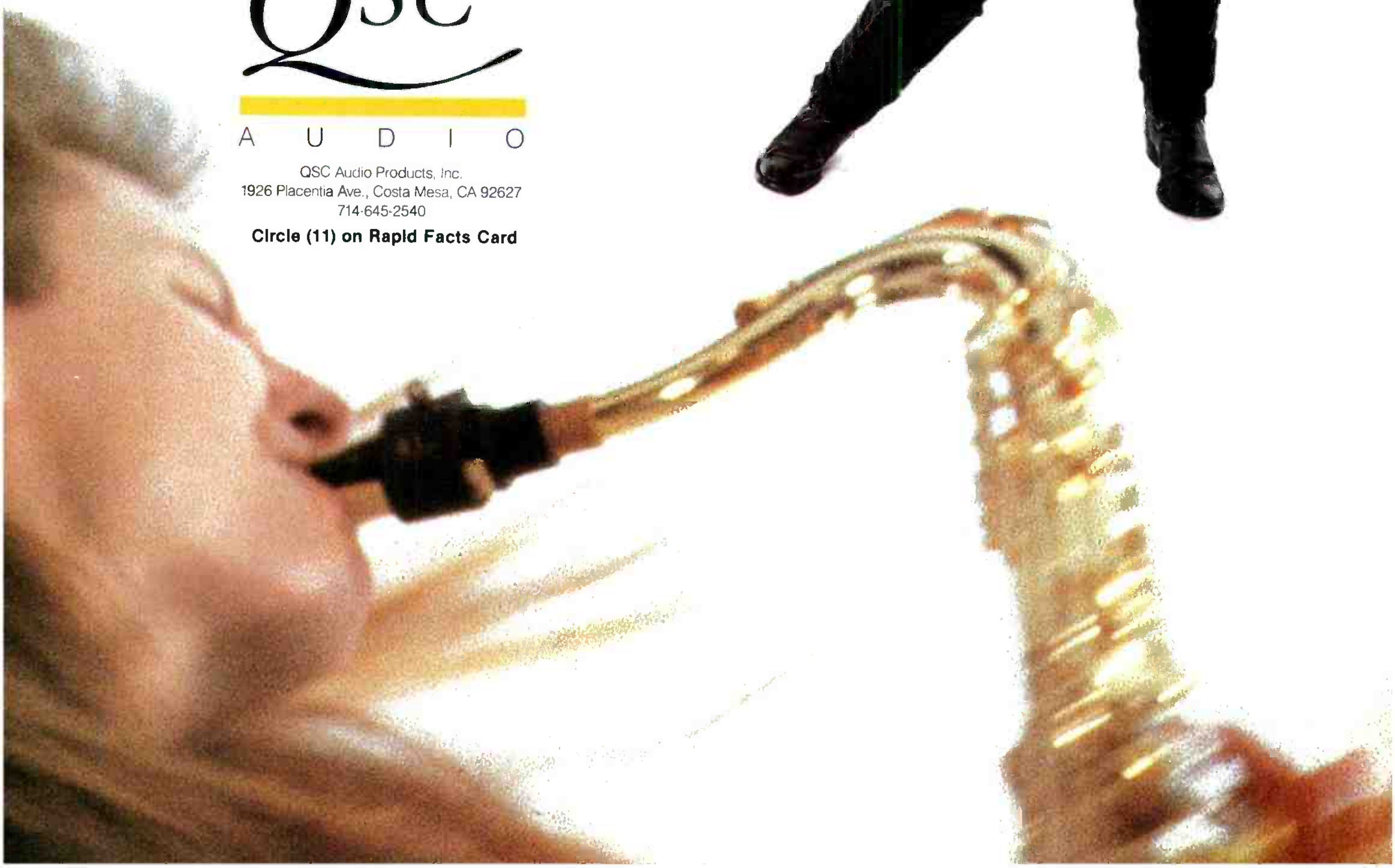

SPARS ON-LINE

By Richard Trump

Exploring Sound Reinforcement Opportunities

There is a tendency among the general public to see the audio industry as a whole, without fully understanding the specialty fields that we, as professionals, have chosen to pursue. It's not unusual for someone on a church board to equate your recording business with being able to solve the audio problems of a highly reverberant sanctuary. Or, a friend with a bar might expect you to be able to solve his sound system problems. Perhaps your best ad agency client has a presentation to make and wants to rent the speakers from your studio. Sometimes I feel like a doctor whose friends are always mentioning their mysterious aches and pains.

When confronted with such queries, the most obvious and wisest response should be, "Sorry, can't help you. That's beyond my expertise." Unknown territory is best left unknown. On the other hand, such inquiries usually trigger a different response: "Sure, let's give it a try." There's nothing like the temptation of extra cash to shift the direction of a business. Somewhere in between lies the indecision—"Gee, I hadn't really thought of doing that before. Let me think about it."

If a more qualified organization is available, let that company handle the job.

Having given all three responses in the past, I intend here to relate some background and reasoning that has worked well for the long-range goals of my business.

Richard Trump is regional vice president/treasurer of SPARS and president of Triad Productions, Des Moines, IA.

Do you have the knowledge and experience necessary to tackle a sound reinforcement job? Even if you know studio acoustics quite well, the demands of a reverberant hall, noisy night club, outdoor setting or large arena offer considerably different challenges. Instead of facing the studio problems of proper imaging, extending bass response or providing the right outboard gear, you will need to deal with speech intelligibility, loudspeaker driver alignment, coverage and acoustic gain.

Designing and installing a good sound system is an increasingly complex process.

Sometimes, because of market size or the lack of other resources, you may be the only person for the job.

Increased industry knowledge, not changes in the laws of physics, has made system performance more predictable. Make sure that you have the knowledge necessary to meet the needs of your customer.

Do you have the appropriate equipment? One reason that system performance has become more predictable is the availability of new tools to measure and analyze the difference between "adequate" and "excellent." These tools are generally not part of a recording studio's facilities, so your results are left to be judged subjectively. If the results are truly good, you are probably safe, but when they are not, corrective measures can be difficult to prescribe.

Do you have something special to offer? You may have unique equipment or special qualifications for a particular job. If a system's primary function is in your area of expertise, you have good reason to be involved. For example, a sound reinforcement project may have some incidental need for recording.

Are you an "expert" by default? Sometimes, because of market size or the lack of other resources, you may be the only person for the job. If you are a courageous soul and really can't think of someone better suited for the task, you may be the best person for the job. But always weigh the customer's options carefully and try to make the decision that is best for them. A wrong move could seriously damage your overall reputation.

Are you desperate? You just had a major session cancel, the equipment payment is due next week and a client wants to rent a playback system. Saving your own hide may be reason enough to cross over into a new business. Just remember that the precedent is likely to create more requests and will make refusals more difficult in the future.

Annually, we provide the on-air mix for a two-site, 21-hour regional Variety Club telethon. It is necessary to supply house and stage mixes at both sites. At one venue, the room is small enough for us to handle all audio. At the larger facility, we understand our limitations and subcontract reinforcement mixing and reproduction.

We keep a couple of small presentation playback systems available for client demos. The equipment requirements for the playback of tapes in a typical boardroom setting are similar to those found in a production studio. Since we often have future work at stake, it is appropriate that we provide a system that properly displays our product.

Saving your own hide may be reason enough to cross over into a new business.

My company certainly doesn't profess to fill the role of expert that some people assume we hold in our market, but we have found a niche that encourages us to continue expanding. Since our reputation is always at stake, we refer more sound reinforcement work than we accept. We also have learned the benefits of collaborating with other professionals when it is called for. A general guideline might be: If a more qualified organization is available, let that company handle the job. If not, be aware of your limitations, and consider carefully whether you are capable of satisfying the customer's needs. Then, proceed with cautious enthusiasm.

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UNDERSTANDING COMPUTERS

By Jeff Burger

A New Year's Resolution

Welcome to 1989! As Father Time takes us from one chapter of history to another, at this time of year, we often make New Year's resolutions. This year, I'd like to suggest to those of you who haven't already: Resolve to abolish your computerphobia.

I am constantly amazed at the number of people who say, "I don't understand computers," and are seemingly content with that disclaimer as a justification for not dealing with them. You don't have to understand how cars operate to drive, nor do you have to comprehend the inner complexities of a microwave oven to make a hot meal. They are just tools with little information overhead or learning curve to get the desired result; the caveat is simply that the more you know about these tools, the more you can coax out of them.

Like the other aforementioned tools, to be productive with a computer really doesn't require that much agonizing. John F. Kennedy's historic statement, "The only thing we have to fear is fear itself," is a very valid concept when applied to computers.

Why are people so fearful of them? Perhaps because they see so much potential power that the misconception of complexity is perpetuated. Perhaps it's because of horror stories that still linger from the primordial days of personal computing.

Computers are not incomprehensible alien creatures—they are incredible machines. We shouldn't lose sight of the fact that computers are also the product of human minds. Further, human minds are continually working to reshape hardware and software to do our bidding with the least amount of effort. As an extension of the human mind, computers work in very logical ways. With each successive generation, they are evolving to operate in much the same way that our brains allegedly function.

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

Computers often take the rap for inferior software designs. In other words, the hardware has very little personality of its own—the programs we feed them provide the majority of the human interface characteristics that we like or dislike. A bad sequencing package may make you hate a given computer, while a good one might turn you into a silicon evangelist! This is part of the GIGO principal—Garbage In, Garbage Out. Humans still determine the way computers respond. And, yes, computers do force you to be accurate and somewhat detail-oriented—which is another possible reason why people still avoid them.

In a recording studio, you're already dealing with computers every day. Most of today's equipment incorporates some type of computer.

Another way that manufacturers try to overcome resistance to computers is through development of improved interface methods. As an example, the overall subject of an evolved human interface is currently being addressed by both the hardware and software communities. Interface options have already been developed to allow the handicapped to use computers. Hardware has been designed to allow artists to draw with the computer and for musicians to enter music with their native instruments.

As for generic human interfacing, the mouse-driven point-and-click approach, within a graphics-oriented on-screen environment, has taken over in most of the personal computers on the market. Even Big Blue (IBM) has gravitated this way. The main reason for this shift seems to be that it is a much more intuitive way of interfacing with the computer. Another reason is that the ability to type well has become less of a distinguishing factor in how comfortable we feel with computers. Simultaneously, the need to remember arcane commands and syntaxes has lessened greatly with this popular interface.

As always, the future is right around the corner. As an example, speech input and talking computers are a reality now. But two things have slowed the progress in getting these features to the consumer: the evolution of artificial intelligence and the price of hardware. The technology for artificial intelligence has come a long way but still has quite a way to go. For scientists to understand it completely is to understand the human brain completely.

However, significant progress has been made in recent years, enough so that we'll soon be seeing the practical reality of a voice interface. Some programs are already becoming pretty good at interpreting commands typed in plain English as opposed to verbatim commands.

With hardware prices, RAM is the big Catch-22. Chances are that the more sophisticated a program is, the larger it is. That means more memory. For years, the cost-to-function ratio has been plummeting to offer about twice the memory for half the price of the previous year. The catch is that programmers keep writing larger, more sophisticated programs, thus requiring more memory. My first computer (in 1980) was an Apple II with 48K of RAM. I thought I was going "big time" when I expanded to 64K. Today, having only 1Mbyte of RAM cramps many people's style!

The result is that entry-level computer prices are still decreasing, while the sophisticated "memory hogs" are remaining at about the same price—but gaining sophistication. The memory required for artificial intelligence and speech synthesis/recognition is formidable, but coming into monetary reach. When this is finally accomplished, one of the last frontiers will be a brain-to-computer interface; we'll just have to think about a task and it will be performed. And that doesn't appear all that far off!

If you are still a computerphobic, the point of all this is two-fold: First, we all have a fear of the unknown that is often easily conquered by familiarity. Remember taking your first driving lesson...how awkward and scared you were backing out of your parents' driveway for the first time. Driving quickly became second nature—the same is true of computers. Second, the computer of today is not the computer of 10 or even five years ago. Hardware and software developers have had to make their prod-

90 ways to depart from the straight and narrow.

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ucts more friendly, intuitive, reliable and cost-effective to reach the broadest possible market.

When this column first started, I brought out a point that merits repeating. If you work in a recording studio, you're already dealing with computers every day. Most of today's consoles, tape decks, signal processors and instruments have some type of computer incorporated in the heart of the machine, if they aren't actually full-fledged computers. Trident's Di-An console and the Emulator III, to mention just two, are really computers with specialized interfaces and additional hardware. The Casio Data Bank on your wrist is a computer. Your programmable VCR and microwave...well, you get the picture.

The mouse-driven point-and-click approach has taken over as a more intuitive way of interfacing with computers.

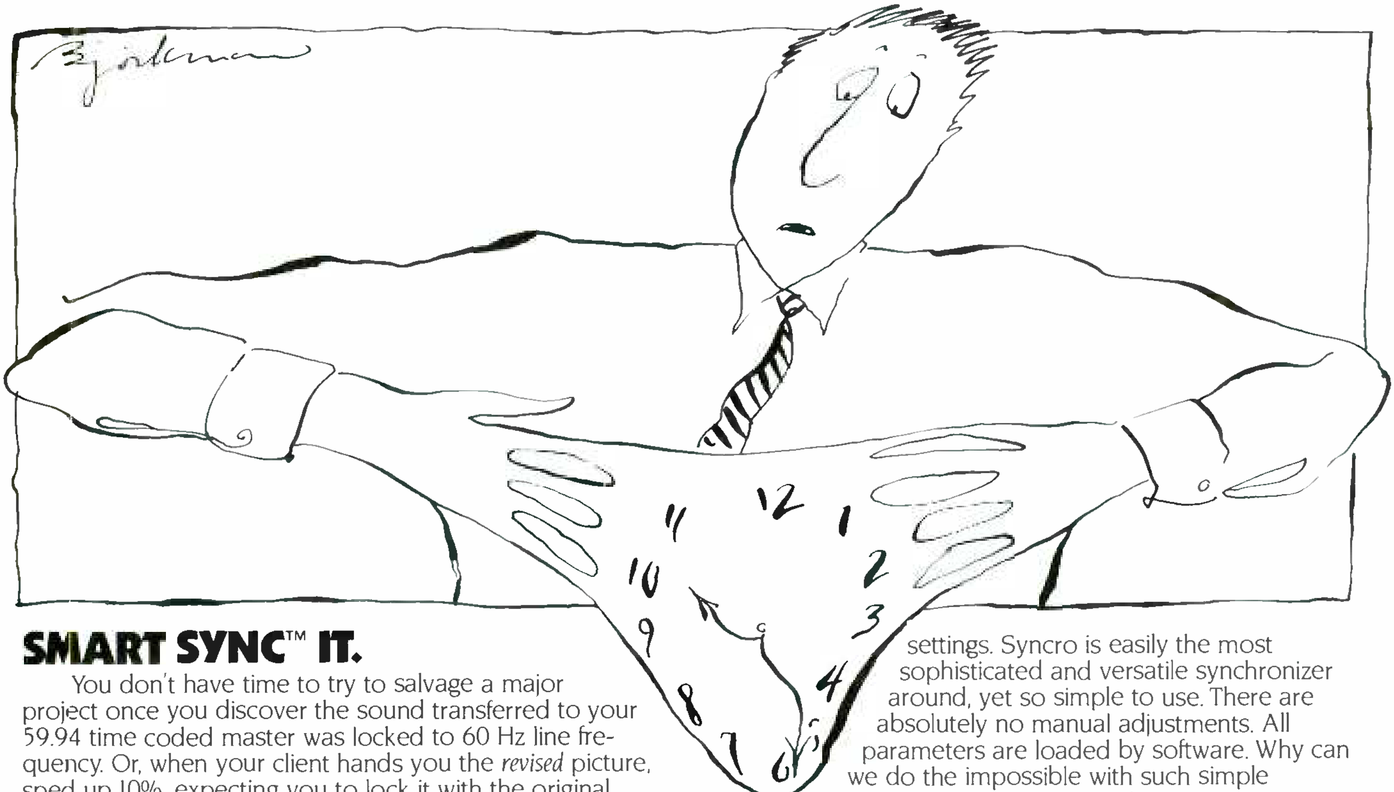
Computer technology is giving us the ability to tap instantly into the entire wealth of man's accumulated knowledge. CD-ROM and on-line databases allow us to access encyclopedias, dictionaries, reference materials and, indeed, entire specialized libraries that contain staggering amounts of information.

Modern audio and video technologies have blurred, resembling computer technology more and more with each day. The computer has become teacher, entertainer, business partner, muse, artist and much more. An emerging technology, of special interest to many people, is interactive education with the computer providing audio-visual tutorials tailored to each user. With the wealth of information available and the almost limitless learning capabilities of the human brain, this technology is pushing us head-over-heels into our future, whether we like it or not.

If you still have a touch of unfounded computer fear, stay tuned to this column—it was developed to help demystify computer terminology and technology. Next month, we'll return to our discussion of telecommunications.

RE/P

DON'T GET STUCK WITH TIME ON YOUR HANDS.



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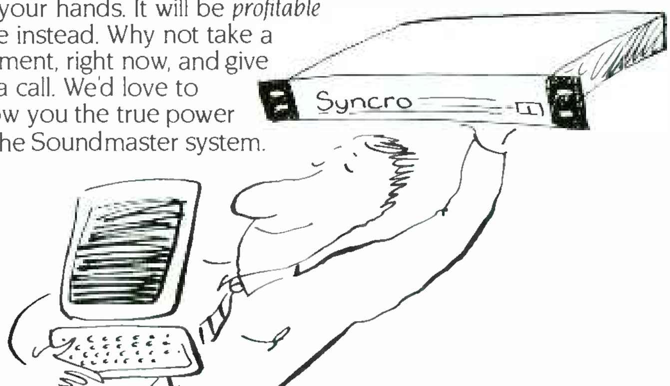

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RE/P

Electronically Controlled Speaker Systems

By David Scheirman

A look at the assets and liabilities of integrated controller/speaker systems.

In less than 10 years, an ever-expanding group of commercially available electronically controlled loudspeaker systems has appeared in the sound reinforcement industry. Ranging from lightweight, compact audio-visual boxes to large-scale modular enclosures suitable for concert sound use, this new breed of speaker systems is marketed as a cure-all in some circles. The various system performance parameters and design philosophies differ significantly, as a closer investigation will reveal.

Ever since the development and introduction of commercial sound systems for use with public address, live music and playback reproduction in sound reinforcement applications, designers have searched for the optimum way to package the loudspeaker components.

An incredible array of loudspeaker systems is available today, in every conceivable size and shape. The desire for compact, high-performance systems has led many manufacturers to concentrate on the development of integrated systems that not only match the transducer components to the box enclosure, but also rely on dedicated electronic circuitry packages to wrestle every last ounce of "woof" and "tweet" from the loudspeakers.

These electronically controlled speaker systems vary somewhat in design philosophy. What they all have in common is that the user has more than just speakers and a box to deal with; a proprietary active signal-processing device is usually needed to make the speaker system operational. (See Photo 1.)

Some manufacturers use the term *processor* to describe their dedicated electronics packages. While taken from the commonly used term *signal processing*, this nomenclature also provides a convenient marketing "hook." It links us with the technologically minded terms *data processing* and *microprocessor*, both of which have the subliminal implication that our all-knowing and error-free friend, the computer, is hard at work on our team, although this is not necessarily the case.

I will use the term *controller* to describe these dedicated electronics packages. Whatever a manufacturer's name for a special signal processor, its purpose is to control the performance of the speaker system, using predetermined operating parameters and/or real-time *sensed* information.

A number of functions are being assigned to these different black boxes, but a careful, logical study of a system manufacturer's literature should provide a reasonably good idea of the design philosophy behind a particular system. Some are merely electronic crossovers with equalization enhancement. Others incorporate group signal delay, phase inversion, bandpass limiting, component protection functions, and more. All, however,

David Scheirman is *RE/P*'s live sound consulting editor and president of Concert Sound Consultants, Julian, CA.

are controllers in the sense that their purpose is to instruct the loudspeaker system how it is to perform. The controller takes an input signal, divides, filters, and cuts or boosts it, and offers the system user more optimum loudspeaker performance, in most situations, than could be otherwise obtained.

The controller's task is to integrate a set of carefully selected speaker components with a wide variety of input program material. A combination of loudspeakers that is optimum for reproducing just the human voice may have very different characteristics from those most suitable for certain types of instrumental music. And, different loudspeakers respond in varying ways to different input signals. For example, drum rimshots, brass groups and cymbal clashes can lead to problems with loudspeakers whose impedance drops to low values at high frequencies (as will happen with low-frequency components operating in the upper end of their response region).[1]

The controller is called upon to deliver the best possible sound under all conditions automatically, with any program input. Users of such systems often include rental companies that provide sound equipment temporarily for public speaking, acoustical and symphonic music, or rock concerts. The ideal control electronics package will take it all in stride.

Indeed, some of the systems detailed here are severely compromised in terms of design intention if the prescribed controller is not used. While it also may be possible to wire the output of any crossover to the power amplifier channels and drive a given speaker system in that manner, performance of the whole system will suffer. Understanding a controller's intended functions helps up to realize its true value.

Is there really a concerted effort on the part of system designers to find ways of making packaged loudspeaker systems smaller, more efficient and better-sounding—all in the interest of good audio? Or is a conspiracy under way to force end-users to purchase expensive, mysterious and unwanted black-box "controllers" with each different brand of speaker system, with the motive being higher profits for the manufacturers?

You be the judge. Don't rely on this feature, alone, however. And don't rely solely on the makers' specification sheets and advertisements. Listen to these systems and use them in professional situations. Compare them with traditional systems and with each other. Keep the laws of physics in mind as you listen to the marketing claims, and make certain that a specific system will meet your needs in the real working world.

Speaker systems are available today that definitely offer higher average sound pressure levels, less mass and more intelligence than traditional, uncontrolled speaker systems. You alone must decide whether or not the intelligence, or signal-shaping ability, that's built into these electronic controllers is compatible with your requirements.

History

Early speaker systems were unavoidably bulky. The large two-way, horn-loaded bins such as those developed at RCA and Western Electric in the 1930s (for

Photo 1. *Electronically controlled speaker systems are currently available from many manufacturers. Shown here are a pair of compact enclosures with controller from Apogee Sound.*

Photo 2. *The UltraMonitor (now the UM-1A) is a compact two-way floor monitor developed by Meyer Sound Laboratories and first released in 1979. This product proved to the industry that electronically controlled speaker systems are marketable.*

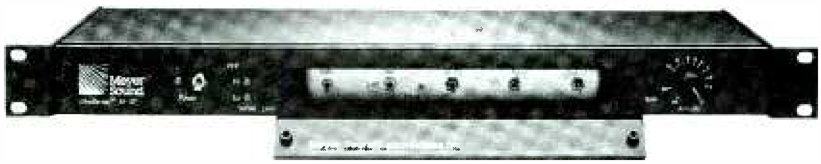


Photo 3. *Some controllers, such as Meyer Sound Laboratories' M-3T, offer signal delay for optimum high-frequency phase alignment.*

use in movie theaters) are a startling contrast to some of the compact speaker systems available today. The system package for two cone-loudspeakers, a single compression driver mounted on a multicelled horn and a passive, internal crossover would fill an 8-foot truck bed—but was intended for permanent installation. While overly large, such speaker systems were relatively efficient, and were capable of producing respectable sound pressure levels with only a 10W or 20W power amplifier.

As packaged loudspeaker systems became smaller, speaker components became more durable and amplifiers became more powerful, some designers began to focus on an integrated systems approach. With this design direction, the optimized crossover frequency and slope configuration for specific loudspeakers was sought out and provided. In the search for a good compromise between transducer reliability and sonic fidelity, the causes of loudspeaker failure under stress were examined. Means of offering both fail-safe system operation through speaker protection, and enhanced performance through variable, level-sensitive equalization were sought.

In the late 1960s, Bose Corporation pioneered the use of dedicated, active equalization for use with specific compact speaker enclosures. A low-frequency enhancement circuit packaged in a separate black box was available for use in conjunction with small-cone loudspeaker systems. Even in this electronically controlled age, Bose prefers to stick to this concept of active EQ with no motional-feedback or sensing circuits in its Acousti-Mass system.

In the mid-1970s, McCune Sound (San Francisco) fielded the proprietary JM-3 system, perhaps the first integrated, full-bandwidth controller/speaker system to see professional use in sound reinforcement. Developed by John Meyer for McCune, the innovative system saw much use in Broadway musical productions and live concert applications. Meyer began independent development projects on subwoofers and other speaker systems that required special signal processing.

In 1979 his company, Meyer Sound Laboratories, introduced the UltraMonitor with its own control electronics unit. (See Photo 2.) Combining crossover, equalization and loudspeaker system protection functions in a single package, this system created quite a stir upon its introduction. System users noticed that it offered relatively high average sound pressure levels for such a compact enclosure, along with seemingly extended low- and high-frequency response characteristics.

A new standard was being set in both the high average sound pressure level and the extended response available from compact speaker systems. A new standard was also being set in pricing for loudspeakers in a box. Accepted into the professional marketplace, the Meyer Sound products established a reference point that other manufacturers began to notice.

In the 10 years or so that have followed, the group of suppliers that offer electronically controlled speaker systems for sound reinforcement use has expanded dramatically. Let's take a look at the various operating functions attributed to

Photo 4. *Celestion's SR speaker system includes compact enclosures housing two 8-inch wide-range cone loudspeakers. Optional subwoofers and unique mounting hardware round out the line.*

Photo 5. *This controller (Renkus-Heinz X31) has LED displays for temperature and excursion protection circuitry activation in low, mid and high bandpasses.*

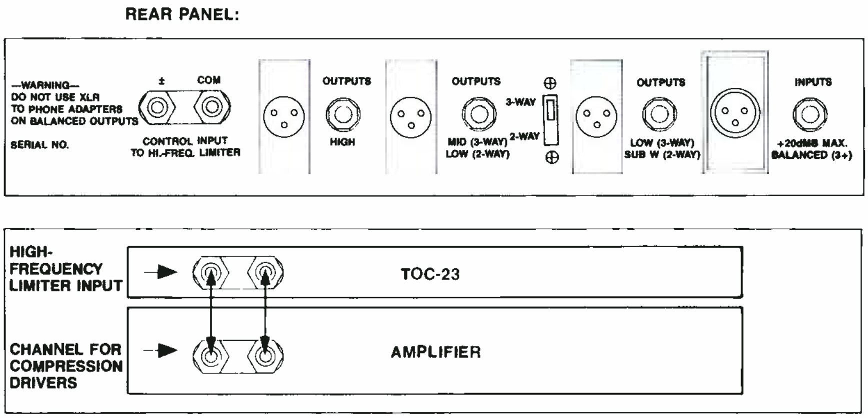
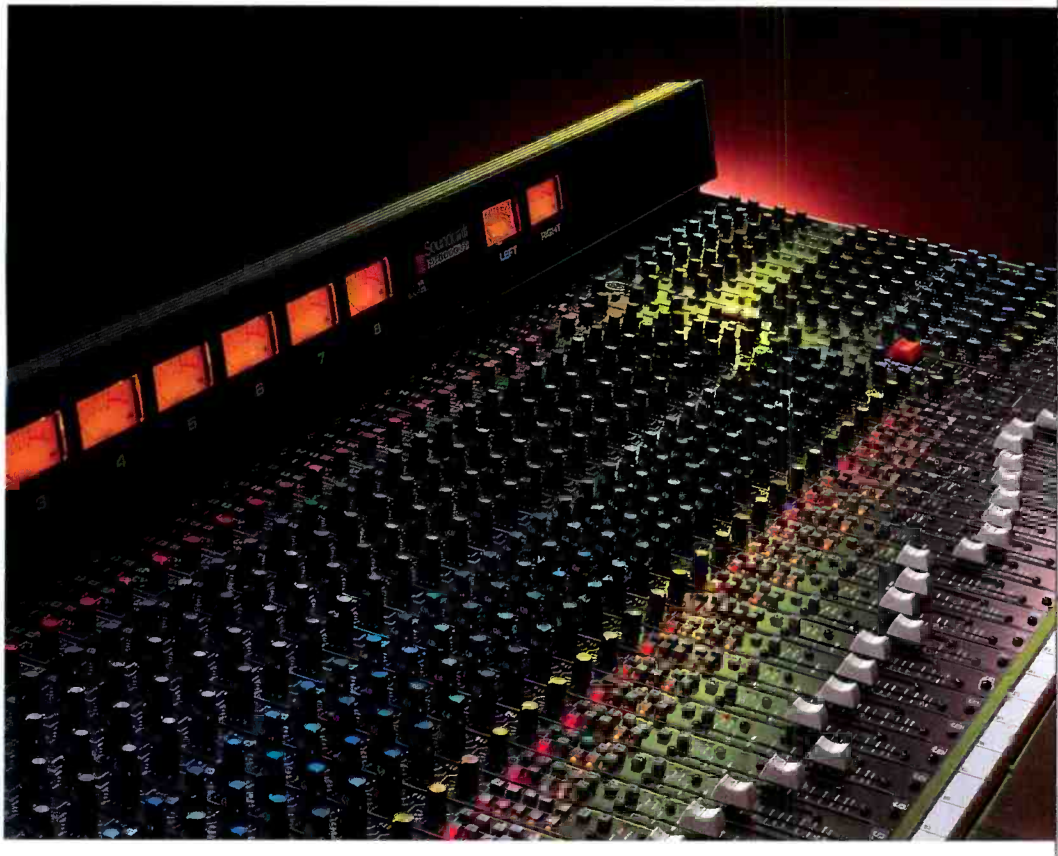


Figure 1. *Controllers from PAS use sensing connections from the output of the high-frequency power amplifier channel to trigger limiting function.*

Installed Or On The Road, They're Clearly Superior Consoles.



The Series 8000 input module is available with LED metering or VCA subgrouping, and incorporates four-band parametric EQ giving you the subtle control needed to produce dramatic results.

The Soundcraft Series 8000 consoles are clearly superior. Designed to meet todays complex production requirements, the Series 8000 incorporates the features you'd expect from consoles costing much more. No matter which configuration you choose, you can be assured your Series 8000 console has a clear and uncluttered audio path.

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The 8-buss design includes sophisticated ground-compensated summing and a grounding system that's hum and interference free. And the custom extruded aluminum rigid frame is very durable and extremely light.

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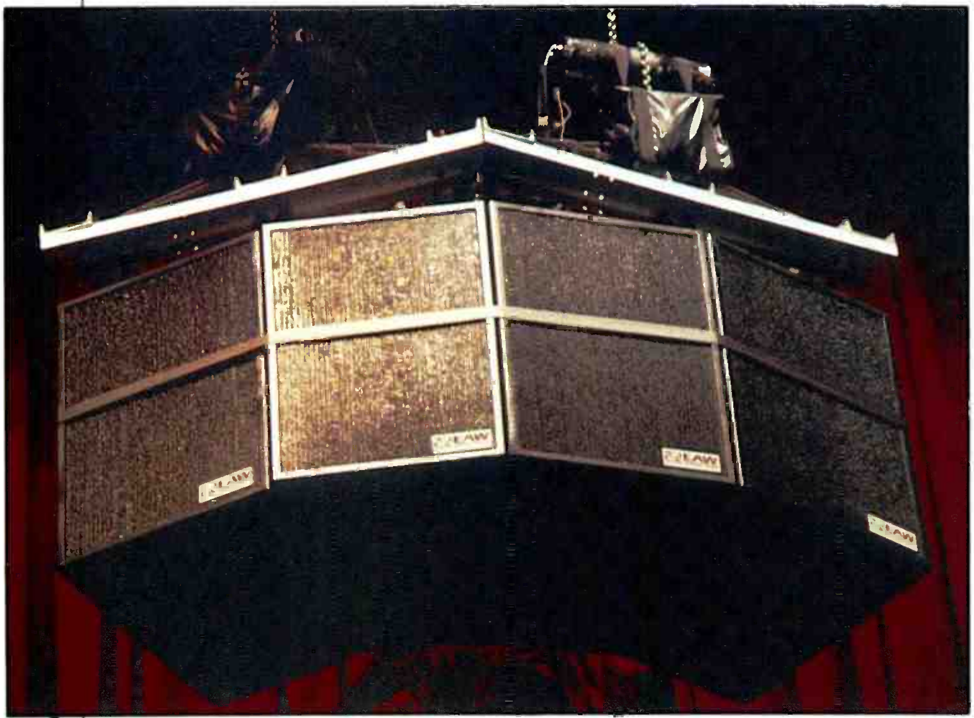

Photo 6. *EAW's KF850 system, shown here in a central point-source array at Radio City Music Hall in New York.*

some of these controllers. Then let's examine some of the more commonly used systems, along with a few new introductions.

The crossover/equalizer

As a system designer chooses the individual loudspeaker components, attention is paid to the crossover frequency, and the effect that a particular frequency choice will have on the power distribution to the various sections of the speaker system. A major purpose of the active filter is to confine the electrical energy that is sent to a given speaker to its own narrow operating bandwidth.

Most electronic controllers also incorporate specially tailored EQ "shaping," including narrowband filtering, to match the control circuitry to the specific speakers in use, and to offer an average frequency response that is nominally "flat." Narrowband filtering should be used carefully in such situations, however, since any filter/equalizer operating in the analog electrical domain can introduce aberrations to system phase response and transient-signal integrity.[2]

At high operating levels, when speakers are pushing up to their safe working limits, crossover point selection and slope are critical. For this reason, some system designers choose to work with variable, or floating, crossover points. This type of processor allows a system to get louder by shifting a greater portion of the power response away from fragile compression drivers and down to more-rugged cone loudspeakers. This spectral shift can also affect both the system's coverage-angle characteristics and aural fidelity. For this reason, it may not be desirable in some situations. This is particularly critical in systems that use combinations of multiple loudspeakers, where the dispersion characteristics of the loudspeaker combination is different from that of the individual units.[3] (See Photo 3.)

Level-sensitive, active equalization can be used to compensate for this phenomenon. It can also be used to take advantage of the Fletcher-Munson curves, based on the average human ear's sensitivity to particular frequencies, to provide more impressive bass and treble output when the system is working at lower levels. Thus, some listeners judge the system to be more "hi-fi" (or perhaps "disco"?) even when it is receiving a very low input signal. This form of equalization functions like the loudness contour button on a home stereo pre-amplifier.

Also incorporated in the crossover/equalizer section of some controllers is signal delay. (We can't delay time, but we can delay the audio signal.) High-level multiway systems often use signal delay to improve the system's coverage pattern at or near the crossover point by providing electronic compensation for driver phasing aberrations or mounting location. (See Photo 3.)

Loudspeaker protection circuitry

Different methods have been developed to protect loudspeaker components as packaged systems have become more compact. Loudspeakers fail either from thermal damage caused by excessive voltage input or from over-excursion, which can lead to rubbing and voice coil misalignment. Protection methods are based on sensing impending voltage overload or modeling voice coil temperature from the measured impedance readings of the speaker.

A single loud pulse, or transient, can tear a speaker apart, leaving us with mechanical failure. Heat damage tends to occur when a high average power level is maintained at the mid to higher frequencies of the speakers' rated operating range.[4]

Controllers can use electronic circuitry to limit input voltage by gain reduction. To avoid voice coil failure caused by thermal overload, compression can be used to prevent the temperature from rising too quickly in the component. Some controllers use "sensor" circuits to monitor the power amplifier's output voltage, and to limit the signal going to the particular loudspeaker components that are beginning to show signs of thermal stress. (See Photo 5.)

Frequency-dependent limiting based on impedance-characteristic measurements is one of the more sophisticated methods used to guarantee the highest possible average sound levels without component failure. Both low-cut and high-cut filtering reduce the voltage input to the loudspeaker system. They also narrow the system's audible frequency bandwidth as the sound pressure level increases.

Most controllers rely on sensed information from the power amplifier's output terminals. Products from Apogee Sound, Celestion, Electro-Voice, Meyer Sound Laboratories, PAS (Professional Audio

Photo 7. *E/V's new Delto-Max two-way bi-amped speaker systems come in 12-inch and 15-inch versions. Dedicated controllers are available, and the manufacturer discourages its use with other crossovers and/or controllers.*

Photo 8. *The new KP-600 series system from Klipsch features modular loudspeaker enclosures with identical exterior cabinet measurements. The tri-amped four-way system has a controller with user-selectable crossover parameters.*

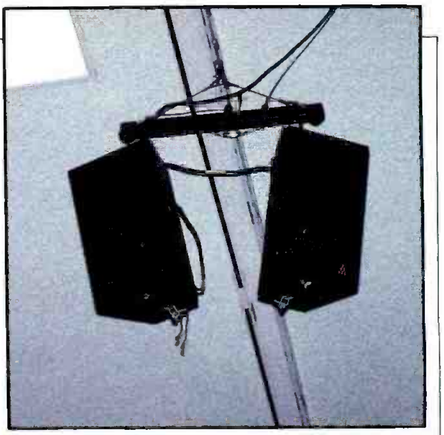

Photo 10. *A pair of Meyer UPA-1A enclosures are easily suspended from simple hanging hardware in a low-ceiling ballroom. The UPA-1A is typical of the single 12-inch box with horn/driver components in a trapezoidal package, available from several manufacturers.*

Photo 9. *The PM31-15 circuit module plugs into the front panel of the Renkus-Heinz system controller. Different modules are available for various speaker systems.*

Table 1. *Size, weight and cost comparison of four compact trapezoidal speaker systems, each with one 12-inch speaker and a compression driver with horn.*

	Apogee Sound AE-5	Electro-Voice DML-1122	Meyer Sound UPA-1A	Renkus-Heinz SR-121A
weight (pounds)	78	68	66	52
size (H×W×D) (inches)	23×14×14	23×14.6×14	22.5×14.5×14.5	24×15.5×14
peak output (decibels)*	130	129	132	132
retail price, single box (dollars)	2,045 (with fittings)	1,860 (add 348 for fittings)	2,390 (add 100 for fittings)	1,530 (add 296 for fittings)
retail price, controller (dollars)	1,850 (A-5)	1,632 (DMC 1122)	1,312 (M-1A)	1,650 (X22, add 250 for module)
total (dollars)	3,895	3,840	3,802	3,726

Notes: *The peak output is taken from manufacturers' specifications sheets. All figures are reportedly measured at a distance of 1 meter on axis. An independent verification under controlled conditions is recommended.

The total price for a single compact enclosure and necessary controller includes rigging fittings, top and bottom, which are optional on three of the examples. The Renkus-Heinz X22 controller requires the additional plug-in module for operation. Thus, the total cost in all four examples is for a single enclosure with rigging fitting and a functional controller.

The four examples are listed alphabetically by manufacturer. The fact that the total list prices are in descending order is coincidental. The model of the appropriate controller is in parentheses in the controller retail price column.

Systems), and Renkus-Heinz all include back-panel sensing-circuit line connectors, providing an information loop that enables the controller to function automatically in response to the amplifier/speaker system. Most units use both a low-frequency and high-frequency sensing signal path; the PAS unit senses high-frequency amplifier output level only. (See Figure 1.)

"We are not really big on limiters, because we feel that's a large part of what users refer to when they object to a 'processed' sound," explained John Kirkland, product manager for PAS. "However, when the limiter's input is tied directly to the output of the high-frequency power amp, you only get limiting when the drivers are in danger of being harmed. In effect, the limiter tracks the output of the HF power amplifier, using opto-couplers. So, the protect function is only engaged when high levels of certain types of program material are encountered."

In most protection systems, the goal is to maximize the output of the speaker components without destroying them. But, there is usually some sacrifice in signal integrity. Several different methods can be used to increase the average attainable sound pressure level of a speaker system.[5]

Controllers may use one or more of the following functions searching for the ideal combination of equalization and speaker protection:

• *Loudness Compensation*—Low (and sometimes high) frequencies are boosted when the system is operated at lower levels, and the effect is removed as the level is increased.

• *Variable High-Pass Filtering*—While often audible as a loss of bass response when the program material changes, this function will reduce both thermal and mechanical stress.

• *Variable Crossover Frequency*—The floating crossover point can protect high-frequency components from damage at

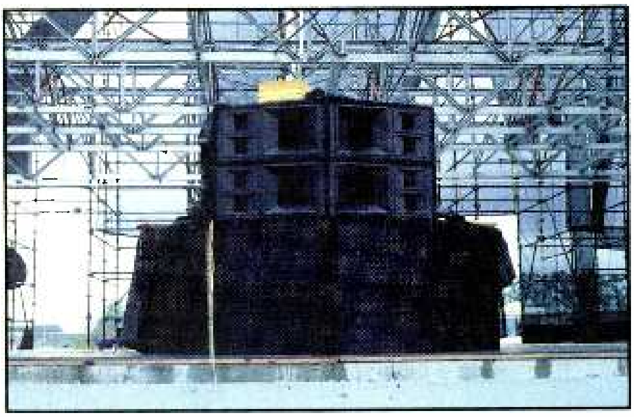

Photo 11. *After proving that electronically controlled speaker systems could be accepted in the marketplace, system manufacturers worked on development projects for larger, arrayable components for sound reinforcement use. Shown here, the first outdoor field test of the Renkus-Heinz MR-1/LR-2M system (on stage at Irvine Meadows Amphitheater in Southern California).*

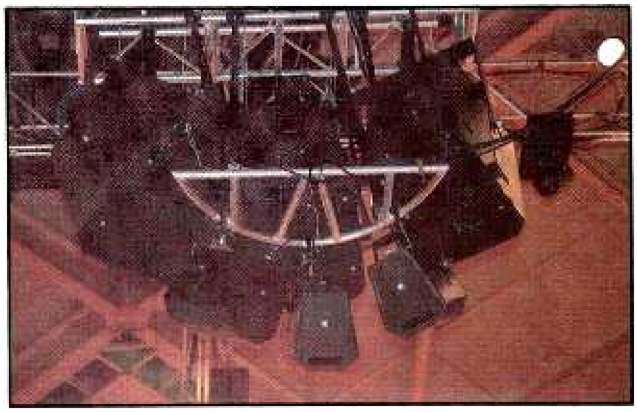

Photo 12. *Manufacturers intent on producing large-scale systems for the professional market must offer enclosures that are easy to use in arrays in hanging situations. Shown here, Apogee Sound's 3X3 and AE-5 enclosures in an array by Burns Audio at the 1988 Democratic National Convention.*

While electronically controlled speaker systems have been proliferating in recent years, some companies have decided not to jump aboard this trend too quickly. JBL Professional, for example, has offered a card-programmable electronic crossover since 1973. Models 5231 and 5232, introduced in 1973, evolved into the present model 5235, which features equalization for specific systems, including choice of crossover slope, power response correction and selectable high-pass filtering for low-frequency driver protection. Switchable monaural summing for subwoofers can be internally preset, along with selectable equalization for low-frequency alignments. These features allow the crossover to operate in a fashion similar to more expensive "electronic controllers."

To this manufacturer, "smart" electronics means something entirely different from what has been brought to market so far. JBL system designers maintain that they would never use electronics as a substitute for a larger power class in transducer selection, or use them in a manner that grossly altered the artistic intent of the performing musicians. The following statements from JBL establish that company's position (page 34).

—David Scheirman

high operating levels, but off-axis frequency response can vary considerably.

•*Soft Clipping*—Gently attenuates transient peaks, and allows protection from over-excursion without dropping the overall gain.

•*Compression and Limiting*—Overall gain reduction is an effective way to prevent thermal overload that leads to speaker failure. Variable ratio compression circuits can work well. High compression ratios in limiting circuits can be very audible.

Controller comparisons

In searching for ways to evaluate one system controller in relation to another, the most obvious direction to take first is what each sounds like. If we have the luxury of auditioning several different systems in the same listening environment, with the same program material, it is relatively easy to develop a working knowledge of each system's characteristics.

Since this is often not possible, an awareness of the audible characteristics of the previously mentioned signal-processing methods can still give us valuable information. By recognizing the methods used in a particular controller, we can gain some general knowledge prior to hearing it in use.

Apogee—Apogee Sound's controller is a crossover/equalizer with fixed crossover points and corrective equalization. Apogee has labeled its sensing circuitry PAF (positive amplifier feedback). A limiter circuit is used for driver protection.

The models are keyed to different loudspeaker systems (such as the 3X3, the AE-1, the AE-5 and the AE-10 subwoofer). RV designates road version for touring; PV designates permanent version for installations. Thus, a model A-1 RV would be dedicated for use with the compact AE-1 enclosure, with input and output connectors on the front panel for easy access during setup and testing.

Celestion—Celestion offers a crossover/equalizer, with a fixed-point 150Hz second-order filter for low-frequency (subwoofer) enclosure operation. Low-frequency protection is through compression, adjusted by power amplifier output monitoring. A three-stage thermal protection circuit for the extended-range speakers is based on voice-coil temperature modeling.

This device is named the SRC1 Electronic Controller. It features a back-panel bi-amp switch and phase reversal for the subwoofer section. It is designed for use with the SR speaker system. (See Photo 4.)

Eastern Acoustic Works—EAW offers a crossover/equalizer with signal delay and loudness compensation (low-

Electronically Controlled Speaker Systems: A JBL Viewpoint

By John Eargle, Mark Gander and Garry Margolis

"Electronically controlled speaker systems" is one of the catch phrases in current pro sound. The term refers to electronics that are dedicated to a given loudspeaker and that process the sound for enhanced presentation over that loudspeaker.

One of the stated premises of such systems is that they are basically pre-adjusted and ready to work, with little in the way of user adjustment.

Often, the underlying premise is that these systems have marginal transducers, and the real purpose of the "smart" electronics is to prevent overdriving or stressing those components.

Another premise is that the packaged speaker-plus-electronics should sound good at moderate to low levels when auditioned on the sales floor, since that is where sales are generally made. Therefore, a little loudness compensation (bass boost) may help make the system stand up better in competitive comparisons. Also, some degree of power response correction in the compression driver circuitry will help the overall balance.

As we see it, the big problem with these systems is that most of the correction, beneficial as it may be, is dynamic in nature, and it is gradually removed as the system is driven at higher and higher output levels.

John Eargle, Mark Gander and Garry Margolis are with JBL, Northridge, CA.

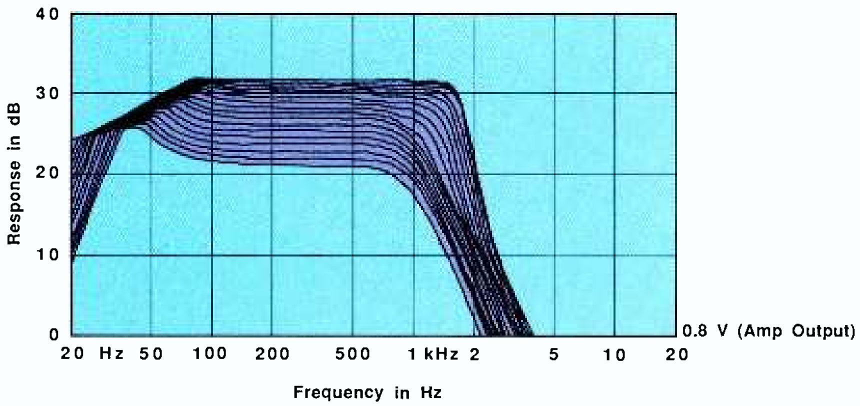

Figure 1. *Low-frequency output of processor, 16dB input range.*

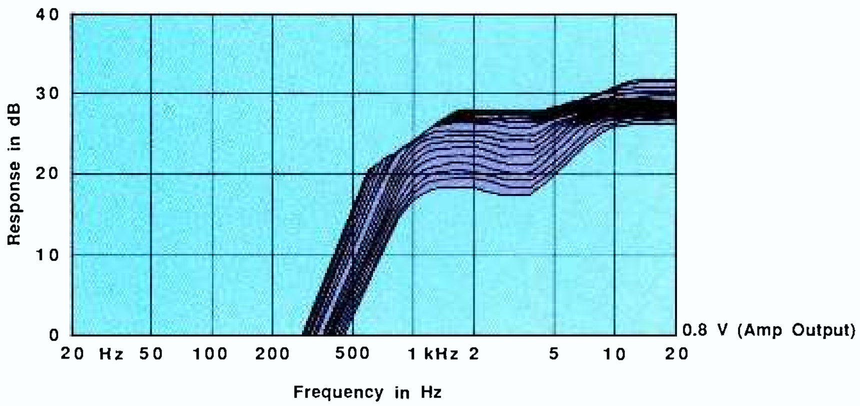

Figure 2. *High-frequency output of processor, 16dB input range.*

Examining one model

We have measured a number of such systems to determine how they typically operate. Figure 1 shows the electrical drive to a woofer in one such two-way system. The lowest curve in the graph shows a mild bass boost around 40Hz, providing some degree of loudness compensation for moderate acoustical levels. The curves were all incremented upward in 1dB steps.

Note that the boost is quickly removed, and in its place, at an input level 16dB higher, there is actually a bass roll-off below 80Hz. Note further that the actual drive level at 40Hz never varies more than 2dB over the entire input range of 16dB. Note also that the bass low-pass crossover frequency moves from about 1kHz up to 1.6kHz over the total input range shown.

Figure 2 shows what happens in the same system at high frequencies. Beginning at the bottom curve, note the generous 9dB power response boost above 4kHz. However, over a total input range incremented upward 16dB, the power response boost is reduced to only 3.5dB. Furthermore, the high-pass crossover point moves from 1kHz at low levels to about 1.6kHz at the highest input level shown here.

Benefits and liabilities

The system we have described exhibits three major departures from normal signal linearity:

1. The bass content at 40Hz remains virtually the same (1dB) over a program input range of 16dB. Music thus loses some important dynamic information.

2. The acoustical crossover point shifts from about 1kHz at low input levels up to 1.6kHz at the highest input levels shown here. There will

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be a change in system coloration and directional characteristics in the mid-band as the woofer is required to work at higher frequencies. In this case, the woofer is 380mm (15 inches) in diameter, and such drivers should normally not be crossed over higher than 1kHz because of the inevitable response and directional anomalies above 1kHz in drivers this size.

3. The overall change in high-frequency balance of about 5.5dB can rob music of brilliance as the program level goes from soft to loud.

In addition, there may be audible pumping in the dynamic control of the system as higher levels are reached.

On the positive side, such systems sound good at lower levels, and that is probably where they will be used much of the time. It is difficult for the system to go into gross distortion because of the overall compression activity in the processor. The systems are, in fact, relatively foolproof and quite easy for the novice user to set up. While some operators may prefer the relatively dense, compressed sound of the system, many musicians will not.

As an overall philosophy, if the system has inadequate high- and low-frequency drivers, the money spent on the processor might be better spent on more robust internal componentry. Transducers with smaller diaphragms and smaller voice coils than might otherwise be chosen will suffer from dynamic non-linearities such as increased harmonic and intermodulation distortion, and greater power compression. While electronics can try to prevent these overtaxed units from breaking and allow them to generate maximum output, the result is often not maximum fidelity.

frequency enhancement). The overload circuitry offers distortion protection through level-sensitive bandpass limiting. The unit is called the MX800 CCEP (closely coupled electronic processor).

A plug-in module allows the system to be used with the KF850 loudspeaker system, for which the MX800 is optimized. The KF850 system does not require this specific controller for operation. The manufacturer advises that electronic crossovers from other manufacturers can be used with excellent results. (See Photo 6.)

Electro-Voice—E/V's Delta-Max systems use a crossover/equalizer with signal delay circuitry. Crossover points are fixed, and high-pass filtering does not shift. Optimized equalization, including low-frequency enhancement (+8dB boost at 50Hz) is included. Speaker protection is achieved through an automatically ratio-variable compression circuit. Speaker modeling circuits provide thermal overload protection and amplifier anti-clip limiting. There is independent LF and HF soft-clipping for excursion protection.

Named the DMC-1122 and DMC-1152, the pair of controllers are intended for use with the respective 12-inch and 15-inch loaded Delta-Max systems. (See Photo 7.)

Klipsch—Klipsch has recently introduced a crossover/equalizer with user-selectable high-pass filtering in three bands. Automatic bandpass gain compensation is based on individual driver sensitivities. High-frequency equalization for flat response is incorporated.

Designated the KP-600-EC, the manufacturer calls this an electronic crossover/processor. Features include mute and phase reversal switches. It is designed for use with Klipsch's new KP-600 modular loudspeaker system. (See Photo 8.)

Meyer Sound Laboratories—Meyer Sound offers crossover/equalizers that use separate limiters for LF and HF outputs. The low-frequency component is protected by a variable high-pass filter. Meyer terms this circuitry Speaker Sense. Frequency and phase response alignment circuitry for specific loudspeaker systems is prealigned by the manufacturer.

Meyer calls its controllers "Control Electronics Units" and offers several versions, dependent on the loudspeaker system in use. These range from the P-1A (for the small UPM-1 column loudspeaker) to the M-1A (for UPA-1A and UM-1A UltraMonitor). The B-2A is designed for use with the USW-1 subwoofer. It features a Bass Extender circuit, offering transient-dependent "fuller bass" effects.

The M-3T incorporates logic circuits in the sense lines that monitor amplifier gain, providing automatic speaker output disconnect in case of overload or amplifier failure. A time correction circuit is switchable in and out. This controller is for use with the MSL-3.

Professional Audio Systems—PAS features crossover/equalizers with EQ compensation for constant-directivity horn designs and a subsonic filter. TOC (time offset correction) is employed as signal delay.

High-frequency component protection is achieved through a fixed-ratio soft-knee-type limiter circuit. Sensing lines are connected from the HF amplifier output to the controller's limiter input.

The units are designed to be used with PAS's Modular Reinforcement Systems (MRS-1, MRS-2).

Renkus-Heinz—Renkus-Heinz's Smart Systems use crossover/equalizers with automatic loudness (bass-boost) compensation. This changes to bass-cut equalization at very high levels to maximize average sound pressure levels.

High-frequency protection relies on variable crossover frequency points and over-easy-type compressor circuitry. Low-frequency protection includes variable high-pass filters and overall level compression. The manufacturer calls this process SPT (spectrum power transfer).

The Model X-31 controller is adaptable to different speaker systems. It includes front-panel plug-in modules, so that a controller can be quickly changed for use with a compact two-way system to a large three-way system. Three different modules are available. The PM31-15 is for use with SR-1, W-1 and SR-2 speaker systems. (See Photo 9.)

Compact loudspeaker system packages

In examining the various controller/speaker systems that are available, accurate, fair A-B comparisons can be difficult since such a wide variety of loudspeaker enclosures is available for many different uses.

We can, however, focus on just one type of speaker enclosure and check out a few vital statistics. Let's look at four examples of a compact trapezoidal enclosure, each housing a single 12-inch speaker and a compression driver with horn.

This style of speaker system is extremely popular with a wide variety of users. It is compact enough to be placed on small stages and light enough to be tripod-mounted. It lends itself easily to use as a suspended speaker in low-ceiling meeting rooms and hotel ballrooms. Found in most rental sound company inventories today, this style of speaker is quite versatile. (See Photo 10.)

Since we'll need a controller even if we are only using a single speaker enclosure, we'll take a look at the current retail price (as of October 1988) of one box and one controller. (See Table 1.)

Large-scale applications

With electronically controlled speaker systems firmly established in the smaller rental system market, manufacturers have begun to expand those technologies into larger system formats. Attempts have been made to develop integrated, full-scale speaker systems for use in bigger venues. (See Photo 11.)

Many of these same controllers are used with arena-sized, full-range systems for audio-visual, public address and live concert applications. To be successful in this part of the industry, speaker system manufacturers must offer products that are easy to use in arrays. (See Photo 12.)

Many companies include built-in hanging fittings and even loose rigging parts or flying bars with their product lines. Those manufacturers that are serious about the professional marketplace often work closely with touring sound firms in concert situations to develop products that are acceptable for this field, including high-powered subwoofer enclosures. (See Photo 13.)

A glimpse into the future

Power amplifiers can vary a great deal in their operating characteristics, and electronically controlled speaker systems can be very sensitive to the input signal as they work right up against their safe operating limits. Some speaker manufacturers prefer to see their customers use one design type, or specific brand of power amplifier with their controlled speaker systems. Other firms claim that any amplifier meeting the suggested power ratings will work fine.

Perhaps one of the next steps in the development of electronically controlled loudspeaker systems will be the integrated controller/amplifier/speaker system, in which the power amplifier is ideally matched to the signal processing and loudspeaker components. Perhaps the front-end circuitry will be included in plug-in modules that are easily inserted into power amplifier frames for use with different speaker systems. Several manufacturers are presently involved in development projects of this type.

Along this line, Yamaha Corporation of America has recently introduced the AST-P2602 power amplifier, which contains integral signal processing circuitry. A plug-in cartridge gives optimized equalization for the AST-S30 loudspeaker system, which comprises a trapezoidal enclosure housing a single 15-inch speaker, a compression/driver and horn mid-range section, and a ring-type compression tweeter. Yamaha says its Active Servo Technology allows the loudspeaker port to become an independently adjustable source of low-frequency energy, thus providing improved bass response from a compact loudspeaker system. The new AST amplifier does not require an extra sensing wire connection, however. (See Figure 2.) When the AST cartridge with its 800Hz crossover is removed, the AST-P2602 reverts to operation as a standard power amplifier.

We can also expect to see an ever-expanding range of tripods, clamps, flying fittings and portable accessories such as cable-sets to be developed and marketed by various speaker system manufacturers in an attempt to become a one-stop supply source for professionals.

The value of dedicated signal processing in the form of electronic controllers will become more evident as technically advanced manufacturers begin to provide computer-interface data ports, coupling their signal processing to a central control microprocessor with VCA circuits. (A system manufactured in the Netherlands by Stage Accompany already offers two-way, full-range speaker enclosures with built-in microprocessor control and the ability to link these boxes to a master control computer.)

One day soon, perhaps even the simplest setups will require more technical expertise to hook up—but once the system is wired and in place, many more operational variables will be in the hands of the sound system, not the human operator.

How does all of this affect the major touring sound industry? A few of the bigger concert-system rental companies stock large inventories of electronically controlled speaker systems, and these are in use with live musical concerts as much as they are for industrial and trade show events. Other firms refuse to bring this type of system into their operations. Most of the larger firms still field proprietary speaker systems.

Traditional electronic-crossover, multiway systems (with optimized equalization and bandpass limiting) will continue to be employed in applications where complete control of the operating parameters must be kept in the hands of musically oriented technicians at the mixing position. This is evidenced by most major touring concert systems in use today.

However, electronically controlled speaker systems have definitely found their niche in other parts of the professional sound reinforcement industry. Understanding the signal processing functions available within specific controllers is a must for any user needing to make certain that a given system will be appropriate for use in different applications.

RE/P

This feature is written with reader interest and education in mind. The mention of specific brand names and products does not imply an endorsement by the author, *RE/P*, or Intertec Publishing.

Photo 13. *Large-scale concert use requires speaker system components that can fill huge performance spaces. Shown here, Meyer Sound Laboratories' 650-R2 subwoofers as fielded by Ultra-Sound at Meadowlands Arena in New Jersey.*

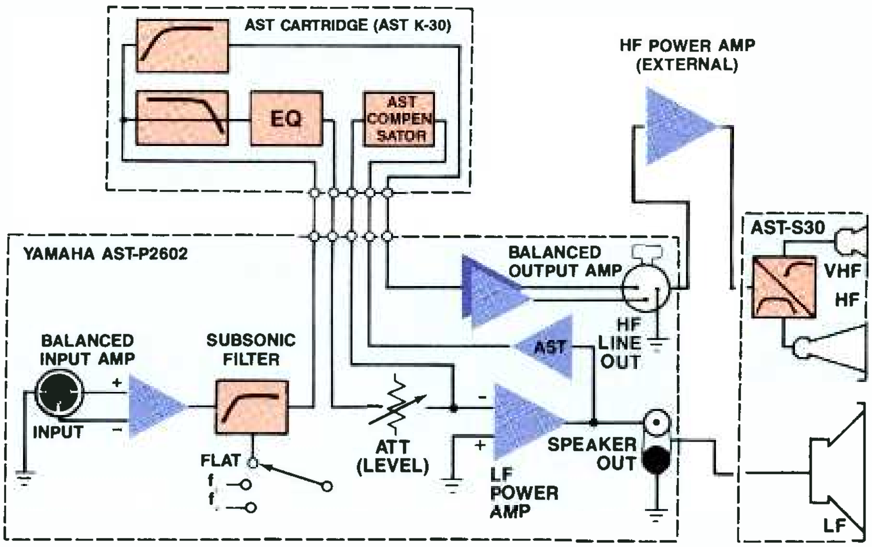

Figure 2. *Detailed block diagram of an AST amplifier/loudspeaker system.*

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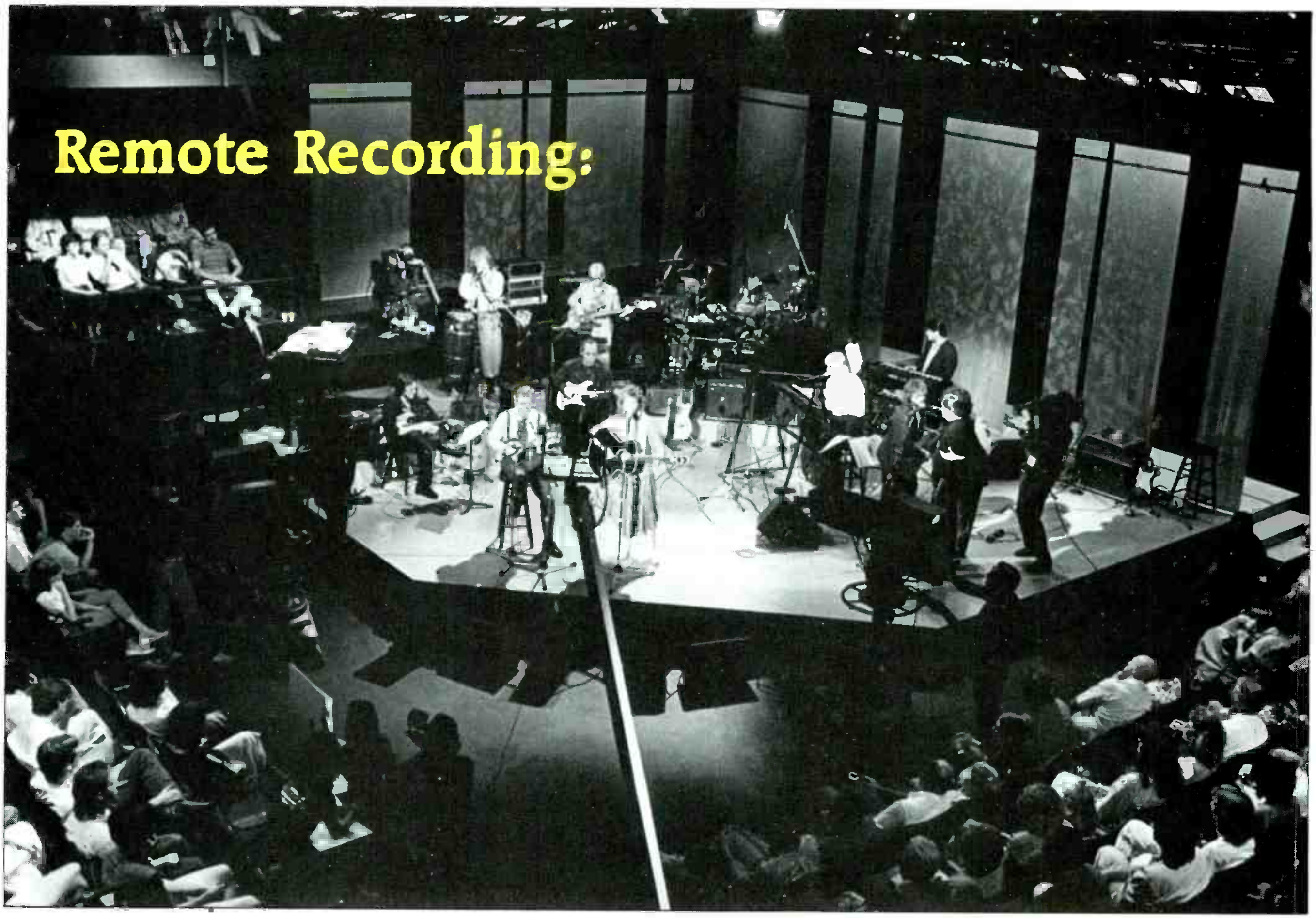


Chet Atkins HBO/Cinemax "Sessions Series"

By Johnny Rosen

Remote music recording for television presents some unique challenges. This "Session Series" demanded a particularly high level of audio quality.

Chet Atkins is not just another guitar player. He's also a song writer, producer and former record company executive. Segovia thinks he is wonderful; Mark Knopler thinks of him as a hero; Earl Klugh makes jazz records with him; and just about every guitar player from Eddy Van Halen to Bill Monroe has something good to say about him. When winning one of his many Grammy awards he said, "I'd like to thank RCA for letting me make my records at home." He won so many Country Music Association awards that they redesigned the rules so that somebody else would have a chance to win. He won another anyway, as a backup player on someone else's country album.

Susan Hackney and Associates, a TV production company in Nashville, started putting together a "Sessions Series" show for HBO/Cinemax. This series provides a close-up look at an artist from a performance point of view. Obviously, Chet presented a special criterion. He demanded sound that was both artistically and technically of high quality. HBO also has a definite set of technical guidelines that are listed in a standards book. The criteria are well defined and easy to follow because they are based on common sense and the use of readily available audio and video technology. What these technical standards cannot cover is the artists' feelings and the emotional level of every performance.

It's easy to capture that "special magic" when you can record 10 or 15 different shows on a tour. The odds are that out of several performances, you can find what you're after. TV shows don't have that latitude. Production costs are just too high to warrant that kind of approach. To meet all these technical and artistic requirements, a special crew of technicians was assembled for what the audience and the artists perceived as an effortless day of great music in an intimate environment.

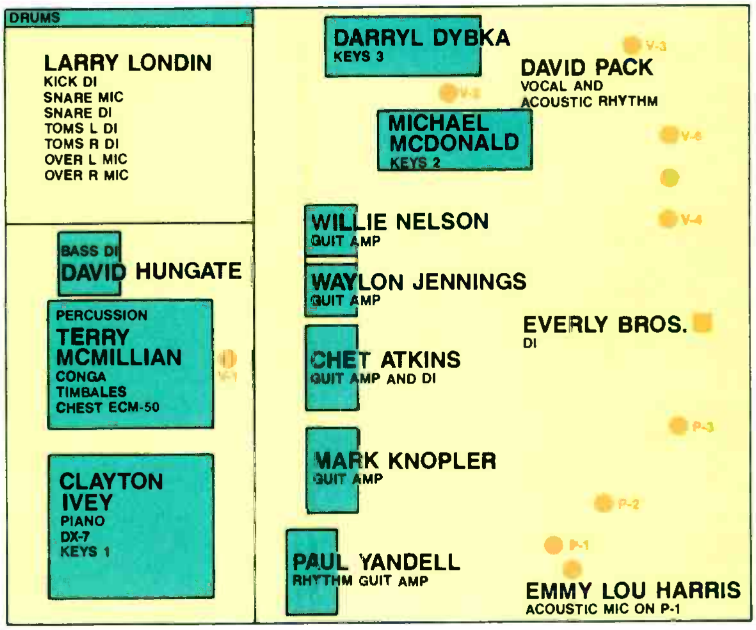
The venue was Neely Auditorium at

Johnny Rosen is owner of Fanta Professional Services in Nashville.

Vanderbilt University in Nashville. This small, black-box, multiform, theater was perfect. The acoustics were pretty dead and the audience could be placed anywhere. No seat was more than 30 feet from the stage. That meant the sound reinforcement system would not have to be large and that the sound of the instruments from the stage would seem a little more natural. The importance of the house and monitor mixer was paramount. The mobile audio facility was Fanta Professional Services, and the video facility was The Nashville Network, Unit 1. The main audio crew consisted of Mark Repp, Johnny Rosen, David Palmer, Robin Victor, Shipley Landis, Jamie Shankland and Billy Saurel. Here is how the jobs broke out.

Mark Repp: In charge of the audio inside the Nashville Network Unit 1 truck and the recording of all audience and effects microphones. Remember that the audience was very visible in this show, and their reactions were important to the overall feel. The live stereo mix had to be available for production personnel and had to be shipped to all the recorders.

All video machines had three audio tracks; the same mix was not sent to every



P-# = PRODUCTION MIC FOR DIALOGUE AND SINGING (USUALLY PLACED UP FRONT)
V-# = VOCAL SINGING MIC
BV-# = BACK-UP VOCAL
KEYS-# = KEYBOARD (USUALLY A DI)
A-# = AMP ON STAGE
PERC-# = PERCUSSION
DI-# = UTILITY DI (USUALLY MOVES)

Figure 1. *Generic stage plot.*

REC	PA	SNAKE	INPUT DESCRIPTION		SUB MSTR	TYPE
1	1		KICK	LARRY LONDIN		DI
2			SNARE	SUB-MIX C12-1		
3			LEFT DRUMS	SUB-MIX C12-2		
4			RIGHT DRUMS	SUB-MIX C12-3		
5	5		BASS	DAVID HUNGATE		DI
6			PERCUSSION	SUB-MIX C12-4		421
7	6		RHYTHM GUIT AMP	PAUL YANDELL		421
8	7		GUIT AMP	MARK KNOPLER		421
9	8		GUIT DIRECT	CHET ATKINS		DI
10	9		GUIT AMP	CHET ATKINS		421
11	10		V-1	TERRY MCMILLIAN		SM-58
12	11		V-2	M. MCDONALD		SM-58
13	12		V-3	DAVID PACK		SM-58
14	13		V-4	BACKGROUND VOCALS		SM-58
15	14		V-5	BACKGROUND VOCALS		SM-58
16	15		V-6	BACKGROUND VOCALS		SM-58
17	16		P-1	PRODUCTION 1		SM-58
18	17		P-2	PRDDUCTION 2		SM-58
19	18		P-3	PRODUCTION 3		SM-58
20	19		ACOUSTIC RHYTHM	DAVID PACK		ECM-50
21	20		KEYS 1	CLAYTON IVEY DX-7		DI
22	21		KEYS 2	M. MCDONALD		DI
23	22		KEYS 3	DARRYL DYBKA		DI
24	23		ACOUSTIC MIC ON P-1			
25	24		GUIT AMP	WAYLON JENNINGS		421
26	25		GUIT AMP	WILLIE NELSON		421
27	26		ACOUSTIC GUIT	EVERLY		DI
28	27		PIANO MIC	CLAYTON IVEY		
29						
30	28		SPARE MIC	CAMERA LEFT		SM-57
31	29		SPARE MIC	CAMERA RIGHT		SM-57
32			TONE GENERATOR			
33	2		SNARE MIC	421	C12-1	
34	3		SNARE DI		C12-1	
35			TOMS LEFT DI		C12-2/3	
36			TOMS RIGHT DI		C12-2/3	
37			OVERHEAD LEFT	SCHOEPS CARDIOID	C12-2/3	
38			OVERHEAD RIGHT	SCHOEPS CARDIOID	C12-2/3	
39			CONGA	421	C12-4	
40			TIMBALES	421	C12-4	
41			CHEST	ECM-50	C12-4	

RETURN LINES TO THE STAGE

#	PA	SNAKE		LEVEL
44	4		DRUM MIX FROM FANTA TO PA	+4
52			FANTA TALKBACK TO SPKR-AMP	

TIE LINES TO REAR PATCH PANEL

JACK	SNAKE		JACK	SNAKE		JACK	SNAKE	
1	1	LEFT MAIN	3	3	LEFT SPARE	5	5	L. RETURN
2	2	RIGHT MAIN	4	4	RIGHT SPARE	6	6	R. RETURN

TRACKS

1 KICK	2 SNARE
3 LEFT DRUMS	4 RIGHT DRUMS
5 BASS	6 PERCUSSION
7 PAUL RHYTHM	8 KNOPLER GUIT
9 CHET DI	10 CHET AMP
11 V-1	12 V-2
13 V-3	14 V-4
15 V-5	16 V-6
17 P-1	18 P-2
19 P-3	20 RHYTHM DAVID
21 KEYS-1 & PIANO	22 KEYS-2 MCDONALD
23 KEYS-3 DARRYL	24 TIME CODE
25	26
27	28
29	30
31	32

FANTA
PROFESSIONAL SERVICES
1213 18TH AVE. SOUTH
NASHVILLE, TN 37212
615-327-1731
JOHNNY ROSEN

PRODUCER	SUSAN HACKNEY
	CINEMAX/HBO
	CHET ATKINS +
5/2/87	F# F#5308
MIXER	JOHNNY ROSEN
STAGE BILLY SAUREL	TAPE ROBIN VICTOR
A-28	
TAPE TYPE AND LEVEL	456 250nWb +3
NR	# OF REELS

AS OF DATE AND TIME BY	6/1/87 7:47 AM

Figure 2. *Track layout sheet.*

machine. Some got stereo, some got just the audience or just the wireless lavalier on Chet. Mark also had to place and record all the audience microphones used for the original stereo mix onto the videotapes. A 24-track recorder was used, and each of the 18 microphones was recorded on a separate track for later use in post-production sweetening.

Johnny Rosen: The audio engineer in charge (A-1). Usually an A-1 develops the audio plan, mixes the show and everyone follows along. The crew on this show didn't require that much supervision. Instead, the Fanta office became the communications hub for all the audio people. Having a single person for all other artistic and technical teams to communicate with simplified the process and avoided redundant and frustrating meetings. Johnny mixed the live recording, which was recorded on two analog 24-tracks, and sent a simultaneous stereo mix to the video truck for distribution to all the various video machines. That ranged from 1-inch C format to 8mm home video format.

David Palmer: David had previously mixed an Earl Klugh-Chet Atkins album that was very successful. After listening to the album, we were sure that he would be an asset, but didn't know how to use him. Though he was not a regular with the Nashville crew, he was willing to fit in and find a special niche that wasn't being covered by other members of the audio team.

When he arrived in Nashville, he started by hanging out with the musicians. He had worked with several of them but never all of them as one group. Starting with the rehearsals at Studio Instrument Rentals, Nashville, David made elaborate notes about every song, from both the technical and artistic angles. Eventually these notes would help the rest of the audio crew the time to concentrate on their individual chores for the show.

The job that David finally filled was remixing the show for the final post-production audio mix. He had followed the music from the beginning of the rehearsals through to the live recording. Because he did not have to mix the live recording, he could concentrate on the artistic, rather than technical, qualities, which allowed him to emulate the live feel better in the final remix. Sometimes, when you are tracking a live show, it is very difficult to listen for the musicality of the songs or the overall energy in the performance. David could concentrate on the music while Johnny made sure that the tracks and the live mix were cut correctly.

Robin Victor: Coordinated all of the audio people and made sure that all of the logistical efforts flowed in the same direction. She kept track of all the audio team members by phone, pager, mobile phone and walkie-talkie. She documented every audio change in the production. As each deviation from the original plan occurred, she evaluated who needed to know and what people were affected.

The notes that Johnny, David and Robin made were all complied into an Audio-Master notebook. Various parts of the notebook would later be distributed to the crew for use during the show and in post-production. If a problem affected other parts of the crew, Johnny would work out the details with each group. That way, other production members only had to deal with one person if problems arose. We wanted to avoid multiple answers to the same questions.

As the show was being rehearsed, Robin called cues to the audio team via a headset intercom system. Usually changes during the show happened out on the stage and were supervised by the audio stage manager. All changes were logged in her book. Having a person at the location, dedicated to handling communications and documentation proved essential. Working from her updated Audio-Master notebook, she called the audio cues during the live recording. The version of the Audio-Master that was used on the set was called "a short run-down."

Billy Saurel: As audio stage manager, he interfaced directly with the talent, sound reinforcement team, TV stage manager, stage hands and camera team. He handled all the microphone placement and audio-cable patching in the stage area. A major part of his job was to communicate the needs of the musicians to the monitor mixer. The audio stage manager needs to understand exactly how the multitrack audio and video recording is done. Sometimes he has to make fast changes in the plan; unless he understands every monitor mix and audio track, his decisions could prove disastrous. Improper tracking makes remixing nearly impossible, improper microphone technique yields horrible quality, and choosing an improper input could make mixing the monitors very difficult. Basically, if the talent doesn't have to think about the audio system, the overall quality of the show will be much greater.

Jamie Shankland: Jamie's job was to supply and mix the stage monitors and the house system. He was picked because of his experience doing TV shows and his ability to cope with an unusual mixing environment. His monitor system was small, loud and of high quality. He would listen to requests carefully and only act when necessary. Before he made any changes, he checked in with Johnny or Robin so that everybody was aware of any critical changes.

Shipley Landis: Past experience has shown that on every complex job, it is essential to have a utility person who can fill in for any of the other workers. His skills need to include mixing, production knowledge, electronics, audio/video interfacing and, especially, the ability to think fast and maintain a sense of humor while under fire. While interfacing three different communication systems, Shipley had to spot problems and alert the correct people. This job requires a jack of all trades, a master of all trades and a candidate for the "Henry Kissinger" diplomat award. We follow the philosophy that all of the equipment will break at a critical moment, and when that moment arrives, we need to know how to get around the problem rather than through it.

Equipment setup

The audio technical setup consisted of three main parts: the sound reinforcement system, the audio recording system and the audio tracks on the video recorders. The only part of the video system that matters here is how the sound was shipped to the video truck for its recorders and monitors.

All music and dialogue mixing was done in the Fanta truck and sent to the TV truck via a single audio snake. Coming back from the Nashville Network truck was a stereo signal that could come from many sources in the Nashville Network truck. The controller for that crossbar switcher, that fed audio and video back to the audio truck, was located in the Fanta truck and the signal that controlled it traveled on a single coax cable into the Nashville Network truck. The following were the channel assignments for the Fanta to TNN Snake A:

Line 1—left stereo mix

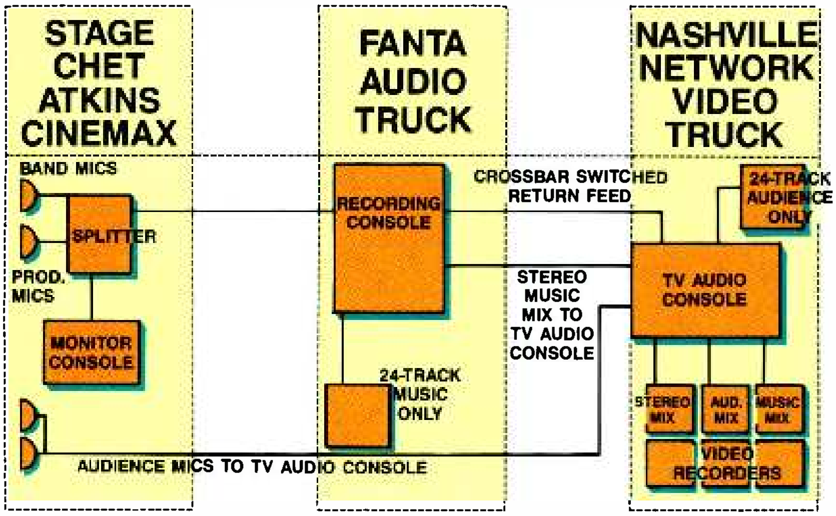


Figure 3. *Audio path for live recording.*

Line 2—right stereo mix
Line 3—left back up
Line 4—right back up
Line 5—TNN RTS intercom channel 1 = director, channel 2 = audio
Line 6—spare

Channel assignments for TNN to Fanta snake B (made of individual cables) were:

Line 1—left return line
Line 2—right return line
Line 3—spare
Line 4—output from return line crossbar switch controlled in Fanta truck (BNC coax)
Line 5—crossbar switcher control signal (BNC coax)
Line 6—SMPTE timecode

The stereo mix from audio truck was sent to the TV truck where it was distributed, along with the audience mix, to the various audio tracks of the video recorders. Obviously, we had to rely on Mark in the TNN truck to sweeten the live mix flawlessly because it really affected the way we cut the 24-track tapes. In this case, sweetening means the adding of audience and effects sounds to the stereo mix. We could hear the dry mix without audience by listening to the feed that left the audio truck, the sweetened mix by listening to the return feed from the TNN truck, or we could listen to just the audience mix by selecting the correct signal on the crossbar switcher. The advantage of doing it this way is that we had 23 tracks for audience alone that were recorded on a third 24-track and a mixer who could babysit them. Also, Mark was the engineer who would later have to sweeten the show when the final remixed tracks were laid back to the final edited video tapes.

Splitting the signal between the PA and the audio truck was accomplished with our 54-input × 108-output splitter box. We use Jensen transformers with three windings, one microphone input and two microphone level outputs. It is built into a steel Haliburton suitcase. Individual grounding switches are available for the recording and PA feeds. 48Vdc phantom power is always "on" for each input. There is also a master ground switch that can lift the entire splitter system from the truck's audio ground.

Microphone technique

Good microphone technique is the most important part of multi- track recording. Microphone technique for recording and PA is almost always slightly different. For this particular show, we leaned toward the needs of the recording, but usually the PA company has the most say when it comes to selecting and positioning microphones. The attitude of the audio crew was that we needed to cater to the talent. If we did that, all the other pieces would fall in line. It turned out that there were only a few special or unusual problems.

The microphone and monitor system for drums was a hybrid of Larry Londin's equipment and the regular system. Larry's drum kit can be configured in many ways. He used D-drums for kick and toms while using his Drum Workshop 7"×15" snare to trigger an electronic snare, as well as other sounds. All this is mixed through a 8×4×2 Soundcraft mixer. He sent us, via Countryman direct boxes, a stereo drum mix minus kick and snare, a separate kick mix and a separate snare mix.

The cymbals were miced using two Schoeps SKM-541 cardioids overhead. We rarely use large diaphragm condensers in live recording because we get better phase response with the smaller diaphragms. Larry mixed his own headphones and did not want a big loudspeaker for his drum monitor. He takes a special mix-minus from the PA system that has everything, including talkback from the audio and video trucks, minus his own drums. That special mix-minus from the PA was brought into an input of his mixer where he could balance his drums and a click track.

We were lucky that he would do it this way. Because there was no monitor speaker near him, we could vary the level of the over-heads without changing the apparent stereo perspective. It also helped us on snare because we could use a Sennheiser 421 instead of a Shure SM-58. The 421 is not as directional as the SM-58, and we could get by without a hi-hat mic. He played a lot of very soft cymbal parts and used his brushes often—the choice of microphones really made a difference.

Terry McMillian, the percussionist, is an enthusiastic and animated player. It was apparent from the rehearsal notes that he plays lots of different instruments with lots of different physical contortions. His congas and timbales were covered by 421s. All his other hand percussion toys were covered by an ECM-50 that he wore on his shirt. Having him wear the microphone provided us with one in-phase input that was always in the correct place. Terry understood how this worked and used good playing technique for the microphone he was wearing. His harmonica and vocals were handled with one SM-58 on a boom. All the percussion microphones were mixed to one track. The vocal microphone was assigned to a separate track. Again, the rehearsal notes came in handy.

Limiters and compressors

Limiters and compressors can ruin a live recording fast. We were careful not to do anything in the live recording that would prevent us from remixing and sweetening the audio. All the main production microphones were recorded through ADR Vocal Stressors. They have been our standard lead-vocal processor for years. This was necessary to avoid harshness from the monitors and provide good presence in the TV mix. Using the vocal stressor as a frequency-dependent limiter makes it appear to roll off the high end when it has to limit a lot. The monitors usually get harsh during loud passages; the vocal stressor makes that less apparent. All other vocals were compressed with dbx-160 units set at just under 1.5:1 ratio and almost no meter movement.

The bass was not limited in the recording chain. David Hungate has an elaborate rig, and we figured that he would set it up better than we could.

All the electronic keyboards were peak-limited with a Valley International Dynamites just to protect against runaway short-term levels. The acoustic piano mic was limited with a UREI 1176-LN set at 4:1 with the meter swinging 6dB to 12dB. The meters are very slow, so we weren't real sure how much limiting that was. If it sounded OK, we left it alone.

The kick DI was also limited using a 1176-LN. Larry often plays very fast parts on kick drum, so it was set for peak limiting with the fastest release time. It helped contain the beginning of a note without losing its body or the next note. The left and right toms from the on stage drum mixer were compressed with dbx-160s set on 6:1, with about 3dB to 5dB of meter movement.

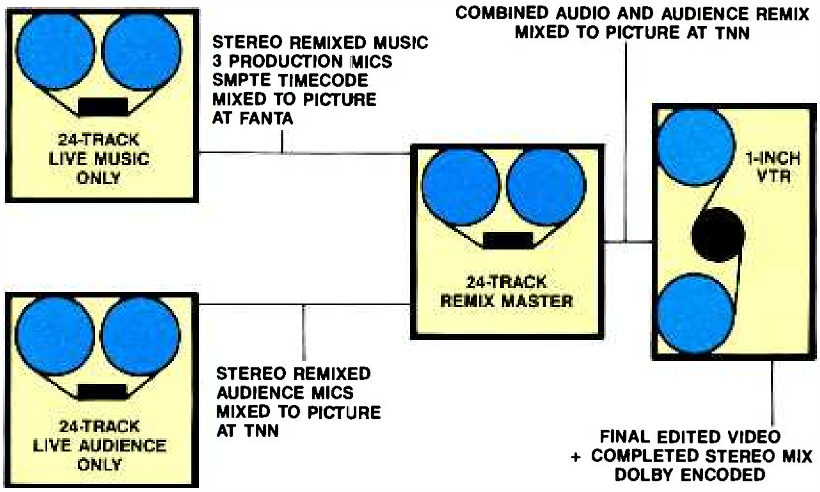


Figure 4. *Remix audio path.*

Effects

The effects used during the live recording were very simple: two Yamaha SPX-90IIs, one for vocals, set on preset program 1, and the other for drums, set on a preset random gate. An Eventide Harmonizer was used to get a delay of 30ms and 70ms from a mono electric guitar signal, just to open up the sound.

After the audience and effects microphones were added in the TV truck, the overall mix was run through an Aphex Compeller to smooth out the mix and make it easier for the editors. This allowed them to hear the soft passages more easily without having to deal with distorted loud passages.

The live mix

The mix of the live recording was the first thing that anyone would hear in post-production. If it was bad, it would affect the judgment of the editors. There was no time to remix before the editing began. After the show was edited, the new, remixed sound was added to the final video tape. Smoothness and simplicity got us through the original live mix. To avoid burning out the musicians, we decided to forego a long sound check. There was enough time during camera blocking to hear everybody play.

Each musician had played on stage many times, but they had never played together as a complete band. To encourage that mood, we felt strongly that all technical services should be as invisible as possible. The stage monitor system had to be physically unobtrusive, loud and very full-fidelity. The monitor mixing system would also have to supply the house sound. That worked fine in this case because the stage was at least as big as the audience area. (See Figure 1.) Finding a place to put the large monitor console, amps and peripheral equipment was hard because we didn't want to see the engineer and his setup on camera, yet he had to be able to hear both the house and the monitor mixes. He got stashed at the edge of the stage area with a direct view of everything.

He was a little too far away for quick access to the microphones, so that put extra pressure on the audio stage manager. We tried to keep motion around the stage and chatter on the intercom minimal. We were trying to use the KISS method, which stands for "keep it simple, stupid." We never identified microphones by their input channel numbers. This avoided any confusion about what we were working on because the monitor and recording inputs were different. On the track layout sheet (see Figure 2), notice that Chet's guitar amp mic is input #10 in the recording system and #9 in the monitor system. By just calling it "Guit Amp Chet," we all knew what we had to find. We used Fanta's layout sheets for everyone because it has spaces for both system inputs and the on-stage sub-snake assignments. If the PA people understood what we were doing, we could all work together better. A group of us that have worked together over the years have settled into this system of labeling because it is simple and functional.

The site

The show was taped about one week before graduation, at Vanderbilt University, which caused some special logistical problems. The university is particularly proud of its grounds and buildings. The theater we were using is in the middle of the campus, surrounded by large magnolia trees and manicured lawns. The problem was how to pull the audio cables into the building without damaging the landscaping. The audio cable run was about 500 feet and needed to cross walkways, flower beds and lawns. After considering our various options, we finally decided to run a guy wire through the trees that the cables could be attached to.

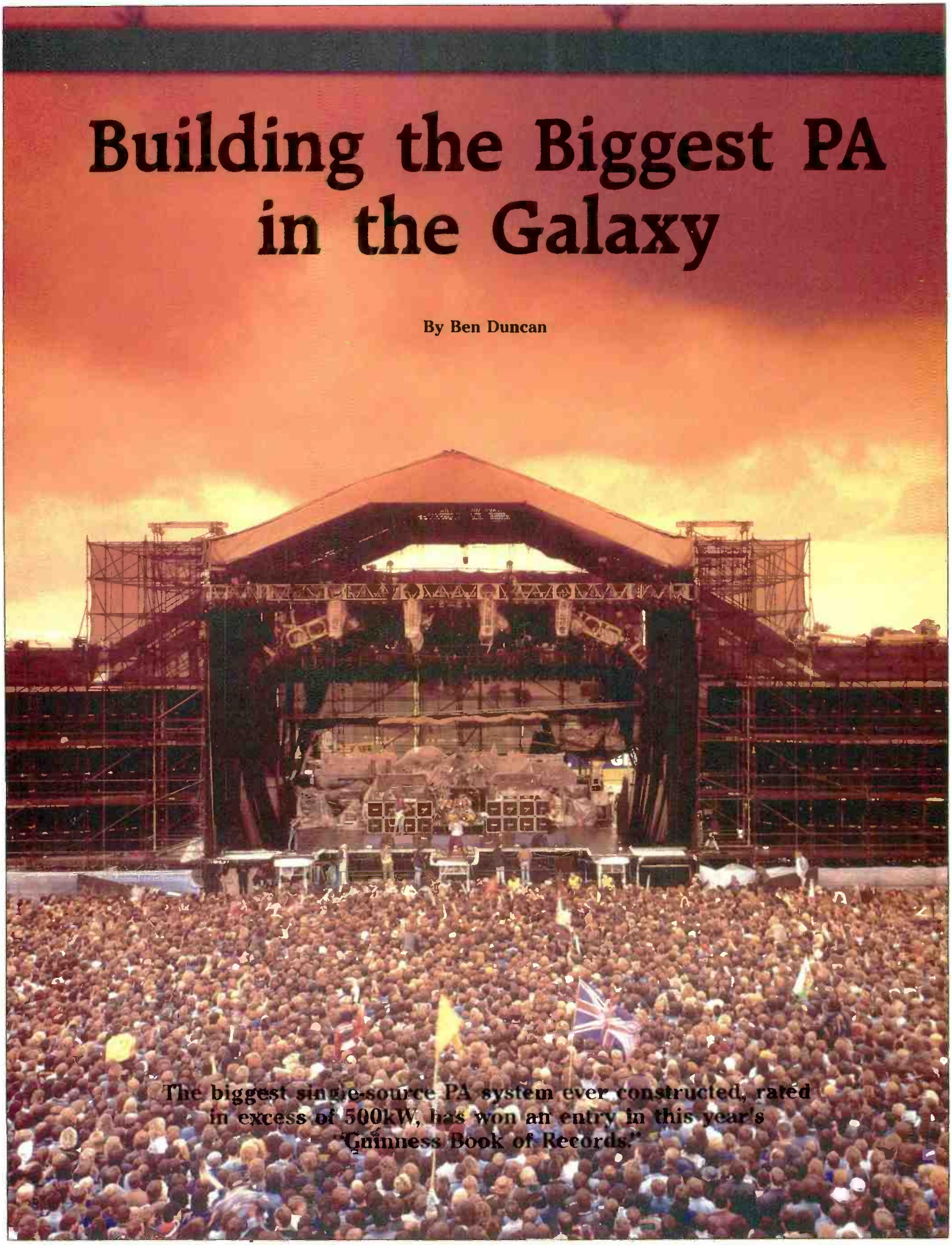
The power capacity of the building was quickly used up by the band's equipment, the lights and sound system. The audio truck, the main video truck and the extra tape machine truck were powered from a silenced, portable generator supplied by Citation Film/Tape Support. The unit had an operator with it whenever it was on and was available to us 24 hours a day if necessary. It had a crystal-controlled governor and very high current capacity, so the small changes in the load didn't affect the voltage or frequency. The only problem that we might have run into was a possible ground difference between the generator-powered equipment and the house-powered equipment. Fortunately, all the power and technical systems were wired correctly, so we had no problems. We attributed the lack of interfacing problems to detailed pre-production work and good communication.

Remixing this show was different from most TV shows. The show was planned around the audio, so there were great expectations. Chet lives in Nashville and would drop in from time to time to check on the progress. The producers and directors were editing all day, every day at Post Masters with Terry Climer. The producers/directors rarely made visual edits that would have an adverse affect on the music. The remix method was simple and easy to use. The original live tracks were cut on an Ampex 1200 24-track, and they were remixed to another Ampex 1200. The original tracks were locked to a Sony 5400 3/4-inch VCR with a BTX 4600 synchronizer, so that David Palmer could mix the show while looking at the pictures.

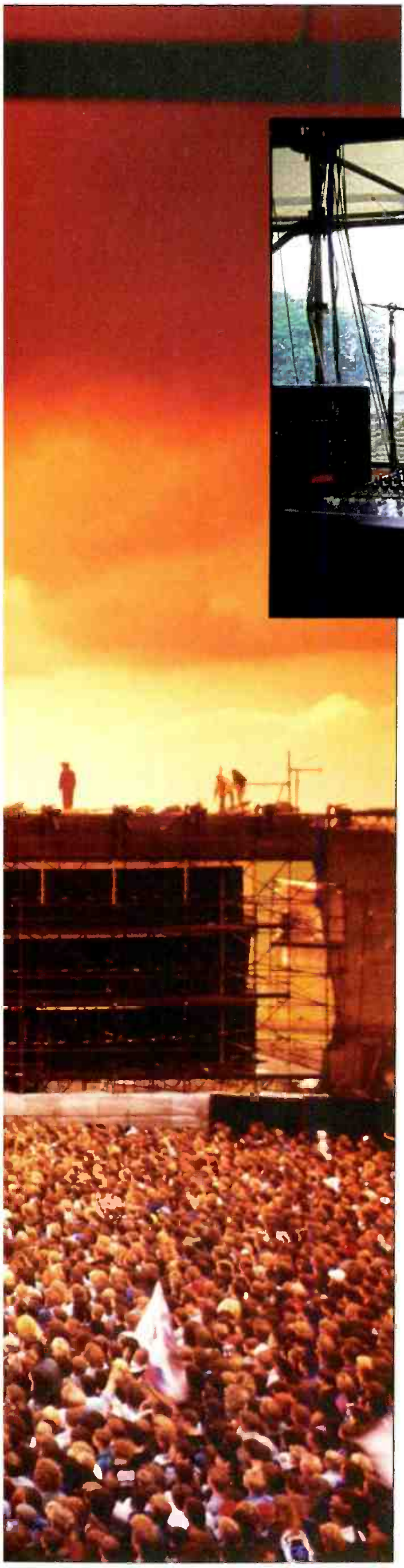
Continued on page 70

Building the Biggest PA in the Galaxy

By Ben Duncan



The biggest single-source PA system ever constructed, rated in excess of 500kW, has won an entry in this year's "Guinness Book of Records."



The motorsport racetrack at Castle Donington in the English midlands has been the location for a nine-year series of annual Heavy Metal, one-day outdoor concerts. At the last festival, on August 20th, 1988, eight bands played, headlined by Iron Maiden. They used the biggest single-source PA system ever constructed. Rated in excess of 500kW, it's won an entry in this year's "Guinness Book of Records."

Throughout the 1980s, UK touring bands have had to weather (for diverse reasons) tough economics. Most of the 1970s headline bands that owned their own PA systems have long ago "realized their assets."

Iron Maiden is an exception. In 1983, after two years of successful touring with Turbosound's own PA rental company, they decided to buy the rig they were then using, including 24 TMS-3 boxes for the main house system.

At the time, the idea of investing in such a huge (and partly "second owner") PA would have looked outrageous to record company accountants. However, Iron Maiden's management had some massive world tours planned. Their decision has paid off because over the past six years, the band has averaged seven out of every 12 months on the road—including one two-year period in which they toured continuously for 13 months! Since then, the original PA has grown substantially to include 100 TMS-3s and 24 TSW-124 subwoofers.

Doug Hall, Iron Maiden's house engineer, saw at Donington the opportunity to use their PA for the whole event, but knew they'd need more equipment. "I like a system to exhibit comfortable headroom, about three-quarters of its total output," he explained.

Logistics

Dick Bell, Iron Maiden's production manager, then contacted Mike Low at Britannia Row, a major UK PA rental company, who was coordinating the sound at Donington. Mike set about arranging a further 400kW array with the help of Samuelson's Concert Productions (another UK rental company that, incidentally, was bought by Britannia Row the week after the concert).

Extra cabinets and amplifier racks were brought in from Regiscène in France and Ampco in Holland. Meanwhile, in Brit Row's warehouse, a section of the planned

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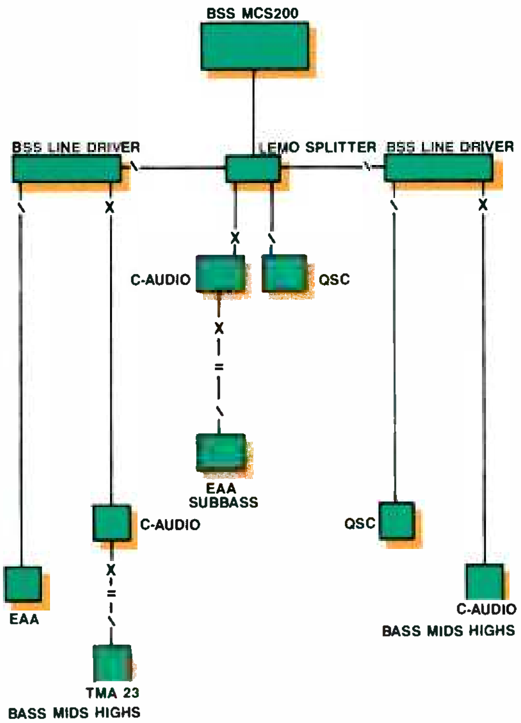

Figure 1. *General interconnect scheme.*

Table 1. *House PA amplifier checklist.*

Quantity	Model	Source
72	QSC 3800	Iron Maiden
102	C-Audio SR-707*	Samuelsons
6	Turbosound TMA-23**	Samuelsons
18	EAA 1000	Regiscène
6	Carver PM 1.5	Regiscène
14	HH V800	Ampco

* modified SR-606s
** Each tri-amp has three stereo channels.

total = 460 amplifier channels in use.

array was evaluated to simulate the intended coverage two months beforehand.

The setup

On-site setup took five days. Monday was spent erecting the main speaker array. The consoles, stage monitoring and the single delay tower array were rigged on Tuesday. Wednesday was "tech day": The system was fired up, hums were exorcised, the polarity of more than 1,700 individual components was checked, and the cabinet array in the PA wings' upper tier was adjusted. Thursday and Friday were reserved for sound checking for Iron Maiden and the support bands. This meant the instrument backline could be left in place backstage, positioned in order of appearance.

Before 1986, Iron Maiden guitarists Dave Murray and Adrian Smith had used the traditional Marshall amp stack (the kind that's permanently cranked up to maximum), together with daisy-chained FX pedals. The sound was neither clean nor quiet enough, so Hall generally used a clean feed from the two guitarists by micing individual cabinets backstage. At Donington, however, stacks of Galen-Kruger amps and a rack of TC Electronics guitar FX cleaned up the sound sufficiently for Hall to mic the on-stage amps directly.

The mics used for Iron Maiden included a Shure SM-7 on bass guitar; Shure SM57s for the two guitars; an SM98 on the hi-hat and another on the tubular bells; AKG D-112 on the kick drum; Neumann KM-84s over the kit, together with SM57s for the remainder. Sony diversity wireless mics were used for the vocals. All the mic lines were fed into BSS MSR-604 active microphone splitters, which provided buffered feeds for the three house and three stage monitor consoles, as well as an isolated feed for the BBC's recording truck. A multicore was also laid from the house consoles to provide effects returns for the BBC. From the splitter rack to the house consoles were three snakes apiece, each with 19 pairs, supplying 3×40-channel capability, with spare pairs.

As usual, the aim was to keep show downtime to a minimum, with each successive instrument/amp array and drum-riser fully miced and waiting behind the backdrop. In theory, it was only necessary to plug the multicore into the next stage box. The actual duration of the seven changeovers ranged between 20 and 25 minutes. To achieve the rapid changeover while keeping Iron Maiden's own control gear intact required an on-stage crew of 24 and three substantially independent monitor systems. The support bands' monitoring alternated between a pair of Soundcraft series 4, 40/16 consoles (supplied by Samuelson's), with their own EQ racks, and Carver PM 1.5 amps. John Shearman and Ed Wilson from Samuelson's did alternate duty on the support bands' monitor mix. Alongside was Maiden's own 24/10 Midas, with a third set of control gear. Their power amplifiers were mostly QSC 3800 (lo + mid), with a few Turbosound Fan Amps, and Turner B502s (on the vocal wedges).

The support bands "A-B"ed between two sets of custom stage monitor wedges, designed by Pete Brotzman when he worked at Turbosound Rentals. They all used TMS-3 side- and drum-fills from Iron Maiden's 40kW monitor system. Maiden used their Turbosound vocal wedges along with vocalist Bruce Dickinson's sidefills, which are rated at 12kW and combine a pair of Martin 215 bins with Community's M4, a 4-inch exit mid-range compression driver noted for its ability to deliver crushing SPL.

House control

In the mix tower were three consoles, each with its own outboard gear. Iron Maiden used its own Amek M1000 48/8/2, with custom modifications and a 16-input add-on stretch-frame for the drums. The support bands alternated between two Soundcraft Series 4, 40/16/2.

Hall's extensive FX racks contained Eventide SP2016 (used on the drums); Eventide 949 and 969 Harmonizers; dbx 160 and 165 compressors; two ADR Compex limiters; two UREI 1176 compressor-limiters; an Aphex C; an AMS RMX-16; an ADR vocal-stresser and dbx's RTA-1 analyzer. Hall also used a Revox B-77 for intro sequences and voice-overs. He says, "Iron Maiden does a lot of their own FX on stage. Being outdoors, I'm using some general 'small room' ambience to add some lushness. I compress the bass with the UREI 1176, followed by the ADR Compex and some post EQ. I compress the vocals by 5dB, so when Bruce goes berserk on the mic, he doesn't kill people!" After 8 years of touring, Hall knows the cues well enough to mute all the unused mics in a flash. This minimizes spillage from Maiden's enormous monitoring and on-stage amp array.

The main PA

Table 1 shows the 460 amplifier channels in use on the two PA wings. These were divided between stereo amps, half-used stereo amps and some TMA-23 stereo tri-amp units (a Turbosound market research product no longer manufactured). Owing to their diverse sourcing and the differences in rental companies' interconnect standards, the amplifiers needed very, very careful matching.

John Newsham from Turbosound and Julian Tether (then Samuelson's chief technician, now working for Britannia Row) spent an evening religiously checking and correcting sensitivities and normalizing the polarity of every amplifier. Five more hours were spent adjusting the system grounding to reducing residual hum and noise.

At Hall's insistence, the 400+ amplifier inputs were driven from a single BSS MCS series crossover. This approach promised sonic coherence, but with incompatible amplifier racks supplied by four different companies, it also presented a nasty interface headache. *Any* failure would silence the main PA outright. Stress on the crossover outputs was avoided by using BSS's newly developed high-power line-driver "booster" package, made to plug into the MCS series crossover "mainframe." It was first devised in association with Concert Sound to drive 200+ amplifier channels at the Nelson Mandela birthday bash. At Donington, two line drivers were placed at the stage-end of the returns

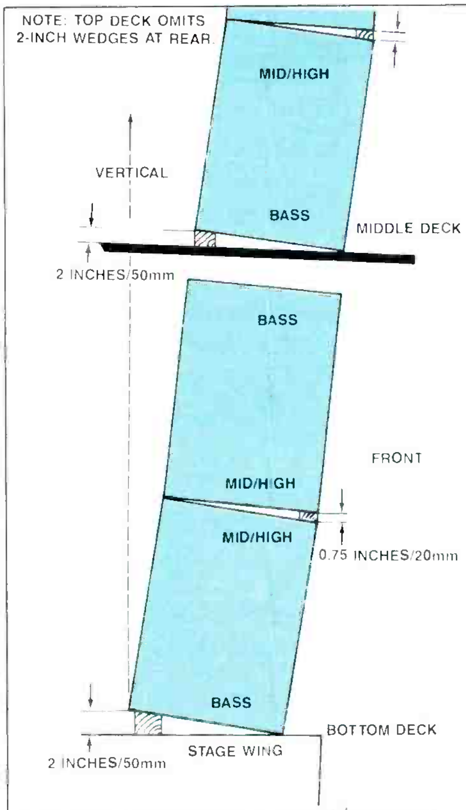

***Figure 2.** Every cabinet was tilted down with lumber wedges acting as "acoustic compensators." Smaller wedges at the front fine-tuned the coupling between adjacent mids.*

multicore. Figure 1 shows the general interconnect scheme.

The main house system comprised 304 Turbosound TMS-3 enclosures. Each 72-foot stage wing (see Photo 1) had three tiers. Each tier had 48 Turbosound enclosures in two rows, totaling 144 per wing. The remaining 16 were placed under the front of the stage, with a separate vocals-only mix to balance the plainly audible instrument amp stacks and side- and drum-fill monitors. In the wings, the enclosures were stacked vertically: first, to gain the longest throw from the HF horn, and second, to close-couple the 10-inch mid drivers. On the ground, under the wings, were 60 Turbosound TSW-124 subwoofers, all arrayed for maximum coupling.

The TMS-3 is normally configured as a medium-throw "box." Nevertheless, the capabilities of this giant array were clearly proved during initial sound-checking—by the PA's clarity in a large village 2 miles away.

At Donington, flying the PA with the available scaffolding would have left a big gap between cabinets, ruining the coupling. Instead, every cabinet was tilted down (see Figure 2), with lumber wedges acting as "acoustic compensators." Smaller wedges at the front fine-tuned the coupling between adjacent mids.

Each tier was built 18 inches in front of the next, while the enclosures in each wing were arrayed in an arc, receding 10 feet at each "wing tip," to produce a pair of phase-coherent, virtual point sources. This allowed a relatively constant SPL to

"With this system, you can get the level without distortion that hurts your ears."

be maintained over the 60-meter span, back to the mix tower.

The small delayed PA, located immediately behind the mix tower, was rated at 50kW. Because of the more-than-adequate bass and subbass projection from the main system, the delay tower's low end was barely "ticking over," and it needed no subwoofers. John Newsham said, "It's not there to start the sound all over again, it's there to give a gentle lift to the mids and highs." A look at the site

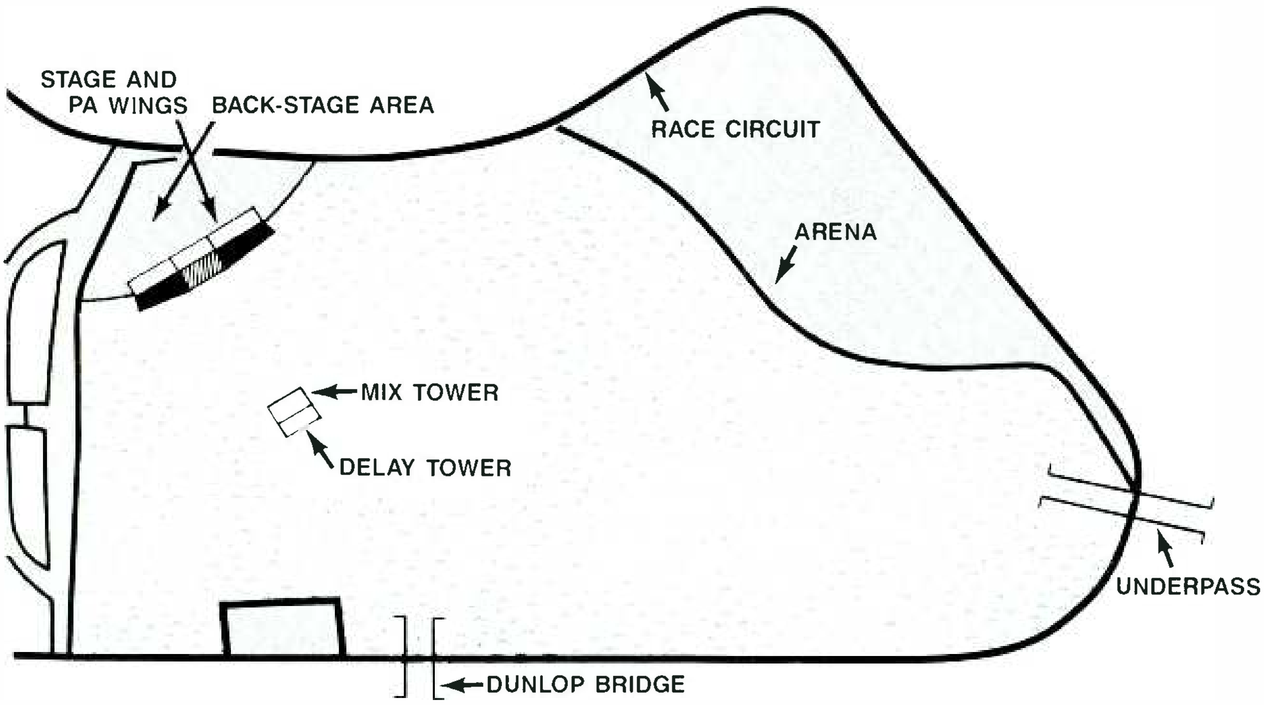

Figure 3. *Castle Donington site plan.*

plan (see Figure 3) shows why: The land rolls away from stage-left into bowl-shaped dip, then up into the main bowl and goes on rising gently to a ridge well beyond. Donington's infamous "dog leg" means a major part of the audience is well off-axis from the stage.

The delay array consisted of 30 MSI (Maryland Sound Industries) HF and 16 LF cabinets. In addition, eight JBL long-throw horns at the top of the tower provided additional reach for the HF material. This array used the stage-right mix to "fill out" the stereo image 500 yards from stage-left.

During Iron Maiden's set, the instantaneous peaks reached 124dB at the mix tower.

This is a technique first applied by Malcolm Hill, a UK PA rental company that has provided sound at Donington on past occasions. The delay tower components were driven off their own crossover, through a Klark-Teknik DN-700 delay line. Stereo balance, horn aiming and delay times were first estimated; the delay was 1ms/foot. With sound crew radioing back the results from a golf-cart, the fine-tuning was accomplished fairly easily.

The event

The warm day began with hazy skies and later clouded over. By 6 p.m., attendance had reached 107,000. The inevitable rain came after nightfall, during Maiden's set. The good side is that wet air seems to improve the transmission of sound.

After six years on the road and literally thousands of gigs to his credit, Doug Hall (who's been with Iron Maiden since their first UK tour in 1980) said, "I don't want a PA to sound like 'big speakers.' For me, it should be an extension of the stage, like it's coming straight from the band. The PA should disappear. With this system, you can get the level without distortion that hurts your ears." The philosophy is also supported by Motley Crue's engineer Mark Dowdell.

Since the support bands' sound peaked at between 6dB and 8dB below full output (which means one-fourth power, or about 125kW), the remaining headroom was more than comfortable. Under these conditions, the short-term rms SPL, monitored at the mixing tower, was 118dB. It was 108dB at the Dunlop bridge, 185 meters (more than 606 feet) from the stage, and nearly on-axis.

During Iron Maiden's set, the short-term rms SPL was similar, but there were more dynamics; instantaneous peaks reached 124dB at the mix tower, compared to 119dB during KISS's set. 124dB corresponds to the system being driven briefly to within 0.5dB of full power.

What was achieved? Well, I believe that this is the first time a group of Heavy Metal bands have played (in the open air to more than 100,000 people) with enough power headroom to play *loud* and develop a peak-to-mean ratio (i.e. "dynamics") of more than 10dB—without stunning the limiters.

RE/P

Coping with Wireless Microphone Systems

By Bill Mayhew

Help is here for those who need to overcome wireless system difficulties.

Wireless microphones have become a convenient way to deal with many difficult production situations involving actors and musical artists who need freedom of movement or concealed audio pickup. Setting up a single wireless mic is a relatively easy task. Yet behind the scenes, in the invisible world of RF, trouble lurks.

Outside interference

To start, let's talk about interference that is not caused by other wireless microphones being used at the same time and at the same location. For lack of a better term, let's call it *outside* interference.

Most people would say that you should have no trouble if you have taken the usual precautions in selecting the transmit frequencies (for the given area) of the wireless mics you are going to use. The real trouble is in the use of the word *usual*. Most manufacturers design their wireless mics to operate in the frequency group between 154MHz and 216MHz. Of course, within this group of frequencies are walkie-talkies and the VHF TV channels. Table 1 shows the relationship between VHF channels 7A through 13F and the licensable wireless frequencies.

The unused TV channels within a given area are a good place to start when selecting a workable frequency. I've found that if you are further than 75 air-miles from a station's transmitter, you should be able to use wireless within that channel's bandwidth. For instance, in the Los Angeles area, the unused channels are 8, 10 and 12. The problem is that if you were to take a wireless that was set on the frequencies for 8, 10 or 12 to San Diego, it would not work because the San Diego area has TV stations broadcasting on channels 8, 10 and 12.

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Happily, the FCC has opened up a new group of frequencies that allow for clear-channel operation within the United States: 169.445MHz, 169.505MHz, 170.245MHz, 170.305MHz, 171.045MHz, 171.105MHz, 171.845MHz and 171.905MHz. (At least, I think that was the intent of the group that convinced the FCC that clear-channel wireless mic frequencies are really needed. The truth is that the FCC took the list of frequencies submitted and changed it a bit.)

At first glance, we all yelled, "Eureka!" But then we realized that the spacing was far too close. In fact, you can only get three of the above frequencies to work together at the same time. The rest of the bad news is that in some areas of the country, wireless users have had interference on these frequencies, too. So far, in the Los Angeles area, I have found the following frequencies to be free from interference: 169.445MHz and 171.105MHz. The reason these new frequencies don't work out as planned is that the spacing between frequencies is for walkie-talkies, which is too close. The FCC didn't take the wireless mic deviation of 15kHz into account.

What interference sounds like

If you are lucky, the interference will be "understandable" in your wireless receiver; you will be able to hear the station clearly enough to get the call letters of the station that is causing the interference. With this information, you or your service tech will be able to tell if you have a fault with the wireless or if you are, in fact, on the wrong frequency for a given area.

Most of the "outside" interference in wireless microphone receivers comes blasting through—usually causing the receiver's VU meter to peg. It's usually a commercial TV broadcast station (and to a lesser degree, FM radio) that causes the interference. Since they use FM modulation with a deviation of 75kHz, you can see why the wireless mic receiver (12 to 15kHz of deviation) really grabs on to the outside signal.

To see if you are going to have outside interference, a good on-site test is to turn on the receivers and open the squelch. Now listen to the spectrum noise. If you hear a carrier, you might want to drag out that old mic cable.

If you suspect outside interference and you enjoy playing Russian Roulette with a shotgun, there are a couple of things you can do. First look at the receiver's meter. If the interfering carrier signal reads higher than −20VU in the RF position (meters vary by brand), you will probably have trouble. Even though FM wireless mics work on the capture principle, (the strongest signal is captured by the receiver), this is not much help because your transmitter is moving about the performance area. How strong must a signal be to interfere? I found that a 2μV (0.000002V) can cause interference problems.

With movement, the transmitted signal is being received at varying strengths by the receiver. During a period of weak reception, the interfering signal could be received by the wireless receiver. If the interference is −20VU or less at the receiver, try moving the receiver a few feet. This could make all the difference in the world.

The idea is to find a location with the least amount interference. You might also try different antenna polarizations. Sometimes a 45° angle works well. Remember that the wireless transmitter and receiver antennas must be polarized in the same direction (See Figure 1.) There are no hard and fast rules for polarizing a hand-held system. Typically, a 45° angle works best. With an antenna diversity system or a switching diversity system, one antenna should be aligned vertically, and the other at 45°.

Antenna diversity and switching diversity systems

To capture the strongest possible signal and minimize dropouts, two reception techniques are available: antenna diversity and switching diversity. In an antenna diversity system, typically two or three antennas are employed, each positioned in a different dimensional plane. The antennas are combined in an antenna combining box/splitter, which feeds up to four standard radio receivers. As the transmitter moves about the performance area, the presence of a strong signal is reasonably assured at one of the antennas.

With antenna diversity, the signals are not switched as they are in a switching diversity system. With careful antenna placement, the antenna diversity method can provide the best possible coverage, no switching noise and a minimum of potential phase cancellation problems. Also, this method generally has no limitations as to brand type and is less expensive than a switching diversity system.

In a single switching diversity receiver there are two antennas. The RF signal-strength from each IF amp controls the audio switch. When one side of the switching diversity receiver senses a weak RF signal, the switch switches to the other side of the receiver, which, at that instant in time, should be receiving a strong signal. (See Figure 2.) While switching diversity systems have a convenience advantage, they are generally more costly and slightly less reliable than a carefully installed antenna diversity system.

Another obvious (but often overlooked) key to successful wireless operation is maintaining a minimum distance between the transmitter and receiver. This helps ensure that the receiver will only "capture" its transmitter and not the interference. One last point, make sure that the receiver is turned off when the transmitter is not in use. This prevents unwanted transmissions from getting into the receiver and possibly leaking through the audio system.

A very common misuse of wireless systems occurs when the engineer thinks the receiver should be at the console. While this may or may not be important, the antenna most certainly should not be there. The antenna and receiver should be as close to the performing area as possible. This means there will be either antenna coaxial cable or audio lines going to the stage. "Wireless" refers only to the signal path between the performer's mic and the antenna.

Frequency selection

If all you need is one or two wireless microphones, then the frequency selection process is quite simple. The problem becomes more complex when you require many more channels. They have some

Table 1. *VHF TV channels and the corresponding wireless frequencies.*

Channel	Frequency (MHz)
7A	174.8
7B	175.4
7C	175.8
7D	177.0
7E	178.0
7F	178.8
8A	80.8
8B	181.4
8C	181.8
8D	183.4
8E	184.0
8F	184.8
9A	186.8
9B	187.4
9C	187.8
9D	189.4
9E	190.0
9F	190.8
10A	192.8
10B	193.4
10C	193.8
10D	195.4
10E	196.0
10F	196.8
11A	198.8
11B	199.4
11C	199.8
11D	201.4
11E	202.0
11F	202.8
12A	204.8
12B	205.4
12C	205.8
12D	207.4
12E	208.0
12F	208.8
13A	210.8
13B	211.4
13C	211.8
13D	213.4
13E	214.0
13F	214.8

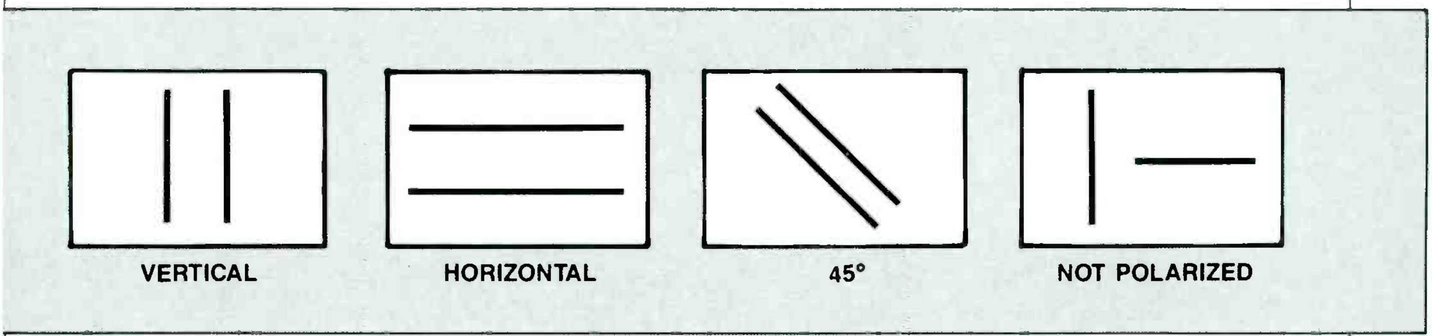


Figure 1. *Antenna polarization options.*

degree of compatibility, but as I said before, the problem can get quite complex.

If you consider IF frequencies, bandwidth, spurious emissions and image rejection, the problem becomes apparent. The sidebar "Frequency Selection" on page xx shows a formula that allows up to 22 wireless channels to be used at the same time, in the same location. To do the math long hand involves thousands of addition, subtraction and comparison problems. Of course, with the aid of a computer, the tedious task is reduced to just entering the parameters you wish to use. If you are not up to doing the calculations yourself, the answer for frequency selection can usually be found with the dealers and/or manufacturers of the wireless equipment.

Squelch

The purpose of squelch is to mute the spectrum noise received when no carrier is present (similar to a noise gate). There seem to be two schools of thought when it comes to squelch controls in wireless microphones. The first is that the end-user should not have to adjust the squelch on the receiver, so the manufacturer puts the squelch control inside the receiver, away from the end-user. This is done to avoid potential misadjustment of the squelch by the end-user.

The second approach suggests that a preset squelch control does not allow for the most sensitive receiver. Further, it does not allow the operator to disable the squelch, if desired. Also, with a user-adjustable squelch, it is possible, at times, to squelch out faint interference.

One additional thought. You should readjust the squelch whenever a unit has been in for repair, when you change locations or if you have reason to believe that someone has changed the control.

The antenna system

Quite often I am asked how much coaxial cable can be connected to a wireless receiver to extend the antenna. I usually suggest 50 feet of RG-58 or 100 feet of RG-8. It seems that everybody has a different idea when it comes to coax length. In trying to find the standard losses of coax cable, I looked into a number of publications, all of which gave me a different answer. So I decided to try some measurements in the *real world*. I set up the test at home, in an open area to help reduce reflections.

First I connected a dipole antenna to our spectrum monitor using 3 feet of RG-58, which allowed me to polarize the antenna for maximum reception. Figure 3 shows a typical dipole antenna. The signal was measured with the transmitter placed 40 feet from the spectrum monitor. Then we disconnected the 3-foot section and in its place connected 25 feet of RG-58. To compensate for the loss in the additional cable, we had to move our dipole antenna 10 feet closer to the signal source. Figure 4 shows a graph of what we found.

This exercise shows that coax cable is better than air. It was quite surprising how little loss the cable had in this real-world application. What this means is that if you can improve your line of sight by relocating your receiver antenna, by all means, do it! As the data points out, a small distance closer to the transmitter more than makes up for the loss caused by additional cable length.

Now before everybody goes out to install coax, there are some finer points to consider. First, you should use a *dipole* antenna. If you just try to remote the whip antenna that came with your receiver, it usually will not have a good ground plane. Next, be very careful about the type of coax used. Appearance itself is no guarantee. Using an ohmmeter, you should read close to 0Ω center to center and also from shield to shield. You should read an open between the shield and the center conductor.

If you suspect trouble with the antenna, check the coax first—it may save time and money at the repair bench. Last but not least, select the location of your remote antenna carefully. It should be at least 2 feet from any metal, the coax leading to the dipole should not parallel the antenna elements and finally the antenna should be polarized to match the polarization of the transmitter antenna.

With all of the wireless microphone ad-

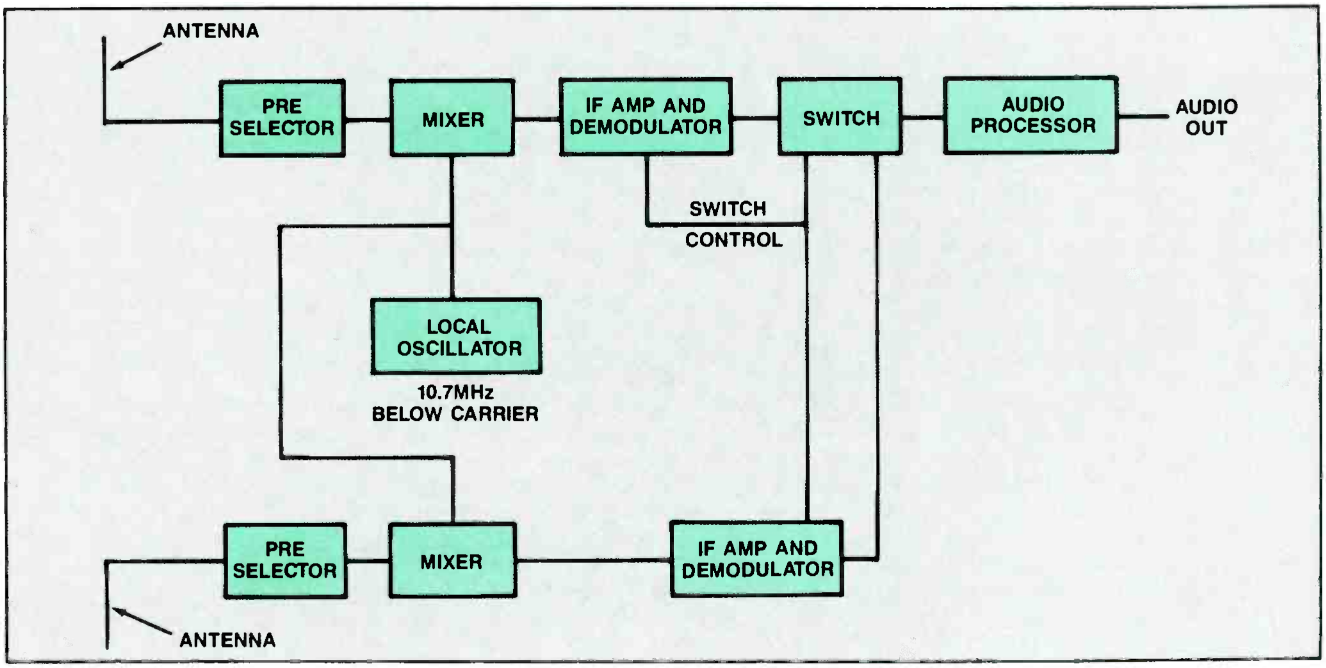


Figure 2. *A typical switching diversity receiver.*

Frequency Selection

The frequency selection formula reads as follows: IFI (intermediate frequency interference) equals *f* (frequency of the wireless) minus the IF, which is 10.7MHz, and products of the IF. Subtract multiples of the IF from the original frequency a total of four times (−IF, −2IF, −3IF and −4IF). Then add multiples of the IF (10.7MHz) to the original frequency four times (IF, 2IF, 3IF and 4IF).

I have chosen three regular frequencies to work with in this example. There is a column for each frequency (174.0MHz, 184.8MHz and 195.4MHz). (See Table A.)

The next step is to see if any of the frequencies in any of the columns are within 0.7MHz of each other. If they are, then you will have some problems. The wireless system will be unusable if the numbers fall within 0.7MHz of the frequencies marked with an asterisk (*), that are below the carrier frequencies marked with an *f*. In other words, the numbers in the three columns will all generate some interference; those marked with an asterisk definitely will.

It is important to understand that this formula only tells you the frequencies that will cause problems; it doesn't calculate the workable choices. As you can see, 174.0MHz and 195.4MHz will not work together, nor will 184.8MHz and 174.0MHz or 184.8MHz and 195.4MHz. The only variable in this is the amount of tolerance you are willing to put up with. 0.7MHz seems to work very well—however, I'm sure you can lower that number and still have some degree of success. In this example, it will be necessary to change two of the carrier frequencies to have three usable mics.

This method only takes into account the local oscillator and the harmonics of same. In doing so, though, it happens to take into account other methods and computations. This is a very conservative method of frequency selection—that happens to work.

Table B shows three frequencies that *will* work well together, providing there aren't other outside interferences to contend with. Notice that more than 0.7MHz separates the frequencies in this example.

Table A. *174.0MHz, 184.8MHz and 195.4MHz should not be used together in a wireless microphone system. A calculation of their harmonics shows there will be interference.*

216.8	227.6	238.2	(*f* + 4IF)
206.1	216.9	227.5	(*f* + 3IF)
*195.4	*206.2	*216.8	(*f* + 2IF)
184.7	195.5	206.1	(*f* + IF)
174.0	184.8	195.4	(*f*)
163.3	174.1	184.7	(*f* − IF)
*152.6	*163.4	*174.0	(*f* − 2IF)
141.9	152.7	163.3	(*f* − 3IF)
131.2	142.0	152.6	(*f* − 4IF)

Table B. *Harmonics show that 180.8MHz, 181.8MHz and 184.8MHz are compatible in a three-mic wireless system.*

223.6	224.6	227.6	(*f* + 4IF)
212.9	213.9	216.9	(*f* + 3IF)
202.2	203.2	206.2	(*f* + 2IF)
191.5	192.5	195.5	(*f* + IF)
180.8	181.8	184.8	(*f*)
170.1	171.1	174.1	(*f* − IF)
159.4	160.4	163.4	(*f* − 2IF)
148.7	149.7	152.7	(*f* − 3IF)
138.0	139.0	142.0	(*f* − 4IF)

The Ten Commandments of Wireless Microphone Use

1. Keep the distance between the transmitter and the receiver's antenna to a minimum.
2. Maintain a direct, unobstructed line of sight between the transmitter and the receiver's antenna.
3. Keep the antenna and mic cables on the transmitter separated.
4. Place the transmitter body pack as high up on the body as possible.
5. Have both the receiver and transmitter antennas polarized.
6. Place the receiver at least 4 feet high.
7. Keep the receiver a minimum of 3 feet from any other receiver or antenna.
8. Keep the receiver a minimum of 3 feet from any metal.
9. Use only fresh batteries.
10. Contact the manufacturer or a qualified consultant if you have trouble determining the usable frequencies.

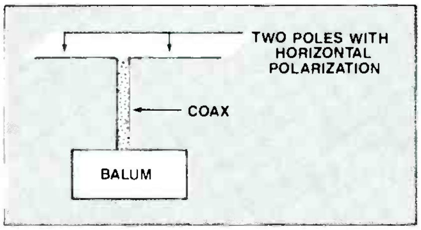


Figure 3. *A typical dipole antenna.*

vances that have taken place in recent times, a new gremlin has raised its ugly head to unforeseen heights. The gremlin is *clothing noise*. Yes, that's right; finally a gremlin that is not the wireless microphone's fault. A number of times I have had A-1s (video's terminology for first audio engineers) call me and complain about drop-outs. After investigation, more times than not, the problem proved to be clothing noise.

Ironically, with clothing noise, the prime cause is usually inexperience. Consider this: In both TV and motion picture production, the A-1 and/or the mixer is usually the most experienced person on the audio crew. In most production situations, the A-1 is removed from actually "dressing" the talent, i.e. attaching, placing and concealing the mic, transmitter and antenna in the talent's clothing. Therefore, dressing the talent is usually left to the A-2—or even a wardrobe person, whose main concerns are not audio quality. Figure 5 shows a "dressing" mic placement method that has proved quite useful.

Take a 2"×1/2" piece of gaffer's tape. This forms the cross arm of the T. A second 1/2"×3" or 1/2"×4" piece of gaffer's tape forms the staff of the T. Then place the omnidirectional mic onto the sticky side of the tape. The top of the mic should reach just to the top of the cross arm and align with the staff of the T. The mic cable should be dressed down the staff of the T. Then slip this entire mass of tape, microphone and cable under the talent's shirt, blouse or coat with the sticky side of the tape toward the outside of the garment.

To help eliminate microphonics caused by the cable, tie the cable in a snug half hitch. In effect, this dressing system causes the microphone to become part of the clothing. The clothing noise is greatly reduced because the microphone now moves with the clothing instead of against it. One word of caution: This system works well on most fabrics, but is a real mess on some synthetics. Use a small piece of gaffer's tape and test delicate or expensive fabrics *before* installing this system. The adhesive may cause irreparable damage.

Neon interference

I really can't think of any other word that strikes as much fear into the hearts of knowledgeable users as *neon*.

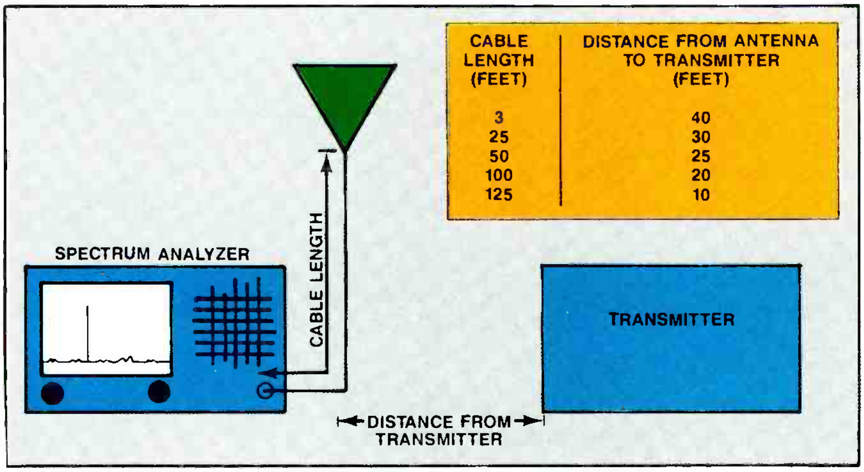
Some time ago, I was given the task of making a number of wireless mics work in the presence of lots of neon. The neon person managed to enclose the proscenium of a theater completely with neon tubes. As I remember, there may have been three different-colored tubes running in parallel around the proscenium arch.

As I recall, I didn't find out about the neon until I arrived at the theater. I suppose the producers didn't want to hear any negative thoughts, especially since they had spent *big* bucks on the set.

Originally, I set the receivers up on the stage and connected them to an antenna diversity system using three dipoles, located stage left, stage right and center stage in the footlights. I arranged them thus to take advantage of the capture effect of using three receivers. If properly placed, using three antennas removes 97% of the dropouts.

Remember the old Frankenstein movies with the "Jacobs ladders" in the background? They consist of two wires set vertically. At the bottom of the ladder, the space between the two wires is rather close, and at the top, the space between them is wider. When a high voltage is applied to the two wires, the arc starts at the bottom and "climbs" to the top, where it finally cannot bridge the gap any longer, and the spark is squelched, usually with a loud snapping sound. The whole process continues to repeat itself until the power is turned off.

This adequately describes the sound that was coming from the wireless channels. It was not in the background; when it was heard it usually competed with the program audio. Also, the interference seemed to be greater when the wireless transmitter was close to the neon. Of course, the director wanted the MC to be next to the neon when he introduced the



CABLE LENGTH (FEET)	DISTANCE FROM ANTENNA TO TRANSMITTER (FEET)
3	40
25	30
50	25
100	20
125	10

Figure 4. *Relative distance from transmitter to antenna vs. coax cable length/feet.*

various acts or when going to a commercial.

I went back to the office to pick up the spectrum analyzer. When I hooked it up on site I found that the effective radiated RF signal from the neon tubes was, at worst, equal to the wireless transmitters'. In addition, the neon bandwidth was full-spectrum. It went from 5MHz to 1GHz. I am sure it was wider, but the spectrum analyzer only measures from 5MHz through 1GHz. By connecting a whip antenna to the spectrum analyzer with a piece of coaxial cable, I could see standing waves along the tubes, as well as from the high-voltage leads feeding the tubes. The strength of the neon signal was equal throughout the entire bandwidth.

I also found that the neon RF field was also being coupled to the entire building's 110Vac line through the primary side of the high-voltage neon transformers. I noticed that the high-voltage wires were just twisted together at the splice points and not shielded. Nor were the transformers grounded. In effect, I had a very broadband (dc to light), high-level signal source that was using the entire proscenium arch, plus the high-voltage feed lines, as an antenna. Couple this with the fact that the broadband signal was also traveling down the ac lines, and I think you can see why I started to think about moving to Hawaii and opening a reef-shoe rental business.

First I met with the neon person. He seemed to become quite excited and happy when I showed him the field that his neon was radiating. He started to jump up and down and mutter utterances similar to "Far out, man!" I suggested that he ground the cases of his eight transformers. This actually started to have a positive effect because he started the grounding process by screwing a rather long sheet-metal screw into the transformer case and then connecting it to the ac ground. After a very short time, the transformers would short out, the neon would stop working and the wireless system would start working.

It took him a couple of hours to replace the transformers. When he grounded the new ones, he attached the ground by using a nut and a bolt through the transformers' mounting ears. I also suggested that he solder all splices.

With all the interference, I decided to move the receivers and the antenna diversity off of the stage and into the balcony. I also hoped that I would be on a different ac circuit.

Almost all professional wireless receivers have good RFI (radio frequency interference) filters built into the ac circuit. However, the filters only remove the RFI from the ac line after it gains entrance to the receiver. If the RFI is strong enough, it can radiate from the ac line just outside the receiver and cause big problems.

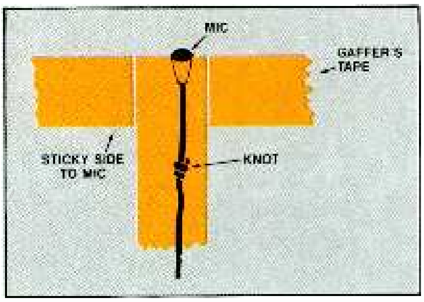


Figure 5. *Attaching a wireless mic to the talent's clothing.*

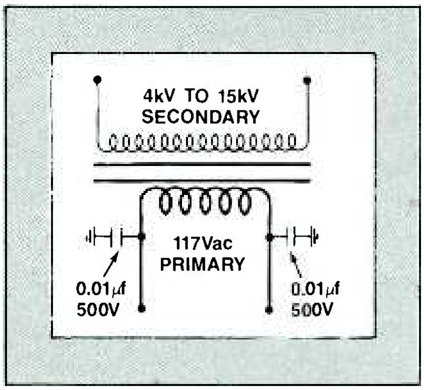


Figure 6. *By-pass capacitors used to buffer the ac line.*

So, I decided to run the receivers on dc. This seemed to help. I have since learned that a high-quality power strip/surge protector/noise filter (the type you would use to protect your PC) helps clean up the ac line radiation. Come to think of it, if you added these surge protectors on the primary side of the neon transformers, you could reduce the ac line radiation even further. If the surge protectors are not available, you can use some bypass capacitors hooked up as shown in Figure 6. The value of the caps should be 0.01µf and at least 500V. This is not as good as the off-the-shelf protectors, as they also have in-line inductors to shunt the RF.

I also ended up removing the antenna diversity system and running the receivers off of their own whip antennas. With all of the RFI in the air from the neon, I was sure that it was mixing with the transmitter carriers and producing products that would be impossible to calculate. Sure enough, it did improve the operation. Up to this point, I was just working on removing the RFI that was getting into the receivers. What about the transmitters?

The transmitters had metal cases with additional metal shields built into the cases. The interference was most apparent when the MC stood right next to the vertical neon tubes located stage right and left. The RFI was getting into the transmitter through the transmitter's antenna. I found that by having the transmit antenna positioned horizontally on the MC, I again could gain marked improvement. I also rotated the receiver antennas to horizontal, but found that they worked best at a 45° angle.

One additional point of RFI intrusion would be the microphone cable and head that plugs into the transmitter. Since I was using ECM-30 mics and the transmitters had all of the latest RFI modifications on the microphone jack, there was not much I could do. However, I would like to have tried a dynamic mic. I have a feeling that the electret has some real problems rejecting a strong RFI field.

Did it work? Most of the time, it did. On a scale of 10, we started with dismal 3, and after all the troubleshooting and tweaking, it was a tolerable 7.

Here are two more suggestions that should help in the event you are faced with a neon nightmare:

1. Enclose all of the neon tubes in 1/4-inch hardware cloth. (Nobody said it would be easy or cheap.) Ground the hardware cloth to an *earth* ground, not the U-ground of the ac system.
2. Use shielded high-voltage cables to feed the tubes. Since the highest voltage used for most neon is 15kV, this can also be difficult. The cable's shield should also be connected to the *earth* ground.

It wasn't that long ago that most wireless users were more concerned about the system "just working at all," even if it did have a very limited dynamic range, frequency response, noise floor and so on. In recent years, however, this has all changed. Now, some systems have over 115dB of dynamic range and an overall frequency response ±3dB from 100Hz to 15kHz. This is more than the dynamic range of the input channels on most mixers. And, the use of switching diversity receivers has eliminated all but the most stubborn drop-out problems. RE/P

Acknowledgments: Special thanks to Ken Mclaughlin for his tips on dressing the talent.

Where Have All The Sprockets Gone?

By Scott Gershin

As alternative methods of sound editing become accepted, more and more production companies are starting to integrate tape and electronic editing with that of 35mm mag strip film.

Alternative methods of sound editing are becoming accepted in the broadcast and film community, with more and more companies starting to integrate tape style and electronic editing with that of 35mm mag strip film. In television, the trend seems to be that a majority of broadcast shows are having their post-production done on tape, while film companies/studios (theatrical releases) are still reluctant to embrace tape-based editing because of a lack of standardized hardware, operating procedures and terminology.

The two most common editing devices for 35mm mag strip film are the moviola and the flatbed (see Photo 1), while editing on tape consists of myriad synchronizers, tape machines, mixers and other peripheral devices that have their own operating languages and user interfaces. Fortunately, those in the electronic editing industry are trying to work with manufacturers to create some kind of consistency and standardization, but this can be difficult when technology advances faster than the industry.

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So far three styles of non-film editing have evolved: 1/4-inch or 1/2-inch tape, RAM-based sampling, and editing using direct-to-hard-disk recording.

All three styles compile the edited information on a multitrack format that can easily be brought to the dubbing or mixing stage. While most mixing facilities use a 2-inch 24-track format, other formats include 1/2-inch 8- or 16-track, 1-inch 16-track, 32-track digital or 2-inch 32-track analog.

As for synchronizers, it's still an open market, with everyone either praising their own units or creating a new vocabulary to describe them.

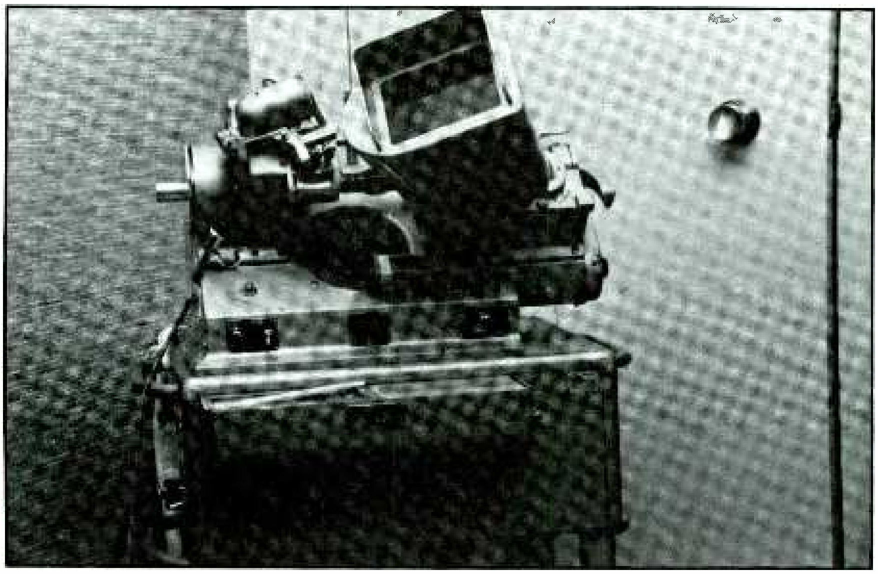


Photo 1. *A moviola, used for editing 35mm mag strip film.*

Tape editing

Tape editing was the first of the three non-film styles to evolve. Borrowing technology from the record and video industries, editing rooms were created, using a multitrack machine locked to picture with the use of a synchronizer reading time code from a video playback unit. For video playback, a 3/4-inch cassette machine, with time code printed on its audio track (Channel 2) and a window burn (character insertion displaying time code information) corresponding with the printed time code is considered the video master unit. The 24-track is thought of as the audio master and is slaved to the video master with a zero offset. At this point, all other machines are slaved to the video master with an offset corresponding to the particular effect and the desired edit point.

Many facilities have their effects library stored on 1/4-inch tape. A technique for laying in effects from that format (as well as other formats, such as 1/2-inch 4-track, carts, CD) is to use trigger starts to "fly-in" the effect. When doing this, the 1/4-inch machine is not in hard lock. For short effects, most tape machines will stay in sync. When an effect is triggered, the editor can also take advantage of the VSO and half-speed tricks that are not possible without transferring between several tape machines—creating considerable generation loss.

It is advisable to trigger the 1/4-inch source machine slightly before the desired cut-time, enabling the machine to get up to speed. The effect, as well, has to be offset (back-timed) one second. The best way to do this is to mark the 1/4-inch machine permanently at a one-second preroll point for both 7.5ips and 15ips, as this is the speed of most SFX libraries. Then just line up the one-second mark on the machine with the mark on the tape and fly the effect in.

Another technique is to store the SFX library on 1/2-inch 4-track (or 1/4-inch 2-track with center track time code) with effects on Tracks 1 through 3, leaving Track 4 for time code. Note: To avoid time code crosstalk onto Track 3, print the time code at −7VU instead of 0VU. With this technique, the effect will always be in hard lock.

A drawback to this is that your synchronizer setup may take time to accomplish a three-machine, or more, lock-up, and with deadlines, the trigger method might be quicker. An advantage to the 4-track lock-up method is that BGs (background SFX) can be laid in three at a time instead of one or two at a time with 1/4-inch machine. For example: You may have to lay in typewriters, phones and off-stage traffic effects that can't be married to each other. By using the 4-track technique, the BGs for a whole scene can be laid in simultaneously.

Many facilities are currently taking advantage of both methods of laying in effects and use multiple 1/4-inch and 1/2-inch machines. For quick access to effects, time code can be printed on the entire SFX tape, which then can be shuttled to a desired location within seconds. This is done by placing a *go-to* command into the synchronizer for that machine.

A common procedure, which is becoming a standard, is to print time code on Track 24 and a 60Hz neo-pilot tone on 23. This leaves Track 22 as a buffer track, which can also contain the production audio—for reference only. Printing a 60Hz tone on Track 23 aids the synchronizer in resolving to other formats and, in case of time code drop outs, it is available as a secondary guide track for jam sync.

One of the major complaints that film editors have with tape is that each of the 24 tracks are physically joined in sync with each other, creating a problem when only a single track needs to be offset. One solution for this problem would be to strip off the effects to a precoded roll of tape, then synchronize the tape with the desired offset and lay it into the mix as another slave unit.

Another solution is to use cart machines. Carts can be timed to be just about any length before the loop occurs. Common cart lengths are 40 seconds, 1.5 minutes and 2.5 minutes. Carts use a 1/4-inch lubed stock and can be recorded in stereo or mono. For effects that have a natural repetition, such as crickets, a shorter cart will work sufficiently. Many electronic editing rooms have three cart machines and a wall full of carts.

Dialogue editing

In the realm of dialogue editing, it is common for the editor to use the 1/4-inch dailies and trigger in sections of dialogue that are phased against a 0 track. The 0 track is usually the audio track that has been edited by the picture editor. If the production audio sounds OK, then the editor will bounce the track to a dialogue track, but when a problem occurs, such as a transfer problem or a piece of dialogue clipped at the end of a sentence, the sound editor must go back to the dailies and recut the line into the dialogue track.

It is important for the dialogue editor to smooth out the dynamics and background ambiences in the production audio. This can be very time-consuming if each of the actors' lines were shot at different times of the day and/or in different environments. Since it is impossible to get rid of the environmental sounds entirely, the editor may well steal a couple seconds of background fill and lay it under the other actors' lines, creating a smooth transition between the dialogue of each actor.

Samplers

The second form of non-film editing involves the use of samplers. They can either be used by themselves as an on-line system or in conjunction with a tape setup. An editor using a sampler still needs to transfer his edits to a 24-track machine, so the product can be brought to the dubbing stage for final mixing.

A disadvantage of using RAM-based samplers is the lack of sampling time. A 4Mbyte sampler holds approximately 40 seconds of effects (10 seconds to each meg of RAM). That works great on effects that are short, such as door slams and gun shots, but when accessing effects that are long, such as several car moves, the sampler can dedicate itself only to one sound or subject matter at a time. In those instances, using a sampler in conjunction with tape machines makes a more versatile and powerful editing setup.

Patches

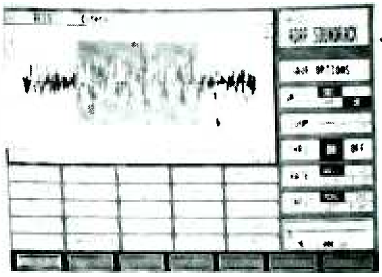
A patch consists of numerous sampled effects. A patch named Mustang might consist of samples such as: car-in, idle, car-out, car-start, brakes and doors. This is used to build the sound elements of the vehicle. The editor can audition any combination of sounds by downloading the necessary patch.

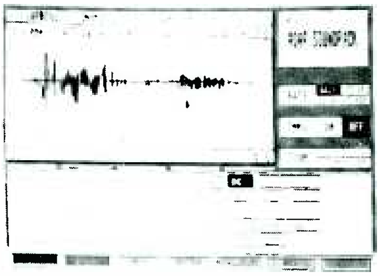
When using a musical keyboard, the editor can manually play-in the sound. For example, to lay-in a car effect, the editor only has to press three keys: one key to start the car, a second to activate the idle (which is looped and can last indefinitely) and the third that starts the car-out. Each sample can be programmed to fade in and out, creating smooth transitions between each effect. Compared to editing tape, this process can take just a few minutes. With tape, you have to lay-in each effect on a separate track and smooth out the transitions, creating loops by checkerboarding from one track to another.

Sequencers

Sequencers in conjunction with samplers make a very powerful combination. Sequencing-in effects is the "Foley by fingers" style of laying in effects. The editor triggers the effect manually, using a keyboard (or any other MIDI controller). That information is recorded in a computer, including the volume and pitch changes that might have occurred during the "playing-in" of that effect. When you roll back the video, the computer will exactly reproduce all the moves that you played. (Note: The sequencer must be able to read all the formats of time code and to edit using those time code numbers.)

When sequencing-in the sample, the editor can contour the effect by using the pitch wheel, velocity control or other MIDI data controllers. An example is in the TV series "Beauty and the Beast," with pipe





Photos 2 and 3. *Many samplers are offering visual representations of the samples to make cut-and-paste techniques more accurate.*

effects constantly occurring in the background during the underground scenes. Velocity control aided the editor when laying-in sampled pipe effects, taking care not to clash with the dialogue. This was done by sequencing-in the effects to play louder in between lines and lower during the actual dialogue.

Changing the pitch also helped when trying not to clash with the tone of an actor's voice. For example, if the voice has a low-end presence with a whisper, the editor uses a high-end bright sound to cut through between lines, and then a duller, lower-pitched version of the effect when concerned about clashing with the actor's lines.

The same technique holds true for Foley-style effects, such as grabs, falls, and punches. The editor can often play-in the Foley effects manually, creating a more natural feel to the scene while saving time—relative to entering the time code locations for each effect. When laying-in Foley effects, it is important to have a large variety of sounds to simulate the random nature in which living creatures make real-time sounds. Samplers are helpful because of their audition features. A 4Mbyte patch can store 20 punches, a dozen body falls and various grabs on dirt, cement, wood, and grass. All of a sudden, you have a full fight-sequence library at your finger tips. To audition these effects, the download time can be 20 to 40 seconds. This is considerably faster than calling the transfer department or shuttling through several reels of effects—before finding the right combination of sounds.

Once the effect is played into the sequencer, the editor can fine-tune the sync of the performance by adjusting the start time of each effect, as each effect has its own sequenced track. Most sequencers have between 40 and 200 tracks that the editor can preassemble and then transfer the edits to one track on the 24 track machine. This saves track space; instead of car moves taking up three tracks (car-in, idle, car-out,) they can be preassembled to one track. The dubbing mixer will thank you.

EDL

Another way of preassembling edits on a sampler is to use an edit decision list (EDL). This list lets the editor choose the desired effect and type into the computer terminal a start, end, duration and pitch number. The list can read time code from the video master, and at the specified mark, the EDL will trigger the sampled effect.

At that point, the assembled edits from the sequencer or EDL can be recorded on tape. This frees the RAM in the sampler to load another patch or EDL, such as tire squeaks and skids. This is the process of off-line editing. Each element or preassembled element has to be laid onto a different tape track.

When designing an effect for picture, the use of envelopes are handy to create wipe-ins and wipe-outs. A neat trick is to lay-in a gun sound at the natural pitch and then, on another track, lay-in the same sample pitched down—with a slower attack time. Then tell the mixer to pan the slower track slightly to one side. The effect is a gun with more bottom end that rings out and decays off to one side.

In the realm of sound designing, the sampler offers a quick, easy way to cut, paste, reverse and loop effects. Many samplers offer visual representations of the samples to make cut-and-paste techniques more accurate. (See Photos 2 and 3.) This technique allows the creation of new, unique sounds that, if done on tape alone, would be quite time-consuming. The editor can listen to the completed edits while laying down new ones. When the editor can monitor all his tracks, it enables the effects to be balanced to the picture, as well as to each other.

Hard disk

The third form of non-film sound editing uses hard disk recording systems. This technology is still in its infancy, but it holds great promise for the future. Hard disk recording is the most similar to the moviola style of editing in that you place a prerecorded effect to time code.

Some of the newest workstation systems have DSP cards that allow reel-rocking techniques. Since effects are stored on hard disk and play directly from the disk, the editor has easy access to the full memory capacity of the hard disk.

Many systems can store between two and four hours of linear recording time. This gives the editor a well containing thousands of effects that can be called upon within seconds. (See Photo 4.) Each effect is assigned to a track output. For example, on the AMS AudioFile, when an effect is recorded, it is called a *cue,* and when it is placed on one of the 8-track outputs, it turns into an *event.* An event can be digitally manipulated without effecting the cue. Currently, most hard disk recording systems come with eight tracks, so for most projects, the hard disk recorder will have to download its edits to a multitrack to allow for more tracks to be digitally edited. The style of editing that most hard disk systems are supporting is the EDL.

Unfortunately, the hard disk recording system is not designed with as much precision for audio manipulation as the sampler. However, when both systems are combined, the editor can have a full, on-line editing system with all the versatility of digital editing.

Transferring from R-DAT or Sony F-1 to a hard disk system (through the use of the AES/EBU ports) is becoming possible. So, the effects will never have to leave the digital domain, and productions can be edited without discernible generation loss.

Most facilities are creating hybrids of the different technologies to meet the needs of their clients and budgets. As hardware, software, operating procedures and terminology standards are established, the popularity of the three non-film editing methods will probably replace the "industry standard" film-style editing techniques of the past. RE/P

Photo 4. *An audio-for-video-post suite at Soundelux in Hollywood. Notice that the AMS AudioFile is built into the mixing console.*

Photos by Cindy L. Manekofsky

Sockhops to Woodstock:

An Interview with Bill Hanley

By Barry McKinnon

Bill Hanley, a pioneer in the field of large-scale sound reinforcement, including systems for the Beatles, the Filmore and Woodstock, reflects on the early years of rock and roll.

Every industry has its pioneers, those individuals who took the first stab at a field no one else considered or wanted to try, and the sound reinforcement industry is no different. Bill Hanley has been in the mobile sound reinforcement business for more than 30 years and has been involved with major artists and events such as the Beatles and Woodstock. He began all of this when off-the-shelf products for large-scale sound reinforcement were not available, and innovation and adaptation were necessities.

Hanley's interest in audio began in high school, oddly enough, through his interest in roller skating. The roller rink where he spent so much time had an incredible organ sound system and great acoustics. In his early teens, he studied radio, television and electronics, which led to his interest in the hardware being used in the rink. The 400W sound system, combined with the energetic playing style of the organist, contributed to the enjoyment that the patrons of the rink felt. "I fell in love with big high-fidelity sound, on a big scale for concerts; I found a lot of joy and happiness in it." Hanley said.

In 1952, he started doing school record hops, and then in 1954, started Hanley Sound with his brother Terry. Bill and Terry did some work for the Boston Arts Festival and ended up buying a system from them. His next project was to get that system into the Newport Jazz Festival when the film "High Society" came out, as the festival became a bigger event after being exposed to the mass audience of the film. Soon afterward, he purchased another system from an electrical engineer in New York. He had been mixing the live sound at Newport since 1960 and then took over the live recording in 1962-1963 to help alleviate the forest of microphones that had been growing on the stage.

The early 1960s

In the early 1960s, Hanley began providing the house sound for Madison Square Garden as well. "They were getting a lot of complaints about bad audio, and we came in and fixed that," said Hanley. He had a lot of problems dealing with the IBEW local at that time because he felt that they didn't understand the show-business nature of this scale of sound work and were slow setting up and tearing down the systems. The promoters were getting mad at Hanley because his more-elaborate systems took longer to set up. "It took a long time to make sure there was a squawker pointed at every ear in the house.

Hanley considers his work on the Beatles' sound system to be a turning point in sound reinforcement. (Photo: The Bettman Archives)

Barry McKinnon is sales manager at BSE Production Equipment and Services Ltd., Calgary, Alberta.

"The road shows would come in and throw up a system on stage, and that would be it." The promoter was paying thousands of dollars in labor for the electricians, and Hanley was getting $500 to $800 per night. "They didn't have a great foreman, and I wasn't a great foreman either, so that was part of the problem. They just didn't understand the business." He continued for three or four years until he was putting so much gear in there, he was losing money. His drive to put up the best sound system possible, no matter what, led to some philosophical shocks.

The Beatles years

Bill was traveling with George Wiens' Jazz Festival, bringing high-fidelity sound to stadiums and fields in Chicago, Cincinatti and Cleveland, when his search for larger shows led to his involvement with the Beatles. "I got to the Beatles when electricity became an extension of the musician, I had learned about and put together an enormous amount of gear. I had four RCA 600W amps that came off of a battleship. They weighed two or three hundred pounds each and didn't have great high-frequency response. You had to drive the inputs with 100W, but it was not enough. When those boys came out on stage, it was absolute pandemonium: you couldn't hear the sound system, you couldn't hear yourself think, 46,000 teenage girls screaming at the top of their lungs, 120+dB ambient noise."

The Beatles' system consisted of 12 Altec 210 low-frequency enclosures and Altec 203B two-cell multicells with 290 drivers. The 288 drivers had better high-frequency response but "a 288 didn't last the night. We were using the 290s, even though the 4Ω voice coils were an inconvenience," said Hanley. The mixing console was a custom-built unit, using 16 channels of Langevin modules.

The Beatles marked a turning point in sound reinforcement in Hanley's view. "Sound changed from being high-fidelity (faithfulness of reproduction); since the Beatles, it's become a battle of levels. With the Beatles, the on-stage levels were getting higher and higher, and the supporting sound system had the grow just to keep up."

The psychedelic 1960s

Hanley was doing permanent gigs for three years at the Filmore in San Francisco, from its opening in 1967, again using the Altec 210s and 203 and 1003 multicells with 290 drivers, set up with left and right stacks downstairs and a counterweighted center cluster, with four of the 210s in it, hung 4 or 5 feet out in front of the proscenium. He used a custom 20-input console, again made up of Langevin AM16 pre-amps and *featuring* EQ on every input. This fed the system through LA2A limiters into passive low-level crossovers and then into MacIntosh MI-200 amplifiers. "In 1965 or 1966, we began bi-amping because we started to use so much power that we blew up the capacitors in the passive networks. They'd begin making some real strange sounds, then—Bang!—nothing.

"A great deal of the joy and happiness that happens for people at concerts is in the intensity and the fidelity. I was one of the first guys to really go bananas with it and try to bring it outside to the masses." Hanley's views were shaken up by two experiences, the first being exposure to Grand Funk Railroad. They had a sound man out of Detroit who had assembled a system using Electro-Voice full-range speaker systems assembled into boxes, augmented by bass speakers.

"Grand Funk shows up at Randall Island, with this kid and his boxes and a pile of DC300 Crowns and associated stuff and sets up beside my system. We didn't 'A-B' them, he just set up and when the band came out he turned it on. Christ, was it loud—distorted, but loud. Here I was, proud of my fidelity, but I stood in the audience and listened to it and realized it was a valid experience—distortion!

> ***"I stood in the audience and listened to it and realized it was a valid experience—distortion!"***

"It was valid for what the band wanted for their audience. Up to that time I had been successful in promoting fidelity, but who was I to superimpose my values on their music? I really had to scratch my head and wonder how much money to spend on fidelity. Here was this kid with his system full of $28.50 EV horns, and piles of them, two or three hundred of them. He got heavy bass out of his bass cabinets and had incredibly efficient output, and it was a valid experience.

"The bands weren't hiring me, they were booting me out of jobs because it wasn't what they wanted. They were getting into this distortion thing; everybody was loving it. Grand Funk was the top group in the country at the time. If they didn't want to hire me, I had to alter my thinking.

"One of the other things that drove me bananas happened in Newport. I remember having a constant tone on the system. It was four or five 210s on each side with some two-cells and 10-cells for inside close, and I walked across the stage and there was no sound. I walked a little further and there was sound again. It was going on and off with phase cancellation. You beat your head on getting everything flat and then you find out it depends where you stand whether you hear it or not. I will never forget that day when I walked across the stage. I knew about phase cancellation indoors, but I never thought about it in an outdoor situation."

Supporting the separation of church and state

Hanley was not doing concerts exclusively, however. In addition to college and university commencements, he provided sound for the presidential inauguration in 1968. In doing so, he introduced the Whitehouse to high-fidelity sound and Shure microphones.

"They had a whole bunch of those University spun-aluminum speakers with the 12-inch woofer and the tweeter in the center, up on scaffolding towers. They were awful. I put up six 210s, three on each side, and blew them away. They had taken pictures at Kennedy's inauguration, and the head sound man for the Catholic churches wrote the specs for the sound system from looking at the pictures. They had lights marked down as speakers and all kinds of things. They threw the Mormon Tabernacle Choir at me at the last minute. This was a great opportunity to bring in my 210s. We set up two systems and 'A-B'ed them, and never listened to his system again. That was the last time I did an inauguration; he never let me come back."

Hanley did not jump on new technology just because it was new. Other than equalization in the channel strips, Hanley had not yet found any pressing need for more elaborate equalization, such as the Altec Acousta-Voice process introduced by Don Davis. "I had talked to him a couple of times out in Anaheim, but it looked like something more for voice systems, not music. I wasn't having any feedback problems. I maximized the front-to-back ratios and tight-miced everything. I never got into EQs; I never learned to use them until later. Then I used them in ways not originally intended, more for artistic purposes." It's not surprising then that Hanley's favorite feedback story doesn't involve feedback at all.

"I was doing Satchmo's birthday party at the Sugarbowl. It was a rush job, and I had just built a new console, checked it out on the bench, and it looked OK. In the middle of the introductions, all of a sudden, it squealed like crazy. Here we are, in front of 20,000 people, and George Wiens is on stage, screaming at me, 'Don't you know what you're doing after all these years?!' He's standing on stage, all but call-

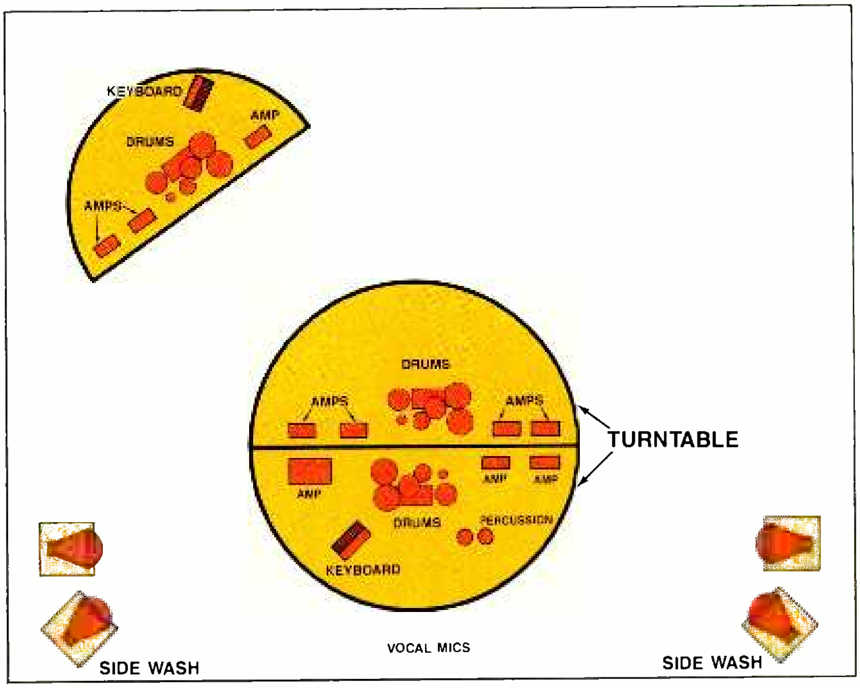


Figure 1. *Woodstock stage (before the turntables broke).*

ing me a f------ a------ in front of these people. I didn't know if there was a problem with the console I had never heard before. It sounded like it was going into oscillation: It wouldn't snap on, it kind of rose up in level. Here I am, banging on the console. Someone spotted this guy standing up by the mic box. Here was this jerk from a recording studio, plugging his signal generator into the mic inputs. Luckily, someone caught this guy. Was I embarrassed!"

Woodstock

Hanley's experience with large outdoor festivals such as the Miami, Atlanta and Dallas pop festivals and the Randall Island festival had established the reputation that led the promoters of Woodstock to him. Michael Lang, the executive producer, wanted a sound system with no screwups, regardless of cost. At that time, there were no precedents for a venue as large as the one contemplated on Yasgur's 700-acre field. After some inquiries, several sources recommended Hanley, because of his involvement in the other festivals.

Because of the scale of Woodstock, Hanley built custom speakers for the event. The low-frequency systems were similar to Altec 210s, but contained four Above those were boxes that each contained eight direct-radiator JBL 15-inch drivers. There were two of each system per side on the lower tier of scaffolding and four each per side on the highest tier. Multidriver 10-cell Altec multicells with 290 drivers provided highs.

"We had done a lot of tests and had a lot of failures with JBLs on the high frequencies, and they wanted a fortune to rediaphragm them for us. We got into the 290s because they would stay together longer. Then we used 288s turned around with the back covers off as direct radiator HF because that's all there was. I wasn't into speaker design." The system was passively bi-amped and driven with a combination of MacIntosh M1200 and M1350 amplifiers and the then-new Crown DC300 for a total power of more than 10,000W.

Stage monitoring was done with a pair of side-wash monitors, each consisting of a JBL 4530 LF enclosure with a JBL driver, and an Altec 311-60 horn with 290E driver for highs. These were driven by DC300s.

The console that he had planned to use for live sound had to be diverted to recording at the last minute. Hanley had chartered a plane to bring a console up at the last minute. Typical of his consoles, they used Langevin pre-amps and EQ. The 20-input house sound console was augmented by Shure M67 mixers as required and fed the system through LA2A limiters. These limiters were also used on the vocal tracks that were layed down on the 8-track Scully recorder.

The stage featured a turntable to allow quick changeovers of the acts. Two 19-pair snakes, custom-built by Bill and Terry, were fed into a custom switchover box that allowed fast changeover at the console. "We were mixing blind. We were way out from the stage, and by the time someone got out to you through the audience with the microphone list, they were already playing." Hanley credits mix engineer Lee Osborne for much of the success in these difficult circumstances, citing his quick grasp of problems under pressure, and his knowledge and experience in live sound work.

Despite the well-publicized problems at Woodstock, the torrential rain and the financial problems of the promoters, it went well for Hanley. "The only things that didn't fail were the sound system, the water supply and the stage security. We'd have been OK if the turntables had stayed together. There were three half-moons that could be set up and hooked to the turntable. They used these cheap 8-inch casters, and the plates eventually tore out of the platform." Hanley had no equipment failures, having been prepared for the worst that Mother Nature and rock music could muster. "We'd done lots of outdoor shows; we were prepared. It was a matter of good planning.

"Woodstock didn't feel like a historic moment. I never thought about it in those terms from an intellectual, historical point of view. You're busy making everything work, making it sound good. It wasn't as dramatic for me as it was for everyone else, it was old hat. Being a technocrat as I am, I was shut out of the political and historical significance."

That's not to suggest that he was unmoved by the experience. "All those people, the joy of the people that were there, the idealism of all those people—it was a

Constructing the stage at Woodstock.

culmination of that kind of thing. A lot of high idealism was why I got into it in the first place."

The console Hanley had built was curved, with inputs in groups of 10. Multiple planes allowed the mix engineer to see and feel every knob. The work with this console led him to start work on an even-more-elaborate console. "Right after Woodstock, I had started to build a console in the round, a semicircular shape that had three planes, two two-band EQs per channel, and presets. No one's figured out yet that what's important in a sound reinforcement console is speed. Get everything into the audio engineer's field of vision; you've got to be able to touch and feel everything. This thing where it takes two or three guys to mix or you need roller skates is crazy." The console remains three-quarters built, a victim of the cash crunch Hanley encountered after Woodstock.

"After Woodstock, the festival markets dried up. I lost a quarter of a million bucks worth of business in three weeks, and I was already three or four hundred thousand dollars in debt." Competition was starting to heat up; other people like the Claire brothers were getting into the market. "I was believing my own bull----, I wasn't countering the competition."

The 1970s

Hanley was still doing touring work in 1970. He was involved in Festival Express, probably the only major rock tour to travel by train. "That was a lot of fun. It was a really great time." He recalls an incident with Peter, Paul and Mary: "They had a really sharpy sound man, they had to mix their own thing and just send me a feed. This was the same guy who did 'Jesus Christ Superstar.' They had to close the show for three weeks because the sound was so bad. One night it sounded terrible, and their manager, Albert Grossman, was screaming at me, 'None of my acts will ever use your system! I'll send letters! You'll never work again!' It was just luck that I had a Nagra on the limiter input from their feed that night, and it came to me distorted. Garbage in, garbage out. So, I'm saying, 'Wait a minute, wait a minute.'

"Back in those days at the festivals, you never really got a chance to do a sound check. The pressure was really intense. I guess that's what I liked about doing it live instead of in the studio."

Hanley has many memories of the developing concert sound business. His position as one of the earliest participants gives him a unique perspective. "We were the first people who were temporary sound reinforcement engineers. I had to learn the hard way that in dealing with live performance, I was dealing with sound reinforcement instead of sound reproduction. It's a different school of thought.

"There wasn't anyone else when I started. I started with six, eight, 10, and finally something like 30 Altec 210s in running inventory. That took some serious trucking." It took quite some time before Hanley was forced to think about the need to make money at it, and not do it just for satisfaction. "I got so much positive feedback from what I was doing that I never came to terms with the need to make money at it."

For Hanley, strong idealism often led him into conflicts with his employers. "When you take the laws of physics as your ideals, you get overconfident. It's still the musicians that call the shots on who gets the jobs.

"Arthur Fiedler used to get mad at me because I wanted to put up lots of microphones, get as much direct sound as I could get, and then add reverb artificially. That way, everyone would hear what's going on in context. When you have a solo flute playing, you drop 30dB in level, and the people 300 feet away are now in the ambient noise. I used to think that was poor. The conductors at commencements would ask for one mic for a 35- or 40-piece orchestra, but now they often go along with multiple-microphone techniques."

In the early days of touring sound, the manufacturers did not look at it as a serious market. "I had a good relationship with MacIntosh, but other manufacturers weren't as cooperative. I had talked to JBL about getting into it, but they thought it was stupid." Now, of course, portable sound reinforcement markets likely make up a significant portion of professional audio sales. At that time, it wasn't seen as a growth market or a good test bed for new products. The needs and demands of touring sound were not yet a concern.

Current projects

Hanley's current project is an automated stage, a folding structure that would contain a stage prewired for mics, monitors and ac. It would include ac distribution, monitor power and light dimming equipment in it. In conjunction with that, lift systems to hoist sound systems into place, prewired and configured. "I've wanted to mechanize this for 13 or 14 years. It's insane: you hump all the stuff into and out of the truck; you do the same thing all the time. All these men doing all this work, two-day marathons to set up a stadium system. I just wanted to push buttons and make it happen." It's an ambitious project that has been consuming a lot of time and capital, but it gets closer all the time. "It would be nice to go to the store and buy one," Hanley said.

Bill Hanley, now 50, still does a few commencements, such as MIT's, and other interesting sound jobs, operating from Hanley Sound, in Peabody, MA. His brother Terry operates Terry Hanley Audio in Cambridge, MA.

The history of the concert sound reinforcement business has never been as well-documented as the movie sound business, but it is every bit as interesting, and, in some ways, more relevant to many of us, amplifying, as it did, the soundtrack of our lives. RE/P

Northeast

Hit Factory (New York) has purchased four ANT Telecommunications E-413/24 telcom noise reduction systems, providing the studio with 96 channels of noise reduction. *237 W. 54th St., New York, NY 10019; 212-664-1000; fax 212-246-2252.*

Blank Productions (Stamford, CT) has purchased three Sony R-DAT machines. Other recent additions include a Roland D-550 with Opcode Librarian software, an MKS8 patcher and a 16-bit upgrade of the facility's Akai S-900 sampler. *1597 Hope St., Stamford, CT 06907; 203-968-2420; fax 203-329-7193.*

Editel/NY Sound Suite (New York) has installed five Magnatech film recorder/reproducers for synchronization with Editel's 24-track system.

LRP Video (New York City) has appointed Marjorie Myers as account supervisor. *3 Dag Hammarskjold Plaza, New York, NY 10017; 212-759-0822.*

Island Media Studios (W. Babylon, NY) has purchased two Valley 430 Dynamites and one Gatex from Manny's Music. *841 Sunrise Highway, W. Babylon, NY 11704; 516-669-1872.*

Eastern Standard Productions (Buffalo, NY) has expanded its real-time duplication and tape loading facility. Duplication production has been expanded by a factor of more than two, and loading capability has been tripled. Direct on-cassette printing has been added, allowing cassettes to be produced with a variety of colors and foils. New real-time equipment includes Denon DRM-44HX units, modified by ESP with a proprietary record system to improve recording accuracy. *26 Baxter St., Buffalo, NY 14207; 716-876-1454.*

Forge Recording Studios (Malvern, PA) has added Sony's line of pre-mastering and digital editing equipment, including the DAE-3000 editor, PCM 1630 processor, two DMR 4000 recorders, DTA-2000 digital tape analyzer, DAL-1000 digital limiter and a PCM-2500 R-DAT recorder. According to the studio, the additions make it the Philadelphia area's only recording and CD mastering studio. *119 Great Valley Parkway, Malvern, PA, 19355; 215-935-1422.*

Times Square Studios (New York) has named Jorge Silva chief engineer. He will also continue his current responsibilities as director of audio. *1481 Broadway, New York, NY 10036.*

New York Audio Productions (New York) has completed a three-month renovation with the installation of a 16-input TAC Scorpion console. The addition will enlarge the studio's music production and voice-over capabilities. Other new equipment includes Drawmer DS 201 gates and Yamaha Q 2031 equalizers. *140 W. 22nd St., New York, NY 10011; 212-243-6826.*

STUDIO UPDATE

Sound Logic Studios (Silver Spring, MD) is the Washington, DC, area's newest recording studio. Designed by John Wesley Gardner, the facility has 15'x18' and 10'x7' studios with variable acoustics and a 15'x25' control room. The console is a 32x8x24 Soundcraft Series 600 console. Tape machines are a Sony/MCI JH 24 24-track, and Studer A810 and Teac 35-2 2-tracks. the heart of the studio is a Kurzweil K250 interfaced with a SMPTE-locked Apple Macintosh Plus running Mark of the Unicorn Performer software. A variety of outboard gear is available. Hank Bartlett is the owner; studio manager is Mark Greenhouse. *3209 Birchtree Lane, Silver Spring, MD 20906; 301-871-0200.*

Roar Productions and Musical Services (Columbia, MD) has added Larry Adler to the staff. During the fall, the facility offered special seminars in live sound, video production techniques, MIDI and careers as a voice-over talent. In conjunction with the Gerard Co., Roar was a finalist for its first entry in the Telly Awards competition. The entry was a TV PSA for the Federal Aviation Administration titled "Discover Today's FAA." 6655-H Dobbin Road, Columbia, MD 21045; 301-596-2600.

Southeast

Woodland Sound Studios (Nashville) has made several additions to the staff. Katie Dungan is studio manager. Barbara Smith is assistant manager. Suzi Ragsdale has been appointed secretary/receptionist. Tim Farmer continues as chief engineer, and is also acting in a consulting management capacity. *1011 Woodland St., Nashville, TN 37206; 615-227-5027.*

Reelsound Recording Co. East (Nashville) is a 24-track remote unit housed in a 1948 Flxible road bus. The unit is equipped with an MCI 428LM console and a JH 24 24-track recorder. Outboard gear includes units from Valley People, dbx, AMS, Yamaha, Teletronix, UREI and RTS. Producer Dave Perkins manages the bus for studio and live projects. *1701 Green Hills Drive, Nashville, TN 37215; 615-385-0220.*

Midwest

Covideo, a subsidiary of Anthony M. Franco, Inc. (Detroit) has been selected as a finalist in the "Guidance and Counseling" category at the 1988 International Film & TV Festival of New York, for its documentary production of "Wasted Dreams." The documentary deals with six Detroit-area young people who have been injured by gunshots. *400 Renaissance Center, Suite 660, Detroit, MI 48243; 313-567-2985.*

Minnesota Public Radio (St. Paul, MN) recently recorded the 200th show of "St. Paul Sunday Morning" in the facility's Studio M. The show was broadcast on Christmas Day. *45 E. Seventh St., St. Paul, MN 55101; 612-290-1500; fax 612-290-1243.*

TRC Studios (Indianapolis) has upgraded Studio B to a fully functional film-video mix room. New equipment includes an MCI JH110 1-inch video layback machine with Dolby SR; a center-track time code update for the JH110B; a Wide Range Electronics 16/35mm mag transfer machine; an Adams-Smith 2600 synchronizer with remote control; an automation upgrade to its Digital Creations Diskmix automation; a Valley People rack with two Gaingrain IIs, two MaxiQs, four Kapex IIs, and two de-essers; a Klark-Teknik DN780; a TC Electronic 2290 sampler and 2240 stereo parametric EQ; and the Sound Ideas compact disc sound effects library. *5761 Park Plaza Court, Indianapolis, IN 46220; 317-845-1980.*

DRC Studio (Springfield, IL) has expanded its inventory. Additions include the Tascam MS-16 recorder with dbx noise reduction, MegaMix automation, Atari Mega ST2 computer, Yamaha REV-5, dbx 160X compressor/limiters, BBE 822 Sonic Maximizer, AKG 414 B-ULS microphones, Sound Ideas sound effects on CD series 1000 and 2000, Simmons SDS1000 MIDI drum kit and an E-mu Systems EIII with more than 200Mbytes of sounds on hard disk. *2416 S. Walnut St., Springfield, IL 62704; 217-753-0409.*

Paisley Park Studios (Minneapolis) hosted the first Sound Stage 88 in early October, featuring two days of music, pro-audio and video equipment exhibits, seminars and instruction. The studio's 12,000-square-foot sound stage housed representatives from more than 70 exhibitors, including Lexicon, Grass Valley, 3M and Ampex. Paisley's three studios were used for instruction and public viewing. About 2,000 people attended, and plans are underway to host the event in 1989.

Sean McMahon, engineer at **Smith/Lee Productions** (St. Louis) has recently purchased new outboard equipment, including an Eventide H-3000 Ultra-Harmonizer, Lexicon LXP-1 digital effects processor, Lexicon MRC MIDI reverb controller, a software update for a Yamaha SPX-90, and a Valley International PR-2 mainframe with two Commander modules.

Ron Rose Productions (Royal Oak, MI) has named Anita Lanning as studio manager. *25885 York Road, Royal Oak, MI 48070; 313-545-1696.*

Mountain

Coupe Studios (Boulder, CO) has upgraded to 24 tracks with the installation of an Otari MX-80 recorder. The facility specializes in radio and television broadcast commercials and album production. *2888 Bluff St., Suite 115, Boulder, CO 80301; 303-447-0551.*

Southwest

Production Masters (Phoenix, AZ) has named Bruce E. Reid as general manager. *834 N. Seventh Ave., Phoenix, AZ 85007; 602-254-1600; fax 602-495-9949.*

Southern California

Todd-AO Glen Glenn Studios (Hollywood) has entered an agreement with Comlogic to supply six custom designed Neotek Essence consoles.

Northern California

The Saul Zaentz Film Center (Berkeley) has acquired an 8Mbyte version of E-mu Systems' Emulator Three digital sound production system. The system was purchased from Audio Images Corp., San Francisco.

Northwest

Spectrum Sound Studios (Portland, OR) has added several staff members and expanded its equipment. New to the staff is Dick Starr, system design sales representative, Matthew Tonjes, maintenance technician, and Kellie Hager, part-time assistant. Spectrum's SSL console has been expanded to 40 channels, and a Total Recall computer has been added. Apogee filters have been added to its Mitsubishi X-86 recorder. Also purchased were TimeLine Lynx synchronizers and a Yamaha G7 grand piano. *1634 SW Alder St., Portland, OR 97205; 503-248-0248.*

Hawaii

Audio Resource Honolulu (Honolulu) has opened just outside Waikiki. The 24-track facility contains an automated Harrison MR-4 console, Sony/MCI tape machines, a Macintosh-based MIDI/SMPTE sequencing system, Westlake and Yamaha monitors, and outboard gear by Eventide, UREI, Orban, Pultec, Drawmer and Yamaha. The studio offers music recording and audio-for-video post-production. Tony Hugar is the studio manager, and Milan Bertosa is the chief engineer. Audio Resource is affiliated with The Audio Lock Up, a Chicago facility. *Century Center, 1750 Kalakaua, Honolulu, HI 96826; 808-944-9400.*

Canada

Comfort Sound (Toronto) has renovated its mobile unit to improve monitoring. Changes have also been occuring in the studio, including the installation of a wall in the recording room, which created a more intimate voice-over area and facilitated better isolation for music bed-track sessions. An Eventide Ultra-Harmonizer has also been purchased. *26 Soho St., Suite 390, Toronto, Ontario, Canada M5T 1Z7; 416-593-7992.*

Metalworks Studios (Mississauga, Ontario) has opened a second studio, SongLab. The facility is a MIDI room with up to 48-track capability. *3611 Mavis Road, Unit 5, Mississauga, Ontario, Canada L5C 1T7; 416-279-4008.*

Pathe Sound and Post-Production Centre (Toronto) has named Ted Rouse as vice president and general manager. The facility has also added a Magnatch High Speed Dubber System and a THX-Lucas Monitoring System.

Seacoast Sound (Victoria, British Columbia) has appointed Geoffrey Bate as general manager. *825 Broughton St., Victoria, British Columbia, Canada V8W 1E5; 604-386-1131; fax 604-386-5775.*

Manufacturer announcements

Tascam has sold a DAT recorder/player to Showco Sound, Carrollton, TX; and a T2640 recording duplicator to Hummingbird Recordings, Melbourne, FL.

Amek has sold a G2520 master recording console to The Church, London, owned by Dave Stewart and Annie Lennox of the Eurythmics.

New England Digital has sold a 32Mbyte, 64-voice Synclavier to Flyte Tyme, Minneapolis, owned by Jimmy Jam and Terry Lewis.

Otari has announced the following tape machines sales: Dave Stewart, two MTR-90s for his personal-use studio in Southern California; George Tobin Studios, North Hollywood, DTR-900; Design FX Audio, Los Angeles, MTR-90; Sunset Sound Factory, Hollywood, DTR-900; LA Studios, Los Angeles, 12 MTR-12 2- and 4-tracks; Front Page Recorders, Costa Mesa, CA, DTR-900; Turner Broadcasting, MTR-12CT; Curtis Mayfield, an MX-80 for his personal use studio; Edit Works, MX-55TM, the first unit delivered in the U.S.; NBC Sports, New York, 16 MKIII 8-track ½-inch machines, used at the Summer Olympics; Howard Schwartz Recording, New York, two MTR-90IIs; Island Media Services, West Babylon, NY, MX-80; If Walls Could Talk Studio, North Caldwell, NJ, MX-80; composer/musician Bob Telson, MX-80; guitarist Jimmy Ryan, MX-80; Manhattan Center Studios, New York, DTR-900; Superdupe Creations, MX-80; Wild Twin Recording, New York, MTR-90II; Backer and Spielvogel Advertising, two MTR-10s; Nutmeg Music, New York; MX-80; and the Hit Factory, New York, two MTR-90s.

Alpha Audio has sold 2,000 square feet of Sonex acoustical foam to Will Vinton Studios, Portland, OR.

Sony has sold a PCM-3324 recorder to Professional Media Services, Gainesville, FL, owned by Grammy-winning engineer Mark Pinske. It is the first 3324 in Florida.

RE/P

THE CUTTING EDGE

By Laurel Cash

Advances in Tape Machine Technology

Otari announces first of B series digital multitracks shipped

Otari shipped the multiple units of its new DTR-900B series digital multitrack recorder last month. According to John Carey, marketing manager for Otari Corporation, "The B series includes all of the changes developed since our earliest machines were delivered and offers the user higher performance, greater reliability and an enhanced ease of use." The new DTR-900B features entirely redesigned Auto-Locator/Remote software and hardware, new proprietary VLSI technology, in-house manufactured heads, and upgraded power supplies to accommodate the use of optional Apogee Electronics low-pass filters in the A/D and D/A sections. Otari has also announced the availability of two new accessories for its DTR-900 machines: the EC-104 plug-in chase synchronizer module and the CB-503 PD (PRODIGI)-to-DASH format converter. The format converter allows bidirectional digital transfers between the DTR-900 and any DASH multitrack machine. No price increase is expected for the new model, and most of the new features of the B series can be retrofitted to the earlier models of the DTR-900 machines, according to Carey.

Circle (170) on Rapid Facts Card

Mitsubishi introduces X-880 digital multitrack recorder

Mitsubishi Professional Audio has announced the new X-880 digital multitrack recorder, an upgraded version of the X-850. Circuitry enhancements include a new, convenient pull-down front panel that provides easy access to audio monitor, ping-pong jacks, emphasis/de-emphasis switches and status switches. The design also adds a comprehensive system status display on the front panel. This display provides visual confirmation of external sync, sampling frequency, playback servo and other system status requirements.

Other circuitry improvements include newly designed linear-phase active analog filters, which are reported to improve the sound quality substantially. These are Mitsubishi-designed, Murata-manufactured and are said to be similar to those of Apogee Electronics. Mitsubishi has elected to use Burr-Brown monolithic A/D and D/A converters. This is also said to increase reliability and further improve the sound quality.

The company has also redesigned the auxiliary analog track circuitry for a higher S/N ratio and lower distortion. A master safe switch has been added to prevent accidental erasure during mixing sessions. The transport enhancements include faster winding modes and a motor driver that has been redesigned both electronically and mechanically for higher performance and reliability.

The X-880 comes prewired for the optional plug-in CS-1 chase synchronizer. The CS-1 resolves to a digital sample frequency that is equivalent to 20μs. According to Mitsubishi, absolute phase-accurate 64-track lock-up or intermachine editing is assured when using the CS-1. It is claimed that when the transport is under synchronizer control, a varispeed resolution to 0.01% eliminates audible pitch changes. Also, redesigned transport control circuitry includes switches and trim pots, allowing you to optimize the transport for 7-inch, 10.5-inch and 14-inch reels.

The X-880 is available now, and has a suggested retail price of $180,000.

Circle (171) on Rapid Facts Card

Sony introduces analog 24-track

Sony Professional Audio has introduced the APR-24 multitrack recorder, which is said to be available for shipment immediately. It is reported that many of the APR-24's features, which are included as standard equipment, are optional accessories on machines available from other manufacturers. Some of the features that come standard with this machine include an on-board time code generator, reader and synchronizer. As a result, the remote unit provides audio, transport, locator and synchronizer control in a single package requiring only one connecting cable.

The synchronization facilities allow you to synchronize to longitudinal time code and VITC in various formats. You can resolve to time code, tone (59.94Hz or 60Hz) and house sync, "burst" time code output during fast-wind modes and "bit bump" with subframe offset accuracy.

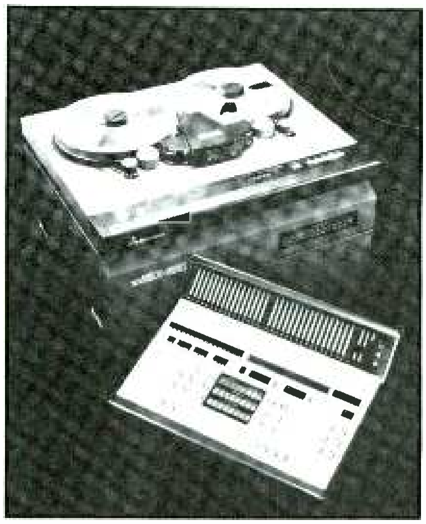
On the remote control are edit keys that allow the actuation of rehearsed and externally programmed "punch in" and "punch out" operations. Sony is said to be targeting the APR-24 to video post-production houses, film audio production and music recording studios. As of press time, the suggested retail price is reported to be $45,500.

Circle (172) on Rapid Facts Card

48-channel DASH recorder launched by Sony

The AES convention in November was the first glimpse most of the industry had of Sony's new 48-channel, DASH-format digital multitrack recorder, the PCM-3348. This unit is reported to be upwardly and downwardly compatible with Sony's PCM-3324 and 3324A. It is said that you can record on channels 1 through 24 on the PCM-3348 and play that master back on a 3324/3324A. 96kHz oversampled electronics are featured on the inputs and outputs. Other features include newly designed digital filtering, real-time ping-pong (with no time delay) and variable crossfade times.

The advanced digital output allows for up to 250 words of digital audio (about



Mitsubishi X-800 digital 32-track.

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

5ms) to be output from the recorder (before the analog output) with one-word resolution, allowing for compensation through other digital devices, thereby assuring absolute phase on the tape. The PCM-3348 also has 20 seconds of solid-state RAM on board with 16-bit resolution. This allows you to store 20 seconds of audio from the tape into RAM and re-insert the RAM data elsewhere on tape. It can be triggered externally either by MIDI, by a gate or manually from the Auto-locator. You can select any two channels to be AES/EBU I/Os. There are a remote channel arming interface and a remote 48-channel multifunction meter system. As of press time, the suggested retail price is reported to be $240,000, and the PCM-3348 should be available for shipment this month.

Circle (173) on Rapid Facts Card

Studer previews new multitrack at AES

Studer Revox previewed its new multitrack recorder, the Studer A827, at AES in November. This is the first multitrack addition to its A820 line since the A820-24 and is based on the A820 transport.

Like its top-of-the-line brother, the A820-24, the A827 offers many of the features possible with microprocessor technology. Some of its more notable features include a 14-inch reel capability, three tape speeds with integrated varispeed controller, two tape types storable for each speed, phase-compensated MDAC-controlled amplifiers with switchable Dolby HX Pro and an optional internal synchronizer. Also included are parallel and serial RS-232/422 control ports for easy system integration.

Priced beneath the A820-24, the A827 is said to incorporate the latest technology and efficient production techniques to reduce construction costs. Formal introduction of this product is planned for the AES in Hamburg, with the first deliveries expected in May 1989.

Circle (174) on Rapid Facts Card

RE/P

Sony APR-24 analog 24-track.

Continued from page 45

The 2-inch remix master track-format was:

Track 3—left audience to be inserted later

Track 4—right audience to be inserted later

Track 5—next pair left audience to be inserted later

Track 6—next pair right audience to be inserted later

Track 7—left stereo music remix

Track 8—right stereo music remix

Track 9—mono music remix

Track 17—P-1 production microphone

Track 18—P-2 production microphone

Track 19—P-3 production microphone

Track 23—timecode (copied from live original 24-track)

Track 24—timecode (reshaped and copied from live original 24-track

The three production microphones were included to support the sweetening process. If a dialogue part needed to be pulled up over the applause, it could be done at the very last minute. We never recorded the applause and effects microphones on the original live-music multitrack, so it was difficult to judge the proper ratio of dialogue to the applause sound. Basically, David Palmer had to concentrate on making the music remix as pretty as possible. A normal axiom for mixing TV sound is if you can see it, you had better be able to hear it—loud! David did not subscribe to that philosophy. He normally mixed albums, not TV shows.

The show was mixed in the Fanta truck, which has a permanent setup for mixing to picture. Since we recorded there, we thought there would be less of a learning curve when mixing. Also, David could concentrate on the job, but still be near enough to the Fanta studio crew to get help, if needed. Instead of making the mix jump up and down to match the picture, we discovered that the smooth playing allowed subtle changes in the mix. If you saw a guitar player soloing, you could hear the solo as well as the other music that fit around it.

Sweetening

Sweetening this show was in the hands of Mark Repp and editor Tom Edwards at TNN. To put the final version of the show together, they synchronized the 24-track remixed master with the final video edit of the show. They also synced up the original audience-only 24-track tape. As they watched the video, they mixed the appropriate audience sound on to the 24-track remix master. When they were finished, they had a 24-track remix master that had the proper music remix, the correct applause and audience sounds, and any other special elements that were necessary to complete the audio (possibly an announcer or special effect).

The 24-track remix master, which now had every song and every piece of dialogue, was then conformed or transferred to the edited master videotape. The audio was recorded, Dolby encoded, on tracks 1 and 2 of the edited 1-inch video master tape.

Some aspects of this show were special. The editor was consulted from the first pre-production meeting to the final cuts. The music remix engineer was involved from the first rehearsal to the final sweetening. And all the team members were truly dedicated and honored to be working on "A session with Chet Atkins, C.G.P." RE/P

NEW PRODUCTS

As reported in the October issue, the recently completed AES Convention in Los Angeles was a bonanza of new products. In compiling the show preview, many companies gave us advance information on their product introductions. Product Preview showcased these products. Other companies, however, chose to wait until the show opened to announce their new products. This month's "New Products" is expanded and contains products introduced at the show but not included in October's "Product Preview." More products will be featured in the February issue.

AMS/Calrec Edit 1 mixer

The Edit 1 mixer uses digital signal processing and consists primarily of eight input faders plus two group or output faders. There are four outputs, two stereo or two double mono; it is possible to mix the two output faders together. Any reasonable number of inputs can be accommodated because the input matrix system eliminates the need for a patchfield. The mixer also features four assignable Logicator (rotary) controls, each with an arc of LEDs projected through light-guides, and an associated four-character alphanumeric display showing the function of the control.

Circle (100) on Rapid Facts Card

Tascam DA 800/24

Tascam's first DASH machine incorporates proprietary ZD circuits in the opto-isolated A/D, D/A converters, along with 2x oversampling in the record and playback. A special pinch-roller drive system provides precise tape-handling and stability. With two digital format I/Os for S DIF-2 PCM equipment and AES/EBU systems, the machine allows digital transfers to various digital devices. The meter bridge also accommodates remote installation, and a second meter bridge may be added for remote use. Suggested retail price is $99,000.

Circle (101) on Rapid Facts Card

Mitsubishi CS-1 chase synchronizer

Designed to facilitate Mitsubishi 32-tracks in sound-for-image projects, the CS-1 allows two machines to be locked up for 64-track capability. Resolution is accurate within the digital audio "word" (±20 microseconds) with two X-880s. It can be interfaced to VCRs, VTRs and ATRs, and can also be programmed with other master machine parameters. The unit consists of a plug-in circuit card for the new X-880 transport and a remote control; an optional card rack allows the CS-1 to be used with the X-800 and X-850.

Circle (103) on Rapid Facts Card

Lexicon MRC MIDI remote controller

The MRC is an intelligent MIDI remote designed to be used with Lexicon's LXP-1 and PCM 70 effects processors. With the LXP-1, the MRC takes uses beyond the front panel to six "hidden" parameters in each of the LXP's 16 programs. The parameters can be altered in real time, and personal setups can be stored in memory. With the PCM 70, the MRC's faders can be used for real-time MIDI System Exclusive control of 12 key parameters for each algorithm, allowing users to tailor factory presets or user registers. The unit can also be used with Yamaha's DX-7s, TX-816 or TX-802 as a patch modifier.

Circle (104) on Rapid Facts Card

NED Synclavier 9600

New England Digital's largest system, the 9600 features 96Mbytes of RAM, 96 polyphonic stereo voices, nearly 3Gbytes of on-line sound storage, SMPTE and VITC synchronization, and a 76-note velocity/pressure keyboard and button control panel. The system supports multiple user interfaces, such as the customized Macintosh II graphics workstation, the keyboard and button control panel, the Digital Guitar interface, a remote controller/editor/locator, or any other MIDI device. Open architecture allows for future expansion and upgrades.

Circle (102) on Rapid Facts Card

Sanken CU-44X mic

Available from Audio Intervisual Design, the Sanken CU-44X is the first transformerless model that features dual-capsule condenser design in a mono microphone. The cardioid pattern uses 1-micron titanium diaphragms, which are corrosion-free and impervious to temperature and humidity changes. The dual-capsule design provides handling of both low and high frequencies, 20Hz to 20kHz within +1dB. Self-noise is inaudible (18dB or less) and maximum SPL is 140dB.

Circle (105) on Rapid Facts Card

Soundcraft MIDI computer

For use with the Soundcraft 6000 console, the computer is a mute control system with a non-volatile RAM memory that stores up to eight songs, each containing 100 patches of complete mute settings. By using the keypad and 10 numeric buttons, each song and patch can be individually named. The supertwist backlit LCD allows information to be viewed from a variety of angles and in varied lighting conditions. When used in a MIDI studio, the computer records mute settings as they are made, at the correct moment within the sequence.

Circle (106) on Rapid Facts Card

Turbosound TXD-520

For small venues or areas that cannot be directly reached by the main loudspeakers, the TXD-520 is designed to handle high power and provide coherent sound over a side area. Universal mounting and fixing hardware allow for easy installation and enclosure orientation. Components used are one 10-inch drive and one soft dome tweeter. Frequency response is 100Hz to 18kHz, and average dispersion is 90°×60°.

Circle (107) on Rapid Facts Card

dbx RT-60 option chip set

An option for the RTA-1 analyzer, version 2.0 of the RT-60 option chip set allows measuring the reverberation time of a space in each of the RTA-1's 31 bands. In addition, it provides enhanced room-response curve capabilities, and adds a

NEW PRODUCTS

microphone calibration with database for 10 microphone sensitivity memories, absolute SPL measurements capabilities and a custom printout feature. In the room response curve mode, 15- and 30-second functions have been added. Retail price for version 2.0 is $499.

Circle (108) on Rapid Facts Card

Tascam ATR-80/32 multitrack

The 32-track, 2-inch machine accommodates 14-inch reels and has reel and PPL capstan motors that use Samarium Cobalt, which reduces mass and produces higher torque than in conventional motors. The result is a fast-wind speed of up to 280ips, more responsiveness and more reliable lockup. A rehearse feature enables users to preview edits without affecting the master. Also included is a backup memory for tape time, pitch control and amplifier mode settings. Retail price is $45,000.

Circle (109) on Rapid Facts Card

Aphex 612 expander/gate

In addition to gating functions, the 612 also features downward expansion with a variable ratio, allowing a wide variety of dynamics control. The unit is also a "ducker," allowing a key input to lower the level of the audio input. The 612 is the first product to incorporate the Aphex VCA 1001. Suggested retail price is $795.

Circle (110) on Rapid Facts Card

DAR SoundStation II configurations

The two- and eight-channel configurations of the SoundStation II feature stereo time-warp, fully animated playback display, punch-in record, long crossfades, and full chase synchronization. Combined with SoundStation II's sound segment cutting and trimming, edit sliding, track slipping and reel-rocking editing, these features make the systems the most powerful multichannel digital audio sound editing and processing systems available, the company says.

Circle (111) on Rapid Facts Card

CTI/dbx A/D chip set

The F410/D20C10/A1520 chip set, an 18/19/20-bit analog-to-digital converter, uses noise-shaped oversampling at 6MHz and flash four-bit conversion. A two-stage digital filter/decimator reduces the sampling rate to 48kHz, increases resolution to 20 bits and eliminates high-frequency noise. The basic configuration of the A/D converter delivers greater than 105dB signal-to-noise ratio; less than 0.005% THD at maximum input, 0.01% maximum THD (untrimmed) at −20dB and −40dB input; and 0.000000076% differential linearity. Price is $130 in quantities of 100.

Circle (112) on Rapid Facts Card

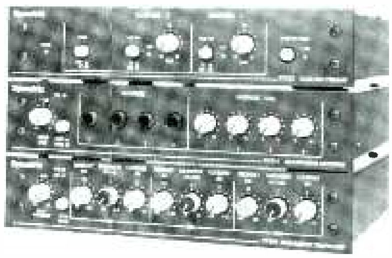
Symetrix "Half-Rack" SX200 series

All of the products in the SX200 Series feature high headroom, balanced inputs, low noise, wide dynamic range, low distortion, studio-quality circuitry, and low-impedance, high-current balanced and unbalanced output line drivers. The SX201 parametric EQ/pre-amp includes +15dB boost and −30dB notch filter capability, with unbalanced pre-amp input, balanced/ nbalanced line-level input and balanced line driver output. Retail price is $239.

The SX202 dual microphone preamplifier features two microphone preamplifiers with variable gain, 15dB pad, +48V phantom power, and left, right, and left + right outputs. Retail price is $219.

The SX204 headphone amplifier includes a one-in, four-out amplifier with proprietary high-voltage converter technology to drive high-impedance headphones like a big power amp, while providing ample power for low-impedance phones. Retail price is $269.

Circle (119) on Rapid Facts Card



Court Signature Series loudspeakers

The soft dome studio monitor loudspeakers include three models. All feature a 1"×8" homopolymer mid and 1"×1" ferro fluid soft dome HF. In addition, the 150W SN20 features 40Hz-20kHz ±3dB frequency response, two-way passive crossover. The 250W SN30 includes a 1"×15" roll surround LF, 30Hz-20kHz ±3dB frequency response and a three-way passive crossover. The 400W SN60 features a 2"×15" roll surround LF, 25Hz-20kHz ±3dB frequency response and a three-way passive/active crossover.

Circle (113) on Rapid Facts Card

Soundtracs In Line series

This console series, initially available in 36- and 48-input mainframes, is now available in a 32-bus format. A number of options, including stereo inputs and effects return modules, are available. Analog specification matches or exceeds the ERIC console without the digital control.

Circle (114) on Rapid Facts Card

Steinberg tape controller TC 1

The TC 1 can remotely control all functions of the Fostex A and B series recorders, except the built-in auto-locator. It positions the tape with an accuracy of ½-frame for up to 255 programmable punch-in/out points. The control time code can be either SMPTE or MIDI; switching between the two is automatic.

Circle (115) on Rapid Facts Card

FM Acoustics Forcelines

Forcelines are high-energy transfer cables designed for lowest loss of wide-band power transfer between amplifiers and loudspeakers. They feature AWG 5 size, are rated for a current of 200A rms continuous and 1,200A peak, and have a minimal resistance of 1.9Ω/km throughout the audio band. They are recommended for smaller monitors and close-field monitor systems, and for wiring mid- and high-frequency drivers in multiway systems.

Circle (116) on Rapid Facts Card

ProCo Patchmaster patchbay

The Patchmaster Series model PM-148 is a single-space, 48-point, unbalanced patchbay designed for use in recording studios, A/V production facilities, commercial sound installations and portable audio systems. The Selectapatch Switch provides easy user modification of the signal flow. Setting the switch determines whether a set of jacks is full-normalled; half-normalled; parallel or open; and no soldering or change in wiring is necessary. Retail prices is $375.

Circle (117) on Rapid Facts Card

Soundtracs FM range console

This broadcast version of the FM range can be configured to double as a small production console for program editing. Available with mono, stereo and telco input modules, the FMB features clean feeds, signal ducking, a program timer and comprehensive monitoring within a choice of two frame sizes. An AFW feature is also available for post applications.

Circle (118) on Rapid Facts Card

Klark-Teknik DN726 delay

The stereo delay line accepts two inputs and provides stereo, in-phase outputs. The frequency response is 20Hz-20kHz. Specifications on the range of delay for the unit is 0 to 1.3 seconds, adjustable in 20s increments, accomplished with 16-bit linear processing for superior performance in demanding situations. A backup battery for memory retention is featured. Retail price is $3,900.

Circle (120) on Rapid Facts Card

Summit Audio Warm Interface

Used as an interface between CD players, R-DAT recorders and analog equipment, the unit provides a tube sound and level matching. The interface features two channels, electronically balanced input/ output, +25dBm maximum output, input design to work with −10dB or +4dB systems, screwdriver trim gain adjustment from the front panel, maximum gain of 20dB, maximum input of +24dBm, dynamic range of 110dB, harmonic distortion of less than 0.1%, and frequency response of 3Hz to 90kHz. Suggested list price is $950.

Circle (123) on Rapid Facts Card

Apogee 924 analog filter/ signal processing module

The 924 analog filter/signal processing module features programming pins that enables the formation of complete analog processing in the front or rear end for digital audio. Also included are an input buffer, an RF filter, a linear phase filter, pre- and de-emphasis, dc servo and an output buffer.

Circle (121) on Rapid Facts Card

ProCo Multiface series switchers

Two new additions to the Multiface series are the RMS-1 and RMS-2 Remote Monitor Switchers, designed to switch power amplifier outputs to alternate speaker systems. The RMS-1 uses one stereo pair of inputs with switches for main/alt and alt A/alt B. The RMS-2 operates in an identical manner, but accommodates two stereo pairs. Suggested price is $275 for the RMS-1, and $389 for the RMS-2.

Circle (122) on Rapid Facts Card

NED Synclavier 3200

The 3200 is a modular compact workstation aimed at customers that could not afford a regular Synclavier system.

NEW PRODUCTS

The 3200 provides up to 32Mbytes of RAM, 32 mono voices and 720Mbytes of hard disk sound storage capacity. The customized Macintosh II graphic workstation, which is standard, controls the system. Other figures include 16-bit, 100kHz multirate sampling, 200-track sequencing and SMPTE/VITC synchronization.

Circle (126) on Rapid Facts Card

Otari MX-50 two-track recorder

Available in both 15/7.5ips and 7.5/3.75ips versions, the MX-50 features built-in tape timer displays with search-to-cue and search-to-zero, front panel ±7% varispeed, 10.5-inch reel capacity, headphone amplifiers, electronic lifter control and a dump edit function. An optional Voice Editing Module (VEM) is available, which provides 2X playback without pitch shift. Price is $2,495.

Circle (124) on Rapid Facts Card

Midas XL2 mixing console

For touring sound, theater and broadcast applications, the XL2 comes in frame sizes of 24, 32 and 40 inputs. A 16-input channel expander section couples to the existing frame via a single, multipin connector. In addition to eight subgroups, there are eight aux sends with full pre/post switching, which enables the console to provide both main or monitor mixes without requiring a separate, dedicated monitor console. Two matrix outputs can receive a submix from any or all of the eight subgroups.

Circle (127) on Rapid Facts Card

Tannoy TPI reference monitor

The TPI features a high-frequency unit developed as a result of the company's Differential Material Technology. Using a deep drawn duralumin diaphragm and skirt with a separate silicone-based suspension gives the piston rigidity associated with titanium but without HF breakup modes in the passband. The coil is ferro fluid-cooled and the driver uses a sculptured asymmetrical phase plate. Both are knitted together with a hardwired crossover.

Circle (128) on Rapid Facts Card

API 525B compressor/limiter

Based upon the 525C design, the 525B has an added motorized gain-reduction device. Three modes can be selected: the standard 525C configuration, a passive element with an adjustable output stage for additional gain, and a completely passive element with no electronics in the circuit. According to the company, the passive mode allows for compressing and limiting with no distortion or coloration.

Circle (129) on Rapid Facts Card

Tannoy SGM-15B monitors

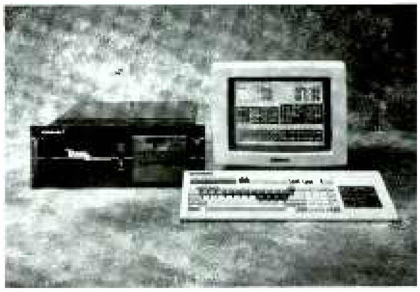
The SGM-15B is designed for users needing a small, yet powerful monitoring system. The monitor contains a 15-inch K-3809 dual-concentric in a cabinet measuring 26½"×19¾"×18½". The crossover uses the same hard-wire construction and a high-current EQ used in the Super Gold Monitor series.

Circle (152) on Rapid Facts Card

Alpha Audio Boss/2 automated audio editor

The successor to the Boss 8400, the Boss/2 features digital waveform editing and concurrent multiprotocol communication using RS-422, RS-232, SMPTE and MIDI. The unit can talk directly to any machine that speaks Sony, Ampex, ES-BUS or other serial protocols. Users can also select whatever combinations of synchronizers are best for their system.

Circle (125) on Rapid Facts Card



Milab D-37 mic

The D-37 is a dynamic cardioid mic designed for rugged use while maintaining high performance standards. The mic is constructed of solid brass and includes such features as a heavily shock-mounted moving coil element for minimum handling noise and built-in pop protection. Frequency response is 50Hz to 20kHz, with a favorable boost in the vocal/presence range.

Circle (132) on Rapid Facts Card

Technics SL-P1300 CD player

The pro unit incorporates 8x oversampling, four A/D converters—two for each channel—and 18-bit technology. Using four A/D converters allows using one for each half of the analog waveform, improving the digital processing of low-level signals that are difficult to capture. Features include two-speed search dial cuing, a rocker control that moves the laser by one pit track, cue point memory and a numeric 10-keypad. Suggested retail price is $1,800.

Circle (130) on Rapid Facts Card

ASC Quick Sound Field

Acoustic Science's Quick Sound Field consists of ½-round Tube Traps on the walls or ceiling of a studio or voice-over booth. The effect is a fast decay rate that is consistent with all frequencies and very diffuse at high frequencies. Additionally, the company says, there is no loss in the upper frequencies and no bass boom associated with smaller rooms. The QSF is available as a retrofit for existing spaces. Retail price is $22 per square foot. Custom free-standing booths are also available, and a 3'×4' QSF baffle for portable use is also available at $375.

Circle (131) on Rapid Facts Card

Hardware and software updates

NED PostPro enhancements

The 8-track Direct-to-Disk multitrack recording and editing system now features a dedicated remote controller/editor/locator, which features programmable buttons that can define command functions. Software enhancements include time compression, direct digital transfer, VITC/SMPTE synchronization and CMX edit list conversion.

Circle (161) on Rapid Facts Card

Sony upgrade to APR-5003 recorder

The upgraded version of the two-channel recorder, called the APR-5003V, features a nine-pin serial cable interface, allowing the recorder to be connected to Sony BVE-900/9000 video editors. The synchronizer features the ability to externally reference a video signal of 50Hz and 60Hz tone, and both LTC and VITC and also accepted. The "bit bump" feature allows for bit-accurate manual adjustment in synchronous mode. List price is $11,950.

Circle (162) on Rapid Facts Card

Impedance alternative for JBL 2204, 2123

Previously available only in 8Ω versions, the 2204 12-inch LF loudspeaker and the 2123 10-inch midrange transducer are now available in 16Ω versions, allowing more flexibility in the design and installation of sound systems. The 8Ω version is designated by H (as in 2204H); J refers to the 16Ω impedance.

Circle (163) on Rapid Facts Card

New software for Lexicon 2400

Version 2.20 expands the 2400 audio time compressor/expander's interfacing capability to include several new videotape machines, including the Ampex VPR-6/VPR-80, Sony BVH-3000 and Panasonic AU-660. A Bypass Play command issues a servo-locked play command from the front panel. Software-assignable relays and inputs provide remote control of this feature, as well as remote control of the 2400's machine control capability. Version 3.0 software provides all of the machine interfaces in V 2.20, as well as dc servo capability for controlling the Panasonic AU-650 MII VCR. Price for V 2.20 is $125. V 3.0 is priced at $250.

Circle (164) on Rapid Facts Card

EQ option for Lexicon Opus

The EQ/filter option allows the Opus to provide 12 channels of digital equalization. The option is comprised of three basic elements. The DSP modules are installed in the host card cage. The EQ control strip is installed in the workstation. EQ application software drives the system. Each of the 12 channels contains four independent filter section, each covering 20Hz to 20kHz. Each of the channels can be assigned to one of six different filter characteristics, allowing flexibility and control over the program material.

Circle (165) on Rapid Facts Card

Updated version of Otari DTR-900

The 900B features a redesigned autolocator/remote software and hardware, proprietary VLSI technology, in-house manufactured heads and upgraded power supplies to accommodate the use of Apogee Electronics low-pass filters in the A/D and D/A sections. No price increase is expected, and the new features can be retrofitted to earlier models. Two accessories have also been introduced: the EC-104 plug-in chase synchronizer module, and the CB-504, a PD-to-DASH format converter.

Circle (166) on Rapid Facts Card

Improvements to Agfa tape and reels

Agfa's PEM 469 mastering tape has a new backcoat, base film and oxide improvements, enabling the tape to withstand more tape passes. The PEM 291D digital audio master tape has a new formulation designed for the current generation of digital recorders and ensures low error rates. Also, the company has redesigned the flange on the reels of the 468 and 469 tape lines, making it easier for users to thread and load the tape.

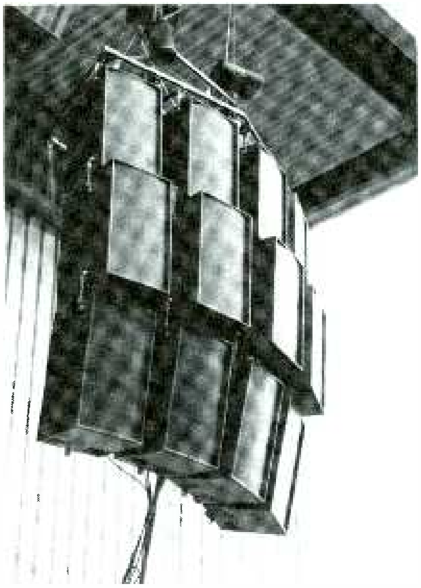
Circle (167) on Rapid Facts Card

NEW PRODUCTS

Martin Audio F2

The F2 is a two-box touring sound system that features full horn loading. The top is configured as a rack-mount shell that can accept different horn and driver combinations, depending on the application. A rack can be made up of only mid or high horns, for long-throw use, or can be configured as a mid/high pack for other applications. A rigging system enables arrays to be built with extended columns of bass, mid and high horns, which can provide clean, high-level music to all audience areas, while using a minimum number of cabinets and amplifiers.

Circle (133) on Rapid Facts Card



Audio Precision DSP-1

For use with the System One, the DSP-1 provides signal generation and analysis in the digital domain, as well as providing feature enhancements for the System One's analog measurement capabilities. For digital applications, the DSP-1 provides digital waveform generation and digital analysis capable of different functions via downloadable software. Direct digital I/O ports are optional. Additional analog measurements for the System One include FFT-spectrum analysis, individual harmonic distortion and total harmonic distortion without noise.

Circle (135) on Rapid Facts Card

Turbosound TXD-530

The 530 is designed to provide extra-wide dispersion in applications that require coverage over a wide area from one source, such as an under-balcony fill. Components used are two 10-inch drivers and one slot tweeter. Dispersion is 120°×50°, with a frequency response of 100Hz to 18kHz.

Circle (136) on Rapid Facts Card

Intelix Studio Psychologist

The remote-controllable matrix mixer/router system allows individuals to mix their own headphone mix. Actual mixing takes place in a modular rack-mounted unit. Each module has 16 mixer nodes using digital attenuators with 85dB control range and a 1.5dB resolution. Inputs and outputs are TRS-balanced. The mixer may also be controlled by a computer with an RS-232 interface. Applications include recording studios, broadcast routing, sound contracting and theatrical sound.

Circle (137) on Rapid Facts Card

HME additions to 700 series

The RP743 and -753 four-channel power stations are the newest additions to the 700 series cabled intercom line. With the 742, two headsets can have communication access to any of the four independent channels. The 753 is a four-channel mix matrix power station with a panel of matrix switches that assign 12 stations or groups to one of four independent intercom channels or two private lines. Both are compatible with 700 series line, as well as most three-wire intercom systems.

Circle (138) on Rapid Facts Card

Dolby 363 two-channel noise reduction

The 363 allows Dolby Spectral Recording and A-type noise reduction to be switchable. The unit contains two channels in a 1U frame. Both channels are equipped with a built-in record/playback changeover capability, allowing a single unit to be used in stereo applications. The 363 is normally supplied with two Cat. 300 modules, which contain both SR and A-type. Optional SR-only or A-type-only versions can be ordered.

Circle (139) on Rapid Facts Card

Hard disk version of Publison Infernal Machine

The Infernal Machine's hard disk version allows one hour and 48 minutes of recording in mono mode, and 54 minutes in stereo mode. Sampling rate is 50kHz, with 16-bit linear A/D conversion. When used with the IM80 color editing software, the Infernal Machine is connected to an AT computer via RS-232 for the creation of editing lists, non-destructive editing and real-time chains. The optional DAB-3 rack allows up to six hours of digital sound to be recorded on a high-speed, high-capacity digital cassette.

Circle (140) on Rapid Facts Card

JBL 12SR loudspeaker system

Featuring a 12-inch LF transducer and a 1-inch exit compression driver on a Flat Front Bi-Radial horn, the Control 12SR is designed for high sound quality at high sound-pressure levels. The LF transducer features a 3-inch, diameter edgewound aluminum voice coil for clean response and high power handling. Using the Bi-Radial horn allows HF energy to be dispersed in a 90°×40° pattern.

Circle (134) on Rapid Facts Card

Chrystal Semiconductor A/D converter

The CSZ5126 is a low-cost A/D converter designed for a variety of applications such as DAT decks and digital audio

EQ systems. Manufactured in low-power CMOS, the chip features a patented self-calibration circuitry, which results in improved performance, according to the company. Dynamic range is more than 92dB in stereo mode and more than 95dB in 2x oversampling mode. The converter is initially available in 28-pin DIPs. Lots of 1,000 are $27.20.

Circle (141) on Rapid Facts Card

Sonic Solutions Desktop Audio system

The workstation is designed to allow users to prepare master recordings with a computer and applications software in the same way as a desktop publishing system works for writers. The system includes a Macintosh II, one or more hard drives and a Sonic Signal Processor circuit card. Software used is CD PreMastering Desktop, which provides all-digital editing, mixing, EQ, dynamics and project management functions for mastering. A new version of NoNOISE software is available as an option. Basic system cost with the premastering software is $44,100.

Circle (142) on Rapid Facts Card

Turnkey systems from Offbeat Systems

The turnkey systems for computer-based music scoring include the computer, circuit boards, and Offbeat's Streamline or Clickstation software. Three configurations are available: the Transportable Streamering System, Laptop Clickstation and Starter Station. Also available is the MIDI Synthesizer Drive, an optional PCB plus software that allows MIDI-based sequencers and devices to be directly controlled by the Streamline or clickstation systems.

Circle (143) on Rapid Facts Card

Agfa-Gevaert PE 647-947 tape

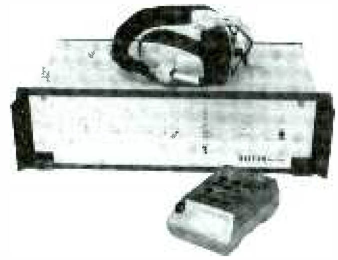
The chrome bulk tape is designed for users who need a high-performance, Bias II cassette formulation. Features include superior electroacoustic specifications, durability, consistency and slitting characteristics, according to the company. The tape is available in 8,200-foot lengths for C-60s and 11,500-foot pancakes for C-90s.

Circle (145) on Rapid Facts Card

Diless ProCom intercom

The full-duplex intercom features push-button dialing for each station, allowing users to make the required connection by pushing one button. Any station may be used as a beltpack or table-mounted unit, and stations are connected with standard three-core mic cables. The automatic central unit can handle 11 simultaneous connections, and no line is ever busy. A special remote station is also available with four public address outputs.

Circle (144) on Rapid Facts Card



Beyer DT 770/990 headsets

Both headsets feature a frequency response of 5Hz to 35kHz to reproduce complex waveforms and upper-end harmonics accurately. The 770 features a large, but low-mass, diaphragm embedded in a bass-reflex system. The 990 is identical but features a circumaural, semi-open-air design.

Circle (146) on Rapid Facts Card

ADx-22 synchronizer

The synchronizer offers synchronization and emulation as a menu selection in the same unit. The unit will lock to a single time code bit in less than five seconds, making it possible for unmodified pre-roll times. Other features include two independent time code readers, one time code generator, an RS-422 data port, an optically coupled parallel port, four DPI relays, an external audio triggered mark function and multiple time code memory scratchpad.

Circle (147) on Rapid Facts Card

ProCo junction boxes

The model HJ-6 allows splitting of a mono or stereo amplifier output to feed

NEW PRODUCTS

six sets of 150Ω-600Ω headphones. It may be used in conjunction with the RMS-1 through a special output jack provided on the switcher. The HJ-4P splits the output of a stereo amplifier to four sets of 150Ω-600Ω headphones, but uses XLR-type connectors on the input and master left and right level controls. Retail prices are $89 for the HJ-6 and $149 for the HJ-4P.

Circle (157) on Rapid Facts Card

Teac LV210A videodisc recorder

The unit is the first 12-inch two-sided laser videodisc recorder for less than $20,000, according to the company. The unit retains most of the interactive features of the LV200 recorder, eliminating the CLV mode, and has the same external computer protocol. Also available is the LV210P videodisc player, which is compatible with the LV200, minus the CLV mode. Price for the LV210A recorder is $17,995; price for the VL210P player is $4,995.

Circle (148) on Rapid Facts Card

Sony TCD-D10 PRO DAT recorder

A scaled-down version of Sony's PCM-2000 DAT recorder, the TCD-D10's design criteria is based on the TCD-DP PRO analog cassette recorder. Recording is at 48kHz, with playback at 44.1kHz. Also included are XLR I/O connectors, internal clock for tape location, AES/EBU digital I/P and 2x oversampled digital A/D and D/A converters. The recorder can operate up to two hours in the field with an internal rechargeable battery.

Circle (149) on Rapid Facts Card

Aphex 124 audio level interface

The 124 is designed to combine −10dBV consumer IHF audio equipment with +4dBm or +8dBm professional/industrial audio equipment, allowing both systems to operate at optimum performance levels, matching impedances and operating levels. The unit features transformerless, servo-balanced inputs and outputs for extended frequency response and exacting transient response. Suggested retail price is $219.

Circle (150) on Rapid Facts Card

Synergy One from Analog Digital Synergy

The Synergy One is an in-line, modular digital mixing console available in frame sizes from four to 64 channels. Standard features include digital four-band EQ, high- and low-pass filter, 1Hz offset filter, de-emphasis, remaining headroom indicator, digital bar graph metering for each channel and group faders. Each module is equipped with a true 16-bit fader, providing 65,536 level steps. Available options include time code-based full-function automation, total automation data reset and format conversions.

Circle (151) on Rapid Facts Card

Sony DAU series ¾-inch tape

The DAU series ¾-inch digital audio master cassettes feature the company's Vivax magnetic particles for clear digital sound production. Carbon mirror backcoating reduces the error rate to a minimum. The tape is available in 30-, 60- and 75-minute lengths. The tape comes in an anti-static shell to eliminate static electricity.

Circle (153) on Rapid Facts Card

Real World Research Audio Tablet

The Audio Tablet is a random-access audio editor designed for various applications, including dialogue and music editing, and CD and 12-inch mastering. The tablet is a two-channel system that encodes audio signals in a linear 16-bit format at 48kHz, 44.1kHz or 32kHz. Both analog and digital I/Os are available. The system consists of three components: the interface, the processor rack and a peripheral rack. The rack components occupy 8U of space, and the tablet interface is an ASCII keyboard and touchscreen.

Circle (154) on Rapid Facts Card

Lexicon LXP-1

The unit is a multi-effects processing module that provides a variety of digital reverberation and effects in a cost-effective format. Its 16 programs include halls, rooms, plates, gates, inverse reverbs, delays and choruses. Stereo inputs and outputs are available. Using a two-level control system gives users access to more than 4,000 sounds. Also available are 16 factory presets and 128 user registers for storing personal setups.

Circle (155) on Rapid Facts Card

Beyer MCE 86 shotgun mic

Weighing 95 grams, the MCE 86 is one of the lightest shotgun mics available. Frequency response is 50Hz to 18kHz, with a maximum SPL of 148dB. The unit can be powered by any phantom power source generating 12V to 48V. Two suspensions are available, one for all purpose use and another for use with a video camera.

Circle (156) on Rapid Facts Card

Gentner Electronics EasyTerm

The rack-mount wiring termination system allows users to open the termination like a door to gain access to equipment inside the rack. Wires are run from inside the rack and are connected using an impact punch tool. An optional "kickbar" holds the unit open while running and punching cables. Two configurations are available. EasyTerm/66 uses a 66-style punch block, for use with solid wire only. EasyTerm/FB uses Gentner's Flexiblock, and is designed for stranded and solid wire.

Circle (158) on Rapid Facts Card

Soundtracs Tracmix

The centrally controlled, stand-alone fader and mute automation system operates via a remote keyboard and color monitor. The automation allows for up to 64 channels of fader and mute automation using dbx VCAs for low noise and distortion. Mix data is held in RAM to eliminate unnecessary disk storage, but it may be saved along with grouping information, tack listings and MIDI song data onto a 3.5-inch disk. A time code generator is also built-in.

Circle (160) on Rapid Facts Card

RE/P

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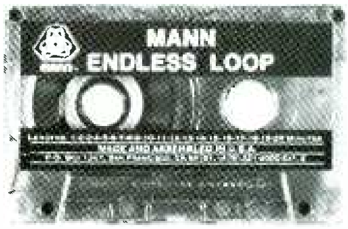

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ENGINEER/PRODUCER INDEX REPLY CARD

January 1989

This is a FREE editorial listing.

This card is for individual engineers and producers. Facilities should fill out the Tracks Reply Form.

1. Fill out your name and address. Include the appropriate title code.

Name ______________________ Title Code [ ]

Address ______________________

City/State/ZIP ______________________

Phone ______________________

2. List your most recent project credits, using the appropriate credit code and subcode. You may include up to five.

1. Client/artist ______________________

______ Credit code [ ] Subcode [ ]

Project title ______________________

Studio(s) used ______________________

2. Client/artist ______________________

______ Credit code [ ] Subcode [ ]

Project title ______________________

Studio(s) used ______________________

3. Client/artist ______________________

Title Codes
E (Engineer) P (Producer) E/P (Engineer/Producer)

Credit Codes
CP (Commercial/Advertising Production)
Subcodes: R (Radio) TV (Television)

MP (Music Production)
Subcodes: S (Single) A (Album) AT (Album Track)
AR (Album Remix) SR (Single Remix)

PP (Post-Production)
Subcodes: F (Film) MV (Music Video)
CI (Corporate/Industrial)

______ Credit code [ ] Subcode [ ]

Project title ______________________

Studio(s) used ______________________

4. Client/artist ______________________

______ Credit code [ ] Subcode [ ]

Project title ______________________

Studio(s) used ______________________

5. Client/artist ______________________

______ Credit code [ ] Subcode [ ]

Project title ______________________

Studio(s) used ______________________

3. Mail this form to Engineer/Producer Index, RE/P, Box 12901, Overland Park, KS 66212. Allow two months from the time you send your entry in until the time it appears in print.

ENGINEER/PRODUCER INDEX REPLY CARD

January 1989

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______ Credit code [ ] Subcode [ ]

Project title ______________________

Studio(s) used ______________________

2. Client/artist ______________________

______ Credit code [ ] Subcode [ ]

Project title ______________________

Studio(s) used ______________________

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Title Codes
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Subcodes: F (Film) MV (Music Video)
CI (Corporate/Industrial)

______ Credit code [ ] Subcode [ ]

Project title ______________________

Studio(s) used ______________________

4. Client/artist ______________________

______ Credit code [ ] Subcode [ ]

Project title ______________________

Studio(s) used ______________________

5. Client/artist ______________________

______ Credit code [ ] Subcode [ ]

Project title ______________________

Studio(s) used ______________________

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STAFF EDITOR
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Recording ENGINEER/PRODUCER

TRACKS REPLY CARD

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Credit Codes
CP (Commercial/Advertising Production)
Subcodes: R (Radio) TV (Television)

PP (Post-Production)
Subcodes: F (Film) MV (Music Video)
CI (Corporate/Industrial)

1. Fill out your name and address. Include the appropriate title code.

Facility Name ____________________

Address ____________________

City/State/ZIP ____________________

Phone ____________________

Your name/title ____________________

2. List your facility's most recent project credits, using the appropriate primary code and subcode. You may include up to five.

1. Client ____________________

__________ Credit code [ ] Subcode [ ]

Project title ____________________

Recording engineer(s) ____________________

____

Mixdown Engineer(s) ____________________

____

Assistant engineer(s) ____________________

____

2. Client ____________________

__________ Credit code [ ] Subcode [ ]

Project title ____________________

Recording engineer(s) ____________________

____

Mixdown Engineer(s) ____________________

____

Assistant engineer(s) ____________________

____

3. Client ____________________

__________ Credit code [ ] Subcode [ ]

Project title ____________________

Recording engineer(s) ____________________

____

Mixdown Engineer(s) ____________________

____

Assistant engineer(s) ____________________

____

4. Client ____________________

__________ Credit code [ ] Subcode [ ]

Project title ____________________

Recording engineer(s) ____________________

____

Mixdown Engineer(s) ____________________

____

Assistant engineer(s) ____________________

____

5. Client ____________________

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Project title ____________________

Recording engineer(s) ____________________

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Mixdown Engineer(s) ____________________

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Assistant engineer(s) ____________________

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ENGINEER/PRODUCER
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Recording ENGINEER/PRODUCER

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Please print or type:

Name ____________________________

Title ____________________________

Company ____________________________

Address ____________________________

City ________________ State ______ Zip ______

Phone No. (_____) ____________________________

Do you wish to continue to receive Recording Engineer/Producer? Yes ☐ No ☐

1. Signature ____________________________

2. Date ____________________________

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Please send me information about the following items.

1	11	21	31	41	51	61	71	81	91	101	111	121	131	141	151	161	171	181	191
2	12	22	32	42	52	62	72	82	92	102	112	122	132	142	152	162	172	182	192
3	13	23	33	43	53	63	73	83	93	103	113	123	133	143	153	163	173	183	193
4	14	24	34	44	54	64	74	84	94	104	114	124	134	144	154	164	174	184	194
5	15	25	35	45	55	65	75	85	95	105	115	125	135	145	155	165	175	185	195
6	16	26	36	46	56	66	76	86	96	106	116	126	136	146	156	166	176	186	196
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10	20	30	40	50	60	70	80	90	100	110	120	130	140	150	160	170	180	190	200

RJ9

RAPID FACTS

For Issue of January 1989
Use until May 1989

3. Please check below the ONE category that best describes your primary job function:
   - 06☐ **Audio Engineer/Producer:** Audio producer, sound engineer, mixer, post-production engineer, mastering engineer, staff/independent sound engineer, producer, editor or other audio engineering titles.
   - 07☐ **Technical Management:** Maintenance engineer, engineering manager or other technical titles.
   - 08☐ **Corporate and Operation Management:** President, owner, partner, vice president, director, general manager, studio manager and other corporate and financial officials, as well as consultant, professor or instructor.
   - 09☐ **Musician/artist**
   - 10☐ Other (please specify) ________________

4. Please check the ONE type of facility or operation that best describes your business classification:
   - 25☐ **Recording studio:** (commercial fixed location and mobile recording trucks, systems and studios)
   - 26☐ **Sound reinforcement:** (installers, operators and designers of concert and other temporary sound systems)
   - 27☐ **Radio broadcast production:** (including non air-chain station-based and independent radio production facilities)
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   - 30☐ **Video sweetening/post-production facility:** (independent audio for video production and post-production service companies)
   - 31☐ **Film re-recording:** (in-house and independent film sound production and post-production companies and facilities)
   - 32☐ **Disk/CD mastering**
   - 33☐ **Audio/acoustic/technical consultant:** (system and equipment consultant, studio and production facilities designer)
   - 34☐ **Corporate, educational, government:** (audio recording facilities in manufacturing and service companies, and other businesses, schools, universities, governmental facilities)
   - 35☐ **Audio equipment retailer/dealer/distributor/rental company:** (sales and/or service of professional audio product lines)
   - 36☐ Other (please specify) ________________

5. What statement best describes your role in purchasing professional audio recording equipment, systems and software?
   - A☐ Make final decision to buy specific makes, models, services or programs.
   - B☐ Specify or make recommendations on makes, models, services or programs
   - C☐ Have no part in buying or specifying.

Recording ENGINEER/PRODUCER

Complete the following OR provide address label from cover: TO ENSURE SERVICE, YOUR NAME AND TITLE MUST BE INCLUDED.

Please print or type:

Name ____________________________

Title ____________________________

Company ____________________________

Address ____________________________

City ________________ State ______ Zip ______

Phone No. (_____) ____________________________

Do you wish to continue to receive Recording Engineer/Producer? Yes ☐ No ☐

1. Signature ____________________________

2. Date ____________________________

(If you are not a current subscriber and wish to apply for a subscription, please see the NEW SUBSCRIBER APPLICATION elsewhere in this issue.)

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Special Instructions:

- You must answer all questions in order to qualify to receive Recording Engineer/Producer.
- You must sign and date the card.

Do you wish to receive Recording Engineer/Producer? Yes ☐ No ☐

1. Signature ______

2. Date ______

SJ9

NEW SUBSCRIBER APPLICATION

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Recording ENGINEER/PRODUCER

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TO ENSURE SERVICE, YOUR NAME AND TITLE MUST BE INCLUDED.

Please print or type:

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Title ______
Company ______
Address ______
City ______ State ______ Zip ______
Phone No. ( ) ______

Special Instructions:

- You must answer all questions in order to qualify to receive Recording Engineer/Producer.
- You must sign and date the card.

Do you wish to continue to receive Recording Engineer/Producer? Yes ☐ No ☐

1. Signature ______

2. Date ______

AJ9

3. Please check below the ONE category that best describes your primary job function:
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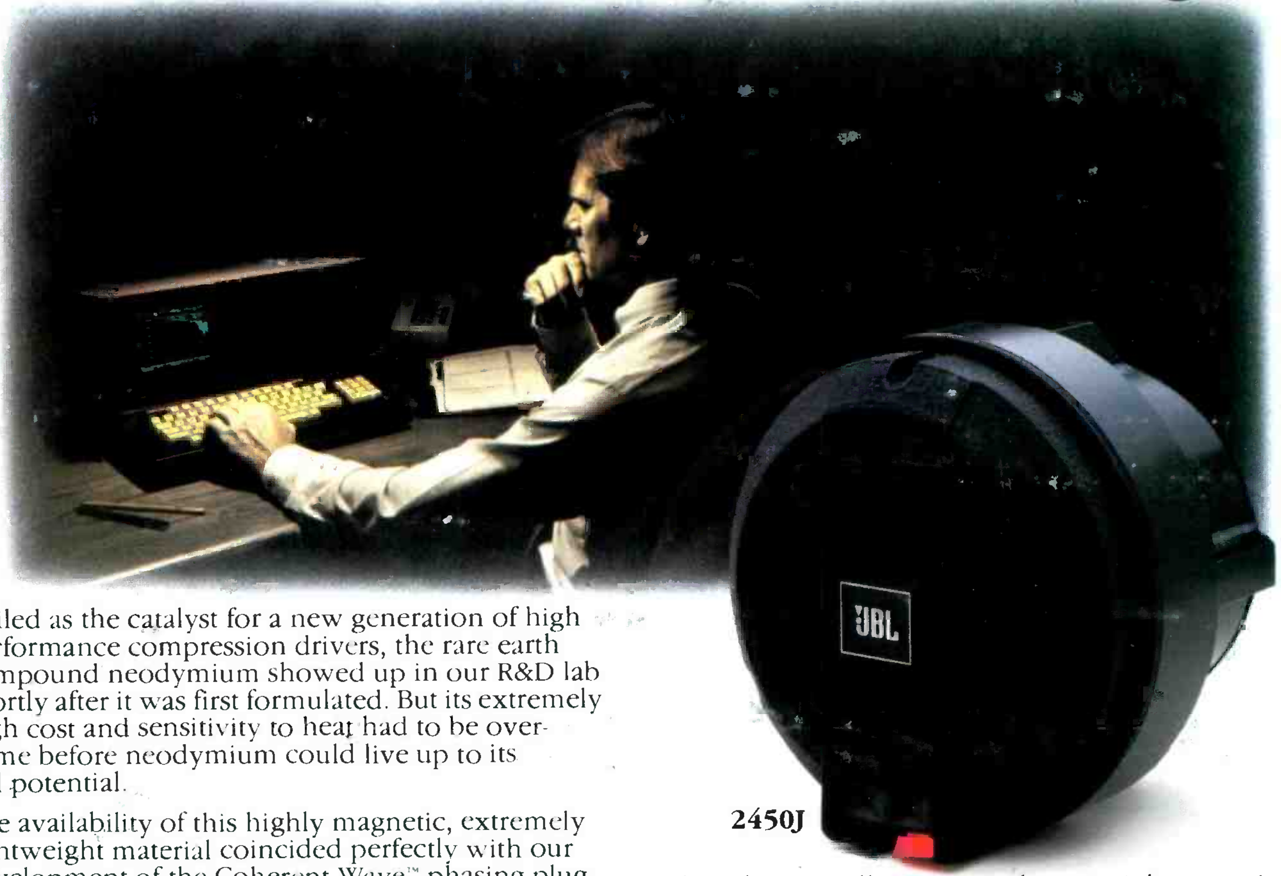
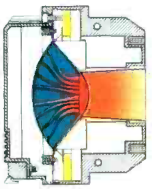
It Takes More Than A Little Neodymium To Change The Face Of Driver Technology.



Hailed as the catalyst for a new generation of high performance compression drivers, the rare earth compound neodymium showed up in our R&D lab shortly after it was first formulated. But its extremely high cost and sensitivity to heat had to be overcome before neodymium could live up to its full potential.

The availability of this highly magnetic, extremely lightweight material coincided perfectly with our development of the Coherent Wave™ phasing plug.



JBL's 2450J Neodymium Compression Driver with Coherent Wave™ phasing plug provides in-phase combining of sound waves for extended high frequency performance. JBL's patented diamond surround titanium diaphragm with new embossed dome greatly reduces distortion and damage at high SPL.

This new design, a phasing plug with annular apertures of constant path length, uniformly directs sound through to the throat providing in-phase combining of sound waves for extended high frequency performance. This new technology is combined with our patented diamond surround titanium diaphragm, incorporating a new embossed dome, to reduce the possibility of distortion or damage at high SPL.

The 2450's smaller size translates to tighter spacing of horn arc arrays, more even and precise coverage and greatly reduced requirements for delay. Plus, the 2450 nets out at a mere 4.8 kg (10.5 lb). The benefits of this dramatic weight reduction include lower shipping costs to the site or on the road and significantly less load bearing requirements for both structures and rigging. With built-in mounting points, the 2450 will take much less time to install.

Yes, it took more than a little neodymium to change the face of driver technology. But we're confident you will find the breakthrough results were certainly well worth the wait and the effort.

JBL

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