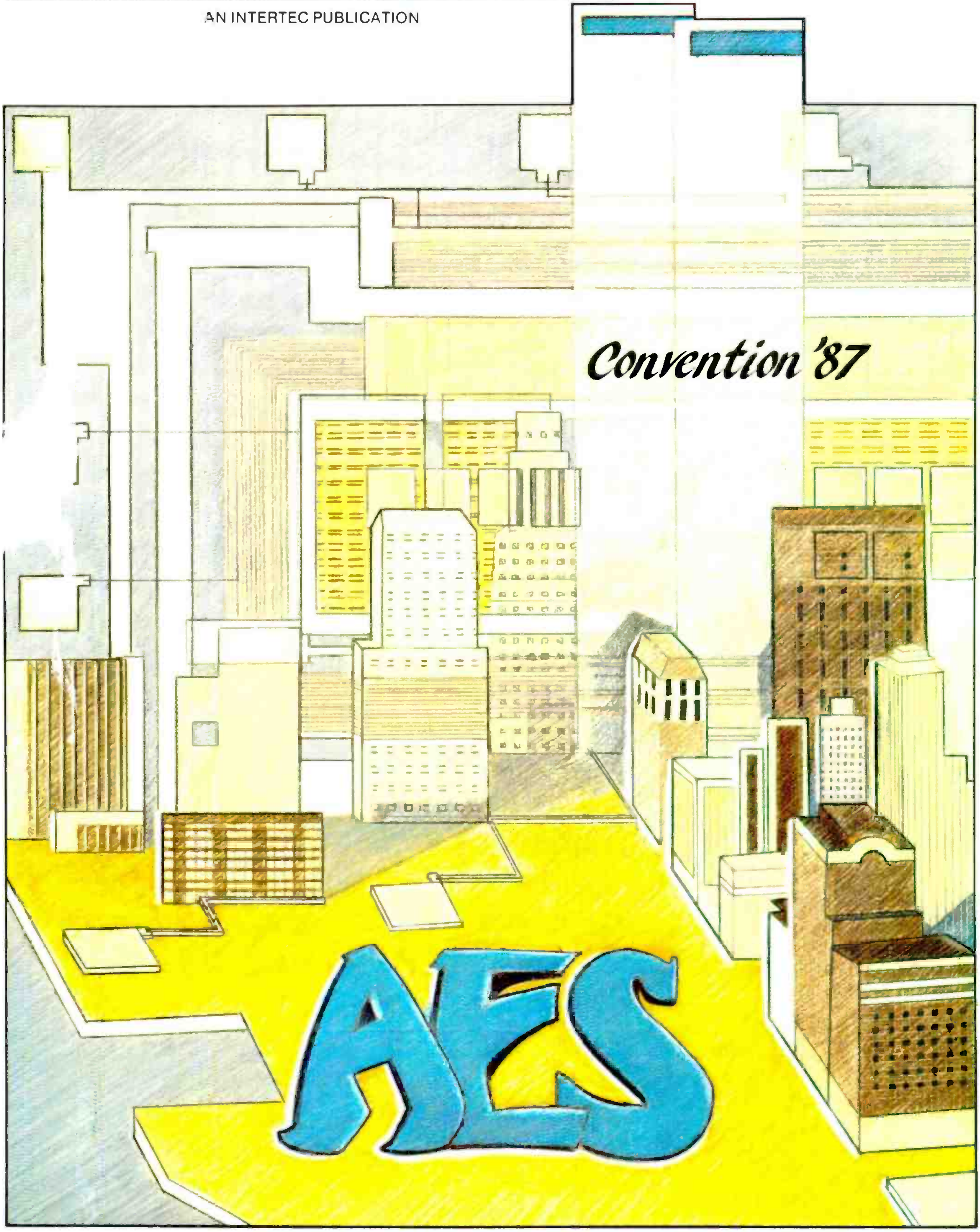
October 1987

Recording ENGINEER/PRODUCER

The Technical Journal for Audio Professionals

AN INTERTEC PUBLICATION

RE/P Readers to Government: Butt Out—No Copycode Pg. 8



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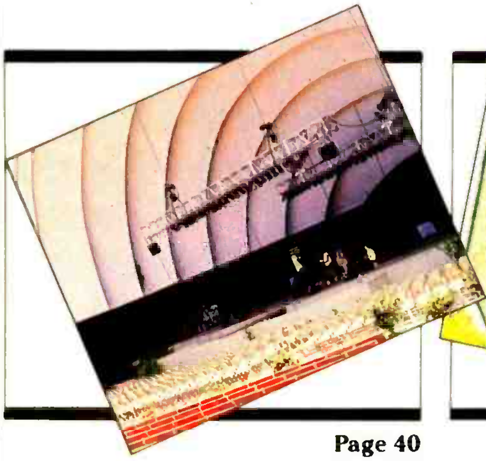

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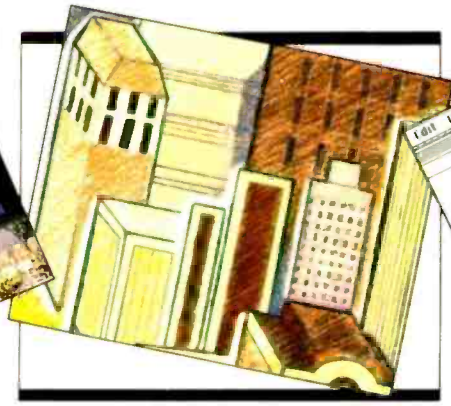

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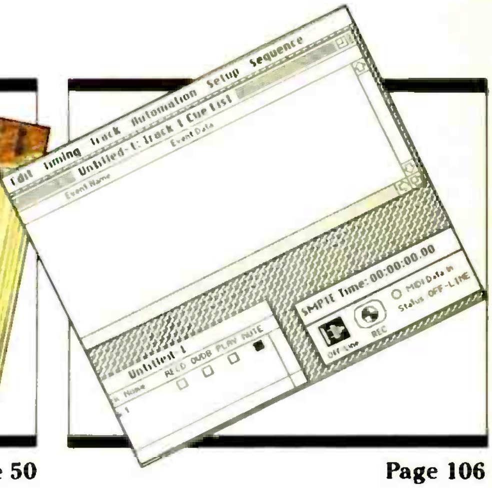

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On the Cover

Pictured are portions of the TRW TDC1034 4-bit D/A converter chip and the New York skyline, which represents this year's AES theme of Audio and Video: Analog Present, Digital Future. Cover design by Alecia Wright, **RE/P's** graphic designer.

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Editorial

The Day the Music Died

Once upon a time in America, there were these guys in "suits," who worked for record companies. They were very worried. What was worrying them was a new technology. It seems that record companies are always worried about new technologies. We remember the hysteria they generated during the introduction of the stereo, the compact audio cassette, the L-cassette, the home video recorder, the CD and so on . . .

The most current incarnation of this worry syndrome revolves around an as yet basically unavailable system most commonly known by its acronym R-DAT.

It appears from all available information that the "suits" are worried about huge numbers of consumers running out, plunking down over $1,000 to buy a machine, hooking it up to their Compact Disc system, inserting the required sampling rate converter (a pricey, and not readily available item in its own right), and spending hour upon hour (in real time) dubbing copyrighted material for later distribution. We find this sequence of events both highly unlikely and not just a little disturbing.

Because of the above, we as recording industry professionals are being asked to give our blessings to a system created to solve this fictional problem and one that will, without question, seriously degrade the audio performance levels we all strive so hard to produce and maintain.

Let's see if we understand this issue correctly . . . copycode proponents want us to cut a very deep notch in our masters centered at 3,840Hz; (At least in the current versions of the copycode system). And they say that absolutely no one will notice. Wait . . . , we've got a better idea—why don't the "suits" let us boost that same area by 60 or so decibels. They say all the mixes we've given them lately have sounded a bit dull anyway.

As Fletcher/Munson demonstrated eons ago, the human ear is highly sensitive to sound in this region. Even good old Ma Bell has taken advantage of this fact by rolling off the upper and lower ends of the audible spectrum to maximize the S/N ratio and clarity of phone conversations, not exactly a hi-fi medium.

Several recent articles in the audio trade press have detailed the available technical aspects of the copycode chip. There are, however, some points from a musical perspective which haven't been discussed quite as much. One is the fact that there is no fixed frequency band in the usable audible spectrum which can be arbitrarily cut or boosted without affecting the original sonic blend of any program material.

The second is the fact that CBS claims to have selected a center frequency that does not fall exactly on fundamentals, demonstrating just how low-fi their thinking is on this whole subject.

Maybe we can help them out of their hole with a short and certainly not complete list of real world musical instruments that can produce fundamentals, second or third harmonics, at or near "notch central":

Acoustic Piano
Violin
Cymbals
All Synthesizers
Xylophone (and other high bells)
Triangle (and other metallic percussion)
Trumpet
Piccolo, and so forth . . .

The list could go on but the point is obvious. These or most other instruments, when tuned to a reference of A-440 and processed at normal machine speeds, are capable of sounding notes very close to the notch, if not in or on it. By using any number of production techniques such as variable speeds, non A-440 tuning references, or one of our industry's favorite microtunable FM synthesizers (with or without the use of a pitch wheel) it is quite easy to produce material exactly at the proposed notch point.

Even more importantly, the whole sequence of events leading up to the current congressional hearings once again is involving the federal government in an area where it truly has no rational reason to be. We should have learned by now that once you involve big brother, you instantly acquire a massive bureaucracy, vague and often complex regulations, and little of a positive nature. In addition, no industry should be allowed to use the law to prevent a technology from being implemented, unless irrefutable proof has been offered that the technology will cause serious harm to the public at large. This is most certainly not the situation with regard to R-DAT.

In fact, given the previous history of the cassette, exactly the opposite, in all probability, will occur. We all have memories of what was said about the cassette, how it would kill record sales, allow mass private copying of material, cripple profits, and on and on. It is clearly evident that none of these horrific scenarios occurred.

If, in the final analysis, there absolutely has to be some sort of copy protection (and this has yet to be demonstrated) beyond the already existing sampling rate incompatibility, why hasn't there been more mention of the totally transparent write/copy/protect flags which are currently incorporated into the digital subcodes of CDs and digital audio tapes? Could it be that CBS has something to gain by owning the copycode system?

What bothers us most is that "the suits" and the government are trying to implement this without consulting any of the professionals that would have to use this system. *That's you.* And judging by some preliminary data we've received in response to our R-DAT survey, you—our readers, this country's working audio professionals—feel the same way.

As the story on page 8 details, an overwhelming majority of you think that the government should stay out of the issue altogether. We've spent our professional lives trying to create the best-sound audio possible, and now legislation is pending that would prevent just that.

As one of our readers commented, "the copycode scheme is like trying to prevent the theft of the Mona Lisa by cutting a hole in it."

We Agree!

Michael Fay
Editor

Frederick J. Ampel
Editorial Director

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

Now We've Got A Small One.

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

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

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

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

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

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

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UREI

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News

RE/P names new editor, announces staff additions

Michael Fay, an independent recording engineer and producer from San Diego, has been named the new editor of **Recording Engineer/Producer**. The change, announced by Cameron Bishop, **RE/P**'s vice president, was effective Sept. 1.

Fay has 17 years' experience as an engineer/producer in all areas of audio production. Since 1979, he has been the owner/manager of Michael Fay Productions, providing engineering, production and consulting services, with a main focus on advertising and corporate/industrial clients.

"Michael's addition to the staff strongly augments **RE/P**'s position as the pro audio industry's technical journal," said Dennis Milan, **RE/P**'s publisher. "He is a working professional with a clear idea of how to present the best possible material for our readers."

Fay will work out of **RE/P**'s Hollywood office.

In addition, two other staff changes were announced. Frederick J. Ampel has been named **RE/P**'s editorial director. Ampel, the editor of **Sound & Video Contractor**, another Intertec publication, will be editorial director for both magazines. Dan Torchia, formerly the managing editor for **RE/P** and Intertec's electronics magazines, is now the full-time staff editor for **RE/P** and **S&VC**.

Please send News items to the editorial and production offices: Michael Fay, RE/P Editor, 1850 N. Whitley, Suite 220, Hollywood, CA 90028 and Dan Torchia, RE/P Staff Editor, Intertec Publishing, 9221 Quivira Rd., Overland Park, KS 66215.

Marcus-Lambert PR begins new venture

Mel Lambert, former editor of **Recording and Engineer/Producer,** and Bobbi Marcus have announced the formation of a new marketing and public relations company for the pro-audio industry. The new company to be known as Marcus-Lambert, will look to service the needs of high-end equipment manufacturers and recording and production facilities.

Marcus, a 10-year veteran of public relations and marketing has handled a variety of recording and production facilities and clients.

SPARS offers studio business cassettes

SPARS is offering a series of 90-minute cassettes and a planning notebook from the recent Studio Business Conference at the UCLA School of Business. Topics include: "Constructing a Business Plan," "Opening a New Studio," "Evolution of the Multi-Studio Operation," "Adding a New Location," "Entry into Video," "Adding a Synthesizer Room" and "Getting into the Rental Business."

Each cassette is $15 for SPARS members and $18 for non-members. The notebook is $30 for members and $40 for non-members. The entire cassette series and notebook can be purchased for $120 for members and $150 for non-members. For ordering, contact SPARS at P.O. Box 11333, Beverly Hills, CA 90213; or call 818-999-0566.

The organization also met in New York during June for the SPARS regional meeting and to celebrate the eighth anniversary of SPARS.

Recording ENGINEER/PRODUCER

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

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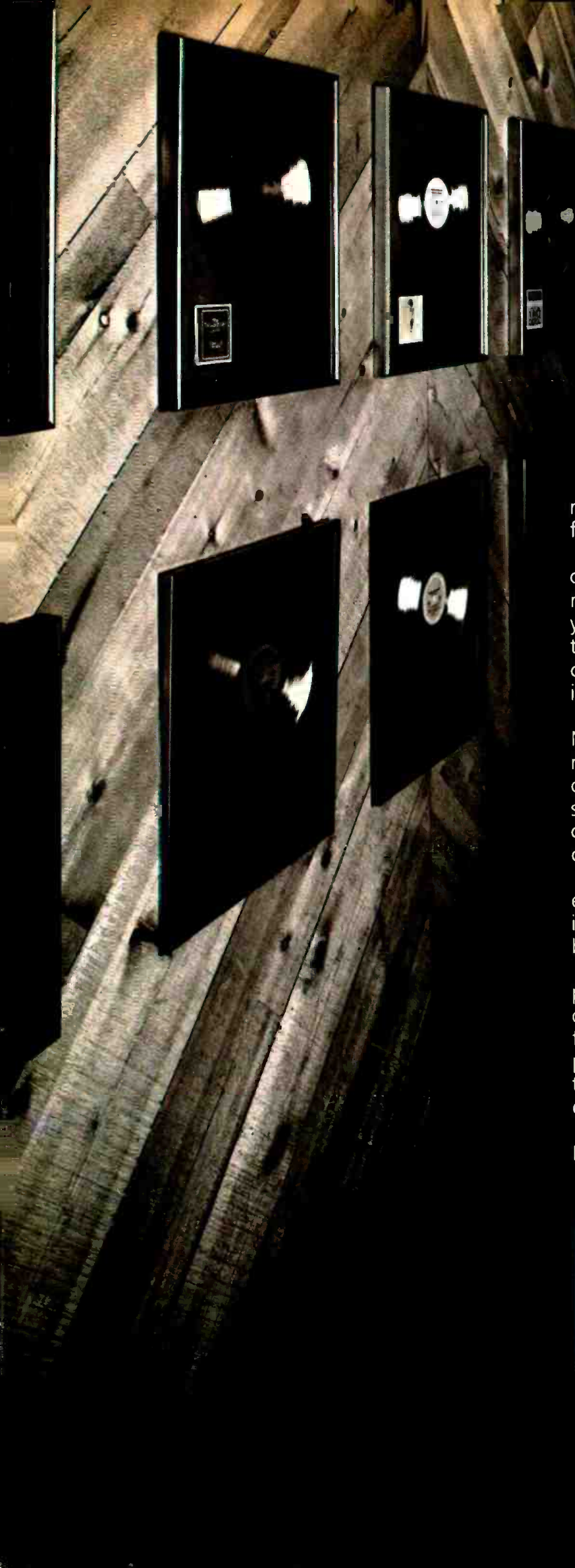


WHAT YOU DO WITH THE M-600 MIXER IS YOUR BUSINESS.

That's why we've designed it to meet or exceed your most demanding requirements. And made it the easiest, most flexible professional mixing console you'll ever work with.

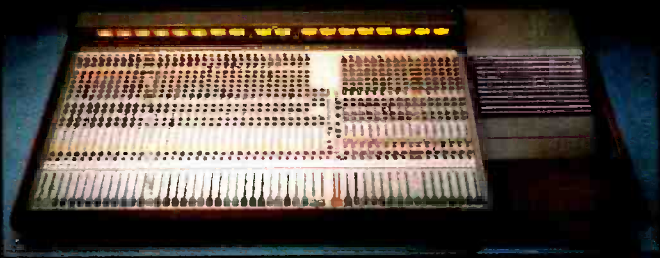
The M-600 is modular. Which means you can custom configure the console to *your* audio or video production needs. The M-600 lets you choose up to 32 input channels, or you can start with 16 or 24 input channels and expand the board as your needs change. Optional stereo modules can also be added to provide even more line inputs for MIDI instruments and video production convenience.

Installation and wiring is exceptionally easy. The M-600 is the only modular mixer that's available with all the necessary finished cables and installation hardware. And that can eliminate a lot of installation hassles and expense. At the same time, no other mixer at its price gives you multi-pin, computer-type connectors for quieter, more secure connections.

But the real pleasures of the M-600 will only be evident after it's in your studio. Up to 64 stereo or 128 mono inputs can be accessed directly from the top panel. A patch bay can be added for fast, flexible routing. That's convenience.

The M-600 has all the features you'd expect in a professional mixing console. Like balanced insert patch points on all inputs, PGM busses as well as the stereo master buss for increased signal processing capability. Plus sweep-type parametric EQ, balanced inputs and outputs, phantom power, talkback/slate channel and all the audio performance you'll ever need. Without the exorbitant price you don't need.

So check out the M-600 modular mixing console. It's ready for fame when you are.



TASCAM

Circle (6) on Rapid Facts Card

News

Readers disapprove of Copycode legislation

By Dan Torchia, staff editor

RE/P readers overwhelmingly disapprove of attempts to regulate R-DAT machines, and think that inserting a notch in digital material cannot be accomplished without compromising audio quality, according to preliminary results of an **RE/P** reader survey.

In the first attempt to accurately gauge how working engineers and producers feel about the R-DAT-Copycode issue, **RE/P** surveyed its readers about R-DAT, Copycode, home taping, piracy and government intervention.

Although the results are preliminary, they portray an industry highly aware of the issues and overwhelmingly non-supportive of attempts to regulate technology.

A summary of the results to date:

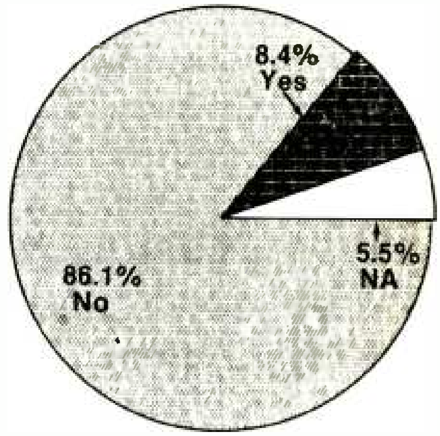
- Almost everyone surveyed (92.3%) was familiar with proposed Copycode legislation.
- A total of 92.3% think that consumers should be allowed to copy analog or digital material for personal, non-commercial use. Seven percent said they should not, while 0.7% did not answer.
- While almost three-fourths of respondents, 73.6%, did not approve of unauthorized duplication for commercial purposes, almost as many, 61.9%, thought the Copycode would not solve the problem.
- Most are doubtful of Copycode's effectiveness; 86.1% thought that it was not possible to insert a notch between 3,500Hz and 4,100Hz without affecting quality. Another 8.4% said that it was possible, while 5.5% did not respond.
- Finally, an overwhelming majority of respondents do not approve of pending legislation or government intervention. A total of 89% of respondents opposed pending Copycode legislation, with 8.1% approving. And 93% said that there should be no government intervention in the development of recording industry technology; 3.7% said there should be, and 3.3% did not answer.

Judging from the response rate, R-DAT is an important industry issue. At presstime in early September, 27% of those receiving a survey had already returned it. The survey was mailed on Aug. 26. If the response rate continues, the response rate could reach 60%, a phenomenal return for a survey, said Cameron Bishop, group vice president of Intertec

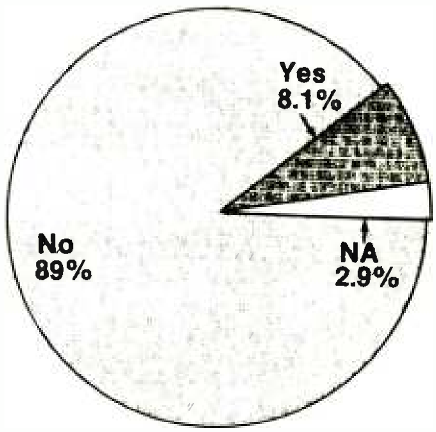
Want to comment?

*If you are against Copycode and want to make your views known to members of Congress, a special card is included on page 123, the last page of the magazine before the back cover. The card, preaddressed to **RE/P**, contains a short letter highlighting some major concerns about the Copycode system. There is also space for your own comments. If you are against the legislation, fill the card out, attach postage and mail it in.*

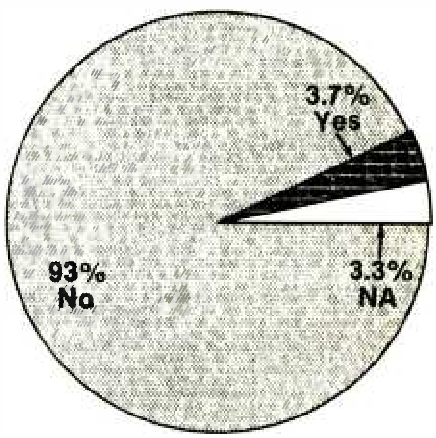
***RE/P** will collect all responses and forward them to the appropriate congressional leaders involved in the legislation. A mass response is often better than many individual responses, and returning the card to **RE/P** will allow you to collectively voice your opinion along with your peers in the industry.*



Is is possible to eliminate or "notch" digitally recorded material in a band somewhere between 3500 and 4100 Hertz without affecting audio quality?



Should the proposed R-DAT Copycoding legislation be passed and implemented?



Should there be governmental intervention in the development of any recording industry technology?

Figure 1. Partial results of RE/P's reader survey on R-DAT and Copycode.

News

Publishing, publishers of **RE/P**.

"I think the results speak very highly of our readers," said Michael Fay, **RE/P**'s editor. "The pro machines they are going to use are exempted from the legislation, so they could have ignored this whole thing if they wanted to.

"But they're not concerned with the money issues, they are concerned about the art of recording. It is their work that would be compromised, and that's why they are responding as strongly as they are."

The majority of respondents have been technical management and engineering, totaling 60.8% of total respondents. Another 23.8% were company management. Operations and productions management accounted for 1.47%, while 0.7% listed miscellaneous titles.

Legislation banning all consumer R-DAT machines without an encoding chip is pending in subcommittees in both houses of Congress. The National Bureau of Standards started testing the Copycode system in early September, with results expected in a few months.

RE/P's reader survey was conducted by Intertec Publishing's research department, under the direction of Katie Smith. Questionnaires were mailed on an nth-name, or random, basis to readers on **RE/P**'s circulation list. Final results will be published in the November issue.

Reader comments

*At the end of the survey that **RE/P** sent out, respondents were asked to comment about the pending Copycode legislation and its potential impact on the pro audio industry. As might be expected with a controversial issue, many people stated their views. Some comments from the batch of initial returns appear below; more will appear in the November issue.*

"If passed, the pending legislation will compromise the technology and quality, thus limiting the ultimate potential and sales. The real answer is enforcement of existing copyright law."

"It is very dangerous to have non-musical or non-technical people making decisions governing a musical and technical industry."

"How can something that removes more than ½k of bandwidth be called 'hi-fi?' The problem is not being attacked at the source."

"Empty barrels make the most noise—let's move forward in our industry and forget this nonsense!"

"I personally have spent years developing my craft and recently purchased equipment at substantial expense to improve the audio quality of our facilities. I feel the legislation will destroy the end results of my efforts."

"It will ruin my company."

Why your next console should be as difficult to hear as it is easy to operate.

The studio is more complex and less forgiving.

Electronic production techniques using MIDI and SMPTE sync require more control than a "wire with gain" can provide. But as functions and components accumulate, the console's signal path has grown more complex, and its audio performance has suffered. On analog recordings, higher levels of crosstalk, noise and intermodulation were an acceptable price for additional control. On digital multitrack, however, these flaws become glaringly obvious.

Crosstalk blurs the stereo image.

Now that digital recorders have virtually eliminated crosstalk, this is an especially annoying problem. *The AMR 24 matches the channel separation performance of digital multitracks* because it employs balanced buses that eliminate crosstalk the same way mic inputs do. This radical design approach takes full advantage of digital's more coherent stereo imaging.

Balanced buses also eliminate the intermodulation that plagues the sound of conventional "virtual ground" mix amps. *The AMR 24's noise floor is constant whether you route one input to a group, or thirty six.* So you can concentrate on the music without distractions from the mixer, even on digital multitrack.

Features shouldn't degrade audio performance.

Automation widens creative possibilities — and narrows the margin for console error. For example, FET mute switches that are "silent" individually can produce audible glitches when grouped. The AMR 24's carefully controlled switching time constants eliminate this problem.

Every circuit in the AMR 24 has been calculated with equally close attention. Each stage has at least 22 dB of headroom; total dynamic range is over 100 dB. Even so, *unused stages are bypassed to produce the shortest effective signal path in every operating mode.*

Perhaps the AMR 24 is a product of extremist engineering. But as we see it, optimum audio performance, not simply a revised layout, is what makes a console automation - and digital-ready.

The feel is familiar, the functions are unprecedented.

The AMR 24 facilitates innovative production techniques within a classically split configuration. Master Input Status switches select mic inputs or line returns on all input channels simultaneously. In its mixdown configuration, the AMR 24 will handle up to 60 tracks, because the 24 Track Select switch changes the monitor returns to line returns normalled to your second 24 track (or to synchronised "virtual tracks" from synthesisers and samplers). The monitor returns have aux buses, solo and mute, plus four bands of EQ and long throw faders, so this flexibility is achieved with no loss of audio quality. For additional effects returns, the Fader Reverse function creates an additional 24 patch points through the cue send faders.

Imaginative design and uncompromising construction give the AMR 24 flexibility and sonic transparency that represent clear achievements: especially clear on digital recordings. For all the facts on this innovative console, send your business card or letterhead to:

DDA AMR 24

SPARS

Klark-Teknik Electronics Inc., 30B Banfi Plaza North
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Circle (9) on Rapid Facts Card

Letters

Music software

From: Larry Polansky, assistant professor of music, Music Department, Mills College, Oakland, CA.

Paul D. Lehrman's "Managing MIDI" column in the June issue points out some important questions and issues in the continuing development of music software. He very correctly emphasizes that there continues to be a greater and greater need for expandable, *flexible* software.

The next generation of software will almost certainly lean more toward language design, and less to restrictive, and inherently limited, applications. Expanded system exclusive implementations in commercial hardware, a more sophisticated user base and the collective creative urge of musicians and producers will, I hope, encourage software designers to more and more often leave behind their imaginary end-user who is, it appears, seen to be impatient with complex machine intelligences, and unwilling to accept open-ended designs which encourage creative user interaction. To implement the kinds of important things that Paul suggests, MIDI software needs to be user-definable and, of necessity, design issues need to be concerned with what many of us refer to as "music languages."

There are already several such environments, including MIDILisp/FORMES from IRCAM, our own HMSL (Hierarchical Music Specification Language). Dan Kelley's MASC, and Ron Kuivila and David Anderson's FORMULA, to name just a few. All of these "languages" are available for standard personal computers (Macintosh, Amiga, Atari, IBM, etc.). All of these environments are also characterized by a high degree of generality and a correspondingly high learning curve. In fact, for three of these environments, the user needs to be a reasonably competent FORTH programmer and, for the fourth, a LISP programmer.

Power and generality are often proportional to ease of use, yet ease of use is also directly related to the general sophistication of the user base. This sophistication will only improve if software designers recognize the tremendous untapped abilities of composers, musicians, producers and engineers, and give them programs and languages worthy of their talents. Software designers will not leave the music world behind; they will bring it forward with them, happily in tow.

Random-Access Editing

From: Bob Katz, New York.

I just finished reading your remarkable July issue on digital technology. I attempted to speed read to avoid future shock, but succumbed nevertheless. My first reaction to the issue was that my article on advanced 3-machine digital mixing has become instant "primitive" history, in the light of the AMS AudioFile (also reviewed in the July issue) and similar disk-basked editing systems.

As a matter of fact, random-access editing *a la* AudioFile would prove to be *exceptionally* efficient in editing the spoken word for commercials, films, radio, etc. One fact that I did not cover in my article was that ¼-inch editing of spoken-word audio normally involves removing and adding many tiny pieces of tape, containing "lip smack," extraneous noises made by the actor, and room tone. Typical ¼-inch spoken-word edits contain, on the average, splices about every five seconds, often pulling very short pieces out of the tape.

It is easy and quick to make such splices on a ¼-inch tape machine. It is almost ridiculous, however, to attempt this type of fine editing on a VCR-based system, with its time-consuming rehearsal process, and difficulty of pulling pieces from within the middle of an already-edited program.

Clearly the AMS AudioFile will provide an efficient, razor blade-less method of cutting spoken word. (We were very lucky that Christopher Plummer has a pretty "noiseless" mouth, or the voice design alone on *Nutcracker* would have taken several days via the Sony DAE-1100 editing system.)

I would like to know how the price of the AMS AudioFile compares with the unique complement of equipment I assembled for that 3-machine mix: three BVU-800's, three PCM-1630's, one DAE-1100 and a TimeLine Lynx synchronizer. I would also like to know whether the AudioFile can control the gain of each D/A output in the digital domain, because the level-change information it stores could allow the unit to perform automated mixing. [Currently, AMS doesn't provide digital control of level changes, this capability may be added later—*Editor.*]

Then, its eight outputs could connect into a very simple production-type audio console at unity gain, without passing through VCAs or other signal degrading devices. It would be only one step from there to feed a small digital mixer avoiding eight D/A conversions.

The July report of a "Transcontinental Digital Overdub" by Paul Lehrman and David Rideau is also future shocking. However, I should inform you of a recent technical development that will allow multiple musicians throughout the country to simultaneously perform and overdub via satellite, without experiencing the time delay problems mentioned in the article.

The device is called a Digital Advance Line (DAL), now under construction at a U.S. lab and incorporating the latest in superconductivity and time-predictive techniques.

Soon, a singer will be able to send his voice via satellite to a remote site and, by inserting the DAL into the satellite return, can hear his own voice in the headphone mix without echo problems. In fact, the DAL actually *anticipates* what the singer will sing *before* he sings it.

To encourage sales of this time-advancing unit, the manufacturers, in a unique marketing ploy, are asking for their $1 million price to be paid in 1950s dollars!

Thanks for giving me the opportunity to comment. R·E/P

News, continued

DDA delivers in Europe and United States

DDA has installed DDA consoles at the following locations:

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— Sound Rental Services, Parkersburg, WV; two D series consoles for house sound and monitors.

—Saban Productions, Studio City, CA; AMR24 36x24 console and a D series for its post-production room.

— Abbey Road Studios, London; DDA console and D series 16x2 console for mobile digital recording.

— Peter Rafelson, composer/producer; AMR24 36x24 console.

— David Dundas, London; AMR24 28x24 console.

— Tape One Studio, London; S series 6x2 console.

— Scacco Matto Studios, Lavagne, Italy; ARM24 44x24x2 console with 64-channel Audio Kinetic Mastermix.

— Orinoco Studios, London; ARS24 36x32 console with 36-channel Audio Kinetics Mastermix and remote patchbay.

Continued on page 95

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Managing MIDI

By Paul D. Lehrman

Because I spend an inordinate amount of my time working with electronic music, when I go out at night the last thing I want to hear is *more* electronic music. I listen to acoustic music: jazz, classical, folk and New Age material. Being overtrained and over-left-brain-oriented, I often find myself analyzing what I'm listening to, rather than just enjoying it.

During a recent performance of an acoustic trio playing California hot-tub music, a thought occurred to me. Although it was almost unrelievedly mellow, there were moments in the performance of palpable tension. The tension wasn't caused by harmonic motion; there wasn't any. The tension was only partially caused by dynamic changes; there's just so far you can go with an oboe.

No. Generating the tension were minute tempo changes and the way the players interacted with each other—some pulling, some pushing. There was no melody to speak of either. Whatever feeling of phrasing existed was due almost solely to the rhythmic interaction. The tempo changes were very subtle, probably less than 2%, but they were used so effectively, it made for some truly exciting moments.

As usual, I tried to apply what I had just heard to my own music, and thought about how I can make a sequenced piece do that, which led me to thinking about the whole issue of timing in computerized music.

Sequencers lend themselves, as we all know, to perfect rhythms. Some sequencers don't let you do anything *but* perfect rhythms, which is a drag if you want to write anything but dance music. On the other hand, there are plenty of sequencers that divide a beat into 100 or more "ticks"; that kind of resolution is sufficient to reproduce most of the rhythmic subtlety of a real performance.

Trouble is, too many programmers don't take advantage of that subtlety. Instead, they record everything in "step time," or quantize the daylights out of any track they lay down as soon as it's done.

Step time is a useful crutch for those with no keyboard chops, but it should be used sparingly. If you have a 16th-note line that you cannot possibly play in real time, it'll sound a lot more musical if you slow down the recording tempo by a factor of four and lay it in that way, than if you were to put it in as metrically perfect 16th notes in step time.

Quantizing is also a useful tool, if used correctly. Some sequencers make you quantize an entire track, but they should be avoided. Quantizing even an entire phrase is an invitation to boredom—leaving a few notes a little ragged creates a break from regularity that the ear finds welcome.

Some sequencers feature a sensitivity control within their quantization functions: notes that fall close to a beat, say within 3 or 4 ticks, are not quantized, while those outside that window are. Occasionally you'll see this function reversed, and notes far away from the beat are left alone while those close in are lined up. Both of these methods can be helpful, and judicious application of either or both can smooth out a line without pounding it into submission.

Quantizing even an entire phrase is an invitation to boredom.

You often come across a quantization function that allows you to advance or retard the quantized notes slightly with regard to the beat. If you want your snare drum to lag behind the beat, you can quantize it separately and place it a few ticks late.

Another feature that's beginning to appear is a reverse quantization function, which introduces small random changes in the placement of notes.

Trouble is, too many programmers don't take advantage of that subtlety.

All of these options are good, but they don't go far enough. To create that true rhythmic tension I was hearing that night, you need a function that advances or retards a beat *progressively* over time, so that the snare drum that starts in bar 38 is two ticks behind the beat but, by the time it gets to bar 56, it's 11 ticks ahead.

Even better would be a "range of ranges" function, in which you could specify a random placement of the beat at one point, a different random placement at another, and have the sequencer interpolate everything in between. To use a similar example, specify a range of −1 to −5 ticks in bar 38, and a range of 7 to 14 ticks in bar 56. The snare drum will get progressively earlier, but not in a lockstep manner.

Sequencers are finally beginning to appear that automatically calculate tempo change slopes over time—specify one tempo here, another there and the hardware takes care of the rest. A popular technique for humanizing tempos is the use of a tempo loop, in which each phrase contains subtle tempo changes that repeat over the length of a track.

For example, I recently wrote a love theme based on a 6-beat phrase that was basically at a tempo of 128bpm. At every sixth beat I lowered the tempo to 120, which created a nice sense of tension and anticipation at every phrase. (Oh, Heathcliff!)

Some sequencers, however, have trouble working at this level of sophistication with tempos. When you're trying to do this sort of thing with a SMPTE-to-MIDI converter, all bets are off. All of these converters have a built-in tempo map to specify where tempo changes occur in the time code bit stream, and these maps have a finite number of entries. Even the newest boxes allow fewer than 200 entries. So, if you're throwing tempo changes at it at the rate of four a bar, it will run out of room very quickly.

When MIDI Time Code-based sequencers arrive, in which the hardware converters do not handle tempo information, this should not be a problem. Already one sequencer about to be released has a clever mode in which clocks coming from a SMPTE-to-MIDI convertor are used not as a tempo reference, but instead as an "absolute" timing pulse, and the sequencer generates its own internal tempos from it.

Devices that take external human input—whether in the form of switch closures ("taps") or audio triggers (live or on tape)—and generate tempo information from it have been with us for a while. They are getting better, however, and now can help humanize sequenced tracks to a great degree.

An idea I played with a couple of years ago, but which got lost somewhere, is the ability to manually insert beat and bar lines, either with a tap button or on a computer screen, *after* a sequence has been recorded, with the sequencer calculating the tempos accordingly. The arithmetic might get a little hairy, but for manipulating tracks in a truly human fashion, it would be worth it.

Remember that, as any performer will tell you, timing is everything. R·E/P

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

Sound on the Road

By David Scheirman

Touring sound system technicians are often confronted with an interesting dilemma while traveling with a concert system on the road. Should the entire speaker system be set up and used because it has been rented and paid for by the client (typically a touring entertainment group), or can a partial system be set up and integrated into a temporary hybrid system by making use of available house facility speaker clusters and other audio gear?

The former choice can mean longer load-ins and load-outs, and sometimes a reduced stage performance area. The latter can save on the local stage crew labor rates, but it can greatly increase the workload on the touring sound system techs as they attempt to interface with unfamiliar gear. And finally, the results may be questionable if the house gear is not thoroughly examined and tested ahead of time.

This dilemma is encountered more frequently than you might think, and making the proper decision in this regard during that first hour as the truck is being unloaded can ultimately mean the difference between a smooth production and a shaky one. There are a variety of parameters to consider, including the labor costs, available setup and teardown time, state of repair of the installed or locally offered gear, and the employing artists' wishes. Touring sound company pride, interestingly enough, can be a consideration as well.

House systems today in many concert venues that regularly cater to touring musical shows are much better than those encountered a decade ago. This fact is due, in part, to the positive results that can take place when touring sound companies begin to focus some of their efforts toward the permanent installation field. For example, Audio Analysts has been involved in an impressive sound system installation at a civic arena in Montreal; Clair Bros. has installed a full hanging system at a performance auditorium in Anchorage, AK; Maryland Sound Industries has contemporary system installations in the D.A.R. Constitution Hall and the refurbished Baltimore Arena; and Stanal Sound has installed systems for the Greek Theatre and the Universal Amphitheatre.

Such installed systems can present a touring sound technician with valuable resources for making the show the best that it can be, that night, in a particular venue. The difficult point about this issue, perhaps, is the question of whether such an installed system will really fill the bill for every show that comes through, regardless of program content and musical type, or whether provisions must be made from night to night to interface with additional touring sound gear that travels with different shows.

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

For stage monitor systems, the issue is fairly clear-cut: most traveling artists who are paying to carry a touring stage monitor system from venue to venue will certainly intend to set it up, whether or not a stage monitor system is included in the house sound system package. Such monitor systems used today are so specialized, and typically custom-tailored to each artist's needs, that it can be considered unreasonable to ask an entertainment group to leave their own stage monitors out in the truck.

House systems today in many concert venues are better than those encountered a decade ago.

With house sound mixing packages, the same holds true. If a group is carrying a specific mixing console and signal processing package in the truck, it will almost certainly come in through the loading dock door, as the consistency of the show's sound from night to night is dependent on one person and his or her mixing tools that are available. It is only natural to prefer to use the same tool from job to job.

Today, however, with loudspeaker systems becoming more linear in terms of power response, regardless of the manufacturer or system designer, it is entirely possible to formulate a concert mix with signal processing that will sound reasonably consistent from night to night on different speaker systems in different venues.

While I am not trying to suggest that there is no future in stockpiling portable loudspeaker systems, it should be evident to participants in the live sound field that many of today's installed house systems deserve a second look when it comes time to ponder the production requirements for a concert at a particular venue. The time that might be spent unloading and reloading a truck may instead be spent becoming acquainted with a new facility's system and its sound technicians, and doing a bit more patching, equalizing and balancing than usual if one were using only the trucked portable system; such time can be well worth spending.

One area this touring season where the question of whether to *leave it in the truck* has come up quite frequently is the fair circuit. A state or regional fair will typically put a production contract out to bid for stage, sound and lighting systems. The chosen gear will be brought in and set up for the duration of the fair and, from the fair staff's perspective, used in each of the dozen or so major productions staged that season at the site.

Excellent results can be achieved today by combining trucked-in gear, including stage monitor systems, house mix positions and cabling systems and reliable, local loudspeaker/amplifier setups. The touring sound technicians may find it worthwhile to survey the offered gear ahead of time, and even perhaps traveling to that city ahead of the show to observe and test the system prior to the performance if the show is of such a magnitude to support such trips in the production budget.

A recent effort by Paul Simon's entourage, touring the world on the *Graceland Tour,* did just that. The show's regular touring sound system was being trucked around Europe, yet two dates were booked in Zimbabwe, Africa. Time was not available to ship the system between continents, yet the shows were booked and would take place.

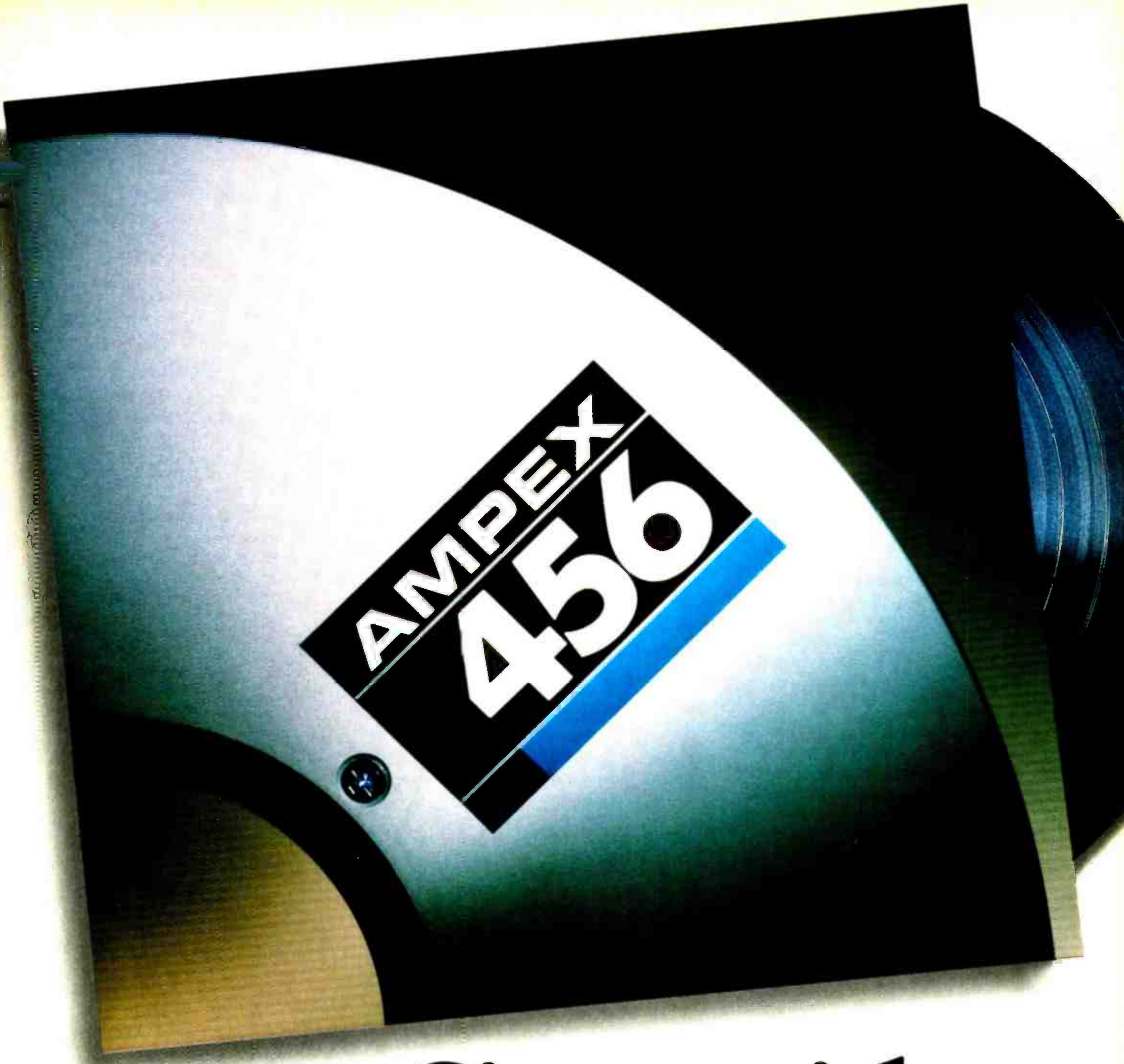
"We had our sound company send down two technicians a week ahead of time," said house sound mixer David Morgan.

"They went in with an oscilloscope and phase checker, and went down to the rental system's shop in Johannesburg, South Africa. Every amplifier, every cable and every speaker component was carefully inspected.

"Even though the speaker system was made up of items over a decade old, including some odd brands of gear, the overall result was quite magical. We had thousands and thousands of people dancing in the aisles outdoors for two nights; the concerts were tape-recorded and they were some of our best shows on the tour. If we'd tried to insist on having our regular speaker system there, that probably wouldn't have happened at all."

The decision of whether to *leave it in the truck* should rest solely with that individual who is responsible to the artist for the sound of the show. Budgetary restraints, labor pressures and scheduling logistics will all raise their head from time to time, but the sound of the show should ultimately be the deciding factor when at all possible. And, in different situations, this may mean leaving it in the truck, or bringing it *all* in.

Every show will be different. And that is one thing that makes live sound an ever-changing challenge. R-E/P



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Film Sound Today

By Larry Blake

The first response my long-suffering editor had to my last column, which I raved about the sound of the eight-year-old *Apocalypse Now*, was: "I thought you were supposed to write about film sound *today*." I agree. This time instead of praising the past, I'd like to take a hard look at two of my present-day pet peeves. First I'll comment on the ways that people relate the financial success of a project (film, TV show, record) to a technical development. And second, I'll comment on the misconceptions about pros and cons of multitracks vs. editing on mag film, especially in regard to editing precision.

I find it hard to understand how anyone can try to draw parallel lines between the almighty dollar and matters such as signal-to-noise ratio or picture resolution. First of all, this thinking contradicts the reason people pay their hard-earned money at a record store or a film box office: to be entertained. Maybe there are people out there who get a kick out of quiet tracks or sharp images, but they constitute a microscopic percentage of the paying public.

I'm convinced that the main reason technical developments are hyped on the basis of their money-making powers is to appease the "men in the suits." You know, the muckety-mucks who sign checks and who see and hear nothing but the bottom line. Perhaps this is an obvious—if not naive—statement, but I think that expanding the boundaries of the craft of film-making is reason enough for technical improvements, be they better microphones or consoles or lenses or recording and editing formats. To my way of thinking, the imprimatur of the box office doesn't legitimize technical matters, it *condescends* to them. Such thinking is not unlike the common backhanded way in which some film critics compliment movies by comparing them to ballet, opera, sculpture or other fine arts. A good film, like any useful doodad, is *sui generis*, and comparisons and justifications are beside the point.

Let's face it, the majority of the extra cost involved in making *great* as opposed to good, sound for records or films cannot and never will be financially justifiable. In the final tally, we go the extra nine yards primarily in the spirit of professionalism and pride in a job well done. There is also the spirit of competition, which I believe is the reason that a large percentage of first-class film soundtracks are edited and mixed by "hungry" independent companies not associated directly with a major studio.

Perhaps a better explanation for using a world class console with an expensive digital multitrack, is that work will be more conducive to neat tricks, more fun, or of higher quality. Pride in one's work is nothing to be ashamed of, and until people start buying records for something other than the song and the performance, or start going to movies for something other than a great story and believable acting, we're going to have to be content in the knowledge that we gave the public our best. Now, on to Gripe Number Two.

I wish I had a nickel for every time I've heard: "Yeah, you think film is precise, I'm accurate to a one-hundredth of a frame with my multitrack and time code synchronizer." Sure, your high-tech synchronizer might be capable of such accuracy, but are you patient enough to take advantage of it? What matters is not the theoretical accuracy of a system, but how *operationally conducive* it is to precision. Where film might be only accurate to a quarter frame, it is *easy* to manipulate and slide small segments. Even after 60 years, sprocketed film remains the medium of choice for precision editing of dialogue and effects. At this point it is probably best to recite the classic benefits of both multitrack and mag film techniques.

Sprocketed film remains the medium of choice for precision editing of dialogue and effects.

Multitracks can also be a big help in foley and ADR recording.

Film sound editing is, first and foremost, a manual craft that allows great flexibility in shifting tracks relative to each other. Because the sound in each edited mag sound "unit" is physically separate from the other tracks running at any given moment, slipping sync is a simple, fast matter. Not only is this a creative tool during the mix, but it also permits sound editing to proceed while picture editing is on-going. This latter ability is a double-edged sword of sound editorial crews, and they frequently spend as much time keeping up with picture changes as they do with cutting the sound in the first place. In this context it should be stated that a major user of multitracks in post-production, standard video sweetening of television shows, is almost always done to the final edit master, which is of a fixed length that will not change.

The closest that the world of multitrack tape can offer is the buzz word "offset." Sure, you can slip a track, copy it over, losing a generation or two in the process (with analog recording), but how easy is this to do? Also, this is an awful lot of trouble and time to go through just to get a door slam in sync.

There is one area that multitrack tape offers a speed advantage in regard to slipping of tracks and that is when *digital* multitracks are used and the picture is recut. One could clone a copy with offsets conforming to the new picture cut by changing groups of 24 or 32 tracks in a fraction of the time that sound editors would require to manually shorten or lengthen dozens of film units.

Multitracks can also be a big help in foley and ADR recording. Which is to say that one can audition a large number of tracks at the same time, something that just isn't feasible with standard 3-track mag recording. However, it is important to note that here I am speaking of multitracks for recording *only;* copying the individual tracks to mag "string off" units allows sync to be fine-tuned in the best film fashion. The "massaging" of string offs is standard practice in theatrical features, while TV shows often dub directly from the original multitrack tapes, uncut string-offs or mixdowns.

The support machinery of mag editing is huge—assistant and apprentice editors, Moviolas, dozens of film units and thousands of feet of "fill." It would be more depressing were it no so flexible. Multitracks *do* offer great potential at low cost and size compared to a room full of 35mm mag dubbers whose insides look like something out of *Dune*.

But when the task at hand is sound editing for film, the neat and easy aspects of multitracks shouldn't blind one to the benefits of mag. Of course, I can't wait for this whole discussion to become moot once random-access digital sound editing becomes an everyday reality. R·E/P

Larry Blake is RE/P's film sound consulting editor.

Living with Technology

By Stephen St. Croix

I have decided to abuse the presses this month with details of some miscellaneous little concepts that have popped up during all the hours spent polishing all the other "Living With Technology" columns that have appeared within the covers of this magazine. They are totally unrelated subjects and, while each one is not really big enough to warrant an entire column, put together they can use up at least five gallons of ink.

Ready? Here are five pointers (deep secrets of mine) on how to be a success. Each of these has been personally tested by me and is guaranteed to work (for me).

For all you lawyers out there, I trust you'll understand that all of these stories, all of these references and all names used (including even my own) are fictional; I don't even really play guitar, I am a certified heli-arc welder. Now that that is clear, I'll continue.

• If you are a session player, a producer or whatever, from time to time you may find yourself in a position where it is difficult to figure out exactly how much to charge a real heavyweight for your services.

Here is how I once solved this dilemma.

I was doing some session work for a very well known Superstar in a Very Good Studio, in the exact magnetic center of Nowhere. I was there without a personal following: no wide-eyed little St. Croix fans. (I guess they missed the plane; the connections were very tight.)

So, I was stuck there with only the people actually working on the album. The lady that I was missing was thousands of miles away. It was not too bad, however: Luckily, the telephone had been invented years before. So had the telephone bill.

I don't even play guitar; I'm a certified heli-arc welder.

It worked out so that I really was only needed from time to time to do my stuff, so the rest of the time I spent talking on the phone. I mean the *rest of the time*—in between playing, during meals, as I tuned up, and as I fell asleep. Needless to say, I managed to generate an impressive total bill.

My work on the album was also impressive and, because I backed it with the guarantee that if the tunes that I worked on did not hit the charts I would return the bucks, I felt that an impressive bill from me was appropriate. After much thought, I took the phone bill, multiplied it by 10 and presented it to the client.

Nobody even asked why the figure was so strange.

The songs hit, the famous rocker paid and we all lived happily ever after, with some of us driving extremely fast cars.

The songs hit, the famous rocker paid and we all lived happily ever after, with some of us driving extremely fast cars.

• When a person reaches a certain level of success (measured, of course, by the most noble of all achievement evaluation references: Money), that person's name will be sold by their credit card company—or some other company with whom they or their bank does intimate private business—to any number of boiler houses.

For those of you who might not know what boiler houses are in this context, they are the places that have some "operator" call your company, ask for you and then ask you to hold while they put someone on that you have never heard of before. If you are stupid enough to wait, then the new guy comes on and, with a voice coated in 30-weight, tells you how important you are because you made so much money last year, and then tells you that you should give it to them for vinyl futures or something equally as cool.

It doesn't take long to figure out what these companies sound like, but it can take quite awhile to figure out how to get them to go away.

If your secretary tells them that you are too busy to talk to them right now, they simply call back again and again (and again). If she says that you won't talk to them, they actually argue and say you have to tell them that yourself, because it is personal business. To hell with them.

Every one of you must have some callers like that, or callers that you don't need for whatever reason, and you may have been searching for the proper way to deal with them. Here it is.

Just have your secretary respond with this scientifically researched response: "I'm sorry, but Mister St. Croix is too cool to speak to you right now". This is so simple, direct and shocking, so deeply foreign and offensive, that it works almost 100% of the time. Please note that it might prove more effective for your use if you substitute your own name for mine. Adams honored.

• For those of you who are players, this is the hot tip. If you can't play better than the competition, at least play louder. It seems that in some circles the two cannot be told apart.

• Always ask for 30-day billing when you buy gear, but pay in 10. This course of action totally confuses everybody. They begin to wonder why you do this, and they eventually come to the conclusion that, whatever the reason, they don't relate, so they become just a little afraid and will do whatever you ask, because they feel that you are too unpredictable to say no to.

• If you write a flaky column for a serious magazine, always come up with little things that happen to force you to turn in the column each month about 22 minutes before the presses roll. This makes it very difficult for the editor to perform massive surgery on your work, and you can get the really weird stuff printed.

I must admit that there is a down side to this, however. If you say something totally stupid, it also prints. And you have to go to the next AES Convention and wonder if the guy is looking at you funnily because he has read your latest offering. R-E/P

Stephen St. Croix, RE/P's technology developments consulting editor, is president of Lightning Studios and Marshall Electronic, Baltimore.

SPARS On-Line

By Nick Colleran

In 1979 SPARS was a new organization and Joe Tarsia, the first president, invited my company Alpha Audio to join the association. We had been in business for nearly 10 years and were prepared to pay $2,000 (the original dues), but only *one* of our four rooms was 24-track. So, we didn't qualify to join the organization.

I wrote a letter; SPARS listened and the rule was changed.

My company joined the association and, within a matter of weeks, we had received back many times the cost of our dues in information of direct benefit to our studio operation and overall company.

Early in the life of the organization, I had a member from Nashville tell me that he wrote off his SPARS dues to advertising. Why advertising? Simple. If you, as a studio owner, are asked to recommend a remote truck or a studio in another city for your client to overdub, what do you do?

Are you more likely to refer them to someone you've met, or throw a dart at the latest studio directory and pray they don't destroy your client's tape? (Leading the client to destroy you at his earliest convenience!)

The first cousin to advertising is, of course, public relations. With all of the trade magazines needing good material to fill editorial space, why is it that some facilities are always being mentioned and others never are? The answer lies partially in the concept of *safe* editorial material.

Established, well-known facilities are safe to write about. If you are hesitant to recommend a facility you have never seen, why should a magazine print what it cannot easily verify? Membership in SPARS is one step to becoming *safe*.

The SPARS Business Conferences often address the subject of public relations from both the owner's and public relations professional's points of view. These meetings can help a studio get it right the first time, concentrating on what has worked for others and not repeating their mistakes.

Through SPARS, we have met people around the country whom we have recommended, as well as received calls from 3,000 miles away to record a group touring in our region.

Among the benefits are the SPARS Business Seminars presented twice a year in New York and Los Angeles. In April, the West Coast Business Conference was presented at UCLA. Industry leaders such as Chris Stone, president of Record Plant, Los Angeles; Murray A. Allen, Universal Recording, Chicago; and Guy Costa, Motown Hitsville Studios, Los Angeles gave their insights to developing business plans for recording and production facilities.

No matter how large your operation, "Big Brother" is bigger when it comes to taxes. Those of us who have clashed with them over the years know that the primary motivation of any tax department is to collect the most money for the least hassle. And the ratio of hassle can be infinite. When is the owner/engineer going to deal with it? Between takes?

Our company is now faced with a potential sales-tax application to studio time and all components of billing for studio services. This application is due to a case where a television station was required by the Supreme Court of Virginia to collect sales tax on its entire production bill. The 1-inch stack of information we recently received from SPARS on sales tax in California and other major production areas is proving useful.

> ***SPARS can imply that a large national resource is available to the single studio owner to fight back.***

Although tax laws vary in different states, we now have the information essential to clearly outline the different forms of studio work and their taxability, and also enable us to make a convincing argument that we do not function like film, video and television facilities.

> ***Membership in SPARS is one step to becoming safe.***

Such an argument could not effectively be made against an apparent conflict with a Supreme Court case. SPARS is helping this studio owner protect his wallet sufficiently to pay his dues for the next few years!

Recording facilities are often told by manufacturers that their problem is unique. Through SPARS you can find other facilities that have the equipment and who maintains it.

In addition to the benefits that SPARS provides for its members, the organization also provides benefits for the pro-audio industry. The best examples are the SPARS Exam cosponsored by Sony Corporation and the Database project funded in part by 3M Company.

The exam, developed in cooperation with the Educational Testing Service of Princeton, NJ, is the first effort to measure the knowledge needed in our business. The Database, when complete, will provide information on our industry, thereby making studio financing easier and market analysis less a matter of guesswork for manufacturers.

While the bulk of this column has addressed the most direct benefits of SPARS as a trade organization on the bottom line, there are many more reasons for membership:

- Is a local bank in South Carolina going to finance studio gear it has little chance of reselling in the local want ads? Is a New York bank going to finance a business that far from home? SPARS' referrals have successfully overcome these obstacles.

At least one SPARS member has used the organization's leverage to secure a refund from a manufacturer. When a piece of gear failed to perform and, worse, caused significant losses, the manufacturer was unresponsive. Things turned around when the studio owner mentioned that he was sure other SPARS members would be "interested in his experience."

If a studio from our neck of the woods can get SPARS to listen, change the rules, and have the studio founder go on to become president of the organization, there is surely a benefit available to any studio that joins and participates, regardless of size, location or market served. The entire organization has always been greater than the sum of its parts, and while many of us have invested considerable time and resources over the years, the benefits have always exceeded the cost.

For more information regarding SPARS membership, write to: SPARS, P.O. Box 11333, Beverly Hills, CA 90213 or call 818-999-0566.

R·E/P

Nick Colleran, president of Alpha Audio, Richmond, VA, is the departing president of the Society of Professional Audio Recording Studios.

3. Neve V Series' Necam 96

The Sandbox has it all. Both MIDI and live recording environments tucked away in a secluded Connecticut retreat. Luxury boarding and day accommodations. An enviable and diverse client roster. And when it came to choosing a console, only Neve's V Series with 48 inputs and Necam 96 would do.

Why Neve? Necam 96, Neve's moving fader automation system, was definitely a deciding factor, says studio owner Spencer Taylor. "The V series had the finest sound and was technologically the most uncompromising console for the complex configurations necessary to set up the MIDI apparatus we envisioned."

Necam 96 is the world's first instinctive moving fader automation system, capable of controlling up to 96 faders. Keyboards, drum machines and other MIDI gear can come into the patch bay, eliminating loose cables. And super touch sensitive servo driven faders give ultimate control.

The Intelligent Rollback feature lets you play back the moves you just made by simply pressing a single key. The tape or film moves back to exactly where you want it, faders move to the right spots and your updates are played for you. The Auto Merge feature will automatically merge changes with the rest of the mix. Easy? You bet.

No need to keep after every pass either. Necam 96 keeps the pass as a Virtual Mix for review or update. But, every so often, it's good to keep the mix for archive purposes.

Plain English instructions presented on a full color, high resolution video display tell you system status quickly, in clear easy-to-read labels. Smart Keys cut down on complexity even more.

But what about the bottom line – sound? "To say it's clean is an understatement. Neve's qualities speak for themselves," says Taylor. "We have been ecstatic with our choice from day one! For the Sandbox, Neve was the best choice. The only choice."

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Circle (15) on Rapid Facts Card

Exposing Equalizer Mythology

By Dennis A. Bohn

Despite their wide use in just about every aspect of audio production, graphic and parametric equalizers are still widely misunderstood.

John Roberts is one of my heroes. Several years ago, John wrote a regular column in **RE/P** titled "Exposing Audio Mythology." The subtitle was, "Laying to rest . . . or at least exposing the false premises upon which they are based . . . some of the pro-audio industry's more obvious 'Old Wives Tales.' "

Great stuff; you could almost hear the theme music and see the masked rider off in the distance.

Originally, John intended to do a few columns on the most flagrant myths. That was in early 1983. He continued until mid-1986.

Every issue, without fail, John waged war on the mythmakers. John is resting now; myth-exposing is maybe too much for one person. But I'd like to help out by exposing some of the more popular myths about equalizers.

• Myth #1: There exists such a thing as a combining filter.

Many engineers are confused about *what* a combining filter is. So am I. Filter designers have many names for different types of filters—Butterworth, Chebyshev, Bessel, etc.—but combining isn't one of them.

The problem here is with the use of the word "filter." We must distinguish between what is being thought and what is being said. Within the context of using this phrase lies the real intent, i.e., how much ripple exists in the output. (The output being a combination of all of the filter's outputs, or combined output.)

The outputs from filter banks combine (or, in reality, recombine) to form a resultant curve characterized by an overall shape and a ripple content with associated phase shift. How this combining takes place, and the bandwidth of the individual filters, will dictate how much ripple exists. The type of filter used has *nothing* to do with it.

Dennis Bohn is vice president of research and development at Rane Corporation, Mountlake Terrace, WA.

Combining is done by electronically summing together all of the filter outputs. It is not a filter at all; it is a means of summing the outputs of individual filters. All equalizers combine their filter outputs. It is wrong to say that an equalizer is non-combining. The *only* examples of non-combining multi-band filters are real-time analyzers and active crossovers.

An example of the misuse of the term "combining filter" concerns the comparison between constant-Q and conventional graphic equalizers. (Conventional, as used here, refers to any graphic equalizer that is not constant-Q.) The popular, albeit false, belief is that conventional equalizers use combining filters, while constant-Q designs use non-combining filters. Both designs sum their outputs together. Examples of both designs exist using one or more summers. The difference lies in the smoothness of the combined curves. The fallacy lies in taking the answer out of context.

Setting a conventional equalizer to exhibit the same bandwidth as a constant-Q design produces a combined result *exactly the same* if the number of summers is the same. However, the only condition where this occurs is either full boost or full cut.

Most users do not understand this is the *only* position where the affected bandwidth is ⅓-octave wide (for ⅓-octave designs). At all other boost/cut settings, the bandwidth degrades to over 1-octave wide.

There is no doubt that if two adjacent filters located a ⅓-octave apart degrade to where each is 1-octave wide, then the summed result will be very smooth. There is also no doubt that this is no longer a ⅓-octave equalizer; it now acts as an *octave* equalizer. If that is what is required, then a conventional equalizer is the correct choice; however, if ⅓-octave control is required, then *only* a constant-Q design will do.

• Myth #2: Minimum Phase behavior is an important criterion when buying an equalizer.

Minimum phase is one of the few things you *don't* have to worry about when buying an equalizer. Not that MP isn't important, because it is. It's just that no known examples of commercial equalizers that are *not* minimum phase even exist. Forget all the marketing hype to the contrary.

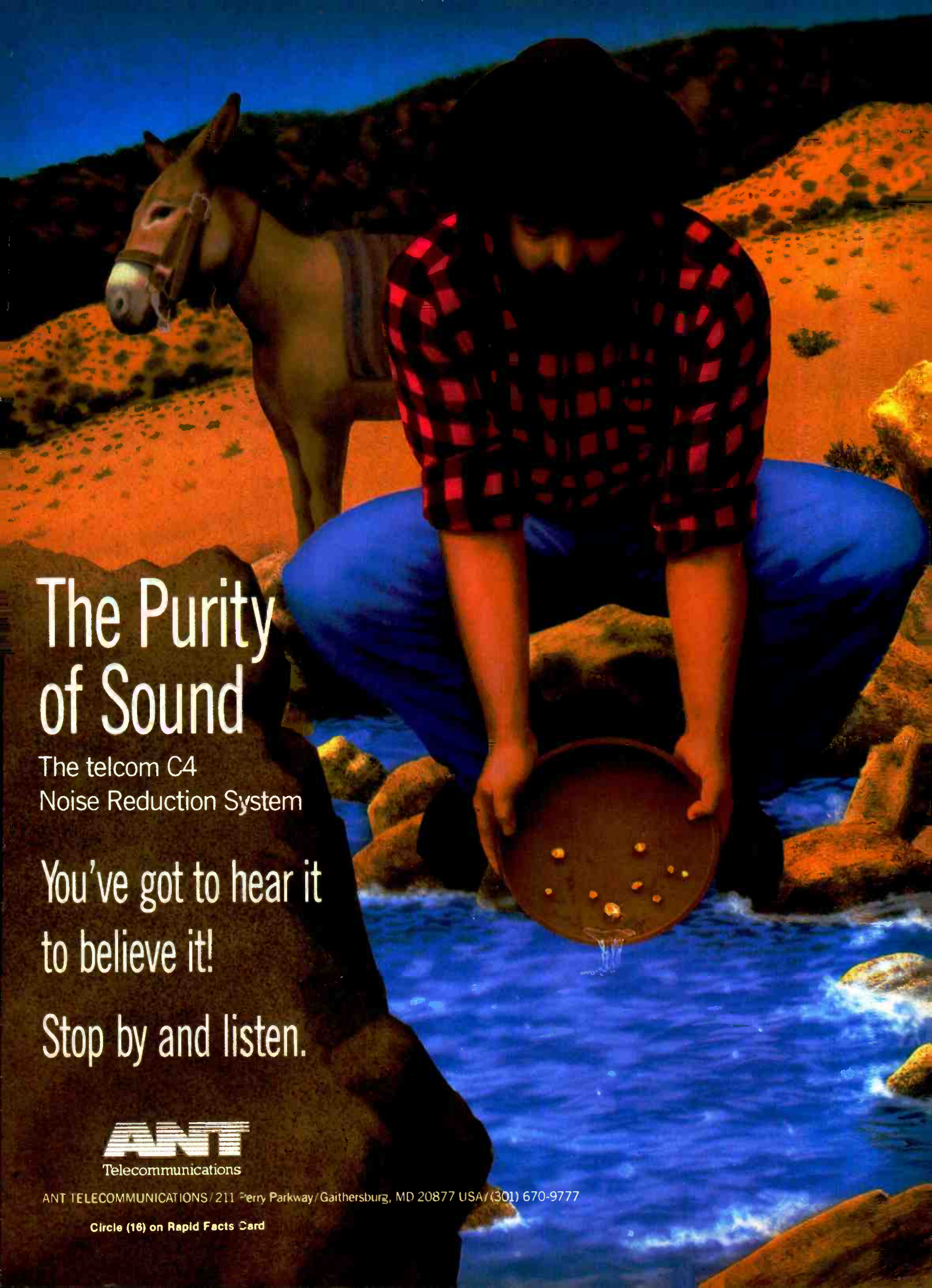
A precise definition of minimum phase is a detailed mathematical concept involving positive real transfer functions, i.e., transfer functions with all zeros restricted to the left half s-plane. If the last sentence produced a zero in the middle of your brain, don't worry. All you need to know is that minimum-phase behavior is not a problem in *any* equalizer you may consider purchasing.

Here again is an example of sloppy rhetoric; a failure to communicate clearly what is being thought. Somewhere years ago some marketing type needed a buzz word for distinguishing his company's equalizer from everybody else's. Some engineer dropped the term minimum phase and the marketing guy went nuts.

That's it, he thought; never mind that it doesn't fit what is trying to be said . . . it sounds good. Nice and high-tech, so use it to try to build a smoke screen between comparable products.

What they *wanted* to say was that their product could create boost/cut curves with *less* phase shift than their competitors, and that this was a good thing. The problem was that here comes the engineer again to say this simply wasn't true. Any two equalizers producing the same curve do so with *exactly* the same phase shift. Same universe, same physics, same results—much to marketing's chagrin.

So they compromised on claiming their product had *Minimum Phase* charcteristics. Never mind that all the competition also had minimum-phase behavior. The cus-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tomer wouldn't know that. The promotion implied that the other products didn't. Let the buying public figure out otherwise.

OK, now you know otherwise. Don't be hoodwinked by this year's buzz word.

• Myth #3: Only one brand of equalizer exhibits complementary phase performance.

Speaking of buzz words, here's a beaut: Complementary Phase Shift. Somebody worked overtime on this campaign. I guess what gets me so angry about this issue is the arrogance of the manufacturer. The underlying premise is that the pro-audio public is so gullible they will believe anything, if presented profoundly. Well, they are wrong. All of you are a whole lot smarter than they give you credit for. Street smarts go a long way in solving problems.

Complementary phase shift means nothing more than the fact that an equalizer displays symmetrical boost/cut curves (and is minimum phase). In other words, the boost curves are mirror images of the cut curves, which means that the phase shift of the boost curves are also mirror images of the cut curves. If two things are mirror images of each other, they are complementary. Nothing too profound here.

Now, it is *not* true that all equalizers exhibit symmetrical boost/cut curves. Therefore, not all equalizers have complementary phase shift. At least two of the more popular brands do not. So, if you perceive this to be an important parameter when buying an equalizer, you are correct in asking whether the unit has symmetrical boost/cut curves; I can give you a list of a dozen manufacturers whose equalizers do.

In truth, every example of graphic equalizer I'm familiar with has symmetrical boost/cut curves, as well as most of the parametric equalizers on the market. In fact, you have to look long and hard to find examples of equalizers that are *not* complementary phase performers. As I said, I know of two but there may be more.

The correct question at this point is why do you care if the equalizer has complementary phase shift? Damned if I know. I can tell you why they say it is important, and I can tell you why they are misleading you.

The popular demonstration involves setting up one channel with an arbitrary curve and then adjusting the other channel for the opposite response. Passing a signal through both channels in series produces a flat frequency response. No phase shift. No time delay.

Now this result seems to have overwhelmed them: they describe the results as bizarre, remarkable and baffling. I can find no one else who is the least bit surprised. This is one of the few places where your intuition is correct.

If you take two equalizers set for complementary curves and put them in series you get a response of *unity*. However, you do *not* get an all-pass response, as they claim. There is no amplitude variation, no phase shift, and no time delay.

Basic sophomore electrical engineering tells us why. Something called a transfer function represents each channel. This mathematical equation completely describes the amplitude, phase and time response of a signal passing through that channel. The complementary channel's transfer function is the reciprocal of the first. Putting them in series causes the two transfer functions to multiply.

If two equalizers do not produce the exact transfer function, then they will definitely sound different.

Anything times the reciprocal of itself produces the answer of unity. Nothing too difficult here. One is *not* the transfer function of an all-pass filter; one is the transfer function of a piece of wire.

So what does all this have to do with what kind of equalizer you may want to buy? Not much, really. The implication is that you must have a complementary phase equalizer to correct for a room's frequency anomalies. Not true. Any equalizer that produces the opposite room response will work, and work just as well.

• Myth #4: Constant-Q means non-symmetrical boost/cut curves.

Until recently, I wouldn't have considered this an official myth. Last year Tab Books published a new book by F. Alton Everest, titled *Successful Sound System Operation*. The book comprises a well-written introduction to the business of sound reinforcement, and I recommend it to anyone just starting out. His treatment of constant-Q equalizers, however, needs some revising.

Mr. Everest states erroneously and unequivocally that constant-Q equalizers characterized themselves by having asymmetrical boost/cut curves. (Which occurred from a misreading of a popular parametric equalizer's data sheet.) This myth involves a mixing of two separate issues.

Reciprocity of boost/cut curves and constant-Q have *nothing* to do with each other. You can find constant-Q symmetrical and non-symmetrical equalizers, just as you can find non-constant-Q symmetrical and non-symmetrical equalizers. The terms characterize two *different* aspects of an equalizer. Constant-Q refers to the bandwidth behavior for different amounts of boost or cut. If the bandwidth stays constant as a function of boost/cut amounts, then it is constant-Q. If it does not, then it is not a constant-Q design.

If the cut curves are mirror images of the boost curves, then the equalizer has symmetrical (or reciprocal, the terms are interchangeable) response. If the curves are not mirror images of each other, then the equalizer is of the non-symmetrical school.

Two separate issues, both available in any combination from several manufacturers. Your choice.

• Myth #5: Given identical equalizers, one passive and one active, the passive unit will sound different.

The key to whether this is a myth involves the crucial word, "identical." If two equalizers do not produce the *exact* transfer function, then they will definitely sound different. That is not the issue here. At issue is whether there exists some sound quality attributable to active or passive circuits per se. There does not.

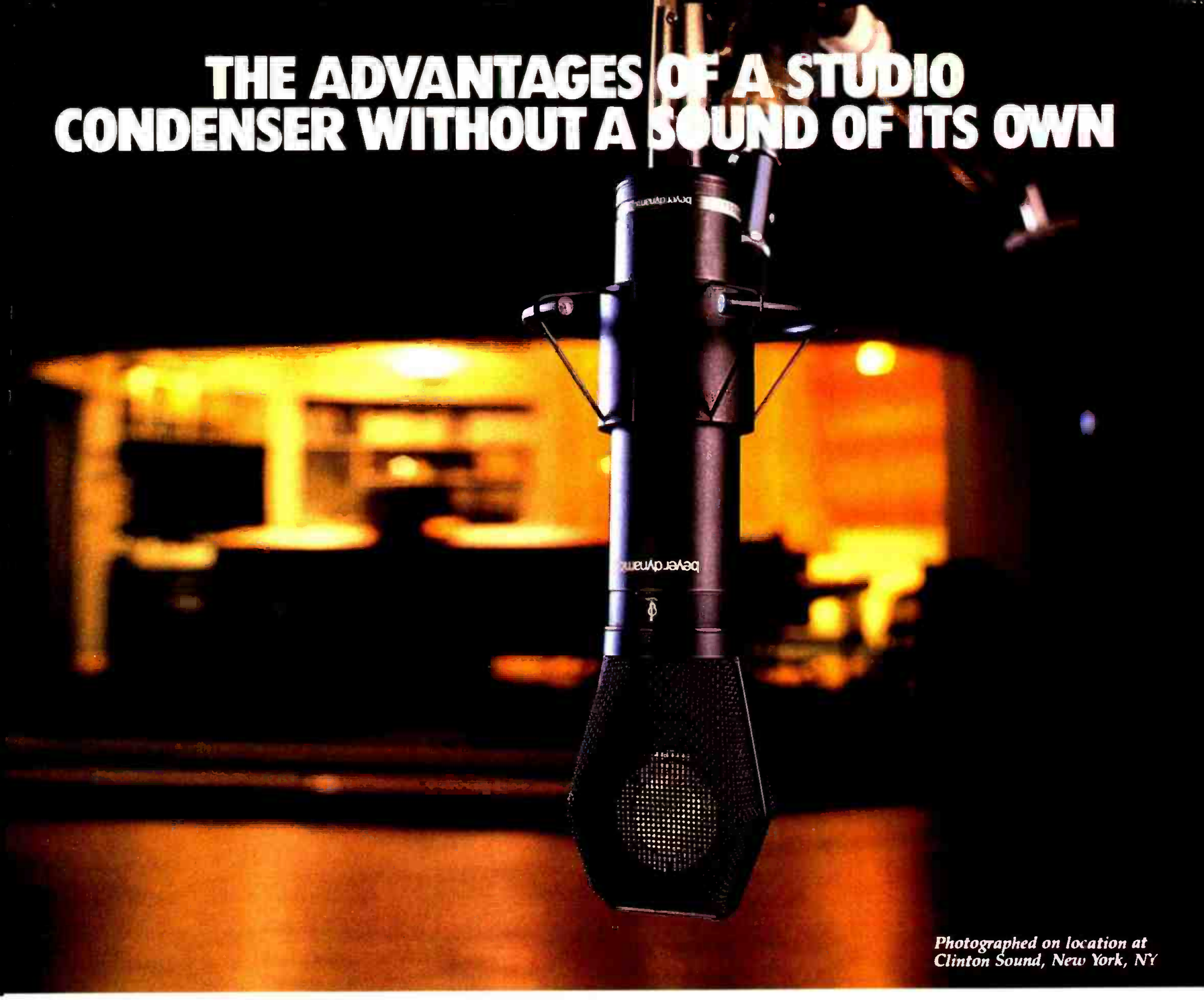
A transfer function exists that characterizes every equalizer's output behavior to a given input change. Any two equalizers with the *same* transfer function, when operating within the constraints necessary to behave according to that function, will give the *same* results no matter what physical form makes up the equalizer.

In general, any equalizer response can be implemented by many different types of circuits, both active and passive. The perceived differences between equalizers designed for the same response function must be explained by factors other than whether the equalizer is active or passive.

Some characteristics that can contribute to the misbehavior of the circuit are nonlinearities that occur because the components are being used improperly or stressed beyond their linear operating region. Sometimes the perceived differences are nothing more than one circuit is quieter than another.

Any two equalizers with the same frequency-domain transfer function will behave the same in the time domain. The transfer function determines responses such as overshoot, ringing and phase shift regardless of implementation.

Nothing mysterious exists within the realm of active and passive equalizers. Simple electronic theory explains all differences between these two, if differences exist. If not, they will perform and sound the same to the objective observer. Never assume that because an equalizer is active or passive it is automatically better or worse for your application. Study your needs and consult with knowledgeable people to make the correct equalizer selection.



Photographed on location at Clinton Sound, New York, NY

For all of its virtues, the typical studio condenser imparts a definite character to any recording. These impositions are often considered inevitable technical imperfections: accepted, ignored or tolerated by audio engineers.

Characteristic anomalies of condenser performance such as exaggerated high end response or distortion have even been rationalized as compensation for the high frequency losses inherent in typical analog formats. Nowadays, however, they are increasingly viewed as unnecessary intrusions in critical analog and digital recording situations.

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Additional comments

*Following the appearance of this article in **Sound & Video Contractor** in October 1986, Robert Orban, chief engineer of Orban Associates provided the following comment. Also appearing is a reply from Dennis Bohn.*

From: Robert Orban, chief engineer, Orban Associates.

Dear Mr. Bohn,

I believe that it is necessary to define our terms regarding Myth #4. Constant-Q means non-symmetrical boost/cut curves.

As far as I know, I was the first to use the term "constant-Q" back around 1976. I described the boot-cut curve family produced by the Orban 621-series equalizers with this term, for which I had a very specific definition. Specifically:

> *"constant-Q" refers to a family of equalization curves characterized by an unchanging "Q" of the s-plane poles of transmissions, in both boost and cut modes.*

This curve family is most easily produced by adding (boost) or subracting (cut) the output of a two-pole bandpass filter with its input. Such summation affects only the s-plane zeros of transmission of the resulting symmetrical bi-quadratic transfer function.

It is readily shown that if the equalizer is reciprocal and constant "Q" boost curves are produced, then the cut curves which are reciprocal to the boost curves must be generated by changing the "Q" of the poles of the resulting biquad. Therefore, by my original strict definition of "constant-Q," "myth" #4 is in fact true.

Your definition of "Q" appears to be based upon the shape of the resulting quadratic curve, rather than upon pole "Q." However, "Q" cannot be defined as the reciprocal of the fractional bandwidth in the case of a biquad, because (among other reasons) "Q" becomes undefined with equalization of less than 3dB. (In this case, the "-3dB" point does not exit.) In fact, I would submit that the term "Q" cannot be meaningfully applied to a biquadratic transfer function in any other way than as the "Q" of the poles.

Mathematically, if the transfer function H(s) of a bi-quadratic function is as follows:

$$H(s) = \frac{as\pm + bs + c}{as\pm + ds + c}$$

Hence, the "Q" of the poles is defined as Q=SQR(ac)/d.

From: Dennis A. Bohn, vice president research and development, Rane Corporation, Mountlake Terrace, WA.

Dear Mr. Orban,

I welcomed your letter and the chance to further clarify our use of the term "constant-Q." Thank you for your thoughtful comments.

I, indeed, acknowledge your first use of the term "constant-Q" in 1976, and welcome it as a succinct delineator for a complex issue. You are also correct in recognizing our using the term in a wider context; however, our definition would include yours, word for word.

In the case of equalizers, the informed customer needs to know whether the resultant curves exhibit constant bandwidth for all slider positions or not. We use the term "constant-Q" equivalently for constant bandwidth. I apologize for not making this clearer. Our feelings are that the end-user doesn't need (or care) to know what happens to the s-plane poles of transmissions. And, yes, there can be no definition of Q if the amplitude response does not change at least 3dB. While true, that is not relevant to the user's understanding of the product's behavior. Loose rhetoric? Perhaps, but clear rhetoric.

As for the mathematical details, I refer you to my recent paper in the Journal of the Audio Engineering Society. I demonstrate that with proper topology, reciprocal curves can be produced without any change to the Q of the bandpass function. Hence, the term "constant-Q" applies to symmetrical curves as well as non-symmetrical curves, if the appropriate circuit configurations are used. The only difference is whether the bandpass function is subtracted from (non-symmetrical), or put into the feedback loop with (symmetrical), the original signal.

Orban's products are examples of constant-Q equalizers that exhibit non-reciprocal curves; Rane's products are examples of constant-equalizers that exhibit reciprocal curves. Applications exist for both types of products.

I hope the foregoing clarifies our use of the term "constant-Q" as meaning constant bandwidth and demonstrates its appropriateness.

• Myth #6: An ideal equalizer would add no phase shift when boosting or cutting.

Phase shift is not a bad word; it is the glue at the heart of what we do, holding everything together. That it has become a maligned term, is unfortunate. Such a belief stands in the way of people really understanding the requirements for room equalization.

The frequency response of most performing rooms looks like a heart attack victim's EKG results. Associated with each change in amplitude is a corresponding change in phase response. Describing them as unbelievably jagged is being conservative. Every time the amplitude changes so does the phase shift. In fact, it can be argued that phase shift is the stuff that causes amplitude changes. Amplitude, phase shift and time delay are all inextricably mixed by the physics of sound; one does not exist without the others.

An equalizer is a tool that allows you to correct for a room's anomalies. It must be capable of reproducing the exact opposite response of the one being corrected, a criterion that requires precise correction at many neighboring points with the associated phase shift to correct for the room's opposing phase shift. It takes phase shift to fix phase shift. Simple as that.

One way people get into trouble when equalizing rooms is using the wrong type of equalizer. If an equalizer is not capable of adding the correct amount of phase shift, it will make equalizing much more difficult than it has to be. The popularity of the many constant-Q designs has come about because of this phenomenon.

Equalizers that produce broad, smooth curves for modest amounts of boost/cut make poor room equalizers, and good tone modifiers; they lack the ability to make amplitude and phase corrections close together. Lacking the ability to make many independent corrections with minimal interference to neighboring bands, restricts their application primarily to giving a shape to an overall response rather than correcting it. Serious correcting requires sharp, constant-Q performance, among many other things.

Only by adding many precise, narrow phase shift and amplitude corrections do you truly start equalizing a system's blurred phase response. You do not do it with gentle, smooth curves that lack the muscle to tame the peakedness of most rooms.

It's just that simple: you must pre-shape the signal in both amplitude and phase. And that requires narrow filters that preserve their bandwidths at all filter positions. R-E/P

The opinions expressed in this article are those of the author and are presented in the interest of stimulating comments from readers. These opinions do not necessarily reflect the opinions of the editor, **RE/P** or Intertec Publishing. Your comments are welcome.

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Part 3

Time Code and Synchronization:

Time Code Myths, Misconceptions and BooBoos

By Larry Blake

Don't be fooled by the following myths, which can make the task of properly using time code extremely difficult.

Sometimes it seems as if everyone is an expert in time code. Undoubtedly, there are many people working within the audio, video and film industries who do indeed know what they are talking about. Nevertheless, there has to be some way of explaining the almost pathological spreading of misinformation about time code and synchronization procedures.

This final installment of a 3-part series is intended to serve as a "spotter's guide" for some of the more common misconceptions in the world of synchronization. If someone attempts to convince you that one of these myths is, in fact, the gospel truth, thank them for their time and slowly but surely walk in the *other* direction. I will also take a look at vertical interval time code (VITC) and applications of user bits.

Time base references

The lexicons of video and film production contain many phrases to describe the "clock" that controls the speed of a particular system. In film, this clock speed is called either the pilot or reference frequency, while video users refer to it as the "vertical field rate," or variations such as "field rate" or "vertical drive."

The essence of these different speed designations is that the given number expresses the relation of the synchronizing system to *one second of time*, thus the use of the term Hertz (which is, of course, cycles per second).

With film, knowing the reference frequency does not tell you everything about the speed and time base during *original* film photography/sound recording. For example, although the line frequency in England is 50Hz, most feature films are shot at 24fps. Many would consider this to be incorrect, assuming that 25 is a more logical multiple of 50 and should be the frame rate. (Of course, EBU time code is 25fps/50Hz, and television films in the United Kingdom are shot at that rate.)

However, it must be understood that picture and sound in professional filmmaking is handled double-system—they are recorded and handled separately until the final release print. The only time that the worlds of sprocketed film and sync-pulsed ¼-inch tape meet prior to a composite print is during mag transfer.

During transfers the only considerations are to flip two switches on the resolver: one set at the frame rate of the film camera, dictating the speed of the mag machine as a function of frames per second; and the other at the reference frequency of the field recorder's crystal, allowing the ¼-inch tape to be resolved at its correct speed.

Thus a 50Hz Nagra tape could be used to make a sprocket-accurate transfer for two cameras that ran at either 24fps or 25fps, the only difference being on the frame rate setting of the mag resolver.

In the case of video recording, the speed number indicates the field rate, or the number of video fields that will be presented in the course of one, real-time, clock-on-the-wall second. Because the image on each video frame is spread out over two fields (the odd lines on the first field, and the even lines on the second field), the frame rate is always half the field rate. Thus in video there is always a direct relationship between the reference frequency and the frame rate.

The 29.97fps frame rate of NTSC video is not the misnomer that it might at first seem to be: While no image recording system (video or film) records *n* frames plus a fraction, after the passage of one real-time second the NTSC video system will not have fully counted 30 frames. Thus, the American National Standards Institute specifications for time and control codes make a distinction between "real time" and "color time," with the latter defined as "the time elapsed during the scanning of 60 fields [or any multiple thereof] in a color television system at a vertical field rate of 59.94 fields per second."

The term "vertical drive" or "v-drive" is often used to express the recording of a signal with a 59.94Hz time base (assuming NTSC color video) onto an audio track as a backup for time code. Technically, this is not the video vertical drive signal itself, but is merely a sinewave generated from it. (Note that people sometimes refer to a field rate sinewave as "60Hz," regardless of whether it is 60 or actually 59.94Hz.)

Presence of this signal on a multitrack tape will provide a backup; it is usually recorded on the track adjacent to time code and which might otherwise be left blank as a guard band.

Case in point: During the dubbing of a film based on a hit album by a well-known rock group, someone accidentally erased the time code track on the multitrack tape containing the master mix. The group's resourceful engineer managed to resurrect and restripe the time code using two mic pre-amps connected in series, aided by an oscilloscope and careful equalization. The presence of field rate sinewave on another track would have made his job easy.

Larry Blake is RE/P's film sound consulting editor.

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Circle (19) on Rapid Facts Card

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Table 1. Relationship of film camera frame rates to common pilot frequencies and time code standards.

Pilot Frequency/ Field Rate	Film Camera Frame Rate: Pilot Sync	Film Camera Frame Rate: TC Sync	Time Code Frame Rate/ Standard
48Hz	—	24fps	24fps/film
50Hz	24/25fps	25fps	25fps/EBU
59.94Hz	23.97/29.97fps	23.97/29.97fps	29.97fps/NTSC color
60Hz	24/30fps	24/30fps	30fps/monochrome

Note: the 23.97/29.97fps frame rates are used primarily when photographing video monitors.

Table 2. Relationship of real time to SMPTE 29.97/30fps drop and non-drop time code.

Time Code Display				
01:00:00:00	01:00:00:00	00:59:56:12	01:00:03:17	01:00:00:00
Real Time	*29.97fps DF*	*29.97fps NDF*	*30fps DF*	*30fps NDF*

When video monitors are to be photographed in a film, it is increasingly common to reference the film cameras to 59.94Hz video sync. This procedure will result in the lack of a video roll bar, so long as the camera shutter is phase-locked to the video frame edge. During the process, a standard U.S. film camera will run 0.1% slower as a result of the difference between the 59.94Hz reference it is receiving and the 60Hz reference of the camera motor. The camera speed will then be 23.97 rather than 24fps.

During photography of these scenes, the ¼-inch recorder must *also* run at a 59.94Hz reference. If a mono Nagra is used, then the production mixer can either receive sync from the video personnel, or simply insert a 59.94Hz crystal into the Nagra's pilot input. Either way, the SK jumper plug *must* be removed lest the Nagra print its standard 60Hz neopilot signal. (With a time code Nagra IV-S TC, the changeover is simple: just switch the internal frame rate setting to 29.97fps.)

Myth No. 1

"The difference between 59.94 and 60 is academic." This is probably the all-time champ, and can be uttered only by someone who has never experienced the subliminal nagging that occurs when picture and sound start to slide slowly out of sync, having begun dead on at the beginning. While the rate of slippage is slow and gradual—you are 1.8 frames out every minute—such differences simply cannot be tolerated in professional post-production practice.

One reason for the omnipresence of this mistake is undoubtedly the frequent participation of recording studio personnel in music-video production. Where proper sync technique is a matter of survival for those in film and video production and post-production, sync pulses and time code are often treated by recording engineers like a bratty younger brother. (Said group often has a similar reaction to monitoring a mix-down of film music to standard film specifications, especially in regard to monitor level and encoding matrix requirements.)

The only answer is that sooner or later they will have to bite the bullet, and either have tracks in sync or out of sync.

> *When making anything from an average rock video to a simple 16mm documentary, proper sync can be obtained by answering a few simple questions:*
> - *What is the time base of the sync pulse on the playback audio master (if any)?*
> - *What will you be shooting on (film or video) and at what frame rate?*
> - *What medium is the project going to be edited and mixed on?*
> - *What medium is the project going to be released on?*

Most of the time, the editing format is also the release format. In any event, the need for distinguishing between the two formats concerns not sync but instead whether the pitch of the original recording will match that of the final release film print or videotape. In this regard, editing and releasing on NTSC video would demand that the time code be recorded in a 59.94Hz time base, while a U.S. theatrical motion picture requires 30fps/60Hz time code.

More important than matching the time base of the audio master to that of the release medium is making sure that the pilot frequency of the playback audio is the *same* as the camera (video or film); this is also true of production recording. You cannot go wrong no matter how convoluted or bizarre the production flow chart.

For example, if you are planning a film release, printing a 60Hz tone on the audio master (which, hopefully, is the original multitrack tapes; see below) will guarantee proper pitch in theaters regardless what happens between photography and release printing. If principal photography is on videotape, then the slow down resulting from resolving to the 59.94Hz video reference will perfectly anticipate and cancel out the speed-up that will occur in tape-to-film transfer.

Myth No. 2

"Drop-frame time code and 59.94Hz are synonymous." In color time, the lower reference frequency (time base, field rate . . .) is "slow," resulting in the number of frames per second being 0.03 less than that in the "ideal" (the word used in the ANSI spec) video system with 30 complete frames per real second.

After one hour, 29.97fps SMPTE, non-drop time code will have counted 108 frames less than a time code generator referenced to 60Hz. Therefore, only when the time code display reads 01:00:03:17 (assuming it started at 00:00:00:00) exactly one hour will have elapsed according to the clock on the wall.

Drop-frame time code was developed to make the time code display in NTSC video match the clock on the wall. As a result, the 108 frames that we never got a chance to count (by the time the real-time clock reached the 1-hour mark), we simply *won't* count. Instead, frame numbers will be dropped at various points during the hour ("the first two frame numbers [0,1] at the start of each minute, except minutes 0, 10, 20, 30, 40 and 50"), thus getting the time code display to that hour mark faster *without changing the speed of the tape.*

It becomes apparent that the only reason this mutant species of time code called drop frame exists is to allow machines to automatically calculate the length of NTSC color programs. *Humans* can figure out the length of non-drop NTSC programs simply by reducing the final time code number by 0.1%.

For example, if the time code out point on an EDL is at 00:30:00:00, then multiply 30 (minutes) by 60 (seconds per minute) by 30 (frames per second). From this figure, you then subtract 0.1% (54 frames) to get the actual clock-on-the-wall program length of 00:29:58:06. (This is, of course, what the time code display would have read had you used drop-frame code in the first place.)

Why use drop-frame time code? While easy count of program length is, understandably, of great concern to broadcasters, an accurate clock match is of little or no use to anyone using time code as a tool to interlock audio, video and film transports.

In short, unless you are contractually bound to deliver drop-frame code, and someone insists, just go with non-drop and make life easier on your editor (the person, not the machine), who will always know that the next frame will have the next frame number.

It should be noted that EBU time code used in Europe has none of these problems; the same 50Hz reference frequency applies

both to film running at 25fps and to videotape. No conversions or counting cartwheels are ever necessary.

Myth No. 3

"You don't need a sync pulse on the multitrack master." The following scenario has been played out dozens of times: A musical is shot to playback, and editing proceeds using the scratch mix on the ¼-inch playback tape for all transfers. Everything is in sync throughout editing.

It eventually comes time to do music premixes on a dubbing stage and, lo and behold, the multitrack master doesn't match the editor's workprint.

While there are any number of specific reasons why this has happened, all of them relate to one fundamental mistake: the chain of sync was broken somewhere between the original recording and picture editing. Just having a sync pulse or time code on the multitrack master is not enough. This reference must be *copied* onto the ¼-inch playback tapes at the *same* time as the audio.

The preferred technique is to resolve the multitrack tape to an external reference that matches the pilot frequency (59.94 or 60) of the sync pulse or time code on the multitrack. (Remember to always reshape time code during transfers.)

Resolving the multitrack master is absolutely necessary when time code is involved, although if the multitrack contains just a sinewave sync pulse, then a "just press play" transfer should work out. You should get into the habit of resolving the master during *all* transfers, much as it is always prudent to regard a gun as being loaded.

In discussing the 59.94/60Hz issue earlier, an important question was raised: What is the time base of the sync pulse on the audio master? In the best of all worlds this is not an issue: *You* put it on the multitrack tapes and, therefore, can be certain of what your reference frequency is.

War stories in the Tri-state area: Interlocking film, audio and video Case story #3

Orchestral music was recorded for eventual playback during a theatrical film shoot. Some of the prerecords were done in England with a 50Hz sync pulse, and some in Los Angeles with a 60Hz reference on track #24. The original 2-inch tapes were edited at this point and temp mixes were made to 35mm, resolving both the 50Hz and the 60Hz references to 24fps. The ¼-inch field playback copies were then made from the mag temp mixes.

After shooting and editing, the filmmakers wanted to pre-mix the edited 24-track tapes onto 6-track 35mm in the dubbing theater. Because the 24-track tapes only had a sinewave sync pulse, and not time code, the latter had to be striped onto an open track. To accomplish this, the edited multitrack master was resolved so that its original record speed would be duplicated. Simultaneously, the synchronizer was recording 30fps/60Hz NDF time code for use in interlocking to the film chain.

A few reels were striped without a problem; once the appropriate time code offset had been found, the original 24-track tapes, the picture editor's worktrack on 35mm and the picture itself were all in sync. Then, on one scene, there was a serious wow problem when they cut to a close-up of someone playing the piano.

The multitrack tape was inspected and, indeed, there was an edit at that point. However, none of the other many edits had caused a problem.

What had happened? *Answer: There was no sync pulse on the "incoming" tape after the edit. (As it turned out, this was a pickup recording done after the original England/Los Angeles sessions, and a few subsequent scenes also had no sync reference.) Because they wanted to stripe the 24-track tape with continuous time code, a method had to be found to allow resolving to take place even during non-synchronous sections.*

The solution that they arrived at was to put the original 24-track 50Hz sync pulse on the X-axis of an oscilloscope and the 50Hz crystal output from a Nagra on the Y-axis. Using an oscillator set to 9.6kHz, the capstan servo of the 24-track was varied to lock the two 50Hz references by keeping a 45° Lissajous pattern on the scope.

During rehearsals, the changes in the oscillator settings throughout the reel were noted. Finally, in one pass the 50Hz Nagra crystal output was kept in-phase with the original sync pulse, thus striping the length of the tape with a new, continuous sync pulse. Then, it was a simple matter to resolve the 24-track and continue with the time code recording.

Moral of the story: A sync pulse generator costs about $10 a day to rent. Need we say more?

The worst that can happen—if the sync pulse is 59.94 and you think it's 60 or vice-versa—is that the final version (film or video) might be slightly off-pitch from the original. (Again, this assumes that the playback deck runs at the same pilot frequency as the camera.)

To reiterate: Where people run into trouble is not which sync pulse is on the master—having one is good enough—but failure to copy that pulse when copying audio.

If you are concerned about keeping correct performance pitch on the final version, then it is a simple matter to match the time base of the playback audio master to that of the preferred release medium. "Preferred" means that you have to decide whether you want correct pitch in movie theaters or on home video cassettes; you cannot have it both ways. In this regard, if your primary release will be on television, then make sure your master has time code recorded in a 59.94Hz time base. A theatrical film will require a 60Hz reference frequency, with 30fps NDF time code. (Film 24fps code was discussed in part two of this series, published in the September issue of RE/P.)

It should be noted that during these discussions, it has been assumed that the choice of sync as limited to standard 60Hz or NTSC 59.94Hz, where the only difference is the reference frequency and thus the "speed" of the time code. The drop-frame issue is really a non-issue because, as must be constantly emphasized, you can have DF or NDF time code in either 59.94Hz NTSC color *or* 60Hz monochrome systems. Otherwise the two share the same 30fps structure, the only difference being how one second is defined.

Myth No. 4

"You really don't need a sync generator." In the course of researching the material for this article, I asked many industry professionals the following question: What is the most common mistake that you run across? Close behind "Confusion about the proper frame rate" and "Breaking the sync chain" was "Failure to use a sync generator." This complaint has many different meanings.

A common problem is the striping of code onto videotape without locking both the time code generator and the VCR to the same NTSC sync source. (See *War Story #4* on page 36.) The ANSI specification has the start of each time code address at line 5 in field 1, plus or minus one line.

Locking the VCR and the generator to the same master will assure that the sync word will come up at the same place relative to the video frames. The sync word is comprised of bits 64–79, and marks the end of each frame in a set pattern of ones and zeros that cannot be duplicated at any other point in the 80-bit word. In addition, it indi-

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cates forward or reverse motion.

An expensive NTSC composite sync generator plus a distribution amp is not necessary for many simple synchronization tasks. In the example cited above, the composite video signal itself can be looped through the time code generator before being fed into the VCR's video input.

A first cousin of the no-sync-generator problem is the failure to resolve audio tape recorders during transfers. Resolving—locking a sync signal on a tape to a stable reference, resulting in playback at original recording speed—is a simple matter with VCRs and VTRs, because all professional video machines contain external sync inputs. The same can be said of professional digital recorders in the DASH, PD, 1610/1630, etc., formats. Plugging a feed from the house-sync DA into the BNC sync inputs on digital and video machines will lock them together.

Analog ATRs are a different matter, there being no pin-out standardization in the industry for control connectors that access the signals needed to servo-control the motor. (Indeed, one of the many reasons for the Nagra's popularity in film sound is the ease with which it can resolve.)

Apropos the problem with asynchronous time code on videotape, if an analog ATR is not resolved during transfer to a Nagra IV-S TC, the sync word will probably "slide" and thus won't come up in the same place every real- or color-time frame. The processing inside the IV-S TC that takes place in order to make the time code coincide with the audio will then become confused, resulting in ambiguous frames.

War stories in the Tri-state area: Interlocking film, audio and video Case story #4

A feature motion picture was shot on NTSC ¾-inch videotape, which was then "bumped up" to 1-inch tape during the on-line edit.

At this point, ¾-inch copies were made from the 1-inch master for use in sound-effects assembly. In addition, production sound from the original ¾-inch tapes was checkerboarded across approximately five tracks of a 24-track tape, using as a guide the offsets indicated on the final EDL.

Sound was also prepared on 35mm and 16mm mag film, all of which would be interlocked at the final mix on a film dubbing stage.

Concurrent with this sound preparation was the transfer of the 1-inch master to 35mm film and a print made for use in dubbing.

Everything seemed fine on the first reel until 200 feet into the dub (a little more than two minutes), when some dialogue seemed out of sync. The director said that they had cheated the line from another take, so it was possible that sync might be a little off. Then, at around the 500-foot mark, another tight-sync scene was loose.

It turned out that sync was rubbery on all reels, with an 8- or 9-frame sync error over the course of 10 minutes. The error was generally in the same direction, although sometimes it varied.

Almost immediately this ruled out any classic mistake regarding the 59.94/60 difference between video and film, because such an error would have added in one direction only and at a much faster rate.

A week of phone calls and questions ensued, during which time the post-production crew learned "more than you ever wanted to know about time code."

What went wrong? *Answer: During time code striping of the ¾-inch tape from the 1-inch edit master, prior to sweetening and dialogue assembly, the time code generator was not locked to the composite sync generator that was being fed to the video machines.*

The facility thought they had *been locked; close inspection revealed a broken sync cable. The time code generator had reverted to either line or its internal crystal upon loss of an external reference.*

As a result, the time code was asynchronous and was not locked to the video signal. This was verified on a dual-trace scope, where the time code sync word could be seen slipping and sliding away from the video vertical interval.

Thus, although the correct time code offsets had been applied when laying the sound from the ¾-inch tape onto the 24-track, the time code was moving relative to the picture. When the 24-track was locked against a 35mm print that had been made from the "stable" 1-inch master, there was no way to keep in sync.

To get everything back in step during the final mix (too much sound preparation work had been done to consider starting again), the mixers were constantly calculating the correct offsets. If there were long stretches of dialogue, the offsets would sometimes have to be punched in between words and sentences.

Moral: Always lock time code generators to the same composite sync source that is being fed to the video machines.

Vertical interval time code

This series of articles has, for the most part, used the phrase "time code" in the generic sense, meaning to include all varieties: 24-frame film, 25-frame EBU, in additin to standard SMPTE 29.97fps drop and non-drop and 30fps non-drop. (It should be noted that while it is not covered as part of any time code standad, 30fps drop-frame time code is frequently used when shooting film for videotape editing and release that will be done in the drop-frame mode.)

When we talk about audiotape recorders and time code, whatever the flavor, we are actually referring to longitudinal time code (LTC). This is the same time code recorded on videotape recorders, either on a standard audio track or on a dedicated channel such as the address track on ¾-inch U-matic cassettes. (Note that address-track LTC must be recorded at the same time as the video, and cannot be subsequently altered or "jam synced." Also, be warned that the address track location of JVC and Sony ¾-inch decks are different, with the JVC heads being several frames later than those on a Sony.)

There is another type of time code called vertical interval time code (VITC, pronounced "vit-see"), which allows time addresses to be written as part of the video signal, and not recorded continuously onto an audio track. Another major difference is that VITC records 90 bits per frame, as opposed to the 80 in standard SMPTE/EBU longitudinal time code.

Primary among the benefits of VITC is that it identifies (on bit 35) the video field, whereas this is impossible with LTC, which is "unrelated" to the video signal and is unable to distinguish between fields. (The 80-bit word of LTC is spread out over one frame.) VITC's field identification feature has potential application in translating the 30fps edit decisions lists of 24fps material originally photographed and to be released on film.

User bits

The address provided by SMPTE/EBU time code takes up only 26 of the 80 bits in the word covering each frame. Among the remaining bits are eight groups of 4-bit words that are left open for additional information anyone might want to add. In standard form, these 32 user bits result in eight hexadecimal (values 0 thru 9, A-F) characters that can be set to note, in a limited fashion, scene/take, date, roll number, etc.

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Distributed in Canada by evolution technology user bits is the placement of a second set of time code numbers or film edge numbers. Thus the location of original source material, be it film or a playback tape, can be noted by switching the time code reader to display the current user bits status.

A time code generator must be able to encode the user bits during original recording. For example, the internal user-bit settings in the Nagra IV-S TC do not include the ability to internally generate continuously counting, parallel time code. Only static information can be recorded into the user bits.

There is currently much attention being paid to pushing the 8-digit hexadecimal boundary by multiplexing the data. The use of time code on film will present the need to cram complete production information into the 80-bit word. Eight-digit hexadecimal notation is unable to deal with, for example, scene R329AB (reshoot of scene 329, camera setup AB), take 82, camera roll number, sound roll number, production number, camera number, etc.

While no official standards or working procedures have been established, anyone interested should refer to the proposed SMPTE recommended practice 135 published in the August 1985 issue of the *SMPTE Journal.*

Golden Rules for Keeping the Sync Chain Unbroken:

- *Always have a time base reference—field rate sinewave or time code—on your multitrack master.*
- *Always resolve the multitrack master when making transfers.*
- *When mixing/transferring to mag film machines and video decks, lock them to the same reference being fed to the multitrack's resolver.*
- *When mixing/transferring to analog tape recorders, transfer the sync reference—time code or sync pulse—at the same time as the audio.*
- *When striping time code from a synchronous master containing a sinewave sync pulse, drive the time code generator from the reference being fed to the master's resolver.*

The bottom line

Just as you can be an excellent driver, even if you don't know how to fix a carburetor, proper use of time code has nothing to do with one's technical abilities. Synchronization fundamentals rely much more on common sense than they do on the knowledge that the 11th bit in an 80-bit time code word indicates the presence of drop-frame code.

The value of clear lines of communication cannot be overemphasized: Provide detailed logs of your work and demand them of others. Be your own best devil's advocate and assume that what you assume is in fact wrong. (Translation: Never assume anything.)

Should you deplete your knowledge of synchronization fundamentals before you understand what you are doing, ask someone who knows where the time code bones are buried. Most truly knowledgeable people are secure in their abilities, and are neither possessive about what they know nor do they look down on anyone asking basic questions.

Which is the same as saying that they have probably made the mistake that you were about to make before you asked.

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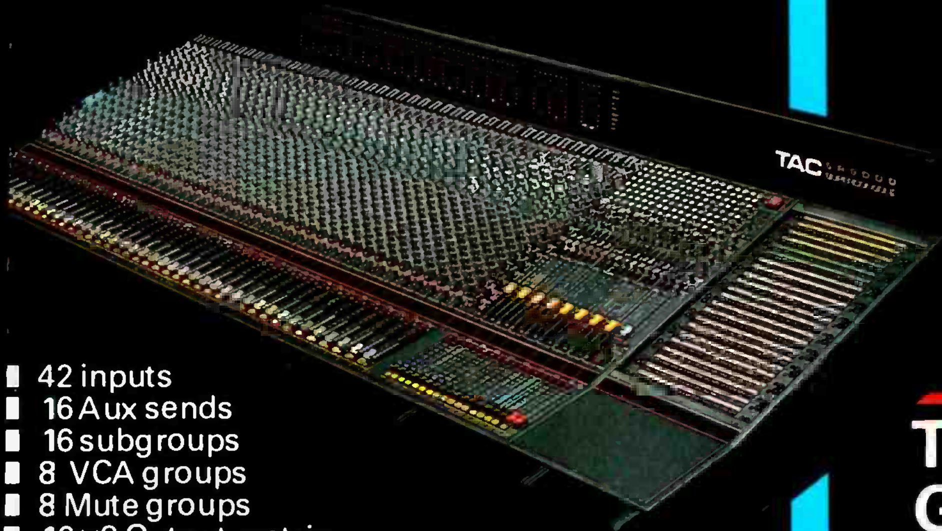

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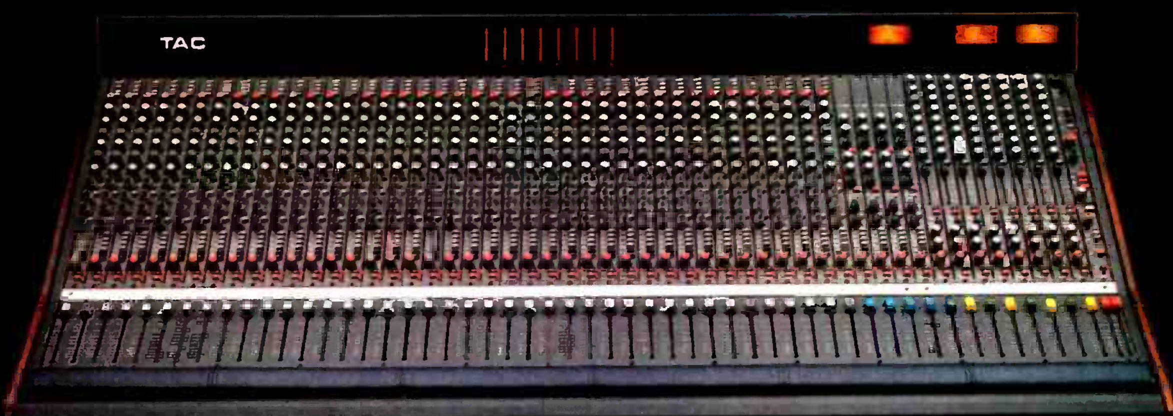

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Regional Sound System Design:

A Profile of Baus Engineering and Vista Sound

By David Scheirman

Two regional sound companies operating on the Pacific Rim detail their efforts to fine tune and enhance the art of live-sound engineering.

In his recently published book *Pacific Shift*, futurist William Irwin Thompson has stated that the virtual center of the global economy is moving gradually westward. His opinion is that, by 2000, the most rapid economic growth will be taking place in the Pacific Rim, those geographical regions that are located in or border the Pacific Ocean.

How does such a theory affect the concert sound business? Recent work-related travel to such areas as Hawaii, Singapore, Japan and along the West Coast of the United States has shown that there is indeed renewed activity in the live-sound system industry throughout the Pacific. While this positive growth is certainly not confined to the Pacific region (Europe and the American East Coast, for example, are busier than ever before), it's useful to examine some of the trends emerging on a regional level.

There does appear to be true growth in the large-scale PA market around the Pacific. For example, Clair Bros. Audio recently sold off a full S-4 system and has shipped it to Hawaii. Tasco has sent a large hanging sound system with Turbosound TMS-3 enclosures to a base of operations in Australia. Japanese rental companies are now able to offer high-quality concert systems with hardware that is a match to products available anywhere in the world. Along the U.S. and Canadian West Coasts, sound reinforcement companies of all sizes report that most available equipment is booked, and that new systems are under construction.

It is on the regional sound-system level, however, that the long-term pulse of a geographical area can be sensed. Companies that design, build and work on a daily basis with only one or two systems must make careful decisions regarding both equipment purchases and system operating techniques.

In Hawaii, there is an obvious increase in the awareness of quality sound for public events of all types, from symphonic performances to rock festivals. As the region strengthens its abilities to serve the growing convention and industrial trade, the need for high-quality systems increases. Baus Engineering, Inc., based in Honolulu, has been fine-tuning a system to serve those markets for several years now.

In Washington, over the past few years there has been a dramatic increase in the number of community events that feature live music. Fairs, festivals and other family entertainment functions are growing in both quantity and frequency. When coupled with the ever-present need in the Pacific Northwest for high-quality touring rental systems to serve the rock-concert industry, a pattern of new opportunity emerges. Vista Sound, of Mukilteo, WA, has been assembling a sound system in recent years to serve this purpose.

Both companies concentrate primarily on fielding a *single* rental system that has been carefully assembled after a close examination of different, commercially available hardware, which is then coupled with custom-built devices and packaging hardware to ensure ease of set-up and trouble-free operation. Let's examine the companies' respective philosophy in detail.

Baus Engineering

Based in Honolulu, many would consider Baus Engineering to be a unique sound system rental and consulting company. Owner Randy Bauske organized the company approximately eight years ago to implement his audio design ideas. Beginning with a small, custom-built loudspeaker system, Bauske considers that he has begun to achieve recognition for his high-quality work in venues from Guam to California.

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

Figure 1. A Mark Levinson ML-3 power amplifier used by Baus Engineering with Phase Linear amplifier for size comparison.

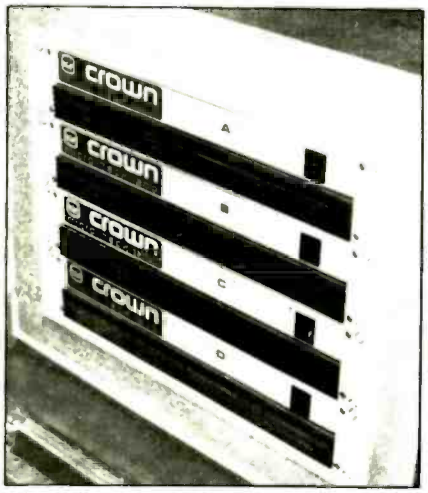


Figure 2. Four Crown Microtech power amplifiers are loaded in a protective, foam-lined road case. Baus Engineering often ships such cases by air freight between islands.

Figure 3. On-stage monitor mix position for a Windham Hill concert at Waikiki Shell, Honolulu.

Figure 4. Front view of Waikiki Shell. Note the Meyer UPA speakers suspended from the lighting truss.

"There has been a lot of recent growth in Hawaii, which brings about a need for quality sound systems," he notes. "We are the primary sound system rental contractor for the Honolulu Symphony Orchestra; we do stadium and arena shows with everyone from Al Jarreau to ZZ Top. And a large portion of our sound system rental projects involves convention work."

As business increased, Baus Engineering made a commitment to purchasing premanufactured, modular loudspeaker systems.

"We eventually chose system components from Meyer Sound Laboratories," Bauske says. "After turning around our own custom-built speaker enclosure inventory, I wanted something that was easy to use, that gave us very little grief in terms of being able to order it, take it out of the box and start using it. The Meyer system gave me that ability."

The company uses both Meyer MSL-3 enclosures and the smaller UPA-1A cabinets.

"For their size, the UPAs are ideal," Bauske says. "They sound great and can be placed practically anywhere with a minimum effort."

Meyer UM-1 stage monitors are also used and, for low-frequency reinforcement, USW-1 subwoofers are used. Bauske chose the latter for their compact size, because much of his company's work in the inter-island region requires shipment by air cargo.

"The USW-1s are also easier to hide when the visual aspect of a show's setup is important, particularly in smaller venues," he says. "The 650-R2 develops more bass, because it is a larger enclosure.

"However, we do a lot of convention work, and the larger boxes can have more of a 'heavier' sound that does not work as well as the USW-1. We keep enough of them so that we can do the large venues easily, as well—for ZZ Top, the system developed massive amounts of bass. Everyone was very pleased."

Baus Engineering's ability to develop "massive bass" stems, in part, from the company's use of high-power, high fidelity amplification. The USW-1s are driven by Mark Levinson with ML-3 units. Weighing in at 145lbs. each, these amplifiers are usually packaged one to a road case by Baus Engineering for small system portability (Figure 1).

"The engineers at Mark Levinson think that we are pretty *radical*," Bauske confides. "I don't think they really understand the type of applications for which we're using their products!"

The ML-3 features two, independent, high-current power supplies, each containing a custom-made 1.2KVA toroidal power transformer. The device is biased to operate in the Class AB_2 mode. Their high price (approximately $6,000 each) probably puts the ML-3 out of reach for most concert-sound companies, but Bauske feels the investment in quality is worth it.

"The Levinson people have done some pretty interesting tests with these things," he states. "They have actually started an automobile with the output of an ML-3."

Rated at 200W per channel, continuous sinewave power at 8Ω, the ML-3 will typically deliver 800W into a 2Ω load. Baus Engineering also uses the Mark Levinson ML-9, which is rated at 643W per channel into a 2Ω load.

"The circuit design is pretty good; I've never had a single failure, and these amps have done approximately 200 shows by now, including flying over to places like Guam for concerts," Bauske says. "They hold up well."

For driving mid- and high-frequency components, Baus Engineering keeps a stock of both Crown Microtech and Yamaha M-80 stereo power amplifiers. All units are well-protected in custom built racks with heavy duty connectors and switching panels (Figure 2).

Baus Engineering's primary sound reinforcement system includes 16 Meyer MSL-3s and 16 USW-1 subwoofers. The MSL-3s are rigged to hang, with eight boxes flying from each aluminum hanging plate. Easy-to-wire "spider" cables distribute the signal to the enclosures from the amplifier racks. Socapex 19-pin, gold-plated connectors are used, with a single cable run going from each amp rack to the loudspeaker positions.

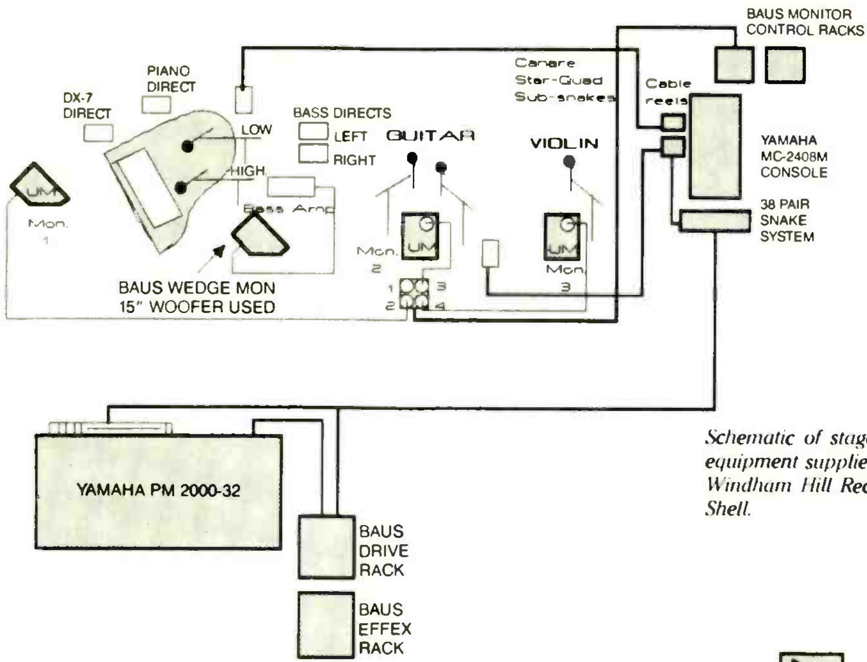
"We borrowed these connectors from the lighting industry," Bauske says. "They are very rugged, and this system is easily wired up by stage hands."

The Mark Levinson power amplifiers draw plenty of current: as much as 20 amps per unit under full load. Bauske designed and constructed a custom-built electrical power distribution system to handle the sound system's needs. A total of 150A is available per leg on the three-phase distro, with massive Crouse-Hinds CF-91 connectors being used for the main feeder cable. Three satellite panels, each handling 100A, can be placed up to 100 feet out from the primary electrical panel.

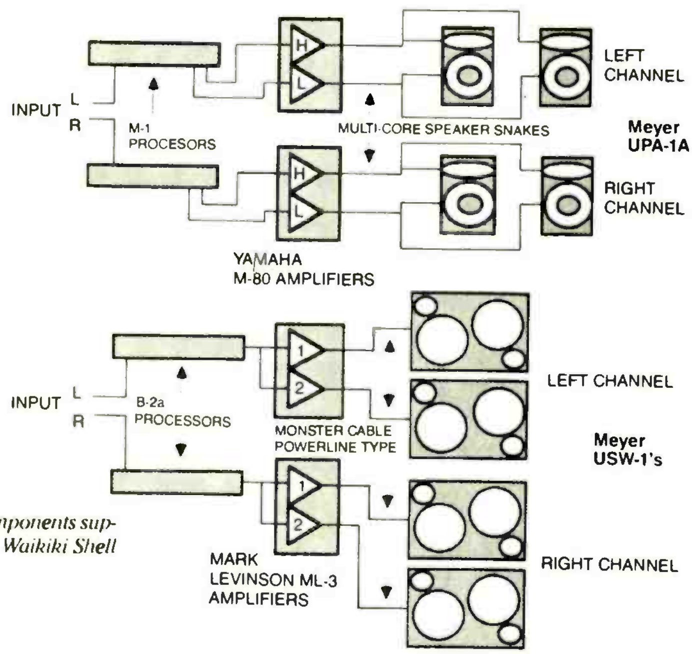
The typical main system drive components used by Baus Engineering for a sound rental date include a Yamaha PM-2000 mixing console, with Yamaha M1516s being available for submixers. The main stereo outputs of the PM-2000 are connected to SAE model 2700 stereo half-octave graphic equalizers.

"The SAE half-octave is what I would call a 'minimum-phase' device," Bauske says. "EQ adjustments can be made and the signal path will still sound musical."

A Yamaha C-2A stereo pre-amplifier is used as a gain device and line driver; it is usually set at +10dB. A dbx model 165A compressor is inserted on the main pro-



Schematic of stage and front-of-house mixing equipment supplied by Baus Engineering for a Windham Hill Records concert at the Waikiki Shell.



Block diagram of sound system components supplied by Baus Engineering for the Waikiki Shell concert dates.

gram as needed, with additional dbx model 160 units on hand for individual channel insertions.

"I've noticed that many companies seem to use the 160s to drive the crossover outputs; they attempt to get their line driver function there, as well as go for bandpass protection," Bauske says. "I don't think that is the way to go on the main outputs if you are *really* concerned about audio quality.

"I go for keeping the original signal as *clean* as possible, and then the Meyer processors handle my component protection functions."

For on-stage monitor mixing, Baus Engineering offers either the Yamaha model 2408 console for simpler events, or the new 32-input/20-output Meyer/ATL monitor mix console with transformerless, balanced input channels and an electronic output assignment system. SAE graphic equalizers and Yamaha power amplifiers are available for each mix (Figure 3).

The system in use

While Randy Bauske noted that it would be difficult to describe a "typical" sound rental date for Baus Engineering—mainly because the workload varies considerably, depending on the event and client—his unique approach to sound reinforcement at the Waikiki Shell deserves mention. The Shell serves a potential audience of 9,000, and its high, arched roof presents a natural projection characteristic for any sound on stage into the audience area. For this reason, on-stage levels must be relatively low to insure a successful event.

"When we do the symphony here," he continues, "we use our temporary loudspeaker system to enhance the natural sound that the shell is projecting. Trying to fight it or overpower it is a poor judgment, particularly in light of the civic-enforced sound level limitations."

At a recent concert date for Windham Hill Records, featuring the band Montreux, Bauske set up a pair of Meyer UPA speakers on the left and right ends of the downstage lighting truss; subwoofers were placed on the ground in front of the stage (Figure 4). For events requiring more level or coverage, Bauske employs the larger MLS-3s; he will often place additional UPAs atop poles in the audience area, feeding them a delayed signal.

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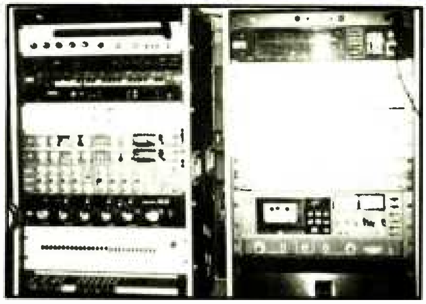

Figure 5. Vista Sound's house electronics racks.

"To sum it up in a single word, I guess I'd say quality," Bauske says. "You need a high-quality input signal, high-quality electronic components, and a quality speaker system. When you combine quality gear with competent technicians and correct operating procedures, there is *no* reason not to have events of all types that sound excellent."

Vista Sound

Vista Sound was set up in 1981 by owner Neale Frazier to provide high-quality sound for entertainers in lounges, fairs, concerts and festivals. Starting with a band-owned, semi-professional system, Frazier has gradually expanded the company's operations to provide full sound production services for venues of up to 10,000 people.

"My philosophy has always been to supply for even the smallest shows the *same* type of quality production that you would find at a major concert," he says. "In 1982 we purchased some speaker system components from McCauley Sound, a local supplier here in Washington state. We have built the system up gradually, and we have tried to build quality in at each step."

After many successful regional tours with artists such as Tower of Power, John Hammond, the Fabulous Thunderbirds and the Robert Cray Band, Vista Sound began to eye the national touring market. In 1984, the company purchased a Kenworth diesel tractor with a custom-built Freuhauf 45-foot trailer.

"Up here in the Pacific Northwest, there really is not a major trucking company that is dedicated to the entertainment business," Frazier says. "Every sound or lighting company has to have instant access to reliable trucking to get the gear around to major concert venues."

Vista's overall sound-system package was carefully designed around the truck's packing dimensions.

"We've gone to great lengths to put this package together as a *system*, so that it goes together as quickly as possible and travels well," he says. "Truck space is something that *can't* be wasted."

Vista Sound employee Fred Micera carefully examined the company's available hardware, and looked at commercially available components, before embarking on an expansion program in 1985.

"I had a big picture in mind; I had been working on ideas for a full system package concept for several years and wanted a chance to develop it," Micera recalls.

"We were particularly interested in the idea of providing equal service to the various opening acts that we encounter; they often get the short end of the stick.

"In the past, when mixing for an opening act on tour, I noticed that a 'major' PA company used a custom active splitter for the headline act, which was their client; the opening act was given a multi-pair cable that was a passive split.

"That meant there was an immediate 5dB-6dB loss for the openers on the show; when the main act came on, the sound was louder, cleaner and unrestricted.

"We just don't agree with that philosophy. In fact, we *always* supply separate, dedicated main and monitor consoles for opening bands so that they can feel comfortable."

Micera chose to remain with McCauley loudspeaker components, and used direct-radiating cabinet loading techniques for the model 6246 15-inch speakers and model 6334 12-inch speakers. TAD 4001 compression drivers were mounted on the McCauley model 472 constant-directivity type horn.

"It is easier to execute direct-radiating designs when building cabinets, than it is to custom-fabricate complex wooden horns," he notes. "I also think that direct radiating enclosures offer better fidelity."

To ensure an easy changeover for high-frequency horns in the future, and to make driver replacement and maintenance simple, each modular loudspeaker enclosure was provided with a removable horn baffle plate. Sessions handles were installed for ease of handling; Micera found that drilling holes and injecting liquid Neoprene stopped buzzes and rattles, a common problem when hardware is attached to speaker enclosures.

"Many small things like that can combine to make a big difference in the way a system sounds and looks," Micera says. "We've found that every band we work with is very responsive to the quality that we attempt to put into both our service and our sound system. People do notice."

To power the modular loudspeaker system, Vista chose Peavey DECA-1200 amplifiers.

"There are many parameters to evaluate on this part of a system, including cost, sound quality, power consumption, and size and weight," Micera advises. "We were looking for a lot of horsepower, and wanted to save space and have a minimum current draw. Peavey specifies 15A of current under full load, and yet we have never been able to measure more than 2A per amplifier, even on heavy metal shows! These devices seem to be electrically very efficient."

For distributing electrical power, Vista built a custom PD system designed by Fred Micera. The system features sub-panels for electrical distribution to the stage, amp rack areas and monitor position; each sub-panel is housed in a custom Starflight road case and can be used as smaller, stand-alone power distribution panels. 220V is available at each amp rack, with 30A per leg at each rack via twist-lock connectors.

Heineman magnetic/hydraulic circuit breakers are employed; according to Micera, these were chosen due to their relatively slow "trip" curve; "They'll take a high initial power surge without tripping prematurely."

Each main distribution panel and sub-panel has both test points for each leg and voltage meters; hinged safety covers protect all 'hot' ac points. The various phases, ground and neutral are consistently color-coded throughout the system.

"We'll give a band up to 120A on 10 different quad boxes," Micera explains. "Usually though, at most regional shows, it only takes about 30A on stage for band gear."

Micera feels that the electrical distribution system is the crucial "heart" of a portable sound system; he spends time at each event checking the distro and monitoring the system's use of ac power.

"My main decisions have been based on experience and on listening."

"One interesting thing I've noticed is that our Crown and Halfer amplifiers in the monitor system draw *more* current than the DECA-1200s in the main system."

In building up Vista's system, Micera professes to have kept an open mind in searching for equipment to purchase.

"There are political ramifications, of course, when it comes to equipment choices," he offers. "Who you buy it from?—and whether or not you can get it—are all part of the equation.

"We have tried to go for what sounds best, not just what looks best in the marketing literature. My main decisions have been based on experience and on listening, *not* on salesmanship or marketing trends."

Vista's primary front-of-house mixing console is a Midas 32×8×2×24 (a complex bus configuration depicting a desk that was orginially designed for both sound reinforcement and simultaneous live recording). A set of 19-pair multicables carry signal lines to and from a well-stocked effects, which includes devices from Orban, Ursa Major, Yamaha, Lexicon, Symetrix

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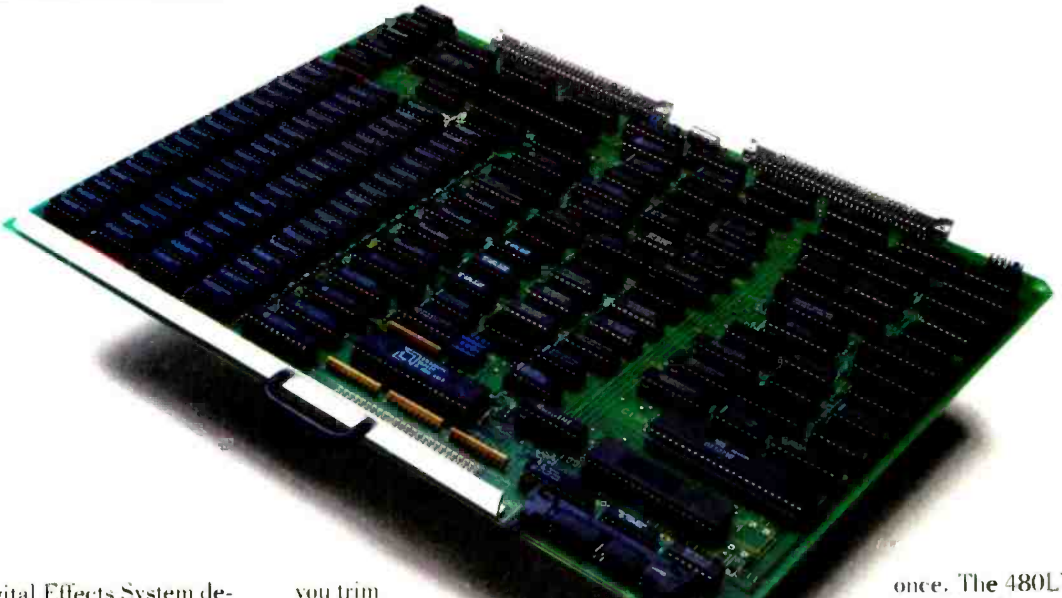

The 480L Digital Effects System delivers audio performance that surpasses conventional digital recorders: true to life sampling is a prime example of its advanced engineering. With the optional Sampling Memory Expander, the 480L becomes an astonishingly practical way to copy or move several seconds of audio from point A to points B and C.

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Figure 6. Vista's stage monitor houses a single LF driver and a JBL 2425 compression driver on a McCauley 421 radial horn.

Neale Frazier.

David Scheirman [left], Randy Bauske [far right] and associates.

and dbx. A 9-pair multicable connects the console with a drive rack that houses Klark-Teknik graphic equalizers, Brooke-Siren electronic crossovers and a Gold Line real-time analyzer (Figure 5).

A custom-built splitter panel receives input signals from 9- and 12-pair sub-snakes on stage; dual 19-pair cables connect the house-mix position with the stage-located splitter rack. Twin 16-pair cables connect the monitor mix position, which features a Soundcraft 24×10 model 400B, with each monitor mix including an inserted AudioArts 2700 equalizer, a Brooke-Siren System FDS-320 crossover and Hafler DH-500 amplifiers for low frequencies and Crown DC-300A Series 2s for highs.

McCauley stage monitors are provided, with both single 12- and single 15-inch versions in stock. JBL model 2425 compression drivers on McCauley model 421 radial horns are used (Figure 6).

"We like to have the input patching and monitor assignments pre-set as much as possible before we even put the system in the truck," Vista stage monitor technician Steve Gregory says. "We press groups in advance for as many details as they can give us about their stage setup. It saves time for everybody.

"And, if a monitor system is properly set up in the first place and well-maintained, we can spend *our* time getting the *right sounds*, instead of messing with cable patches. We recently did 190 different acts in three days at a large outdoor festival at the Seattle Center; advance planning made a *big* difference."

Fred Micera noted that, in the course of building up the Vista concert sytem, he came to realize that it is wise to carefully scrutinize every piece of gear that goes into a system, whether new or used, regardless of brand name.

"Every item we got in from a manufacturer was taken apart and inspected," he recalls. "It is *surprising* what can get by their quality control inspectors."

All loose connections were secured; nylon cable ties were used to give extra support for internal wiring harnesses. Relays and other heavy PCB components were secured to the boards with extra epoxy cement and all ICs were visually checked and fully seated in the consoles and processing equipment.

"On some of the new, compact special effects devices, it is important to beef up the frame supports, especially on the units that are only one rack space high, and maybe fifteen or sixteen inches deep. A lot of parts are being crammed in a small space there."

Any pieces of equipment with rear-panel terminal strips were given extra wire tiedowns to prevent flexing of signal wires; bevelled, compression-rubber washers were used for rack-mount screws. 18-gauge mic cable was used for internal rack wiring signal leads instead of the thinner, 20-gauge wire often seen.

"Each electronics rack has a swinging back door with a perforated metal screen. This door cuts down on the dust that gets inside, yet still provides good ventilation," Micera explains. "We can get inside easily for maintenance."

Ultimately, the test of a sound system's roadworthiness is a tightly-scheduled national tour.

"That is the direction we are heading," Vista Sound owner Neale Frazier says. "We're pretty confident that this system has been put together well and will see us through."

Both of these concert-sound systems have been assembled by optimistic sole proprietors working primarily on a regional basis; a general feeling of optimism in their respective geographical areas has led to significant investments in new hardware to upgrade existing systems. Those devices still not commercially available, such as power distribution systems and electronic rack panels, have been designed and fabricated on a custom basis to best suit the companies needs.

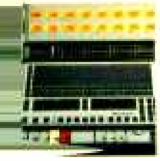
Both companies profess a strong interest in audio quality, and are seeing a positive market response to this philosophy. While highly visible marketing programs by pro-audio equipment manufacturers have served to inform these companies of available products, neither has based its purchasing decisions on sales programs or marketing images alone. Evaluation of the system's intended use, and of specific available products, has led to eventual purchase of gear for immediate use.

While it is difficult to project the actual amount of time needed for the recovery of capital investment in sound system components when assembling a competitive concert-sound system, it is fairly certain that those systems assembled with quality components by knowledgable technicians, and equipped to operate in a specific market, stand a better chance of success than do those pieced together in haphazard fashion.

Both companies represent regional rental sound systems for concert use that are in touch with the needs of the contemporary entertainment industry. R·E/P

For more details, contact: **Baus Engineering, Inc.**, 1684 Kalakaua Ave., Honolulu, HA 96826; 808-949-1969 or **Vista Sound**, 8229 44th Ave. W. Suite C, Mukilteo, WA 98725; 206-743-6811.

The mention of specific manufactured products should not be taken as an endorsement. This feature has been written with reader interest and education in mind— *DS.*



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83rd AES Convention Preview

As it enters its 40th year and presents its 83rd convention, the Audio Engineering Society, along with the entire industry, is in the middle of a period of change—in the way technology is used, in which mediums the final product will be presented and in what sort of facilities the work will be completed.

This year's theme takes the idea of change and sums up the state of the industry: "Audio and Video—Analog Present, Digital Future." According to AES, this year's convention will have the largest collection of technical papers and workshops and the largest product exhibition in its history.

On the following pages, **RE/P** presents its preshow convention coverage. If you're going to the show, this section will give you an idea of what to expect. And if you're not going to the convention, the section is a good way to keep informed on the industry's latest developments. Product information from all exhibiting companies is available via Reader Service Numbers on all entries in the Exhibitor Listings, and the "New Products" section beginning with the December issue will feature products that were introduced at the show.

A final word about this section: Every effort has been made to make it as accurate and up-to-date as possible. Because of our press deadlines, some changes and additions may not have been included. For final details, check your show program when you get to the convention.

At A Glance:

Event: AES 83rd Convention.

Dates: Oct. 16-19, 1987.

Location: New York.

Number of exhibitors: 225.

Estimated attendance: 12,000.

Exhibit hours:

Friday, Oct. 16
12 p.m.-7 p.m.

Saturday, Oct. 17
10 a.m.-6 p.m.

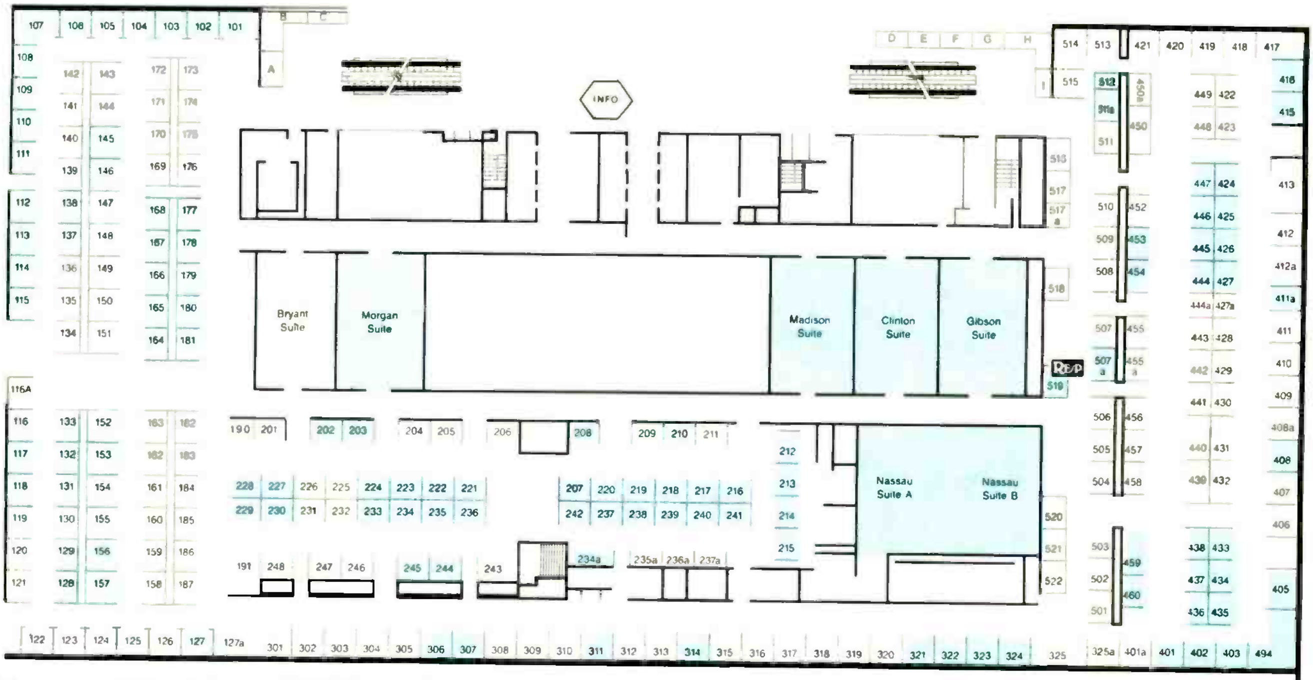
Sunday, Oct. 18
10 a.m.-5 p.m.

Monday, Oct. 19
10 a.m.-4 p.m.

More information: 212-661-8528.

REP's AES Coverage:

Hilton Hotel Exhibitors



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Issue advertisers and their booths are printed in blue.

ACO Pacific (515)
Adams-Smith (227-230)
Advanced Music Systems/Calrec (413)
Agfa-Gevaert (169-176)
AKG Acoustics (216-217, 240-241)
Allen & Heath Brenell (409-410)
Alpha Audio (112-115)
Amber Electro Design (206)
Amek Systems & Controls/TAC (131-133, 152-154, 116-118)
American Multimedia/Concept Design (319-320)
Ampex (107-110)
Analog Digital Synergy (325-325A)
Anchor Audio/ROH (208)
ANT Telecommunications (459-460)
Aphex Systems (202-203)
Applied Research & Technology (210)
Audico (452)
Audio Developments (160, 185)
Audio/Digital (305)
Audio Precision (306-307)
Audio-Technica (191, 248)
Audiotechniques (513-514)
Audio Video Consultants (301-302)
BASF Corporation (158-159, 186-187)
BSS (221-224, 233-236)
Beyer Dynamic (507A)
BGW Systems (189)
Bruel & Kjaer Instruments (507-509)
Bryston Vermont (130)
Cal Switch (235A, 236A)
Calzone Case Company (411)
Cetec Gauss (119-121)
Cetec Ivie (126)
Cetec Vega (127)
Cipher Digital (322-324)
Connectronics (303-304)
Countryman Associates (234A)
Crest Audio (147-148)
Crown International (504-506)
DDA (221-224, 233-236)
Design Direct Sound (408A)
Digital Creations Corporation (318)
DOD Electronics (456-458)
Dolby Laboratories (141-144)
Eastern Acoustic Works (190, 201)
Electro Sound (246-247)
Explorations (321)
Eventide (455-455A)
FM Acoustics (237A)
Gold Line (211)
Gotham Audio (161-163, 182-184)
David Hafler Company (243)
Harrison Systems (428-432, 439-443)
Heino Ilsemann (124-125)
Innovative Electronic Design (450A)
JBL Professional (Clinton Suite)
JRF Magnetic Sciences (111)
JVC Company of America (417-420)
Keyboard Technologies (309-310)
King Instrument Corporation (122-123)
Klark-Teknik Electronics (221-224, 233-236)
Lenco (311)
Lexicon (511A-512)
Magnefax International (127A)
Marshall Electronic (421)
Martin Audio Video Corporation (412, 412A)
Metro Audio (208)
Meyer Sound Laboratories (128-129, 156-157)
Milab International (321)
Minim Electronics (116A)
Mitsubishi Pro Audio Group (Bryant Suite)
Monster Cable (137-138)
Mosses & Mitchell (406)
Music Services International (308)
Neotek Corporation (404-405)
Rupert Neve Inc. (Morgan Suite)
New England Digital (Nassau Suite A&B)
Orban Associates (415-416)
Otari Corporation (424-427, 444-447, 453-454)
Panasonic/RAMSA (212-215)
Penny & Giles (407)
Professional Audio Services and Supply (315)
QSC Audio Products (244-245)
Recording Engineer/Producer/Sound & Video Contractor (519)
Renkus-Heinz (501)
RPG Diffusor Systems (314)
Saki Magnetics (408)
Samson Technologies (312-313)
Selco/Sifam (316)
Sennheiser Electronics (502-503)
Shure Brothers (422-423, 448-449)
Solid State Logic (Gibson Suite)
Soundcraft (Clinton Suite)
Sound Ideas (411A)
Sound Technology (145-146)
Soundtracs (207, 218-220, 237-239, 242)
Sound Workshop Professional Audio Products (450)
Stanton Magnetics (510)
Studer Revox (164-168, 177-181)
Studio Sound (317)
TASCAM (432-438)
Telex Communications (401-403)
3M/Magnetic Media Division (101-106)
Trident Audio USA (134-136, 149-151)
Turbosound (225-226, 231-232)
UREI (Clinton Suite)
Whirlwind (139-140)
Wireworks (204-205)
Xedit Corporation (155)
Yamaha (Madison Suite)

Audio Transformers

Choose from a wide variety of types and packages

REVISED SEPTEMBER 1986

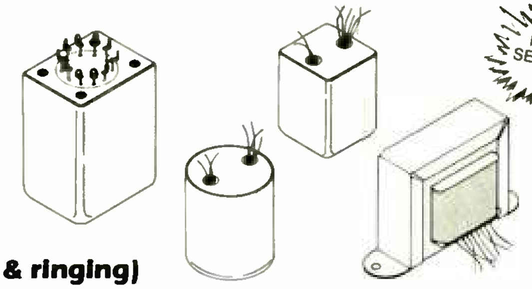
Computer optimized design

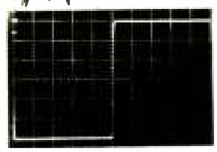
100% tested – consistent quality

Low distortion

Wide bandwidth

Minimum transient distortion (overshoot & ringing)





JE-16A
2 kHz Square Wave

INPUT TRANSFORMERS AND SPECIAL TYPES

	Model	Application	Impedance Ratio Pri-Sec	Turns Ratio Pri:Sec	20Hz Max Input Level[1]	Typical THD Below Saturation (%) 20 Hz / 1 kHz	Frequency Response (dB ref. 1 kHz) 20 Hz / 20 kHz	Band-Width[2] −3 dB @(kHz)	20 kHz Phase Response (degrees)	Over-Shoot (%)	Noise Figure (dB)	Magnetic Shield[4] (dB)	Number of Faraday[4] Shields	Package[5]	PRICES 1-19	PRICES 100-249	PRICES 1000
	MICROPHONE INPUT																
	JE-16-A JE-16-B	Mic in for 990 opamp	150-600	1:2	+8	0.036/0.003	−0.08/−0.05	230	−8	<1	1.7	−30	1	A = 1 B = 2	75.42 82.89	49.87 54.81	34.40 37.81
	JE-13K7-A JE-13K7-B	Mic in for 990 or I.C.	150-3750	1:5	+8	0.036/0.003	−0.09/−0.21	85	−19	<2	2.3	−30	1	A = 1 B = 2	75.42 82.89	49.87 54.81	34.40 37.81
*	JE-115K-E	Mic in for I.C. opamp	150-15K	1:10	−6	0.170/0.010	−0.50/+0.10	100	−16	<7	1.5	−30	1	3	54.81	36.24	28.39
	LINE INPUT																
	JE-11P-9	Line in	15K-15K	1:1	+26	0.025/0.003	−0.03/−0.30	52	−28	<3		−30	1	1	122.22	80.82	55.75
	JE-11P-1	Line in	15K-15K	1:1	+17	0.045/0.003	−0.03/−0.25	85	−23	<1		−30	1	3	52.32	34.59	27.10
	JE-6110K-B JE-6110K-BB	Line in bridging	36K-2200 (10K-600)	4:1	+24	0.005/0.002	−0.02/−0.09	125	−12	<1		−30	1	B = 1 BB = 2	73.95 85.59	48.90 56.59	35.88 39.04
	JE-10KB-C	Line in bridging	30K-1800 (10K-600)	4:1	+19	0.033/0.003	−0.11/−0.08	160	−9	<2		−30	1	3	53.17	35.16	24.53
	JE-11SSP-8M	Line in/ repeat coil	600/150-600/150	1:1 split	+22	0.035/0.003	−0.03/−0.00	120	−9	<3.5		−30	1	4	194.63	128.69	88.78
	JE-11SSP-6M	Line in/ repeat coil	600/150-600/150	1:1 split	+17	0.035/0.003	−0.25/−0.00	160	−5	<3		−30	1	5	98.39	65.06	44.88
	SPECIAL TYPES																
	JE-MB-C	2-way[3] mic split	150-150	1:1	+1	0.050/0.003	−0.16/−0.13	100	−12	<1		−30	2	3	44.85	29.65	23.24
	JE-MB-D	3-way[3] mic split	150-150-150	1:1:1	+2	0.044/0.003	−0.14/−0.16	100	−12	<1		−30	3	3	76.19	50.37	39.42
	JE-MB-E	4-way[3] mic split	150-150-150-150	1:1:1:1	+10	0.050/0.002	−0.10/−1.00	40	−18	<1		−30	4	1	114.40	75.64	52.18
	JE-DB-E	Direct box for guitar	20K-150	12:1	+19	0.096/0.005	−0.20/−0.20	80	−18	<1		−30	2	6	54.56	36.07	28.23

1. (dBu) Max input level = 1% THD; dBu = dBv ref. 0.775 V
2. With recommended secondary termination
3. Specifications shown are for max. number of secondaries terminated in 1000 ohm (typical mic preamp)
4. Separate lead supplied for case and for each faraday shield
5. Except as noted, above transformers are cased in 80% nickel mu-metal cans with wire leads.

PACKAGE DIMENSIONS:

	W	L	H
1 =	1 5/16" Diam.		× 1 9/16"
2 =	1 3/16"	× 1 3/16"	× 1 5/8"
3 =	1 1/8" Diam.		× 1 1/16"
4 =	1 1/2"	× 1 3/4"	2 1/2" w/solder terminals
5 =	1 5/8" Diam.		× 1 3/4"
6 =	1 1/8" Diam.		× 15/16"

NICKEL CORE OUTPUT TRANSFORMERS[6]

Model	Construction	Nominal Impedance Ratio Pri-Sec	Turns Ratio Pri:Sec	20 Hz Max Output Level[7] (dBu)	across (n) windings	600 Ω Load Loss (dB)	DC Resistance per Winding	Typical THD Below Saturation (%) 20 Hz / 1 kHz	Frequency Response (dB ref. 1 kHz) 20 Hz / 20 kHz	Band-Width −3 dB @ (kHz)	20 kHz Phase Response (degrees)	Over-Shoot (%)	Package	PRICES 1-19	PRICES 100-249	PRICES 1000
JE-11-BMCF	Bifilar 80% nickel	600-600	1:1	+26	1	−1.1	40Ω	0.002/0.002	−0.02/−0.00	>10MHz	−0.0	<1[9]	7	81.55	53.92	37.76
JE-11-DMCF	Bifilar 80% nickel	600-600	1:1	+21	1	−1.0	38Ω	0.004/0.002	−0.02/−0.00	>10MHz	−0.0	<1[9]	8	56.32	37.24	25.69
JE-123-BLCF	Quadfilar	600-600 150-600	1:1 1:2	+32	2	−1.1	20Ω	0.041/0.003	−0.02/−0.01	>450 170	−1.9 −4.0	<1[8]	7	73.85	43.14	29.76
JE-11SS-DLCF	Bifilar split/split	600-600 150-600	1:1 1:2	+27	2	−1.0	19Ω	0.065/0.003	−0.02/−0.01	>10MHz 245	−0.0 −2.5	<1[8]	8	53.62	35.45	24.46
JE-11-ELCF	Bifilar	600-600	1:1	+23.5	1	−1.1	40Ω	0.088/0.003	−0.03/−0.00	>10MHz	−0.0	<1[9]	9	36.36	24.04	16.59
JE-11-FLCF	Bifilar	600-600	1:1	+20.4	1	−1.6	58Ω	0.114/0.003	−0.03/−0.00	>10MHz	−0.0	<1[9]	10	27.36	18.09	12.48
JE-112-LCF	Quadfilar	600-600 150-600	1:1 1:2	+20.4	2	−1.6	29Ω	0.114/0.003	−0.03/−0.01	>450 205	−1.2 −3.2	<1[8]	10	32.80	21.69	14.96
JE-123-ALCF	Quadfilar	66.7-600	1:3	+26.5	3	−1.3	8Ω	0.125/0.003	−0.04/+0.06	190	−4.6	<6[8]	8	50.96	33.69	23.24
JE-11S-LCF	Bifilar w/ split pri.	600-600 150-600	1:1 1:2	+30	1 (sec)	−1.7	63Ω	0.058/0.002	−0.02/+0.01 −0.02/−0.05	>10MHz 155	+1.1 −4.1	<1[8]	8	50.96	33.69	23.24

6. Multifilar construction has no faraday shield: cannot be used as input transformer. All specifications are for 0 Ω source, 600 Ω load.
7. Max output level = 1% THD; dBu = dBv ref. 0.775 V
8. Source amplifier −3 dB @ 100 kHz
9. Source amplifier −3 dB @ 200 kHz
10. Output transformers are horizontal channel frame type with wire leads, vertical channel frames available. PC types available.

PACKAGE DIMENSIONS:

	W	L	H	Mounting Centers
7 =	1 1/2"	× 2 5/16"	× 1 15/16"	2 13/16"
8 =	1 5/16"	× 1 15/16"	× 1 5/8"	2 3/8"
9 =	1 1/8"	× 1 11/16"	× 1 3/8"	2"
10 =	1 1/16"	× 1 7/16"	× 1 3/16"	1 3/4"

* IMPROVED PERFORMANCE

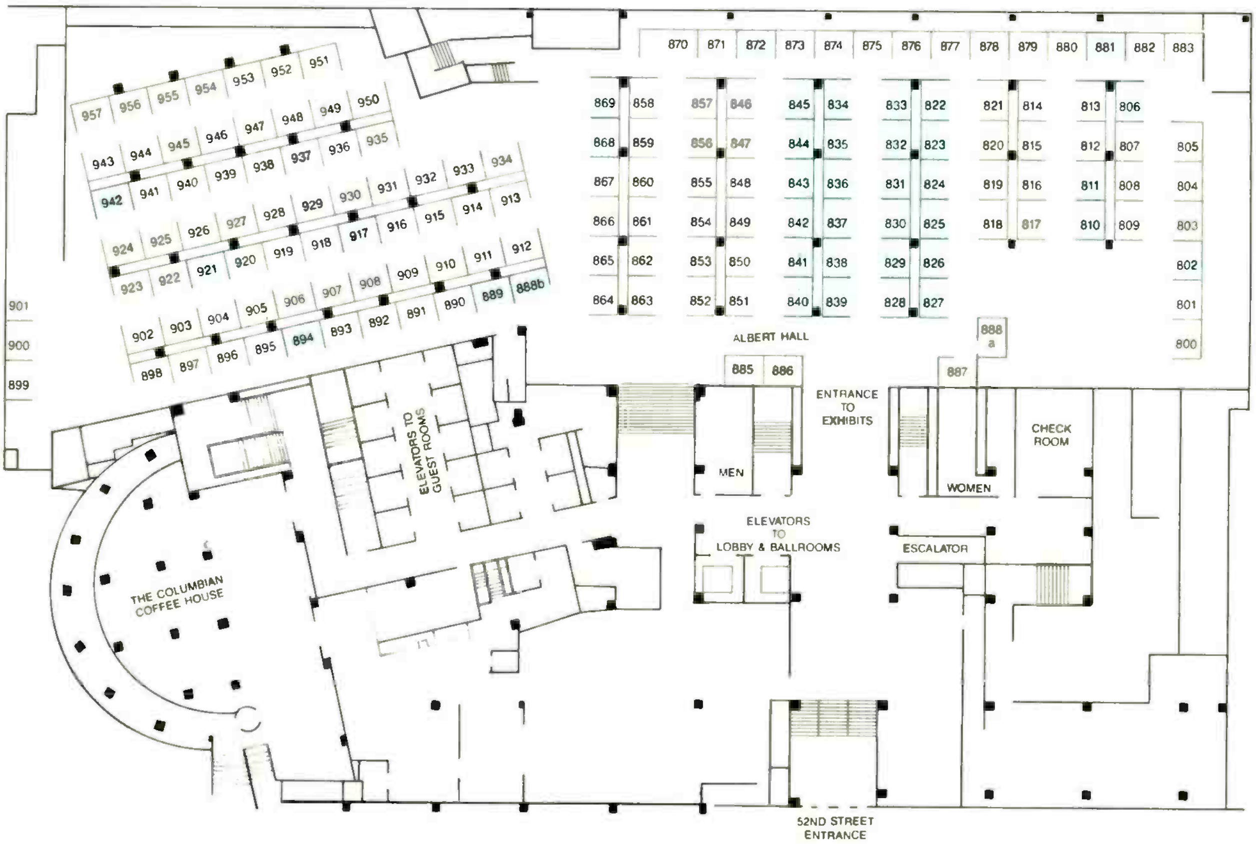
These charts include the most popular types which are usually available from stock. Many other types are available from stock or custom designs for OEM orders of 100 pieces or more can be made to order. Certified computer testing is available for OEM orders. Call or write for applications assistance and/or detailed data sheets on individual models.

Prices shown are effective 9/15/86 and are subject to change without notice. Packing, shipping, and applicable sales taxes additional.

jensen transformers
INCORPORATED

10735 Burbank Boulevard • North Hollywood, California 91601
(213) 876-0059 • TELEX via WUI 6502919207 MCI UW
Closed Fridays, visitors by appointment only.

Sheraton Centre Exhibitors



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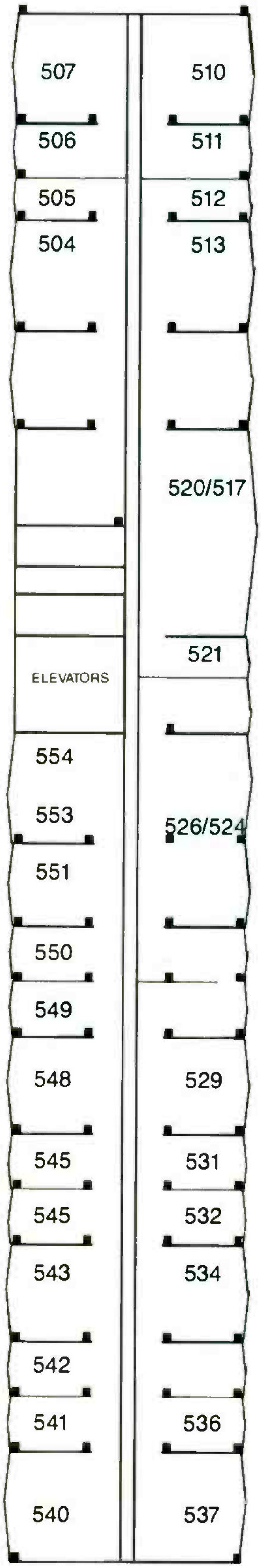
AB International Electronics (887)
AEG (873-876)
Aircraft Digital Music Library (902)
ALD Lab (856-857)
Alesis Studio Electronics (894)
American Modular Power (914)
Apex Machine Company (809-810)
API Audio Products (893)
Apogee Electronics (879)
Apogee Sound (942)
Audiocast (858)
Audio Accessories (920)
Benchmark Associates with
Downtown Design (805)
Berklee College of Music (938)
Brainstorm Electronics (931)
Canare Cable (802)
Carver Corporation (881)
C Audio (908)
Celestion Industries (866-867)
Clarity (863)
CMX Corporation (812-813)
Community Light & Sound (888A)
J.L. Cooper Electronics (807)
Court Acoustics Sales (927)
CST Manufacturing & Sales (880)
C-T Marketing (943)
Bill Daniels Company (951)
Digital Audio Research (820-821)
Digital Signal Processing & Control (918)
Dorrough Electronics (804)
Philip Drake Electronics (925-926)
Editron (885-886)
Evertz Microsystems (868-869)
FM Tubecraft Support Systems (917)
Forward Technology (924)
Full Compass Systems/Richmond Sound (954)
Full Sail Center for the Recording Arts (907)
Howe Technologies (912)
ILP Manufacturing (944)
IPS (909)
IQS (910)
KABA Research and Development (872)
Kenwood USA Corp. (923)
Thomas Klotz (905)
Leonardo Software (889, 888B)
Marshall Electronics (808)
Media Week Limited (937)
Micro Audio (921)
Micro-Point (911)
Music Maker Publications (915)
Musitech (952)
Nakamichi America (811)
Neutrik/Dialight (928)
Opcode Systems (932)
Optical Disc Corporation (849-848)
Oxmoor Corporation (939)
Pearl Microphone Laboratory (956)
Penn Fabrication (847-846)
Philips Subsystems and Peripherals (864-865)
Power Solutions (935)
PPG America (945-947)
Pro Co Sound (922)
Publison America (859-862)
Quested Monitoring Systems (903-904)
Research Technology International (953)
Saje (870-871)
Sanken Microphone (810)
Schoeps/Posthorn Recordings (884)
Shape Inc. (850-853)
Shep Associates (896)
Simmons Electronics (895)
Sonic Research Associates (916)
Sonosax (815)
Sony (822-845)
Soundmaster International (913, 934)
Southworth Music Systems (897)
SPARS (936)
Star Case (955)
Stramp USA (892)
Strand Magnetic Tapes (882-883)
Studio Master Systems (919)
Sunkyong (816-819)
Symetrix (933)
Tape Automation (948-950)
TC Electronic (890-891)
Technics (803)
Timeline (877-878)
Toolex Alpha (929-930)
Troisi (898)
27th Dimension (806)
VCL Audio (854-855)
Voyetra Technologies (940)



5th Floor Hilton Exhibitors

Akai/IMC (Rooms 512-513)
Audio Media Research (Room 550)
E-mu Systems (Room 542)
Fairlight Instruments (Rooms 517, 520-521)
Fane Acoustics (Room 531)
Fostex Corp. of America (Rooms 524/526)
Korg USA (Room 534)
Kurzweil Music Systems (Room 504)
Lenco (Room 549)
Lexicon (Rooms 506-507)
New England Digital (Room 540)
Panasonic/RAMSA (Room 510)
Peavey Electronics (Room 551)
Technical Audio Devices (Room 537)
Technos (Room 543)
WaveFrame Corporation (Room 529)

6th Floor Hilton Exhibitors

American Multimedia/Concept Design (Room 613)
Audio Design Associates (Room 650)
Music Services International (Room 604)
Roland Corp. USA (Rooms 616-617)
Tannoy North America (Room 622)
Westlake Audio (Room 620)
Zimbelman (Room 624)

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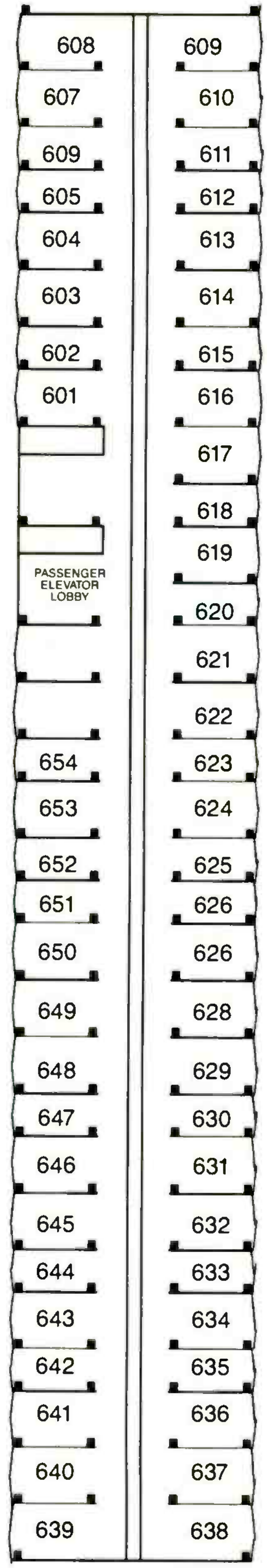
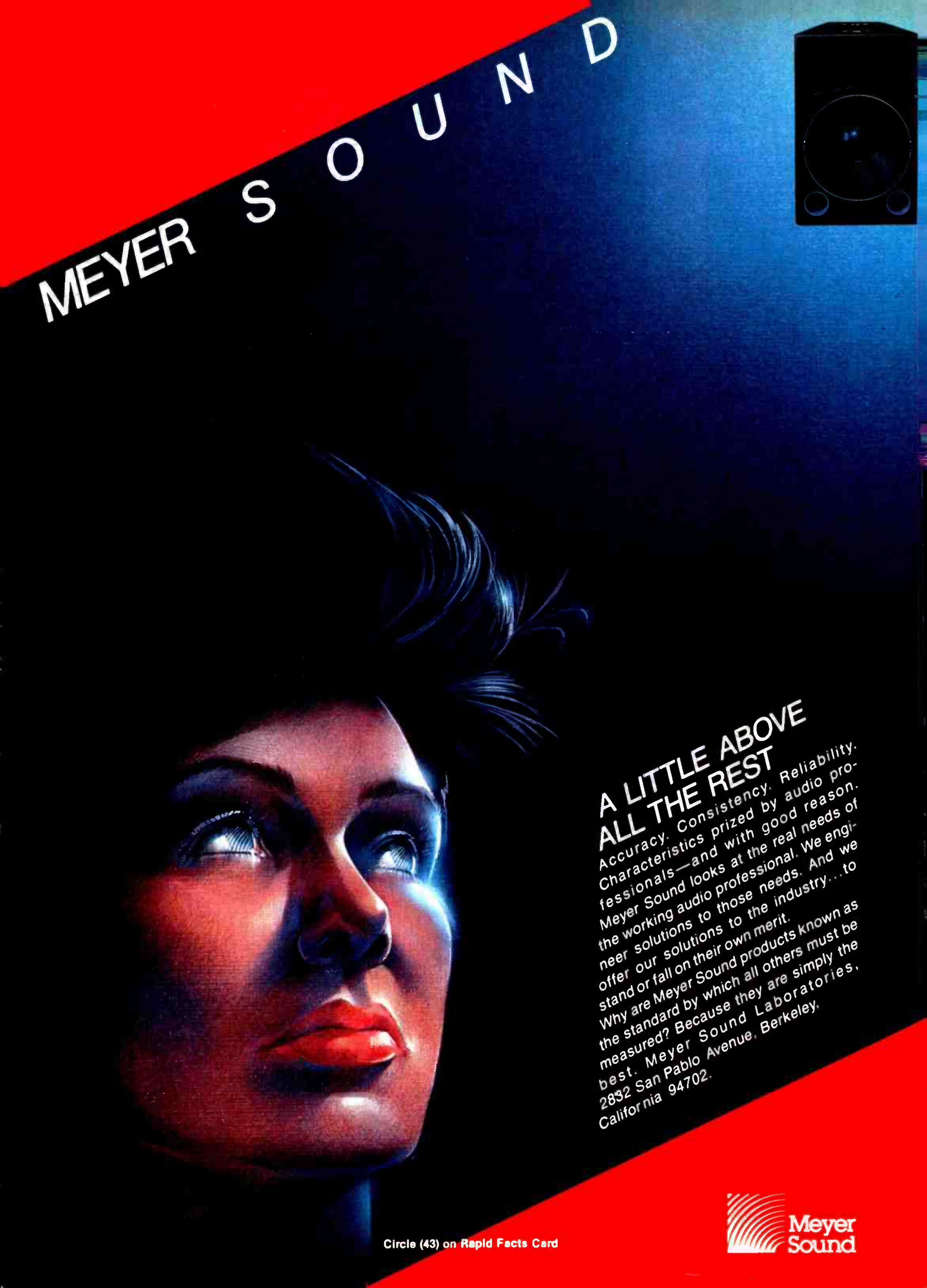



MEYER SOUND
A LITTLE ABOVE ALL THE REST
Accuracy. Consistency. Reliability. Characteristics prized by audio professionals—and with good reason. Meyer Sound looks at the real needs of the working audio professional. We engineer solutions to those needs. And we offer our solutions to the industry…to stand or fall on their own merit.
Why are Meyer Sound products known as the standard by which all others must be measured? Because they are simply the best. Meyer Sound Laboratories, 2832 San Pablo Avenue, Berkeley, California 94702.
Circle (43) on Rapid Facts Card
Meyer Sound

Exhibitor Listings

With a split venue and a variety of demo rooms and suites at this year's convention, there may be some confusion in locating where everything is. **RE/P's** pre-show coverage can help you make the most of your time if you're attending the show. If you aren't attending, then the coverage will keep you up-to-date with what is happening in the industry.

This alphabetical listing of exhibiting companies is the result of more than two month's worth of work and planning. We mailed a form to nearly all of the exhibitors, asking for new products that will be introduced and product lines that will be exhibited. From the forms that were returned to us, as well as from phone calls to companies, we obtained the information that is presented here.

Each listing is presented in several parts. On the first line is the company name and its booth number. To save space, companies that have more than one booth is notated by an asterisk. To easily show where the company is located, the booth number is preceded by an H or an S—H for the Hilton Hotel, and S for the Sheraton Centre.

If the companies could give us new product information, model numbers or names and a short description are listed. Product lines that are going to be exhibited at the show are listed below the new products.

For each entry, a circle number has been assigned. If you want more information on any of the companies, circle the appropriate number on the Rapid Facts Card, which is bound into the back of this issue.

And for companies that have advertised in this issue, the company names are listed in blue. On the same line as the circle number, the appropriate ad pages for that company are listed in blue ink, allowing you to check their ad for more immediate information.

Although this list is as comprehensive as possible at press time, there are changes and additions occurring every day. Every effort has been made to make this as complete and up-to-date as possible, but for final show information, refer to the official show program at the convention site.

If you are looking for a specific piece of equipment and want to find out which companies make it, refer to the product directory on page 82, which lists companies and booth numbers in 68 product and service categories.

And to locate exhibitors once you get to the show floors, maps of the Hilton, Sheraton and Hilton demo rooms begin on page 52.

AB International Electronics **(S-887)**
Product line
Power amplifiers.
Circle (125)

ACO Pacific **(H-515)**
New products
ACM48UP cardioid microphone; PS9048 battery-powered 48Vdc phantom supply.
Product line
Consoles; microphone accessories and preamplifiers.
Circle (126)

Adams Smith **(H-227)**
New products
Zeta Three audio-video-MIDI synchronizer; editing and graphic display upgrade of 2600 A/V.
Product line
Editing systems; time code synchronizers.
Circle (127) **See ad page 91**

Advanced Music Systems/Calrec **(H-413)**
Not available at press time.
Circle (128)

AEG Corporation **(S-873*)**
Not available at press time.
Circle (129)

Agfa-Geyaert **(H-169*)**
Product line
Recording tape; tape duplication.
Circle (130)

Aircraft Digital Music Library **(S-902)**
Not available at press time.
Circle (131)

Akai Professional/IMC **(H-Rm. 512*)**
New products
MPC60 MIDI production center; ASO10 MIDI production sequencer; DP2000 video and audio patchbay and distribution system; DP3200 audio patchbay and distribution system; PG2000 computer controller; MZ1000 RGB color monitor; EWI1000 electronic wind instrument; EVI1000 electronic valve instrument; MR76 programmable mix bay; PEQ6 programmable 7-band EQ.
Product line
Consoles; drum machines, equalizers; MIDI devices; patchbay and jack panels; signal processors; sound samplers; synthesizers and keyboards; tape machines.
Circle (132) **See ad page 117**

AKG Acoustics **(H-216*)**
New products
ADR68K digital reverb; DSP610 Delta Stereophony system; C-522 stereo ENG microphone; C-562 boundary layer microphone; C-410 headset microphone; K-260 professional semi-open-air-circumaural stereo headphones; CK-62DF diffused-field omnidirectional capsule for modular C-460B system.
Product line

AKG, continued

Microphones and accessories.
Circle (133) **See ad page 35**

ALD Lab **(S-856)**
Not available at press time.
Circle (134)

Alesis Studio Equipment **(S-894)**
New products
HR-16 16-bit digital drum machine; MMT-8 8-track MIDI recorder; Micro Enhancer, Micro Limiter and Micro Gate signal processors.
Product line
Compressors; drum machines; limiters; noise gates; reverb devices; signal processors.
Circle (135) **See ad page 15**

Allen & Heath Brenell **(H-409)**
New products
CMPTE hardware/software package for CMC series of mixers; SIGMA console with LED metering and I/O modules in small- and large-frame versions; CMC series of microprocessor-controlled mixers.
Product line
Console automation systems; consoles; MIDI devices; patchbay and jack panels; time code synchronizers.
Circle (136)

Alpha Audio **(H-112)**
Product line
Acoustic design and construction; acoustic treatment materials; editing systems; time code synchronizers.
Circle (137) **See ad pages 8, 67**

Amber Electro Design **(H-206)**
New products
AudioCheck software for 5500 programmable audio measurement system; new measurement options for models 5500 and 3500.
Product line
Test and measuring equipment.
Circle (138)

Amek Consoles/ Total Audio Concepts **(H-131*)**
New products
APC 1000 assignable production console; G2520 master recording console; Amek Classic broadcast TV production console; TAC SR9000 sound reinforcement console.
Product line
Consoles; crossovers and frequency dividers.
Circle (139) **See ad page 39**

American Modular Power **(S-914)**
Not available at press time.
Circle (140)

American Multimedia/ Concept Design **(H-319*, Rm. 613)**
New products
Digital Audio Analog Duplication System (DAAD); 790-2 dual-pancake loader modification; Signature series; quality control equipment.
Product line

American Multimedia, cont.
Cassette tape duplication.
Circle (141)

Ampex Magnetic Tape Division (H-107*)
New products
467 digital U-matic audio cassettes and shippers.
Product line
Recording tape; tape duplication.
Circle (142) **See ad page 17**

Analog Digital Synergy (H-325*)
New products
Multiformat digital mixing console.
Product line
Consoles.
Circle (143)

Anchor Audio (H-208)
Product line
Loudspeakers.
Circle (144) **See ad page 110**

ANT Telecommunications (H-459)
New products
E413 24-channel multitrack noise reduction system.
Product line
Noise reduction systems.
Circle (145) **See ad page 25**

Apex Machine Company (S-809)
New products
CA-15 1-color semi-automatic cassette printer; T-8 water-washout plate making machine; videocassette screener; semi-automatic cassette packaging machine.
Product line
Tape duplication.
Circle (146)

Aphex Systems (H-202*)
New products
110 Aural Exciter Type E; ESP-7000 surround-sound processor.
Product line
Equalizers; limiters; noise gates; signal processors.
Circle (147) **See ad page 75**

API Audio Products (S-893)
New products
550A equalizer; 560B graphic equalizer; 553 program equalizer; 5502 dual 4-band rack-mount with 550A-type EQ; 3124 4-input mic pre-direct box; 3124M 4-input mic pre/direct box; 312 mic pre-amp; 325 line amp; 701A power amp; 940M motorized fader system; 4832 input module for discrete console series.
Product line
Amplifiers; compressors; console automation systems; consoles; disc-mastering systems; equalizers; faders; limiters; pre-amplifiers; signal processors.
Circle (148)

Apogee Electronics (S-879)
New products
Audioscope integrated information display mainframe; enhancements of 944G and 944S linear phase filters for digital multitracks; Quested Monitoring Systems H405 close-field monitor.
Product line
Loudspeakers; tape machines; test and measuring equipment; VU/PPM meters.
Circle (149)

Apogee Sound (S-941)
New products
3x3 3-way tri-amped speaker system; AE-12 dual 18-inch high-power subwoofer system.
Product line
Loudspeakers.
Circle (150) **See ad page 92**

ART—Applied Research and Technology (H-210)
New products
⅔-octave and ⅓-octave intelligent equalizers with Smart Curve; ⅔-octave and ⅓-octave IEQ satellite; rack-mount integral video monitor.
Product line
Delays; equalizers; reverb devices; signal processors.
Circle (151) **See ad page 13**

Audico (H-452)
New products
System III videocassette tape loader, reloader and unwinder.
Product line
Peripheral and miscellaneous devices; test and measuring equipment.
Circle (152)

Audiocast (S-858)
Not available at press time.
Circle (153)

Audio Accessories (S-920)
Not available at press time.
Circle (154)

Audio Design Associates (H-Rm. 650)
Not available at press time.
Circle (155)

Audio Developments (H-160*)
New products
AD066 Portaflex audio "tools"; AD145E editing mixer; AD062H console version of AD062; AD150 rack module of two mic-line modules with EQ.
Product line
Compressors; consoles; editing systems; limiters; VU/PPM meters.
Circle (156)

Audio/Digital (H-305)
New products
ADD-3US industrial digital processor; ADD-3PG industrial digital processor with page mode.
Product line
Delays; signal processors.
Circle (157)

AMR—Audio Media Research (H-Rm. 550)
New products
DSR 100 digital stereo reverb; SyncController SMPTE time code generator/machine synchronizer; MCR 4/S 4-track cassette recorder; MIDI manager; PME 4 4-band parametric equalizer; CDS 2 dual-channel compressor/limiter/de-esser; NGT 2 dual-channel noise gate; PMA 70+ power amplifier.
Product line
Amplifiers; cable and connectors; compressors; consoles; delays; equalizers; limiters; loudspeakers; microphones and accessories; MIDI devices; noise gates; peripheral and miscellaneous devices; reverb devices; signal processors; tape machines; time code synchronizers.
Circle (158) **See ad page 29**

Audio Precision (H-306)
New products
DCX-127 and BUR-GEN, both for use with System One audio test system; "A" version of System One.
Product line
Test and measuring equipment.
Circle (159) **See ad page 73**

Audio-Technica U.S. (H-191)
New products
AT4071 condenser shotgun microphone; AT4073 condenser short shotgun microphone; AT4031 unidirectional condenser microphone; AT8506 4-channel phantom power supply.
Product line
Cable and connectors; headphones, headsets and intercom systems; loudspeakers; microphones and acccessories; MIDI devices; peripheral and miscellaneous accessories; phonograph cartridges.
Circle (160)

Audiotechniques (H-513*)
New products
Lydkraft PE-1B, MP-1B and PE-1A tube equalizers; Audiotechniques custom mic and cue boxes.
Product line
Cable and connectors; compressors; equalizers; equipment distributors; peripheral and miscellaneous accessories.
Circle (161)

Audio Video Consultants (H-301*)
New products
Double-pancake video loader.
Product line
Audio and video loaders.
Circle (162)

BASF Corporation (H-158*)
Product line
Recording tape.
Circle (163)

Benchmark Associates with Downtown Design (S-805)
Product line
Microphone pre-amps.
Circle (164)

Beyer Dynamic (H-507A)
New products
M700 supercardioid dynamic vocal microphone;

Beyer, continued

MCE80 supercardioid condenser instrument/vocal mic; MCE81 supercardioid condenser vocal mic; M58 dynamic ENG/EFP mic; MPC40 boundary layer condenser mic; MCE10 miniature hypercardioid condenser mic; OT990 semi-open-ear studio monitoring headphone; OT770 sealed-ear studio monitoring headphone; MC736-PV short shotgun condenser mic; MC737-PV long shotgun condenser mic; M2A717 fishpole boom.
Product line
Amplifiers; cable and connectors; consoles; crossovers and frequency dividers; equalizers; headphones, headsets and intercom systems; limiters; microphones and accessories; peripheral and miscellaneous accessories; racks and stands; signal processors; test and measuring equipment.
Circle (165) See ad page 27

BGW Systems **(H-209)**
New products
BGW GTA touring amplifier; BGW SPA1 signal processing subwoofer amplifier; BGW SPA3 signal processing 3-way amplifier.
Product line
Amplifiers.
Circle (166)

Brainstorm Electronics **(S-931)**
Not available at press time.
Circle (167)

Bruel & Kjaer **(H-507*)**
New products
4011 cardioid studio microphone; 2231 + BZ7104 portable reverb measuring system.
Product line
Microphones; test and measuring equipment.
Circle (168)

Bryston Ltd. **(H-130)**
New products
10B 2-way stereo and 3-way mono electronic crossover.
Product line
Amplifiers; crossovers and frequency dividers.
Circle (169)

BSS **(H-221*)**
New products
MSR-604 active mic splitter system; DPR-502 dual-channel noise gate; 360EQ internal EQ modification for FDS-360.
Product line
Crossovers and frequency dividers; limiters; maintenance equipment; noise gates; peripheral and miscellaneous accessories; signal processors; audio transformers.
Circle (170)

Cal Switch **(H-235A*)**
Product line
Cable and connectors; microphone accessories; patchbay and jack panels; test and measuring equipment.
Circle (171)

Calzone Case Company **(H-411)**
New products
Escort, Proline, Convoy and Ultima series cases.
Product line
Equipment cases.
Circle (172)

Canare Cable **(H-802)**
New products
4S11 heavy-gauge quad speaker cable; A2U2-L camera remote cable for audio and video; F-10 RCA plugs; F-11 mini phone plugs; D403-AT flexible MIDI cable; MR202-AT mult cable; L4E4-AT star quad mult cable; R300 stackable cable reels; GS-6 guitar/instrument cable; L-2B2-AT mini patchbay interface cable.
Product line
Cable and connectors; patchbay and jack panels.
Circle (173) See ad page 99

Carver Corporation (S-881)
New products
PM2.0T, PM1.5, PM350 and PM175 power amplifiers; PMX stereo crossover.
Product line
Amplifiers; crossovers and frequency dividers.
Circle (174) See ad page 37

C Audio (S-908)
Product line
Power amplifiers.
Circle (175)

Celestion Industries (S-866*)
New products
SR compact modular sound reinforcement system.
Product line
Loudspeakers.
Circle (176)

Cetec Gauss (H-119*)
New products
5220 2-way loudspeaker system; 5350 3-way loudspeaker system; 2480 master/bin tape duplicator; 1100 test equipment.
Product line
Loudspeakers; tape duplication; test and measuring equipment.
Circle (177)

Cetec Ivie (H-126)
New products
2502 and 2503 7-input, 4-channel automatic mic mixers; PC-40 portable computer-controlled ⅓-octave and 1-octave spectrum analysis system.
Product line
Amplifiers; compressors; crossovers and frequency dividers; delays; equalizers; limiters; microphone pre-amplifiers; signal processors; test and measuring equipment.
Circle (178)

Cetec Vega (H-127)
New products
R-32 wireless microphone receiver; PRO 2 wireless mic system; R-33 miniature portable wireless microphone receiver; Q-Plus wireless intercom system; T-39 handheld wireless microphone transmitter; 67B portable diversity wireless microphone receiver.
Product line
Headphones, headsets and intercom systems; microphones.
Circle (179) See ad page 93

Cipher Digital (H-322*)
New products
4810 Phantom VTR emulator; 4825 Shadowpad-mini offset entry keyboard; 4835 Shadowpad-maxi edit controller; 4890 Softouch-PC edit system.
Product line
Editing sytems; time code synchronizers.
Circle (180) See ad page 51

Clarity (S-863)
New products
XLV effects automation interface for Lexicon 224XL and 480L, AMS rmx-16, Quantec Room Simulator, Yamaha REV-1 and all voltage-controlled devices.
Product line
Console automation systems; MIDI devices; peripheral and miscellaneous accessories; reverb devices; signal processors.
Circle (181)

CMX Corporation (S-812*)
New products
CASS 1 computer-aided sound system; CASSIE edit-only version of CASS-1.
Product line
Console automation systems; editing systems.
Circle (182)

Community Light and Sound (S-888A)
New products
CS52 3-way sound reinforcement/keyboard loudspeaker.
Product line
Loudspeakers.
Circle (183)

Connectronics (H-303)
Product line
Cable and connectors.
Circle (184)

J.L. Cooper Electronics (S-807)
New products
MAGI mixer automation gain interface; SAM 3.5-inch disk for automatic archiving of automation passes.
Product line
Console automation systems; MIDI devices; time code synchronizers.
Circle (185)

Countryman Associates (H-234A)
Product line
Headphones, headsets and intercom systems; microphones and accessories.
Circle (186) See ad page 115

Court Acoustics Sales (S-927)
Product line
Loudspeakers.
Circle (187)

Crest Audio (H-147)
New products
8001 and FA800 power amplifiers.
Product line
Amplifiers.
Circle (188)

Crown International (H-504*)
New products
CM-100, -200 and -300 microphones; Select ser-

ies microphones; PB-2 amplifier; Macrotech series amplifiers; FFX-2 stereo electronic crossover.
Product line
Amplifiers; microphones.
Circle (189)

CST Manufacturing and Sales **(S-880)**
Not available at press time.
Circle (190)

C-T Audio Marketing **(S-943)**
New products
C-DUCER MIDI acoustic percussion trigger; OHM PA speaker; GT4A 4-channel noise gate.
Product line
Amplifiers; loudspeakers; microphone preamplifiers; microphones; noise gates; racks and stands.
Circle (191)

Bill Daniels Co. **(S-951)**
Not available at press time.
Circle (192)

dbx **(H-dbx Suite)**
New products
RTA-1 real time analyzer system; 929 single-ended noise reduction module for 900 series; 1531 graphic equalizer.
Product line
Compressors; equalizers; limiters; noise gates; noise reduction systems; signal processors; test and measuring equipment.
Circle (339)

DDA **(H-221*)**
New products
DCM-323 console; D series console with tape returns expanded to 24.
Product line
Console automation systems; consoles.
Circle (193)

Design Direct Sound **(H-408A)**
New products
80 series II live mixing desk; CFDI-81 small close-field horn; CFDI-90 small horn; CFD2-51 tour pack horn.
Product line
Consoles; loudspeakers.
Circle (194)

Digital Audio Research **(S-820*)**
Not available at press time.
Circle (195)

Digital Creations **(H-318)**
New products
Diskmix moving faders automation system.
Product line
Console automation systems; console faders.
Circle (196)

Digital Signal Processing & Control **(S-918)**
Not available at press time.
Circle (197)

DOD Electronics **(H-456*)**
New products
DSP128 signal processor; software development system for DSP128; Audio Logic SC31 graphic equalizer; Audio Logic R203 digital delay; Audio Logic PA86 psychoacoustic signal processor; Audio Logic PQ52 parametric equalizer.
Product line
Compressors; consoles; crossovers and frequency dividers; delays; educational courses and programs; equalizers; expanders; limiters; microphone pre-amps; MIDI devices; noise gates; noise reduction systems; phasers; pitch shifters; reverb devices; signal processors.
Circle (198)

Dolby Laboratories **(H-141*)**
New products
365 2-channel interface for SR or A-type NR; Cat.

Dolby, continued

280 SR module for 360 and M series frames; Cat. 431 SR module for multitrack applications.
Product line
Noise reduction systems; signal processors.
Circle (199)

Dorrough Electronics **(S-804)**
Product line
Compressors; limiters; signal processors; VU/PPM meters.
Circle (200)

Philip Drake Electronics **(S-925*)**
New products
6000 series intercom/talkback system.
Product line
Amplifiers; headphones, headsets and intercom systems; patchbay and jack panels.
Circle (201)

Eastern Acoustic Works **(H-190*)**
New products
KF850T and KF160T Virtual Array loudspeaker systems for concerts and installations; SB850C subwoofer loudspeaker system; MX800-8T electronic crossover system; MH102-60T, MH102-90T and MF102T 60-, 90- and 120-degree mid/high frequency reproducer.
Product line
Loudspeakers.
Circle (202)

Editron **(S-885*)**
Not available at press time.
Circle (203)

Electro Sound **(H-247*)**
New products
ES8000 modified slave; ES4800 digital control slave.
Product line
Tape duplication; test and measuring equipment.
Circle (204)

E-Mu Systems **(H-Rm. 542)**
New products
Emulator III digital sound production system; EMAX HD digital sampling system; SP-1200 sampling percussion system.
Product line
Sound samplers; synthesizers and keyboards.
Circle (205)

Eventide **(H-455*)**
New products
To be announced.
Product line
Delays; MIDI devices; pitch shifters; reverb devices; signal processors.
Circle (206)

Evertz Microsystems **(S-868*)**
New products
7100 audio transport; 7000 audio/video chase synchronizer; 120 multifunction display; 4010 LTC and VITC time code generator, reader and character inserter.
Product line
Time code synchronizers.
Circle (207) **See ad page 86**

Explorations **(H-321)**
New products
Milab LC-28 transformerless condenser microphone.
Product line
U.S. distributor of Milab microphones.
Circle (208) **See ad page 116**

Fairlight **(H-Rms. 517, 520/521)**
New products
MFX hardware/software package and Bernouli and WORM options for Series III.
Product line
Music production libraries; sound samplers; synthesizers and keyboards.
Circle (209) **See ad page 19**

Fane Acoustics (H-Rm. 531)
Product line
Amplifiers; loudspeakers.
Circle (210)

FM Acoustics (H-237A)
New products
FM216 precision line level interface.
Product line
Amplifiers; cable and connectors; crossovers and frequency dividers.
Circle (211)

FM Tubecraft (S-917)
New products
AF series acoustical foam; ETR series tubular racks.
Product line
Acoustic treatment materials; racks and stands.
Circle (212) See ad page 10

Forward Technology (S-924)
Not available at press time.
Circle (213)

Fostex Corporation of America (H-Rm. 524/526)
New products
4011 video character inserter.
Product line
Consoles; crossovers and frequency dividers; equalizers; headphones, headsets and intercom systems; loudspeakers; microphones and accessories; MIDI devices; patchbay and jack panels; peripheral and miscellaneous accessories; recording tape; tape machines; time code synchronizers.
Circle (214) See ad page 45

Full Compass Systems/ Richmond Sound (S-954)
Not available at press time.
Circle (215)

Gold Line (H-211)
New products
ASA30B portable ⅓-octave real-time analyzer; LM27P5 rack-mount ⅓-octave real-time analyzer.
Product line
Crossovers and frequency dividers; lighting equipment; limiters; noise gates; signal processors.
Circle (216)

Gotham Audio (H-161)
New products
Neumann/Teldec DDM CD Compact Disc mastering system; Neumann RSM 190 stereo shotgun mic system; Harmonia Mundi Acustica digital equalizer and mixer system; EMT246 digital reverb; Audio Developments AD 145 Pico stereo mixer.
Product line
Consoles; delays, disc-mastering systems; equalizers; limiters; loudspeakers; microphones and accessories; noise reduction systems; phonograph cartridges; phonograph turntables; reverb devices; signal processors; tape machines; test and measuring equipment; VU/PPM meters; distributors of Neumann, Teldec, EMT-Franz, Harmonia Mundi Acustica and Audio Developments.
Circle (217)

David Hafler Co. (H-243)
New products
P-230 MOSFET amp.
Product line
Amplifiers.
Circle (218)

Harrison Systems (H-428*)
New products
PRO-790 audio console; AIR-790 on-air radio console.
Product line
Console automation systems; console faders; consoles.
Circle (219)

Heino Ilseman (H-124*)
Product line
Cassette loading equipment.
Circle (220)

Howe Technologies (S-912)
New products
2300A phase chaser.
Product line
Consoles; signal processors.
Circle (221)

ILP Manufacturing (S-944)
Not available at press time.
Circle (222)

Innovative Electronic Designs (S-450A)
New products
4000 automatic mixer system; 5000 audio processing system; 6000 power amplifier system; 500AC/500 representative computer programs of Announcement Control System and Audio Control System.
Product line
Amplifiers; compressors; delays; signal processors.
Circle (223)

IPS (S-909)
Not available press time.
Circle (224)

IQS (S-910)
New products
System 416 modular signal analyzer; 416-2a dual-channel audio signal analyzer; 401 FFT spectrum analyzer; 416-SSM-2 arbitrary waveform signal source; 416-VSP vector signal proces-

"RAMSA calls their WR-8428 a post-production recording console. I call ours terrific. And use it to record Superior Court, ESPN Sports, People's Court and other national TV shows. Why? Because it performs like consoles that cost twice the price. And I've had zero complaints. Crosstalk is inaudible. Love RAMSA's mix matrix, too. It lets me assign busses and mix to feed different areas of program to different destinations–even at different levels, as needed." *Dick Liebert, Chief Engineer, The Production Group, Los Angeles.* For more information contact RAMSA at 6550 Katella Avenue, Cypress, CA 90630 714-895-7277.

RAMSA

Panasonic
Industrial Company

Circle (39) on Rapid Facts Card

IQS, continued

sor; 302 and 312-S reference microphones; Spectrum Analyst Pak 1 software; Signal Analyst Pak 1 software; Computer Aided Speaker Design software; Direct-to-Disk software.
Product line
Microphone accessories; signal processors; test and measuring equipment.
Circle (225)

JBL Professional (H-Clinton Suite)
New products
Control 5 2-way structural-foam molded enclosure; 2426 1-inch throat exit compression driver; 8330 surround-sound loudspeaker system.
Product line
Amplifiers; crossovers and frequency dividers; delays; equalizers; loudspeakers; signal processors.
Circle (226)

JRF Magnetic Sciences (H-111)
New products
PLX series replacement heads for ¼-inch machines; PLX replacement heads for 16mm and 35mm and magnetic film equipment.
Product line
Magnetic recording heads.
Circle (227) **See ad page 66**

JVC Company of America (H-417*)
Not available at press time.
Circle (228)

KABA Research and Development (S-872)
New products
Turnkey system for production of audiophile cassettes; Cool Power Tower oak equipment cabinet.
Product line
Racks and stands; tape duplication.
Circle (229) **See ad page 61**

Kenwood USA (S-923)
New products
DA-3500A CD encoder; DR-3552 CD decoder; DB-3545 CD jitter analyzer; DR-3750A R-DAT decoder; DA-5730 R-DAT encoder; DR-5740 R-DAT jitter analyzer; DG-3400A digital audio signal generator; super high-resolution CD standard player.
Product line
Test and measuring equipment.
Circle (230)

Keyboard Technologies (H-309*)
Not available at press time.
Circle (231)

King Instrument Corp. (H-122*)
New products
2797 dual-supply audio loader; 2500 dual-supply VHS videocassette loader.
Product line
Tape duplication.
Circle (232)

Klark-Teknik Electronics (H-221*)
New products
DN-405 mono parametric EQ; DN-410 2-channel parametric EQ; Jadel MkII close-field monitor system; DN780MIDI, for MIDI operation of DN780 program sequence.
Product line
Console automation systems; consoles; crossovers and frequency dividers; delays; equalizers; limiters; loudspeakers; maintenance equipment; MIDI devices; noise gates; peripheral and miscellaneous devices; reverb devices; signal processors; test and measuring equipment; audio transformers.
Circle (233) **See ad page 11**

Thomas Klotz (S-905)
Not available at press time.
Circle (234)

Korg USA (H-Rm. 534)
New products
DRV3000 digital reverb; DRV2000 reverb/multi-effects processor; DRM1 digital rhythm module; DDD5 drum machine; DSM-1 digital sampling synthesizer module; DSS memory expansion; Soundesigner software for DSS-1; Macintosh computer with Opcode interface and cables.
Product line
Delays; drum machines; headphones, headsets and intercom systems; reverb devices; signal processors; sound samplers; synthesizers and keyboards.
Circle (235) **See ad page 31**

Kurzweil Music Systems (H-Rm. 504)
1000 PX Professional Expander; 1000 GX, -HX and -SX guitar, horn and string expanders; K1000 keyboard version of 1000 PX; output option for K250 that adds 12 direct mono outputs.
Product line
Synthesizers and keyboards.
Circle (236) **See ad page 89**

Lenco (H-311, Rm. 549)
New Products
MPA-2300 high-definition monitor power amplifier; MPA-2100 high-definition power amplifier; 600 series audio distribution products; 300/400 video terminal modules.
Product line
Amplifiers; peripheral and miscellaneous accessories; test and measuring equipment.
Circle (237) **See ad page 103**

Leonardo Software (S-889*)
New products
L-001 Professional Librarian sound effects and music library system; L-008 Cueprinter re-recording cue sheet printing software; L-009 Spotmaker A&R paperwork organizer.
Product line
Libraries; software.
Circle (238) **See ad page 64**

Lexicon (H-511A*, Rm. 506*)
Not avaailable at press time.
Circle (239) **See ad page 47**

Magnefax International (H-127A)
New products
3800 3-slave audiocassette duplicator.
Product line
Recording tape; tape duplication.
Circle (240)

Marshall Electronic (H-421)
Not available at press time.
Circle (241)

Marshall Electronics (S-808)
New products
PJM-18, -24 and -36 high-definition patch cords; 2921, 2919 and 2941 speaker cables; speaker cable in 8-, 10-, 12- and 16-gauge.
Product line
Cable and connectors; patchbay and jack panels; peripheral and miscellaneous accessories; racks and stands.
Circle (242)

Martin Audio Video Corp. (H-412*)
Not available at press time.
Circle (243) **See ad page 92**

Metro Audio (H-208)
New products
CT2 cable tester; PP1 phase detector; LM1 headphone bridging amp.
Product line
Maintenance equipment; peripheral and miscellaneous accessories; test and measuring equipment.
Circle (337)

Meyer Sound Labs (H-128*)
New products
500 R loudspeaker system; 500 RW stage monitor system; P-1A control electronics unit for UPM-1 loudspeaker.
Product line
Amplifiers; equalizers; loudspeakers.
Circle (244) **See ad page 58**

Micro Audio (S-921)
New products
IBM computer interface card and software for programming to Micro Audio ⅓-octave equalizer.
Product line
Equalizers; test and measuring equipment.
Circle (245) **See ad page 78**

Micro-Point (S-911)
Not available at press time.
Circle (246)

Minim Electronics (H-116A)
Not available at press time.
Circle (247)

Mitsubishi Pro Audio Group (H-Bryant Suite)
New products
Westar 8300 film re-recording console; Westrex RA-1739 digital stereo photographic film sound recorder; Mitsubishi X-400/8 8-channel DAT recorder; Mitsubishi X-86 2-channel DAT master-

ing recorder; Mitsubishi XE-2 2-channel digital audio editor; ACS audio crossbar mixer/switcher.
Product line
Amplifiers; compressors; console automation systems; consoles; console faders; delays; editing systems; equalizers; expanders; faders; limiters; noise gates; peripheral and miscellaneous devices; signal processors; tape machines.
Circle (248)

Monster Cable (H-137)
New products
Prolink cable for interconnections, patchbays, instruments and tube microphones.
Product line
Cable and connectors.
Circle (249) See ad page 112

Mosses & Mitchell (H-406)
Product line
Patchbay and jack panels.
Circle (250)

Music Services International (H-308, Rm. 604)
Not available at press time.
Circle (251)

Musitech (S-952)
Not available at press time.
Circle (252)

Nakamichi America (S-811)
New products
MR-1 3-head master recorder cassette deck; MR-2 2-head master recorder cassette deck; SP-7 stereo headphones.
Product line
Headphones, headsets and intercom systems; recording tape; tape duplication; tape machines.
Circle (253) See ad page 33

Neotek (H-404*)
Product line
Consoles.
Circle (254) See inside back cover

Neutrik/Dialight (S-928)
New products
NC3FX-HD and NC3MX-HD male and female weatherproof XLR connectors; NP3TT bantam plug; XSR connectors; NL4FC 4-pin speaker connector; NL4MP 4-pin speaker receptacle; NJ3FC6 locking ¼-inch jack; NC3FX-S XLR connector with built-in rotary switch; NC3FPP and NC3MPP male and female plastic panel receptacles; circular connector system.
Product line
Cable and connectors; loudspeakers; microphone accessories; patchbay and jack panels; audio transformers.
Circle (255)

Rupert Neve (H-Morgan Suite)
New Products
Enhanced V series 60-input console; Necam 96 computer-assisted mixdown system; 8248 multitrack recording and mixdown console; Digital

Neve, continued

Transfer Console for Compact Disc mastering.
Product line
Console automation systems; console faders; consoles.
Circle (256) See ad page 23

New England Digital (H-Nassau Suite A&B, Rm. 540)
New products
Software updates and optical disk storage drive for Synclavier; self-contained remote configurations and new software for Direct-to-Disk Multitrack Recorder.
Product line
Sound samplers; synthesizers and keyboards; tape machines.
Circle (257) See ad page 65

Opcode Systems (S-932)
Not available at press time.
Circle (258)

Optical Disc Corporation (S-848*)
New products
530 Compact Disc mastering system; 533 CD analyzer.
Product line
Disc-mastering systems.
Circle (259)

Orban Associates (H-415*)
New products
787A programmable mic processor; 642B parametric equalizer; 222A stereo spatial enhancer; programmable parametric equalizer.
Product line
Compressors; limiters; reverb devices; signal processors.
Circle (260) See ad page 68

Otari Corporation (H-424*)
New products
MX-55 ¼-inch 2-track tape machine with time code, also in ¼-inch 4-track version.
Product line
Tape duplication; tape machines; time code synchronizers.
Circle (261) See ads pages 3, 117

Oxmoor Corporation (S-939)
New products
DCA-2 digital control attenuator; RC-16 remote control; DEQ-29 ⅓-octave programmable equalizer; 4x4 4-channel buffer amplifier.
Product line
Equalizers; faders; signal processors.
Circle (262)

Panasonic/RAMSA (H-212*, Rm. 510)
New products
WR-S852 40-input modular sound reinforcement mixing console; WR-M840 40-input modular stage monitor mixing console; WU-8119 surround sound module for WR-8428 console; WP-9440 power amplifier; WS-SP2 subwoofer processor for WS-A240 subwoofer.
Product line
Amplifiers; consoles; crossovers and frequency dividers; loudspeakers; microphones.
Circle (263) See ad page 71

Pearl Microphone Laboratory (S-956)
Product line
Microphones and accessories.
Circle (264)

Peavey Electronics (H-Rm. 551)
New products
HDH-3 subwoofer; HDH-4 mid/hi enclosure; HDH processing controller; HDH-2 2-way enclosure; HDH-M 2-way monitor enclosure; CS-900, M-7000 and DECA 2451 power amps; MD 421-24 stereo mixing console; PVM-45 cardioid electret condenser mic; UniVerb and AddVerb reverb processors; PKM 8128 and -8128E keyboard mixer and 8-channel expansion module; DEP 3.2S sampling digital delay effects processor.
Product line
Amplifiers; cable and connectors; consoles; crossovers and frequency dividers; delays; equalizers; equipment cases; loudspeakers; microphones and accessories; MIDI devices; peripheral and miscellaneous accessories; racks and stands; reverb devices; signal processors; sound samplers; audio transformers.
Circle (265)

Penn Fabrication (S-846*)
New products
Flight case and speaker mounting hardware; rack-mounting equipment; connectors; jack sockets and electrical fittings; knobs and sliders; microphone speaker and lighting stands; sound absorbing foam.
Product line
Acoustic treatment materials; connectors; lighting equipment; peripheral and miscellaneous accessories.
Circle (266)

Penny & Giles (H-407)
Product line
Faders.
Circle (267)

Philips Subsystems & Peripherals (S-864*)
Not available at press time.
Circle (268)

Power Solutions (S-935)
Not available at press time.
Circle (269)

PPG America (S-945*)
Product line
Synthesizers and keyboards.
Circle (270) See ad page 107

Pro Co Sound (S-922)
New products
TT-448 patchbay system; DB-1 direct box; MS-2 and -3 mic splitter boxes; MC-2 mic combiner box; AV-1 A–V interface box; HJ-6 and HJ-4P headphone junction boxes; DB-4 4-channel direct box; MS-42 and -43 4-channel mic splitters; IT-4 and -8 4- and 8-channel line-level out-

put isolation transformers; RMS-2 recording monitor switcher; lines of mic and speaker cable.
Product line
Cable and connectors; patchbay and jack panels; peripheral and miscellaneous accessories.
Circle (271)

Professional Audio Services and Supply **(H-315)**
New products
802 BBE signal processor; Neumann microphones; Passco direct box.
Product line
Equipment distributors.
Circle (272)

Publison America **(S-859*)**
Not available at press time.
Circle (273)

QSC Audio Products **(H-244)**
New products
MPS 2300 2-zone music and paging system; A2150 and A2300 power amplifiers.
Product line
Amplifiers.
Circle (274) See ad page 9

Quested Monitoring Systems **(S-903*)**
Product line
Studio monitoring systems.
Circle (275)

Recording Engineer/Producer
Stop by booth 519 in the Hilton Hotel to meet RE/P's sales and editorial staff, including Cameron Bishop, group vice president; Dennis Milan, publisher; Fred Ampel, editorial director; Michael Fay, editor; Dan Torchia, staff editor; Stan Kashine, East Coast sales representative; Mary Tracy, Midwest sales rep; and Herb Schiff, Jason Perlman and Chris Woodbury Leonard, West Coast sales reps.

Renkus-Heinz **(H-501)**
New products
B series Dynagard speaker system.
Product line
Amplifiers; loudspeakers.
Circle (276)

ROH **(H-208)**
New products
302 master party line station.
Product line
Headphones, headsets and intercom systems.
Circle (277)

Roland Corp. US **(H-Rm. 616*)**
New products
D-550 linear synthesizer module; MT-32 MIDI sound module; S-550 digital sampler; Maestro "S" sequencing software; S-220 digital sampler; MC-500 Microcomposer Performance Package; MC-500 Microcomposer Bulk Librarian; MC-500 rhythm bank; VP-70 voice processor; M-240 24-channel line mixer; M-160 16-channel line mixer.
Product line
MIDI devices; signal processors; sound samplers; synthesizers and keyboards.
Circle (278)

RPG Diffusor Systems **(H-314)**
New products
Acoustical treatment system.
Product line
Acoustic design and construction; acoustic treatment materials.
Circle (279) See ad page 94

RTI—Research Technology International **(S-953)**
New products
TapeChek DII dropout analyzer.
Product line
Test and measuring equipment.
Circle (280)

Saje **(S-870*)**
Not available at press time.
Circle (281)

Saki Magnetics **(H-408)**
New products
Replacement heads for various brands of tape machines.
Product line
Tape machine replacement heads.
Circle (282) **See ad page 100**

Samson Technologies **(H-312)**
New products
BR-3 synthesized true diversity selectable wireless receiver; BH-3 synthesized hand-held transmitter; BT-3 synthesized belt-back transmitter.
Product line
Wireless microphones.
Circle (283)

Sanken Microphone **(S-810)**
New products
CMS-7 MS stereo portable microphone.
Product line
Microphones and accessories.
Circle (284) **See ad page 43**

Schoeps/Posthorn Recordings **(S-884)**
New products
MK 21 subcardioid capsule; The Box real-time soundstage analyzer.
Product line
Equipment distributors; equipment rental and leasing; microphones and accessories; VU/PPM meters.
Circle (285)

Selco/Sifam **(H-316)**
Product line
VU/PPM meters.
Circle (286)

Sennheiser **(H-502*)**
New products
MKH 30 figure-eight studio condenser microphone; MKE-4032 handheld condenser stage mic; MKE 48 drummer's headset; HD 250 studio headphone monitor.
Product line
Consoles; headphones, headsets and intercom systems; microphones and accessories.
Circle (287)

Shape Inc. **(S-850*)**
New products
VHS loading system; CD lift/lock storage case; CD flip file storage rack; CD retailer packaging format.
Product line
Peripheral and miscellaneous devices; recording tape; tape machines; test and measuring equipment.
Circle (288)

Shep Associates **(S-896)**
Not available at press time.
Circle (289)

Shure Brothers **(H-422)**
New products
SM84 unidirectional condenser lavalier mic; FP51 4-input, 1-output gated compressor/mixer; BC70, -80 and -90 broadcast series phono cartridges; PDP 1000 Compact Disc player for broadcast; SM15 head-worn condenser mic.
Product line
Compact Disc players; consoles; headphones, headsets and intercom systems; microphones and accessories; phono cartridges; audio tranformers.
Circle (290)

Simmons Electronics **(S-895)**
Not available at press time.
Circle (291)

Solid State Logic **(H-Gibson Suite)**
New products
G series studio computer; new EQ and input cards for 4000 E series consoles.
Product line

 Circle (53) on Rapid Facts Card

Console automation systems; consoles.
Circle (292) See ad page 108

Sonic Research Associates (S-916)
Not available at press time.
Circle (293)

Sonosax (S-815)
Not available at press time.
Circle (294)

Sony Professional Audio (S-822)
New products
MXP-2036 broadcast/video post-production console; MXP-3036VF recording/remixing console; MXBR-2009E stereo module; DAL-1000 digital limiter; new editing software for PCM-3202; new software enhancements for APR-5003 and ADS-3000.
Product line
Compact Disc players; console automation systems; consoles; disc-mastering systems; editing systems; limiters; microphones; tape machines.
Circle (295) See ad page 87

Soundcraft USA (H-Clinton Suite)
New products
624 addition to Series 600; TS12 FAME real-time automation system; Digitor tapeless audio editing system; 200 BVE video editing console; 8000 sound reinforcement console.
Product line
Consoles; tape machines.
Circle (296) See ad page 49

Sound Ideas (H-411A)
New products
Sound effects library on 50 CDs with more than 5,000 sound effects; sampler library on 6 CDs with 3,100 instruments.
Product line
Sound effects libraries.
Circle (297) See ad page 115

Soundmaster International (S-913*)
New products
Soundmaster Integrated Editing System.
Product line
Editing systems; time code synchronizers.
Circle (298)

Sound Technology (H-145*)
New products
Model 3000AR audio testing system; MSAT audio switching system.
Product line
Maintenance equipment; test and measuring equipment.
Circle (299) See ad page 79

Soundtracs (H-218*)
New products
ERIC mixing console; FME modular mixer; MRX console.
Product line
Console automation systems; consoles.
Circle (300) See ad page 55

Sound Workshop Professional Audio Products (H-450)
New products
Diskmix moving fader addition for 34C console; VD-3 video dubbing stage console; ADR/Foley console.
Product line
Consoles.
Circle (301)

Southworth Music Systems (S-897)
Not available at press time.
Circle (302)

Stanton Magnetics (H-510)
New products
30M/SR single cup or shoulder rest headphone; DSM-1 disco slip mat; SRS-215 stereo headphone; SRS-225, -245 and -265 headphones; ST-PRO stereo headphone.
Product line

Stanton, continued

Headphones. headsets and intercom systems; phonograph cartridges.
Circle (303) **See ad page 111**

Star Case **(S-955)**
*New products*___
Computer-aided design program for designing complete cases.
*Product line*___
Amplifiers; Compact Disc players; consoles; drum machines; editing systems; equalizers; equipment cases; loudspeakers; pitch shifters; phonograph turntables; racks and stands; recording tape; reverb devices; signal processors.
Circle (304)

Stramp USA **(S-892)**
Not available at press time.
Circle (305)

Strand Magnetic Tapes **(S-882*)**
*Product line*___
Recording tape.
Circle (306)

Studer Revox **(H-164*)**
*New products*___
A820-A 1-inch multichannel mastering recorder; A727 professional Compact Disc player; SC4008 system controller; 963 mixing console.
*Product line*___
Compact Disc players; consoles; headphones, headsets and intercom systems; loudspeakers; noise reduction systems; phonograph turntables; tape machines; time code synchronizers.
Circle (307) **See back cover**

Studio Master Systems **(S-919)**
*New products*___
Studio Master Plus console mixing logging system; Studio Master studio billing system; Track Master track sheet and label generation program; Outboard Master outboard equipment documentation program.
*Product line*___
Business software; consoles; VU/PPM meters.
Circle (308)

Sunkyong **(S-816*)**
Not available at press time.
Circle (309)

Symetrix **(S-933)**
*New products*___
511A noise reduction system.
*Product line*___
Amplifiers; compressors; equalizers; expanders; limiters; microphone pre-amps; noise gates; noise reduction systems; patchbay and jack panels; signal processors.
Circle (310) **See ad page 105**

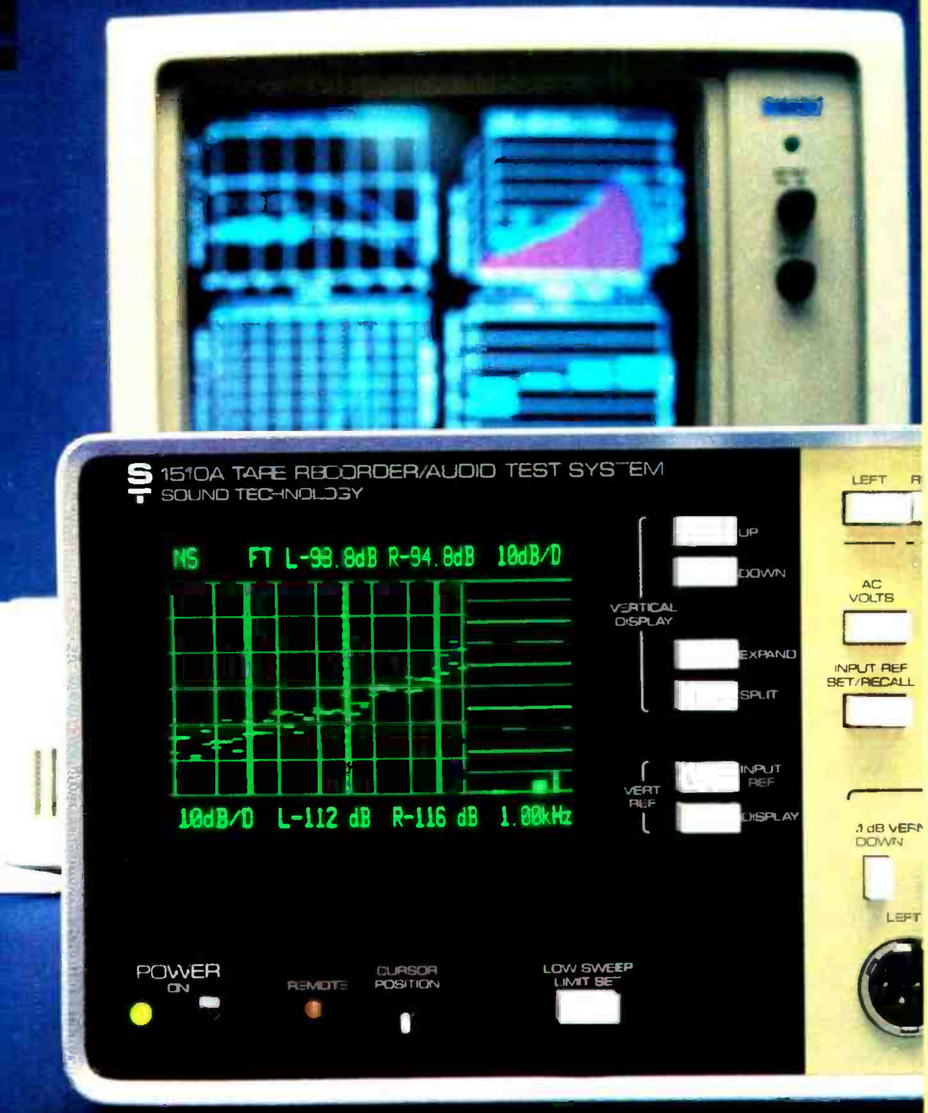
Tannoy Ltd. **(H-Rm. 622)**
*Product line*___
Amplifiers; Loudspeakers.
Circle (311)

Tape Automation **(S-948*)**
Not available at press time.
Circle (312)

Tascam **(H-433*)**
*New products*___
ATR 80 2-inch, 24-track recorder; ATR 60-16 16-track recorder; ES 50 SMPTE time code synchronizer controller; M-600 32-input, 16-bus recorder.
*Product line*___
Compact Disc players; consoles; equalizers; MIDI devices; tape duplication; tape machines; time code synchronizers.
Circle (313) **See ad page 7**

TC Electronic **(S-890*)**
*Product line*___
Signal processors.
Circle (314)

The best of both worlds:



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THE SOUND TECH 1510A AUDIO TEST SYSTEM

Until recently, most Audio Test Systems have been either manual stand-alone systems or external-computer driven automated systems.

Engineers have long enjoyed the portability, ease of operation and cost effectiveness associated with manual stand-alone systems. Unfortunately, these systems have always lacked speed and documentation capabilities.

On the other hand, the ideally configured external-computer driven test system can provide speed, data analysis, documentation, graphics and integration with other GPIB test systems. Unfortunately external-computer driven systems are designed for production testing and are not suitable for troubleshooting or field work.

However, Sound Technology has combined the Best of Both Worlds into a portable, intelligent stand-alone system with complete PC compatibility: the Model 1510A! The Model 1510A can be used as an intelligent stand-alone system, and when connected to an external computer it becomes the ideal Automated Test System.

The ST1510A as a Stand-alone System

- Built-in Intelligence
- Portable
- Semi-automatic and Manual Test modes for fast troubleshooting
- Built-in CRT/Graphics

The ST1510A as an Automated Test System

- Graphics
- Test Chaining/limit testing
- Industry-standard Computer Interface (GPIB)
- User Choice of PC
- Bundled software for IBM, HP
- Compatible with Automated Switchers
- Production Testing

Contact us now for full technical and applications information. Learn firsthand why the ST1510A is the only test system that combines the best of both worlds!

SOUND TECHNOLOGY
1400 Dell Avenue
Campbell, California 95008
(408) 378-6540 Telex: 357445

Put The Best To Test!

AES Booth 145 & 146

Circle (41) on Rapid Facts Card

Technical Audio Devices **(H-Rm. 537)**
Product line
Loudspeakers and components.
Circle (315)

Technics **(S-803)**
New products
SL-P1200X professional Compact Disc player.
Product line
Amplifiers; Compact Disc players; loudspeakers; phonograph turntables.
Circle (316)

Technos **(H-Rm. 543)**
Not available at press time.
Circle (317)

Telex Communications **(H-401*)**
New products
6120 XLP high-speed in-cassette duplicator; MagnaByte electronic imaging devices for Apple or IBM.
Product line
Headphones, headsets and intercom systems; microphones; tape duplication.
Circle (318) **See ad page 63**

3M Company **(H-101*)**
New products
R-DAT cassette; AUD digital U-matic cassette; improved 350 Cenetrak magnetic film; Pro II chrome bias audiocassette; recording reels.
Product line
Recording tape.
Circle (319) **See inside cover, 1**

Timeline **(S-854*)**
New products
Lynx systems controller; Lynx film interface module.
Product line
Time code synchronizers.
Circle (338)

Toolex **(H-929*)**
Not available at press time.
Circle (321)

Trident Audio USA **(H-134*)**
New products
DI-AN digitally controlled analog recording console; 80C console with 48-track capability; Trident 24 recording/post-production console.
Product line
Consoles.
Circle (322)

Troisi **(S-989)**
Not available at press time.
Circle (323)

Turbosound **(H-225*)**
New products
TFM-2 floor monitor; V-2 HF manifold; TSE-211 mid/hi enclosure; TSE-115 and -215 bass enclosures; TSE-260 HF enclosure with V-2.
Product line
Loudspeakers; test and measuring equipment.
Circle (324)

27th Dimension **(S-806)**
New products
Holophonic sound effects library.
Product line
Music product libraries.
Circle (325) **See ad page 112**

UREI **(H-Clinton Suite)**
New products
6210 accessory power amp; 6211 accessory power amp with mic pre-amp.
Product line
Amplifiers; crossovers and frequency dividers; delays; equalizers; loudspeakers; signal processors.
Circle (326) **See ad page 5**

VCL Audio **(S-854*)**
Product line
Consoles.
Circle (328)

Voyetra Technologies (S-940)
New products
Seq+ Mk I, II and III software; Conversion+ software; Sideman DTX software; OP-4001 and Op-4000 MIDI interfaces; PatchMaster+ software.
Product line
MIDI divices; synthesizers and keyboards; software.
Circle (329)

WaveFrame Corporation (H-Rm. 529)
New products
AudioFrame digital audio workstation.
Product line
Editing systems; sound samplers; synthesizers and keyboards.
Circle (330)

Westlake Audio (H-Rm. 620)
Not available at press time.
Circle (331) See ad page 74

Whirlwind (H-139)
New products
TEST-1 cable tester; IMCOM direct interface; MASSCON 176-pin connector.
Product line
Cable and connectors; patchbay and jack panels; audio transformers.
Circle (332)

Wireworks Corporation (H-204)
New products
CR1207 cable reel; CR1808 cable reel with larger capacity; T series rack adapters for individual mic splitter boxes, direct boxes, amplifier input boxes.
Product line
Cable and connectors.
Circle (333)

Xedit (H-155)
Not available at press time.
Circle (334)

Yamaha (H-Madison Suite)
New products
DMP7 digital mixing processor; REV 5 digital reverb; REX 50 digital multieffects processor; C300 stereo cassette deck; MT2X multitrack cassette recorder; NS10H monitor loudspeaker system; MZ203 and -204 dynamic vocal and instrumental microphone; MZ205Be dynamic instrumental mic; MC series sound reinforcement mixing cosoles.
Product line
Amplifiers; compressors; consoles; crossovers and frequency dividers; equalizers; loudspeakers; microphones and accessories; MIDI devices; reverb devices; signal processors; tape machines.
Circle (335) See ad pages 80-81

Zimbelman Ltd. (H-Rm. 624)
New products
ISA 110 microphone amplifier/equalizer module: ISA 115HD dual-channel microphone amplifier/equalizer module; ISA 113 microphone amplifier; ISA 116 remote-control microphone amplifier.
Product line
Compressors; consoles; equalizers; limiters; microphone pre-amplifiers; noise gates; signal processors.
Circle (336)

R·E/P

Product Directory

This directory of products to be exhibited at the AES Convention lists the companies that manufacture equipment or offer services in 68 categories.

To find the companies that make a product that you are interested in, find the appropriate category printed in bold type. The companies are listed in alphabetical order, along with their booth numbers.

Because companies are exhibiting in two venues, each entry has an H or an S preceding the booth number—H for the Hilton Hotel, and S for the Sheraton Centre. To save space, companies that were assigned more than one booth number are listed only with the first booth number of their space. These companies have an asterisk following this number.

Information on new and established products for the companies listed in the directory is available in the exhibitor listings.

Because of our press deadlines, new exhibitors or booth changes from AES may not be included. Be sure and check your convention program for updated information.

Acoustic design and construction
Alpha Audio (H-112*)
RPG Diffusor Systems (H-314)

Acoustic treatment materials
Alpha Audio (H-112*)
FM Tubecraft (S-917)
Penn Fabrication (S-846*)
RPG Diffusor Systems (H-314)

Amplifiers
API Audio Products (S-893)
Audio Media Research (H-Rm. 550)
Beyer Dynamic (H-507A)
BGW Systems (H-209)
Bryston Ltd. (H-130)
Carver (S-881)
Cetec Ivie (H-130)
Crest Audio (H-147*)
Crown International (H-504*)
C-T Audio Manufacturing (S-943)
Philip Drake Electronics (S-925*)
FM Acoustics (H-237A)
David Hafler Co. (H-243)
Innovative Electronic Designs (H-450A)
JBL Professional (H-Clinton Suite)
Lenco (H-311, Rm. 549)
Meyer Sound Labs (H-128*)
Mitsubishi Pro Audio (H-733, Bryant Suite)
Peavey Electronics (H-Rm. 551)
QSC Audio Products (H-244*)
Panasonic/RAMSA (H-212*, Rm. 510)
Renkus-Heinz (H-501)
Star Case (S-955)
Studer Revox (H-164*)
Symetrix (S-933)
Tannoy Ltd. (H-Rm. 622)
Technics (S-803)
UREI (H-Clinton Suite)
Yamaha (H-Madison Suite)

Cable and/or connectors
Audio Media Research (H-Rm. 550)
Audio-Technica (H-191*)
Audiotechniques (H-513*)
Beyer Dynamic (H-507A)
Cal Switch (H-235A*)
Canare Cable (S-802)
FM Acoustics (H-237A)
Marshall Electronics (S-808)
Monster Cable (H-137*)
Neutrik/Dialight (S-928)
Peavey Electronics (H-Rm. 551)
Penn Fabrication (S-846*)
Pro Co Sound (S-922)
Whirlwind (H-139*)
Wireworks (H-204)

Compact Disc players
Shure Brothers (H-422*)
Sony (S-822*)
Star Case (S-955)
Studer Revox (H-164*)
Tascam (H-433*)
Technics (S-803)

Compressors
Alesis (S-894)
API Audio Products (S-893)
Audio Developments (H-160*)
Audio Media Research (H-Rm. 550)
Audiotechniques (H-513*)
Cetec Ivie (H-126)
dbx (H-dbx Suite)
DOD Electronics (H-456*)
Dorrough Electronics (S-804)
Innovative Electronic Designs (H-450A)
Mitsubishi Pro Audio (H-Bryant Suite)
Orban (H-415*)
Symetrix (S-933)
Yamaha (H-Madison Suite)
Zimbelman (H-Rm. 624)

Consoles, automation systems
Allen & Heath Brenell (H-409*)
API Audio Products (S-893)
Clarity (S-863)
CMX (S-812*)
J.L. Cooper Electronics (S-807)
DDA (H-221*)
Digital Creations (H-318)
Harrison Systems (H-428*)
Klark-Teknik (H-221*)
Mitsubishi Pro Audio (H-Bryant Suite)
Rupert Neve (H-Morgan Suite)
Solid State Logic (H-Gibson Suite)
Sony (S-822*)
Soundtracs (H-207*)
Studer Revox (H-164*)
Studio Master Systems (S-919)

Consoles, digital
Amek Consoles/TAC (H-131*)
Analog Digital Synergy (H-325*)
Gotham Audio (H-161*)

Rupert Neve (H-Morgan Suite)
Star Case (S-955)
Yamaha (H-Madison Suite)

Consoles, faders
API Audio Products (S-893)
Digital Creations (H-318)
Harrison Systems (H-428*)
Mitsubishi Pro Audio (H-Bryant Suite)
Rupert Neve (H-Morgan Suite)
Soundtracs (H-207*)
Star Case (S-955)

Consoles, portable
ACO Pacific (H-515)
Allen & Heath Brenell (H-409*)
Amek Consoles/TAC (H-131*)
API Audio Products (S-893)
Audio Developments (H-160*)
Audio Media Research (H-Rm. 550)
DDA (H-221*)
Design Direct Sound (H-408A)
Fostex (H-Rm. 524*)
Gotham Audio (H-161*)
Harrison Systems (H-428*)
Klark-Teknik (H-221*)
Neotek (H-404*)
Rupert Neve (H-Morgan Suite)
Peavey Electronics (H-Rm. 551)
Panasonic/RAMSA (H-212*, Rm. 510)
Shure Brothers (H-422*)
Soundtracs (H-207*)
Star Case (S-955)
Studer Revox (H-164*)
Yamaha (H-Madison Suite)

Consoles, recording and production
Akai/IMC (H-Rm. 512*)
Allen & Heath Brenell (H-409*)
Amek Consoles/TAC (H-131*)
API Audio Products (S-893)
Audio Developments (H-160*)
Audio Media Research (H-Rm. 550)
DDA (H-221*)
Design Direct Sound (H-408A)
DOD Electronics (H-456*)
Fostex (H-Rm. 524*)
Harrison Systems (H-428*)
Howe Technologies (S-912)
Klark-Teknik (H-221*)
Mitsubishi Pro Audio (H-Bryant Suite)
Neotek (H-404*)
Rupert Neve (H-Morgan Suite)
Panasonic/RAMSA (H-Rm. 510)
Solid State Logic (H-Gibson Suite)
Sony (S-822*)
Soundcraft (H-Clinton Suite)
Soundtracs (H-207*)
Sound Workshop Professional Audio Products (H-450)
Star Case (S-955)
Studer Revox (H-164*)
Tascam (H-433*)
Trident Audio USA (H-134*)
Yamaha (H-Madison Suite)
Zimbelman (H-Rm. 624)

Consoles, sound reinforcement
Allen & Heath Brenell (H-409*)
Amek Consoles/TAC (H-131*)
Beyer Dynamic (H-507A)
DDA (H-221*)
Design Direct Sound (H-408A)
Harrison Systems (H-428*)
Klark-Teknik (H-221*)
Mitsubishi Pro Audio (H-Bryant Suite)
Neotek (H-404*)
Rupert Neve (H-Morgan Suite)
Peavey Electronics (H-Rm. 551)
Panasonic/RAMSA (H-212*, Rm. 510)
Soundcraft (H-Clinton Suite)
Soundtracs (H-207*)
Star Case (S-955)
Studer Revox (H-164*)
Tascam (H-433*)
Trident Audio USA (H-134*)
Yamaha (H-Madison Suite)

Crossovers and frequency dividers
Amek Consoles/TAC (H-131*)
Beyer Dynamic (H-507A)
Bryston Ltd. (H-130)
BSS (H-221*)
Carver (S-881)

Crossovers and frequency dividers, continued

Cetec Ivie (H-126)
DOD Electronics (H-456*)
FM Acoustics (H-237A)
Fostex (H-Rm. 524*)
Gold Line (H-211)
JBL Professional (H-Clinton Suite)
Klark-Teknik (H-221*)
Peavey Electronics (H-Rm. 551)
Panasonic/RAMSA (H-212*, Rm. 510)
UREI (H-Clinton Suite)
Yamaha (H-Madison Suite)

Delays
ART-Applied Research & Technology (H-210)
Audio/Digital (H-305)
Audio Media Research (H-Rm. 550)
Cetec Ivie (H-126)
DOD Electronics (H-456*)
Eventide (H-455*)
Gotham Audio (H-161)
Innovative Electronic Designs (H-450A)
JBL Professional (H-Clinton Suite)
Klark-Teknik (H-221*)
Korg USA (H-Rm. 534)
Mitsubishi Pro Audio (H-Bryant Suite)
Peavey Electronics (H-Rm. 551)
UREI (H-Clinton Suite)

Disc-mastering systems
API Audio Products (S-893)
Gotham Audio (H-161*)
Optical Disc Corporation (S-848*)
Sony (S-822*)

Drum machines
Akai/IMC (H-Rm. 512*)
Alesis (S-894)
Korg USA (H-Rm. 534)
Star Case (S-955)

Editing systems
Adams-Smith (H-227*)
Alpha Audio (H-112*)
Audio Developments (H-160*)
Cipher Digital (H-322*)
CMX (S-812*)
Mitsubishi Pro Audio (H-Bryant Suite)
Sony (S-822*)
Soundmaster International (S-913*)
Star Case (S-955)
WaveFrame (H-Rm. 529)

Equalizers, graphic
Akai/IMC (H-Rm. 512*)
API Audio Products (S-893)
ART-Applied Research & Technology (H-210)
Cetec Ivie (H-126)
dbx (H-dbx Suite)
DOD Electronics (H-456*)
Fostex (H-Rm. 524*)
JBL Professional (H-Clinton Suite)
Klark-Teknik (H-221*)
Micro Audio (S-921)
Mitsubishi Pro Audio (H-Bryant Suite)
Oxmoor (S-939)
Peavey Electronics (H-Rm. 551)
Star Case (S-955)
Tascam (H-433*)
UREI (H-Clinton Suite)
Yamaha (H-Madison Suite)

Equalizers, parametric
Aphex Systems (H-202*)
API Audio Products (S-893)
Audio Media Research (H-Rm. 550)
Audiotechniques (H-513*)
Beyer Dynamic (H-507A)
Cetec Ivie (H-126)
DOD Electronics (H-456*)
Gotham Audio (H-161*)
Klark-Teknik (H-221*)
Meyer Sound (H-128*)
Mitsubishi Pro Audio (H-Bryant Suite)
Star Case (S-955)
Symetrix (S-933)
Tascam (H-433*)
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Equipment cases
Calzone (H-411)
Peavey Electronics (H-Rm. 551)
Star Case (S-955)

Equipment distributors
Audiotechniques (H-513*)
Professional Audio Services and Supply (H-315)
Schoeps/Posthorn Recordings (S-884)

Equipment rental and leasing
Schoeps/Posthorn Recordings (S-884)

Expanders
DOD Electronics (H-456*)
Mitsubishi Pro Audio (H-Bryant Suite)
Symetrix (S-933)

Faders
API Audio Products (S-893)
Mitsubishi Pro Audio (H-Bryant Suite)
Oxmoor (S-939)

Headphones, headsets and intercom systems
Audio-Technica (H-191*)
Beyer Dynamic (H-507A)
Cetec Vega (H-127)
Countryman Associates (H-234A)
Philip Drake Electronics (S-925*)
Fostex (H-Rm. 524*)
Korg USA (H-Rm. 534)
Nakamichi America (S-811)
ROH (H-208)
Shure Brothers (H-422*)
Stanton Magnetics (H-510)
Studer Revox (H-164*)
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Libraries, music production
Fairlight Instruments (H-517*)
27th Dimension (S-806)

Libraries, sound effects
Leonardo Software (S-888B*)
Sound Ideas (H-411A)

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Gold Line (H-211)
Penn Fabrication (S-846*)

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Alesis (S-894)
Aphex Systems (H-202*)
API Audio Products (S-893)
Audio Developments (H-160*)
Audio Media Research (H-Rm. 550)
Beyer Dynamic (H-507A)
BSS (H-221*)
Cetec Ivie (H-126)
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DOD Electronics (H-456*)
Dorrough Electronics (S-804)
Gold Line (H-211)
Gotham Audio (H-161*)
Klark-Teknik (H-221*)
Mitsubishi Pro Audio (H-Bryant Suite)
Orban (H-415*)
Sony (S-822*)
Symetrix (S-933)
Zimbelman (H-Rm. 624)

Loudspeakers, component
Audio-Technica (H-191*)
Celestion Industries (S-866*)
Cetec Gauss (H-119*)
Community Light & Sound (S-888A)
Design Direct Sound (H-408A)
Eastern Acoustic Works (H-190*)
Fostex (H-Rm. 524*)
JBL Professional (H-Clinton Suite)
Neutrik/Dialight (S-928)
Peavey Electronics (H-Rm. 551)
Renkus-Heinz (H-501)
Tannoy Ltd. (H-Rm. 622)
Technics (S-803)
Turbosound (H-225*)

Loudspeakers, sound reinforcement
Apogee Sound (S-941*)
Celestion Industries (S-866*)
Cetec Gauss (H-119*)
Community Light & Sound (S-888A)
C-T Audio Marketing (S-943)
Design Direct Sound (H-408A)
Eastern Acoustic Works (H-190*)
Fostex (H-Rm. 524*)
JBL Professional (H-Clinton Suite)
Meyer Sound Labs (H-128*)
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Panasonic/RAMSA (H-212*, Rm. 510)
Renkus-Heinz (H-501)
Star Case (S-955)
Tannoy Ltd. (H-Rm. 622)
Turbosound (H-225*)
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Eastern Acoustic Works (H-190*)
Fostex (H-Rm. 524*)
Gotham Audio (H-161*)
JBL Professional (H-Clinton Suite)
Klark-Teknik (H-221*)

Meyer Sound Labs (H-128*)
Star Case (S-955)
Studer Revox (H-164*)
UREI (H-Clinton Suite)
Yamaha (H-Madison Suite)

Maintenance equipment
BSS (H-221*)
Klark-Teknik (H-221*)
Metro Audio (H-208)
Sound Technology (H-145*)

Microphones, accessories
ACO Pacific (H-515)
AKG Acoustics (H-216*)
Audio Media Research (H-Rm. 550)
Audio-Technica (H-191*)
Beyer Dynamic (H-507A)
Cal Switch (H-235A*)
Countryman Associates (H-234A)
Fostex (H-Rm. 524*)
Gotham Audio (H-161*)
IQS (S-910)
Neutrik/Dialight (S-928)
Peavey Electronics (H-Rm. 551)
Sanken Microphone (S-810)
Schoeps/Posthorn Recordings (S-884)
Shure Brothers (H-422*)
Yamaha (H-Madison Suite)

Microphones, booms and stands
AKG Acoustics (H-216*)
Audio-Technica (H-191*)
Beyer Dynamic (H-507A)
Gotham Audio (H-161*)
Peavey Electronics (H-Rm. 551)
Panasonic/RAMSA (H-212*, Rm. 510)
Schoeps/Posthorn Recordings (S-884)
Shure Brothers (H-422*)

Microphones, pre-amplifiers
ACO Pacific (H-515)
API Audio Products (S-893)
Beyer Dynamic (H-507A)
Cetec Ivie (H-126)
C-T Audio Marketing (S-943)
DOD Electronics (H-456*)
Gotham Audio (H-161*)
Schoeps/Posthorn Recordings (S-884)
Shure Brothers (H-422*)
Symetrix (S-933)
Zimbelman (H-Rm. 624)

Microphones, studio and PA
Audio Media Research (H-Rm. 550)
Audio-Technica (H-191*)
Beyer Dynamic (H-507A)
Bruel & Kjaer (H-507*)
Countryman Associates (H-234A)
Crown International (H-504*)
C-T Audio Marketing (S-943)
Explorations (H-321)
Fostex (H-Rm. 524*)
Gotham Audio (H-161*)
Peavey Electronics (H-Rm. 551)
Panasonic/RAMSA (H-212*, Rm. 510)
Sanken Microphone (S-810)
Schoeps/Posthorn Recordings (S-884)
Sennheiser (H-502*)
Shure Brothers (H-422*)
Telex Communications (H-401A*)
Yamaha (H-Madison Suite)

Microphones, wireless and RF
AKG Acoustics (H-216*)
Beyer Dynamic (H-507A)
Cetec Vega (H-127)
Countryman Associates (H-234A)
Peavey Electronics (H-Rm. 551)
Samson Technologies (H-312*)
Schoeps/Posthorn Recordings (S-884)
Shure Brothers (H-422*)
Sony (S-822*)
Telex Communications (H-401A*)

MIDI devices, add-on
Akai/IMC (H-Rm. 512*)
Allen & Heath Brenell (H-409*)

MIDI devices, continued

Audio Media Research (H-Rm. 550)
Audio-Technica (H-191*)
Clarity (S-863)
J.L. Cooper Electronics (S-807)
DOD Electronics (H-456*)
Eventide (H-455*)
Fostex (H-Rm. 524*)
Klark-Teknik (H-221*)
Peavey Electronics (H-Rm. 551)
Roland (H-616*)
Tascam (H-433*)
Yamaha (H-Madison Suite)

Noise gates
Alesis (S-894)
Aphex Systems (H-202*)
Audio Media Research (H-Rm. 550)
BSS (H-221*)
C-T Audio Marketing (S-943)
dbx (H-dbx Suite)
DOD Electronics (H-456*)
Gold Line (H-211)
Klark-Teknik (H-221*)
Mitsubishi Pro Audio (H-Bryant Suite)
Symetrix (S-933)
Zimbelman (H-Rm. 624)

Noise reduction systems
ANT Telecommunications (H-459*)
dbx (H-dbx Suite)
DOD Electronics (H-456*)
Dolby Laboratories (H-141*)
Gotham Audio (H-161*)
Studer Revox (H-164*)
Symetrix (S-933)

Patchbay and jack panels
Akai/IMC (H-Rm. 512*)
Allen & Heath Brenell (H-409*)
Cal Switch (H-235A*)
Canare Cable (S-802)
Philip Drake Electronics (S-925*)
Fostex (H-Rm. 524*)
Marshall Electronics (S-808)
Mosses & Mitchell (H-406)
Neutrik/Dialight (S-928)
Pro Co Sound (S-922)
Symetrix (S-933)
Whirlwind (H-139*)

Peripheral and miscellaneous accessories
Audico (H-452)
Audio Media Research (H-Rm. 550)
Audio-Technica (H-191*)
Audiotechniques (H-513*)
Audio Video Consultants (H-301*)
Beyer Dynamic (H-507A)
BSS (H-221*)
Clarity (S-863)
Fostex (H-Rm. 524*)
Klark-Teknik (H-221*)
Lenco (H-311, Rm. 549)
Marshall Electronics (H-808)
Metro Audio (H-208)
Mitsubishi Pro Audio (H-Bryant Suite)
Peavey Electronics (H-Rm. 551)
Penn Fabrication (S-846*)
Pro Co Sound (S-922)
Shape Inc. (S-850*)

Phasers
DOD Electronics (H-456*)

Pitch shifters
ART-Applied Research & Technology (H-210)
DOD Electronics (H-456*)
Eventide (H-455*)
Star Case (S-955)

Phonograph cartridges
Audio-Technica (H-191*)
Gotham Audio (H-161*)
Monster Cable (H-137*)
Shure Brothers (H-422*)
Stanton Magnetics (H-510)

Phonograph turntables
Gotham Audio (H-161*)
Star Case (S-955)
Studer Revox (H-164*)
Technics (S-803)

Racks and stands
Beyer Dynamic (H-507A)
C-T Audio Marketing (S-943)
FM Tubecraft (S-917)
KABA Research and Development (S-872)
Marshall Electronics (S-808)
Peavey Electronics (H-Rm. 551)
Star Case (S-955)

Recording tape, audio
Ampex (H-107*)
BASF (H-158*)
Fostex (H-Rm. 524*)
Magnefax International (H-127A)
Nakamichi America (S-811)
Shape Inc. (S-850*)
Star Case (S-955)
3M Company (H-101*)

Recording tape, U-matic
Ampex (H-107*)
Star Case (S-955)
3M Company (H-101*)

Reverb devices
Alesis (S-894)
ART-Applied Research & Technology (H-210)
Audio Media Research (H-Rm. 550)
Clarity (S-863)
DOD Electronics (H-456*)
Eventide (H-455*)
Gotham Audio (H-161*)
Klark-Teknik (H-221*)
Korg USA (H-Rm. 534)
Orban (H-415*)
Star Case (S-955)
Yamaha (H-Madison Suite)

Signal processors
Akai/IMC (H-Rm. 512*)
Alesis (S-894)
Aphex Systems (H-202*)
API Audio Products (S-893)
ART-Applied Research & Technology
Audio/Digital (H-305)
Audio Media Research (H-Rm. 550)
Beyer Dynamic (H-507A)
BSS (H-221*)
Cetec Ivie (H-126)
Clarity (S-863)
dbx (H-dbx Suite)
DOD Electronics (H-456*)
Dolby Laboratories (H-141)
Dorrough Electronics (S-804)
Eventide (H-455*)
Gold Line (H-211)
Gotham Audio (H-161*)
Howe Technology (S-912)
Innovative Electronic Design (H-450A)
IQS (S-910)
JBL Professional (H-Clinton Suite)
Klark-Teknik (H-221*)
Korg USA (H-Rm. 534)
Mitsubishi Pro Audio (H-Bryant Suite)
Orban (H-415*)
Oxmoor (S-939)
Peavey Electronics (H-Rm. 551)
Roland (H-616*)
Star Case (S-955)
Symetrix (S-933)
UREI (H-Clinton Suite)
Yamaha (H-Madison Suite)
Zimbelman (H-Rm. 624)

Software, business and studio applications
Leonardo Software (S-888B*)
Studio Master Systems (S-919)

Sound Samplers
Akai/IMC (H-Rm. 512*)
E-mu Systems (H-Rm. 542)
Fairlight Instruments (H-Rm. 517*)

Sound samplers, continued

Korg USA (H-Rm. 534)
Peavey Electronics (H-Rm. 551)
Roland (H-616*)
WaveFrame (H-Rm. 529)

Synthesizers and keyboards
Akai/IMC (H-Rm. 512*)
E-mu Systems (H-Rm. 542)
Fairlight Instruments (H-517*)
Korg USA (H-Rm. 534)
Kurzweil Music Systems (H-Rm. 504)
New England Digital (H-Nassau Suite A & B)
Roland (H-616*)
WaveFrame (H-Rm. 529)

Tape duplication, cassette
Agfa-Gevaert (H-169*)
Ampex (H-107*)
Apex Machine Co. (S-809*)
Electro Sound (H-246*)
KABA Research and Development (S-872)
King Instrument (H-122*)
Magnefax International (H-127A)
Nakamichi America (S-811)
Otari (H-424*)
Tascam (H-433*)
Telex Communications (H-401A*)

Tape duplication, reel to reel
Agfa-Gevaert (H-169*)
Ampex (H-107*)
Otari (H-424*)
Telex Communications (H-401A*)

Tape machines, analog
Akai/IMC (H-Rm. 512*)
Fostex (H-Rm. 524*)
Gotham Audio (H-161*)
Otari (H-424*)
Sony (S-822*)
Soundcraft (H-Clinton Suite)
Studer Revox (H-164*)
Tascam (H-433*)

Tape machines, cartridge
Otari (H-424*)

Tape machines, cassette
American Multimedia/Concept Design (H-319*)
Audio Media Research (H-Rm. 550)
Fostex (H-Rm. 524*)
Nakamichi America (S-811)
Shape Inc. (S-850*)
Studer Revox (H-164*)
Tascam (H-433*)
Yamaha (H-717*, Madison Suite)

Tape machines, digital
Apogee Electronics (S-879)
Mitsubishi Pro Audio (H-733*, Bryant Suite)
Otari (H-424*)
Sony (S-822*)
Studer Revox (H-164*)

Tape machines, mag film
Mitsubishi Pro Audio (H-Bryant Suite)

Tape machines, replacement heads
JRF Magnetic Sciences (H-111)
Saki Magnetics (H-408)

Test and measuring equipment
Amber Electro Design (H-206)
Apogee Electronics (S-879)
Audico (H-452)
Audio Precision (H-306*)
Beyer Dynamic (H-507A)
Bruel & Kjaer (H-507*)
Cal Switch (H-235A*)
Cetec Ivie (H-126)
dbx (H-dbx Suite)
Electro Sound (H-246*)
Gotham Audio (H-161*)
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KURZWEIL™
Music Systems

RANDOM SAMPLING

Volume I | Notes & News from Kurzweil Music Systems | Issue No. 4

KURZWEIL 250™ AT USC..."**Seminar in Electronic Music, Computers, and MIDI,**" sponsored by the **USC School of Music,** included lectures by **Clark Spangler** and **David S. Mash** and hands-on instruction on the **250. SPANGLER,** internationally-known synthesist, talked about **digital synthesis and sampling** adding, "The 250 is one of the great orchestration tools. You've got the freedom to do what you like to do." **MASH SAYS**... "I spoke almost exclusively about the 250 and demonstrated one. It excels for composers, arrangers, and orchestrators." Mash, who's **Chairman of the Music Synthesis Department of Berklee College of Music,** wrote the new **Kurzweil 250 User's Guide,** so he definitely knows the 250. **250 USER'S GUIDE REVISITED**...Still don't have one? See your Kurzweil dealer. **HAVE A MODEM?** Dial into the Kurzweil User's Group on **PAN,** a VideoText network that you can access with your Macintosh,™ a terminal program and a modem. **You're just a phone call away from the factory and from other Kurzweil users.** Swap sounds, trade information and more. To sign up, refer to your manual or write to us. **IF YOU'RE REALLY INTO MIDI**...The **Kurzweil MIDIBOARD™** is the ultimate studio and performance controller. 88 weighted wooden keys give you the feel of an acoustic piano, with **attack and release velocity plus mono- or polyphonic afterpressure.** Program it to transmit on eight MIDI channels. **Play eight different patches on eight different MIDI modules...simultaneously!** Arrange your setups to switch from module to module with

one touch of a button or pedal switch. Front panel sliders give you real-time control over the MIDIBOARD's response. Twelve assignable controllers let you modulate any function your synth can receive over MIDI. **SHORT ON MEMORY? SUPERAM I and II** expand the **KURZWEIL 250's** sampling memory to one or two megasamples. With **SUPERAM II,** you can have four banks of up to 100 seconds sampling time each, depending on sampling rate. **Load four times as many diskette-based sounds into RAM** for immediate access. And remember, the 250's sampling memory is completely separate RAM...all samples can be used together and in layers with the **KURZWEIL 250's** preset sounds in ROM. **150FS CONTINUED**...Did you get the free **Version 1.6 software upgrade?** Or the free **Voice Block B upgrade?** If you didn't, see your Kurzweil dealer. QUESTIONS?...**Write to us at: Kurzweil Music Systems, Inc., 411 Waverley Oaks Road, Waltham, MA 02154. Over.**

Circle (45) on Rapid Facts Card

Workshops, Papers and Special Events

Aside from the exhibits, a variety of other activities are scheduled during the convention, including an awards banquet, technical tours, and workshops and papers. The listing below gives basic information; final information will be available at the show.

Awards banquet

The annual awards banquet will be Sunday, Oct. 18. A pre-dinner reception begins at 7 p.m., with dinner following at 8 p.m. After an awards ceremony, entertainment will be provided by Broadway Limited.

Tours

Three technical tours have been planned, providing a glimpse into real-world situations that cannot be duplicated on a convention floor. Registration for all three tours can be made at the registration desks starting at 3 p.m. on Thursday, Oct. 15.

Tour No. 1
Friday, Oct. 16
1 p.m.

Kaufman-Astoria Studios, Astoria, Queens, is one of the most famous production stages in the world, starting with its beginning in the 1920s as the site of many famous motion pictures to its recent role in a digital overdub via satellite (see the June **RE/P**).

Tour No. 2
Saturday, Oct. 17
10:30 a.m.

This tour will feature a behind-the-scenes look at Radio City Music Hall, said to be one of the most flexible performance complexes in the world.

Tour No. 3
Monday, Oct. 19
10:30 a.m.

The Rodgers and Hammerstein Recording Archive at Lincoln Center houses more than 500,000 recorded items of historical significance, and this tour will feature how items are catalogued and stored.

Papers and workshops

During the convention's four days, a total of 96 papers and 10 workshops will cover the entire spectrum of audio production. At press time, final arrangements were being made, and some details may have changed by convention time. For final details, check the convention schedule.

Workshops

Dates and times for the 10 workshops listed below were not available at press time.

- "Disc-based Audio Editing," Bill Foster, Tape One, London.
- "User Interfaces for Electronic Music," Dr. William Bixton, University of Toronto, Toronto.
- "Techniques for Subjective Listening Evaluation," David Clark, DLC Design, Farmington Hills, MI.
- "Sound Reinforcement Workshop," Dave Kaye, Boston.
- "Pre-emphasis and De-emphasis in Digital Recording," Albert Grundy, Institute for Audio Research, New York.
- Education Workshop, AES Education Committee, Martin Polon, chairman.
- "Management in Recording Studio Operations," SPARS workshop.
- "Motion Picture Sound," John Allen, New York.
- "Tape Recorder Maintenance," Gregg Hankes, NY Technical Support, Chappaqua, NY.

Papers

A total of 96 papers will be presented in 15 technical sessions. At press time, names of the people presenting the papers were not available.

Session A
Friday, Oct. 16
9 a.m.-12:30 p.m.

"Advances in CD and DAT Multimedia," part 1, will be chaired by Ken Pohlman of the University of Miami School of Music, Coral Gables, FL. Seven papers will be presented:

- "Compact Disc Manufacturing, 'As the Bubble Bursts.' "
- "Techniques for the Quality Analysis of Compact Discs."
- "An Extension of the CD Mastering System Format for CD-ROM."
- "Multi-media Applications on CD-ROM: The Information Exchange Protocol (IXP) and Other Standardization Developments."
- "Compact Disc Video and Audio Engineering."
- "Compact Disc Video Signal Optimization."
- "The CD-I Authoring System."

1:30 p.m.-3:30 p.m.

Part 2 of "Advances in CD and DAT Multimedia" will contain four additional papers:

- "Optical Disc Mastering Technology: From Laservision to CDV and Beyond."
- "CD Mastering: Advancing the State of the Art."
- "Application of Oversampling A/D and D/A Conversion Technique to R-DAT."
- "R-DAT and Professional Audio."

Session B
Friday, Oct. 16
9 a.m.-12:30 p.m.

"Acoustics and Intelligibility" will be chaired by D.B. Keele Jr. of Techron, Elkhart, IN. Seven papers will be presented:
• "Time Response: Magnitude and Phase."
• "Sound Intensity and Interaural Cross Correlation Measurements Using Time Delay Spectrometry."
• "Investigating the Early Sound Field."
• "Decay Characteristics of Coupled Room Systems."
• "Environmental Effects on the Speed of Sound."
• "A Computationally Efficient Method of Predicting Speech Intelligibility."
• "Development of a New Algorithm for Predicting the Speech Intelligibility of Sound Systems in Rooms."

Session C
Friday, Oct. 16
1:30 p.m.-5:30 p.m.

"Psychoacoustics" will be chaired by D. Deutsch of the Department of Psychology at the University of California at San Diego. Seven papers will be presented:
• "On the Behavior of Listeners to Stereophonic Sound Reproduction and the Consequences for the Theory of Sound Perception in a Stereophonic Field."
• "Subjective Quality Assessment Methods . . . the Old International Standards are Changing."
• "Statistical Analysis of Double Blind Tests for Multiple Audiences."
• "Aural Acuity and the Meaning of Sound Quality: A Cultural Approach."
• "Perception of Synthesized Timbres: Approximations to Selected Targets, and Level-dependent Effects with Reproduction Systems."
• "The Perception and Measurement of Resonances in Audio Components."
• "Results of the 1986 AES Audiometric Survey."

Session D
Friday, Oct. 16
7 p.m.-10 p.m.

A technical council special session on transmission will be chaired by D. Gravereaux, New Caanan, CT. In addition to a technical committee meeting, one paper will be presented:
• "Electronic Distribution of Personalized Music in the Retail Environment."

Session E
Friday, Oct. 16
7 p.m.-10 p.m.

A technical council special session on acoustics and sound reinforcement will be chaired by K. Jacob, Bose Corp., Framingham, MA. In addition to a technical committee meeting, four papers will be presented:
• "Physical Measurements vs. Subjective Testing."
• "Choice of Sample Size in Listening Tests."
• "A Systematic Method for the Aural Analysis of Sound Sources in Audio Reproduction/Reinforcement, Communications and Musical Contexts."
• "NLSI—The Biointerferometric Paradigm for Binaural Acoustics."

Session F
Saturday, Oct. 17
9 a.m.-12:30 p.m.

"Recording and Playback Technology" will be chaired by L. Boden, Glendale, CA. Seven papers will be presented:
• "1888-1988: A Hundred Years of Magnetic Sound Recording."
• "Azimuth Measurement in Audio

Cassettes."
• "Azimuth Reference Housing for the Compact Cassette System."
• "Optical Playback Cartridge for Stereo LP Records."
• "Modular Digital Multitrack Recorder."
• "A New Audio Digital Filter with Compensation of Phase for A/D and D/A Conversion."
• "An 18-bit D/A Converter for High Performance Digital Audio Applications."

Session G
Saturday, Oct. 17
9 a.m.-12:30 p.m.

"Transducers," part 1, will be chaired by J. Vanderkooy of the University of Waterloo, Waterloo, Ontario, Canada. Six papers will be presented:
• "The Theory of Acoustic Waveguides."
• "The Application of an Inductively Coupled Shorted Turn and the Dual Coil Loudspeaker System."
• "Improved Hands-Free Microphone for Automotive Communications."
• "Subwoofer Performance for Accurate Reproduction of Music."
• "The Control of Sound Reradiation with Constant Sound Intensity Helicoids."
• "Power Transmission Through Crossover Networks."

2 p.m.-4:30 p.m.

Part 2 of "Transducers" will contain five additional papers:
• "A Loudspeaker Design for Reduced Reverberant Sound Power Output."
• "Anomalies of Wavefront Reconstruction in Stereo and Surround-Sound Reproduction."
• "Generalized Design Method of Lossy Passive-Radiator Loudspeaker Systems."
• "Problems Related to Military Specifications for Audio Transducers."
• "A Loudspeaker Motor Structure for Very High Power Handling and High Linear Excursion."

Session H
Saturday, Oct. 17
2 p.m.-5:30 p.m.

"Studio Digital Recording, Mixing and Editing" will be chaired by L. Boden of Glendale, CA. Seven papers will be presented:
• "Progress in Digital Audio."
• "Digital Time-Alignment Recording Techniques."
• "A Magnetic Storage Disk-Based Digital Audio Recording, Editing and Processing System."
• "Optimizing Audio Data Transfer from Direct Access Media."
• "Twin-DASH Stationary-Head 2-Channel Recording at 15 IPS."
• "An Overview of the PD (Prodigi) Format."
• "Discussion of the Technical and Operational Characteristics of the Two Digital Audio Multichannel Formats: DASH and Prodigi."

Session I
Saturday, Oct. 17
7 p.m.-10 p.m.

A technical council special session on signal processing will be chaired by D. Eger of Techron, Elkhart, IN. In addition to a technical committee meeting, three papers will be presented:
• "Signal Processing Applied to the Modification of Sound."
• "The Design of an Audio Limiter Using Digital Signal Processing."
• "Forensic Audio, Theory and Applications."

Session J
Saturday, Oct. 17

7 p.m.-10 p.m.

A technical council special session on transducers will be chaired by J. Bullock of Shure Brothers, Evanston, IL. In addition to a technical committee meeting, one paper will be presented:

- "Tutorial on Computer-Aided Design Techniques for Transducers."

Session K
Sunday, Oct. 18
9 a.m.-12:30 p.m.

"Analog and Digital Signal Processing" will be chaired by R. Adams of dbx, Newton, MA. Seven papers will be presented:

- "The Design of High Performance Voltage Controlled Equalizers."
- "An Ultra-Low Noise Monolithic Microphone Pre-Amplifier."
- "A Low Distortion Transimpedance Amplifier Realized with N-ch MOSFETS."
- "Variable Analog Filters for the Support of Variable Sampling Rate Digital Audio."
- "A Digital Signal Processing Algorithm for Digital Audio Disk Recording."
- "DSP Architectures for the Digital Audio Workstation."
- "A Digital Signal Processing System for Automatic Dialogue Post-Synchronization."

Session L
Monday, Oct. 19
9 a.m.-12:30 p.m.

"Sound System Engineering" will be chaired by Daniel Queen, Daniel Queen Associates, New York. Seven papers will be presented:

- "The Avery Fisher Center at NYU: Hi-Tech Comes to the Library and Education."
- "Acoustics and Sound System for the new Parliamentary Assembly Hall of the Palatinate in Mainz."
- "Public Address for Carnival Parades."
- "Large Multichannel Wireless Microphone Systems: Meeting the Need for 20 Channels in Theater Applications."
- "A TV Station Low-Noise, High-Quality Audio System."
- "Controlled Reflection Isolation Booth."
- "Sound System Engineering with Electronic Spreadsheets."

Session M
Sunday, Oct. 18
1:30 p.m.-5:30 p.m.

"Digital Signal Processing Chips for Music Synthesis, Recording and Processing" will be chaired by J. Strawn of S Systems, San Rafael, CA. In addition to a discussion and question-and-answer period at the end, six papers will be presented:

- Presentation on signal processing products by representatives of Texas Instruments, Motorola and Analog Devices.
- "Processing Music with the TMS-32020."
- "Design of a Professional Real-Time Signal Processor for Synthesis, Sampling, Mixing and Recording."
- "DSPs for Music at Toronto."
- "Using Digital Signal Processor Chips in Stereo Audio Time Compressor/Expander."
- "Hardware Design of the AudioFrame Digital Mixer and its Applications in Music."

Session N
Sunday, Oct. 18
1:30 p.m.-5:30 p.m.

"Audio Techniques in Film and Broadcasting" will be chaired by S. Lyman of CBC Engineering, Montreal. Eight papers will be presented:

- "Improvements in FMX Technology."
- "Audio Performance of Professional VTRs."
- "Improvement of Digital Audio in M-II

Format VTR."
- "Digital Connections in a Broadcast Studio Center."
- "Sampling-Frequency Synchronization with Minimal Delay."
- "Surround-Sound System for HDTV and Experimental Study on Production Technique."
- "An Ergonomically Designed Control and Display Concept for the Sound Mixing Process in Film."
- "Stereo Audio Television: Practical Problems in Audio Post-Production Techniques."

Session O
Monday, Oct. 19
9 a.m.-12 p.m.

"Audio Measurements and Evaluation," part 1, will be chaired by Richard Cabot of Audio Precision, Beaverton, OR. Six papers will be presented:
- "In Honor of Heyser (1931-1987)."
- "Loudspeaker and Acoustic Measurements Using Maximum-Length Sequences."
- "Measurement of Transducer Motional Impedance—An Update."
- "Time-Domain Measurements Simplified."
- "A New Windowing Technique for Digital Harmonic Distortion Measurement."
- "Measurement of R-DAT Playback Signal."

2 p.m.-4 p.m.

Part 2 of "Audio Measurement and Evaluation" will contain four additional papers:
- "A Musically Appropriate Dynamic Headroom Test for Power Amplifiers."
- "Automated Measurement of the Dynamic Characteristics of Compressors and Expanders."
- "A Different Approach to the Old Problem of Audio Level Monitoring."
- "A Dynamic Phase Meter for Program Material."

R·E/P

Product Directory
Test and measuring equipment, continued

Klark-Teknik (H-221*)
Lenco (H-311, Rm. 549)
Metro Audio (H-208)
Micro Audio (S-921)
RTI-Research Technology International (S-953)
Shape Inc. (S-850*)
Sound Technology (H-145*)
Turbosound (H-225*)

Time code synchronizers
Adams-Smith (H-227*)
Allen & Heath Brenell (H-409*)
Alpha Audio (H-112*)
Audio Media Research (H-Rm. 550)
Cipher Digital (H-322*)
J.L. Cooper Electronics (S-807)
Evertz Microsystems (S-868*)
Fostex (H-Rm. 524*)
Otari (H-424*)
Soundmaster International (S-913*)
Studer Revox (H-164*)
Tascam (H-433*)
Timeline (S-854)

Tranformers, audio
BSS (H-221*)
Klark-Teknik (H-221*)
Neutrik/Dialight (S-928)
Peavey Electronics (H-Rm. 551)
Shure Brothers (H-422*)
Whirlwind (H-139*)

VU/PPM meters
Apogee Electronics (S-879)
Audio Developments (H-160*)
Dorrough Electronics (S-804)
Gotham Audio (H-161*)
Schoeps/Posthorn Recordings (S-884)
Selco/Sifam (H-316)
Studio Master Systems (S-919)

R·E/P

Continued from page 12

Time code book available from Cipher Digital

The bibliography for "Time Code and Synchronization, Part 1," published in the August issue, incorrectly listed a mailing address for one of the books. The "Time Code Handbook" is available in a revised version written by the engineering staff of Cipher Digital.

The book is available from the company at Box 170, Frederick, MD 21701. The cost is $12.95 plus $3.50 shipping and handling.

AES selects educational grants recipients

The AES Educational Foundation has awarded graduate study grants to Richard Karstens, Northwestern; Vincent Luciani, Georgia Institute of Technology; Daniel Powell, McGill University, Montreal; J. Clarke Stevens, Brigham Young University; and David Yuen, University of Miami School of Music.

The grants for the 1987-88 academic year were established to encourage students to enter audio engineering and other related fields. Recipients are selected on achievements in audio and by faculty recommendations.

Additional information and application forms are available from the Audio Engineering Society, 60 E. 42nd St., New York, NY 10165; 212-661-8528.

TAC receives Queen's Award for Export Achievement

Total Audio Concepts Ltd., was presented the Queen's Award for Export Achievement on August 7 by Lord Lieutenant of Nottinghamshire, Sir Gordon Hobday.

Hobday explained, "TAC's achievement has been not only the successful growth of the business, but it has also created worthwhile, real and permanent jobs in the sector of the population where the job market currently offers least hope."

Sales director John Penn says, "I must emphasize that this award is being received on behalf of everybody in the company and is a mark of the achievement of the entire workforce."

People

G. Russell Farrell has joined Renkus-Heinz as director of marketing and sales.

David Wynn has joined American Audio systems as a sales consultant.

Nick Morris has been promoted to vice president/general manager of Nagra Magnetics, New York.

Peter Wellikoff has been appointed as executive vice president and general manager of Celestion Industries.

Yamaha Music Corporation, USA announces the following promotions: **Tom Weeber** has been promoted to general manager of the Drums, Guitars and Amplifiers Division (DGA), **Jim Coffin** has been promoted to the new position of assistant general manager of the DGA Division, and **Steve Thatcher** has been promoted to assistant general manager of the Digital Musical Instruments Division.

Telex Communications has promoted **Dean Flygstad** to senior vice president for science and technology, and **Joseph Winebarger** has been named vice president/engineering.

R·E/P

Anatomy of a Dance-Single Remix

By Bruce Nazarian

Producing extended length or heavily modified dance remixes is rapidly becoming a specialist artform.

One of the hottest trends in the music business during recent years has been the dance-music phenomenon. Although some would argue that dance music is nothing more than "disco with a face-lift," there is no doubt that the idiom has given us some great new stars. Artists like Madonna, Prince and Culture Club would probably not have had such an easy time of breaking across into pop stardom had there not been a strong dance-club base to pave their way to success.

There are other stars in the dance-music world, too: the producers and engineers that have been creating and/or remixing these successful club records. Names like Arthur Baker, John "Jellybean" Benitez, M&M (John Morales and Sergio Munzibai), Mark Berry and Shep Pettibone have become industry standards for record companies looking to maximize the dance floor potential of many recording artists.

This article will outline the various considerations that go into remixing and producing dance music. Presented in a do-it-yourself form, it should help you understand what processes are involved in reshaping an existing record for the dance-music market.

Bruce Nazarian is an independent producer who started his musical career as a session guitarist. Since forming D&B Productions with co-producer Duane Bradley, the duo has had a continuous string of dance music successes over the past three years. Nazarian also owns Gnome Productions, a 24-track MIDI production studio located in the Planet Sound complex, New York.

Author Bruce Nazarian with Orphan Record' artist Jimmy Lifton, during a recent dance-single remix of Lifton's charted release, "I'm a Man," at Gnome Productions/Planet Sound, New York.

Knowing the market

In creating music for the dance floor, or restructuring already existing music for danceability, there are always certain things that producers and engineers strive for. Whether it is a certain rhythmic impact, or a uniquely recognizable drum sound, each of these professionals know what the market is tuned into, and each tries to hit that mark, while keeping their productions and/or mixes original and fresh.

Keeping in mind the end market for this musical style, the desired objective is to get the audience up off their seats and out on the dance floor. To accomplish this, the rhythm is the key element. Although the original mixing engineer or producer may have been more concerned with pleasing pop or urban radio markets, the name of the game in dance music is the *groove*! Precisely for this reason, a dance mix tends to emphasize the rhythmic elements of a production, making the mix "hit" a bit harder than its radio counterpart.

In addition, because the time limitations of radio do not hold true for the dance clubs, the dance remixer often lengthens and changes the original arrangement. A 4-minute album track can easily be extended into a 6- or 7-minute dance track, when you consider that the introduction will be lengthened, one or more verses or choruses repeated, a breakdown placed in the middle, and a lengthy outro added.

In general, club mixes tend to time out in the 5- to 7-minute range, with five or six minutes being a pretty decent length for a club mix. But this is only a guideline; some recent club remixes have timed out at almost 10 minutes, with Prince's whopper 21-minute remix of "America" being the all-time leader.

Reworked, but not too much

While the goal of the remix engineer is to enhance the danceability of the record, there are also some things that generally are not tampered with. It is not the intent to inundate the record with his or her own personality, losing the essence of the original production in the process. Rather, it is the subtle job of enhancement and refinement for a specific market that is the forte of a remix engineer. The desire here is to improve upon the greatness of the original mix, without going so overboard as to make the song unappealing or unrecognizable.

The end result should bring a fresh new view of the song's original good points, while enhancing its danceability and suitability for dance club play.

There are also other considerations: the remixed version of a contemporary single is frequently used to extend the record's chart or radio life and boost its ultimate chart performance. To this end, the remix should be faithful to the original, while providing a sufficiently different view of the song to be almost a "new release," even though it may already be a chart contender. [For additional insight on such factors, interested readers may wish to refer back to Ralph Jones' interview with Louil Silas, Jr., published in the February 1986, issue of RE/P.]

How to begin

As with any job, understanding the desired end result is the first step in getting started. In this case, it means determining the number and format of the desired mixes. In general, 12-inch singles (the format used for dance records) tend to include several different versions of the featured song. They may also contain a second song as a bonus track, but most domestic 12-inch singles do not.

The number of different versions is subject only to the running time limitations of a 12-inch record at either of the two most popular speeds of 45 or 33⅓ rpm. At the latter speed, the physical disc limitations make for a maximum of 20 minutes of music per side; at 45 rpm, that time is somewhat shorter.

Regardless of the speed at which the record is eventually cut, one fact remains the same: the less music put on the side, the greater the level at which it can be mastered, dramatically increasing the record's apparent volume during radio airplay and the frequency response over a club playback system.

Mix treatments

Since dance remixes can take many forms, from subtle reworkings for sonic improvement, to drastic alterations for "freak effect," the first job is to gain an understanding of what specific mix treatments might be needed for a particular record. If we take a typical case—an urban radio single with a good groove, but just a bit too short for a dance 12-inch—we may only need an Extended Vocal mix (lengthened version of the original single, with some musical or vocal sections extended to gain time), and an Extended Instrumental track (basically the same approach, but without the lead vocals).

If the track is a bit more of a "street" record, that is, a grittier, more sparse funk record, then the extended version may take on more of a Dub approach, where crazy edits and other sonic tricks are used to build up the energy from the record's already intense level.

Nazarian's facility, Gnome Productions, houses an impressive array of MIDI-capable sequencers and sampling keyboards used during singles remix.

In any case, the remixer's experience, coupled with the record company's desires, will dictate how to determine the number of mixes needed and the formats for them. Because many remixers either work in collaboration with dance club DJs, or are DJs themselves, it is usually easy for them to get the feedback they need to fine-tune a mix. If you are contemplating doing a dance remix, you

Track sliding with a digital synthesizer

One of the nicest feelings to have when you are in the studio producing a record is the knowledge that everything is under control. With the recent addition of a New England Digital Synclavier to Gnome Productions' arsenal of MIDI-based equipment, this is exactly the case.

In addition to acting as the MIDI control center for all of our existing MIDI-equipped keyboards, the Synclavier also provides us with up to 400 seconds of 50kHz, 16-bit digital sampling in stereo. Because of this capability, I use the unit extensively to slide tracks with precise control.

For example, we frequently record multitracked background vocals (sometimes up to 20 tracks). We can sample the composite mix of those vocals and re-record them precisely into each chorus throughout the song, eliminating unnecessary time spent with background singers duplicating their already perfect performances.

In addition, the transparent quality of samples make it impossible to detect any generation loss between the original track and the duplicates. We have also used this capability to slide lead vocal and instrumental tracks around with great precision.

Assuming that the master tape has already been striped with time code, sliding tracks involves three relatively simple steps:

In Step 1, *the desired tracks are patched to the sample inputs on the Synclavier's sample-to-memory interface. Having activated the appropriate software page, the tracks are played into the unit and recorded into sampling memory.*

In Step 2, *the desired destination tracks are selected on the multitrack machine, and the appropriate Synclavier outputs routed to those tracks. As the master is being played back, the time code reader in the Synclavier is activated to give a running readout of the locations recorded on the tape. At the correct location for the layback, the time code is noted and entered into the unit as the Start time.*

Step 3 *consists of rewinding the 24-track a small amount, activating the Synclavier's external time code mode, and starting. If the programmed time code location is accurate, then the Synclavier will play back the sampled vocals onto the 24-track machine and the transfer is complete. If the vocals are ahead or behind, the time code start time (the "offset") is adjusted to compensate, and step *3 repeated until the transfer is complete.*

Compared to older "fly-in" techniques using an analog recorder, we have effected huge time savings, as well as making track sliding much more easily accomplished.

might wish to strike up a working relationship with a dance club DJ—he'll probably appreciate the opportunity to become involved with the production end of making records, and you'll value his feedback.

Remix example

A good example of creative remixing is Timex Social Club's recent hit, "Rumors/Vicious Rumors." The version heavily played by urban radio and dance clubs had the track stripped down to its emptiest, "street" feel. By removing the instrumental pad from the track, leaning out the lead vocal track, and omitting some lines and adding some "beat-box" percussion, the remix turned the song into essentially a "rap" track. In the clubs, and on urban radio, the effect was sensational.

The pop version, on the other hand, was devoid of the "beat-box" percussion featured prominently in the club version and restored the smooth keyboard pad under the lead vocal, as well as the missing lead vocal lines, in this form, it was very well suited for pop radio play. "Rumors/Vicious Rumors," represents a classic case of tailoring the same song, by judicious remixing to suit the requirements of two different markets, and it worked well.

What mix? How many?

Frequently, club DJ's will use two copies of the same 12-inch record, and mix between sections of different mixes to, in effect, "custom-mix" a brand new version of the song for their particular crowd. (DJ-turned-producer Jellybean Benitez started out doing this with a pair of 7-inch singles for his crowd at the legendary New York dancery, The Funhouse.

To allow the dance club DJs to build their own creative mixes, several different mix versions are usually provided. A typical 12-inch single may include any or all of these various mix treatments:

- **Extended Vocal** or **Instrumental** mixes usually preserve the existing elements of the original album track or single, but repeat musical or vocal sections to extend the overall length.
- **Dub** mixes tend to accent the rhythmic elements even more, taking snatches of vocal or instrumental riffs and echoing them heavily. Occasionally, they even include instrumental or vocal riffs that do not exist on any other mix version. Sometimes multiple dub versions are released on the same record, to allow even more mixing choices to be made by the DJ, or to extend the record's appeal to an even wider audience.
- **A capella** or **Voices only** mixes feature just the vocal tracks of the record, with little or no instrumental backing tracks. This format allows the club DJ to drop in the vocals over the record in places where they might not already exist, or even on top of other records. If a record is popular, the voices can also be mixed in on top of another current record to "tease" the crowd.
- **Bonus Beats** are sections of the basic rhythm track, usually featuring the drums, percussion and other rhythmic motifs used in the record. This mix provides the DJ with an opportunity to make a smooth transition to another record of a similar tempo, or provide a bed from which to drop in the acapella vocals.

A recent trend by some record companies is to feature multiple remixes of the same song, done by different engineers. For example, one remixer's approach may be a more "rock" drum sound, and the other may be a bit more R&B. Such mixes frequently occupy opposite sides of the same 12-inch single.

The differences in each mixer's approach help to expose the record to a wider buying market, and allow the club DJ a greater latitude in what records may be suitable for their crowd. As an additional marketing tool, some 12-inch singles also include the original album or single version of the track, in addition to the various remixed versions.

Planning a typical remix session

Let's take a look at how a typical remix session might be planned out. You may be surprised at how many variables need to be worked out before you even turn on the tape machine. Figure 1 shows the basic steps involved:

Obtain original master or safety.

To properly remix, you need access to the first-generation master, if at all possible. If that is not obtainable (for whatever reason) then the next best thing is a second-generation safety master, transferred directly from the original master. This stage may also be necessary if the studio you're using for a remix does not have the same noise-reduction system used on the original master. In this case, you will need a decoded safety master from which to work. Usually, these master tapes are 24-track, but you can remix from just about any tape format. (It gets hard, however, to get a really killer drum sound off an 8-track master!)

2. *Analyze tape tracks; check track sheet for accuracy.*

This may sound silly, but you'd be amazed at the number of incorrect, or outdated, entries that exist on some track sheets. Because the track sheet is your only guide to what's actually recorded on the master tape, you had best spend a few minutes making sure that what it's telling you is *accurate*, or else you'll spend hours chasing down nonexistent tracks.

In addition, time spent listening while checking the track sheet will help to familiarize you with the structure of the song and the existing tracks. This is knowledge that eventually you must have to make informed judgments about how to restructure the mix as it goes along. Listen to the track several times and take notes as you go.

On most mixes, I find it helpful to make a simple "timing map" of the song structure, relating the musical layout of the song to the running time of the tape. For example:

Time = *Location*
00:00 = Bar 1, Intro.
00:15 = Bar 9, Chorus 1.
00:30 = Bar 17, Verse 1.
00:45 = Bar 25, Verse 2.
01:00 = Bar 33, Chorus 2.
...and so on.

Such a chart makes it very easy to set

autolocator points, or to manually enter a location if you need to refer closely to a certain section on the tape. Another tool that I have found very handy in remixing is a "track map," which graphs the activity of each of the master's 24 tracks against running time.

As can be seen from Figure 2, an entry in the appropriate square means the track is active somewhere during that section. The chart can be very helpful in getting an overall view of what's happening at each moment on the tape, and especially helpful if you are contemplating sliding tracks around (more on this later).

3. Reference the existing album or single mix.

You should be familiar with the existing version of a record, so that you will know what needs improving and what doesn't. In some cases, the record company commissioning the remix may not want to stray too far from the record's original sound. Knowing the sound and concept of the original track will help you stay within their guidelines, whatever they may be.

The time spent researching the original mix will also help you key in on things, such as the various "dimension" effects used—reverbs, delays, echoes and other processing. Some of these effects may be quite suitable for re-use. In short, the more familiar you are with the original, the more it can help during the remix.

4. Determine if any additional tracks are necessary.

Although it may be possible to mix the existing tracks without adding any additional material, and still make a great dance record, a more usual scenario is to replace some tracks or add new parts to existing ones. This technique can range from the simplest drum replacement (either with triggered samples, or re-recorded drum sounds), to a full-blown master recut.

I recently co-produced just such a remix, using a 24-track safety master of an existing European hit, we erased and replaced everything on the record except the original vocals. This was accomplished by laying in a brand new

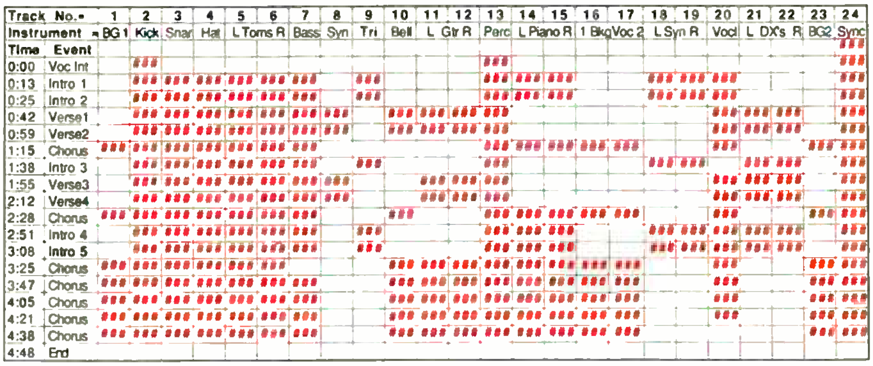

***Figure 1.** Basic flow chart of the various stages involved in the production of a dance-single remix.*

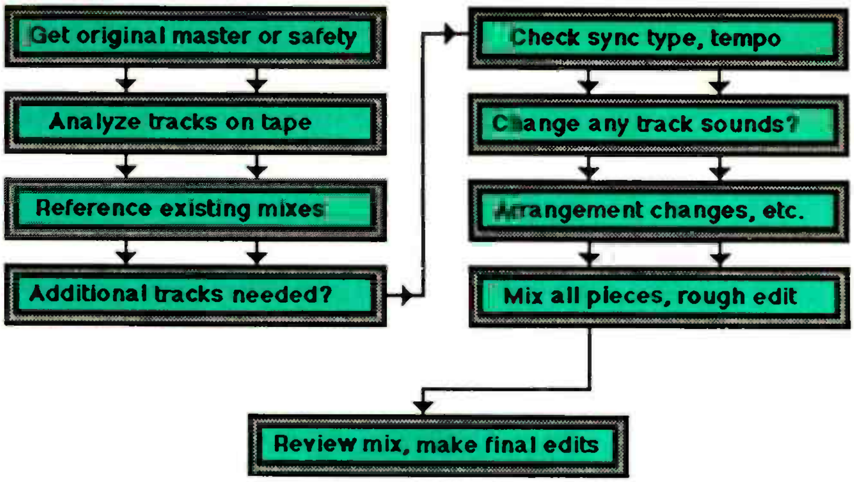

Figure 2: *To simplify the remix procedure, an annotated timing breakdown of the 24-track tape provides an overall view of what's happening at every moment on the master. It can be particularly useful when tracks have to be slid to differnt parts of the tape.*

programmed backing track played by MIDI sequencers and synchronized to the original via time code.

If you opt for recording additional tracks, you have three choices of technique:

1. Erase some of the existing tracks from the 24-track master and overdub new parts (a technique that is frowned upon by almost everyone);

2. Create a 24-track slave reel and overdub additional tracks to complete what you are hearing. (This is a much cleaner way of doing things, although more costly in studio time.); or

3. Find (or create) a sync track on the original master tape and use MIDI sequencers and/or drum machines to add whatever additional parts may be needed. With the advent of computer-generated original tracks, this technique is becoming more and more easy to implement. It's pretty easy to harness modern computerized recording technology if the original producer (assuming it's someone other than you) has left you the necessary information from which to work.

If the tracks was originally cut against a sync or time code track, and the sync track wasn't subsequently erased to add an overdub, you should be in good shape. Just find out what type of sync track it is, or the appropriate time code offset point (the point at which the external time code box began generating the clock signal) and the track's tempo of the track. (Be sure this latter information has been faithfully recorded on the track sheet.)

Given either of these sync methods, it is easy to synchronize new parts to the existing tracks. If you do use a sync track, then there is one more detail:

5. Determine the type, tempo and source of the sync track.

Generally, there are two options for synchronized recording: either some form of proprietary sync code (Linn stripe, Oberheim, Roland and Yamaha FSK code); or time code and an external clock box to read the time code and generate a clock signal for the computers to follow.

To properly remix, you need access to the first-generation master if possible.

Whichever method has been dictated by the original recording, it should be fairly easy to determine what type of code you have and make the appropriate hookup for using it. If no sync code exists, you can even stripe time code onto an unused track and create your own sync track. (Exactly how to accomplish this, however, is a fairly involved process, and beyond the scope of this current article.)

6. Determine any sonic changes required in the tracks.

Now it starts to get subjective. Using experience and taste, the remixer makes a judgment on each track of the existing recording, as to its usefullness in the remixed version. If the musical part is correct, but the sound needs help, then re-recording the track may be the best way. If both the part and the recording are lame, then trashing it and recording a completely new replacement part may be the best solution.

Again, there are no hard and fast rules here, only judgment calls based on experience and taste. If there are any things that need to be added (additional vocals, percussion, etc.), this is the time to record them.

Producing modified dance remixes is rapidly becoming a specialist artform.

7. Plot the final arrangement changes, track sliding, etc.

While determining the necessary sound changes, some thought must be given to the final structure of the remix. Having selected the basic mix treatment—Extended Vocal mixes, for example, have a slightly more conventional structure than, say, a wild and crazy Dub mix—we need an intro, a smooth transition into the body of the mix, a break, another transition into the second body or reprise, and some form of outro.

The intro will consist of some appropriate way for a DJ to mix into the track, sometimes just a stripped-down drum groove. Building from this point into the body of the mix is a matter of taste, and how long you want the record to run. (It may take anywhere from 30 seconds to a minute to get from the intro to the first vocal verse.) The intro may be a perfect place to introduce a new "signature" sound that the remixer has decided to feature in the mix; perhaps a new sampled melody phrase or some other distinctive sound. The idea here is to provide an instantly recognizable "hook" that makes identification of the song easy.

Once you're into the body of the mix, the verses and choruses generally take care of themselves for the next few minutes. Then there is the question of what to do for a break. The track could slide back into the intro, dropping down to the drums again, or it could go in a totally different direction. Again, there are no hard and fast rules here, just taste and judgment.

After the break has been built, a secondary body of verse and/or chorus material will use up the time between the break and the outro. The outro is either similar to the intro, i.e., a stripped down drum groove into a fade to make mixing out easy, or a dead stop.

Another technique that has rapidly come into play with the widespread use of digital-sampling technology is Track Sliding, the process of moving a recorded piece of music or vocal from one location on the multitrack tape to another. Frequently a remixer will find an interesting musical phrase or vocal line that could be usable if it were moved to a different place on the master tape.

A recent trend is to feature multiple remixes of the same song by different engineers.

Prior to the availability of high-quality 16-bit digital samplers, this technique was only possible by a laborious process of dubbing off to 2-track and flying the part back onto the master tape at the desired new location. With 16-bit samplers, track sliding has become an easily-used technique that can assist in the restructuring of the master. A specific example using a digital sampling synthesizer is provided in an accompanying sidebar.

Session Examples:

A conversation with producer Duane Bradley

My partner, Duane Bradley, who also serves as Detroit Club DJ and record producer, describes his remix technique in terms that are close to home: "I always try to envision the final mix as if I was going to play it in the club that night," he says.

"The ultimate goal of a dance remix is to make the record more danceable, while preserving and enhancing the appealing aspects that already exist in the record. In addition, whenever possible, I try to maintain and enhance the record's commercial appeal.

"The first and main area of concentration is, of course, the rhythmic foundation—the drums, percussion and bass line of the record are of utmost importance. Because you can't build a solid record on a shaky foundation, we do our homework to be certain the rhythm track is as solid as a rock. After that, building the rest of the tracks into the mix comes easily.

"Knowing what will work on the dance floor is a great asset. I try to keep my alternate mixes as interesting and predictable as possible."

Different DJs prefer different styles of mixes for their clubs, so what may be correct for one club DJ may not work for another. Providing several different mix treatments on the 12-inch allows each DJ to program the most appropriate version for their particular crowd or club, making the most effective exposure for each particular record.

Another trademark Bradley trick is the inclusion of "wild" vocals (vocals without music tracks) and bonus beat tracks on the 12-inch.

"I know from my own experience that these kinds of little ideas can really be of good use to a creative DJ," he adds. "Especially now, with some DJs beginning to use digital samplers in conjunction with their mixing boards, these little tidbits can be used to maximum advantage."

Bradley also advises some preplanning for intros and outros of 12-inch mixes: "The best dance records for me as a DJ are the ones that are easiest to mix in to and out of. Because it seems like I always have a hundred things going on at once while I'm mixing for my crowd, the easier you can make it for me to mix into the record, the better my reaction to it will be—and the more times it might be played. When I mix a record for others, I pay special attention to these details. It really pays off."

8. Mix all necessary pieces: make rough edits.

At last it's time to begin putting it all together. Building the mix at this point closely resembles the original mixdown session. The various instruments are put up in whatever order the remixer chooses to use (generally drums first) and the remainder of the mix is built from there. As the tape rolls, the predetermined pieces that are needed are mixed and recorded.

To facilitate the eventual assembly of the mix, and save time on non-automated consoles, start-and-stop mixing techniques can be used. One section of the tape is rehearsed until the mix levels are correct, and then that section printed to the 2-track. The next mix section will then be set up and rehearsed and, with an eye toward how the junction edit (the edit that puts them together) will work, that section also will be printed.

As each section is completed, the junction edit may be performed to verify that it will, in fact, work properly. In this manner, the mix is being completed and edited practically as it goes along.

Another nice feature for non-automated consoles are programmable mutes, where certain tracks or groups can be assigned to be simultaneously cut from the mix with one switch. This type of function can make rapid mix changes very easy to accomplish, and is especially handy when you need to mute 13 tracks, and only have 10 fingers.

If no sync code exists, you can even stripe time code onto an unused track and create your own sync track.

9. Review the mix and make any final edits based on its flow and running time.

After all the junction edits have been performed, the mix is played in its entirety and reviewed for overall flow. Any fine-tuning of the mix can easily be done at this point, including additional editing, or even remixing a section again, should a problem be noted. Trimming the various sections to bring the total running time into line with the desired length is all that remains to be done to obtain the final mix.

The last word

There you have it: you've successfully navigated your way through a dance remix. Although the procedure may seem a bit complicated at first glance, it's really quite easy. If you've handled dance mixes before, you will probably be familiar with most of the steps listed here; if not, the best preparation you can get is to listen to some of the better dance records that have been produced lately, and see just how these techniques are executed in practice.

Several of the currently hot producers include Jimmy Jam and Terry Lewis (Janet Jackson, S.O.S. Band), Arthur Baker ("Planet Rock", "Sun City"), Nick Martinelli (Melba Moore, Loose Ends), Jellybean Benitez (Madonna) and Narada Michael Walden (Aretha Franklin)—but there many others. A brief listen to some of their recent productions will introduce you to some of the best work in the business, and give you a head start in the right direction.

Just remember, if your mix doesn't make *you* want to get up and dance, it probably won't have that effect on a club audience either.

R·E/P

The Future of MIDI Time Code

By Paul D. Lehrman

A set of commands within the MIDI spec, MIDI Time Code could emerge as a standard interface and communications protocol for studio automation.

MIDI Time Code has been a popular buzzword in the world of music and sound production for many months. But exactly what it is, what it does and when it's going to do it, have remained something of a mystery to the average recording engineer. MIDI TC has been both praised and maligned, read and misread, quoted and misquoted, and talked about so divisively that it now rates as one of the most controversial "standards" ever proposed.

Simply stated, MIDI Time Code is a set of new commands within the MIDI specification that gives MIDI the potential to perform complex studio automation. Its creators hope that manufacturers in both the musical-instrument and pro-audio spheres will begin a headlong rush to incorporate MIDI TC in their products. Although it is going to be a while before MIDI TC's fate is established, if it does take off, then the commands could substantially affect the way that music and sound—especially in conjunction with images—are produced.

Although a variety of manufacturers and consultants had input into designing it, MIDI TC is primarily the work of two people: Evan Brooks and Chris Meyer of Digidesign, the Menlo Park, CA, company that specializes in voice-editing software for MIDI samplers. Meyer was formerly with Sequential Circuits; it was during his tenure there that the idea for MIDI Time Code was first germinated.

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

What it can do

MIDI Time Code has two purposes. One is to allow a MIDI data line to synchronize a variety of studio equipment—including, but not limited to, tape transports, samplers, CD players and digital sequencers—to complement, and in some ways replace, SMPTE time code.

Synchronization has been part of the MIDI specification from the beginning. MIDI clock commands, which divide quarter-notes into 24 parts, can synchronize devices hooked up to a MIDI cable by making sure that they run at the same tempo. On the other hand, Song Position Pointer (SPP) commands specify time in relative terms (as an offset from the beginning of a sequence, in measures and beats), thereby allowing a sequencer or other device to be told to start from any point within a piece of music. Together, these two commands are already being used in many studios to get sequencers to follow audio tape, videotape or each other. They are simple, fast, and relatively reliable.

MIDI Time Code, however, goes beyond that and, like SMPTE time code, deals with "real-time" information—hours, minutes, seconds and frames—and even has provisions for data type and user bits. In fact, the original name for MIDI TC was *MSMPTE*. This nomenclature was discarded, however, in the belief that the Society of Motion Picture and Television Engineers might not like its name being used for something it had no part in developing.

The second purpose of MIDI Time Code is to act as a common command language for automation and editing; one that is not specific to any one machine, but which instead, like MIDI itself, can be used equally well by any manufacturer's equipment. To accomplish this, the specification defines MIDI Cuing messages that communicate event lists and other information to and from various devices "off-line," i.e., not in real-time.

There are two strong points in favor of implementing MIDI Time Code. The first is that, because hardware for generating and reading MIDI data is generally considered to be easier to build than hardware for SMPTE time code, MIDI TC can provide sophisticated automation capabilities for a much less expensive class of equipment. The second point in MIDI TC's favor is that currently there exists no universal automation standard using any other protocol, such as RS-232 or RS-422. (Although various manufacturers have developed some very sophisticated formats on their own.) MIDI TC offers a "clean slate" for manufacturers to develop such a standard.

Despite its potential, however, there is a certain amount of resistance to adopting the new standard.

Why it took so long

Contributing to the mystery and confusion surrounding MIDI TC is the fact that it took a long time to be approved. Although work started on it in late 1985, and a tentative format was devised in June 1986, it wasn't until February 1987 that the Japan MIDI Standards Committee gave the specification its formal imprimatur and, even then, say some observers, the committee was "reluctant" to do so. While the process was going on, discussion of the idea was conducted, under strict security, by the MIDI Manufacturers' Association and what news did leak out was more tantalizing than informative.

Because of the delay in securing approval, manufacturers were hesitant to develop products using MIDI TC and even speculation about how it would eventually be used was at a minimum. (And for good reason: in the last weeks before final approval, a minor change was made to the spec that would have rendered any already-written software useless.)

Although a few developers outside of the small circle that originally devised the idea became involved in perfecting it, most developers adopted a "wait-and-see" attitude. A few spent their energies on peripheral issues, such as the proper acronym for the code. (*MTC* was objectionable to some manufacturers that were already using it for Master Time Code.) And many wondered whether it was necessary at all, given that the cost of decoding SMPTE time code directly was falling rapidly.

Inside MIDI TC

MIDI TC does not replace MIDI. Rather, it is an extension of MIDI 1.0, and uses existing MIDI message types that were either previously undefined, or were being used for other, non-conflicting purposes, such as the Sample Dump Standard.

The great majority of existing MIDI hardware— synthesizers, processors and samplers—will never use MIDI TC directly for anything. Instead, new devices that read and write MIDI TC will have to be developed.

Most of what MIDI TC does has almost nothing to do with MIDI as we know it today. The decision to make it part of the MIDI spec was based largely on two facts: first, that MIDI is the closest thing the audio world has to a universal communications protocol; and second, that MIDI hardware is well established and inexpensive.

Although a MIDI TC signal can travel the same electrical path as more conventional MIDI information generated by a sequencer or a synthesizer (notes, controllers, pitchbends, etc.), there is no particular reason for it to do so. In fact, to keep from overloading the MIDI stream, it would make sense in most applications for the MIDI TC signal to be kept completely *separate* from other MIDI data.

MIDI Time Code is not a replacement for SMPTE time code either. Because it runs at MIDI frequencies (31.25kHz), it cannot be recorded or sent over ordinary audio lines. SMPTE time code will remain the format of choice for printing onto audio or videotape. SMPTE TC is converted into MIDI Time Code through a special device, and then the MIDI TC synchronization messages can be distributed to other devices around the studio.

MIDI Time Code's synchronization messages come in two flavors. When a tape is wound, cued to a certain spot and stopped, its time code location is read and the information sent to a MIDI TC generator, which regurgitates it as a Full Message. This message includes a complete time code number, plus a tag for the type of time code (24, 25, 30 drop-frame, or 30 non-drop).

While a tape is running at normal speed (or close to it, as in varispeed mode) the MIDI TC generator will not send out such long messages, but instead will send a simple 2-byte message every ¼ frame known, not surprisingly, as a Quarter Frame Message. QFMs have two functions: they act as synchronization pulses, with a specific message occurring precisely on each frame boundary; and they contain pieces of the time code number in the form of ½ bytes or nibbles. When a reader receives a complete group of eight messages, it will know the complete time code number. (See the accompanying sidebar for a more complete discussion of MIDI Time Code data types.)

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MIDI Time Code commands

MIDI Time Code consists of four messages that fit into three categories, or types, as determined by their headers. Two of these message types have not been used before for any purpose, while the third is also used with the new Sample Dump Standard.

Quarter Frame Messages

*Quarter Frame Messages are sent only while the system is running in normal (or varispeed) time, either forward or in reverse, but not in fast forward or rewind, because at high speeds they would clog up the MIDI stream. Each Quarter Frame Message comprises two bytes (all references to MIDI command codes in this sidebar are made in hex), the first being **F1**, which is the new System Common "header," and the second consisting of the bits* **Onnn dddd,** *where **nnn** is the message type (of which there are eight) and **dddd** are the data bits.*

*The eight message types denote least significant (LS) nibbles (or half-bytes) and most significant (MS) nibbles for frames, seconds, minutes and hours. A complete Quarter Frame Message set comprises eight messages, one of each type, to convey a complete Time Code number. The message for "frame LS nibble" **(F1 Ox)** is always sent on a frame boundary. In addition, the hours/MS nibble message includes two bits for specifying SMPTE time code type (24fps, 25, 30 drop-frame, and 30 non-drop).*

A MIDI Time Code reader takes the incoming nibbles and, when a full set of messages has been received (after at least two frames), it can decode a location and achieve timing lock or whatever other function is desired. A reader should also be able to deduce that when Quarter Frame Messages are coming in backward, time is running in reverse.

*Quarter Frame Messages not only provide location information, but also provide the synchronization pulse for a MIDI Time Code system, with a frame LS nibble **(F1 Ox)** message arriving precisely (within the constraints of MIDI) on each frame boundary. QFMs are only sent while a system is operating in real time, so they also serve to inform the whole system that time is running—i.e., tape is rolling.*

Full Messages

Unlike the QFMs, each Full Message contains all of the time code data. Full Messages are not used while tape is running—they would clog the MIDI line—but instead are sent after a tape or other device has been cued to a specific point, to tell the rest of the system where it is.

It is also suggested that during a very long fast wind, if a device needs to be periodically updated, a Full Message can be sent every so often.

The format for the Full Message is as follows:

F0 7F 7F 01 01 hr mn sc fr F7

***F0 7F** is the Real Time Universal System Exclusive Header; **7F** the channel number (in this case, it means the message is intended for the whole system); **01** the "sub-ID 1" identifying the message as MIDI Time Code; **01** the "sub-ID 2" identifying the message as a Full Message; **hr** the time code hour, including the time code type; **mn, sc,** and **fr** the time code minutes, seconds and frames; and **F7** is the "End of Exclusive" (EOX).*

User Bits

These can be used exactly the same way as user bits in SMPTE time code streams. MIDI Time Code User Bits allows four 8-bit characters, plus a 2-bit Format Code. A User Bits message can be sent at any time.

Here is the format:

F0 7F 7F 01 02 u1 u2 u3 u4 u5 u6 u7 u8 u9 F7

F0 7F *is the Real Time Universal System Exclusive Header;* ***7F*** *the channel number (message intended for the whole system);* ***01*** *the "sub-ID 1" identifying the message as MIDI Time Code;* ***02*** *the "sub-ID 2" identifying the message as a User Bits message; and* ***F7*** *is the EOX.*

The User Bits themselves take the form:

u1 = **0000aaaa;**
u2 = **0000bbbb;**
u3 = **0000cccc;**
u4 = **0000dddd;**
u5 = **0000eeee;**
u6 = **0000ffff;**
u7 = **0000gggg;**
u8 = **0000hhhh;**

and are assembled into four 8-bit characters consisting of **aaaabbbb ccccdddd eeeeffff gggghhhh.**

u9 is **000000ii,** which contains the Binary Group Flag Bits, as defined by the SMPTE.

MIDI Cuing Messages

Finally, there are the MIDI Cuing messages which, in preliminary versions of the MIDI Time Code specification, were called Set up messages. They are at least 13 bits long, with provisions for additional information as desired.

The format is as follows:

F0 7E 7F 04 st hr mn sc fr ff sl sm <additional info> F7

where ***F0 7E*** *is the Non-real Time Universal System Exclusive Header;* ***7F*** *the channel number (message intended for the whole system);* ***04*** *the "sub-ID 1" identifying the message as MIDI Cuing;* ***st*** *the "sub-ID 2," or setup type, which we'll get to in a moment;* ***hr, mn, sc*** *and* ***fr*** *hours, minutes, seconds and frames;* ***ff*** *fractional frames (from 0 to 99* ***decimal****);* ***sl*** *and* ***sm*** *Event Number (least-significant byte first); and* ***F7*** *is EOX.*

There are 128 possible setup types within this set, and the current specification defines 19 of them:

01/02 = Punch In/Out;
03/04 = delete Punch In/Out from the cue list;
05/06 = Event Start/Stop;
07/08 = Event Start/Stop using additional information later in the message;
09/0A = delete Event Start/Stop (with or without additional information) from the Cue List;
0B = Cue Point;
0C = Cue Point with additional information;
0D = Delete Cue Point;
0E = There is an event name in the additional information (for the convenience of humans).

There are also five special messages for which **st** = 0. They take the place of the Event Number:

00 00 = Time Code offset;
01 00 = Enable Event List;
02 00 = Disable Event List;
03 00 = Clear (erase) Event List;
04 00 = System Stop.

In MIDI Cuing messages, any "additional information" is nibblized MIDI data, with the LS nibble first. The exception is information following an OE message ("There is an event name . . . "), in which case the additional information is nibblized ASCII, LS nibble first.

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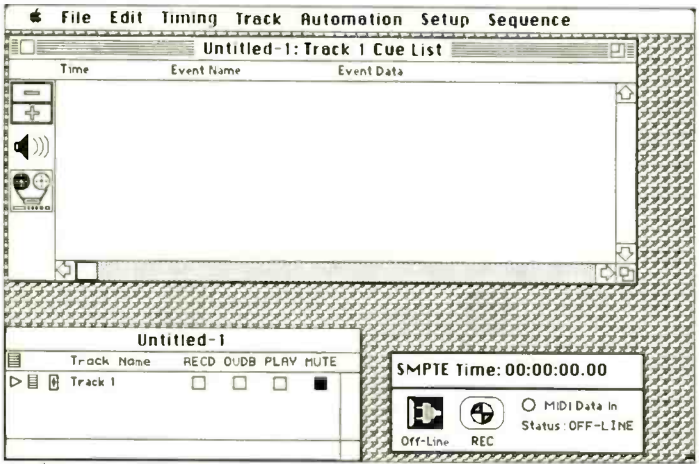
The Cuing messages are more complex. They can be used off-line to assemble an edit list, which consists of instructions for a device to execute a specific instruction (for example: play, stop, punch out, reset) at a specific time. Each instruction within a Cuing message has a number, a time, a name, any amount of additional information and a type. There are 128 possible types of Cuing messages; the current spec defines 19 of them.

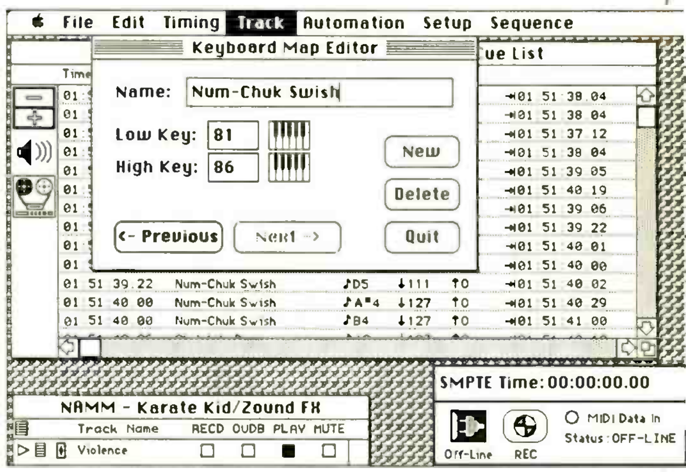
To use MIDI Time Code effectively, a device will need to be able to read and store event lists in the form of Cuing messages coming from another source—similar to the way synthesizers can read patch data from a computer patch editor. However, while synthesizer patch data is in that realm of MIDI call System Exclusive—so that one synthesizer's French horn is another's gibberish—MIDI Time Code's commands are designed to be universal, so that an edit list designed for one type of device can also be used by others.

What it will do

Using MIDI Time Code in an audio/video environment would be much like using any other automation or editing protocol. A central controller would assemble edit lists and send commands to the various devices being used to produce the soundtrack.

There would be several important differences, however. One would be that every device in the studio would talk the same language, and therefore no conversions between codes or formats would be necessary (except for time code-to-MIDI TC conversion at the master and slave tape decks). Consequently, the same piece of software could be used to assemble edit lists for *every*





Reproduced here are four representative screen dumps from Digidesign's Q-Sheet, a Mac-based sequencer that records, edits and plays various kinds of MIDI data using MIDI TC derived from time code as its clock. [Left to right.]

• ***Start-Up Screen:***
When the user first boots the program, they are presented with a blank edit decision list (with one track created and opened) and the tape transport controls.

• ***Keyboard Mapping:***
The sound effects are built up track by track, either by live recording or manually entering SMPTE time code locations and hitting keys on a MIDI-capable device. Naming of cues is simplified by Keyboard Maps. Either before or after entering the hits, the name of sound effects may be entered according to MIDI key ranges, and are then automatically filled in the cue list.

• ***Composite List:***
All the tracks have been entered at this point, including several tracks of sound effects, along with automation of external signal processors, mixers and MIDI switching. These may be viewed per track or (as here) as a master list that merges all the tracks. The composite list can be printed and kept as a permanent record.

• ***Automation Window:***
Viewing information in this program for its other main purpose—mixdown automation—is as a series of moving faders, knobs and buttons. Each may be named and assigned to MIDI controllers per track. Here is shown an assignable layout for controlling an external Yamaha DMP-7 mixer.

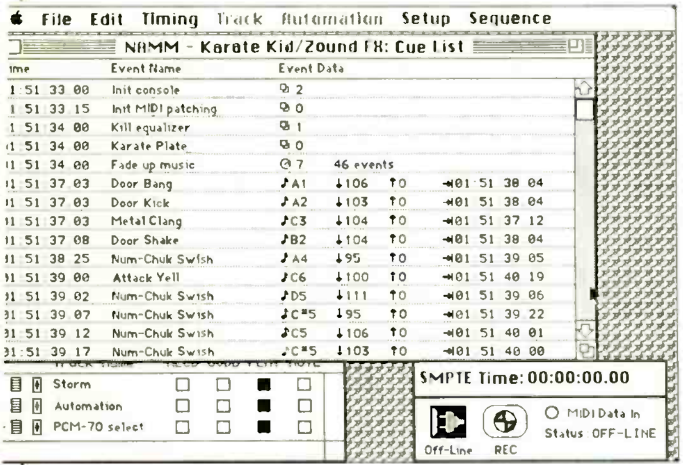
piece of equipment. Also, events could be correlated among different devices through the software. For example, all of the events in a scene—whether they are on a tape, on disc, in a sampler or in a sequence—could be pushed back a specific amount of time with a single command.

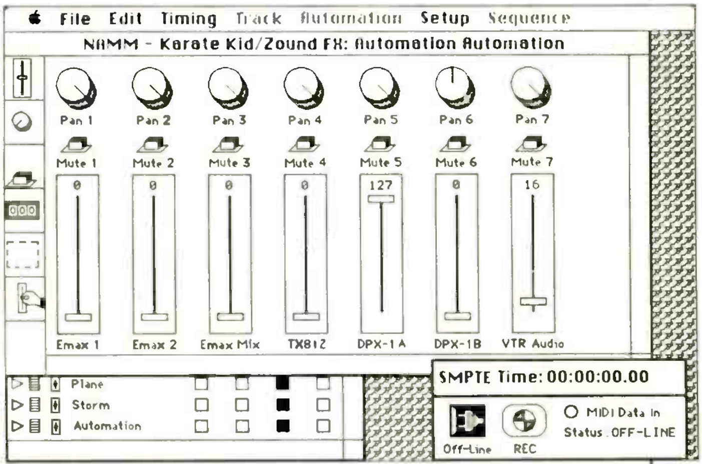
The second major difference is that the user would have direct access to MIDI commands to drive sequencers, samplers, mixers and processors. An engineer could record an effect into a sampler, edit it, move it around and then print it directly to the master tape without leaving the computer. A music track recorded in a sequencer could be started, looped, edited, retimed and remixed in the same program. Foley effects could be recorded live on the fly into a sampler and a sequencer, and then fine adjustments to the timing of each event made afterward. Room ambience could be controlled by sending program and controller changes to a MIDI-programmable digital reverb. The mix for the whole soundtrack could also be recorded as a series of MIDI events and edited as necessary.

With computer control over all of the sound events, there is less need for multitrack tape, thereby cutting down on generation loss and cost. Anyone who has worked with a friendly computer sequencer knows the advantages of using the latter to record multitrack music compared to conventional tape recording; a MIDI Time Code-based editing system will provide those same advantages.

First software packages

The implications of MIDI Time Code are far-reaching and, like most new standards, all of its possibilities haven't yet even been thought of. One of the first software pack-





ages to use MIDI TC, Digidesign's Q-Sheet, which is being readied for release as of this writing, runs on an Apple Macintosh, in conjunction with a time code-to-MIDI TC converter. (As of this writing, there is only one such device on the market: the J.L. Cooper Electronics PPS-1, an inexpensive box that performs just the conversion functions, with no built-in intelligence. A similar device from Opcode Systems, Timecode Machine, is due out shortly, as is one from Sonus that will interface directly to an Atari ST personal computer.

Q-Sheet has been described as a sequencer with a non-musical interface. In its current form, the program records, edits, and plays various kinds of MIDI data, using MIDI Time Code derived from time code as its clock, rather than the usual measures, beats and ticks. The program can handle an infinite number of "tracks," which serve as a cue list for one or more MIDI devices.

The information in a track can be notes, program changes or controllers, and it can be created directly by the software or recorded from an incoming MIDI keyboard or other source. Each track can be independently routed to any MIDI channel and to each of the Macintosh's two serial ports.

The software is designed to be expandable as more devices that will operate from MIDI Time Code become available (as Digidesign fervently hopes). For example, you can enter an event in the sequencer by specifying a particular time first, and then playing a note or moving a slider on a MIDI keyboard.

There is also an automation screen available for each page, which can act as a "console construction set." Any MIDI note or controller can be assigned a pictorial icon, and sets of these icons built up into a "virtual" console. The console's controls can be grouped and adjusted on the screen with the mouse, which simultaneously sends out MIDI data to the appropriate device and records it into the sequencer. If corresponding commands are received from an external controller, the screen faders move and, again, the movement can be recorded.

When the user starts a track from a point other than the beginning, the software will "chase" all previous controller and program change commands, so that all on-line devices will be set correctly for the cue. Finally, the program will read MIDI Files created by other sequencers.

Q-Sheet is just the tip of the iceberg for MIDI Time Code-based automation. For one thing, the program works only with standard real-time MIDI messages; its "cue lists" are simply sequencer tracks of note-ons and -offs, controller and program changes, to be executed in real time, not true MIDI Time Code Cuing messages. Right now, Cuing messages would be useless because there is not yet any hardware to read them.

What remains to be done

As we've seen, MIDI Time Code is currently at the "Chicken-and-egg" stage. Many manufacturers are reluctant to implement a standard that might not prove successful, and so are waiting for others to do it first. Of course, the others are waiting too.

Some manufacturers are ignorant of what MIDI Time Code is all about and, rather than bothering to learn it, they seem to be hoping that either their competitors will show them the way to use it, or it will just disappear. (While researching this article, I was rather amazed at how many otherwise knowledgeable people in the industry were ill-informed about MIDI Time Code.)

Although the currently available software represent good starts, programs for generating and transmitting true Cuing messages still remain to be written. With the large number of undefined Cuing event types that are just aching to be used, such software will have to be capable of being highly customized. Because it will have to work with many different devices from different manufacturers, it may end up looking like that holy grail of software writers, the "universal patch editor."

Hardware products that have yet to be developed include synchronizers and transport controllers, with the ability to store the lists created by the computers, and then issue the proper commands for operating audio and video decks, CD players, consoles, video switchers and effects devices.

One of the problems facing developers of such devices is that any MIDI Time Code-compatible recording or playback unit, like a sequencer, tape transport or editor, will have to learn to interpret Cuing messages in a way appropriate to its operation. An Event Start command may be read and executed immediately by a sequencer, whereas a tape machine will have to take pre-roll into account, and a sampler might have to spend 20 seconds loading a new file from disk before it can make a sound.

Decisions will have to be made as to whether "pre-events" like these will also have to be externally programmed (as "start pre-roll" or "load disk" commands), or whether they will be part of the internal intelligence of the device in question. (For example, the sampler *knows* that 20 seconds prior to an Event Start it has to initiate a disk load.)

Updates will have to be made available for existing devices that are time-sensitive, such as hardware sequencers, synthesizers with built-in sequencers and drum machines, so that they will be able to read MIDI TC directly and fire at pre-programmed times.

This last point raises the question of how current sequencers and drum machines, which are used to reading MIDI clocks that give them tempo information, will deal with a sync signal that gives only *absolute* time, *not* tempo. Where does the conversion between real time and "musical" time take place—in the synchronizer, in the sequencer itself, or in another kind of device entirely?

One solution, which is already being used in conventional time code-to-MIDI converters (i.e., ones that generate MIDI clock and Song Position Pointer), is a tempo map. This comprises an area of memory in the converter that holds a list of all the tempo changes in a particular piece, and the bars and beats (or fractions) on which they occur. The device can then calculate and generate the proper MIDI clock rates and pointers when it receives time code data.

Unfortunately, there is no universal form of tempo map, and each device on the market has its own way of generating them. A proposal has been made to the MIDI Manufacturers' Association for a tempo map standard that will allow different devices to exchange information. However, its approval is uncertain.

The verdict obviously is still out on MIDI Time Code. Although it is very cleverly designed, and shows tremendous potential, it will, like any new standard, require the cooperation of many diverse people and products to make it work.

Unlike MIDI, which seemed from the outside to represent no real threat to anyone and thus was readily adopted by scores of manufacturers, MIDI Time Code will require hardware and software makers to rethink their ideas about synchronization, automation and communication.

It certainly is an appealing idea, having control and processing devices from hundreds of different companies talking the same language, and using an already common electrical interface that adds only a dollar or two to a device's manufacturing costs. But the cost of reading and writing SMPTE time code is also decreasing, and it's hard to say definitely that the use of MIDI Time Code will save money.

If you were hoping that this article would provide the final answer to whether MIDI Time Code will succeed, I'm sorry to disappoint you. I don't know. Some folks claim to know, but at least half of them are wrong.

Just like you, I'm going to wait and see. R·E/P

Part 2

Eight Stand-alone Microphone Pre-Amplifiers

By Bob Hodas and Paul Stubblebine

- Benchmark Media 4x4
- GML Model 8300
- HART MP-501 PC card
- Innovative Audio UTMP-2
- Jensen Twin Servo 990
- Sontec MB-1A
- Studio Technologies Mic Pre Eminence
- Summit Audio TPA2

In the September issue we considered the physical attributes of eight microphone pre-amps to see what they had to offer. This month we'll explore the results of the listening tests, and see if there is any correlation to the measurements presented in the first installment.

For those of you who missed last month's issue, the eight devices under test were the Benchmark Media 4X4, GML Model 8600, Innovative Audio UTMP-2, Jensen Twin Servo 990 (manufactured under license by Boulder Amplifiers), Sontec MB-1A, Studio Technologies Mic Pre Eminence, Summit Audio TPA2, and a custom-designed MP-501 card produced by Harvey A. Rubens Technology (HART).

Bob Hodas is RE/P's evaluations and practices consulting editor, and Paul Stubblebine is a San Francisco-based recording engineer and audio consultant who started as a mastering engineer at CBS and The Automatt.

First, let's look at the procedure used for the tests. Briefly stated, various instruments were recorded through the test pre-amps to a multitrack, and then played back through the studio console. The engineers were then brought in and asked to judge the test pre-amp's performance against a reference pre-amp (not against each other).

Voice		Triangle		Piano	
GML 8300	4	Benchmark 4x4	6.5	Studio Tech MPE	5
Sontec MB-1A	3	Studio Tech MPE	3.5	GML 8300	3
Jensen 990	2.5	Sontec MB-1A	2	Summit TPA2	3
Innovative UTMP-2	2	GML 8300	1	HART MP-501	2
Summit TPA2	2	Innovative UTMP-2	1	Benchmark 4x4	1
Benchmark 4x4	1.5	Jensen 990	1	Jensen 990	1
Studio Tech MPE	1	Summit TPA2	1	Sontec MB-1A	1
HART MP-501	0	HART MP-501	0	Innovative UTMP-2	0

Table 1. *Individual instrument scores.*

Recording sessions

Three separate recordings were made with professional musicians playing piano, triangle and the spoken word. The microphones were connected to the pre-amps with 50 feet of Monster Cable Prolink series 1. Twenty feet of the same cable type connected the pre-amp outputs to the tape machine. The piano was recorded using a pair of Neumann M49s, triangle with a B&K 4007, and the vocal with a Neumann U87. (The latter microphone is a modified version by Klaus Heyne, and the M49s were a matched set that came from Heyne's personal collection.)

Instruments were recorded to a Sony PCM-3324 DASH-format digital multitrack that was equipped with Apogee Electronics anti-aliasing filters (model 940-G "gentle" rolloff on input; model 940-S "steep" rolloff on output).

Some of you may question, based on potential image resolution and aliasing-filter phase problems, the use of a digital rather than an analog machine for this type of test. We have no argument with this and, in fact, are not big fans of digital for the above stated reasons.

We were, however, limited by certain factors in doing tests of this type. First, we must find a studio willing to provide studio time to conduct these reviews. Denny Jaeger was kind enough to make some space available to us, and his studio happens to feature a pair of PCM-3324s. (It should be noted that the replacement Apogee filters made a noticeable improvement in the 3324's phase response.)

Second, the available studio time slot had to be coordinated with the availability of equipment from the various pre-amp manufacturers or their local reps. So, for the sake of science, if anyone out there with a world-class studio and state-of-the-art monitors would care to donate unlimited studio time, please contact us, we would be happy to talk to you.

With the mics in place we proceeded to set a level for the reference pre-amp, and to then match all the pre-amp gains. When we had a good level on the reference pre-amp, a tone was put through, and the output voltage measured. We then set all pre-amps to this output voltage.

Pre-amps that had only stepped gain pots were set to the closest level possible (playback gain was adjusted to compensate). All units were also tested for polarity and adjusted to maintain consistency.

We used the internal phantom power from all of the pre-amps except for the Sontec. On this unit, even though we measured 54V at the outputs, it would not power the B&K. In this case—and with the U87—we used a Neumann phantom power supply.

The musicians were asked to play a specific piece or pattern as we recorded with each pre-amp. For the sake of fair comparison, we listened for consistency and intensity to maintain a similar performance level. We recorded separate passes—rather than splitting the mic to the different pre-amps—because we felt that the splits could lead to loading problems.

For each instrument, the recorded order of pre-amps was scrambled in order to randomize the effects of any variations that could exist in the tape machine. This also made it easier to set up for the subsequent double-blind listening tests.

Listening sessions

The listening tests were also conducted at Denny Jaeger's personal studio in Oakland, CA. Playback through Jaeger's Harrison series 10 console was monitored over a Meyer Sound Labs 833 system in a room that had been tuned with SIM technology.

With the large number of participants involved with the tests, we felt that our technique of record/playback was the most appropriate. Had we been listening for ourselves, our technique would have been quite different. We would have listened to the instruments live as opposed to recorded, and avoided the console altogether by plugging the pre-amp outputs directly to the monitor amplifier. With 16 engineers on call, we had to compromise, however.

The engineers participating in the test were as follows: Fred Catero, John Cuniberti, Karl Derfler, Klaus Heyne, Stephen Jarvis, Ken Kessie, Danny Kopelson, Lolly Lewis, Larry Oppenheimer, Jeffery Norman, Walter Palmer, Michael Raskovsky, Loren Rush, Rick Sanchez, Robert Skye and Jack Vad. These individuals represent a cross section of people from the areas of broadcast, recording and live performance, encompassing the varied musical fields of classical, jazz, pop and rock.

The 16 engineers were seated at the console two at a time while the test operator worked channel mutes. Each pre-amp was compared to a reference model and listeners were asked to judge performance against this reference. The engineers had about 30 minutes to listen to each instrument. Listeners could go back and forth between the reference and DUT (device under test) as often as necessary.

After evaluating all eight pre-amps, the

We feel that personality has a lot to do with the selection of audio hardware.

participants could return again to any one of them for comparison to the reference, or to each other. When they were satisfied that they had heard enough to assemble the scores, we moved on to the next musical selection.

These tests were conducted as double blind; neither the listeners *nor* the test operator knew which pre-amp was assigned to which channel except, of course, for the reference model.

The engineers were asked to judge the pre-amps on several different attributes:

1. *Stereo image* (piano only)—width, depth and definition.
2. *Spectral balance*—linearity, bass extension and treble extension.
3. *Transient handling capability.*
4. *Clarity vs. Obscurity.*
5. *Gestalt* ("gut feeling," "musicality," personal like or dislike).

Listeners were asked to score the device under evaluation from −5 to +5, with the reference pre-amp representing the zero reference. When tabulating the scores, an emphasis was placed on attribute #5, Gestalt. We feel that personality has a lot to do with the selection of audio hardware, and that gut feeling will make the final decision. In order to break the tie in a case where two units may have scored the same on Gestalt, we analyzed the data from

These tests were conducted as double blind; neither the listeners nor the test operators knew where the pre-amp was assigned.

attributes 1 through 4. In those cases, #4 and #3 and linearity (from #2) appeared to be the most significant.

We note that a debate has raged in a number of magazines regarding the audible difference between pieces of equipment that measure similarly, and the proper methods to set up listening tests for these differences. A number of researchers assert that the proper format for such tests involves asking the subject to identify "X" device as either "A" or "B." In other words, each trial is a guess that has either a right or wrong answer.

Our listening tests were set up quite differently; there were no right nor wrong answers. We simply asked the listeners to mark their ballots according to what differences they heard, if any. We were definitely after subjective impressions. We then analyzed the data to see whether there was consistency in their answers.

We asked the listeners to mark their ballots according to what differences they heard, if any.

There is no doubt that our listening group was able to define characteristics of the different pre-amps. Based on the Gestalt scores, 11 of the 16 engineers chose a pre-amp that they liked when listening to two out of the three instruments. This indicates that the pre-amps do have distinct personal-

Need More Information?

For further details of any of the microphone pre-amplifiers evaluated in this Hand-On review, circle the following numbers on the Rapid Facts Card located at the back of this month's issue:

Product	*Number*
Benchmark Media 4x4	*101*
GML Model 8300	*102*
HART Engineering MP-501	*103*
Innovative Audio UTMP-2	*104*
Jensen Twin Servo 990	*105*
Sontec MB-1A	*106*
Studio Technologies Mic Pre Eminence	*107*
Summit Audio TPA2	*108*

ities, and that the test provided sufficient resolution to allow this identification.

We found reassuring the fact that the engineer who provided us with his GML model 8300 pre-amp chose this unit as one he liked in two out of three recordings.

Although the results shown in Tables 1 and 2 are not statistically significant, it's interesting to note that, in several cases, a certain pre-amp may have scored the best on attributes 1 through 4, yet was not the favorite of the engineer. In these cases, some other pre-amp had a certain "savoir faire" that made it more attractive, more musically pleasing.

This leads us to the question of whether the best specifications reflected the best scores? The answer to that question is a definite yes *and* no. The top performers certainly did exhibit quite good measurements (see Table 2 published in the September issue), but good measurements did not guarantee a high score. Of particular interest is HART MP-501, which was close to the absolute top slot in the specs, yet took the bottom slot in the listening scores. This poses some interesting questions.

For the MP-501 we may surmise that specmanship is not everything and personality and musicality must have an influence on our listening. Why else for example, would the Summit Audio TPA2—which exhibits comparatively high distortion, noise and limited bandwidth—score higher in the listenings tests? The use of tube circuitry may present the soft, warm sound that many find pleasing. Of course tubes are not the only answer; the Innovative Audio UTMP-2's score was sitting right down there with the HART MP-501.

***Table 2.** Total scores for eight mic pre-amplifiers.*

Studio Technologies Mic Pre Eminence	9.5
Benchmark 4X4	9
GML Model 8600	8
Sontec MB-1A	6
Summit Audio TPA2	6
Jensen Twin Servo 990	4.5
Innovative Audio UTMP-2	3
HART MP-501 card	2

The Tables provide the order and number of votes cast for each instrument, as well as total votes. Half points were given in the two cases when there was no possible way to discern a winner. We feel that the rankings among the instruments hold more weight than overall points.

You will note that in the instrument listings, the GML Model 8300 and Studio Technologies Mic Pre Eminence each show up in two out of three of the top two positions, whereas the Benchmark 4x4, far and above the pre-amp of choice for triangle, did not fair nearly so well on the other two instruments.

Having had the advantage of recording with all the pre-amps, while setting up the listening test, we mostly chose the units that were ranked at the top by our listening group, with the exception that PS gave a higher than average score on piano. In the case of the vocal, we both felt that, although other pre-amps were more flattering to the voice, the Studio Technologies MPE provided the most accurate reproduction. (Our personal scores are not compiled in the chart; only those of the 16 invited engineers have been included.)

Listener comments also indicated some trends. We had purposely picked a triangle that was extremely rich in harmonics and produced a lot of beat frequencies. This was interpreted by some as IM distortion and was a bit confusing. Several listeners went into the studio and listened to the piano; it was these engineers who felt that we had done a good job of representing the true sound of the piano with our mic placements.

We hope that this information will provide the stimulus for those interested in stand-alone pre-amps to do some additional testing on their own. Several pre-amps came out looking quite nice in these tests. We recommend that they be given careful scrutiny in the comfort of your own studio, to see which best suits your needs and preferences. All of the engineers involved in this listening were all anxious to participate and told us they had an enjoyable time.

With these units available and more coming out on the market, there is a lot of critical listening to be done. R·E/P

The mention of specific products is not to be taken as an endorsement by **RE/P** or Intertec Publishing.

We want to thank Danny Alvarez, piano, Tom Duckworth, triangle, and Orson Wells, vocal, for their help in making the test tape—*BH* and *PS*.

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