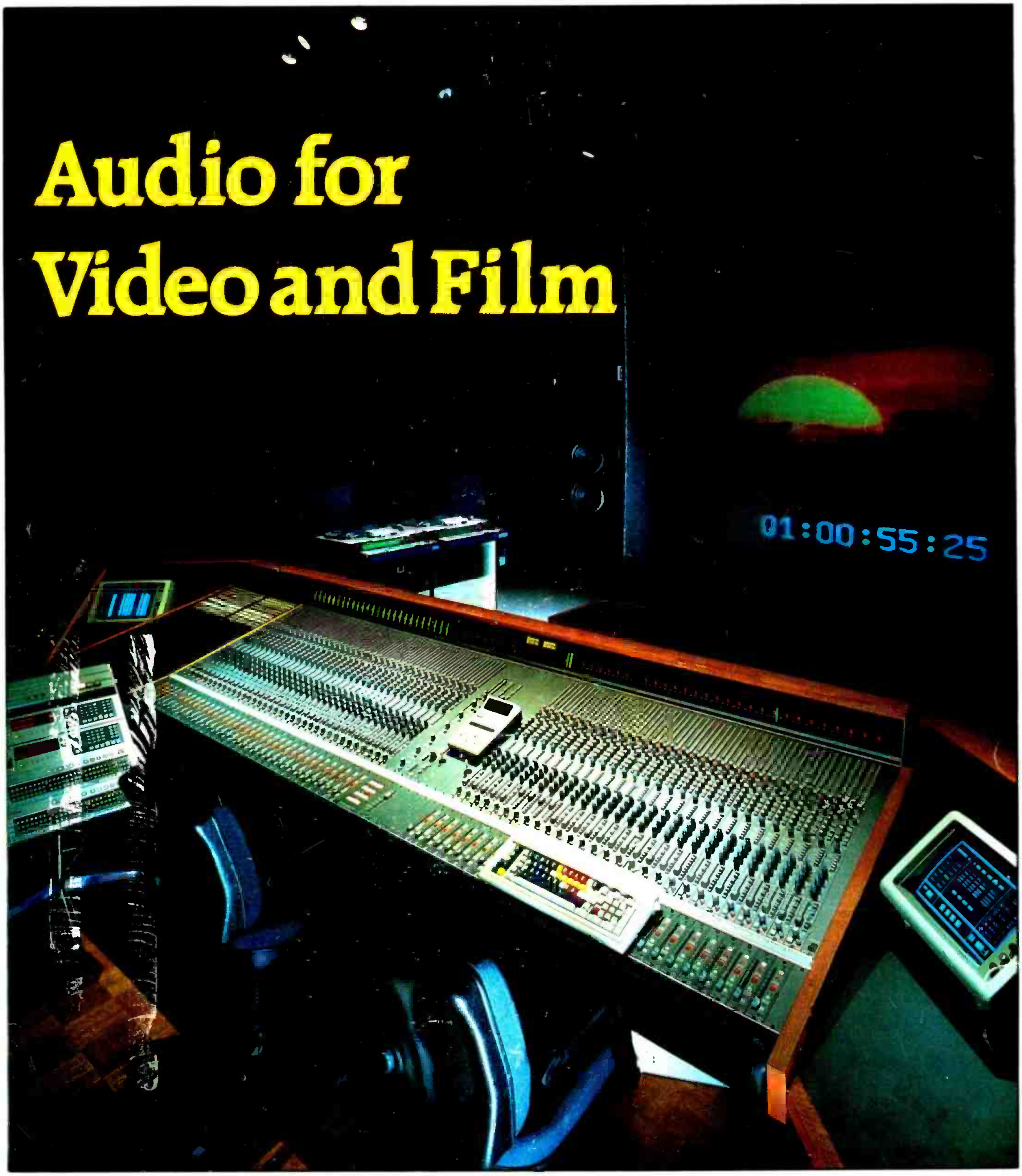
November 1987

Recording ENGINEER/PRODUCER

The Technical Journal for Audio Professionals

AN INTERTEC PUBLICATION

R-DAT Response
Pg. 8



The music industry chose the Ensoniq ESQ-1 as the Most Innovative Keyboard of 1987 ...over the Yamaha DX7II, the Roland D-50 and Korg DSS-1

Each year, members of the music industry are polled for their choices of the most innovative instruments in a number of categories*. For the last 2 years the award has gone to Ensoniq. This year, the Ensoniq ESQ-1 was chosen for its great sound and versatility over some pretty significant competition.

There are good reasons why the ESQ-1 is at the head of the keyboard class. First there's sound—always the top criteria in evaluating a musical instrument. The ESQ's 3 digital oscillators per voice and choice of 32 different waveforms give you an unmatched pallette of tone colors.

If you'd rather just plug it in and play, there are thousands of great sounds available on cartridge, cassette and disk from Ensoniq and a number of other sound developers.

The ESQ-1 is also the only synth in its class to feature an on-board 8 track MIDI sequencer with functions that rival many stand-alone units. And because the ESQ-1 is multi-timbral with dynamic voice allocation, it can play a completely different 8-voice sound on each sequencer track.

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Great sound and versatility. The award-winning Ensoniq ESQ-1 and the new ESQ-M Synth Module. Only at your authorized Ensoniq dealer.

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*The award we're referring to is the 1987 Music & Sound Award for Most Innovative Keyboard/Synthesizer.

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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.

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

TASCAM

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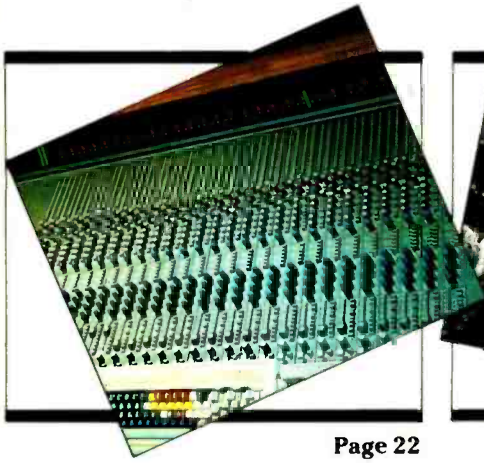

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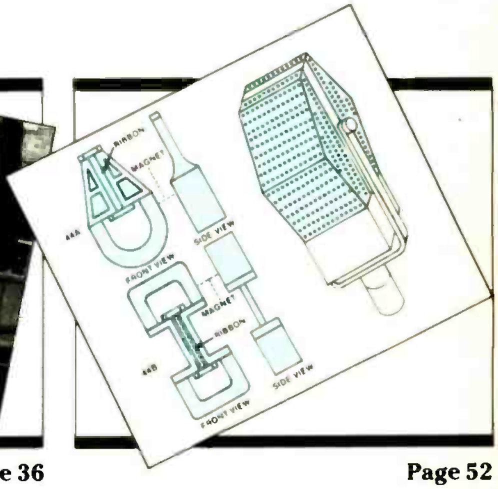

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On the Cover:

Featured is The Master's Workshop, Digital Theatre in Toronto. The facility features a Neotek custom Elite with 64-channels and automated with ARM and disk mix, two Sony 3324 24-track recorders and a Sony 16′ x12′ VPH2002A1 video projector. The room also features 6-channel variable position monitoring. Photo by Sergio Petrelli.

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Editorial

RE/P: The Next Generation

It was Thursday, July 9th. The call came in about 3 p.m., PDT. It was the once-in-a-lifetime, make-me-an-offer-I-can't-refuse call we all hope for.

The caller was Neil Fink from San Francisco, and he had been retained by Cam Bishop of Intertec Publishing (publishers of **RE/P**) to help find a new editor. As he explained, editor Mel Lambert was leaving to pursue other interests, my name had come up as a possible candidate to replace him, and would I be interested in discussing the possibility of a career change?

He went further to explain that Intertec was not looking for a journalist to fill the position, but rather an engineer—someone with working experience in the pro audio environment.

It sounded too good to be true. I knew I was very interested in the position, and I was not going to lose the opportunity through lack of effort. On Sept. 1st, I started my new career.

I was introduced to **RE/P** about 17 years ago. My engineering interests started at the same time in a garage rehearsal room with a 2-track recorder, a tube board with rotary pots, our band's PA equipment and some stereo equipment I'd brought home from Vietnam.

For the past eight years, I've owned Michael Fay Productions in San Diego, which specialized in engineering, production, composition and synth programming services for advertising, broadcast promotion, industrial, corporate and educational clients.

Combined with 32 years of musical experience, I bring to **RE/P** first-hand knowledge and understanding of the equipment, techniques and pressures facing engineers and producers today.

I will use these experiences to focus the editorial content on the real-world topics and problems you face while trying to better serve your clients.

One of my first assignments was to spend a week visiting studios of all types and sizes in Dallas, Nashville, New York, Chicago and Minneapolis. I met with studio owners, managers, engineers and producers, getting feedback on what they felt were the strengths and weaknesses of **RE/P** and other pro audio magazines.

Overall, the responses were very similar to those I had before coming to **RE/P**.

We're all looking for information to better understand our business. We want to know the thinking behind how and why various creative and business decisions are made. Information we can use the very next day. Ideas that stimulate growth, and the application of new techniques.

It is also important for **RE/P** to take a strong leadership role in the professional audio industry. No longer is it enough to simply reflect what is going on around us, as would a newspaper. We must also investigate, digest and disseminate information in a manner that will help guide our industry into the 21st century.

We are going through the most exciting and controversial times in audio history. Exciting new hardware and software systems are being released almost monthly. There is international controversy surrounding R-DAT and the proposed Copycode system. MIDI has not only changed how music and sound effects are made, but by whom. Audio production can no longer be separated into neat sub-classes such as records, jingles, soundtracks, multimedia and broadcast. With the rapidly expanding symbiotic relationship between these areas of production, most studios, engineers and producers must learn to work efficiently and comfortably with each other, in order to survive.

Because of complexity in today's studio environment, it is important that we at **RE/P** stay in touch with all areas of audio engineering and production. I feel it's as important to educate and inform those working with microphones, acoustic spaces and analog tape, as it is with those working direct-to-disk using samplers, sequencers and edit decision lists.

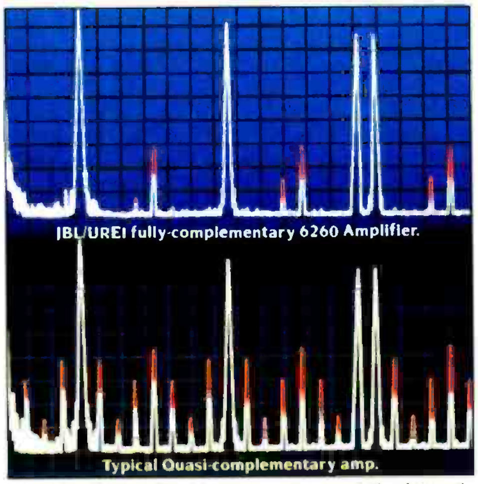
Ultimately, we all speak the same language—sound. **RE/P** will be here to interpret the nuances.

If you have ideas or suggestions on ways we can better serve you, please write to me in care of the editorial office in Hollywood. The address is 1850 N. Whitley Ave., Suite 220, Hollywood, CA 90028.

I'd like to close this first editorial by thanking Mel for his assistance during our transition. Through his efforts and experience, **RE/P** has maintained a high level of integrity and sophistication. We wish him the best in his future endeavors.

Michael Fay
Editor

What To Look For When You Listen To A Power Amplifier.



Red spikes in the TIM Spectrum reveal the dramatic differences in distortion output.

When it comes to evaluating amplified sound, seeing is believing.

In fact, when engineers judge the sound quality of an amplifier, they often rely on *two* precision instruments: the human ear, and the industry-standard Transient Intermodulation Distortion Test, because when measuring sound with T.I.M. what you see is what you get.

And what you see can be eye-opening. Amplifiers that seem to square off evenly spec. for spec., often perform very differently under the scrutiny of T.I.M. Pushed to their limits, many produce brittle, edgy or distorted sound especially during high frequency passages and sharp transients.

Many manufacturers deal with distortion by using massive amounts of feedback through a single overall feedback loop, placing greater demands on the amplifier and producing an inferior sound.

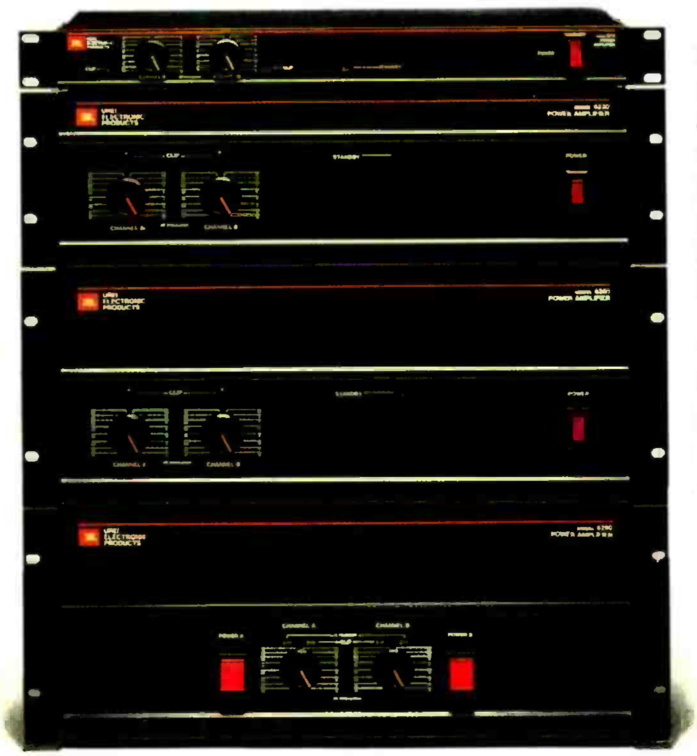
When we built our new JBL/UREI Amplifiers, we committed ourselves to designing the industry's purest-sounding amps that would not only score highest marks on the T.I.M. Test, but deliver the truest amplified sound ever heard.

Instead of sloppily force-feeding massive amounts of output signal back into input stages, and congesting it all into one circuit loop, we've established operating points at *each* gain stage. This allows signal purity to be maintained along the entire circuit. And permits optimized use of the type and amount of feedback for each individual gain stage.

In a simple analogy, the new JBL/UREI Amplifiers do each signal track right the first time, so that you don't have to fix it in the mix. The result is sound far cleaner than typical quasi-complementary and fully-complementary output stages only. And far more pleasing to the ear.

Put JBL/UREI's remarkable new Amplifiers to the test at your local JBL/UREI dealer today. We're confident you'll think it's the finest amplified sound you've ever heard. Or seen.

For an informative Technical Paper on the unique design philosophy behind the new JBL/UREI Amplifiers, please write to:



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News

Live session marks first direct-to-DAT use

Amid continuing debate over R-DAT and Copycode, the industry's first direct-to-DAT recording session occurred in August in Oakland, CA.

The live session with the Blazing Redheads, a jazz-rock group, occurred at Bayview Studios. The session was recorded live to three sources; to a custom analog reel-to-reel recorder for vinyl release master; to a digital recorder for a CD master; and to an Onkyo DT-2001 DAT recorder, creating the DAT master.

Keith Johnson engineered the session, which will be released on vinyl and compact disc on Reference Recordings. The DAT recording was available for listening at audio fairs in New York and Tokyo this fall.

Agfa forms joint ventures

Agfa-Gavaert has formed two joint ventures designed to combat inroads made by Far East manufacturers in the magnetic tape market.

In the first agreement, Agfa has formed a venture with Philips and Du Pont for the manufacture and marketing of audio, video and data magnetic tape, effective Jan. 1. Agfa will own 60% of the venture, and Philips and Du Pont 20% each.

In the second agreement, Agfa will pool resources with BASF for the research and development of selected audio and video products, focusing on future coating technology and cost-effective methods.

The company said that the agreements were made to combat competition from the Far East, which has resulted in a 60% decrease in the price of blank videocassettes in the past five years.

Timeline installations

Timeline has announced the installation of the LYNX VSI video editor interface or LYNX SAL synchronizer at the following facilities: Sheffield Studio, Baltimore; Streeterville Studios, Chicago; MGM-Lorimar, Hollywood; producer Reggie Lucas. The following studios have purchased LYNX modules: Sigma Sound, New York; A&M Recording, Hollywood; and Soundtracks Studios, New York.

Harrison opens West Coast office

Harrison Systems has opened a West Coast operations center, located at 4721 Laurel Canyon Blvd., Suite 290, North Hollywood, CA 91607. Sales, customer support, documentation and spare parts will be located at the office.

People

John E. Stiernberg has been named national sales manager for dbx Professional Products.

James S. Ruse has been named national sales and marketing manager for Stewart Electronics.

John Harris has been named executive vice president/general manager of Centro Corporation. **Curtis J. Chan** has been named vice president of marketing and product development.

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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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Inside THE TAPELESS STUDIO™

Direct-to-Disk™

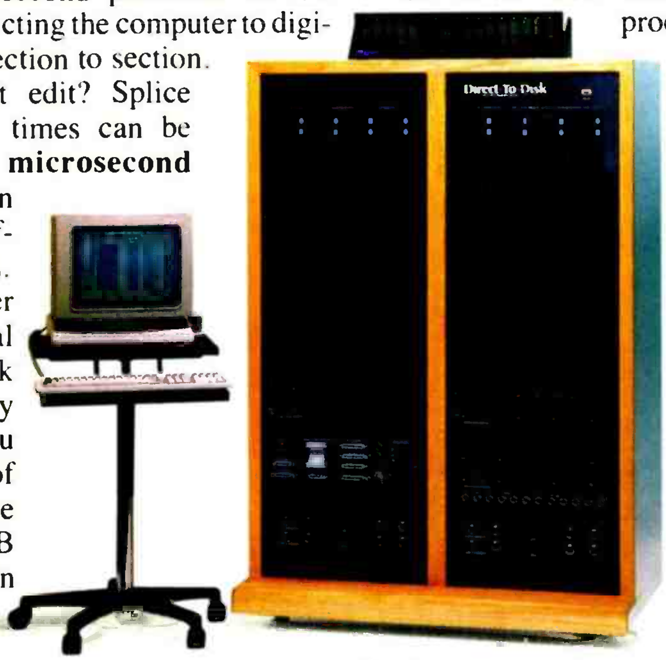
DIGITAL MULTITRACK RECORDER

Already proven in leading studios throughout the world, the Direct-to-Disk Multitrack Recorder is now available in stand-alone, remote operated 4, 8, and 16-track units.

Powerful new software provides fast, flexible **automated editing** features unavailable with conventional tape-based multitracks, such as individual track offsets, auto fly-ins, and multiple loops on every track.

The terminal screen gives a complete, easy-to-read visual display of all track information. Using a mouse you identify splice points with microsecond precision on the display, instructing the computer to digitally crossfade from section to section.

Unhappy with that edit? Splice points and crossfade times can be adjusted with **ten microsecond accuracy**. Or you can define a completely different set of edit points.



Because you never disturb your original tracks, Direct-to-Disk editing is completely **non-destructive**. You can construct dozens of different edits from the same material and A/B each one. Bounce again and again with no loss of fidelity. Even punch-in without erasing. The computer records and logs each move, and can instantly retrieve any pass for comparison.

With Direct-to-Disk, audio information is recorded and stored on a network of reliable, high-speed winchester hard disk drives, which offer not only **superior audio fidelity** and data integrity compared to tape, but superior performance. And because winchester disks are a **random access medium**, rewind, fast-forward, auto-locate and SMPTE lock are instantaneous.

With variable digital sampling rates of up to **100 kHz, 16-bit resolution**, 0.04% distortion and 96 dB signal-to-noise ratio, Direct-to-Disk offers by far the best fidelity of any multitrack on the market today.

The stand-alone Direct-to-Disk is based on the same hard disk storage and proprietary processing technology that has made the Synclavier® the industry standard for reliable performance in the studio and on the road. And like the Synclavier, the Direct-to-Disk system is modular and **software updateable**. As new features become available, you upgrade simply by loading in a floppy disk.

There is only one totally integrated disk-based digital audio recording and editing system for today's music production and audio post-production requirements—the Direct-to-Disk Multitrack Recorder.

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News

Reader survey: 90% oppose Copycode

By Dan Torchia, staff editor

Final results from **REP**'s reader survey on R-DAT and Copycode indicate that readers overwhelmingly oppose Copycode and attempts to legislate its use.

Additionally, the high response rate indicates that the issue continues to be extremely important within the pro audio community.

Bottom line: about 90% of readers who returned questionnaires were opposed to legislation mandating the use of Copycode, and 93% were opposed to government intervention in recording industry technology.

"As the preliminary results indicated, engineers do not want to use anything that would degrade the quality of their work," said Cameron Bishop, group vice president of Intertec Publishing, which publishes **RE/P**. "We think the numbers are significant, and we intend to present the view of our readers to the appropriate people.

"Throughout the entire R-DAT-Copycode debate, the opinions of professional engineers and producers have been unjustly ignored."

Additional results:

- Almost everyone was familiar with the proposed legislation, totaling 92.6% of responses. A total of 7.1% were not familiar, and 0.3% did not answer.
- Although they were opposed to unauthorized duplication of prerecorded material for commercial use, a large majority supported personal, non-commercial duplication. A total of 71.1% were opposed to piracy, while 92.9% approved of private duplication.
- Not only did respondents doubt that Copycode would solve the piracy problem, they also doubted whether the system would work. Only 7.1% thought encoding material would solve the piracy problem, and 85.5% thought that it would not be possible to notch material without affecting quality.
- A majority of survey respondents, almost 60%, listed their title as technical management and engineering. Another 24.1% listed company management, 15.5% listed operations and production management, and 0.5% did not list a title.
- Respondents from the West returned the highest percentage of questionnaires, 31.3%. Returns from the Northeast and South both totaled 26.6%, and 15.5% were returned from the North Central region.

The final response rate was 40.7%, a very good rate for a reader survey, Bishop said.

The survey was conducted by the research department of Intertec Publishing, under the direction of Katy Smith. Questionnaires were sent on an nth, or random, basis to names from the **RE/P**'s circulation list.

Reader Comments

As might be expected with a controversial issue such as R-DAT and Copycode, comments from readers returning the survey were numerous. Many comments were excellent summaries of what readers thought were the most important points of the issue, and combined with the survey information, shed a great deal of light on what the recording community thinks about R-DAT and Copycode.

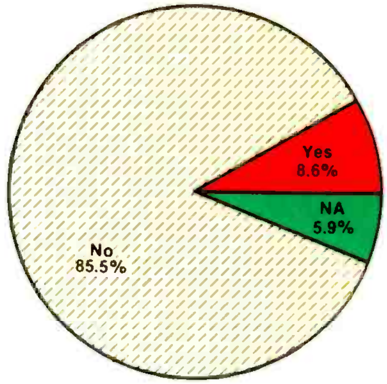
As we did last month, we have listed some of the comments we received below, divided into approval of the Copycode system and disapproval of it.

Anti-Copycode

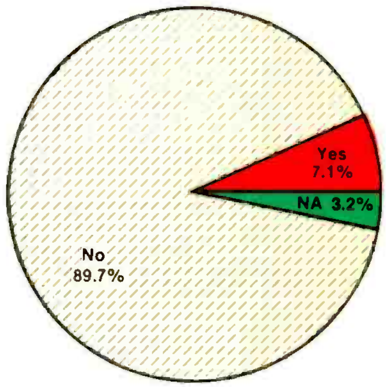
After so many years of striving for a "flat" frequency response for audio production/reproduction, it is crazy to put a notch in it in the "fundamental" range of high instruments. If you must do this "copy guard," find another way.

True pirates will find a way around any copy-coding technology. Look at "black boxes" that unscramble video satellite signals! If copy-coding must be used, don't put it right in the middle of the vocal range. Put it 50 cycles or so below or above 18kHz or 19kHz.

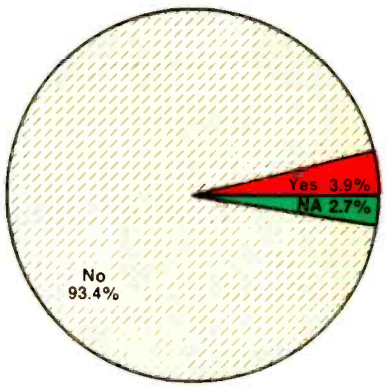
Although I do not believe DAT will lend itself to success in the consumer market, it may prove invaluable to smaller "low budget" studios. If a notch were to be taken out of the bandwidth of the recorder, it would be useless to professional audio studios. The audio industry should be responsible for the quali-



Is it possible to eliminate or "notch" digitally recorded material in a band somewhere between 3,500 and 4,100 Hertz without affecting audio quality?



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



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

***Figure 1.** Final results of the **RE/P** reader survey on R-DAT and Copycode.*

ty standards of audio, not the government.

The movie industry complained about VCRs, [and] wanted legislation similar to R-DAT. The movie industry is quietly happy now that such legislation has not been implemented.

Anyone selling any kind of music (unauthorized) for commercial benefit can be prosecuted under the laws already on the books. The simple fact is the major labels aren't ready yet for DAT—too bad for them.

Since sampling frequency is different than CDs, no direct copies would be possible—if passed, who knows what restrictions would be tried next.

The proposed legislation is analogous to punishing the entire class because of one kid who acts up. It's not fair to compromise the listener's enjoyment. It is well-known that we have a problem with piracy, but to copy-code is to cop out from the task of controlling the pirates. For more than 100 years, the music business and the recording business have been hand in hand—we can't afford to let some well-meaning but misguided legislation destroy this.

Do any proponents of the law realize how many clever techie types there are out there who would (and could) build a box to defeat this protection? For any protection there is at least one person who can break it. And it isn't even a very clever protection scheme.

The major labels should look to themselves for a solution and not put the screws to the audio industry.

Copy-coding could be done using ultra high frequencies that wouldn't disturb the integrity of a mix. Whoever thought up the idea of notching audible frequencies must have his ears up his....

If our industry stops listening, why should anyone else listen to our product. Sonic responsibility is in our laps. If we blindly follow the business end, I fear the worst.

Attempts at legislating ethical behavior have historically failed. Either the innocent are punished or the guilty rule.

This wonderful technology should not be impaired for professional use. Let the pros and consumers use these machines and increase the penalties against convicted pirates.

To limit recording technology advances is a little like regulating the auto industry to protect the horse carriage companies, or regulating GE so that the candle manufacturers could compete with the light bulb.

The beginning of a disaster. There has to be a better way—i.e., tax tape.

Putting a notch in the audio spectrum is a step backward. If the government intervenes in technology, what next? Legislation for "Bring Back Mono" or "Stop MIDI Madness"?

Pro-Copycode

We need to protect our artists from illegal copying by consumers. I am all for any intervention with or without the government, to make consumers buy records, cassettes, R-DAT, etc., without allowing them the opportunity to steal the information by copying. Even if it is merely "non-commercial use," for every copy made, consider that one less sale of a recording project.

If it can be done properly, it should be implemented.

It isn't fair that the industry loses millions of dollars in sales due to consumer copying. No matter what is done, the fact is that copying by the consumer will always occur in one form or another.

Piracy is already threatening revenues to record companies and artists. If illegal duplication of material is not stopped, there won't be as many artists or music.

Home (consumer) duplication of prerecorded analog and digital music should be made impossible to do, by a technical means. Ditto for commercial counterfeiting of prerecorded music.

CBS offered a listening test of the Copycode system on various material. It is good that the system is on a voluntary participation scheme with the engineer and producer determining in the studio if the system affects any program material.

Until digital systems become a home standard, replacing analog recorders, and a digital sub-code command can be established that will limit disc-to-tape or tape-to-tape dubbing, then some "notching/filtering" system should be implemented as a temporary solution.

There is a definite need for copy-coding of some sort, to eliminate commercial copying. This could, however, be handled by a number of means (e.g., parity bit manipulation, filtering at higher frequencies, etc.), which would not adversely affect audio quality for home use.

Write your Congressman

*If you are against Copycode and the law pending before Congress that would legislate its use, you can make your views known. On page 67, the last page of the issue, is a card, preaddressed to **RE/P** containing a letter to Congress. Additional space is included for your own comments. Fill out your name and address, attach postage and mail it in.*

***RE/P** will collect all responses and will present them to congressional leaders involved in the legislation.*


AMPEX
467
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467

Managing MIDI

By Paul D. Lehrman

When is a standard not a standard? In the case of MIDI, it's when a piece of music recorded with one sequencer cannot be played back by another.

In most computer applications, there is a certain degree of compatibility between different software programs, and even between different types of computers, when it comes to exchanging information. Most programs that use text can share it with most others, and binary data can be traded among a variety of normally incompatible software and hardware using communications protocols like Xmodem.

Spreadsheet programs can import and export tables of numbers, and even graphics can be transferred between different computers thanks to "page description" languages like PostScript.

You can't do this yet, however, with MIDI information. If you have an Atari running one sequencer program and an IBM running another, the only way to get them to talk to each other is over a MIDI cable in real time: set one to record and the other to play, set one to be timing master and the other to be slave, cross your fingers and hit the button.

If your software is one of the many programs that doesn't allow multiple-channel recording, you may have to do this 16 times to transfer one song—or even more if you want to keep different tracks that are on the same channel (kick and snare, for example) separate.

If you want to exchange music over a modem, something many artists find themselves doing increasingly often, you're completely out of luck unless you and your exchangee have identical software packages, *and* the communications software you use allows true binary data transmission. If you want to use a network, so that you can upload and your buddy can download at different times, then you have to make sure the network allows that facility as well.

Because all sequencer developers have their own ideas about the best way to structure files, both in memory and on disk, it's no surprise that few programs are compatible. This situation is beginning to change, however, in a movement started about a year ago by one manufacturer of music software for the Apple Macintosh. That company introduced something called "MIDI Files," as a non-software-specific way of storing information about sequences, and included the capability of dealing with MIDI Files, importing and exporting them, into its own software.

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

The idea was submitted to the various powers-that-be for formal approval and inclusion into the official MIDI spec, but in the meantime, the idea of an unofficial standard caught on with other Macintosh developers, and by the time of the June NAMM show, almost all of the Macintosh-based sequencers on display boasted MIDI File capability.

As of this writing, the spec is still not finished. As a consequence, the various programs that use it do so in ways just different enough to make true file exchange between them an iffy proposition—sometimes the files get transferred correctly, sometimes only parts of them do, and sometimes the whole thing crashes.

If you want to exchange music via modem, you're out of luck unless both parties have identical software.

But official adoption is fast approaching, and developers who use other computers (even though there was nothing about the original spec that was overly Macintosh-specific) are looking more seriously at it. There will probably also be a name change, to something like "MIDI Sequence Dump Standard."

One aspect of MIDI data exchange that has been concerning many developers and users is how tempos are described. Ordinarily, a sequencer defines tempos in terms of beats per minute, with tempo changes falling only on individual beats or even bar lines.

For really precise synchronization, especially in conjunction with visuals, this is not really good enough, and the MIDI File spec allows tempo to be described in terms of microseconds per beat, with 24-bit resolution (or one part in approximately 17 million), with changes to occur on every MIDI clock if desired.

In addition, the 24-MIDI-clocks-per-quarter-note rule does not necessarily have to be adhered to in a MIDI File—if a program wants to specify a different time base, it can.

In a multimedia studio, getting tempos to translate from one piece of equipment to another is of paramount importance. It's easy to get tempo changes into a sequence which is being driven by a computer's internal clock, but when an external timebase is acting as the master, the tempo changes have to come from somewhere else. In most cases, that's the responsibility of the device that translates the external clock (i.e., SMPTE time code) into MIDI clocks and pointers that the sequencer can follow.

Such synchronizers contain "tempo maps," which describe the tempo at any particular point in the sequence. Often these maps have to be programmed into the synchronizer by hand through a long and tedious process, with each beat or bar addressed individually. With some of the more inexpensive devices, you are only allowed *one* tempo from beginning to end.

One company that manufactures both synchronization hardware and sequencing software uses system-exclusive data to transfer tempo maps between them, which is one solution, but only if you happen to want to use both their software and their hardware—each is quite useless without the other.

There have been proposals for a universal tempo map standard, but the latest version of MIDI Files includes all the information needed for constructing such maps, and in fact, a MIDI File can consist of nothing *but* tempo information, so the idea of a separate standard is no longer necessary.

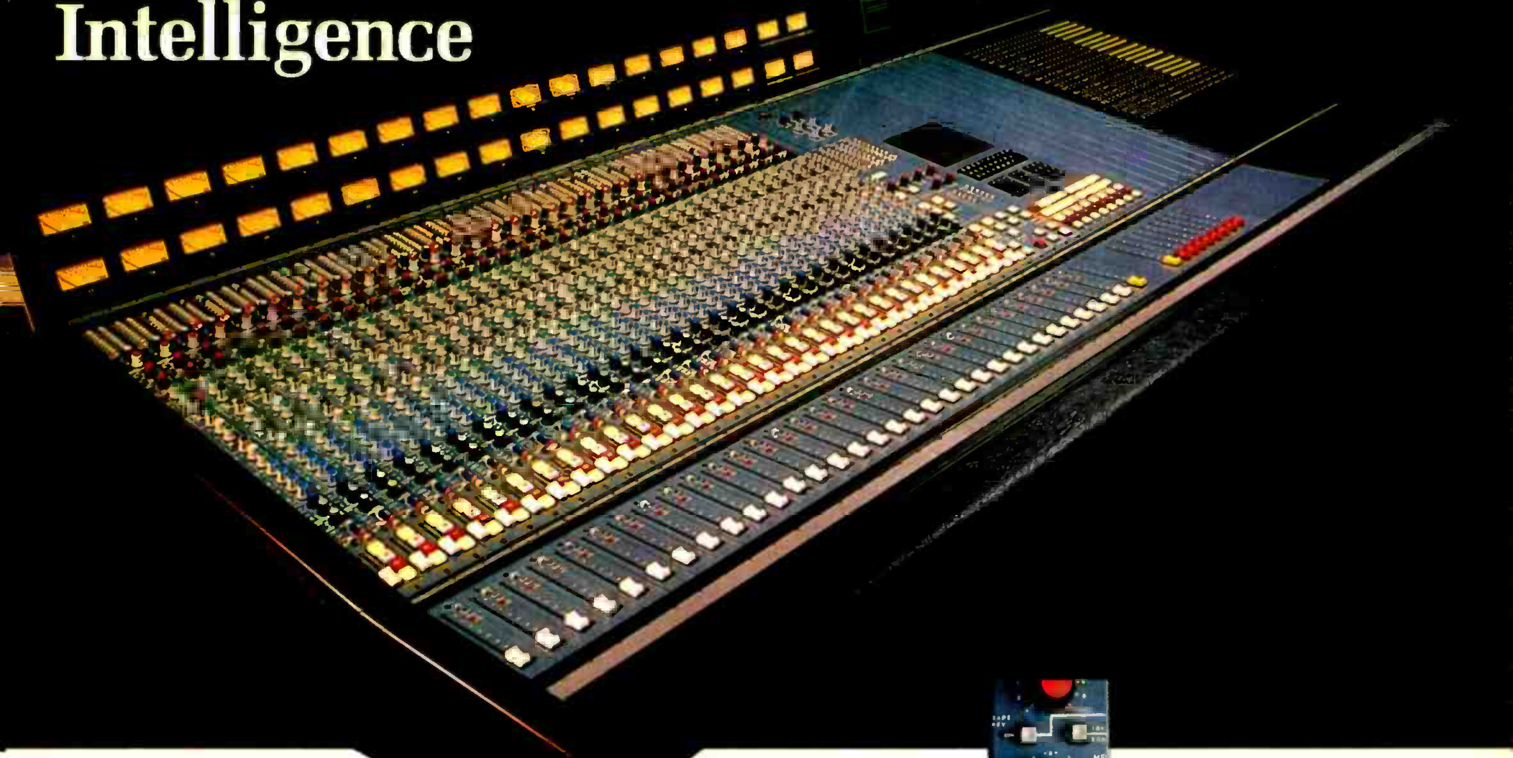
Getting MIDI Files from one place to another on a disc or via modem is only one part of their potential use. There is also the question of transmitting them over MIDI. This might lead some to ask, "Why bother? If you've got two sequencers, why not just transmit in real time from one to the other?"

Well, for one thing, you don't have to worry about getting them in sync, and for another, doing the exchange as a MIDI File can allow you a much higher degree of accuracy.

Suppose you have two events in a MIDI File that are supposed to happen at precisely the same time. If you transfer the file over a MIDI line in real time, those two events will happen *not* simultaneously, but one after the other. But if later you want to use the file in a sequencer that addresses multiple MIDI lines, which *could* send the events simultaneously, you won't be able to take advantage of it—the file will be "smeared." Do this over multiple generations, and you could have serious problems.

MIDI is not music, it's a description of musical events. MIDI Files is not MIDI, it's a description of MIDI events. It may seem like we're moving further and further from the music, but actually we're getting closer all the time. RE/P

Simple Intelligence



Let's assume that results are what's important, not which tools you use.

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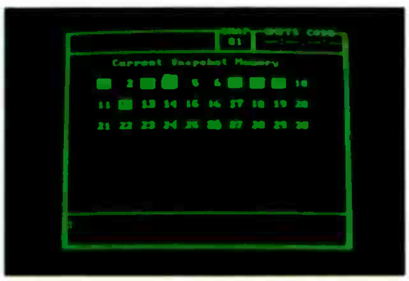

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Sound on the Road

By David Scheirman

Anyone who has ever had to load an expensive concert sound system into a truck and pack into a variety of different venues can attest to the fact that having the sound system is only the first step in the process. The correct packaging of the sound equipment is important not only to make the handling of the gear go more smoothly, but to ensure that such handling does not damage delicate electronics racks and loudspeaker components.

Less than half of the total weight of a typical concert sound system involves the electronics, amplifiers and loudspeakers themselves. The large part of the overall system's weight is represented by custom enclosures to hold and protect the speakers, racks for amplifiers and signal processing, trunks and boxes for stands and cables, and of course the protective outer "road cases."

The value of an investment in proper packaging and road cases becomes clear when accidents happen. The owner of a major concert sound company recently told me the story of a truck full of PA gear that was involved in a semi-trailer highway mishap. Although the truck left the road at full speed and crashed down an embankment, the system itself was entirely useable upon being towed into the loading dock at the next day's venue. The outer shell of many foam-lined road cases was impacted, but there was no significant damage to the system components themselves.

Even such simple items as microphone stands or spare loudspeakers in cardboard

Different types of equipment all require special attention to detail to ensure good "roadability."

boxes can become missiles of destruction if the transport truck is involved in an accident. Proper packaging is necessary from the road technician's perspective, expected by local stage hand crews and is prudent from the equipment owner's point of view.

A variety of manufacturing companies provide the touring industry with custom-built road cases. Most major cities have some type of local plant that builds cases for such items as computers, medical equipment and media gear. Many of these companies are also familiar with sound and musical equipment. Some case companies market their wares on a national basis, setting up distribution networks through music stores and sound companies.

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

Occasionally, as concert sound companies develop a need for greater numbers of road cases and packaging materials, spin-off case companies will be set up onsite to serve both the sound company and the local retail market. Some touring companies have experimented with designing and building their own racks and cases with welded steel or aluminum frames; others

The value of investing in proper packaging and road cases becomes clear when accidents happen.

rely primarily on an in-house woodshop to handle their packaging needs.

Over the years, some innovative ideas have been tried when it comes to packaging sound equipment, ranging from humorous (speaker enclosures built into Anvil-type road cases that were bigger than refrigerators) to dangerous (sharp metal edges on case lids that caused finger and hand injuries).

The existing commercial road case industry has zeroed in on designs that work quite well within the past few years. Even two common objections to the purchase of large quantities of these cases, high cost and the time it takes to build them, are now being overcome.

One case company has recently introduced its proprietary computer software program that is dedicated to the design of travel cases. All measurements for both wood and aluminum materials are calculated by computer and then latch, handle and logo placement are determined. The complete case is displayed on the computer screen, and the case designer can then rotate it to see top, sides and bottom. When the design is finalized, the program generates shop drawings.

Different types of equipment all require special attention to detail to ensure good "roadability." For example, a case for a mixing console will have different requirements than one for power amplifiers.

When cases are designed and assembled for mixing consoles, cabling requirements should be taken into account. If a sound company chooses to not install integral multi-pair connectors on a console for the direct hookup of snake cables, then there will be a wiring harness for plugging into the back panel.

Significant time can be saved if this harness is able to stay plugged into the console from day to day as it is moved. Not having to repatch this harness for each show will cut down on the potential for wiring errors. Thus, a manufactured road case that is custom-fitted to a stock mixing console may be worthless to a professional touring sound company.

Proper packaging techniques for power amplifiers can pose an interesting design challenge. If large amplifier units are stacked up in a single-wide rack, the resulting package can be both top-heavy and awkward. When eight or 10 of today's heavy, traditional technology amps are packed in a rack that is then enclosed in a foam-lined road case, the type and placement of handles that may work on a light guitar case will have a relatively short life expectancy on such a heavy package.

Some of the newer, lightweight amplifiers are helping to make this part of the concert system more easily roadable. One major touring company is able to pack such units double-wide in a hinged case, resulting in a powerful and compact package.

Cable systems can present case designers with challenges. Oftentimes, the best way to carry a cable is not necessarily the best way to package it for daily use. Power distribution systems and snake cable runs both require extensive thought and planning efforts before the construction of road cases. Will the cable be removed from the case entirely for use, or stay partially attached? Is the cabling system set up in modular sections of varying length, making the system more flexible, or must the entire 250-foot length of feeder cable be removed from a case each day even if the power panel is only 10 feet away?

Protecting the significant investment that is represented by a modern concert sound system is an important part of a PA company's business operations, and the correct design and build-up of packaging technologies deserves attention.

The equipment will then not only be protected from "road hazards" such as trucking accidents and loading dock falls, but those persons responsible for moving and setting up the gear will be able to accomplish their jobs in a more safe and efficient manner. RE/P

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Dennis Milan
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SPARS On-Line

By Guy Costa

The industry's most rapidly growing—and evolving—area is audio for video and film. As someone responsible for providing the audio services for a broad spectrum of visual projects, I'd like to share with you some practical insights that I've learned over the years.

This growth area reflects how our industry has been changing. Last year, SPARS decided to change its name from the Society of Professional Audio Recording Studios to the Society of Professional Audio Recording Services. The purpose was to accurately reflect the broadened scope of the audio industry.

We no longer just rent studio facilities—we sell service, and our clients expect these services to be nothing short of professional, accurate and competitively priced.

SPARS didn't decide overnight to make this change. We contemplated it for more than a year and we didn't make a final decision until we were sure it was the right choice.

Likewise, you have to use the same process. Whether you get into audio for film and video as a new venture, or rather as an extension of your present business, the venture must be undertaken with an understanding of the skills, facilities and services required to support this very specialized market and clientele.

There are some hard realities you will encounter if you decide to go into this business, and those of you who do enter it should do so only after evaluating all of the options and developing a sound business plan.

Audio for film and video is the new growth area for the recording industry, and those with future vision are making their five-year plans right now.

Basically, audio for film and video is the business of creating, synchronizing, and recording and mixing sound for use with visuals. It can run the gamut, from jingles and voice-overs for television commercials to soundtracks for sophisticated corporate productions.

With so many areas available for exploration—and exploitation—it is first necessary to identify the specialization that your expertise (or ego) will find most rewarding and the one that will fit your budget.

On the technical side, it requires the mastery of every acoustic discipline and recording technique imaginable, plus a lot of common sense. Most importantly, you need the ability to absolutely synchronize the audio and visual components.

Guy Costa is vice president of operations for Motown Record Corp. and is president of SPARS.

In its simplest form, synchronization might include locking up synthesizers to a ½-inch video workprint (which would include a sync reference or SMPTE time code) and doing a live mix down to one or two tracks, while simultaneously transferring the code to an open track. A more sophisticated system might include 1-inch video, a SMPTE synchronizer, computers and synthesizers with MIDI capabilities, multiple multitracks, 35mag recorders and a variety of digital recorders.

Whatever you use, the primary thing to remember is the KISS formula: Keep It Simple, Stupid. To keep all these elements in-sync, you must be familiar with the numerous analog and digital formats, the nuances of color and black-and-white, the various frame rates and their complex interrelationships.

Audio for film and video is the new growth area for the industry.

Another concern is understanding your particular market. When you identify your niche, research the kinds of technical facilities your clients require for complete "wrap-around" service and support. Once a client has to leave your facility to have additional work done elsewhere, you run the risk of permanently losing that client.

This is not to say that you must have every service available to the client, just those concurrent technical facilities that wrap-around your specialty or niche. An example in audio for video is having a 1-inch VTR available for laydowns and laybacks. If scoring is your chosen niche, then you might consider having the capability to electronically add a "streamer" or "pops" to the video workprint; if it's film, it's being able to mixdown from multitrack to 35mm mag.

Decide whether your marketing strategy will focus on quality or quantity. Determine your profit margins for each type of service you provide, and focus on a service mix that complements each other and caters to similar clientele.

Rock 'n' roll and advertising clients normally don't relate well, and neither do records and scoring. Not only are the clients different, but so are the services and the personnel necessary to support them. This is not to say that you can't build a successful facility catering to diverse clients. It's just easier and more profitable to specialize whenever possible.

At Motown, we have every conceivable A-V format, from 16mm and 35mm telecine through high-end digital synthesizers, direct-to-film mixing, and up to five rooms interlocked with eight 24-track recorders. With all that flexibility, however, our smaller "off-line" rooms are still the busiest and turn the highest profit.

The bottom line is choosing the people and equipment that provide you and your clients with the most cost-effective method of getting the job done professionally. The competition may have more gimmicks or a fancier reception area, but it's quality results that will bring the clients back.

A steady, consistent client base is essential to long-term success. Remember: You are selling highly personalized services, not just studio time.

Whether you are deciding to get into audio for film and video for the first time, planning an expansion of your business or just evaluating the market potential, approach it with a clear business perspective. The choices you make will have repercussions for years to come. Making them in a vacuum, not having accurate information, funding or the proper facilities is foolhardy at best, and stupid to say the least.

Ask yourself these questions:

How much will it cost (including time, energy, money and personal stress)?

How much profit will it generate?

Decide whether your marketing strategy will focus on quality or quantity.

Will it be worth it?

Focus on the realities and determine your success in advance. Don't just jump in and expect to become an overnight success. You are part of a mature industry, where the competition is tough and smart, and you need leverage to get a piece of the action. And there won't be any help from the IRS—you can no longer afford to be in the business for the tax breaks.

Researching all of your options in advance is the easiest way to minimize the risk, identify the realities and maximize your rewards. The best and least expensive method is to create a comprehensive business plan, one that matches your resources and long-range goals. RE/P

Film Sound Today

By Larry Blake

Dear Producer/Director,

At some point, whether you like it or not, you will have to put a soundtrack to your movie or TV show. You probably are more responsible for the overall quality of the sound job than you might think. Hiring an Academy Award-winning crew, releasing your film in 70mm or using digital recorders—these will ultimately have little to do with the quality of the soundtrack.

There are many talented people working in film sound, and they need only half a chance to give you a sound job that everyone can be proud of. What follows are some suggestions that might be of help.

• Establish a dialogue before shooting among the production sound crew, the supervising sound editor and the re-recording crew.

Even one such meeting can save untold time, money and, perhaps most importantly, grief. Classic problems such as improper synchronization techniques in musicals, not recording enough sound effects on location and ordinary production dialogue snafus can be anticipated and dealt with before they cost you, in the broad sense of the word.

• Bring on your supervising sound editor early.

In England, it is standard for the supervising sound editor to come on the show on the *first* day of shooting. Should you be so bold as to do this, you will give that person the priceless opportunity to collaborate with the production team on location effects, wild tracks, etc. The supervisor will not only have the time to search libraries for useful effects, but will also be able to record new material if necessary.

This is especially important, for example, when the company is renting such items as horses, crowds or period autos. It is a lot easier and *cheaper* to record them during production than it is to track them down months later and arrange a recording session.

Although mag-based sound editing has a tough time keeping up with picture changes, the supervisor would still be able to try out broad strokes, especially with background effects, which would be incorporated in and improve the quality of the temp dubs and scratch mixes.

Even if your budget will not allow you to bring on the supervisor during shooting, hire him or her early enough to be able to assimilate the film before having to deal with the day-to-day interruptions brought on once the army of editors and assistants is hired. Also, a rapport with the director can be established only through many meetings.

• Beware of any person or company that claims to be able to give your film an "all-digital" soundtrack.

Current digital technology has not quite caught up to, and shouldn't be confused with, the *potential* offered by digital editing, and brings with it as many compromises as it does benefits. You could be purchasing a bill of goods, not a guarantee of quality.

• Mix in stereo.

There is no excuse for any film to be mixed in academy mono. Here are some of the more popular lame excuses that have acquired a cockeyed badge of truth by virtue of being repeated so often: "This isn't a musical." "We don't have any battles in space." "We don't have the money." "This is a quiet dramatic film; we don't want to obtrude on the dialogue with sounds coming from around the auditorium."

Ear-splitting soundtracks turn the public and many in the industry against stereo mixing.

First, consider the cost factor. The average studio film costs approximately $15 million, not to mention millions more for print and advertising. The surcharge for a stereo mix is about $50,000, including license fees, extra mixing time and mag stock. I simply refuse to see how any film budgeted at $3 million or more cannot find that money somewhere.

The problem is in how stereo mixing is performed, not in stereo mixing itself. I think that ear-splitting soundtracks turn the public and many in the industry against stereo mixing. It is too often assumed that bombastic effects come with the territory, that stereo equals loud and obnoxious.

Thus, reviewers complain about the "loud Dolby music" as if mixers and projectionists push buttons at various points in the film to "Dolby-ize" the track. Stereo and the subtlety of rich, quiet music and effects are not mutually exclusive.

Stereo optical prints don't cost one cent more than mono optical prints, and almost all first-run theaters are equipped for stereo reproduction. The lucrative and inevitable world of home video guarantees that the stereo mix will be heard and appreciated long after the theatrical release. Your money will be well spent.

However, to me and my idealistic ways, the big issue is the sound. First, listen to your favorite film score with the treble control turned down, and you'll get an idea of what will happen to all dialogue, music and sound effects if you mix in academy mono.

Most of the films I see these days are mixed and presented in stereo, and the few mono mixes that I hear are constricted and colorless by comparison. A mono mix cheapens a film, and movies like "Ishtar" and "Children of a Lesser God" have no business being in mono. When watching an academy mono film, I am sometimes confused as to whether the music I am hearing is underscore or source music that was futzed.

• Remember Ecclesiastes.

To paraphrase the Bible (as the Byrds did), there's a time and place for everything. Producers and directors should realize the myriad problems (great cost being only one) that they introduce when continuing to cut the picture once the editing has begun. Likewise, the modern habit of selecting or altering effects while at the dub stage turns it into a Moviola that rents for $800 an hour.

The act of "locking" the picture and making the black-and-white dupes should mark the beginning of the sound editing season. The key word is should. Of course, late-incoming optical effects means that footage will have to change. But if you can avoid last-minute aesthetic cuts, you will maximize the time budgeted for original sound work. Generally, your sound labor budget does not include change work, so track changes mean overtime labor.

Producers, make your directors lock the show. Directors, please learn to let it go. Sound editors frequently spend more days changing tracks than originally cutting them, having been previously told by the producer that "we don't have any money to do this show."

Space considerations prevent my dealing with other, equally important topics, such as giving the production sound crew a chance to do their job properly; hiring a sound effects recordist for a few weeks to record effects specific to the film; and reprinting the dialogue track for the sound editors.

Don't listen only to me; bring on your sound team early and pay attention to what they say. If you've hired good people, you and your film will find the effort worthwhile. RE/P

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

Digital Sound for Motion Pictures

By Doug McKenzie and Bob Predovich

Although digital post-production appears to have found a home in the higher budgets and controlled playback environments of such mediums as exposition films, it could be years before any practical broadcast and theatrical applications will be realized. Despite this, a growing number of facilities are directing attention toward electronic post-production to increase audio quality, as well as to meet deadlines and budgets.

Many producers are already taking advantage of such services. In fact, it would appear that the digital buzz word is more prevalent in such circles than stereo was when first contemplated. Today, there are hundreds of digital 32- and 24-track recorders in facilities around the world.

Obviously, even when combined with the high number of 2-track digital mastering systems, such hardware represents a small percentage of the total number of recorder/reproducers that exist in the field. However, no one will deny that digital sound is a phenomenon whose time has come.

The Master's Workshop's first commitment has been to bring digital sound to the feature-film community. This move is in consideration of the high standards for sound quality that are already prevalent, and the budgets available for such projects. Although a great deal of experimentation and development is necessary to bring about suitable digital playback systems for movie theaters, this hurdle may soon be overcome.

Maintaining digital integrity throughout the various stages of recording, editing, scoring and mixing of the soundtrack can save an estimated six or more generations, given the number of record, transfer and pre-mix stages in the post-production process. Now factor in the number of tracks present in a feature production, which can easily total more than 100, and it's easy to appreciate the potential degradation common to analog sound.

The introduction of digital tape to the electronic audio post-production process, as well as new concepts of machine control, has vaulted the capabilities of sprocketless systems well past those of their traditional counterparts, particularly in the area of sound editing for dramatic television and feature films. At the same time, the evolu-

Doug McKenzie is president of The Master's Workshop, an audio recording and post-production studio in Toronto. Bob Predovich is vice president of The Master's Workshop.

tionary nature of the system design allows for a natural transition from the manual techniques used for decades in the film industry.

Following are case studies of three projects that have been completed using this approach.

Bruce Nyznik, the award-winning sound designer for the film "Discovery," came to us with a unique challenge. As a long-established film sound editor, Nyznik already had people and facilities in place to launch a massive assault on a major project in a minimum amount of time. With the need to finish the film for a rapidly approaching opening day of the Expo '86 in Vancouver, British Columbia, a team of editors was prepared to work around the clock.

There was a major problem, however: This film's soundtrack was to be totally digital. The question for us was how could the producer make use of what was already in place, and still deliver a 100% digital track? The Master's Workshop developed a solution: the proprietary Digital Audio Conform System (D.A.C.). As a co-designer with Andy Staffer, of the Soundmaster Audio Editing System, I knew that the capability existed to auto-assemble a soundtrack from an Edit Decision List. So, we needed to devise a way where the *sound editors* made the edit decisions using traditional film sound hardware and *we* recreated them in a digital format on our 3324s.

D.A.C. involves producing 35mm mag sound work prints of digital time code originals. The time code from the digital master is transferred onto a track that is unused in the 35mm edit process, simultaneous to the dubbing of the guide sound. Traditional sound editing then takes place, with the editor unaware that the time code is also being cut with each splice.

Multiple dialogue and music and effects reels were prepared as in a standard film mix. But instead of these analog tracks being used for final mix purposes, the numeric information they contained generated the edit list. Proprietary systems were developed to accurately determine "entry" and "exit" points in increments of 1/100 of a video frame (1/3,000 of a second) and offset information to position the source reels correctly for every edit.

This time code data was keyed into the edit decision list in an off-line room and stored to disk.

When the lists were complete, we went to the digital "on-line" stage. Here, a film to tape transfer of the picture was interlocked with a pre-striped 3324 tape and the source PCM digital time code masters. Each list had been stored to disk under its film reel number. We would recall "Film X" and command "Execute" for the list. The machines would automatically roll to their respective GOTO points, with appropriate offsets, play and perform the record in and out, just like a 1-inch videotape conform. The guide analog track could be quickly referenced, to ensure that the system was accurately recreating the original.

The process worked flawlessly. The film sound editors were astonished to hear their mono guide tracks transformed sub-frame accurately into digital stereo. What was most gratifying is the precision the system exhibited in recreating the original, whether taking frames out of a long "r" in the middle of a word for proper ADR fit, or in following the numerous edits needed to patch together the sync in a lasso twirling scene.

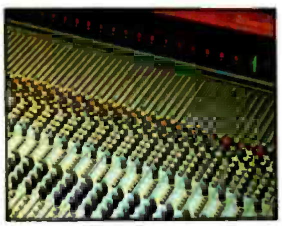
We proved that there is at least the same amount of precision available in our electronic systems as in traditional film methods (technically, there is much more). We were able to exactly duplicate what had been cut on the bench.

The result was a unique travelogue film. "Discovery" is a 20-minute sight and sound experience produced by Independent Pictures, Toronto, for the British Columbian government. In a short time, the film develops characters and tells the story of a young girl who is visited by an adolescent extra-terrestrial in the form of a large self-propelled red ball. From within this mysterious capsule, she discovers things about her province that are as surprising to her as to her inquisitive visitor.

The audience captures aerial views of some of the most beautiful landscapes in the world and also feels the details of acceleration, bumps and near misses during the travel. The clarity can be attributed to the all-digital track and in part to the use of the Showscan process developed by Douglas Trumbull, which enables 70mm film to be handled during the shooting and projection stages at 60 frames per second.

Showscan also provided a system that

Yuri Gorbachow, engineer during ADR session for "Sword of Gideon."



James Porteous and Paul Massey, re-recording engineers, during the final mix of "Sword of Gideon" for HBO.

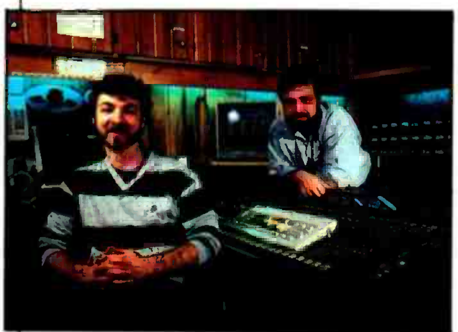
with the guide sound that had been cut along with the 35mm picture. This EDL was loaded and stored to disk in an off-line interlocked six channels of digital sound on three compact disc players synchronized to the 70mm projector.

Digital and dialogue

Though digital technology benefits all aspects of post-production, it is most useful in facilitating the flow of dialogue processes. Typically, sound effects and music can be recorded under controlled conditions, but in original scenes, dialogue recording is often not. Here, the level of quality is directly related to circumstances that are sometimes out of the control of the recordist. Though productions such as "Chasing Rainbows," the first HDTV drama series, are recording their dialogue digitally, most are still analog. This leaves signal-to-noise problems that can compound through multiple analog generations of pre-mixes and mixes.

The ability of digital technology to provide for numerous generations that are theoretical duplicates of the original, without the addition of spurious elements such as tape hiss, opens up a realm of possibilities. Loud and soft sounds recorded on the set can be maintained with the same dynamic range throughout the post process without the fear of noise getting in the way a few generations later. Noisy originals do not compound with every dub. In fact, carefully recorded analog originals married to the final picture via an interlocked digital multitrack can deliver breathtaking results.

Such was the case in "Sword of Gideon," a recent HBO/CTV mini-series starring Rod Steiger and Michael York, on which we conformed the ¼-inch originals directly to digital multitrack. This project had been shot around the world using mono and stereo Nagra recorders, with pilotone referenced to 60Hz and 50Hz. We first restriped the 130-plus ¼-inch reels with synchronous SMPTE time code. Then we developed a proprietary process to produce an edit decision list, matching the ¼-inch takes room, for later recall during the "on-line" conform.



Rick Ellis and Ralph Chiaravalloti creating EDL for digital audio conform process for the film "Discovery."

Master's Digital Mix Theatre incorporating a Soundmaster Editing System, a custom 64 input automated Neotek Elite console, variable 6 channel monitoring and twin 3324 multitracks.

Rather than simply recreate the guide track frame-by-frame, our editors were challenged with correcting clipped words, and other traditional dialogue editing functions as well. We determined that one way to eliminate many historically troublesome areas of dialogue editing/pre-mixing was to provide for ramped transitions in and out of each cut. The criteria we set would allow for variable rate fade-ins and outs, while maximizing "air" at the beginning and end of each slate. In this way, no "foreign" dialogue would be conformed, yet ambience overlaps would occur between most slates, providing for smooth segues. This determination was also made off-line, and GPI events entered into the list to trip an outboard autofade device.

A track layout was developed to facilitate a minimum amount of source reel changes during the conform. This allowed for the ¼-inch sound reels to be spent onto the digital multitrack, and then retired. What resulted was a mosaic of tracks, pieces of a jigsaw puzzle that began to create a complete sound image as we neared the conclusion.

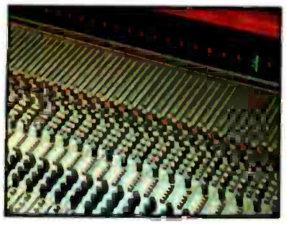
Not only did the digital technology allow us to duplicate (without generation loss) the original master audio tapes, but it allowed another significant benefit to our process. We had used eight tracks and there were obviously huge gaps between slates on the tape. Because every cut already had its own fade in and out, much of the dialogue premix drudgery could be eliminated by simply opening up all eight channels and letting the tracks cross-fade with each other. This was possible without noise concerns since the empty space between cuts on the tape contained digital silence.

The results were unique. Nuances that might have been lost through analog generations were apparent and it felt like the actors were in the room with us. Film dialogue editors who heard the track were amazed, especially with one location scene that had a distant low level drone present. The conforming process was so smooth that multiple slates blended together to keep the drone constant in a faders up interlock.

In our case, we have approached every feature production differently, depending on the technical requirements, problems and budgets. The film "Discovery" is a unique example of enveloping a traditional 35mm film edit process into a complete digital system. This same concept is currently being employed in the production of "Parallel 50," which will be the first all-digital soundtrack developed for an IMAX film.

More typical of our experience in complete electronic or time code based editing and mixing is our work on the digital application for "Chasing Rainbows," produced by

NorthernLight & Picture in conjunction with the Canadian Broadcasting Corporation. The 13-part series is being shot and edited using Sony's High Definition Video process, which produces a stunning 1,125 line resolution, and enables "film style" camera and lighting work. The original sound is shot to a Sony 1630 PCM digital processor and stored on ¾-inch videotape with time code corresponding to the HD video master. In post, the original scene's sound is conformed along with the picture to another 1630 and delivered to us, essentially first generation.



Our process involves transferring this dialogue conform to the 3324 multitrack format and then beginning to layer the many other sound elements necessary to create a seamless dialogue track, a fluid and synchronized movement Foley effects track, a detailed and dynamic specific background effects track and, of course, a digitally recorded and edited score. The work was accomplished using three Sony 3324 digital multitracks for the recording of Foley and dialogue elements and an interlocked BVU 800 linked to a 1630 PCM processor was used to edit original and library sourced digital sound effects with sub frame accuracy.

Ken Nelson, technical director, and James Johnston, technical engineer.

The mix was done on two locked 3324s containing all the original sound components. We mixed within these 48 digital tracks, and additional open tracks were used repeatedly to slip sound during the mix. In some circumstances, we would create an offset between the two transports for the desired time shift, and print the sound from one machine to the other. When reset to a zero offset, the sound would then reside in both the original and new positions with no generation loss due to the digital process. Depending on the length of the sound to be shifted, there were also many occasions when our mixers would simply dump the track into our Publison sampler, and, via Soundmaster's GPI system, trigger playback frame-accurately at a new address. In either case, no one would have to leave his chair to make these adjustments.

For the purpose of the initial broadcast commitment, the final digital master mix was converted to analog, and restriped to NTSC and PAL 1-inch videotape pull-downs of the high definition master.

"Sting" on screen in "The Police Synchronicity Concert," one of the concert specials produced at Master's Workshop.

The aesthetics of digital

In general, we are pleased with the results of our first year of digital sound post-production. The evolution has been natural in that our processes have always been time code based and electronic in nature. There will be need for analog for many years to come, particularly in applications to the broadcast industry. We have found that the digital processing of sound in comparison with analog represents an increase in total cost of approximately 25% on the post-sound budget. It does not necessarily require additional time, although there is certainly an orientation factor present here and a learning curve becoming evident.

Certain assumptions previously held by producers and sound personnel with regard to analog processing need to be re-evaluated. For example, the typical compensations made to overcome degradation through generation loss such as over-emphasized top end and noise filtering are no longer valid. Some feel that digital is less forgiving, more revealing and thereby sterile in delivery.

One can hardly fault the medium for representing the truth. In so doing, it simply becomes transparent, which in any event should be the objective. RE/P

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To boost high-frequency output we utilized a magnet with the greatest flux density available, plus an optimized, balanced magnetic circuit to "stiffen" the coupling between the amplifier and the diaphragm. The resulting increase in high-end response also solved the problem of definition and articulation, so the sound is cleaner and livelier, with better transient-handling capability. As a result, trashy instrumental and vocal sounds are consigned to the trash can, where they belong.

The 10 kHz breakup you've heard in our competitor's driver was eliminated by using a 3-inch diaphragm instead of the other guy's 4-inch component, moving the primary diaphragm breakup point all the way out to 16 kHz, well beyond fundamentals and first harmonics.

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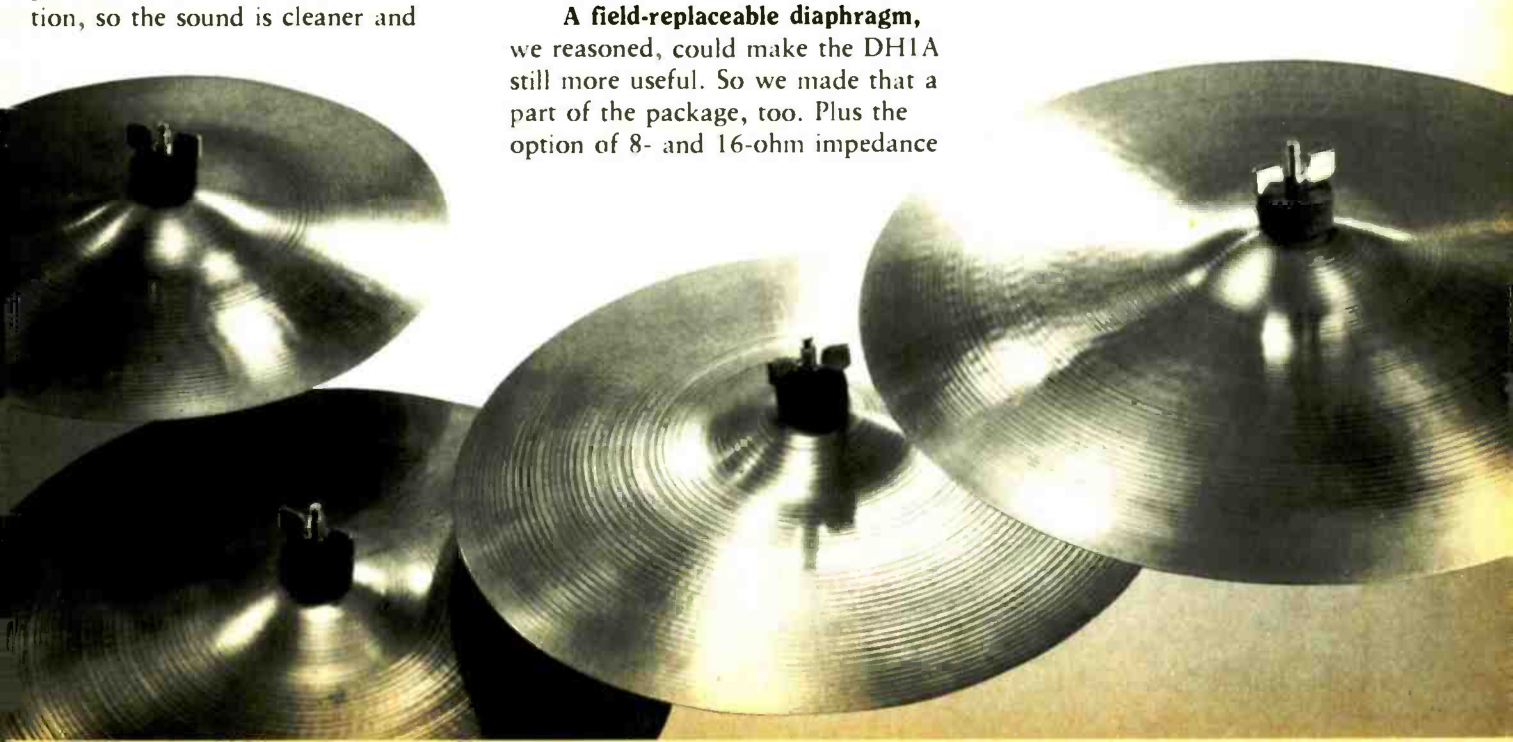


Quantec QRS/XL Real time Digital Signal Processing System

By Bob Hodas

Regular readers of **RE/P** may recall my review of the original Quantec Room Simulator published in the December 1984 issue. Since then, the QRS has become known for its high-quality reverb and ambience simulation. The unit can now be found in many recording, post-production and film rerecording facilities around the world. In the meanwhile, however, Quantec has not been resting on its laurels. Recently, the company introduced the QRS/XL.

The XL is a unique piece of processing hardware in that it is indicative of a future wave in audio processing devices. It is basically a "black box" digital effects unit that can be controlled from an external MS-DOS compatible personal computer. Soon, Atari- and Mac-compatible software will also be available for the XL. The PC connects via an RS-232C serial port on the back panel.

The XL can also be operated without a PC because the internal programs are accessible via a front-panel rotary switch. (If parameters are to be manipulated, or new programs designed, an external PC is necessary, however.)

Bob Hodas is **RE/P**'s evaluations and practices consulting editor.

Internal programs

The 1U unit comes loaded with 90 editable factory presets, plus an additional 30 user storage locations. Of course, with the addition of a PC, storage of programs is restricted only by your floppy disk budget.

The XL not only contains the original QRS program algorithms, as can be seen from Table 1, but many additional new reverbs and effects such as filters, flanging and gating. (It should be noted that Quantec's U.S. distributor, Marshall Electronic, modifies some user ROMS, such as the plates, to better suit the domestic market.) Also included in U.S. versions are 15 extra plates and some other special-effects programs not listed in Table 1. (If you need something specific, I understand that XLs are available with custom programs designed to specifications submitted by

```
———————————— QUANTEC QRS/XL CONTROL PANEL ———————————— 5:37.45p

   Muted          62 FIR:   Telephone Filter 2 (ISO)          Online
                       f[lo] = 226 Hz  f[hi] = 2800 Hz          XL-A

 Midi program configuration
 MIDI       0   10   20   30   40   50   60   70   80   90  100  110  120
    0       0   10   20   30   40   50   60   70   80   90  100  110  120
    1       1   11   21   31   41   51   61   71   81   91  101  111  121
    2       2   12   22   32   42   52   62   72   82   92  102  112  121
    3       3   13   23   33   43   53   63   73   83   93  103  113  121
    4       4   14   24   34   44   54   64   74   84   94  104  114  121
    5       5   15   25   35   45   55   65   75   85   95  105  115  121
    6       6   16   26   36   46   56   66   76   86   96  106  116  121
    7       7   17   27   37   47   57   67   77   87   97  107  117  121
    8       8   18   28   38   48   58   68   78   88   98  108  118
    9       9   19   29   39   49   59   69   79   89   99  109  119

                                              Function Stereo Delay
 MIDI          Configure  Transmit  Receive   Load  Save  Enable  Disable

                                       F7Clear F8Mute F9 -  F10 +
```

Figure 1. *Actual screen dump of the XL's MIDI Mapping function.*

```
———————————— QUANTEC QRS/XL CONTROL PANEL ———————————— 5:43:49p

   Muted          52 SPC:       Reverse Drum 1                 Online
                                Short Attack                    XL-A

 HELP   Function keys for the flanger
 -F1       F1 and Ctrl-F1 increment speed in steps of 1 and 5 respectively.
           Shift-F1 and Alt-F1 decrement speed in steps of 1 and 5.
 -F2/F3 F2/F3 and Ctrl-F2/F3 increment the lower (F2) and higher (F3)
           turnaround frequency of the flanger in steps of 1 and 5.
           Shift-F2/F3 and Alt-F2/F3 decrement the frequencies.
 -F4       F4 decrements the flanger feedback. Shift-F4 increments it.
 -F5       F5 and Ctrl-F5 increment right channel sweeping phase.
           Shift-F5 and Alt-F5 decrement it.
 -F6       Lets you select several modes, see the manual for more explanations
 -F7       Triggers the flanger in MIDI mode. Shift-F7 stops the sweep.
           Ctrl/Alt-F7 increment/decrement the retrigger lock time.

                                              Function Stereo Delay
 HELP F-KEYS   Overview  Qrs  Gated  Delay  Flanger  Chorus

                                       F7Clear F8Mute F9 -  F10 +
```

```
———————————— QUANTEC QRS/XL CONTROL PANEL ———————————— 5 49

   Muted          24 QRS:  St.Peters Cathedral, Rome           Online
                  RT:12.5s Lo:1.6  Hi:0.1   Sz:5 Dn:100%        XL-A

 HELP  Keyboard
 Menu mode                         Input mode
 ESC        exit current menu      ESC        Exit edit or cuurent input field
 TAB        change menu item       Tab        Jumps from left to right column
 ?          Help                   Backspace Deletes one character to the left
 !          DOS command            Delete     Deletes to the right
 Enter      Select menu item       Insert     Toggles between Insert and Overwrite
                                   Home       Go to beginning of input field
                                   End        Go to end of text
 Use the arrow keys to move        Enter      Terminate input
 around in menus or the
 input mask

                                              Function Stereo Delay
 HELP          General  Effects  Mouse  Keyboard  F-Keys
F1Input F2Freeze                       F7Clear F8Mute F9 -  F10 +
```

Figure 2. *Representative examples of two Help screens.*

the user.)

Quantec is not hiding anything in this unit and will provide compilers for any computer hacker who wants to be creative. Not to get ahead of myself, but the owner's manual provides extensive coverage of program creation and language.

The XL design puts the power of program creation directly into the hands of the user. The ability to store new and altered programs on disk means that updates and exchanges may be accomplished via a modem link. The company has already set up user access to its VAX mainframe in Munich, West Germany. The VAX can be accessed through a local data network such as Tymnet or Telenet.

All new programs and updates on the Quantec Universal Access Network for Transmission and Exchange of Code (QUANTEC, get it?) are available free of charge. Program exchange clubs could be established on any E-mail service for interested parties.

Internal architecture

Power is the XL's middle name. The device features 16-bit linear input and output DACs, while internal architecture is 32-bit. Extremely high audio quality is achieved with 2x oversampling on the input, and 4x oversampling on the output. The digital filters exhibit almost no phase shift. Quoted bandwidth is 15kHz, as opposed to 8kHz on the original QRS.

Front-panel features

To the far left of the front panel is a rotary pot controlling the Wet/Dry mix ratio at the outputs. Along with the stereo input meters are separate RS-232C and MIDI communication LED indicators. Parameter information is displayed on an 80-character, reverse-transmissive LCD. The display is quite easy to read and may be adjusted for different viewing angles and lighting situations.

To the right of the LCD window are the stereo output meters and two software-programmable user LEDs. For instance, on flange programs the meters display flange peak and sweep rate, while for gating they indicate opening and closing modes. A separate rotary switch is used to select the desired program for display in the LCD window.

The rear panel houses the left and right channel XLR input and output connectors. A 25-pin RS-232C port enables connection to your PC; baud rate and protocol switches are provided to configure the XL for proper serial-data exchange. MIDI In and Thru connectors are present, along with channel set switches. In addition to an LCD screen control, there is an XL identification switch that allows up to 16 independent units to be controlled from a single PC.

Table 1. *Descriptions of the factory-supplied programs with the Quantec XL digital signal processor.*

Small Rooms

No.	Description
1	Living room, furnished RT: 0.4s
2	Living room, unfurnished RT: 0.63s
3	Wardrobe RT: 0.16s
4	Oil barrel RT: 0.8s
5	Yellow submarine RT: 1.0s
6	Chamber music studio RT: 1.0s
7	Disneyland spaceship RT: 3.0s
8	Bathroom RT: 0.7s
9	Cotton tent RT: 0.2s

Medium Rooms

No.	Description
10	Staircase RT: 2.5s
11	Cave RT: 1.0s
12	Backyard RT: 2.0s
13	Theater RT: 1.0s
14	Concert hall w/audience RT: 2.0s
15	Concert hall w/o audience RT: 2.5s
16	Empty stage RT: 2.0s
17	Muffled cinema RT: 0.2s
18	Railway station RT: 0.2s
19	Natatorium RT: 2.3s
20	Mueller's Volksbad in Munich, Germany RT: 5.0s

Large Rooms

No.	Description
21	Chapel RT: 2.8s
22	Church RT: 3.0s
23	Cathedral RT: 5.6s
24	St. Peter's Cathedral in Rome, Italy RT: 12.5s
25	Taj Mahal in Agra, India RT: 45.0s

Plates

No.	Description
26	Plate 1 RT: 0.5s
27	Plate 2 RT: 0.8s
28	Plate 3 RT: 1.0s
29	Plate 4 RT: 1.8s
30	Plate 5 RT: 3.2s

Reverberation Sets

No.	Description
31	Set 1—Size 3 w/ variable RT RT: 0.2s
32	Set 1—Size 3 w/ variable RT RT: 0.5s
33	Set 1—Size 3 w/ variable RT RT: 0.8s
34	Set 1—Size 3 w/ variable RT RT: 1.2s
35	Set 1—Size 3 w/ variable RT RT: 1.6s
36	Set 2—Size 4 w/ variable RT RT: 0.2s
37	Set 2—Size 4 w/ variable RT RT: 0.4s
38	Set 2—Size 4 w/ variable RT RT: 0.8s
39	Set 2—Size 4 w/ variable RT RT: 1.6s
40	Set 2—Size 4 w/ variable RT RT: 2.4s

Gated Reverbs
41-48

Enhance Patterns
49-51

Softies

No.	Description
52	Reverse Drum 1 Short attack
53	Reverse Drum 2 Long attack
54	Soft attack t=7.5ms
55	Soft attack t=15ms
56	Soft attack t=31ms
57	Soft attack t=62ms

Delay Effects

No.	Description
58	Rock 'n Roll Star Time: 120ms
59	Cosmic Delay Time: 1s

FIR Filters
60-69

Special Effects

No.	Description
70	Dropping Needles
71	Oscillating Crystal Glass
72	Splitting Images
73	Space Voice
74	Space Voice

Chorus Effects

No.	Description
75	2-voice stereo chorus
76	4-voice stereo chorus
77	6-voice stereo chorus
78	6-voice stereo chorus
79	24-voice stereo chorus

Panning Effects

No.	Description
80	Auto Pan
81	Leslie slow
82	Leslie fast

Flangers
83-90

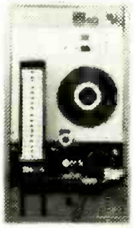
Programs have been condensed due to space limitations. Complete parameters are available from the manufacturer.

LEASE

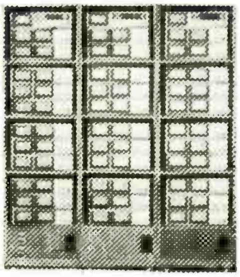
COMPLETE

AUDIOPHILE CASSETTE PRODUCTION SYSTEM

LOAD → DUPLICATE → LABEL → PACKAGE



LOADER



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Technical Specifications

Inputs: *2 (XLR-3 female), balanced with floating grounds +6dBm/−15dBm (selectable via digital attenuators).*
Impedance: *13.2kΩ balanced, 6.8kΩ, single-ended.*
Headroom: *+12dB over 0dB ref.*
HF rejection: *18dB/octave > 100kHz.*
Level display: *−30dB to +12dB (dB-linear).*

A/D conversion: *Two-times oversampling with digital filtering.*
Sampling frequency: *64kHz.*
Resolution: *16-bit (linear).*
Bandwidth: *15kHz.*
Distortion: *0.03%.*

Processor: *16-bit control microprocessor, 32-bit signal processor architecture.*
System clock: *8.192MHz.*
DSP program: *256 steps per sample.*
Working memory:
uP - 128kByte ROM.
64kByte RAM.
DSP - 256kByte RAM.

Programs: *90 preset locations, editable (ROM); 30 writeable locations (RAM).*
Room simulation: *Quantec QRS algorithms.*
FIR filtering: *dual - up to 117 taps. mono - up to 235 taps.*
Subsampling FIR: *various configurations.*
Delay, sampling: *MIDI triggerable; dual - max. 2 seconds. mono - max 4 seconds. Various special effects.*

D/A conversion: *Four-times oversampling with digital filtering.*
Sampling frequency: *128kHz.*
Resolution: *16-bit (linear).*
Bandwidth: *15kHz.*
Distortion: *0.01%.*

Outputs: *2 (XLR-3 male), balanced.*
Level: *+6dBm/−15dBm (selectable via digital attenuator).*
Impedance: *50Ω, balanced, 25Ω, single-ended.*
Headroom: *+12dB over 0dB ref.*
Level display: *−30dB to +12dB (dB-linear).*

MIDI

Connections: *In, Thru*
Channel: *(1-16): selectable.*
Control: *+ program number selection.*
+ recognition of keyboard note and dynamics.
+ PC editing of lookup tables.
+ system exclusive message capability

RS-232 Interface

Connection: *25-pin female, switched as modem (DCE).*
Baud Rates: *300/1200/4800/9600/115k.*

Front Panel

Display: *2 × 40 characters, backlighted LCD.*
Level indicators: *4, 8, LED's at 6dB/step.*
LED: *MIDI active indication.*
LED: *RS-232 active indication, 2 user-programmable LED's, Rotary control for program selection. 11-position detent pot for mixing of direct and processed signals.*

General Electrical Data

Dynamic Range:
(direct) 96dB, unweighted.
>100dB, A-weighted.
(processed) 93dB, unweighted.
>98dB, A-weighted.
THD: *0.01% (d); 0.03% (p).*
Frequency response:
(direct) 20Hz to 20kHz, (±0.1dB).
(processed) 40Hz to 14kHz, (±0.3dB),
20Hz to 15kHz, (±0.5dB).

Miscellaneous

Power supply:
230/115V ± 15%; 30VA.
Packaging: *19-inch standard rack mount, 1 unit high, 320mm deep.*
Weight: *4.5kg.*
Protection:
+ against mains overvoltage.
+ against loss of programs in case of main failure.
+ effective suppression of mains transient noise.

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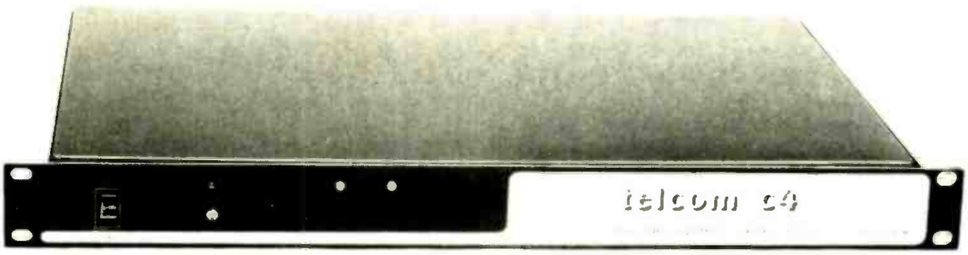

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Circle (18) on Rapid Facts Card

```
---------------- QUANTEC QRS/XL CONTROL PANEL ---------------- 3:54:20p

   Muted          91 QRS:          Plate 4: St.Croix Mod.       Offline
   Freeze          Reverb Time: 1.8 s    Predelay: 50 ms         XL-A
 Input Off

 Program 121     Display upper/left      Stereo Plate: A    t-60= 1.0 seconds
 Channel S               lower/right

      Input from [L/R/S] S                          Density [%] 100

    High end factor [x] 0.75              Reflection delay [ms] 2
      Reverb time [sec] 1                 Reflection level [dB] Off
     Low end factor [x] 1.6
                                              Reverb delay [ms] 18
         Room size [0..5] 3                   Reverb level [dB] 0

                                  Function Room Simulation
 MAIN MENU     ? Edit  Undo  Compile  Online  Program  XL  Files  Misc  Quit

F1Input F2Freeze                              F7Clear F8Mute F9 -  F10 ·
```

Figure 3. *Representative screen dump of a parameter control screen. Above is a delay algorithm (notice the cross coupling of the feedbacks). Note the smaller, upper window shows the parameters of algorithm currently running on the XL. The bottom window, on the other hand, is displaying the parameters currently being manipulated by the controlling PC.*

System operation

Operation without an external PC is extremely simple. Program selection can be performed in real time by scrolling through the unit's internal registers; whatever you display in the window is instantly called up and running. Using a PC as the controller, however, means that new programs may be called up directly. Parameters are displayed on the PC screen for manipulation in real time, or off-line while a different program is running on the XL.

MIDI mapping also allows rapid program loads when operating with an external keyboard or MIDI controller. Figure 1 shows a typical MIDI mapping control screen.

At first glance I assumed that operating the XL with a PC would be difficult. As I got further into the process, however, I was amazed how simple it proved to be. First, let me say that there are several different levels of Help screens that make referring back to the manual practically unnecessary. Figure 2 shows a representative example of an XL Help page.

The screens are laid out logically and the operation is simple. Even when performing parameter edits, I found that the computer would not let me enter values outside of its available ranges.

Figure 3 shows a screen dump representative of a program parameter. The small window at the top duplicates the XL's LCD display and shows the currently running program. To the left and right of that upper window are displayed functions such as input and output mutes; on-line or off-line editing mode; which of the possible 16 XLs you are addressing; and much more.

The large center window displays the control parameters for the program you want to manipulate, and the input configuration for either stereo or mono signal. The window also displays program number and LCD display labels.

The lower window provides menu selection and the program type being manipulated—for example, Room Simulation of FIR (finite impulse response) Filter—beneath which are the real time function key labels.

Listening tests

As always, we now get to the bottom line: How good does the Quantec XL sound? Listening tests were performed in Mix Room A at Lucasfilm's Sprocket Systems post-production facility in San Rafael, CA, a Jeff Cooper-designed room that has a very low noise factor of NC12.

Of course, the first thing I did was hook up the XL's outputs and crank the console gain. The XL is even quieter than the QRS which, until now, was the quietest reverb I have ever reviewed. The unit added no additional noise whatsoever to the system.

I went through many of the programs and found them all to be of very high quality: quite a few of the factory-supplied presets are useful right out of the box.

The *Room Simulation* programs were impressive, and I welcomed the effect of the added high-end response. There is also the added parameter of a High/Low crossover selection, which has become popular in several other types of reverbs.

Having done a lot of work with the original QRS, I was curious to see if the XL would be a whole new ball game. Even though the new unit has a quoted 15kHz bandwidth, Quantec, in an attempt to modify its QRS fans, has added various user-selectable filters that can be inserted in room programs to more closely simulate realistic reverb.

Algorithms have remained the same but, because the XL's reflection density is much higher than the QRS, for a closer match you should lower the density parameter to 50%.

The *Chorus* and *Flange* effects are the best I've heard; talk about clean and quiet . . . "Ichycoo Park" stand back! The Flange program can perform zero crossover and offers two separate channels of independently adjustable flanging.

I'm an old-fashioned kind of a guy and almost always would rather gate a reverb myself; the XL, however, has a couple of *Gate* programs that I would definitely use. The XL has taken a different approach to Gated Reverb—it is not actually a gate, more a crossfade between two reverbs. You can perform a true size change, and even move from a plate effect to a room.

Both Flanges and Gates have a discrimination window, adjustable in cycles, that allows you to select which type of audio will trigger the effect. Short times are used for transients, and longer times for smoother instruments such as voice. Both programs also offer adjustable windows to prevent retriggering.

Delays and *Panning* programs are both useful. The *Special Effects* are a lot of fun. Panning proved impressive in that I achieved not only a left-to-right illusion, but also top to bottom.

The *FIR Filters* are really good; in fact, the telephone filters are the actual curves developed by the phone companies.

Softies didn't do much for me at first glance, and so I spent almost no time with them.

An impressive custom program, *Air*, is shipped with domestic XLs. It provides more punch, high-end and much wider stereo field without the use of EQ, "excitement," distortion or compression. A processed track appeared noticeably louder with no increase indicated on the meters—a useful effect for broadcasters.

I came away from this review totally impressed with the Quantec QRS/XL. The delivered programs are very good. With the power and support provided to write your own programs, the creative possibilities are endless. R-E/P


N/D757

Production Sound Recording

By Fred Ginsburg

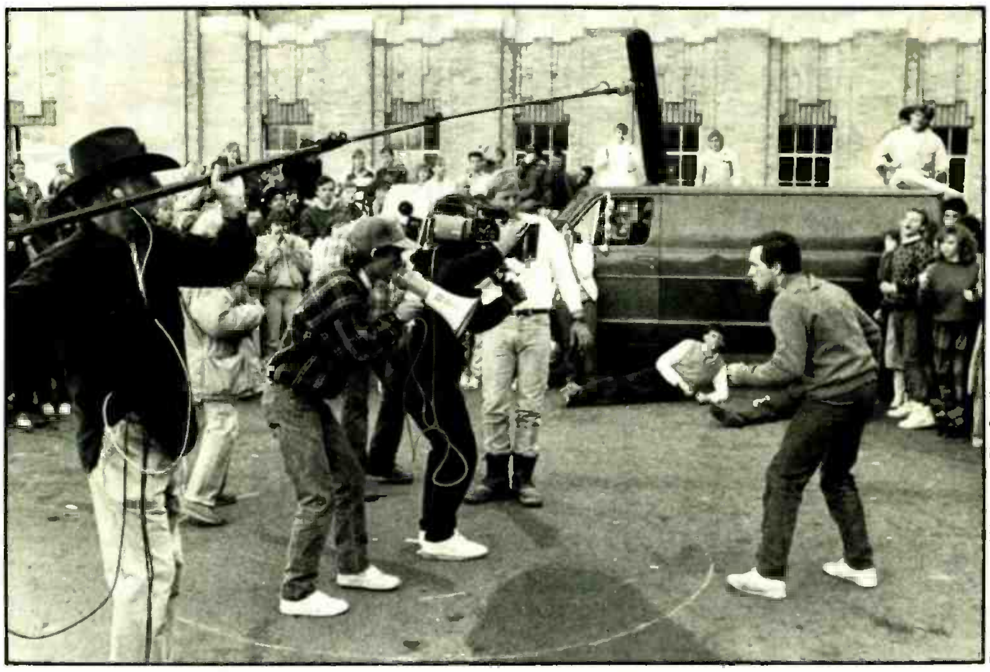


Photo courtesy of Universal City Studios.

Director Phil Joanou (with megaphone) and Don Bolger, boom operator, direct actor Casey Siemaszko during production of "Three O'Clock High," which is scheduled for release this fall by Universal City Studios.

A film set isn't a recording studio on location. Numerous, often uncontrollable, obstacles can impede your attempts to record usable audio.

Production sound mixing is the complex craft of recording live dialogue and sound effects on the set during principal photography of a motion picture or videotape. Many studio engineers wrongfully assume that expertise recording or post-production behind a console automatically guarantees them success out on a movie set. The technology, methodology and working elements out in the film world, however, are a far cry from the sanctuary of a sophisticated control room and acoustically correct studio.

On the set, a production mixer is only one of a number of important craftspeople involved in the making of a film or videotape. Sound must do their thing in harmony with the priorities of camera, lighting and dramatic direction, and must often take a back seat to these other aspects.

Very few production mixers possess more than the most rudimentary knowledge of audio electronics. Although they know how to operate their equipment, it is unnecessary for them to be able to perform any sort of extensive repairs. (Technicians back at the sound shop do that. Besides, on an active set there is neither the time nor the necessary bench equipment to fix anything.)

What sound mixers do have to master, in addition to their own craft, is an understanding of the crafts and techniques *adjacent* to theirs. A mixer cannot choreograph booms and fishpoles without understanding lighting, lenses and camera moves.

Creative decisions regarding the soundtrack itself cannot be made without an innate familiarity with the entire editing and re-recording process. For example, the mixer must be comfortable in knowing when and if the (bad) sound on a wide master shot can be replaced by the editor with the crisp track (though not an exact lip-sync match) borrowed from a later close-up.

Set etiquette is very important. For keeping accurate logs, shot numbers must be received from the scriptperson (and confirmed with the camera assistant). Communication between talent, director and camera must be assertive, yet not overbearing. And rigging microphones under a shy actress' costume requires the utmost tact.

Fred Ginsburg, CAS, is a Los Angeles-based free-lance production sound mixer and technical writer. He is a member of the IATSE local 695, and is a member of the Cinema Audio Society's board of directors.

Filmmaking is a collaborative art; production sound mixing is a part of that process.

Staffing for audio

A major cause for many qualified studio engineers falling flat on their faces when asked to go out on a shoot is that they think (or the producer thinks) that *one* man can do it all, or that an assistant's skill with a soldering iron back at the studio translates into finesse with a fishpole or boom.

Professionalism begins with insisting upon a proper team of personnel. A typical production sound crew consists of between two and five people, two or three being the norm. The mixer is considered the department head and assumes responsibility for the production soundtrack, crew, equipment and set politics.

The second most important person is the boom operator. The mixer and boom person will often collaborate on determining mic placement. Difficult scenes demand an almost telepathic relationship between the boom and mixer, because they play off of each other's capabilities (mic placement vs. mixing panel) in terms of actors and blocking. A good boom operator should also be able to temporarily fill in if the mixer is absent from the set.

Features and television will have a third person, known as the utility sound tech, whose duties include rigging plant mics and radio mics, wrangling cable on complex moves or working as second boom.

Scenes being shot to sync playback may also require the addition of a skilled playback operator.

Myth of One-Man VTR/audio

Low-budget videotape shoots are notorious for expecting 1-man crews to bring back "Hollywood" finished soundtracks.

Typical duties of a 1-person crew include operating the VTR, mixing the incoming mic levels and holding the boom—all at once! Sorry guys, it just doesn't work if you're aiming for anything better than an ENG soundtrack.

There are two types of sound that a 1-man crew can be expected to successfully achieve: ambience and talking head.

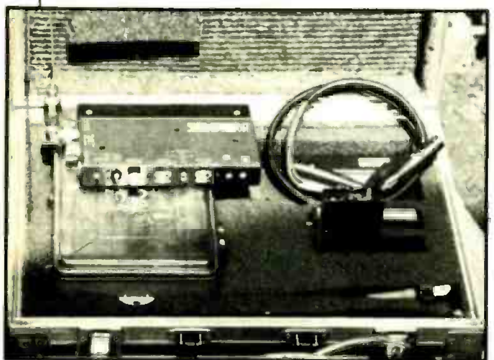
Because the VTR is usually moved on a cart with a monitor for the director, and is physically cabled to the camera, it becomes difficult to position oneself close to the source of the sound. Even the best condenser shotguns cannot discern dialogue from such camera distances. The best that a mixer can be expected to deliver is general background ambience.

As an alternative, lavalier mics can be deployed to capture specific dialogue. However, hardwired lavs are rarely suitable for more than interviews, spokesman standups or limited drama. Although radio mics can provide greater talent mobility, they often have the drawbacks of RF interference and dropout. In general, lavaliers may create a problem with audio perspective—in other words, the intimate, close-up sound does not match the camera angle. There are also the inherent problems of clothing noise and body movement.

On the other hand, a 2-person sound crew would be able to record all but the most complex dramatic scenes. Because the boom operator is a separate entity, he or she is free to move in close and follow talent for continuous optimum mic placement. The mixer is now able to divert full attention to riding gain, playing phasing and mixing inputs. Even when talent is placed on a lavalier or radio mic, the boom operator can round out the track by micing the subtle detail, such as footsteps, hand props and other natural effects.

Portable equipment

It goes without saying that the right tools are a necessity for doing a good job. Start with a lightweight, mobile location-sound



[Clockwise from top]

Seen on close-up detail here is a Perfectone 3-input mono mixer (used for backup duties or as a second unit), plus various Rycote (gray barrels) and Light Wave (slim solid black) windshields with integral shockmounts.

The author's accessory box contains a variety of lavalier microphones, power supplies, related accessories and mounting sundries, plus a selection of audio and video connectors and adapters.

The author's wireless microphone case houses a pair of Coherent Communications Artech series E-250s, plus two Sennheiser MKE-2 and two Sony ECM-30 lavalier models.

cart. Unlike the studio engineer, a production mixer does not try to be isolated from the action, but instead will park the cart on the edge of the set in full view and earshot of all of the action. That way, audio occurrences can be anticipated along with the interaction of scene blocking with mic placement.

The recording medium is usually to ¼-inch mono, ¼-inch 2-track or videotape. Especially with video decks, it is always worth monitoring via the record machine rather than the mixing board, in case of RF interference or ground loops.

The mixing console is of paramount importance. On a set, it is not so often a question of mixing a large number of inputs, as it is being able to exert smooth gain control over one or two. A good console should offer four to eight inputs (rarely will you ever need more than that), limited equalization, lots of input gain and a communications module. Overall size and weight are critical factors, as is the capability of operating the unit from batteries. Generally, ac power causes more problems for sound than it's worth, both in terms of cabling and grounding problems.

EQ is hardly used, except for bass rolloff and a slight mid-range boost to help punch the dialogue; anything more is best achieved during post-production. Signal processing does not belong on the set; it should be done later during final re-recording. Films are shot out of sequence, with shots being torn apart and edited together to create finished scenes. Attempts to prematurely process the track can cause a myriad of matching problems.

Lack of processing applies especially to so-called stereo field recording. Left/right spatial relationships will be created with pan pots during post-production, mainly because the production mixer has no way of anticipating exactly how the final scene will be edited. Two available tracks on a stereo recorder are best used as separate mono channels, such as for dialogue and ambience, or radio mic on actor #1 and radio mic on actor #2.

Much of production mixing consists of riding mic gain, leveling out the extremes of each character as well as balancing between multiple characters. Smooth ramping or slope is essential to accomplish this feat inaudibly. Most of the compact sized ENG-style mixing boards fall short for this function. Not only are their knobs too small for human fingers, but even slight gain adjustments call attention to themselves on the track.

In terms of the communications module, a console should have a slate mic (possibly with sub-tone slating if one is working in film); talkback to the boom man during a take; audio returns for the boom men; and a 1kHz tone oscillator for setting reference levels.

Battery operation of the mixer when used with a Nagra portable is imperative. Not only is clean ac often hard to come by on a set, but Nagra decks have opposite ground and may blow up if the case makes contact with something passing a large differential.

If you are working in video, be particularly cautious of all electrical connections (VTR, monitor and camera) that may cause ground loops and induce noise into the track. (Note: Electrical interference can be induced via BNC *video* connectors, not just power or audio.)

Microphone selections

Microphones should be of true condenser type (T-powered or phantom); good shockmounts and blimp-type windscreens are a must. Patterns should include a unidirectional shotgun, one or two hypercardioids, a wide hypercardioid and a wide cardioid.

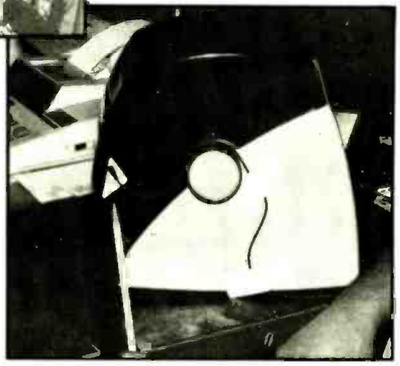
Avoid being saddled with a less expensive, electret-type directional condenser mic. Although low priced, systems lack the sensitivity and reach offered by real condensers.

Lavaliers come in two basic varieties: transparent and proximity-oriented. Until recently, all professional-grade electret lavaliers were proximity-oriented—in other words, they tended to add presence to close dialogue while rejecting background ambience.

Some of the newer mini lavaliers offer a more transparent sound, allowing them to blend in more naturally with overhead boom mics or to function as plant mics. The drawback is that the transparent lavs also

Fred Ginsburg, author, pictured at his mobile sound cart, which houses a 6-input Audio Developments stereo mixer and a Nagra IV-S-TC stereo portable recorder with center-track time code. A Nagra 4.2 mono recorder located on a lower shelf is available for synchronous playback during a film or video shoot.

Sometimes the locations of a microphone can cause problems for a director or camera man, particularly if there can be booms near the talent. Where can the mic be located to pick up both actors in this scene? Standing in for the talent are Ginsburg, [left] and Emmy-winning sound mixer for "St. Elsewhere," Blake Wilcox.



The author's solution was to tape a small Sennheiser MKE-20 lavalier microphone to the side of the Rolodex file, and loop the cable to the portable recorder.

pass more ambience. Depending on the situation, both varieties have their applications.

Radio mics are an option you may want to have available on a shoot. They are expensive, and even the best of them can be unreliable because of RF interference, magnetic voodoo, etc. There will be instances, however, when the sound is either miced with radios or not at all. UHF frequencies tend to be less prone to common interference, but have less range than a comparable VHF unit. If you plan to use radio mics, budget for more units than you expect to use so you'll have spares.

Fishpole selection is very important. They should extend to 12 or 18 feet and be lightweight, but not to the point of bowing under the weight of a mic, shockmount and windscreen at full extension. (Struggling all day with a bending fishpole will wreak havoc on a boomman's back.)

Studio booms are reasonably inexpensive to rent and should be seriously considered for many non-documentary productions. Their ability to telescope over distances and cue make them highly versatile on a set. Add some extra budget for an experienced boom operator to run the studio booms.

Priorities on the set

The job of a production mixer is to record raw material for the editors and re-recording mixers to transform into a finished soundtrack. Therefore, the first priority is to secure usable dialogue any way possible. In the event that laying down usable dialogue is impossible, then you should still try to get as near perfect a track anyway, even if it is only to be used as a guide track for ADR or looping.

Second priority would be to record the dialogue in matching perspective for the camera angle; the final priority is to record sound effects to accompany the shot. This latter might include such things as footsteps, hand props and doors. Sometimes these effects are recorded during the actual take, a rehearsal or after the shot. Some productions may also ask for presence or room tone.

Hint: If you are asked to record presence, arrange to do it just before the camera rolls on the first take. That way, everyone is in position and the sound will be an *exact* match of the actual take. Waiting until the end of the last take means that the production mixer will have to fight the commotion of exiting talent, crew and wrapped equipment.

The first priority is to secure usable dialogue any way possible.

Recording sound effects can be trying. From a political standpoint, editors like sound effects, but they like them on isolated tracks. Producers, on the other hand, are impressed by sync sound effects on the main track during the screening of dailies. Production managers, on the other hand, hate *anything* that slows up the pace of activity on the set, even if it would save money later.

It is essential to determine how the show is going to be handled from an editorial standpoint. For instance, most non-theatrical videotape productions do not have a budget for extensive audio sweetening, other than to lay in some music, narration and a few key sound effects. In that situation, a good mixer might try to pack the track with as much texture and live sound effects as necessary, without endangering either the clarity of the dialogue or the ability to intercut shots.

On the other hand, a feature film editor would prefer a clean soundtrack that can be embellished on the cutting bench.

Microphone technique hierarchy

There are four basic ways to approach micing a subject: boom, plant, lavalier or radio mic.

In most instances, the best dialogue will be achieved by employing a fishpole or boom overhead of the subject. A good condenser, angled 1 or 2 feet and slightly ahead of the subject, will produce a crisp, natural sound. The audio track will not be affected by clothing noise or bodily movement (such as arms folded across the chest). Talent can move around, walk and sit with the mic following overhead. Multiple performers can interact with each other, both verbally and physically, without rustle or phasing problems.

In a pinch, the fishpole can be held at knee level with the mic pointing up.

There is no difference to the mic between your using a fishpole that is not moving, and a C-clamp. Fixed microphones, also known as plant mics, can be strategically deployed around the set to cover isolated characters who would be impractical to reach with the boom.

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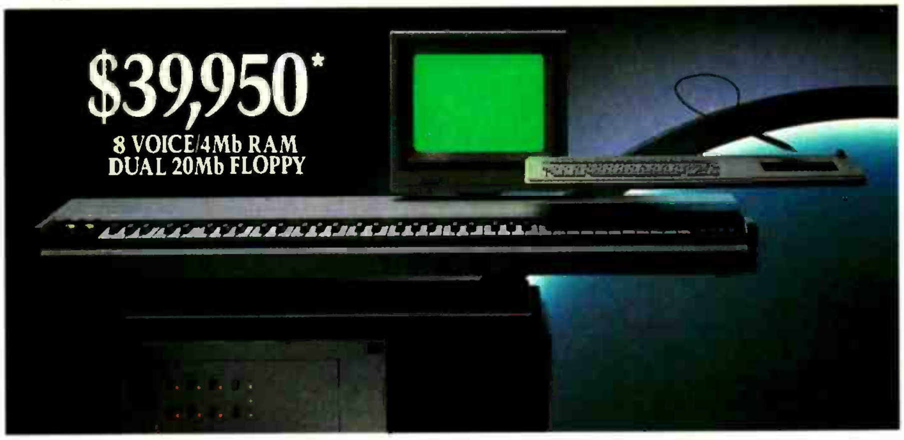

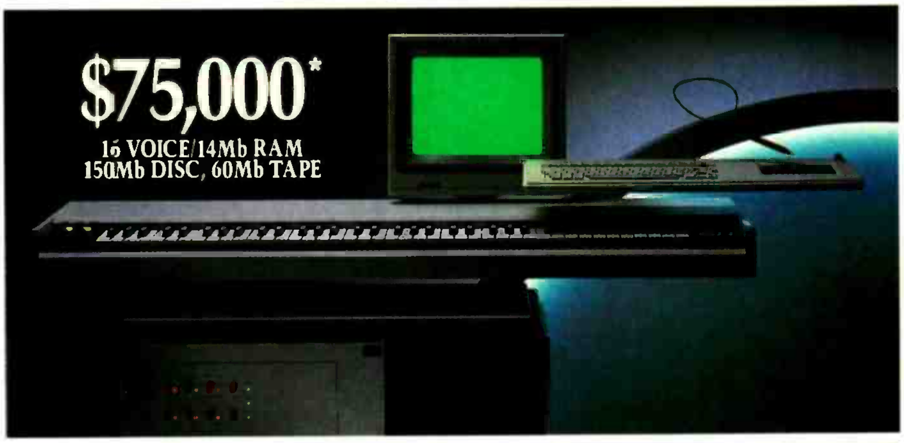

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*Prices effective as at July 1, 1987 subject to change without notice. Systems above 16 voice require external rack-mount units (not shown), delivery late 1987. $39,950 system does not include music keyboard.

Circle (20) on Rapid Facts Card

Plant mics can consist of either conventional condensers or suitable lavaliers. The new minilavaliers, with their improved sensitivity and transparency, make excellent plants. They are so small that they can be hidden right in the middle of the scene and not show up on camera.

Plants can be hidden in doorways, on executive pen sets, on the edge of dressing mirrors, bed headboards, automobile sun visors . . . even in floral arrangements.

Employ shockmounts to avoid noise and vibrations from being directly conducted to the mic. With lavaliers, a small loop of tape works nicely.

Pay strict attention, however, to multiple mic phasing. A plant and a boom overlapping can easily result in mush, unless you keep your hands busy on the pots.

Lavaliers are the next option. Worn on the body, they tend to go (or stay) where the actor is. As mentioned earlier, proximity-oriented mics tend to add presence to the dialogue as well as to reject background noise. Transparent lavaliers blend better with overhead condensers and sound more natural or less "forced." On the other hand, the latter do not screen out background ambience as much.

If an actor is going to be walking, and a cable dragging from his or her ankle is impractical, then radio mics are the final resort.

Selection of overhead mics

Which condenser pattern is best? Like everything else in our industry, it depends on the situation.

Narrower patterns, such as shotguns, provide greater reach but capture more reverberation in a closed interior. For that reason, shotguns are preferred for exterior use or sometimes for use on a very dead soundstage. As well as capturing a build-up of echo in an interior, a shotgun used overhead tends to be physically unwieldy. Their narrow pattern also makes cuing from actor to actor very critical.

Shotguns, like telephoto lenses, will compress the background vs. the foreground. Just as in a photograph, a distant sunset will appear large and close to a foreground sailboat, shotguns will magnify background sounds and ambience in relation to the subject. The best way to use a highly directional microphone is to ensure there is nothing behind the subject—in other words, the mic looks down at the subject, "seeing" only quiet dirt past it, or is aimed upward at the subject, seeing only silent sky. Aiming a shotgun horizontally should be avoided, except for micing certain sound effects.

Wider pattern condensers (cardioids) and ribbon mics provide the mellowest sounds, in terms of picking up reverberation or room echo, but also have the shortest effective range. But, because cardioids are often used in cramped interiors with low ceilings, excessive headroom is not usually a problem.

In between the wide cardioids and narrow shotguns are the wide hypercardioids and the narrower hypercardioids; their selection would be a trade-off between mellowness vs. effective reach.

In addition to echo and reach, another factor that comes into play when selecting a boom mic is that of spread. Scenes involving tricky blocking and/or multiple actors might be better served by a mic that does not require critical targeting, even though it would be a compromise against reach and punching the dialogue.

One very useful trick in balancing a strong voice against a weak voice is to take advantage of the mic's natural pattern. Favor the weak voice on axis, and let the strong voice strike slightly off axis.

A production mixer tries to be on the edge of the set in full view and earshot of all the action.

A word about wind noise. Foam slip-on windscreens should always be used on interior shoots, because condensers are sensitive to even the most minute air movements. Outdoors, use a barrier mesh-style windscreen; wind tends to gust unexpectedly and simple foam is ineffective against anything more than a wisp of a breeze.

Handheld microphones aren't used very frequently in theatrical production, but can have their uses. Because their elements are virtually indestructible, dynamic mics are ideally suited for recording loud explosions. A dynamic mic used relatively close-up (6 to 9 inches) works excellently for isolating speech from a noisy background, such as for on-site voiceovers or talking head (with the mic seen on-camera).

Staged scenes involving the use of a handheld mic (as a prop) should be recorded exactly that way. Use a boom mic to actually record the voice, unless you want to be at the mercy of handling noise and inconsistent mic placement.

For "man-on-the-street" reporter interviews, provide the talent with an omnidirectional condenser or electret mic. It will provide some consistency, regardless of how well the reporter pays attention to cuing the microphone between himself and the interviewee.

Using lavaliers

Correctly rigging body mics on talent requires time and tact because, at some point during the process, the soundperson will have to work inside of talent's clothes.

The microphone capsule itself can be secured either outside of clothing or hidden under wardrobe; the cable and connector will almost always be routed under wardrobe.

If the mic is going outside of clothing, then mounting clips can be used. The proper technique for using a tie-clasp style clip involves looping the cable from the head of the mic, like a "J," through the bend or hinge of the clip. The cable continues up and around—behind the garment—to complete the circle. As the cable makes its way down, it is clenched in the jaws of the clasp, thus providing a strain relief.

The remainder of the cable is run behind clothing so that the XLR connector can be secured at a convenient point, such as at the waist (to a belt or pocket pack) or ankle. Regular mic lines can then be connected or disconnected easily to free the talent between takes.

If needed, an external lavalier can be made quite inconspicuous by camouflaging it to match wardrobe. Colored marking pens can be used on small strips of tape and/or foam windscreens to subdue the appearance of the mic head and clasp. Alternatively, small patches of felt or cloth can be used to cover the mic.

Remember that the camouflaged mic will be so tiny in the frame during a medium shot as to be nearly invisible. On close-ups, the camera can frame out the microphone completely.

A useful trick is to save the foam-tipped tech swabs used for head cleaning. These foam booties make excellent, expendable windscreens for mini lavaliers.

There are two types of clothing noise you can encounter: contact and acoustic. Contact clothing noise is caused by a garment flapping into or rubbing across the mic capsule. The solution is to carefully immobilize all clothing that may create this problem, by taping down everything on either side of the mic. One popular technique is to sandwich the mic between two sticky triangles of tape (formed by folding a strip of tape like a flag, sticky-side out).

Because contact noise can also be caused when clothing rubs against the mic cable, care should also be taken in this area—even with external lavaliers. Form a loop near the mic for strain relief, and then apply a few lengths of tape along the cable. Use double-faced tape or sticky triangles to immobilize clothing to keep it from rubbing.

Acoustic noise can be created from clothing rubbing against itself. Static Guard works well, while a light spray of water can help soften starched fabric. Synthetic fibers tend to be much noisier than naturals and should be avoided whenever possible.

The most important point to remember when rigging radio mics is to never allow the mic line and the antenna to cross. Also, the antenna should be kept somewhat rigid, and never looped over itself. If the antenna has to run in a direction other than straight up or to the side, then invert the transmitter pack and let the mic cable loop rather than the antenna. A good way of keeping the antenna rigid is to affix a rubber band to the tip and then safety pin the rubber band to the clothing. This technique maintains a little tension in the antenna but still provides a safe strain relief if the actor should bend over.

Check the talent regularly. Tape tends to loosen due to moisture and movement. Costumes tend to be adjusted constantly, either by talent themselves or by the costume department.

Never assume that the wardrobe personnel know how to rig either lavaliers or radio mics. Consult with them about costume selection or modification but do not leave the actual wiring up or readjustments of mics or radio transmitters to them. Costumers worry about how the actors look, not how they sound.

Sync playback

The key word here is sync. To have performers dance and mouth lyrics beat for beat with a prerecorded track requires 100% sync at all production and post-production levels.

A master soundtrack is created and then recorded with a sync reference—normally pilot tone or SMPTE time code. This track will generate the final version soundtrack used by the editor in the finished product. Generated from this final version soundtrack will be two or more playback dupes for use on the set.

On the set, the playback dupe is resolved and played back from a portable sync recorder. The track is either amplified through a speaker system or silently broadcast to the performers via induction-loop cuing. The camera films at crystal sync, sound speed. A second audio deck or the VTR records a marker slate and re-records the portion of the playback track (and/or time code, if applicable) being played during the particular take.

The soundtrack from this second recorder is used for the subsequent viewing of dailies. (Without the slate or time code, there would be no easy way of knowing exactly what part of the playback track applied to what footage.) Later, the editor will match this guide track, which is in sync and cued up with the picture, against the complete and uncut, but sync version master soundtrack that becomes the finished version. This, in turn, results in the camera takes now being in proper sync with the complete soundtrack, and the guide track can be eliminated.

None of this will work, however, if any of the following conditions exist:

- The playback dupes are not a frame-for-frame, sync pulse-to-sync pulse *exact* copy of the master soundtrack that will constitute the final edit.
- The playback dupes are not resolved and played back to sync.
- The camera does not run at crystal sync speed.
- Talent is out of sync due to poor acting.

Production sound is not an endeavor to be taken lightly. Knowledge of electronics and a mastery of studio techniques are not necessarily qualifications in themselves for successfully mixing dialogue on a shooting set. Life in the recording or production studio is based on achieving perfection under controlled conditions. Mixing on a set is a matter of generating usable raw material, in what are usually uncontrollable conditions.

But, with the right approach, equipment and trained personnel, a production mixer can achieve excellence. RE/P

Photography by Philip Silver and Fred Ginsburg.

Maintaining Compatibility When Using Multiple Wireless Microphone Systems

By Ken Fasen

Proper frequency selection is the key to successfully using multiple wireless microphones.

The advantages provided by wireless microphones make them popular, and occasionally mandatory, for many broadcast and recording applications. The freedom of movement and benefit of concealment allow production options that would be impossible with wired microphones. Perhaps the most notable advantage is that talent is no longer restricted to a stage, creating many opportunities for flexibility and creative production.

Wireless microphones are radios and, therefore, are subject to the laws of radio-frequency physics. These laws of physics and the regulations of the FCC put practical limitations on the selection of operating frequencies. In addition, when two or more wireless microphone systems are operated simultaneously, a complex set of restrictions applies to the equipment design and selection of operating frequencies. The following is a description of these restrictions, what causes them and how they can be overcome.

Definition

Multiple-system frequency compatibility is defined as the condition of two or more wireless microphone systems operating simultaneously, with no degradation in the performance of one due to the presence of the other systems. A *system* is defined as a transmitter and a companion receiver.

Suppose five wireless microphone systems are operating simultaneously. If each system functions equally well with the other four systems turned on, as well as with them turned off, the five systems are said to be compatible. If, however, the presence of any of the other four systems degrades the performance of the fifth, they are said to be incompatible. Only one of the other systems may be responsible for the interference. On the other hand, a combination of the other systems could be responsible.

Causes of frequency incompatibility

The restrictions on the selection of operating frequencies are imposed both by the FCC and by the limitations of the transmitter and receiver circuits. The FCC reserves frequencies and bands of frequencies for wireless microphone operation. Eight specific frequencies are reserved under FCC Part 90.265(b) for wireless microphone use. Because of the channel spacing chosen, however, only combinations of two of these frequencies are truly compatible in spite of the eight available frequencies. Selecting frequencies in the TV-channel spectrum allows larger compatible systems to be built.

RF physics and the limitations of the equipment circuitry impose restrictions that fall into the following six categories:

- separation between operating frequencies,
- transmitter spurious signals,
- 2-signal intermodulation,
- 3-signal intermodulation,
- receiver local oscillator radiation and
- receiver image-frequency sensitivity.

Each category is distinct and requires its own explanation. All six restrictions concerning system frequencies must be satisfied simultaneously to achieve complete compatibility.

Separation between operating frequencies

The separation between operating frequencies is simply how close together in frequency the systems are spaced. The limiting factor in the receiver design is the selectivity of the intermediate frequency (IF) filter and the dynamic range of the RF pre-amplifier and mixer circuits. The more selective the filter and the higher the dynamic range, the closer the operating frequencies can be. A basic guideline is that all operating frequencies should be separated from one another by at least 400kHz.

A selectivity problem can be identified by turning on all receivers, then turning on only one transmitter at a time. If any receiver other than the companion receiver unsquelches, the operating frequencies may be too close together. Calculating differences in operating frequencies can confirm this possibility.

The best solution to this problem is to change system frequencies. To do this, calculate the differences between system frequencies. If any of them are less than 400kHz apart, they must be changed. Suppose the following system frequencies are being used:

F1 = 174.8MHz
F2 = 175.4MHz
F3 = 175.7MHz

You find F2 and F3 less than 400kHz apart. Therefore, either F2 or F3 must be changed. An acceptable frequency for F3 would be 178MHz. Now all system frequencies are separated by more than 400kHz.

Fasen is engineering manager for HM Electronics, San Diego, CA.

One alternative is to turn on the transmitter companion to the affected receiver in an attempt to *capture* the receiver and reject the interfering signal. Capture is a phenomenon in FM receivers whereby the stronger of two co-channel signals suppresses the weaker one. This alternative may or may not be successful, depending on conditions.

Transmitter spurious signals

In addition to the desired signal, transmitters emit energy on other frequencies as well. Most wireless microphone transmitters use a quartz crystal as a frequency-determining element and multiply it up to the operating frequency. A multiplier circuit is one in which the output frequency is a multiple of the input frequency.

Consider a transmitter operating on 160MHz. Starting with a 20MHz crystal and multiplying by 2 yields 40MHz. Multiplying by 2 again yields 80MHz, and once again yields the desired 160MHz. This is a "times eight" (2 x 2 x 2 = 8) transmitter. However, it also radiates energy at frequencies other than 160MHz. Signals also are present at 80MHz (x4), 140MHz (x7), 180MHz (x9), 200MHz (x10) and so on. Granted, they are significantly weaker than the desired 160MHz signal, but they exist. A receiver operating on one of these undesired output frequencies may receive the transmitter's spurious signal, possibly causing audio degradation.

Assume that a spurious signal is transmitted at a level 70dB less than the desired signal. Its transmitted power might then be −53dBm. This is 57dB above the threshold sensitivity of the receiver and will unsquelch it.

The solution is to select frequencies so these spurious signals do not fall on or near the other operating frequencies. Such undesired signals should be at least 250kHz from any operating frequency. Multiples of 1 through 16 times the crystal frequency should be computed and compared with all the other operating frequencies.

To check for spurious signals, turn on all the receivers, then turn on only one transmitter at a time. If any transmitter unsquelches a receiver other than its companion receiver, you may have a transmitter spurious problem. Calculating the transmitter's undesired crystal harmonics will tell you if this is the case.

Again, the best solution is to change system frequencies. First, calculate the offending crystal harmonic, then change the offending transmitter or the offended receiver. Suppose you are using the following system frequencies:

F1 = 190.8MHz
F2 = 214.8MHz

The ninth harmonic of the crystal for system No. 1 occurs on 214.65MHz, which is within 150kHz of F2. The calculations follow.

1. 190.8MHz ÷ 8 = 23.85MHz (crystal frequency).
2. 23.85MHz x 9 = 214.65MHz (ninth crystal harmonic).
3. 214.8MHz − 214.65MHz = 150kHz (less than 250kHz apart).

An acceptable frequency for F2 would be 214MHz because it is 650kHz away from the ninth crystal harmonic of system No. 1 (214.65MHz − 214MHz = 650kHz). Alternatively, you can keep the F2 transmitter turned on in an attempt to mask the undesired signal from transmitter F1. Results of this approach will vary.

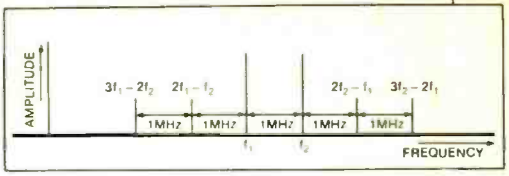


Figure 1. *Two-signal IM does not produce any intefering products when only two systems are used.*

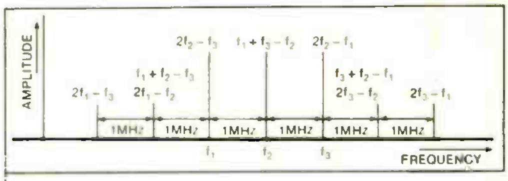


Figure 2. *Three-signal IM can produce interference at a large number of frequencies.*

Two-signal intermodulation

Two signals applied to any non-linear circuit create additional signals, or intermodulation (IM). These signals include the sum-and-difference products of each of the fundamental input signals and their associated harmonics. The following components are produced:

- fundamental: F1, F2
- second order: 2F1, 2F2, F1 ± F2, F2 − F1
- third order: 3F1, 3F2, 2F1 ± F2, 2F2 ± F1
- fourth order: 4F1, 4F2, 2F1 ± 2F2, 2F2 ± 2F1
- fifth order: 5F1, 5F2, 3F1 ± 2F2, 3F2 ± 2F1
- higher-order products

Order is defined as the sum of the numerical coefficients that multiply the F1 or F2 terms. Note that the even-order products usually occur far removed in frequency from F1 and F2 and, therefore, are omitted here for simplicity.

If F1 and F2 are close to each other in frequency, the 2F1−F2 and 2F2−F1 terms also fall close together. If F1 and F2 are separated by 1MHz, those products also will be separated from F1 and F2 by 1MHz. For example, if F1 = 160MHz and F2 = 161MHz, the following intermodulation signals will occur:

- third order: 159MHz (2F1−F2) and 162MHz (2F2−F1)
- fifth order: 158MHz (3F1−2F2) and 163MHz (3F2−2F1)
- higher-order products

An RF spectrum analyzer display depicting these relationships is shown in Figure 1.

If any of these products fall on or near any system frequency, interference and incompatibility will result. The guideline is that these IM products should be at least 250kHz away. Note that 2-signal IM will occur when two systems are operated simultaneously. However, if the two transmitters are separated by the required minimum of 400kHz, the interference is not a problem when only two systems are operating. This is because the close-in IM products (2F2−F1 and 2F1−F2) will be at least 400kHz away from either system frequency. Two-signal IM, however, also can cause interference when three or more systems are used.

IM can occur in the output stages of two closely held transmitters. If this happens, it actually will be retransmitted by both transmitters. The IM also might occur in the receiver RF circuitry due to close proximity of the transmitters to the receiver antenna. In any case, a signal produced on or near a system frequency may unsquelch an undesired receiver.

The problem can be identified in two ways. First, the interference occurs only when two transmitters are turned on. Turning either one of the transmitters off removes the interference. Second, the interference is more severe when the two transmitters are in close proximity to one another or are close to the receiver antenna. The interference may disappear completely when the transmitters and receiver antenna are separated from each other.

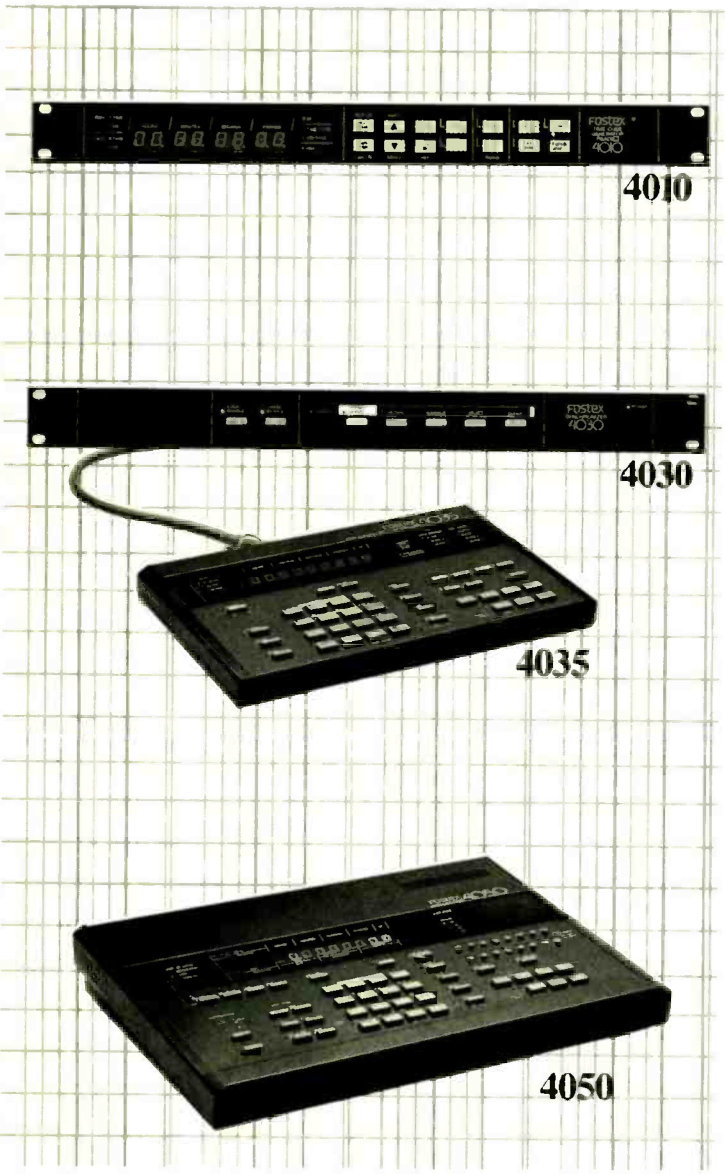
Again, the best solution is to change system frequencies. Calculate the IM products to see if any fall within 250kHz of any system frequency. For example, suppose the following system frequencies are being used:

F1 = 174.8MHz
F2 = 175.4MHz
F3 = 176.6MHz

When tested, with F1 and F2 transmitters both turned on and in close proximity to one another, the receiver on F3 not only unsquelches, but also receives the audio from F1 and F2. Use the following formulas to identify the cause:

1. 2F1 − F2
2. 2F2 − F1
3. 3F1 − 2F2
4. 3F2 − 2F1

Fostex Presents a Series of Timely Investments:



4010 SMPTE Time Code Generator/Reader has all the functions you'll ever need because it's completely software based. As in all high tech digital electronics today, there's a switching power supply. And socketed EPROMs mean your hardware will never be obsolete. Plus, the 4010 has a serial data port on the rear panel which allows communication with (and control by) computers.

4030 Synchronizer with switching power supply and **4035** Controller for all major brands and models of tape recorders—both audio and video. Control up to one master and three slaves, or slave as many transports as you wish, just by adding a 4030 for each slave. A serial interface port allows communication with personal computers to run F.A.M.E., Fostex Automated Media Editing. SMPTE time code based, it works with 24, 25, 30 fps and drop frame, and features resolution to 1/100 of a frame, selectable pre-roll up to 99 seconds, 10 position auto-locate and **programmable, automatic punch in/out with rehearse function.**

4050 Auto Locator is a full function remote control unit for Fostex Models 80, 20 and all E-Series recorders. In addition to tempo control, you can locate to the measure bar and beat, thus the 4050 is the first autolocator to think musically. Up to ten cue points can be programmed and you can punch in and out automatically. The 4050 is also a MIDI Synchronizer, featuring a SMPTE/EBU Time Code generator/reader—all four formats. Any MIDI instrument can be synchronized to this most accurate timing reference. You can also simulate the running of the recorder in order to have the SMPTE/EBU code run MIDI instruments only. There's even a serial communication port which lets you use a personal computer.

Merging Technologies

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The calculations show that 3F2 − 2F1 = 176.6MHz, which is F3. You must, therefore, select a new F3. Recalculate the IM products using the new F3 to be sure no IM products fall within 250kHz of the new F3 frequency. In addition, you must take into account all of the combinations of any two of the three system frequencies. In this case, an acceptable frequency for F3 would be 178MHz.

One option is to keep the transmitters separated from each other and from the receiving antenna by at least 10 feet. Also, turning on the transmitter for the offended receiver may help to mask the problem.

Three-signal intermodulation

Just as two signals combined in a non-linear circuit can cause sum-and-difference products to be created, the same happens with more than two signals. Although not as severe, 3-signal IM also can be a problem. The following signals are produced by 3-signal IM.

- fundamental: F1, F2, F3
- third order: 2F1 ± F2 and F1 ± F2 ± F3

Again, the even-order and higher-order products usually are far removed in frequency and are seldom of interest. For the sake of simplicity, the higher-order products will not be discussed here.

Consider a system with microphones operating on the following frequencies:

F1 = 159MHz
F2 = 160MHz
F3 = 161MHz

Third-order IM products will occur as follows:

Frequency	IM Product Formula
157MHz	2F1 − F3
158MHz	F1 + F2 − F3 and 2F1 − F2
F1 = 159MHz	2F2 − F3
F2 = 160MHz	F1 + F3 − F2
F3 = 161MHz	2F2 − F1
162MHz	F3 + F2 − F1 and 2F3 − F2
163MHz	2F3 − F1

Note that in the example given, third-order IM products fall exactly on the system frequencies themselves. For example, F1 + F3 − F2 = 160MHz, which is F2. Hence, equal spacing of system frequencies results in 2-signal IM interference (2F2 − F1) as well as 3-signal IM interference. These relationships are shown in the RF spectrum analyzer display in Figure 2. The guideline, again, is to keep these IM products at least 250kHz away from any system frequency.

The characteristics of 3-signal IM are identical to those of 2-signal IM. Three-signal IM can occur in the output circuits of transmitters or in the input circuits of receivers. The problem can be identified in two ways. First, turning off any one of the three offending transmitters will eliminate the interference. Second, the interference is more severe when the three transmitters are in close proximity to one another or to the receiver antenna.

As with 2-transmitter IM, the best solution to 3-transmitter IM is to change frequencies. When you add a fourth transmitter, the calculations take on more importance, because you must consider all combinations of three of the four frequencies. Consider the following example:

F1 = 181.4MHz
F2 = 183.4MHz
F3 = 184.8MHz
F4 = 186.8MHz

When transmitters on F1, F2 and F3 are turned on and in close proximity to one another, the receiver on F4 not only unsquelches, but also receives the audio from transmitters F1, F2 and F3. The following formulas are, therefore, applicable:

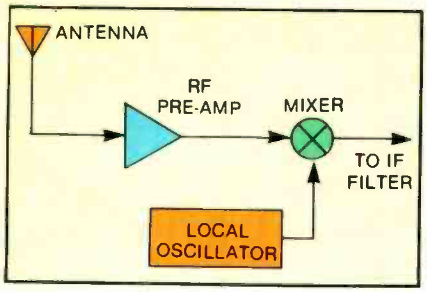

Figure 3. Basic superheterodyne receiver block diagram.

1. 2F1 − F3
2. F1 + F2 − F3
3. 2F1 − F2
4. F3 + F2 − F1
5. 2F3 − F2
6. 2F3 − F1

Note that only third-order product formulas are listed here. Fifth-order formulas (3F1 − F2 − F3 and 3F3 − 2F1) also may be significant, but are omitted here for simplicity.

Calculations show that F3 + F2 − F1 = 186.8MHz, which is F4. You must, therefore, select a different frequency for F4. In addition, you must consider all combinations of three of the four system frequencies, and you must be sure the six formulas don't indicate an IM product within 250kHz of the new fourth-system frequency. In this case, an acceptable frequency for F4 would be 204.8MHz. Alternatively, keeping the offending equipment physically separated and turning on the offended receiver's transmitter may help minimize the problem.

Receiver local oscillator radiation

A basic superheterodyne receiver block diagram is shown in Figure 3. The local oscillator (LO) generates a carrier, which is mixed with the signal received at the antenna. This process generates a new signal at the intermediate frequency (IF). The local oscillator is actually a low-powered transmitter-type circuit, with an output wired to the mixer circuit. If the LO signal is coupled to and radiated by the receiver antenna, interference may be produced.

Consider a receiver operating on 160MHz. If the receiver has an IF of 10.7MHz and low side injection, the LO will operate on 149.3MHz. This frequency can be radiated by the receiver antenna and detected by any receiver operating on or near 149.3MHz.

If a receiver LO signal falls on or near any system frequency, interference (and incompatibility) may result. The guideline is that no receiver LO should be closer than 250kHz to any system frequency. LO radiation can be identified by turning off the offending receiver and noting whether the interference disappears. Of course, this effect is most obvious with all the transmitters turned off.

Assume you have a system operating on the following frequencies:

F1 = 203.3MHz
F2 = 211.4MHz
F3 = 214.0MHz

When the receiver on F3 is turned on, the receiver on F1 unsquelches. You determine that the LO frequency of receiver F3 = 203.3MHz (214MHz − 10.7MHz), which is F1. The solution is to select a new F1 or F3, being sure it is at least 250kHz away from the LO of the other receivers. An acceptable frequency for F1 would be 210.8MHz.

Again, physically separating the offending receiver and its antenna from the offending one and turning on the offended receiver's transmitter should minimize the problem.

Receiver image-frequency sensitivity

Receiver image frequencies also can

produce interference problems. An image frequency is equal to the LO frequency minus the IF frequency. For example, if a receiver operates on 160MHz with an IF of 10.7MHz and an LO of 149.3MHz, then the image frequency is 138.6MHz (149.3MHz – 10.7MHz).

The typical receiver is 70dB less sensitive at its image frequency than at its operating frequency. This provides an image-frequency threshold sensitivity of approximately –40dBm. A transmitter with an output level of +17dBm will generate a signal 57dB above the receiver's image-frequency sensitivity threshold. Interference can result easily. The same basic guidelines apply. Separate all image frequencies from operating frequencies by at least 250kHz.

You can identify the problem by noting whether the offending transmitter is 21.4MHz lower than the offended receiver's operating frequency. Although these calculations are based on low side injection and an IF of 10.7MHz, the principle is valid regardless of the LO frequency.

Degrees of compatibility

As with other forms of interference, the amount of compatibility varies. In some cases, the problem may be mild and, in others, severe. The stronger the interference signal and the closer it is to a system frequency, the more serious the problem. For example, if a system operates on 160MHz, and an IM product is generated on 160.05MHz (50kHz away), the problem is likely to be severe because the frequencies are close together. If, however, the IM product occurs at 160.24MHz (240kHz away), the problem may be mild or even non-existent.

Mild compatibility problems often can be masked by keeping all the transmitters turned on. The transmitter's signal strength usually will be sufficient to capture its companion receiver. If a transmitter is turned off, any signal stronger than 1μV may be heard in its companion receiver. If, however, the transmitter is turned on, an interfering signal may have to be as strong as 1,000μV (60dB stronger) to be heard in the receiver.

Misdiagnosis

Other problems may appear to be caused by incompatible frequencies. Outside RF interference or dropouts are sometimes responsible for poor microphone performance. It is important to be aware of this possibility so you don't waste time looking for a cause in the wrong place.

First be sure that the entire wireless microphone system is operating on compatible frequencies. In the case of a 4-microphone system, this involves more than 14,500 calculations.

To complete such an analysis requires a sophisticated computer program. Fortunately, programs can be written to automatically search for a desired number of compatible frequencies when given lower- and upper-frequency bounds. Such programs may be available through software companies, public databases or computer bulletin boards. When purchasing wireless microphone systems, ask the manufacturer to perform an analysis for you. The company may be able to suggest alternative frequencies if compatibility problems seem likely.

Well-designed, state-of-the-art equipment can maximize wireless mic performance. However, even with the best equipment, there is always the potential for problems. The first step to minimizing frequency incompatibility is to judiciously choose system frequencies.

RE/P

Manufacturers' Reaction to MIDI Time Code

By Paul D. Lehrman

While some manufacturers are using MIDI Time Code in their products, others are waiting to see how the standard is accepted.

Whether MIDI Time Code will fly is a touchy subject for some people, especially those who have a lot to gain from it. Although at this point, the verdict is still out nearly nine months after its adoption as a standard, its chances for success seem to be improving, as more people understand what it is, and how it can best be used.

As my article in the October 1987 issue stated, one of the problems is a reluctance among certain segments of the industry, particularly those companies that are already involved with studio automation, to adopt a new standard. When I was writing this article, I interviewed a number of industry figures to get their opinions on the subject. Unfortunately, because of space limitations, this section had to be omitted, which may have caused the impression that I was saying critical things about MIDI Time Code without any justification.

I want to set the record straight, and make it clear that as a writer I have no personal bias one way or the other on the issue (although as a musician, I want very much for it, or at least something like it, to succeed). What follows are sections of the article that did not appear last month, tweaked slightly to bring them up to date.

In some ways, MIDI Time Code can be considered a solution in search of a problem. It isn't replacing anything that already exists. Instead, it is postulating a new way of achieving studio automation. But with so much to be done before MIDI Time Code can be considered "accepted," inevitably the question is raised, "Is it all worth it?" The answer depends on whom you talk to.

Digidesign, which makes a MIDI Time Code-based automation software program called "Q-Sheet," is very enthusiastic about the standard, as is Opcode Systems, which recently added MIDI Time Code capability to its Cue program, which was originally designed as an off-line film composer's aid.

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

Others, however, are not so sure. This includes many companies who make hardware that interfaces directly with SMPTE, which is only to be expected, but at the same time it is precisely these companies that will have to implement the new standard if it is to survive.

Some critics of MIDI Time Code have objected to the space that the data would occupy in an already crowded MIDI data stream—estimates of the extra overhead the data would need range from 8%-32%—but that argument has essentially been settled by general agreement that MIDI Time Code would in most cases occupy its own cable, separate from other MIDI information.

More serious arguments are raised by those who think it is simply unnecessary, and an extra burden on manufacturers. For instance, Mark Cohen, vice president of Fostex, thinks, "It may be more confusing and expensive to customers than it's worth. Once you have the SMPTE converted to MIDI (clocks and pointers), if everything else is downstream from that, just do it through MIDI. Why convert everyone's MIDI gear?

"There already is a SMPTE standard for video editing communications, which runs at a much higher baud rate than MIDI. There's no reason for video equipment manufacturers to even think about MIDI, and if you tell Sony they're going to have to put a MIDI jack on the back of a BVH-2000, they'll laugh at you.

"At first, we were highly interested, but then we heard about the limitations on baud rates, and the timing errors. People will have to build PLL timing loops into software to compensate for timing errors, and that doesn't always happen the way it should," Cohen says.

At this writing, Fostex has no plans to implement MIDI TC. "Ninety-eight percent of what MIDI Time Code will do is already in our equipment," says Cohen. But they're not counting it out completely. "We could add it as a retrofit, if it settles down, and if we see it as advantageous to our customers," he says. But at the same time, he feels that "SMPTE and video editing stuff can be brought down in price, even lower than MIDI."

Potential timing errors are also of concern to Gerry Lester, of Adams-Smith: "We're not planning to lock a tape machine to MIDI TC—there's too much jitter in a full MIDI line, and if there's another message in the way, the sync message could jump as much as 3ms. You would need very forgiving software to handle that."

Their reservations notwithstanding, Adams-Smith, which was one of the more active contributors to the MIDI TC specification, plans to release a new version of its Zeta Three synchronizer, which will incorporate MIDI Time Code, about the time you read this. The unit will support an editable internal tempo map (which can be saved on tape and reloaded, using the user bits in the SMPTE time code), and several "learning" modes in which tempos are entered externally. When a standard MIDI protocol for exchanging tempo maps (like the MIDI File standard) is approved, "We'll be behind it." The company will not, at first, be supporting Cuing messages, mostly because the level of MIDI control of transport functions has yet to be decided by the tape recorder manufacturers. "We'll serve as a gateway to those machines if and when it happens," Lester says.

Bill Southworth, president of Southworth Music Systems, is skeptical of MIDI Time Code, although he is incorporating it into his JamBox/4, which does direct SMPTE-to-MIDI conversion in conjunction with the company's Macintosh-based sequencing software.

"Our opinion is that the need that MIDI TC tries to satisfy could be done at least as well, and possibly better, within the framework of SMPTE and MIDI," he says. "The real issues for music—bandwidth and tempo maps—are not addressed." He will however, "support additional capabilities as they emerge."

John Carey, marketing manager of Otari, is concerned about MIDI Time Code's limitations as a true automation standard. He believes a MIDI TC-based audio editing system, integrating sequencers and tape, is still "a ways off, because the information resolution of a quarter frame just isn't good enough, unless devices have some kind of offset that can be calculated from the quarter-frame boundaries."

Otari is looking into implementing MIDI Time Code "in the middle future," according to Carey, "which means not in the next year. But we will be integrating reel-to-reel machines more into electronic music environments, and so we're looking at it."

Another company that makes synchronization systems is Alpha Audio. "MIDI Time Code has no impact on us at the moment," says vice president Bob Tulloh. "We've had MIDI on our system for over a year. We send note-on commands to samplers at specific points in time. We treat it like just another device, like a cart machine." Tulloh is also concerned with potential bandwidth problems. "We run at 38.4kbaud (as opposed to MIDI's 31.25kbaud), and we're already up against bandwidth limitations."

Another company spokesman, Carlos Chafin, explains, "At present our synchronizer, The Boss, does not generate MIDI sync, and to do so will require circuit changes due to timing overhead restraints in the communication co-processor scheme. The sync demand option has been a low priority with us as most of our users are engineer/editors, not synth users, and most synth players already have some kind of device for doing SMPTE-to-MIDI conversion. But when we put in MIDI sync, MIDI Time Code will closely follow, and I do feel it is going to happen, probably when someone offers a product that gets our engineers fired up enough to drop what they're doing and deal with more MIDI stuff."

Tascam is adopting a "wait-and-see" policy. The company plans to have a device out around the end of the year that is specifically designed for MIDI/SMPTE synchronization, but they have not yet decided whether it will incorporate MIDI Time Code. The "MIDIizer" will convert SMPTE to MIDI and back again, and synchronize various popular brands of tape decks. According to marketing manager Bill Mohrhoff, "It may or may not have MIDI Time Code. It may put it out and not react to it or, if we start to see other things that use it, we may implement it more."

"We make peripherals," adds David Oren, director of product planning. "Until the software developers decide how they will implement everything, we're not really going to do anything."

Hardware that uses MIDI Time Code is coming to market slowly. As of this writing, except for the new Adams-Smith synchronizer and various peripheral devices made by smaller companies like Southworth, Opcode, and J.L. Cooper, the only device on the market that recognizes MIDI Time Code is Sequential Circuits' Studio 440, a hardware-based sequencer and sampler (designed in large part by Chris Meyer before he left the company), but it uses MIDI TC on an elementary level. As one spokesman put it, "The MTC doesn't do much. You can use it to clock a second 440, but there isn't much call for that.

The Roger Linn-designed ADR15 sampler/sequencer from Akai, due out around the time you read this, will read MIDI Time Code and play from an internal tempo map. On the other hand, Yamaha's new MSS-1 SMPTE-to-MIDI convertor does not implement MIDI Time Code.

This chicken-and-egg scenario is typical of new technologies, especially when people used to working in separate worlds, like composers and film sound engineers, are forced to deal more closely with each other. However things turn out, MIDI Time Code is a noble attempt to bridge a major gap in our industry, and even if it doesn't succeed, it will have had a major impact on the way we think and work. RE/P

Re-evaluating Ribbon Microphones

By Robert H. Lowig

Long respected for their excellent transient response, ribbon designs have often been dismissed because of their supposed fragility to handling and breath noise. Recent developments in ribbon technology may mean that we need to re-evaluate their applications, in particular stereo MS techniques and digital sessions.

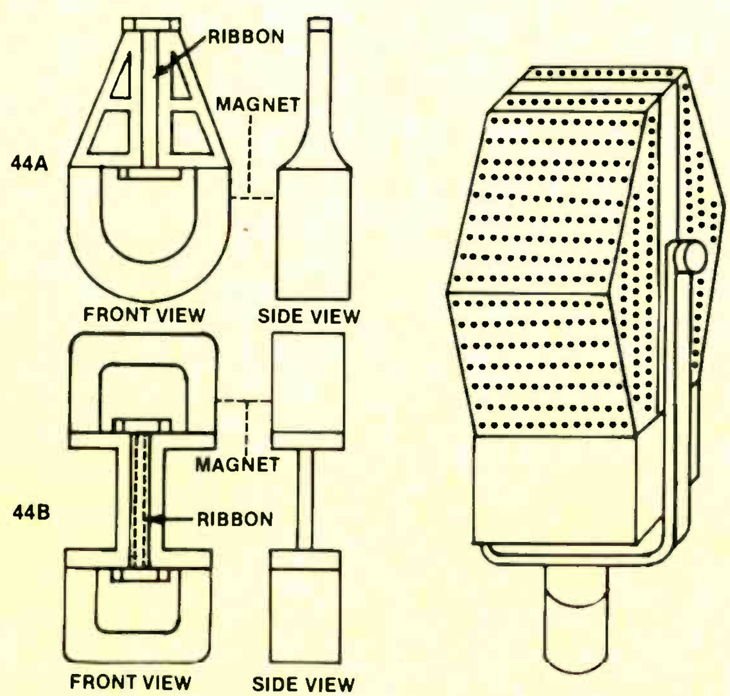

Figure 1. Internal construction of vintage RCA model 44 ribbon microphones. Note the slight differences in magnet shapes and ribbon suspensions between the 44A [top left] and 44B models.

When ribbon microphones were first introduced in the early Thirties, they signaled a radical change in microphone transducer design. This revolutionary approach was greatly appreciated by audio engineers of the time, who delighted in the ribbon mic's ability to produce a warm and natural sound.

Ribbons quickly became accepted as the standard microphone for "broadcast-quality" audio production and, as early as the Fifties, they were used extensively in all major radio and TV studios. Although the initial idea for a ribbon mic was undoubtedly conceived simultaneously in the United States and Europe, the industry's first acceptance of its unique capabilities was a direct result of the research and engineering efforts made by RCA. Developed by engineer Harry F. Olson, the RCA 44-B ribbon mic was first introduced in 1932 (Figure 1).

Bulky and extremely fragile, these early ribbons were nonetheless incorporated into applications where sound quality was a primary concern for professional audio engineers of that era.

Understanding the mechanics of the more conventional moving-coil design is useful in clarifying the operating principle of a ribbon mic. A moving-coil microphone produces an alternating current by moving a coil wire within a magnetic field created by a permanent magnet. If

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the coil is attached to a diaphragm, the structure will track the changes in air pressure that the sound waves produce, generating a corresponding alternating current.

The ribbon design achieves similar results, but incorporates a basic design change that produces sonic characteristics a moving coil could never achieve. In a ribbon design, a thin strip of duraluminum or pure aluminum is suspended between two permanent magnets. The aluminum strip conducts electricity and also acts as the diaphragm (Figure 2).

The ribbon itself will track the changes in air pressure occurring between the front and rear of the microphone, causing the ribbon to move within the magnetic field and generating a current flow across it.

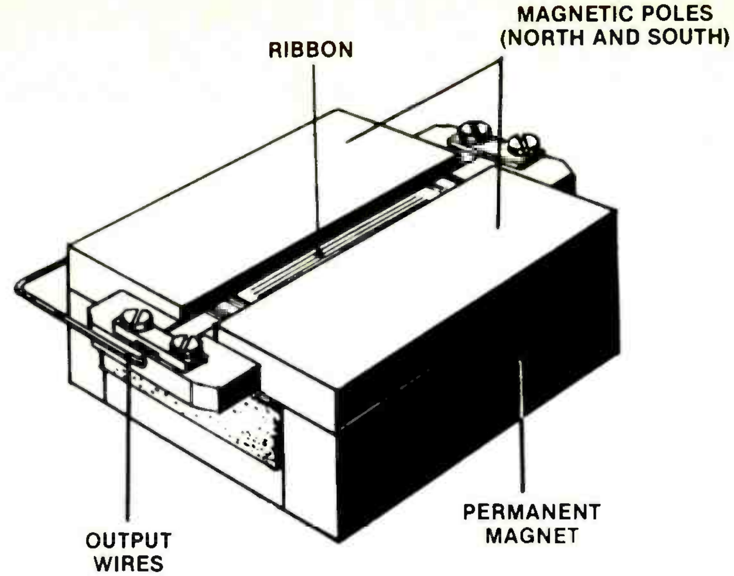

Figure 2. Internal construction of a modern ribbon microphone.

Transient response

Because the moving structure is an extremely small and lightweight strip of aluminum, the specific weight of the element is unparalleled by any other transducer type. This allows the ribbon to overcome inertia and track transients better than moving-coil designs, thus recreating all of the original sound source. Transient response is so fast that the ribbon's warm, natural low-end and clear, transparent midrange and high-end responses become immediately apparent. The heavier weight of moving-coil elements prevents them from reproducing transients as effectively.

The light weight of the ribbon element also allows accurate damping to reduce sibilant sounds. A moving-coil structure continues to move after the sound waves stop, thereby creating sounds that aren't part of the original signal.

However, the original ribbon designs required large magnets to maintain high flux density and large mass ribbons to provide sufficiently high output levels. The increased size and dimensions of these ribbon and magnet structures necessitated large and bulky enclosures, making early ribbon mics such as the RCA 44 undesirable for hand-held use.

In order to move properly, a ribbon had to be corrugated horizontally. This design made it extremely delicate, and the ribbon could be deformed with the slightest burst of wind. Also, singers generating vocal pops at above-average sound pressure levels could damage ribbon elements. These factors led to the ribbon's longtime reputation as a "fragile" microphone suitable only for controlled studio environments, while its use for outdoor film and broadcast location work was out of the question.

However, the distinctively warm and transparent sound produced by ribbon mics was well worth the shortcomings, and such models became a popular choice for a variety of recording and broadcast studio applications.

Contemporary designs

The evolution of contemporary technology has led to a radical change in transducer design. The modern ribbon has undergone a complete transformation, eliminating many of its early drawbacks. Leading European manufacturers experimenting with rare-earth magnets discovered that they could produce higher flux-density magnetic fields equal to the early ribbon designs, but at a fraction of their original size. This discovery allowed the reduction of ribbon size, while still maintaining sufficient output levels.

The actual design of the ribbon also changed. Smaller ribbon strips were corrugated lengthwise to create a more durable element that also would move in a more linear direction. Coupled to a 3-stage blast filter and mounted in a ball-type basket on a conical shaft, the first hand-held ribbon sufficiently rugged enough for live vocal applications came into being.

Characteristically, it produced a better transient response and less distortion than its moving-coil counterpart, yet it could handle up to 120dB SPL.

Double-ribbon mics

Further experimentation produced a double ribbon mic that quickly earned the respect of recording studio engineers all over the world. The double-ribbon design incorporates two aluminum strips situated one on top of the other, a fraction of an inch apart. This configuration allows even higher output levels than single-ribbon designs, and also decreases non-linear distortion and sibilants because of a highly effective damping factor. The design can also handle higher SPL than its predecessors without a blast filter.

The double-ribbon design is now the first choice for studio applications where accurate reproduction is critically important. Because it is so precise, the double-ribbon mic affords more uniform control of the polar pattern at all frequencies. This uniformity allows a ribbon mic with a cardioid pattern to offer greater control over feedback and, in its natural figure-eight form, to achieve nearly 100% cancellation at its 90 and 270 null points.

It is also easier to obtain a specific frequency response with this design. For hand-held applications, this ribbon delivers a smooth boost in the midrange for added vocal presence and efficient rolloff of footfall noise. These characteristics also provide an extremely flat response for recording situations.

Over the years, ribbons have become more popular as their applications potential has grown. A milestone of contemporary mic technology, the hand-held ribbon mic has allowed users to achieve a type of sound never experienced with moving coil models. For recording, the ribbon's fast transient response and improved damping capability makes it ideal for accurately reproducing the sound of percussive instruments that produce transients as part of their initial attack tones. These attributes make ribbons particularly effective for micing instruments like the piano, harp, acoustic guitar, violins and as overhead mics for drums.

MS techniques

More recently, ribbon mics have been "rediscovered" for new applications, almost as if they were specifically designed for these tasks during their creation more than 50 years ago. One of these new areas is MS (mid-side) recording, a technique that combines the cardioid and figure-eight outputs of a mic to create a stereo image.

The growing popularity of stereo TV sound production has created a great demand for stereo program material. Field audio engineers for film and video are now being pressured to produce soundtracks in stereo, and MS mic techniques offer one of the easiest ways to satisfy this demand.

Matched pairs of ribbon mics, one with a figure-eight and one with a cardioid polar pattern, are currently available. Field engineers are recording material with each mic being fed to a separate tape channel, and then phase-combining both tracks to stereo through an electronic matrix system during the post-production process. These matched ribbon pairs are said to produce an extremely "honest" spatial and perceptual stereo image, along with a highly accurate frequency response and reproduction of transients—the end result being truly outstanding sound quality.

Digital applications

Another important discovery is that ribbon mics have the potential to play a crucial role in the still evolving digital recording process. Research by leading recording engineers has established that using ribbon mics for digital recording sessions can provide a more "humanized," musical-sounding finished product.

It is believed that when the moving coil element is subjected to high SPL transients, the resulting signal will cause the center core of its coupling transformer to ring at its resonant frequency. The digital process records this ringing, the final manifestation of which is a harsh and often grainy sound.

Conversely, a ribbon mic passes this transient voltage without consequence through its coupling transformer, submitting only the original accurate soundwave. Because of its exceptional transient response, a ribbon mic is also capable of capturing the specific timbral nuances and dynamic shifts that distinguish a particular performance recorded in digital, yet *without* the self-generated noise and strident sound generally attributed to condenser designs.

The final results of using ribbon mics in digital recording have been characterized by engineers as exceptionally "warm, transparent and musical" without the all too familiar upper-end harshness for which digital is known.

How ironic it is that ribbon microphone technology, developed over 50 years ago and abandoned because of its supposed "fragility" by all but a very few current manufacturers, is providing a wealth of new possibilities in today's most sophisticated studio, live concert and field recording applications. RE/P

Studio Update

Northeast

If Walls Could Talk Studios (Caldwell, NJ) has moved into a new 1,500 square foot, 4-room studio designed by **Herb Pabst** and **Glenn Taylor**. The facility features a 20′x23′ control room, iso booth, MIDI/keyboard room and a reception area.

The studio features a **Harrison Raven** 32x24 mixing console, **Otari MX-80** 24-track recorder, **Sony PCM 501ES** digital mixdown processor, **Studer A-80** ½-inch mixdown with Dolby A units, three **Time-Line "Lynx"** time code modules and a **Yamaha SPX-90** and **Rev-7**. *Box 309, Caldwell, NJ 07006; 201-226-1461.*

Howard Schwartz Recording (New York) has renovated Studio East and now features a **Solid State Logic 6000E** with 24 inputs and Total Recall. *420 Lexington Ave., Suite 1934, New York, NY 10017; 212-687-4180.*

White Crow Audio (Burlington, VT) has purchased a **Studer A820** 24-track recorder. *19 Marble Ave., Burlington, VT 05401; 802-658-6475.*

Newbury Sound (Boston) has installed **Digital Creations "Diskmix"** time code based automation system. The facility also features a **Harrison MR-4** console outfitted with **Sound Workshop's ARMS** automation system, **Lexicon 224XL**, three **Yamaha SPX-90s** and three **Lexicon PCM70s**. *1260 Boylston St., Boston, MA 02215; 617-267-4095.*

Kennedy Music and Recording (Philadelphia) has added an **Apple Macintosh Plus**. Software includes **Mark of the Unicorn Performer** and **DX Librarian**. *5253 Montour St., Philadelphia, PA 19124; 215-533-2380.*

Target Productions, (Boston) has added a 24-track audio suite that features a **Synclavier.** Also featured are voice-over record and editing, audio sweetening and full music and sound effect libraries.

Jeff Largent has joined the facility as sound designer. He was formerly the senior audio engineer for Video One. **Amy Coblenz** and **Sally Fikaris** have been added as account executives and **Amy Kafka** joins the facility as communications director. *529 Main St., Boston, MA 02129; 617-242-1900.*

Cove City Sound Studios (Glen Cove, NY) has upgraded its facility to 48 tracks. New equipment includes a **Neve 8014** and a **Neve 8068** console. *7 Pratt Blvd., Glen Cove, NY 11542; 516-759-9110.*

Southeast

Sound Stage (Nashville, TN) has installed a second **SL4056 E series** console with Total Recall. *10 Music Circle South, Nashville, TN 37203; 615-256-2676.*

Twin Oaks Studios (Rocky Point, NC) has added a **D & R 4000** series console with 40 inputs and shortloaded with 26 input modules. In addition, the studio has added **Stephen Bradley** to the staff as in-house producer and keyboard specialist. *P.O. Box 187, Rocky Point, NC 28457; 919-675-9226.*

Full Sail Recording (Altamonte Springs, FL) has changed its name to **Platinum Recorders** and has updated and renovated its mobile recording facility under the direction of manager **Gary Platt**.

New equipment includes two **Otari MX80** 24-track tape machines with CB120 remote controller, **Sphere Eclipse 48 x 28** console, **Hafler DH 500** amps. Outboard gear includes two **DBX 900** series, including 903 compressors, 904 gates, 905 equalizers and 902 de-essers. UREI and Aphex compressors are also included. *658 Douglas Ave., Altamonte Springs, FL 32714; 305-682-7788.*

Mastertouch Recording Studio (Nashville, TN) has updated its **New England Digital Synclavier II** preproduction/scoring room with the addition of a MIDI interface and 112-point patchbay. The patchbay can be linked to the **Sony** 24-track digital main studio.

Other equipment includes a **Yamaha QX-1** 8-track sequencer with eight TF-1 modules, **Yamaha SPX-90** digital effect processor, a **Fostex 3070** compressor/limiter and an **MXR 129** pitch/transposer. *2714 Westwood Drive, Nashville, TN 37204; 615-297-2246.*

North Central

Metro Studios (Minneapolis) has taken delivery of an **Otari MTR-20** ½-inch track 2-track recorder, **Focusrite ISA 115-HO** dual mic pre-amp and a **Valley People 815** dynamic processor. *200 Third Ave. North, Minneapolis, MN 55401; 612-338-3833.*

StudioStudio (Dallas), a new 24-track studio, features a **Harrison 3232** series console with 48 inputs and Auto Set automation, **Sony JH 24** multitrack and **JH 11** 2-track, **Lexicon 200** reverb and **UREI 813B** monitors.

Other eqiupment includes **Crown DC-300A, D-150 and D-60 amps, UREI 813B** and **811B** monitors and a **Lexicon 200** digital reverb.

The studio features a drum booth, iso booth and a 1910 **Mason and Hamlin** 9-foot grand piano. *4801 Spring Valley, Suite 106, Dallas, TX 75234; 214-960-0381.*

Goodnight Dallas (Dallas) has taken delivery of a **Macintosh Plus** computer and MIDI interfaced with an **Opcode Studio Plus**. Software includes **Mark of the Unicorn Performer** and **Opcode synth librarian** for storing and recalling patches.

New equipment includes **E-mu Emax** sampler and **Yamaha DX-7II** synthesizer and **RX-5** drum machine. *11260 Goodnight Lane, Dallas, TX 75229; 214-241-5182.*

Southern California

Tarpan Studios (San Rafael, CA) has taken delivery of the **Fairlight CMI Series II**. *1925 Francisco Blvd., Suite G, San Rafael, CA 94901; 415-485-1999.*

Northern California

Hyde Street Studios (San Francisco) has renovated the facility and now features four separate studios.

Studio A, now **Powerline Productions**, is the newest addition, offering 24-track recording and **Jessie Bradman** as executive producer. **Studio C** features **Alpha & Omega Recording/Sandy Pearlman,** a 48-track studio. **Studio D** houses **Hyde Street Studios** and features a **Meyer** monitoring system. **Studio E** features **Earwax Productions**, a MIDI production/music composition facility. *245 Hyde St., San Francisco, CA 94102; 415-885-4999.*

Please send Studio Update announcements to both the **RE/P** editorial and production offices. Michael Fay, **RE/P** Editor, 1850 N. Whitley, Suite 220, Hollywood, CA 90028 and Dan Torchia, **RE/P** Staff Editor, P.O. Box 12901, Overland Park, KS 66212.

New Products

Ampex 467 digital audio cassette

The company is introducing an 80-minute play length to its line of U-matic digital audio cassettes for compact disc mastering. The length allows for the full 72-minute CD capacity, plus blank tape at the head and tail to reduce high CRC capacity and to allow for control data.

Circle (157) on Rapid Facts Card

BSS DPR-502 noise gate

The DPR-502 is a dual-channel noise gate that features a bidirectional MIDI analog interface, allowing the unit to transmit and receive MIDI information. Auto Dynamic Enhancement provides two levels of information emphasis, restoring information that can be lost as a gate triggers.

Circle (151) on Rapid Facts Card

Popper Stoppers windscreen

The pop filters/windscreens are available in 3-, 4- and 6-inch sizes, and use a standard ⅝-inch fitting. They are also available with a 15-inch gooseneck and a screw-on clamp.

Circle (152) on Rapid Facts Card

Beyer percussion mic group

The five microphones—the M 422, M 420, M 201, M 380 and MC 713—are for snare, rack toms, floor toms, kick drum and cymbals. They accurately reproduce the sound of modern percussion while absorbing the physical punishment resulting from today's drumming styles, the company says.

Circle (153) on Rapid Facts Card

Beyer CV 720 PV power supply

This is an addition to the MCM system of condenser mic capsules and power supplies. The unit accepts phantom power sources from 12V to 48V, and includes a 10dB attenuator and a 12dB/octave LF filter to eliminate boom noise.

Circle (154) on Rapid Facts Card

Korg DSM-1 digital sampling module

The DSM-1 is a 16-voice multi-timbrel sampling module, with 1meg of 12-bit sampling memory. The unit is compatible with DSS-1 disk, which features samples and programs from the company's DSS-1 disk library.

Circle (155) on Rapid Facts Card

Turbosound TFM-2 monitor

The compact floor monitor is a 3-way, bi-amped monitor rated at 450W. It employs the Turboconcentric loading device, which combined with a specially designed bass section allows sound reproduction at higher power levels, the company says.

Circle (156) on Rapid Facts Card

API 3124/3124M microphone pre-amp

The unit fits in one rack space and has XLR mic inputs that are internally selectable for 150Ω or 600Ω. The 3124M has the same features of the 3124 plus a stereo bus, auxiliary bus, mix level control, panning, an auxiliary send and an optional insert point for each channel.

Circle (158) on Rapid Facts Card

HME RM77 reverb mic

The RM77 has reverb circuitry built into the mic. It comes with an adjustable reverb control, a 3-position switch with mute, echo and normal settings, and a built-in pop filter.

Circle (159) on Rapid Facts Card

Focusrite Sidecar mixer

The unit is a 16-8 mixer, which can be interfaced to main console logic and buses and used as a fully integrated extension. An alternative mic pre-amp and Focusrite equalization are also provided.

Circle (161) on Rapid Facts Card

Circuit Design Technology CGM-2 processor

The CGM-2 single rack space, stereo multifunction processor provides simultaneous or independent compression, gating and dynamic modification. It is accurate over an 80dB range with a total dynamic range of 114dB with a maximum undistorted output of +24dBm. The noise floor is 90dB.

Circle (138) on Rapid Facts Card

Pulizzi Engineering TPC 115-8 system

The TPC 115-8 power distribution and control system is available in two different models, the TPC 115-8-A and TPC 115-8-C. Both are 120V, 12 and/or 24 amps with an EMI/RFI filter for protection against electromagnetic interference and radio frequency interference.

Circle (139) on Rapid Facts Card

Audio Logic MT 66 stereo compressor-limiter

The MT 66 stereo compressor-limiter can be operated as two discrete channels of compression-limiting or linked together in a stereo mode so that the two channels compress without the loss of the stereo image.

The unit is capable of dynamic range compression from 1:1 to infinity:1, with up to 25dB of gain reduction. It incorporates a "soft knee" feature to give a more natural sound while reducing gain, and includes a gate to ensure quiet operation when no signal is present.

Circle (140) on Rapid Facts Card

Atlas/Soundolier MAC-1 omni-purpose adapter

The MAC-1 omni-purpose adapter clamps directly onto any free-standing instrument and can be used to add multiple mics to individual floor and desk stands. The adapter accepts all ⅝-inch, 27 female threaded mic holders and standard accessories.

Circle (141) on Rapid Facts Card

Applied Creative Technology DB8 console interface

The DB8 console interface adapts instruments, effects units, reverbs, signal processors, tape recorders and other line-level output devices to the mic inputs of a console. With the interface, users can connect all instruments, including each output channel of drum machines and samplers, to the mic inputs of their console along with all the outputs of their tape machines.

Circle (142) on Rapid Facts Card

Nady 49 mini wireless system

The 49 mini wireless system operates on 49MHz frequencies, and receiver filtering circuitry allows the simultaneous use of two channels in the 49MHz band at the same location.

The receiver can be powered either by its internal 9V battery or by an external ac/dc adapter. The system is available with a choice of three transmitters: the 49 HT handheld, the 49 GT bodypack, or the 49 LT bodypack with attached lavalier mic.

Circle (143) on Rapid Facts Card

Stewart Electronics MP-2 mic preamplifier

The MP-2 microphone preamplifier is a low-impedance, single-channel mic preamp that sends signals directly to tape. The unit includes rack mounting, LoZ XLR cannon-type connectors, peak overload LED indicator lamp, active 3-band EQ with sweepable midrange, switchable 20dB input pad, phase inversion and EQ defeat switches. It comes with a switchable 48V phantom power supply.

Circle (144) on Rapid Facts Card

Carroll Touch input systems

Scanning infrared touch input systems have been introduced for the Mitsubishi HF1400, HF2400 and HF3400 monitors. The three displays are 13-inch, high line-rate, color monitors with an in-line gun and

New Products

analog input. Resolution is 512x400 on the HF1400, 720x540 on the HF2400, and 1,024x780 on the HF3400. The system fits both the plastic enclosure model and the chassis version.

Circle (146) on Rapid Facts Card

Roland VP-70 voice processor

The VP-70 voice processor converts monophonic sound sources into MIDI information. The 4-pitch shift circuits of the unit convert the signal to any 4-part chord. When assigning a multiple note chord, the user can adjust the internal individually for each note plus or minus two octaves.

Circle (148) on Rapid Facts Card

Stewart UDP-2 preamplifier

The UDP-2 preamplifier will process any signal from acoustic guitar to line level. Complemented by both HiZ and LoZ outputs, the preamp can be used to drive any input.

Features include front- and rear-panel ¼-inch phone-type inputs, bass and treble adjustments with sweepable midrange control, and switchable rear-panel output level. An effects loop has been included to interface with signal processing gear.

Circle (149) on Rapid Facts Card

Audio + Design SoundMaestro and DIGI-4 recording systems

The SoundMaestro (previously called SoundMaster) recording system uses the Atari Mega ST computer with 4Mbytes of memory and the capability of controlling up to 16Gbytes of hard-disk storage. The system has 2-channel full bandwidth 16-bit at frequency of 44.1kHz and 22.5kHz.

The key part of the hardware is the SoundStreamer interface unit with 256kbytes FIFO buffer memory and a control system that links the Atari's ROM port to Sony recorders.

The DIGI-4 recording system provides electronically balanced XLR inputs/outputs, +22dBm clip level, coincident time correction, unity-gain 2dB stepped precision stereo attenuators, pre-emphasis switching and copy prohibit switching and video sync.

Circle (129) on Rapid Facts Card

Wheelock line preamp and page port expander

The PRM-150 line preamplifier and page port expander takes 600Ω input and provides 4Ω output, while boosting low-level audio signals. The unit includes a volume control.

Circle (131) on Rapid Facts Card

Marshall Electronics Mogami Neglex-X multi-cable

The Mogami Neglex-X series multi-cable features a number-coded/color-coded individual pair identification system and a drain wire with each pair. Each pair features two 25AWG conductors or bare oxygen-free copper (OFC) with matching bare OFC served shield.

The cable is available in 4-, 8-, 16- and 24-pair.

Circle (132) on Rapid Facts Card

Innovative Tech Works VTR controller card

New Products

The PC VTR controller card allows simultaneous control of two professional-type VTRs equipped with the SMPTE RS-422 serial communications protocol. The card occupies one IBM PC or compatible slot. Menu-driven software is provided, and the card is designed to be used with PC-based paint and animation systems.

Circle (133) on Rapid Facts Card

Shure 849-LC and 869-LC condenser microphones

The 849-LC is a ball-type condenser mic designed for vocal applications, and the 869-LC condenser mic is best-suited to instrument micing and recording. Both models include heavy-duty shock mounting, complete protection against outside RF interference, and both can be powered by standard power sources or by a 1.5V AA battery.

Circle (134) on Rapid Facts Card

Renkus-Heinz P-1500 Smart power amp

The P-1500 Smart power amplifier is rated at 300W per channel into 8Ω, 500W into 4Ω and 1,500W into 4Ω bridged. The amp prevents hard clipping and includes a high-frequency oscillation protection feature. Input-output connectors include XLR, ¼-inch phone, dual-banana and terminal strip.

Circle (135) on Rapid Facts Card

Black Audio Devices connector templates

Connector templates speed the process of fabricating connector panels by eliminating the extensive drafting required to lay out the panel before drilling.

Made from spring-tempered stainless steel, the templates are for use with D3M- and D3F-type connectors.

Circle (136) on Rapid Facts Card

Applied Voice Technology CallXpress

The CallXpress voice and call processing system combines inter-application linking, user programmability, alphabetical directories and extension specific processing. Callers can move between automated attendant, voice mail and other application modules without having to hang up and redial or go through an operator.

Callers can also use the touchtone keypad to spell out names to obtain directory information. The system also allows each extension to have a customized menu of alternate choices for call routing.

Circle (137) on Rapid Facts Card

Professional Sound SONOSAX SX-PR series mixer

The SONOSAX SX-PR series modular stereo ENG/EFP mixer is available in three sizes—2-, 4- and 6-channels—and features a master module and an input module with two channels in each module.

Equivalent input noise from a 150Ω source is −129.2dB. Frequency response is ±0.5dB from 30Hz to 20kHz.

Circle (124) on Rapid Facts Card

TDM 30GE-1 graphic equalizer

The 30GE-1 2-rack space, mono, 30-band

New Products

graphic equalizer incorporates three notch filters, each having sweepable frequency and variable depth. There is a range switch to select between 6dB and 12dB boost or cut operation.

All inputs and outputs are balanced or unbalanced.

Circle (125) on Rapid Facts Card

Sequential Prophet 3000 digital sampler

The Prophet 3000 digital sampler features 16-bit linear and true stereo sampling capability. Sampling rates include approved industry standards of 48kHz, 44.1kHz, 32kHz and 16kHz. The system also features real-time sample monitoring, preset modes for mapping, layering, switching or crossfading presets, sustain and release loops, loop crossfading and compression facilities.

The instrument has an additive synthesis mode making it possible to create or edit sounds by drawing waveforms using the cursor and varying sine levels while viewing a bar graph.

Circle (123) on Rapid Facts Card

Soundtracs MX series console

The MX series sound reinforcement console is available in three mainframe sizes: 40, 32 or 24 inputs. Four-band equalization (with two sweepable mids) and six dedicated auxiliary sends are incorporated on the input modules without the use of concentric potentiometers, along with a 8x4 matrix and eight dedicated effects returns on the groups.

Four pin XLR sockets are provided in the left- and right-end cheeks for high-intensity "Littlites."

Circle (122) on Rapid Facts Card

Akai MB76 mix bay

The MB76 MIDI programmable line mixer has seven line inputs and six line outputs, with 32 memory banks that can be programmed individually or through bank copy functions.

The 32 banks can be called up via MIDI directly through MIDI program change command, or via footswitch through the bank-up jack. The unit will operate as a sub-mixer to channelize effects.

Circle (101) on Rapid Facts Card

Automatic Connector fiber-optic link

The 40-97003-000 fiber optic link has a transmission capability of 2km and a repetition rate of 64K-bits/s. The unit operates as a link between two RS-232C electric connectors.

An external 12V dc power source can be used, or the link can use power via the RS-232 connection. Output power level is −23dBm, and time delay is $20\mu s$ plus 5ns for each additional meter of fiber.

Circle (104) on Rapid Facts Card

Yamaha QX3 digital sequence recorder

The QX3 digital sequence recorder provides 16-track digital sequencing with up to 48,000 notes recorded with velocity. The unit includes a built-in 3½-inch floppy disk drive and can record anything played on a MIDI keyboard.

The real-time mode includes three types of punch-in options that record only the

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New Products

information between user-set punch-in and punch-out points.

Circle (105) on Rapid Facts Card

Soundtracs MRX series console

The MRX series recording console includes 4-band EQ with two sweepable midranges and six dedicated auxiliary sends on the inputs. Also included is 3-band EQ, six auxiliary sends and 16-track monitoring.

Metering is provided by 10 segment LED bargraphs for 16-track tape, auxiliaries, LR mix and solo.

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Symetrix 511A noise reduction unit

The model 511A 2-channel, single-ended noise reduction unit provides up to 30dB noise reduction without encoding. Each channel includes a high-frequency filter and a downward expander.

An 18dB/octave rumble filter is included for treatment of low frequencies. The unit operates with balanced or unbalanced +4 or −10 nominal levels.

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Audio Services SQN-4S mixer and services

The SQN-4S 4-input mixer has complete microphone powering as well as gain switching at the pre-amps, allowing the inputs to accept a wide range of signals.

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Electro-Voice Extended Range speakers

Extended Range 2-way speakers models SH-1502ER and SH-1512ER feature the DH2010A titanium diaphragm, high-frequency driver, and a high-frequency output that delivers flat response to 20,000Hz.

The low-frequency section of both enclosures is powered by an EVG 15-inch woofer that uses a flat-wire aluminum voice coil and has a long-term power handling capacity of 200W.

Both high-frequency sections use the Time-Path phasing plug and a 90°x40° constant-directivity horn.

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TimeLine film module and interfaces

The new LYNX film module reads and generates biphase film synchronization signals and is compatible with all biphase standards from 2x to 100x frame rate. The module can be used to jam sync time code from a film recorder and can lock video machines to film transports, including reel-to-reel digital multitracks and PCM digital systems using VTRS.

The Sony BVH-2000 serial interface for LYNX SAL synchronizer modules adds a standard P-2 communications/control port to the module for control of a BVH-2000 VTR.

The Sony 5850 interface for LYNX VSI and SAL modules allows video editing systems to control the VTR. The TimeLine 5850 encodes and decodes editor commands for its non-standard communications port.

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R·E/P

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Classified columns are not open to advertising of any products regularly produced by manufacturers unless used and no longer owned by the manufacturer or distributor.

Classified Advertising should be sent to RE/P, Advertising Department, 9221 Quivira Rd., Shawnee Mission, KS 66215.

MISCELLANEOUS

WANTED—OLD SOUND EQUIPMENT (Tubes, Amps, Speakers, Drivers, Horns, etc.), from McIntosh, Marantz, M-Levinson, Altec, JBL, Jensen, Tannoy, Western Electric, Westrex, Langevin, etc., Tel: 818/576-2642 Audio City, P.O. Box 832, Monterey Park, CA 91754. 8-87-12t

WANTED: Pultec EQ's. We will pay $1,000 for almost any Pultec program EQ models EQP1/EQP1A/EQP1A3. Also wanted EQH2/EQH3/MEQ5/MAVEC/MB1/any tube or ribbon microphones and limiting amps. Please call or write to: Dan Alexander Audio, P.O. Box 9830, Berkeley, CA 94079, (415) 474-1625. 11-87-1t

FOR SALE

NEUMANN/SCHOEPS/AKG tube mikes in excellent condition, Gotham checked W. new capsules in most cases; VF-14 tubes for U-47; also 'telcom' N/R cards; Send SASE for complete listing to: Stephen Temmer, 767 Greenwich St., NYC 10014. 11-87-2t

SOUND IDEAS Effects Library. 125 Boxed Reels. 15 IPS. Cross Referenced to over 13,000 entries. Comes with Catalogue and 3-ring binder. $1500.00 OR BEST OFFER. SOUND VENDORS 818-985-9774. 11-87-1t

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4. V Series Central Facilities

"It was twenty years ago today. . ." and at least one thing hasn't changed. Since Record Plant Studios first opened its doors in New York two decades ago, innovation and excellence have forged a legend.

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The V Series Central Facilities section is one of the reasons Neve finds its way into the most discriminating recording environments. Features such as multi-function programmable solo with central reset, split or in-line operation, user-defined multiple master status switching and onboard stereo cue sub-mixers, all bring rave reviews from the people who have experienced the difference.

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When Roy Cicala renewed his commitment to the best in the business, he turned to Neve. "The bottom line. . . Neve got me back."

Roy Cicala

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