

September 1987

Recording

ENGINEER/PRODUCER

The Technical Journal for Audio Professionals

AN INTERTEC PUBLICATION

Lease/Purchase
Decisions
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\$4.00

Console Developments



Take a Neotek to work for a test drive.

No one judges performance just by reading the specs. To appreciate a recording console you have to put it through its paces. Hear what it'll do in your own studio. With your material. Your room. That's where the rubber meets the road.

We're so sure you'll like what you hear, we want to offer you an easy way to compare our consoles to any others: Borrow a Neotek demo module.

Fleetwood Mac did. They ended up cutting an album on it. We found out when they asked for a second one for stereo tracks.

Check out what's under the hood. Hear our mic pre with the finest condenser mics. That's an easy one; manufacturers already use our consoles to demo their best. Try synth inputs and see why New England Digital uses Neoteks to show off the Synclavier. Drive it with a digital track. You'll hear how our consoles make digital recorders sound their very best.

While you're at it, check the construction quality, the control layout, the flexible signal flow, and our very special equalizer.

Then, if you can wait a few weeks, we'll put you in the driver's seat of a new Neotek console, custom built for your personal requirements. Like we did for Stevie Wonder.

The way we figure it, we haven't come in second in a comparison yet. So we can't lose.

Neither can you.

So if you want to hear for yourself what's so special about a Neotek, drive one at work.

And if you're planning to keep our demo module for your next album tracks . . . well, that's OK too.



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Circle (1) on Rapid Facts Card

Listen to what engineers in 47% of all recording studios have already heard.* And what they haven't.



YAMAHA
NS-10M STUDIO



YAMAHA
NS-10M STUDIO

What we're going to tell you about the new NS10M Studio reference monitor may sound familiar, and for good reason.

The NS10M Studio is based on our legendary NS10M which, judging from its popularity in recording studios, delivers the near-field acoustic imaging that most engineers have demanded.

So rather than listen to competitive monitors to improve the NS10M, we listened to professionals like you.

And ended up retaining the best aspects of the NS10M's performance, while enhancing others.

That means you can expect the same smooth frequency response. The same high-power handling capability. And the same ability to take on the stresses of a longer duty cycle. All while maintaining accurate spatial definition without inducing listener fatigue.

Listening to what engineers needed also

meant making refinements designed specifically for the studio environment.

Like connector terminals that accept large-diameter speaker cable for optimum signal quality. A 3.5cm dome tweeter with built-in acoustic damping tailored for near-field monitoring. And a horizontal configuration so the NS10M Studio never gets in your way.

And because it takes a pro to better service a pro, the NS10M Studio is sold exclusively through authorized Yamaha Professional Audio dealers.

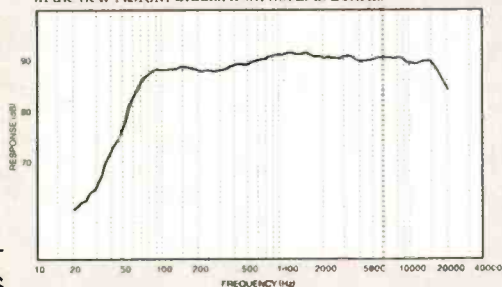
The NS10M Studio.

Proof that at Yamaha, we listen to professionals as much as professionals listen to Yamaha.

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*1987 MIX Magazine Annual Recording Industry Directory

Frequency response remains exceptionally smooth in the new NS10M Studio, from 60Hz to 20kHz.



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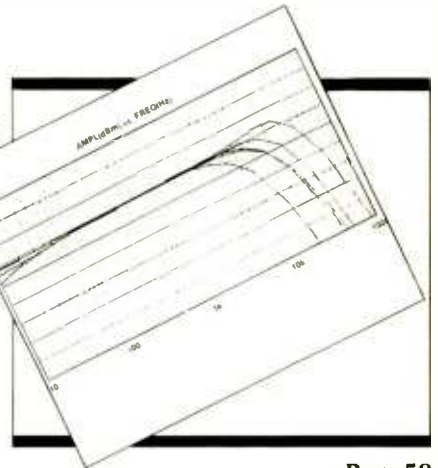
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Console Developments

As we transition from analog designs, through assignable topographies to the all-digital consoles of the next decade, what are the key operational and technical parameters that manufacturers and users alike need to address?

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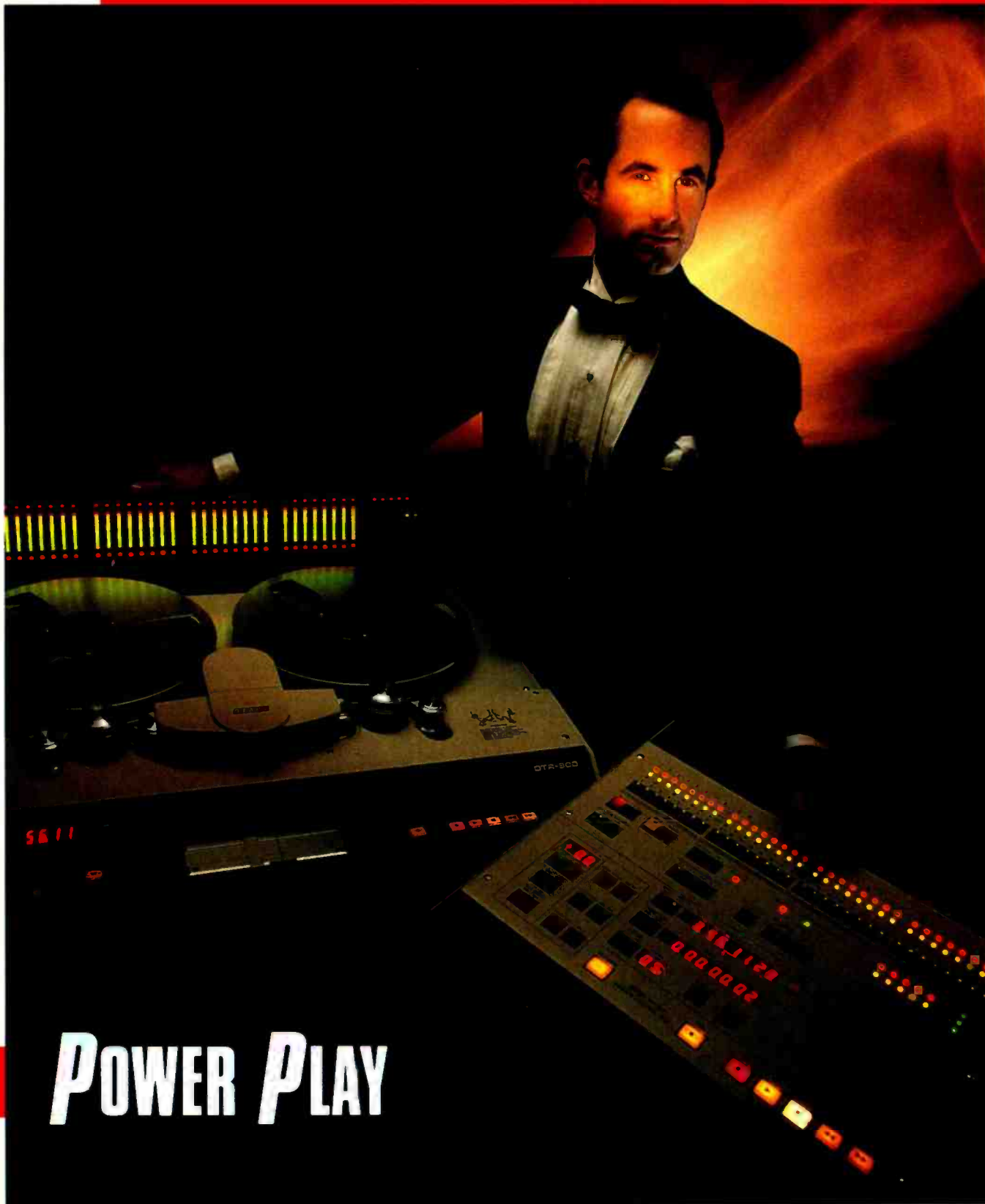
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An Amek APC 1000 console, 80 module mainframe with 56-input channels fitted with GML moving fader automation at Green Street Recording, New York. Photo by Bill Waltzer.

RECORDING ENGINEER/PRODUCER—Volume 18, No. 9—(ISSN 0034-1573) is published monthly by Intertec Publishing Corporation, 9221 Quivira Road, P.O. Box 12901, Overland Park, KS 66212-9981. Second-class postage paid at Shawnee Mission, KS, and additional mailing offices. POSTMASTER: Send address changes to Intertec Publishing Corporation, P.O. Box 12901, Overland Park, KS 66212-9981.



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Circle (5) on Rapid Facts Card

Rational Audio

By Mel Lambert, Editor

A recent experience at what the majority of us would consider a "world-class," high-visibility studio underscored for me the convoluted nature of the recording and production industry.

On the one hand, this multiroom facility was equipped with just about everything that's happening in contemporary consoles, tape machines, monitoring and outboards (and the vintage mic cupboard was particularly well stocked with examples of European and American memorabilia).

On the other hand, I was hearing tape tracks that were so distorted I couldn't believe my ears. No way could anyone have failed to have noticed the whole numbers of THD and TIM on the solo lead vocal and guitar tracks. That the basics were destined for a rock-and-roll album release was incidental; the band in question is well known for its powerhouse lead lines and axe solos.

Buried in a rough mix, the tracks were definitely more acceptable to my ears. And, by the time we reached the final remix sessions, the distortion that had been so offensive a couple of days ago—for those of us with those sorts of ears—was now holding the mix together real well, and adding just the amount of "bite" and "edge" that had been the producer's intention.

So what's the bottom line here? Does distortion really matter? In absolute terms the only answer any of us could give would be a resounding "Maybe!"

Think about it. When did equipment specs and technical data play a major part in the creative process of making music (not to mention movie soundtracks, video post, live sound and the other myriad species of audio production).

OK, I'd be the first to admit that we need to make sure that the overall process of capturing the sound from the studio or a box of electronics, recording it to multitrack (analog or digital) and subsequently playing it back should be as faithful, as linear and as quiet as we can make it.

All of which means paying particular attention to mic quality (and not just how it was meant to sound when new, but how several years of misuse and neglect might possibly affect their performance); console gain structures and headroom (including minimal path techniques, and maybe the possible applications of external mic pre-amps and equalizers) bus output levels and interface to the multitrack (while many of us spend time sorting out the front-end of a

console, fewer will examine its output capabilities, or the headroom performance of the multitrack's input and output amps); tape machine and noise-reduction alignments, including record, bias and EQ calibrations (an optional process with DASH/PD transports).

But if the session chemistry calls for overload—artistically as well as sonically—then I would predict a short career for an engineer who cannot live with the truth that any sound is valid for that particular song or performance.

Although making sweeping generalizations—including this one—is a dangerous thing to do, let me come up with a working definition of what an attentive engineer should be taking care of and which, many of us would consider, is particularly well suited to surviving in today's cost- and talent-competitive industry.

Any engineer, whatever his or her recent background—either having risen through the technical ranks, or began as a musician/performer and made the move to the other side of the glass—has one fundamental task to perform in the control room: make sure that the hardware is being run in such a way that every nuance of the audio performance can be captured. Period.

If the producer—who, in this day and age, is just as likely to be the engineer—wants to saturate the tape and practically scorch the binder off the back...that's what should be done. Just so long as the translation process from studio to tape to playback is as faithful as modern-day technology will allow, then the engineer is taking care of business.

I should also mention that the session I attended just a couple of weeks ago was being tracked to digital multitrack and was destined for remix to 2-track stationary-head digital. I don't think that I am being facetious when I say that what I heard in the control room those evenings was some of the most faithfully recorded whole-number THD that I have heard in a long while. I expect the resultant album release to do very well for all involved.

RE/P

2. Neve V Series Dynamics



Engineer Rex Recker of Photomagnetic post production facility in New York City is used to working with on-location tracks. But he can't imagine working without the Neve 60 channel V Series console with Necam 96:

"It's great for cleaning up tracks and getting rid of unwanted background noise or room tone.

"And noise gating with the V Series is so simple. You can get the quality and sound you want, quickly...when it counts. It's a luxury which, once experienced, becomes a necessity.

"We already have two Neve V Series consoles and we're getting a third."

The signal/noise gate and limiter/compressor combine to produce an unsurpassed in-channel dynamics unit. There is also a "key" input from the patch bay for external gating to give greater control of effects. And all units can be linked together for perfect stereo operation.

Neve makes it easy to remove spill-over, tighten up rhythmic sounds and create dazzling special effects, whether tracks are made in or out of the studio.

There's no clipping. No pumping. Just warm, compelling sound from the most musical dynamics unit yet available.

The Neve V Series Dynamics is only one of the powerful reasons top recording professionals like Photomagnetic's Rex Recker choose Neve.

Neve...Only the Best for the Best!

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Circle (6) on Rapid Facts Card

News

Otari moving to new U.S. headquarters

The company recently started construction of its new U.S. headquarters, with a projected move-in date of January 1988. The new building will be located within the Vintage Park Development of Foster City, CA.

New additions will include a sound room, customer training facilities and a special test room for Otari's laser-based thermal magnetic video duplicator (TMD).

NED opens new center in Chicago

New England Digital's new regional sales, service and training center, located in the heart of Chicago's production district, is the fourth center the company has opened in the last two years.

"Over 40% of all commercial broadcast audio comes out of the Midwest," says Franklin B. Sullivan, NED vice president of marketing and sales.

The new Chicago center is located at

741 N. Dearborn Parkway, Chicago, IL 60610; 312-266-0266.

Loggins tour debuts flying array

Kenny Loggins' 7-week, 30-city tour, which began in mid-July, marked the debut of the A-1 Audio/Meyer Sound Lab integrated flying array. By combining a variety of Meyer speaker cabinets with patent-pending flying apparatus designed and built by A-1, "The entire system has been able to be flown in times with minimal manpower," according to Albert V. Siniscal, A-1 Audio owner.

The new system, consisting of Meyer MSL-3s, USW-1s and 650R-2 subwoofers, remains largely intact for transport, thereby eliminating lengthy assembly time and maximizing truck space. Amplification includes Crest 4001 power amps, plus a mixture of Meyer, Turbo-sound and A-1 Audio monitor enclosures.

Loggin's long-time studio engineer Terry Nelson will be handling the house

console, with monitor mixer Alan Richardson overseeing the stage duties. A-1 technicians Jim Stark and Lee Rickmers provide system support.

NECAM 96 now available for non-Neve consoles

According to John Andrews, sales and marketing director, "The growing popularity of moving fader-systems meant that we have come under increasing pressure to sell NECAM for fitting to other manufacturers' consoles."

Gotham to distribute Audio Developments mixer line

The range of portable mixers, designed for use in both film and television field production, includes the 145 Pico available with 4, 6 or 8 inputs, the 062 in versions from 4 to 16 inputs, and the 160/260 ENG mixers. The Pico is also available in a drop-in desktop version for in-studio applications, including video editing.

Recording

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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.

CORRESPONDENCE

Advertising and Subscription:
9221 Quivira
Overland Park, KS 66215
913-888-4664
Telex: 42:4156 Intertec OLPK
Fax: 913-888-7243


Editorial:
Suite 220
1850 N. Whitley Ave.
Hollywood, CA 90028
213-467-1111
Fax: 213-856-4895
IMC Email: REP-US

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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.

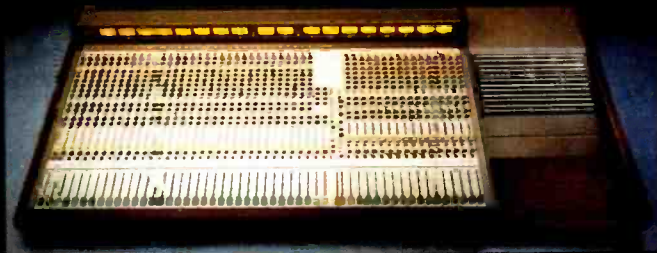
The M-600 is modular. Which means you can custom configure the console to *your* audio or video production needs. The M-500 lets you choose up to 32 input channels, or you can start with 16 or 24 input channels and expand the board as your needs change. Optional stereo modules can also be added to provide even more line inputs for MIDI instruments and video production convenience.

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

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

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

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



TASCAM

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Circle (7) on Rapid Facts Card

Audio Developments recently expanded its line with the Port-A-Flex system of audio processing units. These battery-powered devices include headphone drivers, line splitters, metering, filtering, compressor/limiter and balancing interfaces.

Everything Audio relocates to larger facility

The company's new offices, located at 2721 W. Burbank Blvd., Burbank, CA 91505, include two demo rooms and an expanded stockroom. The new service department accommodates four technicians for "Better QC, service and repair turn-around time."

Sony Broadcast to launch Pro-DAT deck

The company intends to market professional digital audio tape decks throughout Europe, starting in the fall.

The first professional DAT machine to be made available will be the PCM-2500. A 2-unit product, the PCM-2500 fits into a 19-inch rack and is tentatively priced at about \$5,700. Unlike the DTC-1000ES, the single-unit domestic model already available in Japan, the PCM-2500 has no record inhibits at 44.1kHz.

Sampling frequency of the 2500's analog input is 44.1kHz and 48kHz, while the digital input can also sample at a third frequency of 32kHz. All digital inputs and outputs correspond with a variety of formats, including AES/EBU and SDIF-2 (PCM-1610/1630 format).

The unit also incorporates its own built-in error rate indicator in playback mode, and can add a copy prohibit code while recording. Other features include emphasis on/off for analog recording, a master safe record function and a voltage selector.

Sony plans to release a portable DAT recorder at the beginning of 1988. The PCM-2000, first premiered in prototype form at the London AES Convention in March, will offer time code capability. It is provisionally priced at around \$8,600.

New textbook spotlights audio-for-video

Reflecting the increased importance that sound quality now plays in video production, *Audio Production Techniques For Video* is intended to provide an explanation of the basics of audio recording, video recording, digital audio and editing, and is published by Knowledge Industry Publications.

Various chapters cover stages of audio/video production and post-produc-

tion encountered with audio and videotape recorders; the basics of time code, synchronization, edit decision lists and edit controllers; microphones and mic-ing techniques; audio-for-video post-production equipment and techniques; the sound effects library; electronic music production; signal-processing and effects devices; and production consoles.

The 356-page hardcover book costs \$39.95; more details from Knowledge Industry Publications, 701 Westchester Ave., White Plains, NY 10604; 914-328-9157.

Sound Genesis introduces computer music services

The company's first sonic software product line for producers is scheduled for introduction in August, and will be implemented on the Fairlight CMI Series II and Series III digital synthesizers.

According to company president Don Webb, "The Master Sampler Collection will contain 20 or more separate packages of documents, software and media, each of which includes 10 'virtual instruments' from one of three categories: instrumental, vocal and environmental.

"Every virtual instrument carries a pedigree that documents its authorship and prior usage. Its purpose is to help professional sound producers rest easy in the certainty of their explicit right to use the virtual instruments in producing their own commercial products, free from any concern of copyright infringement."

Digital delays used in scientific study

Scientists studying animal behavior at the Rockefeller University, Millbrook, NY, are using digital delays while recording bird calls for studies of learning and behavior in birds and their similarities to humans.

Dr. Stephen Nowicki, studying animal behaviors at the university's Field Research Center, says, "It is usually impossible to tell when a bird will sing, so the normal technique while recording bird calls is to leave a tape recorder running for a long time, and then edit out the dead spaces between the bird songs. It can be a very tedious job."

Rockefeller University scientists use a tape machine equipped with a voice-actuated device that starts the transport when a bird sings. A DigiTech delay line is used to delay the sound long enough for the tape machine to come up to speed, and still not miss the beginning of the bird's song. The result is an automatic editing of the recording more

suitable for study, Nowicki says.

Peirce-Phelps completes sound system for Bucknell University

The company's Audio Systems Division recently completed installation of a specialized sound system for the experimental Tustin Theater at Bucknell University, Lewisburg, PA.

"The theater is essentially a black box with four walls, bare floor and lighting grid that allows for maximum flexibility in putting on productions," says Steve Chene, lighting designer and associate technical director for the university's theater department. "A major advantage for such an avant-garde concept is that the stage can be placed anywhere. For example, we can position the stage in a three-quarter thrust, conventionally, or mix actors and audience together."

People

David L. Prentice has been appointed sales engineer for **New England Digital's** New York sales and service center.

James A. (Andy) Moorer, has joined **Sonic Solutions** as vice president of audio research.

Stan L. Freeman has joined **HM Electronics** as customer service manager. **Al Zang**, a 14-year veteran of the pro-audio industry, joins the company as product specialist. Also, **John Bucknovitz** joins HME as product specialist in the industrial group.

Joe Green has joined **Fender Musical Instruments** as director of marketing for sound reinforcement products, with responsibilities for the company's Pro Sound and SUNN products.

API Audio Products has appointed **Jim Wallace** as vice president of finance and director of marketing. In addition, **Kevin W. Raynor** has been promoted to director of manufacturing and quality assurance manager.

Digital Audio Disc Corporation has appointed **Scott N. Bartlett** as director of sales and marketing. In addition to directing sales activities, he will oversee all advertising, promotion and customer relations for DADC, Sony's CD manufacturing subsidiary. **RE/P**

Please send News updates to the RE/P editorial and production offices. Editor, 1850 N. Whitley, Suite 220, Hollywood, CA 90028 and Dan Torchla, staff editor, Intertec Publishing, P.O. Box 12901, Overland Park, KS 66212.

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.

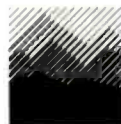
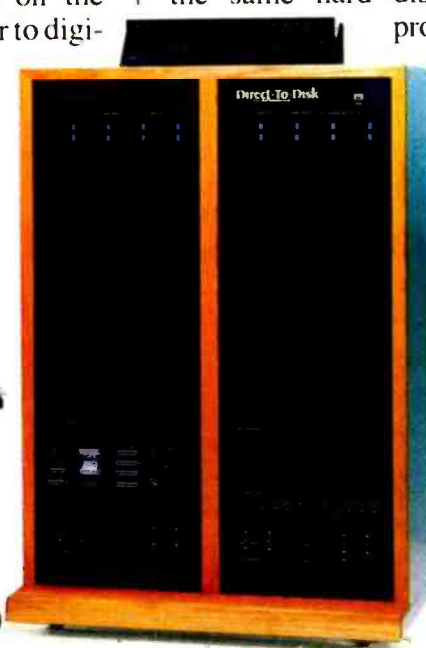
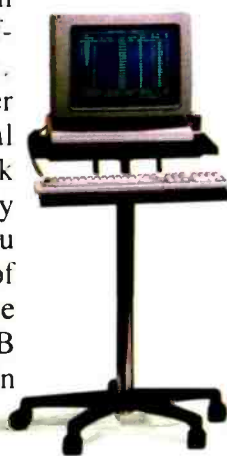
The stand-alone Direct-to-Disk is based on the same hard disk storage and proprietary processing technology that has made the Synclavier® the industry standard for reliable performance in the studio and on the road. And like the Synclavier, the Direct-to-Disk system is modular and **software updateable**.

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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Circle (8) on Rapid Facts Card

Letters

From: Allan Lamshead, vice president, director of engineering, Evertz Microsystems, Ltd., Burlington, Ontario, Canada

With reference to Larry Blake's article in this issue of **RE/P** titled "Time Code and Synchronization: The Link Between Film Production and Post-production," your readers might be interested in the following note of clarification.

The ECM 4010 brings an additional capability to the Telecine-VTR transfer process. The original negative's foot and frame numbers are encoded into the user bits of VITC which is placed on the 1-inch master and all work prints of it. The ECM 4010 quadrature input tracks the 3:2 sequence on a field accurate basis, ensuring that the original negative's frame has been properly identified on the video tape.

When the edited master video tape is complete, the ECM 4010 reader can detect discontinuities in either time or user bits, and generates a field accurate negative cutters list directly on a printer or to a computer. The numbers can be decoded at still frame, so there is no need to burn in.

Successful recovery of VITC after many generation losses is possible with the upgraded VITC reading techniques.

Copycode Scanner

From: David P. Reaves, III, manager of radio operations, WNYC, New York.

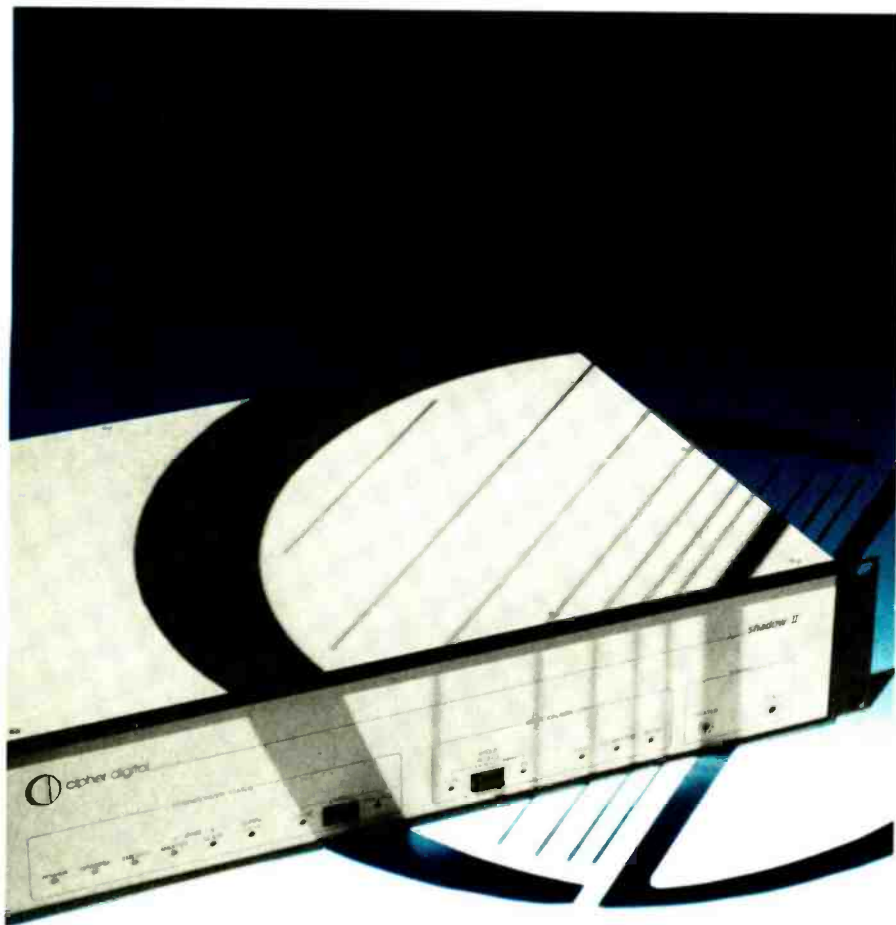
It was with great interest I read the letter from Bart Locanthi in the May **RE/P**. Although the entire DAT/copyright issue is very complex, with a great amount of business at stake, the implementation of the CBS Copycode system seems to 'ring' false.

CBS made perceptual tests concerning the human ear's loudness sensitivity as related to frequency, with several resulting CBS Labs devices that make extensive use of their conclusions. I immediately bring to mind the CBS Loudness Indicator, Dynamic Presence Equalizer and Loudness Controller.

All three of the units' designs were predicated on the idea that human hearing is *most sensitive* in the range several hundred hertz either side of 4kHz. Thus it is surprising to find that Copycode's 80dB notch falls in that critical range.

Seeing Bart Locanthi's graphic presentation of the effects of Copycode [analog notch filters] makes me hope that this audio abomination never gets widespread use among software producers. And that its success mirrors CBS' lesser creations: CX, SQ and the TV color wheel.

RE/P



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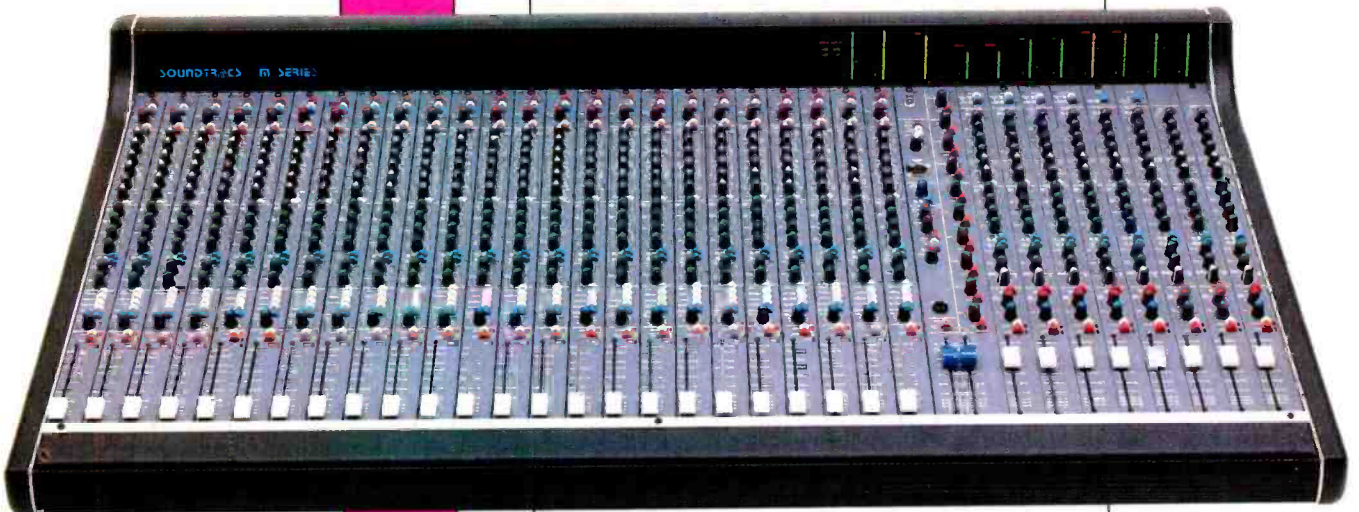
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Managing MIDI

By Paul D. Lehrman

The other day I was telling a client about a software program that would map aftertouch to pitchbend, so that the harder I pushed a key, the higher the pitch would go. I illustrated it with a synthesizer whose pitchbend range had been set to two semitones.

"Yeah," he said, "but I'll bet you can't make it bend a whole octave." After I picked myself off the ground, I pondered what to say. "Be nice," I thought. "He doesn't know how *stupid* he sounds."

Those of us who work with MIDI every day, and who have spent the equivalent of several lifetimes poring over the minute details of the MIDI Spec and its additions and addenda, sometimes forget that the whole world hasn't had the same experience, and that otherwise perfectly competent people sometimes have gaping holes in their knowledge of MIDI.

This is too bad, because such people are cutting themselves off from realizing the full potential of their equipment. For them, I'm going to devote this column to answer some questions about MIDI; questions which might be considered so dumb that anyone who doesn't know the answers is afraid to ask.

• Why is a note-off not always a note-off?

Although there's only one note-on ("key down") command in MIDI, there are two ways to specify a note-off ("key up"). Every note command has a velocity value following it; if a note-on command is followed by a velocity of zero, it's understood that this is to be read as a note-off.

The reason for this is that MIDI allows a sped-up method of data transmission known as running status. In running status, if two commands of the same type are to be sent consecutively, you don't have to send the actual command a second time, just its value. If you send note-ons with zero velocity instead of note-offs, you can use running status.

Unfortunately, not all synthesizers read such data correctly (this is especially true of pre-MIDI synths that have been MIDI-fied). Occasionally, a problem crops up that usually results in stuck notes, because the receiving synth doesn't know it's supposed to shut up.

Note-off velocity values are not particularly important at the present time. There are a couple of MIDI keyboards that generate genuine note-off velocity,

and quite a few sequencers can read it, but no one, to my knowledge, is actually doing anything musical with it yet. This will probably change soon.

• What's the difference between velocity and pressure sensitivity?

Velocity sensitivity means that a keyboard can tell how fast you've hit a key. (Usually this is equivalent to how *hard* you've hit it, but not necessarily.) Velocity data is used to determine a sound's volume, brightness, attack time, or other appropriate characteristic. Pressure sensitivity—also known as aftertouch or

Some otherwise perfectly competent people have gaping holes in their knowledge of MIDI.

Channel Pressure—means that a keyboard knows how hard you are pressing on the key at the bottom of the key-stroke *after* you've played it. This capability is usually used for vibrato depth, or brightness, etc.

• How many instruments can I put in a MIDI system?

There's no limit to the number of synths, processors, and other do-dads that can be loaded onto a MIDI cable. There is a limitation, however, on how many synths receive their own discrete channel, which is 16. You can separately control more than 16 devices, however, by asking some channels to do double duty.

With some synths you can also specify note limits beyond which they will not respond, so that one channel, for example, can play a bass voice which doesn't go above B 59 and, at the same time, play a flute which doesn't go below C 60.

• Why can't I record MIDI sync on tape?

For the same reason you can't put a digital audio signal on analog tape: the bandwidth is too high. Even though the sync word only occurs 24 times per quarter note, the speed of the datastream is still 31.25kHz, and it's a square wave to boot.

Fortunately, you can fairly easily convert MIDI sync into something else that is recordable. One method is to change it into an audio tone that is being modulated by another tone in step with the MIDI sync signal. (This is called FSK sync.) Originally, FSK was "dumb" and only gave information about start and stop

points and tempos. If you were locking a sequencer to a tape striped with FSK, you'd have to start it every time from the beginning.

• OK, what is Song Position Pointer, anyway?

This is a MIDI "word" that some devices send out to tell other devices where they are located in a sequence. A device that reads some form of tape sync (like the intelligent FSK mentioned above) and uses it to generate a Song Position Pointer message, can tell a sequencer exactly where the tape is at any time. In this way, the sequencer can set itself to start at the right measure and beat to match the tape position. Any MIDI sequencer that locks to tape has to use Song Position Pointer in one way or another; don't let anyone tell you otherwise.

By the way, don't confuse Song Position Pointer with Song Select (even though both are referred to as Song Pointer). The latter is a different kind of MIDI pointer, which specifies the number of the song to be played next, assuming there is more than one song cued up to play (like in a MIDI jukebox). This feature isn't used very much.

And here's where we came in:

• Why don't the pitchbends between my synths match each other?

A MIDI pitchbend command, like most MIDI commands, does not contain musical information. Instead, it sends generic information; the musical purpose that particular information serves is up to the receiver. Most synthesizers let you specify pitchbend range. If two synthesizers are set to different ranges, they will be affected differently by a single pitchbend command.

For example, if one synth's pitchbend sensitivity is set to 7 (semitones), then a full-up pitch wheel will change a C to an A flat, while on another synth set to 2, that C will change to a D. All intermediate positions of the pitch wheel will be scaled accordingly.

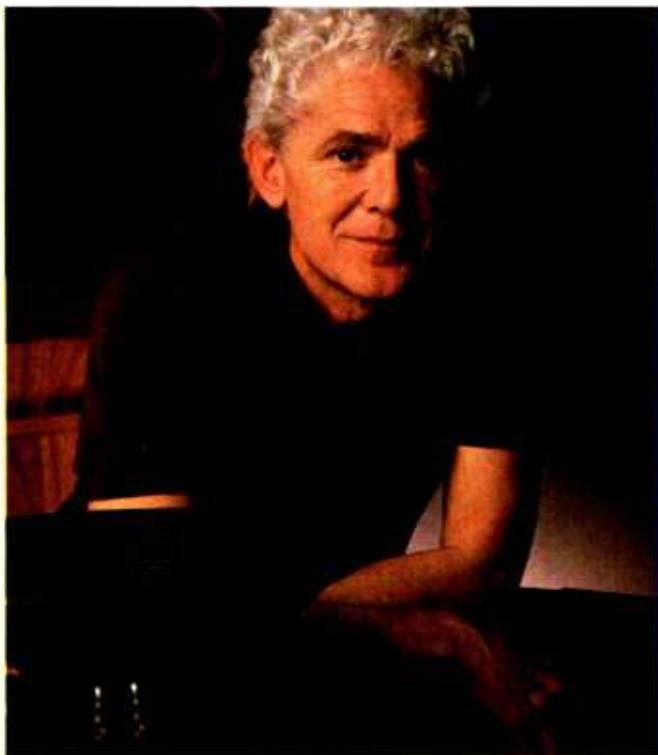
In addition, there are a few instruments whose pitchbend curves are reported to be slightly different from the norm: pitches at the bottom and top of the pitchbend range are correct but, in the middle, the pitch might not agree with another synth's. This is not a healthy development, and should be discouraged.

I'm sure there are many more confusing aspects of MIDI. If you have a favorite, let me know care of RE/P. Anonymity is assured.

RE/P

Paul D. Lehrman is a Boston-based free-lance writer, electronic musician, synthesist, producer and regular contributor to RE/P.

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Sound on the Road

By David Scheirman

While economists are busy wondering which direction the stock market will jump next, the entertainment industry is having a banner year. Historically, times of political or economic uncertainty for other industries have left entertainment untouched. For example, while the nation was battered by the Great Depression following the stock market crash of 1929, more vaudeville entertainers were on the road than ever before.

Whatever the reasons, this year's touring season is one of the most successful yet. Not only is every new act with a video in heavy rotation on MTV hurrying to put together a road show, but many successful name artists from the past two decades are also finding that the concert-going public is still interested in purchasing tickets to see their old favorites perform live in concert once again. The Monkees, Steppenwolf, Three Dog Night, The Temptations, America and the Everly Brothers are a few of the acts that have seen good box-office action this summer.

In addition, many of the industry's major arena and stadium draws from the recent past are finding this to be a good time to tour. Pink Floyd, Supertramp and Yes are preparing to embark on a series of major fall dates. The word is out that even such artists as Paul McCartney and Mick Jagger will be packaging live shows once again after seeing what a successful season is underway.

A healthy summer season for touring acts means a healthy summer season for concert sound companies. The simple fact is that live touring shows still require complex, specialized portable sound systems. Companies in the business of supplying such equipment are finding the demand for systems and personnel is actually beginning to exceed the available supply of road-ready systems and crews.

"Every piece of gear we have is booked into the late fall already," said a rental system manager for a major concert sound company. "We are double-booking some already, because the odds are that a few of these upcoming tours won't quite get it together. There are a lot of crazy deals going on out there right now, according to the booking agencies."

When an active summer season starts to draw in all of the available concert sound hardware from the major companies, many smaller regional companies find themselves in an advantageous posi-

tion. Oftentimes, major companies can only fulfill their scheduling commitments by subcontracting gear (often extra consoles, or "stacks and racks") from smaller companies that are often considered to be a competitive nuisance when work is harder to get.

In an industry that can fluctuate wildly from year to year, assembling an expensive concert-sound system on the outside chance that a bigger company will subcontract it on occasion is certainly not good business sense. However, those regional companies that service their own local

Increased concert activity has made the 1987 touring season one of the strongest ever.

accounts and are in the process of expanding their available stock may profit from establishing a dialogue with major sound companies that may be rental customers in the future. What does it take to make one or two single systems attractive for subcontracting?

A thorough knowledge of the methods and techniques used to prepare a sound system for road travel is one important factor. While the pro-sound manufacturing industry is rapidly approaching the point where an end user can specify and order a completely packaged concert-sound system ready to use right out of the box, there are still many things that a major concert-sound company requires of a system that off-the-shelf products do not offer.

Interfacing cable systems and electronics rack packages, along with proper power distribution systems, is a major priority when rental systems from an outside source are scrutinized. Correct engineering techniques for speaker-system design and assembly are also important because the overall sound quality results of the event are still the responsibility of the primary contracting company for the concert events.

Dramatic improvements in commercially available power amplifiers, cross-overs and speaker enclosures have given rise to a steadily growing group of regional sound companies that are assembling well-considered, competitive systems. Often, the first contact that such companies have with the major touring scene comes through these events,

where their gear is subcontracted by a major sound company (to be used for rear delay speaker tower stacks, for example).

With 1987's summer season hotter than ever, most concert sound companies have been forced to both expand their rental systems inventory to service existing clients, along with the increased new business, and to seek new lines of communications with smaller companies that have additional rental gear available during extremely busy times. Sophisticated, new-generation mixing consoles are in great demand, and many companies are choosing to lease from the manufacturer or to rent from regional companies instead of committing to the bulk purchase of a group of desks.

"It's a frantic scene right now," said the owner of a new 40-input console. "I've had calls from three major companies this week, all wanting to rent it from me at the same time."

While different companies strive to service their traditional accounts with existing resources, and simultaneously seek new touring contracts during such a busy season, the inevitable last-minute system swaps are taking place, with trucking companies being the ultimate winners as major systems crisscross the nation. Billy Idol moved from Audio Analysts to Electrotec Productions; Audio Analysts picked up Run-DMC. Bon Jovi moved from Tasco to Showco. The Moody Blues, a longtime Clair Bros. account, picked Showco this season; longtime Showco account David Bowie moved to Clair Bros.

Both equipment and experienced personnel are in great demand; correct decisions now about the matching of systems and crews with different touring accounts will have a major impact on the future of various sound companies and their business efforts.

The true winner, perhaps, during such an active touring season is the concert-sound industry itself. The improved cash flow and increasing number of live concert events leads to a climate that is ripe for research and development and the advancement of the state-of-the-art. Equipment manufacturers, sound company staffs, touring artists and, ultimately, audiences all stand to benefit as live concert sound gets better and better. The 1987 summer season has been a driving force that is largely responsible for some positive changes.

R/E/P

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

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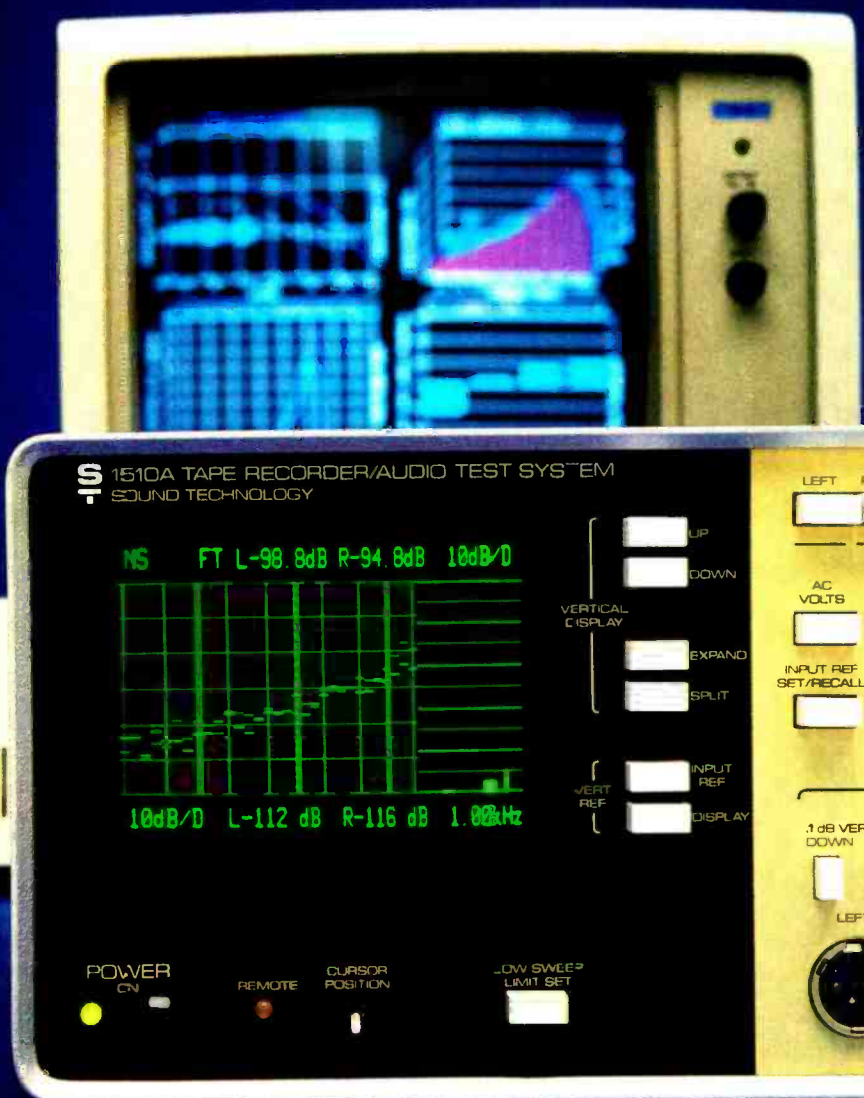
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Film Sound Today

By Larry Blake

Filmgoers in the late Seventies and early Eighties were treated to a string of 70mm sound and light spectacles. Of course, the period officially began with *Star Wars* and continued with *Close Encounters of the Third Kind*, *Days of Heaven*, *Superman*, *The Empire Strikes Back*, *Altered States*, *Raiders of the Lost Ark*, *Dragonslayer*, *Das Boot*, *Tron* and *Blade Runner*.

Your guess is as good as mine as to why the well of instant audio-visual classics dried up five years ago.

In the midst of this period, there was one film that, for my money, stands out above this estimable crowd. To this day, *Apocalypse Now* remains the benchmark by which I judge film soundtracks. Ironically, the story behind the creation contains reasons why the soundtrack should not have been good, much less a classic.

First of all, most of the *Apocalypse Now* mixing and editorial crew had little, if any, experience in stereo films. All post-production sound work (editing and mixing) took place at director Francis Coppola's Zoetrope facility in San Francisco and nobody on the team had been involved with the first two years of Dolby Stereo dubbing as practiced in Hollywood.

Furthermore, the equipment at hand was in many ways not up to the demands of a big stereo dub. The Zoetrope re-recording studio—built specifically to mix *Apocalypse*—was miniscule by Hollywood standards.

The room for equipment at the Pacific Avenue facility was also extremely cramped. Where your average Hollywood dubbing stage of the time had 15 or more dubbers and a large Quad-Eight film console, Zoetrope's machine room contained only seven 35mm dubbers and a pair of 6-track recorders. (Since the late Seventies, the ante of film dubbing stages has been upped to where 25 dubbers, three mag recorders and 70 console inputs are commonplace today.) The console was a 28-input MCI JH-500 that had to be hot-rodded for the unique monitoring and recording demands of film dubbing.

Luckily for us, however, the negative factors listed above were no match for the talents of personnel involved. The sound team for *Apocalypse Now* was led by Walter Murch, who was responsible for the inventive soundtracks of George Lucas' *THX-1138* and *American Graffiti*, in addition to Coppola's *The Conversation* and *Godfather* movies. His contribu-

tion to the soundtrack merited him poster credit on *Apocalypse*, the first time I know of this happening in a major U.S. film.

Working with Murch on the soundtrack was a crew of talented personnel, all of whom would later go on to make their mark in film sound: Richard Beggs, Mark Berger, Jay Boekelheide, Tom Scott, Leslie Shatz and Dale Struppell. The sound editorial team was led by Richard

To this day, Apocalypse Now remains the benchmark by which I judge film soundtracks.

Cirincione, who supervised a platoon of sound editors from the Bay Area, New York and London.

A large number of people were involved not because, as is often the rule with today's big-budget films, the picture had been handed to the sound crew at the last minute. Instead, this was one instance where the sound team was given the time to experiment; by the time the 7-month dub began in December 1978, Cirincione had been working on the film for more than 1½ years, while Murch had been working as one of the picture editors.

Music for *Apocalypse* was given no less attention, with music producer David Rubinson coordinating an army of musicians (mostly synthesists) hired to "realize" a scratch piano track written by Coppola and his father Carmine. By *Apocalypse* standards, this work was done in a short period of time, from January 1979 until the film's premiere in August 1979.

No less effort was spent in ensuring that the presentation was up to the care put into making it. Indeed, one of Coppola's original ideas was for *Apocalypse* to play at one theater in the middle of the country. Talk about making moviegoing an event! On a more practical level, the Northpoint theater in San Francisco was adopted as the benchmark in judging how mixes made in the small Zoetrope dubbing stage translated to the real world.

While the equipment limitations listed above were easily overcome—albeit with the addition of two extra generations, due to extensive pre-dubbing—mention has to be made of the technological perks the *Apocalypse* crew enjoyed. First of all, music was mixed to mag film on

the dubbing stage, against dialogue and effects pre-mixes.

In almost all standard stereo films, music is brought to the dub as a composite mix, with little consideration given to how everything will blend together. Beyond virtually precluding any opportunity for creative blending of music and sound effects, this attitude often finds the composer and sound editors in a tug of war officiated by the director or head mixer.

The film also marked probably the first extensive use of console automation in a stereo film dub. (Automation had previously been used in New York and Hollywood, but never caught on in the latter for stereo dubbing.)

All of these time, money, equipment and personnel issues are really beside the point, because it is the *sound* of the film, how it is used to support the drama, that sets apart *Apocalypse Now* from your average stereo dub.

While the battle scenes justly command a great deal of the attention given to the sound and spectacle of *Apocalypse*, my favorite sonic moments occur elsewhere in the film. First there is the scene early in the film in Martin Sheen's hotel room when he is remembering being back in the jungle. The intertwining of the crickets rising with the music is a thing of beauty.

Later on in the film, at Do Lung bridge, the distinction between music and effects is all but obliterated. Indeed, one of the more noteworthy aspects of the helicopter attack on the village is the weaving of Wagner's "Ride of the Valkyries" in and out of the helicopters and munitions.

Probably my single favorite sonic moment in the film is the helicopter turbine windup that leads us from the campfire to the field of Hueys. Not only does it function as music under Robert Duvall's famous "Charlie don't surf" comment, but it is an inspired use of the old trick of using sound effects to bridge scenes.

The other most frequently mentioned aspect of the *Apocalypse* mix, after the battle scenes, is the 6-track mix with stereo surrounds.

I'm sad to report that I can't foresee a film soundtrack of *Apocalypse Now*'s caliber coming along anytime soon. It's hard enough for the factors of time, money, equipment and talent to get together on any one film; adding a story that is rife with creative sound opportunities and a director who knows how to exploit them, well...it's like waiting for the next Lennon/McCartney team. **R·E·P**

Larry Blake is REIP's film sound consulting editor.

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Living with Technology

By Stephen St. Croix

Welcome to Stephen St. Croix's Official "What You Should and Should Not Buy Advice Column."

We all work pretty hard these days (purple control rooms alone don't seem to bring 'em in any more), and you probably don't have the time now to hang out at your local pro-audio dealer. I sure don't.

It is my job to know exactly what is going on out there, even if it is not going on yet. That used to be a pretty easy thing to do; I know the right people to keep in touch with the newest developments, and I do a fair bit of consulting and designing on today's new toys—I could generally call up a few guys and boxes would be at my door in days.

But things are changing. Today's new gear is falling out of the trucks and onto the street faster than most of us can even make it over to the pile of *yesterday's* new stuff. I find that if I don't stick my head out and check the developments in the technologies that apply to all of us, at least once a week now, a great deal can slip by me totally unnoticed.

OK, so I stuck my head out. In fact, I have just completed a total immersion course in T3 (Today's Techno Toys), and I have made some observations. (This statement comes, no doubt, as a total surprise to my regular readers.) I once told you that technology advances today at a rate that yields twice the power or half the price, every six months. These two advancements can (and are, of course) usually combined by the manufacturer into what he feels is the best marketing ratio.

In looking around at what is going on out there today, however, it seems to me that there are people who are not satisfied with even that rather impressive growth curve. They want all the power of the big gear—those 10 or 15 kilobuck processors—in a little Japanese spin-off, for \$23 and they want a 20% discount. (I *personally* want to win a newer Lamborghini Countach on the "Wheel Of Fortune," and I want Vanna to deliver it herself.)

So back to the people who want all the power for none of the money. Amazingly enough, they seem to get it. Almost. Instead of the technology growth trend advancing along the hoped-for direction—that is, a balance of more features/better performance/lower price—

several companies have figured out that if they offer the expected feature list, but at an unexpected price; enough customers will be wowed into completely forgetting about all that obscure performance type stuff. And real big money could be made.

For the first time in the history of man, technological advancement means more features, but of *less* quality. This is an interesting development. It allows a marketer to bring out a low-cost version of a competitor's mega-pro-processor very

***People want all the power
of the big bear
for \$23 . . . and
they want a 20% discount.***

fast, while the buzz generated by that top-end device is still hot. He does not have to wait until the current techno-advancement curve allows him to offer the same features and quality at a lower price. Instead, he simply copies (sort of) the features that everybody loves, makes the price irresistibly low, and cuts back the quality to make the whole venture profitable.

I am not saying that doing this does not take technological chops; in fact it does take some serious skill to develop the monster VLSIs used in these toys. The difference is that, in the interest of bringing a processor to market with that shocking price that sells, the development time went into drastically lowering manufacturing costs, assuming significant loss in the actual audio quality to be a necessary casualty.

The game is well played by the big boys. They drop the numbers on the most "popular" specs as little as possible, but they take real big chunks out of the others. There is no other way. The more obscure a spec, the more chance there is that it has been carved up pretty badly.

Have you ever wondered why the big-buck 16-bit processor and the little-buck 16-bit processor simply don't sound the same, even though a full page of specs almost match? Well, now you know.

However, all of this does make possible a new phenomenon. Little garage bands may now have pitch-shifted vocals, digital reverbs in their guitar amps, and gated reverb on their \$200 double kick drum kits. Amazing.

But there is a new situation that we,

the recoding engineers and producers, are faced with as a result of this unique interpretation of the potentials of advancement. The same bands are bringing these toys into the studio with them, and expecting us to make saleable, high quality albums with "the stuff that they are used to." Great.

It is sometimes hard enough to teach the talent why they must record with new guitar strings, or why the drums have to sound totally different in the studio than they did on stage. Now we are faced with the additional task of explaining why six tracks with effects from their new Furbisher 6000-SUX (that they just got so that they could do all the cool effects that they hear on records, live on stage for \$10.99), is going to produce enough mud for wrestling; which, by the way, is *too* much mud for rock and roll.

This is not to say that such low cost devices are garbage. Having played with all of them recently, I hereby admit that I have kept three. One extremely small digital reverb will be moving into my Mesa Boogie; one processor hangs onto the back of one of my sampling keyboards; and one medium/low-cost, American-made rackmount digital reverb lives in one of the synthesizer racks.

I use them daily when I arrange or score but, with the exception of the American unit, I do not print them.

I generally print all the synthetic material direct to the digital 2-track. All the synthesizers are SMPTE-TC locked to the multitrack; and computer played. All real (miced) stuff is, of course, simultaneously mixed down from the multitrack.

I have six of those \$10K+ processors, yet I *still* print effects on the multitrack and use the same effects machines again during mixing. I cannot afford therefore any compromise in audio quality of the effects that I use, not when the target is 95dB of 20kHz CD!

Now, as promised, for the official part: what to buy and what not to buy. Buy it all.

Just keep in mind that the Volkswagen is great for getting groceries, but it's the Ferrari that you take to the opera.

RE/P

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

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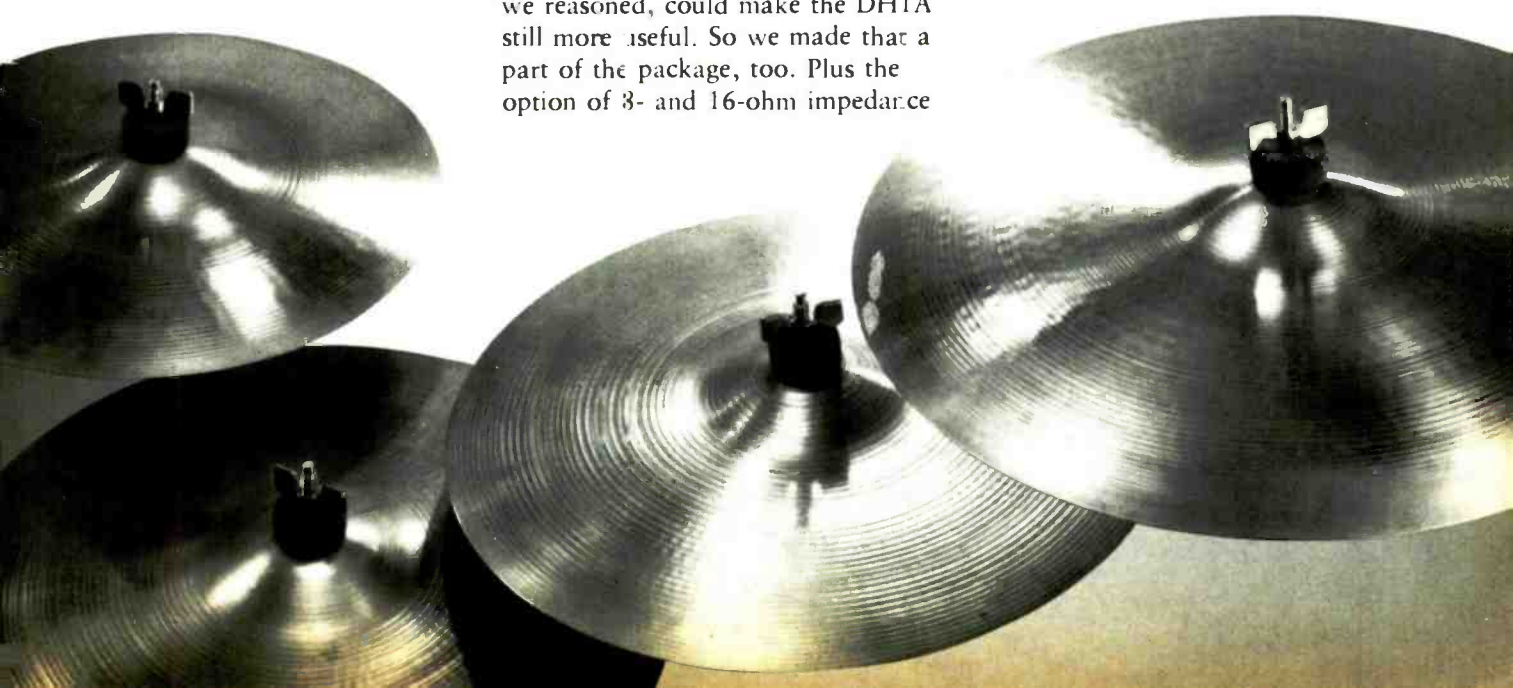
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SPARS On-Line

By Wilbur W. Caldwell

Someday, not far in the future, a very strange thing is going to happen to you. You will walk into a control room somewhere (maybe even your own) and you will discover that all of the equipment is missing.

The console is gone. The tape decks have mysteriously disappeared. Even your new multitrack digital is missing. There is no outboard gear to be seen anywhere. The room is, in fact, empty except for a couch and a few chairs, a large table and a CRT monitor with an alphanumeric keyboard placed between two speakers. The computer terminal will look small, impotent, insignificant and lonely in the vast room that used to burst at the seams with dials, lights, faders and knobs.

What's going on?

This is not a crazy hallucination. Nor is it a nightmare. Far from it. All over the world, people are working diligently to make this strange vision an everyday reality. The technology is increasingly within their grasp. The vision of this stark, empty control room calls up images of gillions of gigabytes of mainframe memory, archives of optical disks and CD WORM drives, and miles of streamed digital information: all in a box so powerful that it is combination console, multitrack recorder, mix automation, outboard signal processing and tape archive file. You might as well throw in billing and accounting for good measure.

The people who are at work developing these all-powerful boxes tell us simply that once sound is in the digital domain, the computer is the ideal tool for manipulating audio, for such processing becomes nothing more than a bunch of mathematical calculations. It's my guess that this is somewhat of a modest understatement. Effective, good sounding dynamics control may prove tricky, and the complexity and quantity of these calculations are mind-boggling, especially when you take into account the real-time requirements involved.

The standards for the coming generation of consoles and recorders are quite high. To put it simply, they must surpass existing analog technology in *every* category of performance: quality of sound;

Wilbur W. Caldwell is president of Doppler Studios, Atlanta, and is assistant to the president of the Society of Professional Recording Studios.

Editor's note: Following the recent resignation of Gary Helmers as executive director of SPARS, we have elected to have individual members of the society address controversial and interesting topics of relevance to the facility owner and operator.

ease and speed of operation; expanded capabilities; increased flexibility; and widespread standardization—to say nothing of cost. This is a tall order, but it must be taken seriously. (While I was musing on this topic of future shock, I asked one of our engineers what he thought about this stark vision of the console. He simply replied: "It will happen." Of course, he's right.)

How should facility owners and operators deal with such sweeping changes? You must first get things into perspective. When I was asked to write about this top-

A new generation is growing up in an age where a computer in the home is as familiar as a television.

ic, one of the first topics to be addressed was: "How important is the console?" Well, most of all, it is certainly never more important than the operator.

In fact, when you put any session in this perspective, the elements of prime importance become the artist, the producer and the engineer (not necessarily in that order, in all cases). These elements have one thing in common: They are *people*. A good engineer can make good recordings on marginal equipment; with extra effort and patience, however, a marginal engineer can never make first-rate recordings, even on the best console in the world.

This is not to say that the console, the recorder and a quality acoustic environment are not important—they are essential. All three elements are links in the chain, but they are merely tools to be used. That they are not creative in and of themselves may be obvious, but it's a key point; people drive the recording industry, not the equipment.

Such considerations provide us with insight on how to deal with the console of the future, and how these changes will develop. We are more in control of our own destinies than we are led to believe.

Because of the human factor in this period of change, we will experience evolution, not revolution. The effect of new digital devices is already strongly felt, and the new consoles have control surfaces that look rather familiar. This is a matter of demand, *not* technology. There is still strong resistance to a console that does not look like a console.

Such resistance can only be overcome slowly by successive generations of consoles, each with a control surface slightly smaller, more compact, more programmable and more foreign than the one before.

Let's forget that several years ago one of the premier console manufacturers introduced a sophisticated automation system with faders that actually moved as the mix transpired. This capability was not a whim, but the result of a careful study that projected what a change-resistant marketplace wanted. Also of note is the fact that several new-generation consoles with programmable or assignable designs have been introduced over the last year or so and, so far at least, we have seen no great rush to embrace these innovations.

The digital multitrack is another example that demonstrates a marketplace in evolution while technology is in revolution. Although such machines have been around for quite a while now, we are only beginning to feel the impact, and only in the high-end of the record business. Certainly, they are now beginning to come on fast, but we have yet to see any shift in the extremely high pricing. We are also a long way from standardization of formats, and thereby far from significant intrusion into commercial recording work.

So what does this all mean to the studio owner? It means that *people* are the key to dealing with future shock. The human is a very adaptable animal. Cross-training in computer sciences, as well as music, electronics and acoustics, will usher in the era of newer technology.

The future will indeed arrive, but not as soon as we might think. There is a new generation of artists, engineers and producers out there, and it is growing up in an age where a computer in the home is as familiar as a television. They will be quite comfortable with a light pen or a mouse. It will be their challenge to deal with the console of the future. It is our job to ensure an intelligent evolution.

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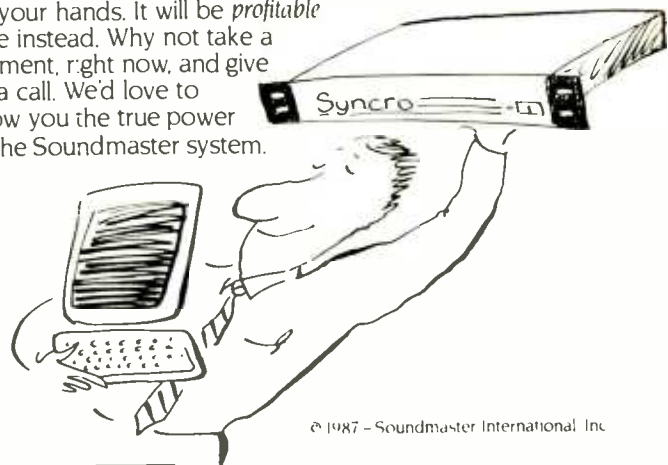
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Future Console Developments: The Impact of Digital Techniques

By Bill Aitken

Console designs are evolving toward all-digital creations in the coming decade, offering operational and sonic advantages and changing the way studios will operate.

As technology accelerates into the Nineties, staying ahead of the competition means that console designers need more financial resources, intellectual capacity and market knowledge than ever before. So, how do console designers know at the beginning of a long and expensive development program that they have got it right?

The ideal audio console has two main facets: highest quality sound and maximum creative potential. Judging how well a console lends itself to creative processes is a complex and interesting argument that we'll examine later. But first

Bill Aitken is with Solid State Logic, England. Over the past several years, he has also written, performed, engineered and produced a variety of free-lance projects for record release, commercials, radio and TV broadcast.

there is the question of audio quality.

On the face of it, audio performance can be measured with relative ease. With the advent of digital control and processing techniques, however, many issues are easily clouded by the slavish application of measuring methods and concepts developed for analog systems to their quite distinct digital successors. Signal-to-noise ratios and distortion measurements are the two most commonly discussed parameters. Although an analog system and a digital system may be quoted as having comparable performance in these respects, it's interesting to consider the implications of what is *not* explained in specifications.

Analog/digital specmanship

When an analog console is over-modulated, it tends to complain gradually and progressively. Recording engi-

neers know about this phenomenon and know just how far to push analog circuits to take full advantage of the available dynamic range. A digital system reacts far more unkindly to this sort of treatment—a growing number of engineers with experience with digital tape recording bear witness to this fact—with the result that, when recording digitally, engineers have to back off modulation levels to the point where they are absolutely sure that even random peaks will not go over the top, and conspicuously mar a precious recording.

The bottom line is that an engineer must leave a generous headroom margin inside the digital recorder to cope with occasional overshoots, which, in turn, means a more restricted dynamic range available for practical use than the number quoted on the specification.

Having backed off the modulation

level on an analog system, the next thing to worry about is noise. When used in tandem with a digital recording, however, there is an additional concern: increasing distortion at low levels. A full-modulation signal will use the full range of quantizing levels/bits available on an analog-to-digital converter; when the signal level is low, however, there are less quantizing levels/bits and the conversion becomes relatively crude. The result is a gritty, unmusical form of distortion, which, sadly, is becoming increasingly familiar to the critical listener.

These simple examples do not mean that digital systems are inherently inferior to analog; in fact, digital techniques will bring new standards of quality and freedom that are inconceivable in analog.

It does mean, however, that people who are designing digital systems must create products that will bring real benefits to the audio control room, rather than produce something that merely mimics or betters on paper the specification of an existing analog system.

Because they are designed primarily for less demanding military applications, currently available 16-bit A/D converters running at the sampling rates required for professional audio use are less than perfect. In practice, instantaneous errors occur in the process, and anything that results in missing codes or loss of linearity in the transfer characteristic will create audio problems, especially on transient material and low-level signals.

In order to develop a high-end digital mixing console, it's not enough for a digital hot-shot to tack a standard 16-bit A/D between the outside world and a computer and start making claims. Good design must start in the analog domain, identifying and eliminating sampling and conversion errors by optimizing the signal before A/D conversion, and by improving and inventing new conversion techniques.

Beyond 16 bits

At this point in the process, if you are simply interested in storing or recording your newly acquired, high-quality digital audio, then you enter the realm of the data storage manufacturer. But a mixing console is about audio processing—filtering, equalizing, compressing, limiting, expanding, mixing and the like. In order to achieve these tasks digitally and satisfy the exacting standards of the professional audio studio, there is a need for a dynamic range in excess of that available using 16-bit technology.

To illustrate the point, consider the problems of a digital equalizer. Audio data may have to be passed through a

multiplier a number of times to achieve a particular digital equalization effect. It doesn't take much imagination to see that if an audio signal represented by a number near the top of the 16-bit range gets multiplied over and over again, the result will soon run out of space inside the computer as the 16-bit limit is reached. In a 16-bit processor, any multiplications that should have produced a number greater than 16 bits will be rounded off to 16-bit.

This particular problem is called rounding error; as out-of-range audio data goes around and around the multiplier, the difference between the number intended and the number produced increases, and so the rounding error increases. The result can be a most unpleasant and unmusical distortion.

The answer is to use a processor with a higher number of bits. Higher bit widths require more processing power, however, and this costs money. But, as with sampling rates, a certain number of bits is absolutely necessary to achieve an audio performance that will meet the requirements of the discriminating listener. There is growing acknowledgment that the appropriate number for digital audio computers is 24-bit, which yields a total dynamic range of 144dB. A properly implemented 24-bit digital system architecture will yield superior results to any analog console and will accommodate all foreseeable developments in converter, storage and transmission technology.

So far, we have only scratched the surface of some of the more quantifiable problems that present themselves to the console designer. However, in order for any console—digital or analog—to be a success, the designer must fully understand the *creative* processes of the professional audio environment and be thoroughly committed to stretching technology to the limit in order to satisfy these needs.

Such matters are more difficult to quantify and often interact or interfere with the more straightforward questions of technical performance. The road to ruin is littered with devices designed by people who found it too easy to design what was technically expedient, at the expense of listening properly to the marketplace. But these are key issues, and the digital console designer will succeed only if he is forward looking, and provides the greatest possible freedom to explore the creative aspects of studio work, in addition to the highest possible audio processing quality.

Man/machine interface

Many of these freedoms will manifest themselves in the design of the man/machine interface. When the word "con-

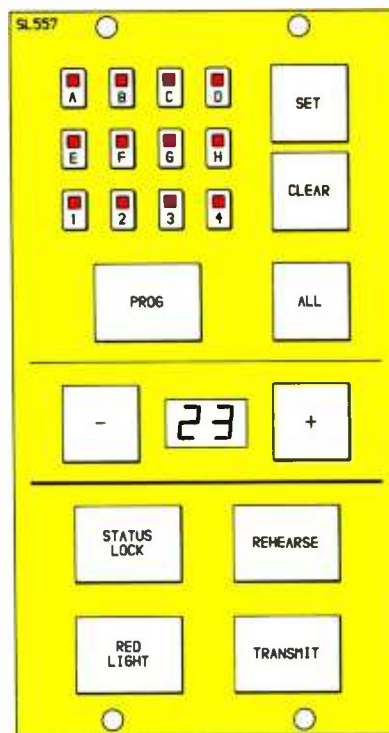


Figure 1. Centrally mounted in the SL5000-M console is a master assign module that features 12 assignable routing buttons. Control from the master module to each channel is selected either by pressing dedicated "set" buttons located close to each channel fader, or by using the "+" and "-" keys in the center of the master module.

sole" is used, most engineers and producers generally think of the front panel or control surface, with its array of knobs and switches, lights and displays. To those who are not inclined to think about what happens beneath the control surface, this is the console.

Not long ago, every knob and button on a console was mechanically linked to the circuitry located beneath the console and was a physical extension of the potentiometer or switch that the engineer controlled by turning the knob or pressing the button. Recent advances in technology have made it possible to physically separate the knobs and buttons from the circuit elements they control, so that they can be mounted relatively far apart and communicate electrically, rather than by more direct, mechanical techniques.

Such an arrangement offers great freedoms to the console designer. Remote control also lends itself to a very attractive parallel control operation. The engineer can manually operate the remote controls, while a computer can share connections to the wiring system and also adjust pots and switches according to the contents of its memory.

This latter capability is often referred

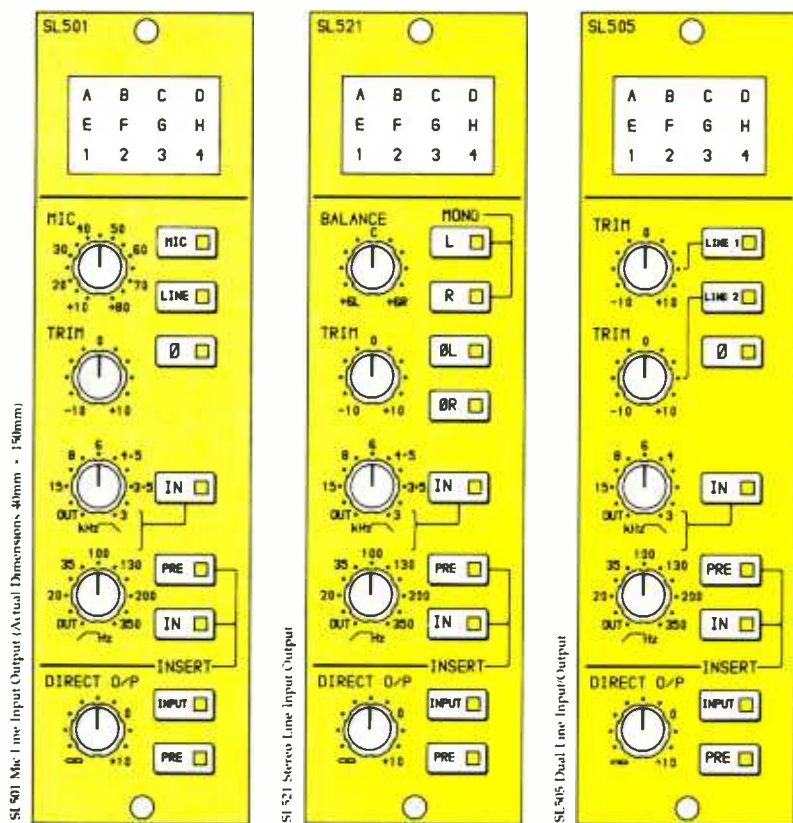


Figure 2. Assignment displays on each of the mic/line, stereo line and dual line-level input modules provide local indications of the routing set up on the master module (Figure 1) for instant verification of input/output assignments.

to as "programmability" and allows the engineer to choose whether to operate controls manually in a spontaneous manner, or to instantly recall a complex switch and control setup for the entire console from the computer memory.

Fully programmable analog consoles apply this philosophy to the extreme, and the attractions of a fully resettable console are obvious. But there is a heavy price to pay. In order to achieve complete programmability in a console employing analog audio processing, there are three broad choices: motorized potentiometers; VCAs (voltage controlled amplifiers); or MDACs (multiplying digital-to-analog converters).

A standard analog console may have 56-input channels with around 30 variable parameters per channel module. Little imagination is needed to project the cost and unwieldy nature of a console equipped with around 1,600 motorized pots or VCAs under computer control. Although such components have proved highly successful for simple level control in fader-automation systems, the scheme we are now projecting makes much higher demands on these components and the computer systems that service them.

For example, the control range stability of VCAs, and the repeat-accuracy tolerances of motorized pots, become much more significant problems when applied to the control of feedback around an amplifier, and consequently severely restrict their usefulness in such applications.

MDACs are suitable for selected programmable applications, but, in context with a fully programmable analog console, they also have their problems. Under certain conditions, switching and non-linearity within the device result in audible distortion at high frequencies. It is possible to minimize this distortion, but at the expense of increased noise. In other words, audio processors using MDACs tend to be noisy. Moreover, this noise is almost constant, regardless of the gain settings, and typically in the range of -7dB to -74dB (CCIR 468, quasi-peak weighted).

The effect that MDAC noise has on minimum noise (i.e. mic amps at unity, faders at infinity) can seriously compromise the usefulness of a console, particularly one with a large number of inputs. While this doesn't preclude the use of MDACs in some applications, it makes them less than ideal as the main building

block of a fully programmable analog audio processor.

Another disadvantage of the cumbersome circuit design that goes with the above techniques is that, in order to keep cost and reliability within a vestige of controllability, resolution of some parameters has to be severely limited. This means that, instead of continuously variable frequency-sweep and bandwidth controls (which have become standard on high-end analog consoles), digitally controlled analog consoles often offer only switched frequencies and bandwidth—and, more often than not, in a very limited way.

A properly designed digital console, which employs digital audio processing, is inherently fully programmable with none of the above problems. But that does not mean that well-engineered programmability must wait for the world of digits. If the functions to be made programmable are confined to switches (which feasibly can be remote controlled without sacrificing audio quality) rather than more problematic, continuously variable parameters (potentiometers), then a console may be designed in which all switched functions can be stored and reset by an external computer—while still maintaining high audio performance. Figures 1 and 2 illustrate one such example of this design philosophy.

In the example illustrated here, an engineer can switch each individual function spontaneously by operating the individual buttons on the control surface, or can command the computer to perform a global reset command, which resets all the switches on the console instantly to a preset pattern.

However, integrity of the audio performance is maintained, because the design does not employ motorized pots, VCAs or MDACs to remote control complex potentiometer functions.

Control assignability

Another by-product of programmability is assignability. Assignability is an extension of the remote-control philosophy, which argues that if a switch can be remote controlled by a button via a length of wire, then there is no reason why there has to be a *different* button for each switch to be remote controlled. Why not save money and space by having only one button and a means of routing the control signal from the single button from one wire to the next, thereby transferring remote control from one circuit to the next?

This design philosophy can also be extended to the lights or displays that inevitably accompany and "echo back" the current state of a pot or switch so that

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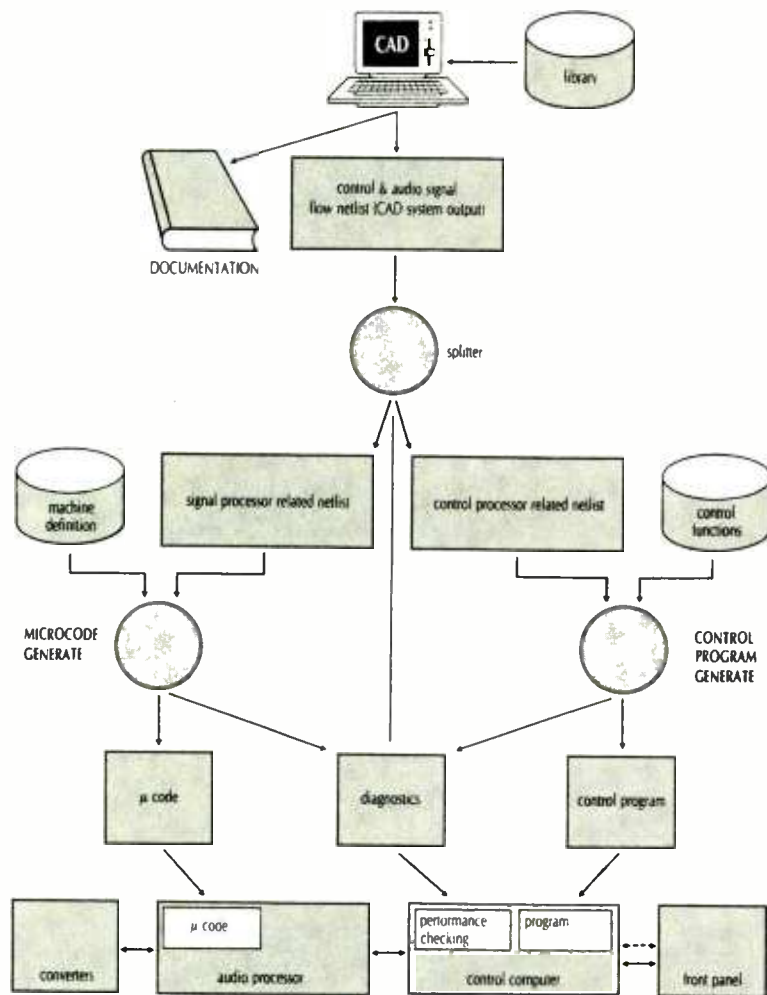


Figure 3. Block diagram of a typical design process to develop and evaluate the basic building blocks and operational parameters of an all-digital console.

the engineer can see the state of a particular circuit when he or she selects the corresponding control line. Extending this argument, it is possible to design a console where thousands of switch functions are controlled sequentially via a single button and thousands of switch states are displayed sequentially via one single light. There are frequent examples of products that attempt to take this philosophy to the extreme: Sometimes the results are elegant; more often, the results are frustrating for the user.

The advantage of assignability is that control panels become smaller and more economical. The disadvantage of assignability is that, by making visual feedback a sequential accessing routine through the use of shared lights and displays, the engineer is robbed of much vital, immediately available information. A 15-foot console may create parallax errors, and not all of the information is in front of the engineer's eyes all of the

time. (How often have you wondered what signal source was causing distortion during rehearsal, only to scan down the input pots and immediately find the one that was in the wrong position?)

Now ask yourself how many manual operations would be involved in interrogating a completely assignable, 56-input board to determine the same data? And, if the input level for each channel was displayed in digital form (i.e. a numerical readout instead of a bargraph), how long would it take you to make sense of the information and react?

The degree to which a console designer implements assignability is, like most things, a matter of balance. A successful designer selects the correct trade-offs for all design parameters, including which controls should be assignable and which should have dedicated controls and displays. Once again, the fully digital system comes out on top, offering the designer a greater chance of getting it

right because of the new freedoms that present themselves.

Responding to market demands

Every manufacturer wants its products to be as immune to obsolescence as possible, thus ensuring long-term commercial success. Digital audio products are no exception, and any investment or commitment in the pursuit of these aims benefits not just the console maker, but also the end user.

SSL's own answer to this challenge has resulted in the development of a digital audio design system* that creates maximum flexibility for the designer and, consequently, will shorten product development lead times, allowing new products to appear more quickly in response to market demands.

This degree of adaptability is achieved by designing tools and aids which, broadly speaking, make it possible to develop three main elements of a digital product completely independently: product specification (including front panel/control surface layout); parameter response; and processor hardware.

As can be seen in Figure 3, SSL's computer-aided design system features pictorial representations of the various elements of a mixing console—such as filters, faders, etc.—that can be called up quickly to the VDU. These visual elements can be taken from the library, displayed on the screen, edited and assembled into something that looks similar to a circuit diagram.

Figure 4 represents a console schematic, and embodies the specification units as conceived by the designer. At this stage in the design process, it is only the console architecture that is being defined; how many channels, what kind of EQ, how many aux/cue sends, pre/post switching details, etc.

It is crucial to appreciate that making such a detailed definition at this stage does not lock the product into any particular processor/hardware system, or control surface/front-panel layout. How and when that happens will be explained shortly. In the meantime, having specified what the designer wants the mixing console to do, the CAD system will produce a corresponding drawing, as shown in Figures 5, 6 and 7, to allow checking of the design and planned or implemented modifications.

***Editorial note:** The basic design parameters and operational features of this design system formed the subject of three technical papers presented at the 81st Audio Engineering Society Convention, Los Angeles, November 1986, by members of the SSL digital design team, including Peter Eastty, William Kentish, David Bell and Chris McCulloch.

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SPECIFICATIONS: CARVER PM-175 Power: 8 ohms, 175 w/channel 20-20kHz both channels driven with no more than 0.5% THD 4 ohms, 250 w/channel 20-20kHz both channels driven with no more than 0.5% THD 2 ohms 300 w/channel 20-20kHz both channels driven with no more than 0.5% THD Bridging: 500 watts into 8 ohms, 400 watts into 16 ohms THD-less than 0.5% at any power level from 20 mW to clipping IM Distortion less than 0.1% SMPTE Frequency Bandwidth: 5Hz-80kHz Gain: 29 dB Input Sensitivity: 1.5 V rms Damping: 200 at 1kHz Slew rate: 25V/micro second. Noise: Better than 115 dB below 175 watts. A-weighted Inputs: Balanced to ground, XLR or TRS phone jacks. Input Impedance: 15k ohm each leg. Compatible with 25V and 70V systems. 19"Wx3.5"Hx11.56"D

SPECIFICATIONS: CARVER PM-350 Power: 8 ohms, 350 w/channel 20-20kHz both channels driven with no more than 0.5% THD 4 ohms, 450 w/channel 20-20kHz both channels driven with no more than 0.5% THD 2 ohms 450 w/channel 20-20kHz both channels driven with no more than 0.5% THD Bridging: 900 watts into 8 ohms, 750 watts into 16 ohms THD-less than 0.5% at any power level from 20 mW to clipping. IM Distortion less than 0.1% SMPTE Frequency Bandwidth: 5Hz-80kHz Gain: 31 dB Input Sensitivity: 1.5 V rms Damping: 200 at 1kHz Slew rate: 25V/micro second. Noise: Better than 115 dB below 350 watts. A-weighted Inputs: Balanced to ground, XLR or TRS phone jacks. Input Impedance: 15k ohm each leg. Compatible with 25V and 70V systems. 19"Wx3.5"Hx11.56"D



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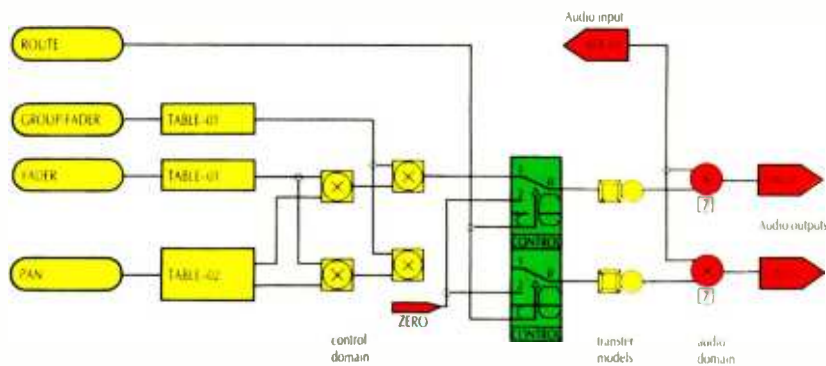


Figure 4. Basic control elements and software interrelationships of a console schematic with control domain emphasis.

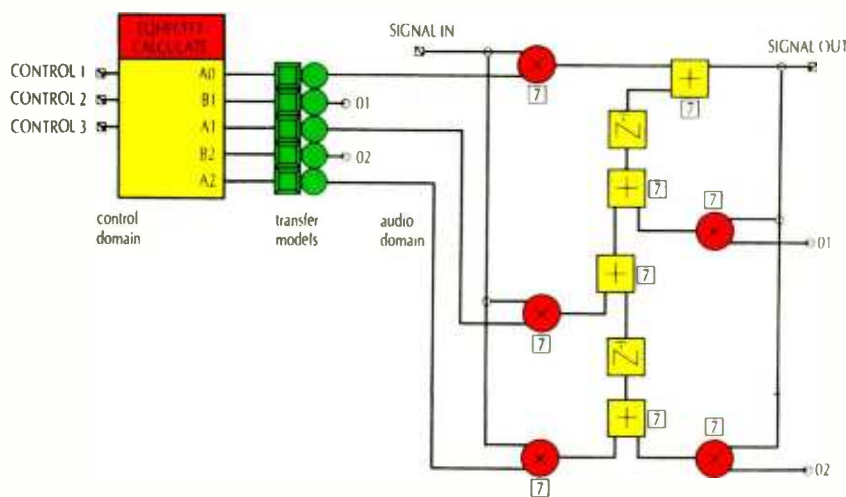


Figure 5. Computer-modelled schematic of a console equalizer section.

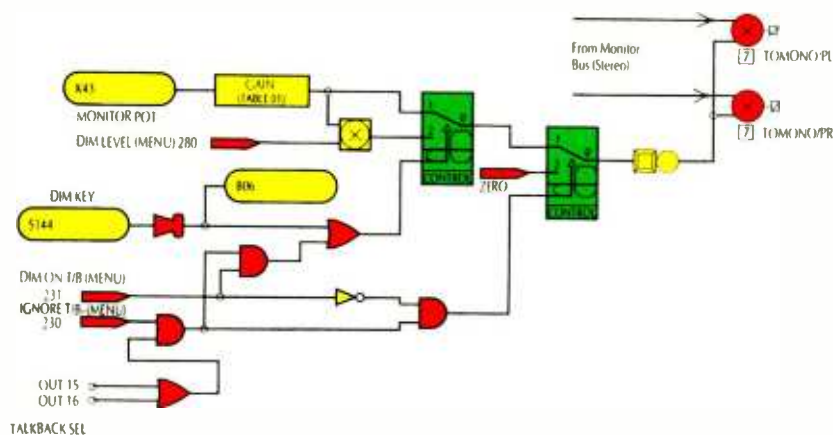


Figure 6. Computer-modelled schematic of a monitor output-gain section.

Audio and control data

So far, the designer has been working with pictorial symbols that represent concepts such as a "fader" or "equalizer." Although this task can be considered a specialized, high-level computer language for console designers, such conceptual blocks contain processing and control elements that have to be analyzed, split up and processed separately. By way of analogy, consider programmable pots where the control element (the knob), is completely separate and independent of the processor (the VCA or MDAC it is controlling). As a result, the console specification data output by the CAD machine is now passed through a program to split audio processor information from control information.

Having split up the two types of data, the designer can now choose the hardware/processor system on which to run the digital audio processing software, and how to lay out the control surface or front panel. First of all, let's consider the audio processing. The block marked "machine definition" is where the designer lets the system know what kind of processor/hardware scheme he intends to use for the audio processing system, and it forms one part of the input data to the microcode generate program; the other part is the processor data split from the CAD output.

The microcode generate program, in essence, says to itself, "The designer has described to me the audio processing he wants [signal processor related netlist output from splitter] and has described the hardware/processor system he intends to use [machine definition], so I shall now generate the microcode to achieve the audio processing required on the audio processor/hardware system defined."

A similar process is pursued to produce a control program to run on the master computer, only this time it's not a hardware/processor machine system that needs defining but the response of the various parameters, such as EQ curves and center frequencies, filter slopes, dynamics attack and decay ranges, etc. This part of the system also tailors the law of each parameter, so that the designer can modify the "feel" of each control in software. The data inputs to the control program generator are the control "parameters" definition, and the control processor related netlist output from splitter.

Fighting off obsolescence

First of all, it is common for product development to involve such a long lead time that, by the time the product is ready for market, the hardware/processor system it uses has been superseded by something better and/or faster. In the

above scheme of things, however, all the designer has to do is to input a fresh machine definition, together with the existing signal processor netlist, in order to generate new microcode suitable for the new machine. (All of which is a far better proposition than spending additional man-years rewriting software for the new and better processor.)

Secondly, there is the matter of product application. Broadcasters, for example, often have to squeeze a lot of equipment into a limited amount of space; mobile broadcast trucks are a prime example. But, if broadcast engineers want higher levels of assignability than are acceptable in a music-recording studio, again the design scheme outlined above is flexible and efficient. In this case, there will be one front-panel/control surface definition of the physically larger music studio console using low levels of assignability, and another for the more assignable, more compact broadcast console. Yet both versions will be based on the same hardware and processor system, having equivalent performance and audio processing power.

Finally, there is the matter of overall technical specification. Even if the user wants to modify something as fundamen-

tal as this, they don't necessarily have to throw away the hardware/processor system or control surface it took so long to develop. Provided that the intended update is conceived within the limits of the processor power and front-panel controls available in the hardware already designed, then "back to the drawing board" becomes "back to the CAD." Having split the data output from the CAD system and generated new netlists, the designer can immediately run the new software on the existing hardware/processor and operate it via the existing control surface.

Although the process might not be quite as simple as that, the advantages of this design approach are undeniable. Product development still takes time, however, and even though we are excited about the future, the industry must make responsibly timed product announcements, so that market expectations will be realistic.

Toward the future

Looking to the future, the professional audio industry should not expect to see a flood of mega-sized digital consoles descending on the market tomorrow. To this observer, the interesting question is:

"How will working practices in the studio change in response to the new methods and techniques made possible by new technology?"

A number of devices already attracting pre-release publicity claim to combine fast, random-access memory of considerable capacity with impressive digital audio processing. Perhaps someone will get this package right? In which case, how will digital tape machines be viewed?

Given the current problems related to varying standards, and the basic limitations of 16-bit digital sound, perhaps there will be a move to selectively combine traditional analog recording techniques (which continue to be very effective for particular tasks) with more adventurous random-access digital recording and editing systems? Various computer interfaces make it easy to achieve such tasks and apply apposite technological solutions in a more cost-effective and creative way.

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Console Topographies:

Live-Performance Vs. Recording and Production Designs

By Douglas Self

What are the key operational and electronic differences between consoles designed for live-sound mixing and those intended for a recording facility? How divergent are their respective technologies?



A great convergence in console design is often said to be taking place these days; it is clear that the technology of live sound is coming closer to that of the recording studio, as the movement toward "doing a show as good as the record" gathers momentum. In some ways, live-audio technology has had to run very hard to stay in the same place because the call is now more likely to be: "Make it sound as good as the Compact Disc."

Standards in all aspects of the audio industry are quite rightly becoming higher and higher, and many companies that have their roots in the design of consoles for live-performance sound are determined that their equipment stay at the forefront of technology.

This article will consider some of the differences—both obvious and subtle—

between live/theater and recording/production consoles, with a view to demonstrating where this technological convergence has occurred, and where it has not.

The first difference is that live consoles come in two distinct species: the main front-of-house console and the (usually smaller) stage monitor board. The purpose of the FOH console is to perform what a recording engineer might describe slightly apprehensively as a real-time mixdown to stereo for the main PA system, with no chance for second thoughts or corrections.

Parallelism is the name of the game here, with all signals being dealt with at once; EQ and outboard-processor settings have been decided long in advance, apart from variations to allow for different auditorium acoustics. The possibilities for creativity and experiment in a live show are, to put it mildly, somewhat limited.

If the FOH console provides sound for the paying customers, then the monitor console provides sound for the on-stage performers, allowing them to hear precisely what they're playing. The resulting monitor or foldback mix may be quite different, though no less complex, from that sent to the PA stacks. In fact, it is not unusual for up to eight or more different monitor mixes to be provided, thereby satisfying the musician's need to hear the overall sound with his or her own contribution dominant.

Although a dedicated monitor mix of sorts can be produced via the auxiliary sends of a FOH console, in practice these are fully occupied as effects sends. Also, for ease of communication, it's useful to have the monitor console located in a physically separate position at the side of the stage. Praise, blame, or desperate pleas for "more guitar on monitors!" can be distributed at once.

Grouping differences

There are subtle but important differences in the architecture of live and recording consoles, particularly with respect to the groups. Figures 1 and 2 show a FOH live console and a split recording console; the in-line format is confined to recording, being based on the needs of tape monitoring. Auxiliary sends and effects returns are not shown, because these are substantially the same for the two types.

A major difference is that tape-replay monitoring is not required on a live console. If the show is being recorded to multitrack with a view to future live albums, the signals are usually taken off

Douglas Self is chief analog design engineer at Soundcraft Electronics Ltd., England.

Figure 1

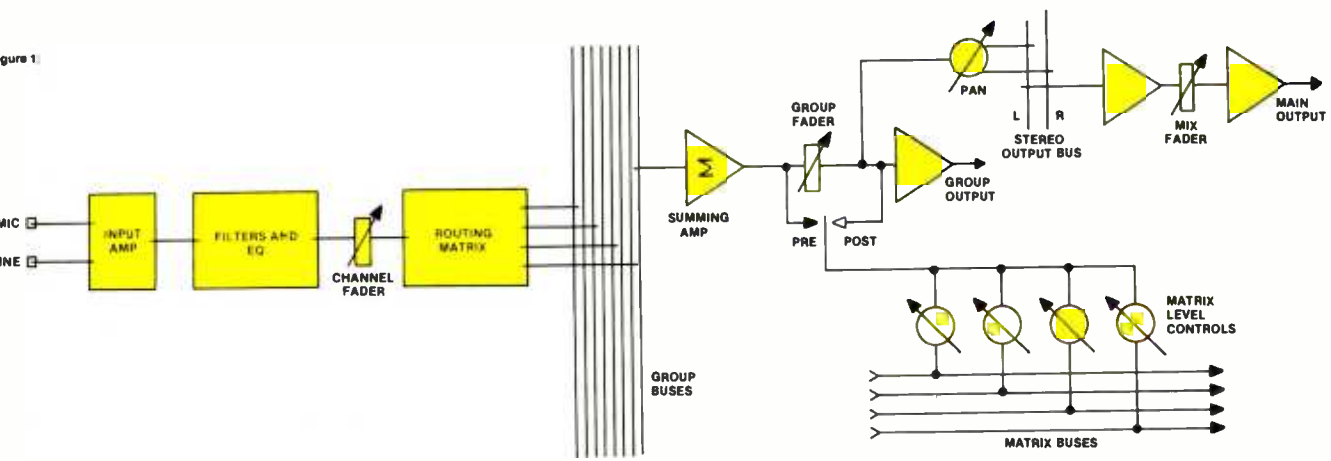


Figure 1. A simplified block diagram of a live-performance console.

prefade from the channel-line outputs. After initial level checking, these outputs are pretty much alone, with little time for critical off-tape monitoring during a performance.

A second architectural difference is that many FOH consoles provide matrix outputs; in other words, as well as mixing the groups to stereo for the main stacks located at either side of the stage, it is also possible to create eight or more extra mono mixes that can be distributed to other banks of speakers in different parts of the auditorium. Because eight or more extra faders per group module is clearly impractical, small rotary controls are used instead.

The detailed differences between live and recording desks start here; with two desks to be driven by the same mic and line inputs, some form of splitting arrangement is clearly necessary. Splitting line-level signals from synthesizers, for example, is simple, because it is at high level and low impedance. Inputs to the two desks are simply paralleled. Microphone signals are rather more delicate, however; although parallel connection is, in general, entirely acceptable, some authorities hold that transformer splitting is desirable.

One rather expensive advantage of this transformer isolation technique is that it can be configured so that a short circuit on one output (located, let's say, in the cable to the monitor desk) leaves the other output to the house desk unaffected.

The placement and use of EQ is rather

different for live consoles. During the tracking stages in a studio, EQ and effects are used with caution to avoid committing a sound to tape that cannot be reversed later. During later mixdown, each channel handles one tape track and sophisticated and flexible EQ is desirable because there is time for a relatively relaxed search for the desired result.

In live work, the intelligent use of a number of noise gates can make a contribution to the sound quality.

During live performances, however, a simpler EQ section may be preferable, because it must fight for space with auxiliary sends and other controls. Also, the need for fast reaction to unforeseen problems, such as acoustic feedback, makes an uncluttered layout highly desirable. If the cost structure of a console restricts it to offering just a few fully parametric sections, these are usually inserted within the group path. In contrast, recording consoles rarely offer EQ built into the groups, although on some desks it can be transferred there from an effects return that shares the same physical module.

Similarly, monitor consoles often feature parametric EQ on the outputs, so that the signals can be fine-tuned to prevent feedback.

Ergonomics

There are other ergonomic factors. Control knobs should always be color-coded, but this is even more important

for live desks operating in a relatively dark environment. Some companies have standardized on color coding—red for gain, etc.—a situation that has helped to make life easier for hard-pressed engineers.

Another possible aid is illuminating pushbuttons. Unfortunately, these devices add significantly to the cost of a console and tend to appear only on higher-priced models. The simple placement of an LED beside the switch is an effective compromise, however. Some designs include a few illuminating buttons for important functions, such as talkback control; it's a nice touch when they also glow dimly when inactive, because it allows instant location with the bonus of extending lamp life.

There are a few live-console facilities that never appear on recording desks. One of these is the "activity" indicator that typically glows when the channel is handling a signal at around the nominal level. A useful extra on monitor consoles is a "-6dB" button, which can be hit very quickly at the onset of acoustic feedback; this is often sufficient to kill the problem without seriously disrupting the monitor mix.

In live work, the intelligent use of a number of noise gates can make a major contribution to the sound quality. In contrast, relatively few gates are required when recording, because each track can usually be dealt with separately. While operating a large live rig, however, connections to and from a bank of external gates quickly become cumbersome and there is a clear need for at least basic noise-gate facilities to be built into live consoles. Although engineers often have

FIGURE 2:

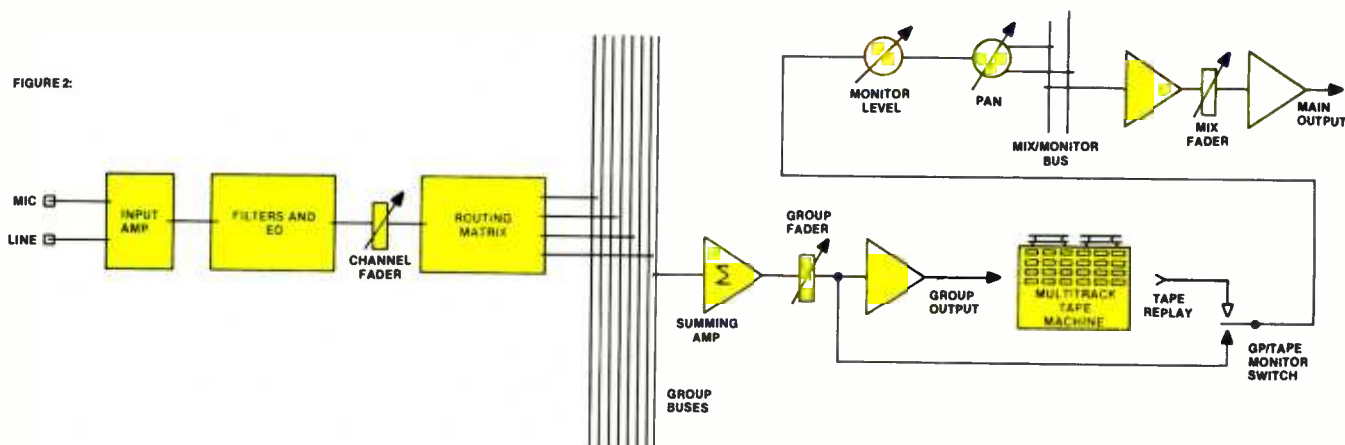


Figure 2. A simplified block diagram of a console designed for recording and production studio applications. It should be noted that the design illustrated here is a "split" configuration, in which the group output and monitor sections are located in separate modules to one side of the bank of input channel modules. The same basic signal flow is found in an "in-line" design, although the physical location of the group summing amps, monitor controls, etc. would be different from that of a split design.

their own favorite models of compressors or limiters, and might well object to being locked into a particular implementation, nonetheless a noise-gate is a simpler proposition. It should be possible to produce a basic design that is acceptable to everyone.

One feature found on most upper-range recording desks, but which is rare in a live situation, is destructive solo-in-place. Because this type of soloing achieves its purpose by muting all channels except those selected, its accidental use during a live performance might prove hard to forgive—to say the least. Nonetheless, SIP can be very useful when setting up a sound system.

One possible solution is to use recessed pushbuttons that are safe from operation by the casual elbow.

A good PFL (and preferably also a stereo AFL system) is essential for quality checks at show time, although meticulous electronic design is required to ensure that not even a trace of a switching transient couples through to the audio outputs.

Demands for increasing complexity and sophistication in making music have resulted in a steady increase in the number of auxiliary sends, effects

returns and insert points provided on consoles of all types. This increase in complexity applies equally to live and recording formats, because recreating a given sound often means that the same outboard processors and signal path must be used in live performance as in the studio. It is now common for large live-performance consoles to offer an integral patchbay that is as comprehensive as that of a recording version.

Everyone connected with our industry has stories of expensive consoles that have been dropped, run over or banged violently into walls.

Hostile environment on the road

The live environment is undoubtedly more hostile than that of the studio. As a result, live consoles must be built to cope both electrically and mechanically. The electrical environment, complete with powerful dimmer packs, grounds of uncertain provenance and often fluctuating supply voltages, tests a console's circuitry to the full.

Consistently high common-mode rejection on all balanced inputs is essential to reject induced noise. The design of an electronically balanced mic pre-amplifier that maintains a good CMRR across the whole audio band, and over a 70dB gain range, is a most challenging problem. Transformer-coupled inputs are unpopular for use on the road, not only because they tend to show LF non-linearity and a doubtful transient

response, but also because their contribution to the weight of a console is considerable.

When ground loops occur, a high CMRR is only half of the story. It is also essential that the input should be able to absorb the ac current flowing in the loop without inducing a buzz into the internal circuitry.

The mechanical aspects of live performance confront a console with stresses rarely encountered in the studio, including regular and sometimes non-gentle movement. Practically everyone connected with our industry has a story of an expensive console that has been dropped, run over or banged violently into every possible wall from parking lot to auditorium. While soundly constructed flight cases absorb the initial impact, it is also vital that a live console is constructed to survive every kind of shock, vibration and torsion.

Rigid aluminum extrusions to minimize weight can be connected with substantial machined end plates to form a structure highly resistant to torsion. It is often appropriate to use flexible IDC cabling rather than rigid motherboards, so that any small deformation that does occur is absorbed without connectors being forced from their housings.

Automated functions

Traditionally, fader automation as found in recording studios has had no place on the road. Not only has the associated computer equipment been bulky and delicate, but a live performance is clearly not suited to replaying fader movements that are rigidly locked to a time reference. VCA subgrouping

systems have been the answer to a lack of synchronized hands. As such, they are usually simple to set up and add negligible extra weight to the equipment.

As live shows become increasingly complex, however, and each audio event has to be right the first time, some form of console status recall or "scene setting" system is becoming necessary to lighten the engineer's workload. One or two recording consoles are available with recall systems that allow the operator to match control positions to those displayed on a VDU; other models are capable of automatically recalling a complete desk setup. This latter capability is extremely useful in a studio, where incoming clients can have their previous mixdown situation recalled and lockouts to preserve desk settings can become a thing of the past. Recall is potentially even more useful in live performance, however, allowing rapid switches of routing and configuration for different effects on different songs, and giving complete freedom to have quite different setups for different performers.

It should be pointed out that there have been some justifiable doubts about subjecting relatively delicate equipment such as VDCs and hard-disk drives to life

on the road; the inability to download data into a synth can be worked around, but a dead console is a major disaster. For this reason some users have been reluctant to commit to such high technology until development has ensured it is sufficiently robust.

Solutions to these problems include the retention of data in battery-backed-up RAM, and using watchdog systems that instantly restart a computer should a momentary power failure cause it to stumble.

Many console manufacturers, including ourselves, are developing a console of this type that provides instantaneous recall of every setting from fader position to mic input gain and phantom power switching. Unlike some design proposals, however, our intention is that parameters such as EQ sweep frequency will be continuously variable in real time, with no need to step up and down via push-buttons. The extensive use of surface-mounting technology in our proposed design offers, we feel, a compact and movable system, composed of a light-weight control console linked to a remote rack—no larger than a bank of power amps—that performs all the audio processing. A high-speed data link be-

tween the control console and remote rack is immune to electrical interference, and allows the audio rack to be positioned at the stage end, eliminating expensive and cumbersome audio snakes.

Technology convergence

It seems clear that there is a distinct technological convergence between live and recording consoles, the architectural differences between the two being less important than the need for comprehensive control and impeccable audio performance in both situations. It is also true that design solutions evolved to deal with the hazards of live operation can often be applied to recording consoles with superb results; for example, the use of rigid extrusions in console frames are also used for recording desks, though the latter may rarely meet with rough handling. The tooling costs of purpose-designed extrusions are such that this is a significant economic consideration; manufacturing costs can be reduced, and therefore prices will be reduced as well.

R&D

I would like to acknowledge that the preparation of this article has benefited from discussions with Graham Blythe, Dave Pallant and Dave Wade-Evans, of the Soundcraft R&D department—/N

A Judgement Rendered on LENCO AMPS

"Performance specifications alone won't reveal all one needs to know in determining the merits or the deficiencies of a power amplifier; but when we found one that had a slew rate of 300 V μ S (8 Ohms), a damping factor that hits the 1000 mark - all the way up to 5 KHz, and a THD of .003%, it came as no surprise to us to find this same level of superb quality in every other department we evaluated on the LENCO MPA POWER AMPLIFIER."

"Its greatest attribute was discovered at the listening tests. This amp brought a dimension in realism, depth, and accuracy to our monitor system none of us had heard before. From the lowest bottom end, through the midrange, and up through the highs, a new clarity and 'openness' revealed every detail of the recorded material. This allowed making evaluations on EQ and dynamics control much easier!"

"I have chosen the MPA-2300 model LENCO AMP for my personal use. It has become my trusted companion in every recording session I engineer. I can trust it to reveal everything I need to know and hear at the console!"

Wiley M. Ross, Recording Engineering
School of Music,
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Making the Lease/Purchase Decision

By Murray R. Allen

There are a myriad of questions you must consider when deciding to purchase a piece of equipment or lease it from a finance company or bank.

When buying recording or production equipment, there are several methods you can use to pay for it. Of course, you could reach deeply into your own pocket and pay with cash; for the remainder of this article, however, I will not discuss this costly, and probably inappropriate, method.

The next method is to borrow the necessary funds from a bank, a *bank loan*. An alternative to this method is borrowing money from a finance company, a *capital lease*. The final method is to simply rent the equipment, a *true lease*. Deciding which of these methods to use is complicated and depends on many variables.

Residual value

The first item of concern is the residual value of the equipment when totally paid off. Musical instruments, such as acoustic pianos and organs, often will be worth *more* at the end of the payoff than when first acquired. Our Bosendorfer Imperial Grand, for example, was purchased eight years ago for \$36,000; it has recently been appraised for \$82,000. Other instruments and electronic keyboards may have little value in five years, simply because they have been replaced with new-

er models with additional features.

High-end consoles will have value after many years, while some lesser quality boards will have practically no resale value. Analog 24-track recorders will be around for a long time and, depending upon their condition, will always be worth something. Digital multitracks will also maintain their value to studios as a work tool. Resale value will depend on what new machines come on-line and at what price.

As a general rule, Universal Recording buys through bank loans or capital leases any piece of equipment we feel will be actively in use at the end of the lease period, and will have at least a *40% resale value*. All other equipment is purchased using true leases. (Peripheral processing gear is the exception: we always paid for such hardware with cash. I have seen this type of equipment go in and out of favor during an 18-month period. Nothing hurts worse than to have to make monthly payments on equipment that is already obsolete!)

The next question is: How does a studio owner anticipate what equipment will be worth in three to five years? One way is to build up good relations with equipment dealers around the country, which is easy to do because every studio could be potential clients for such companies. Ask what they think the item or soon-to-be-acquired piece of equipment will be worth down the road. Take the lowest estimate, cut it by 10% and you

will probably be as close as possible to reality.

If the result is a very low or no residual value, a true lease might be the way to go. If the residual value is somewhere around 30%, other factors may color your decision.

Cash dispersal during payment

Another factor that is extremely important is to calculate the total cash paid out over the payment period. (You will discover large differences between the total cash paid out to different institutions with what, on the surface, look to be similar plans.) Naturally, we will almost always go with the plan that has a *lower* cash dispersal over the payment period.

Sometimes this sum can be difficult to compute, because you must take in the consequence of taxes on your cash dispersal. If you are philosophical and realize that tax shelters are only tax deferrals, and that long range you will have to pay the taxes anyway, you may omit the effect of taxes on your cash dispersal. But if you do omit the tax considerations, you will be making a major mistake that cannot be corrected down the line. In other words, *don't* be philosophical.

Collateral

Let's assume that you are going with a bank loan. Banks like collateral, which may take the form of other assets and personal guarantees, the most popular

Murray R. Allen is president of Universal Recording Company, Chicago, a multiroom facility that specializes in a wide range of audio production, including record sessions, electronic music, video sweetening, commercials, jingles and scoring.

This is the mixing console that will cause a evolution in 24 track studios.

The world's first Dual Mode mixing console.

With the advanced features and performance of desk you'd expect to cost a great deal more than it actually does, it's a remarkable achievement. Take Dual Mode.

When you record with the TS12, its in-line monitoring means you create a control room mix on the main faders.

In mixdown this becomes the basis for your master mix. Saving time and helping you keep a train of thought as to how the final mix will sound.

Unlike other in-line desks, though, the TS12 also allows true stereo subgrouping on long-throw group faders.

In this mode the routing matrix offers either six stereo groups or four extra auxiliary sends – totalling ten sends – plus four stereo groups.

No other console in the world provides such versatility.

The TS12 is an open-ended investment, with optional disc-based SMPTE automation for faders, mutes, EQ in/out and auxiliary on/off. Again, at the price it's unique.

And fittingly, audio performance is superb.

Recent Soundcraft advances in summing amp technology, and in grounding and decoupling systems, make the TS12 one of the cleanest and quietest consoles ever.

The mic amplifiers, a critical factor in console quality, create less than 0.01% distortion at 10kHz at 70dB of gain. (Easily exceeding 16-bit digital specifications.)

Standard features are impressive, to say the least. Six auxiliary sends, seven stereo line inputs or effects returns, a 'musician friendly' headphone mix, an extensive 19" metal frame patchbay – and the option to create a massive total of 102 inputs.

Quite a line-up.

Attention to detail is equally stringent with modular PCBs, no dual concentrics and a clear, logical layout that belies the sophistication inside.

But the most remarkable feature of the new TS12 is without doubt the price.

We suggest you call us today to find out just how remarkable.

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Soundcraft
TS12

MAX 102 INPUTS WITH 36 CH FRAME • SUMMING AMP NOISE -125dBv EIN (200Hz) • MIC AMP -128dBv EIN (200Hz) • MIC AMP DISTORTION <0.01% THD AT 10kHz AT 70dB GAIN • -85dBv AUX SEND KILL AT ALL FREQUENCIES • CROSSTALK TYPICALLY -70dB AT 10kHz (-85dB 1kHz) • MAXIMUM 2560 • OUTPUT

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Figure 1.

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	MONTHS	MONTHS	MONTHS	MONTHS	MONTHS	MONTHS	MONTHS	MONTHS	MONTHS	MONTHS	MONTHS	MONTHS
0	47.08	38.75	33.22	29.28	26.34	24.06	22.25	19.56	17.66	16.26	15.19	14.35
1	46.61	38.37	32.89	28.99	26.08	23.82	22.03	19.36	17.48	16.10	15.04	14.21
2	46.17	38.00	32.58	28.71	25.83	23.59	21.82	19.17	17.31	15.94	14.89	14.07
3	45.76	37.66	32.27	28.44	25.58	23.37	21.61	18.99	17.15	15.79	14.75	13.94
4	45.37	37.33	31.98	28.19	25.35	23.16	21.41	18.81	16.99	15.64	14.61	13.80
5	45.00	37.01	31.71	27.94	25.12	22.95	21.22	18.64	16.83	15.49	14.47	13.67
6	44.65	36.71	31.44	27.70	24.91	22.75	21.03	18.47	16.68	15.35	14.34	13.55
7	44.33	36.43	31.19	27.47	24.70	22.55	20.85	18.31	16.53	15.21	14.21	13.43
8	44.02	36.16	30.95	27.25	24.49	22.36	20.67	18.15	16.38	15.08	14.08	13.30
9	43.74	35.90	30.72	27.04	24.30	22.18	20.50	18.00	16.24	14.95	13.96	13.19
10	43.48	35.66	30.50	26.84	24.11	22.00	20.33	17.85	16.10	14.82	13.84	13.07

5% Residual

0	45.22	37.32	32.06	28.32	25.52	23.36	21.64	19.08	17.28	15.95	14.93	14.13
1	44.78	36.95	31.74	28.04	25.27	23.13	21.42	18.89	17.10	15.79	14.78	13.99
2	44.35	36.59	31.44	27.77	25.03	22.90	21.22	18.71	16.94	15.63	14.64	13.86
3	43.96	36.26	31.15	27.51	24.79	22.69	21.01	18.53	16.78	15.48	14.50	13.72
4	43.58	35.94	30.87	27.26	24.56	22.48	20.82	18.35	16.62	15.34	14.36	13.59
5	43.23	35.64	30.60	27.02	24.35	22.28	20.63	18.19	16.46	15.20	14.23	13.47
6	42.89	35.35	30.35	26.79	24.13	22.08	20.45	18.02	16.32	15.06	14.10	13.34
7	42.58	35.08	30.10	26.57	23.93	21.89	20.27	17.86	16.17	14.92	13.97	13.22
8	42.29	34.82	29.87	26.35	23.73	21.71	20.10	17.71	16.03	14.79	13.84	13.10
9	42.02	34.57	29.64	26.15	23.55	21.53	19.93	17.56	15.89	14.66	13.72	12.99
10	41.76	34.34	29.43	25.95	23.36	21.36	19.77	17.41	15.76	14.53	13.60	12.87

10% Residual

0	43.37	35.88	30.90	27.35	24.71	22.66	21.03	18.60	16.89	15.63	14.67	13.92
1	42.94	35.52	30.59	27.08	24.46	22.43	20.82	18.42	16.73	15.48	14.53	13.78
2	42.54	35.18	30.30	26.82	24.23	22.21	20.62	18.24	16.56	15.33	14.38	13.65
3	42.15	34.86	30.02	26.57	24.00	22.01	20.42	18.06	16.40	15.18	14.25	13.51
4	41.79	34.56	29.75	26.33	23.78	21.80	20.23	17.89	16.25	15.04	14.11	13.39
5	41.45	34.26	29.49	26.10	23.57	21.61	20.05	17.73	16.10	14.90	13.98	13.26
6	41.14	33.99	29.25	25.88	23.36	21.42	19.87	17.57	15.95	14.76	13.85	13.14
7	40.84	33.72	29.01	25.66	23.17	21.23	19.70	17.42	15.81	14.63	13.73	13.02
8	40.56	33.48	28.79	25.46	22.98	21.06	19.53	17.27	15.67	14.50	13.60	12.90
9	40.30	33.24	28.57	25.26	22.79	20.88	19.37	17.12	15.54	14.37	13.48	12.79
10	40.05	33.02	28.37	25.07	22.62	20.72	19.21	16.98	15.41	14.25	13.37	12.68

15% Residual

0	41.52	34.44	29.74	26.39	23.89	21.95	20.41	18.12	16.51	15.32	14.41	13.70
1	41.11	34.10	29.44	26.13	23.65	21.74	20.21	17.94	16.35	15.17	14.27	13.56
2	40.72	33.78	29.16	25.88	23.42	21.53	20.02	17.77	16.19	15.02	14.13	13.43
3	40.35	33.47	28.89	25.64	23.20	21.32	19.83	17.60	16.03	14.88	13.99	13.30
4	40.01	33.17	28.63	25.40	22.99	21.13	19.64	17.44	15.88	14.74	13.86	13.18
5	39.68	32.89	28.38	25.18	22.79	20.94	19.46	17.28	15.74	14.60	13.73	13.05
6	39.38	32.63	28.15	24.96	22.59	20.75	19.29	17.12	15.59	14.47	13.61	12.93
7	39.09	32.37	27.92	24.76	22.40	20.57	19.12	16.97	15.45	14.34	13.48	12.82
8	38.82	32.13	27.70	24.56	22.22	20.40	18.96	16.82	15.32	14.21	13.36	12.70
9	38.57	31.91	27.50	24.37	22.04	20.24	18.81	16.68	15.19	14.09	13.25	12.59
10	38.34	31.69	27.30	24.19	21.87	20.08	18.65	16.54	15.06	13.96	13.13	12.48

20% Residual

0	39.66	33.00	28.58	25.43	23.07	21.25	19.80	17.65	16.13	15.01	14.15	13.48
1	39.27	32.68	28.29	25.17	22.84	21.04	19.60	17.47	15.97	14.86	14.01	13.35
2	38.90	32.37	28.02	24.93	22.62	20.84	19.41	17.30	15.81	14.71	13.88	13.22
3	38.55	32.07	27.76	24.70	22.41	20.64	19.23	17.14	15.66	14.57	13.74	13.09
4	38.22	31.79	27.51	24.48	22.21	20.45	19.05	16.98	15.51	14.44	13.61	12.97
5	37.91	31.52	27.28	24.26	22.01	20.27	18.88	16.82	15.37	14.30	13.49	12.85
6	37.62	31.26	27.05	24.05	21.82	20.09	18.71	16.67	15.23	14.17	13.36	12.73
7	37.35	31.02	26.83	23.85	21.63	19.92	18.55	16.52	15.10	14.04	13.24	12.61
8	37.09	30.79	26.62	23.66	21.46	19.75	18.39	16.38	14.96	13.92	13.12	12.50
9	36.85	30.58	26.42	23.48	21.28	19.59	18.24	16.24	14.83	13.80	13.01	12.39
10	36.63	30.37	26.23	23.30	21.12	19.43	18.09	16.11	14.71	13.68	12.89	12.28

Source publication No. 728, residual lease rates. Financial Publishing Co., 82 Brookline Ave., Boston, MA 02215.

being the equipment itself. If this is the only collateral, the bank will probably want you to make a large down payment, maybe 20%. In this way, the bank is assured that the resale value of the equipment will always be worth more than you owe on the loan.

To keep the payment down as low as possible, try and make a case for the retention of value in the piece of equipment. This is an area of negotiation where you may use the letters you have received from those newly acquired dealers around the country. Although the banker may not fall for this one, it's surely worth a try.

There are also professional equipment appraisers that specialize in this type of work. For a fee, they will provide you with the closest to the truth that is possible regarding value retention. It might be worth your time and money to secure the opinion of such professionals; by the way, most bankers *will* believe what these companies report.

Banks will give 100% loans to companies that have strong collateral in other areas than just the equipment they are buying. This is why it is important to get

to know your banker and, more importantly, to let him get to know you.

Loan interest

Now you have been approved for the bank loan and the question of interest is brought to the table. The easiest loan to understand is the simple interest loan, in which the principal is equally divided over the payment period in monthly installments. Every month, in addition to the principal, you pay the interest on the unpaid balance.

Because the interest declines as the principal declines, your early payments will be the highest, with a continual drop in payments until totally paid off. Today, such loans are tied to the prime rate so that interest may change from time to time. If your desire is to conserve cash in the beginning, this type of loan will not be appropriate because it results in a heavier cash burden up front.

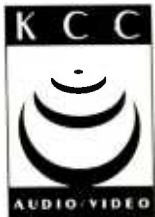
To get an idea of how much cash you will be putting out over the entire loan, divide the interest rate by two and multiply the total principal by this answer. This will give you a close approximate of the total interest you will pay on this

loan. Now add this number to the total principal and you will have the answer. Of course, your banker or accountant can run you an entire payment schedule based on current interest rates.

Because of this formula, a 12% loan is sometimes called a 6% add-on loan. Although you will be paying 12% interest, the banker has left you with the perception that your loan is really at 6%. Be very careful of this trick. I have seen situations in which a studio thinks it has secured a 12% loan but is, in reality, paying 24%.

Another type of bank loan will fix the interest over the entire loan, for which the bank will make your monthly payment the same over the entire payback period. When this happens, make sure there is no pre-payment penalty. Naturally, under these conditions most of the money goes to interest during the beginning months. After the first year, your principal might be higher than expected. Because this type of arrangement borders on what the finance companies offer, let's move to the capital lease.

Capital leases are the same as bank loans to the Internal Revenue Service:



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The background of the advertisement features a large, stylized graphic of a vinyl record with vertical lines representing the grooves. In the bottom right corner, there is a smaller, circular logo of a vinyl record. The text 'FUTURE DISC SYSTEMS' is prominently displayed in a bold, sans-serif font, with 'INCORPORATED' in a smaller font below it. Below the company name, the text 'COMPLETE ANALOGUE & DIGITAL MASTERING SERVICES' and 'FOR COMPACT DISC, RECORD & CASSETTE MANUFACTURING' is presented in a bold, sans-serif font. At the bottom, the address '3475 CAHUENGA BLVD. WEST, HOLLYWOOD, CALIFORNIA 90068' and phone number '(213) 876-8733' are listed.

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you write off the interest and depreciate the principal. Capital leases can be very creative. Some companies will not only loan you money for your equipment, but they will loan you money to pay for it. Such loans usually show themselves as having very low payments during the first year with increasing payments as the payoff period progresses; some even have a balloon payment at the end.

This approach is useful if you really need a piece of equipment, but your present cash flow does not allow you to proceed with the normal financing. The down side is the total amount of cash you'll ultimately require to buy this piece of equipment can be as much as twice by this method. Naturally, if this is your only choice, go for it wisely.

For straight capital lease financing, the interest is usually computed in a complicated way using the Rule of 78 method. (For a full breakdown of what this means, write to the Financial Publishing Company, 82 Brookline Ave., Boston, MA 02215, and ask for publication No. 728. The book will tell you how to compute your interest, and also contains all the tables that finance companies use in setting your interest rate.)

Finance companies will usually make a loan and then sell the "paper" to other investors. As a result, there might be hidden extras, such as a closing cost or administration fees. Make sure you know what you are paying for. Another factor to be concerned with is the date of the interest computation. For example, a loan that reads 10% with the interest being computed on the first day of the month is really a 10.25% loan. If computed on the last day of the month, however, it would be a true 10% loan.

Loans computed on the Rule of 78 usually have penalties for early payoff; make sure this is clearly spelled out in the loan agreement. As I said before, there is no tax difference between a

It is OK to use any of the above methods of financing, as long as it makes business sense.

bank loan and a capital lease, the only difference being that the latter will conserve cash in the early period of the loan. You may have plenty of cash available, but want to reserve your bank lines of credit for something else down the road. This would be another valid reason to go with a capital lease.

Although capital leases are usually more expensive than a bank loan, it is possible to obtain very competitive capital leases if indeed you can prove your capability in obtaining a bank loan.

True leases

With a true lease, the ownership of the equipment is in the name of the leasing company. Your agreement may give you the right to purchase the equipment at the end of the lease for market value; this is legal. However, if the lease gives you the right to purchase the equipment at the lease termination for a fixed amount, such as 10% or some other similar fixed amount, the IRS will rule this to be a *capital lease*. You will then lose any tax benefits available with true leases.

In a true lease, you can write off the *entire* payment. With the new tax law, there are questions regarding an extension of the periods used for depreciation. If this proves to be true, under certain conditions the faster write-off afforded by true leases will be attractive.

Loans computed on the Rule of 78 usually have penalties for early payoff; make sure this is clear.

Another attractive element of true leases is that they are cheaper than capital leases. If you are buying a piece of equipment that will have no resale value at the end of the pay-off period, you'll actually pay *less* for this piece of equipment during its usable lifetime. When the lease runs out, just send it back to the leasing company. If it does still have some value, your ability as a negotiator may buy you something worthwhile at a good discount. (After all, what are outside investors going to do with a 24-track recorder?)

In some states, if you were to buy the equipment under the terms of a capital lease, your sales tax would be much higher than by buying at market value under a true lease. A machine costing \$100,000 new would cost, say, \$8,000 in sales taxes under a capital lease. When bought five years later at market value under a true lease, for say \$10,000, it would attract a tax of \$800—an overall savings of \$7,200. Add the compounded interest on top of this \$7,200, and you have one *nice* saving. (Of course, there is always a risk that the market value will be very high, causing you to pay twice for the same machine.)

Real-life examples

Let me show how Universal handles the above scenario, and for what reasons. When we bought our 3M DMS digital recorders in 1980, we believed the machines would be totally obsolete in five years, and used a true lease. When the term ran out, we placed a blind ad in the trades to see what we could get for the machines. The leasing company placed a market value price far *below* the equipment's going price. This meant that we could exercise our option and turn an immediate profit (or, more truly, a reduction in cost). Because the machine was still in use at the studio we did indeed exercise our option.

When we were considering the purchase of our first New England Digital Synclavier, we went to a leasing company that would give us an expanded line of leasing credit, because we knew that we would probably be expanding our systems. (Expanded credit makes it easy to buy new expansions for the same system.) The leasing company is already familiar with the main equipment suppliers. When going into computers, we feel this is a good idea. Once again our choice of true lease or capital lease was based on the guesstimated residual value left in the equipment at the end of the lease term.

Our bank was always complaining that we rarely used them for this type of financing. We told them that we would use them if they could beat the rates of the leasing company. On our next purchase they beat the leasing company, so we gave them the account.

Enough war stories. Figure 1 represents an example of a page out of the finance company bible. As you can see, it shows the factor to be used for many payout conditions. To use this set of tables, just take the cost of a piece of machinery and multiply it by its correct factor. This will give you your monthly payment. If you send away for publication No. 728 from Financial Publishing Company, you can back your payment into a factor, and look up what your interest rate is in reality. You will also look up the residual value and the influence of advance payments. We use this book all the time—it has helped us with all of our bank and finance company dealings.

It is OK to use any of the above methods of financing, as long as it makes business sense. To buy a machine and then pay for it twice while you may be discounting your studio by 50%...well I guess what I am saying is count your money and make wise decisions.

Real estate: The safety net for studios

If you were to list the pros and cons of owning the building and land occupied by your studio, the pros would win hands down. This is not to say, however, that such an arrangement is not without its own inherent problems.

When my partners and I bought Universal from its former owners in 1975, the money that paid for the studio went to settle long-term debts and taxes. The money paid for the real estate created a profit of \$1.7 million for its former owners. In other words, after being in business for 29 years (1946-1975), the only company profit was from the real estate.

Another associate recently sold his facility for several million dollars to a public company, but retained ownership in the property that houses the studio. You might say he has his cake, and can now eat it too. Yet another studio was able to finance its expansion by borrowing against the increases in property value experienced by the real estate.

On the other side of the fence, a friend bought a large building and a pile of video equipment. He was not able to generate enough cash flow to cover his overhead, loan payments and mortgages. He lost the building and put the company into Chapter 11. He is presently at risk of going into personal bankruptcy.

Another studio owner in a Chicago suburb owes a great amount of money to the banks. He would love to liquidate his property, which is appraised at a high figure, to pay off his debt. The only problem is that the neighborhood is going down, and nobody wants to pay his price.

A careful decision

As you can see from the above stories, some win while others lose. However, the losers all have one thing in common: They do not know any more about the real-estate business than they know about the studio business. Whenever you plan to spend a large amount of money for a studio or a piece of land, you should try to understand the dynamic forces that control the destinies of such ventures.

I believe in what is often referred to as the "Allen Holy Trinity" con-

cept of operating a successful recording studio. This concept revolves around the development of three corporations. The first corporation is in the real-estate business; it owns the property. One of the people involved with this entity should know the property business, including future city plans, future values on various parcels of land and the wherewithal to fight for the most favorable property taxes. The day may come when the studio isn't using this property, and having someone aboard who knows the business of real estate will prove invaluable.

The second corporation will be the entity that owns the equipment. This group will buy and/or lease the technical equipment to a third corporation, the recording studio, which will have no other tangible assets other than its future retained earnings. It will rent equipment from corporation #2, and rent its space from corporation #1, which has included as part of its rental fee the cost of all leasehold improvements.

The master plan behind the scheme is to set the payment schedules in such a way that the recording studio (corporation #3) rarely shows a profit; the "leasing" company (corporation #2) throws off a good cash return to the principles; while the real estate company (corporation #1) absorbs the brunt of whatever profit is created. One way of accomplishing the latter is to write a lease tied to gross sales of the recording studio.

Assuming that all three corporations have the same owners (with maybe one exception: a real estate expert for corporation #1; an equipment financing specialist with corporation #2; and a recording studio wiz with corporation #3), the principles will be able to have the best of all worlds. The recording studio will keep them busy and pay their salaries. The leasing company will create dividends, while offering depreciation as a device to lower net income as well as taxes. The real-estate company will also offer depreciation while, in effect, it may appreciate in value. When it comes time to cash out, it is much easier to sell a piece of commercial real estate that has a good record of profitability than a recording studio.

Of course, to make this all come true, the recording studio **must** be operated properly; it must throw off enough cash to meet all of the leasing and real-estate requirements.

Being in the real-estate business is not everybody's cup of tea, however, if you have a one-room operation in a downtown area, renting makes good sense. If you do not have the ability to provide the necessary funds for real-estate investment, then renting is the only way to go. If you are required to operate in a certain location where there are no buildings available at a price you can afford, then the choice to rent, as opposed to not being in business, is a good one.

Sometimes renting can be profitable. I know of a New York studio owner who had a long-term lease. The owners of the building wanted to terminate his lease early, so they ended up paying him \$2.1 million dollars to let them out of the lease. You could say that, in some cases, renting is a fantastic method of doing business.

Let me throw one more variable on the fire. Let's say you have plenty of excess cash. You would like to buy the building in which your studio is operating. The price is right. The problem is that by doing so would you be putting all your eggs in one basket? By investing in real estate elsewhere would you receive a better return on investment? Would an investment outside of a studio (one-purpose) building offer more long-term security? If you are going into real estate should you approach it like a business and make the optimal use of your money?

Resale potential is the most important factor for the real-estate speculator. When going into commercial real estate, you should make your decisions along business lines; to own property just for the sake of ownership may not always be wise.

When you decide to buy the land and building for your studio, you may be making the most important business decision of your life. Take your time and do it wisely. The correct decision may reap benefits beyond your wildest dreams and make you a rich person to boot.

R·E·D

Time Code and Synchronization:

The Link Between Film Production and Post-production

By Larry Blake

Continuing our 3-part series of articles that concentrates on providing practical, hands-on information to help alleviate most of the common time code and sync problems.

Almost without exception, every U.S. feature film since the coming of sound in the late Twenties has relied on manual labor to synchronize sound and picture. At the beginning (or sometimes end) of a take, a clapper (slate) is held up, displaying the scene and take numbers along with production information. A striped piece of wood at the top is brought down onto the slate proper, creating a visual and aural synchronization point.

In the editing room, a picture "workprint" from the camera negative and a 35mm mag print from the 1/4-inch production sound tape are synchronized by hand. Later, a coding machine applies inked edge numbers to picture and sound at every foot, thus helping the editors identify the origin (scene and take, lab and sound roll numbers, etc.) of the hundreds of small pieces that make up a feature film. The negative is conformed to the workprint with the latent key numbers applied by the film manu-

facturer. Everything is laboriously logged into a series of code books.

It becomes obvious that some form of automation can be brought into the editing fray to reduce the amount of manual work during film post-production. Everyone agrees: Time code and video editing, in some form or the other, are the obvious answers. What is up for grabs at this point is exactly *which* form and how it should be implemented.

Time code slates

As a result of informal standardization of center-track 1/4-inch time code by a variety of manufacturers during the mid-Eighties, time code's presence in film production has increased by leaps and bounds. Concurrent with this development has been the use of time code slates to display to the camera the time code address that corresponds to the 1/4-inch tape. When recording a scene, the slate will usually indicate the time-of-day (TOD) code being generated by the production recorder. On a playback shoot, the slate will indicate the location of a

specific point in the song.

During film-to-tape transfer (telecine), a number is read off the slate and keyed into a device that converts telecine motion to time code, thereby imbuing that film frame with the address visible on the monitor. In effect, time code is synthesized and "follows" the telecine machine in forward or reverse. The 1/4-inch studio deck searches to find the same location on the production or playback tape.

The TOD code is often employed for telecine synchronization only, with new, continuously ascending code printed onto the longitudinal time code track of the videotapes. Thus, if 20 minutes of film were shot that day, the codes on the tapes would run from 01:00:00:00 to 01:20:00:00. The original TOD or playback code numbers can be preserved on the tapes either by printing them either in a second burn-in window or on the off-line tapes or as vertical interval time code (VITC) in the video picture. (VITC will be discussed in part three of this series, to be published in the October issue.)

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

Either way, the edit decision list (EDL) will use the new numbers, because continuous code is much easier for edit systems (and editors) to deal with than TOD code that might have 2-hour jumps between adjacent scenes. Also, creating new time code easily provides unique addresses in multicamera shoots. (Then again, there is another argument that TOD code should be carried into post-production *especially* in concert situations, when one might want to view and edit from multiple angles simultaneously.)

Common problems with center-track time code

The most common mistake cited by telecine operators (AKA "colorists") regarding time code slates is the short pre-roll that many production mixers give when recording. When backing up for the pre-roll, the synchronizer has to read at least 5 and preferably 10 seconds of up-to-speed time code addresses on the 1/4-inch tape. (Pre-roll on sound is not necessary during playback shoots because a tape of the complete song is on the deck during telecine.)

Thus, if the sound person yells "speed" 5 seconds after the flags are up on the

Nagra, then at that point only inexpensive audio tape has rolled, and the camera department can slate the scene with the same speed that they have done for the past 60 years. Of course, pulling this off without making it appear that everyone has been waiting for sound requires that the mixer anticipate when everybody *will* be set and roll the recorder earlier than usual.

Contrary to popular mythology, there is *no* need for a long 5-10 second pre-roll time on the cameras. Because professional film cameras are up to speed within a second, shooting the time code slate for a few seconds is considered sufficient time for the colorist to find a valid frame. "Pre-roll" on the telecine machine is accomplished simply by backing over the tail of the previous take: a sprocket hole is a sprocket hole.

Rich Garibaldi of Telecine Tech, Hollywood, was one of the first to synchronize audio tape and film during telecine in the United States, and has been doing it since the fall of 1983, approximately a year before the time code Nagra and time code slates became commercially available. He notes that, as recently as July 1987, he did a job "where the clapper-loader [the assistant cameraperson hold-

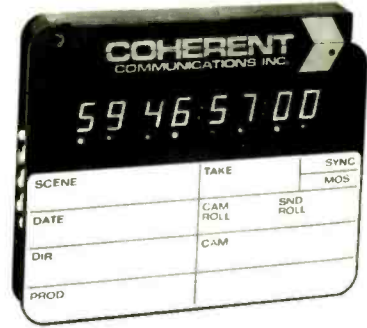


Figure 1. Coherent Communications TC-500A "smart slate" with internal time code generator.



Figure 2. Denecke Dcode TS-1 reader-only slate.

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ing the slate] stood there with the time code numbers rolling and counted 'one, two...' all the way to five, and then clapped the sticks. It was insane. They burned up a lot of film and later on in telecine they complained.

"This is a textbook case of everything we have tried to avoid: disrupting the talent and the flow and rhythm of the production."

There are two anomalies that production mixers have to be aware of regard-

ing the accuracy of slate readings. First, the code output of the Nagra IV-S TC in record mode reflects the delay that is introduced to the time code signal in order to make it time-coincident with the audio at the record head. (At 7.5ips the delay is approximately 4 frames, and at 15ips it is 2-3 frames.)

Some exceptions to this rule should be noted. First, there is no delay on the output either when the Nagra IV-S TC's main function knob is set to the test posi-

tion or during playback. Thus, if one is using a smart slate, such as the Coherent Communications TC-500A (Figure 1) containing a time code generator, the TC should be set to test when jamming code into the slate, which will then always display the correct time.

The second issue that the production mixer should be aware of is that some slates, including Denecke Dcode TS-1 (Figure 2), do not display the current frame, but are instead a frame late. Again, this problem is not present with smart slates. (One must always be sure that the internal generator on a smart slate is set to the same frame rate as the Nagra!) The bottom line is that the venerable clapstick helps the colorist determine the correct offset, if any, on the first few takes of a session.

All of the above assume that the IV-S TC will be used in conjunction with a film camera. Should the TC record high-quality "double system" audio during a videotape shoot ($\frac{3}{4}$ -inch U-matic, Betacam, 1-inch, etc.), the code recorded by the Nagra *must* be sent from a separate external time code generator that is locked to the video. (Remember to always set the TC to the same time code frame rate as the incoming time code.) The IV-S TC's time code should not be distributed to the video decks, not only because of the offset in the record mode, but also because the TC's time code generator cannot lock to external composite sync or composite video.

Probably the most avoidable problem that people make when using time code concerns the frame rate to which the TC's time code generator is set *vis-a-vis* the camera's frame rate. Standard practice today has time code running at 30fps when the film camera is shooting at either 24fps or 30fps. Only when the time code is printed directly on the film in the (24fps) camera should the Nagra time code frame rate match that of the cameras.

One of the primary reasons for 29.97/30fps code being used in normal 24fps film production (with time code slates) is that video and film audio post-production facilities are used to handling sound elements with 30-frame time code. Thus, the 30fps/60Hz field code will match exactly with the 29.97 frames per second/59.94Hz code striped during telecine. After off- and on-line video editing, access to original sound elements will be simple because, once pulled down to 59.94Hz, the latter will be at the same frame rate as the edit master. While most synchronizers allow for the use of 24fps "film" time code, 30fps code is always a safe bet.

War stories in the tri-state area: Interlocking film, audio and video: Case story No. 2

A rock concert documentary was photographed in NTSC videotape, with 29.97fps/59.94Hz time code sent to the multitrack machines in addition to the 1-inch video decks.

After a successful video release, a stereo theatrical exhibition was planned. The edit of the film would remain the same as the video release, and the 1-inch edit master was transferred to a 35mm release negative. During tape-to-film transfer a 0.1% speed up occurs as the time base of the image is changed from 59.94Hz to the 60Hz time base of 24fps film.

Because of the vast differences in the requirements of stereo home video and Dolby Stereo theatrical release, the whole concert was re-mixed. Mixing proceeded in the standard recording studio fashion: the fader and EQ settings were rehearsed and, once everything was right, the mix was printed. The only difference here was that the recording medium was 35mm magnetic film brought to the dubbing stage to make the Dolby Stereo printing master.

On the first day of film dubbing, the soundtrack slowly but surely started trailing behind the picture. What had happened?

Answer: During the mixdown to 35mm from the original multitrack elements, the mag recorder and the multitracks were not locked to the same sync source. While the multitracks were locked to a NTSC sync generator and thus were being resolved, the mag machines, on the other hand, were referenced to 60Hz line. Therefore the mix would play back at its original speed at the dubbing stage, when picture and sound were both locked to 60Hz. The problem is that the

sound, during 60Hz playback, did not undergo the 0.1% speed up that the picture did as the result of the tape-to-film transfer.

The irony here is that the solution—referencing the multitrack and mag recorders to the same source—would also have given them the flexibility to punch-in and update the mix. (By the way, it wouldn't have mattered whether the common sync generator had been referenced to 59.94Hz or 60Hz; the former figure would have resulted in "slow" mag elements that would be "pulled up" during the film mix. Locking to 60Hz would have pulled up the speed of the multitrack and thus the mag elements would "already" be sped prior to going to the dubbing stage.)

The solution to the problem was to transfer all of the mag elements to another set of 35mm units. The original 35mm mix was resolved to its "slow" 60Hz speed while the 35mm recorder was referenced to 59.94Hz. Thus, the speed of the program would increase the proper degree when the mag dubbers were interlocked with the film projector at the mix.

Moral of the story: Make sure that you understand the speed change that results from tape-to-film or film-to-tape transfer. Also, when mixing the audio, locking all machines to the same sync source will almost always result in maximum flexibility, no matter what happens later on in post-production. It is a safe bet simply because one is in effect giving the mix-downs the same sync reference as the original tracks.

It is also wise to continue with either drop-frame or non-drop frame time code throughout a project. Again, many synchronizers will allow the mixing of drop and non-drop, but keeping *one* of them constant from field recording to telecine through to off- and on-line videotapes will make everyone's job easier.

Assuming that the correct time code frame rate has been chosen, camera and sound logs should note what was done, plus what offset the telecine operator (or assistant film editor) should expect, as in: "Sound recorded at 7.5ips with 30fps NDF time code. Camera ran at 24fps. Five-frame offset." ("NDF" indicates non-drop while "DF" denotes SMPTE drop-frame time code.)

Similarly, logs for a playback shoot should reassure the colorist that you were aware of the slow-down that occurs in telecine: "This tape contains 29.97fps/59.94Hz NDF time code and was cross-resolved during playback to 30fps/60Hz. Camera ran at 24fps." (For more information on the use of time code in playback shoots, see Part One published in the August 1987 RE/P.

Telecine operators have to find a valid frame displayed on their monitors that they will then enter into the system. A big problem is that the camera shutter might be open during the changeover between two time code frame numbers. Thus, there would be nothing distinguishing a 01 from a 00, or telling you that 09 was actually created from a combined 03 and a 04. Aware of this, colorists will simply inch their machines frame by frame until they are sure that they see a valid frame change or a number like a 2 (02, 12, 22) that cannot be formed by combining a 7-segment representation of two other numbers.

Garibaldi notes some of the designs of time code slates actually hinder rather than help him find a valid frame.

"The design of the Denecke TS-1 slate that freezes the time code display [for three frames] on the 'clap' is running under the misconception that I use *that* frame as the synchronization frame," he says. "By freezing the display for three frames, I know that two out of three are incorrect and, for that reason, I conspicuously avoid anywhere near the clap. It's an error-creating situation to have the *same* time code displayed on more than one frame of film.

"It also limits where I can synchronize the picture to the track, because I can *only* use those frames that include a photograph of the slate before the clap. If the time code were continuously displayed through and beyond the clap, all of *those* frames would be available to me

for synchronization purposes. If the slate is in the wrong position at the clap, or the camera is not at speed yet, I would have as much as twice as many opportunities to find out the time code number." (The unit's designer, Mike Denecke, says he is aware of this issue, and the slate can be modified to not freeze the reader at the clapstick.)

Garibaldi says that the Coherent Communications TC-500A slate that normally exhibits only even-numbered frames, also ostensibly to aid him in finding a

valid frame, in fact "cuts the accuracy of the synchronization system in half because you still have to find the frame where the roll-over [time code frame change] occurs.

"If you have two frames called 'four' [04], the first one is right and the second one is really five [05]. You have to assure yourself which one you're on before you can use it to synchronize."

Ivan Kruglak of Coherent says that the even-frames display function can be altered by switching PROMs, and that

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Time code on film

While SMPTE/EBU time code was developed to facilitate videotape editing, the motion picture industry has had little incentive to change from its manual procedures. A necessary first step has been taken with the use of video editing in the post-production of music videos, commercials and episodic television series. All of the above employ time code slates to synchronize the worlds of film, audio and video during telecine, after which time the film elements are not used.

Because there is no need to go back to the original film, the considerable cost of making film workprints can be saved.

Theatrical features require hundreds of film prints, however, and the main problem involved with not making a film workprint is that the production company never sees the film projected on a screen until the negative is cut, and the "first answer print" is made. Also, unless inexpensive kinescopes (film copies of videotape) are made, it will be difficult to use standard 35mm sound editing and re-recording techniques. Multitrack and video-based sound editing offer as many disadvantages as they do benefits, and

"hybrid" will be the operative word until random-access sound editing truly arrives.

Thus a system that actually prints time code on the camera negative would allow a video EDL to be used in negative conforming.

Although time code has been recorded on film since the late Sixties, most notably in Europe with the EBU/IRT system, only in the past few years has there been much activity in the filmmaking community at large. Most of the attention today is focused on systems sponsored by the two most prominent 35mm professional camera manufacturers: Panavision, Tarzana, CA, and Arnold and Richter (Arriflex), Germany.

Panavision has been promoting the Film Data Track (FDT) system developed by Jean-Pierre Beauviala of the French camera company Aaton. [In late July, the system's name was changed to Panavision/AatonCode—*Editor*.] Arriflex is using the time code system developed by Coherent Communications, Sylmar, CA.

Briefly, here are the operational similarities between the two systems:

- A small hand-held master clock is used to set the time both on the recorder and all film cameras. The accuracy of any device's generator can be checked relative to the master, and presumably everything would be reset after the lunch break regardless of the potential crystal accuracy. At the end of the day, start/stop and scene/take information can be loaded from the cameras either onto a printer or onto computer disk.

The tough decision is exactly how much of the process will be automated, and how dependent is the operation on information fed into it. To wit: Will everything grind to a halt if crew members are resistant or even merely lazy in their application of the system?

- The presence of time code information on the film negative has the potential of saving money by eliminating the stock, processing and printing costs usually consumed by the camera shooting the slate (either normal clapstick or high-tech time code) for synchronization purposes. This savings alone could pay for extra equipment, although perhaps crews and actors will have to be weaned away from needing the clapstick to call them to attention.

During multicamera concert shoots, camera operators will be freed from the necessity of shooting full film magazines after photographing a time code slate. Instead, they can start and stop at will, knowing that there will *always* be a valid time code address printed on the negative.

- Transfer to videotape will be sped up

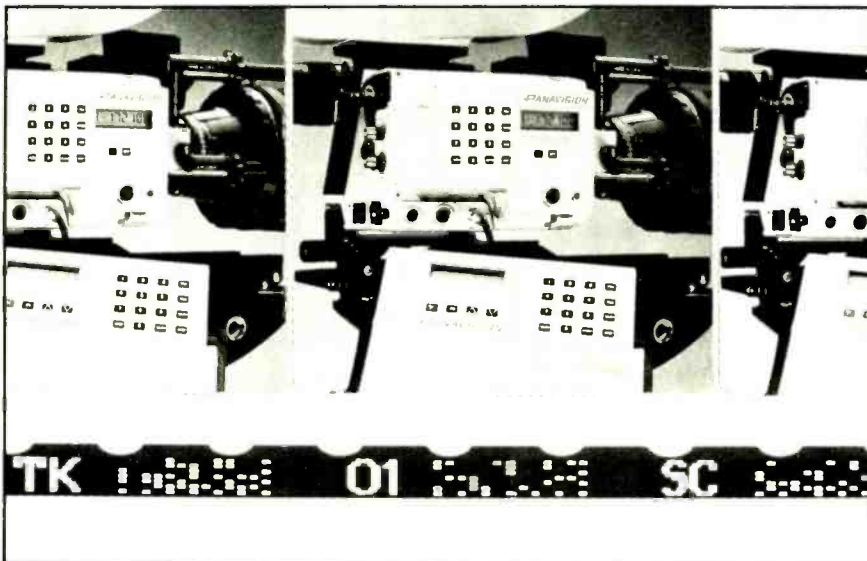


Figure 3. AatonCode on 35mm film, with machine-readable 7x13 matrix and man-readable figures, here indicating scene and take numbers.

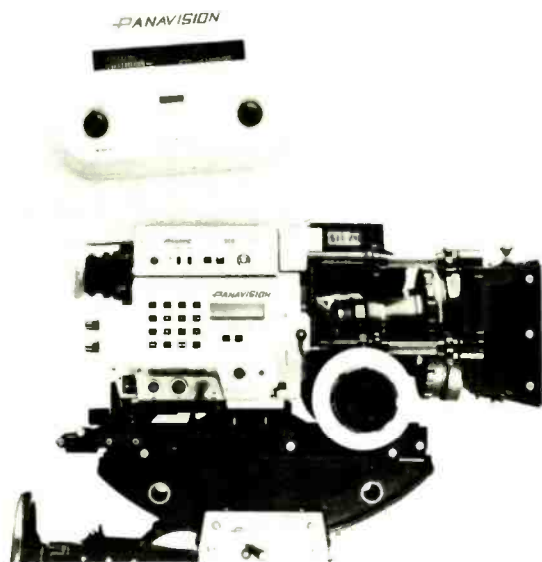


Figure 4. Panavision Platinum 35mm camera with AatonCode module installed.

considerably by a reader in the telecine, which automatically gives time code addresses to the 1/4-inch deck with the production tape. (See the section on the Aaton Film Data track below.)

- Important to the acceptance of time code-on-film systems will be the role of random-access video editing systems, such as Montage and Ediflex, especially in conjunction with 24fps video decks. With current video editing systems, there remains the problem of translating a 30-frame video EDL to original 24fps film negative. Among other problems, the "3:2 pulldown" used in telecine to make the 24-to-30 conversion results in the two fields in many video frames being composed of separate film frames.

A big concern is ensuring that the EDL from the final video edit will be a *truly* frame-accurate guide for the negative cutter. In standard films, the workprint is on one "gang" of the sprocketed film synchronizer, while the negative is laid in to match it on one or two other gangs. The latent edge numbers from the negative are printed through on the print to ensure correct sync.

There has been much thought applied to overcoming this problem. Probably the most popular idea has the film manufacturer's edge numbers burned into a window on the off-line editing tapes, with the counter incrementing every foot along with the numbers on the film.

A more elaborate technique entails printing information from the edited videotape onto clear, "lightstruck" film leader using an ink-jet printer mounted on a 35mm transport. In the case of the Film Data Track system described below, the human-readable time, camera, date, etc. characters could be printed, presumably taking the original 24fps time code numbers from the vertical interval of the edit master. This ersatz workprint can be thought of as a sprocketed EDL or an image-less kinescope.

Editing systems will also have to create a separate EDL for the soundtrack. Not only does sound often cut in a different location from picture, but frequently the sound from one take is used with the picture of another. It will probably be necessary for systems to create *two* sound EDLs because of the need to edit overlapping sound such as narration, music playback with live singing, or simple overlapping dialogue. Compared to the flexibility offered by standard film editing, any less than a two-sound-EDL system would severely inhibit the ability of the picture editor to manipulate sound.

The Aaton FDT is written between the sprockets on 16mm film and outside the sprockets on 35mm by seven LEDs

mounted in the camera's film gate (Figure 3). The system is currently factory-installed in all Aaton XTR 16mm cameras and is being implemented in a number of 35mm Panavision Platinum, Panaflex Gold and Panaflex-X cameras (Figure 4).

The result is both a machine-readable, once-per-frame 7x13 matrix and a man-readable code usually spread out over 1 second. (This latter code is the original Aaton Clear Time Recording system that has seen much use in 16mm

production.) Marking the end of each man-readable sequence is the time code address, with the letter "s" (following the "seconds" digits) indicating frame "Zero."

The system will thus remove the need for assistant film editors to place arbitrary inked edge numbers on the sound and picture workprints. Instead, the production number, roll number, camera number, date and scene/take information, plus the time code address, are visible on the edge of the camera negative

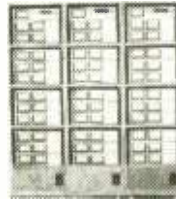


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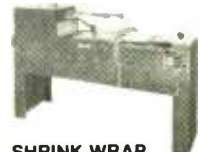
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Figure 5. Nagra T-Audio TC studio deck.

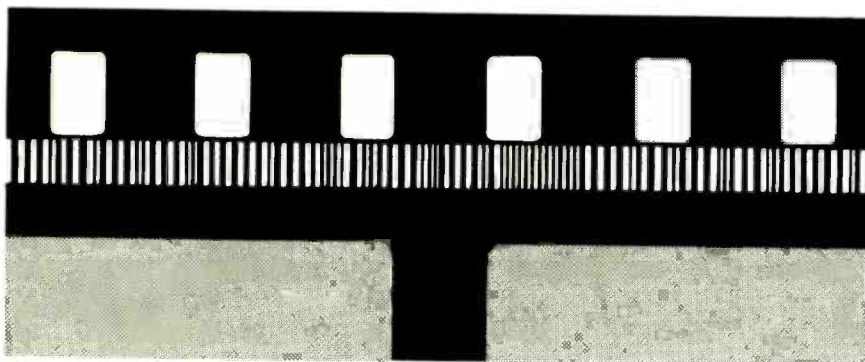


Figure 6. Coherent Communications film time code track on 35mm film.



Figure 7. Coherent Communications time code system, including Nagra time code generator, 16mm camera module and master clock.

and positive film workprint. Readable time address will make it possible to use a video EDL in negative conforming.

To get matching information on the mag transfers, Aaton developed the Adage (pronounced "uh-daj") printing table. The time code recorded on the tails-out roll of 35mm mag (presumably on the balance stripe) is read in reverse and printed on the edge of the mag. The resulting heads-out roll of sound is visually aligned to the picture using the "s" digit.

The first use of FDT in the United States began this spring with a 16mm "the making of..." documentaries on *Harry and the Hendersons*, *Batteries Not Included* and on Steven Spielberg's upcoming *Empire of the Sun*. The new *Star Trek: The Next Generation* television series is being shot with Panavision 35mm cameras recording 24fps time code addresses in FDT on the film negative. Both the telecine and video editing (on a Montage random-access system) are being done at the Unitel Video facility on the Paramount Studios lot, Hollywood.

The Rank Cintel flying-spot scanner at Unitel is one of two in the United States to be equipped with the Aaton telecine reader and controlling software; the other is at Ruxton Ltd., Burbank. By the end of the year, six more FDT telecine readers will be in the United States with five others worldwide.

The 24-frame TOD code from the Aaton controller is sent to the Nagra T-Audio TC studio deck's internal chase synchronizer (Figure 5) and also put in a window on the 3/4-inch off-line cassette for potential future use in conforming the film negative to the final video edit. The original 24fps code is also recorded in the vertical interval of both the 3/4-inch and the on-line master (Betacam or 1-inch). This VITC information will allow the address of the original Nagra IV-S TC field tapes to be followed through on-line video editing, because it is a virtually indestructible part of the video signal.

Steve Buchsbaum, telecine supervisor at Unitel, says that having the production time code on the film, as opposed to having to key it in manually, is a "tremendous" time saver.

"All you do is roll the film, the time code is read and the Nagra [T-Audio] finds it and you're there. An average day's dailies on *Star Trek* takes 2-2½ hours," he explains. Buchsbaum estimates that the same amount of footage, if shot with a time code slate, would take 3-3½ hours.

"It definitely speeds things up, and eliminates the telecine operator from having to worry about finding the number and checking to see if it's in

sync. This way, you check your first edit of the day and, if it's in sync, they're all going to be in sync."

One of the benefits of using FDT in standard film editing is the automatic printing of "meaningful" edge numbers onto picture and sound work elements. A possible use would be the easy matching of reprinted dialogue tracks to the original picture editor's worktrack. Today, a sound editor has to align the two tracks until they phase, whereas with FDT presumably could quickly get within a sprocket hole or two.

However, the expense of an Aaton reader (\$16,000 to \$20,000) will confine automatic machine synchronization using the once-per-frame flag to telecine suites. Automatic synchronization of film and sound elements, on the other hand, is most useful during initial syncing of dailies; during editing all material contains head leader with start marks. You cannot have a machine telling you that your *purposefully* slipped mag element is out of sync.

Coherent Communications Time Code System

Ivan Kruglak has been developing his time code system since the late Seven-

ties, and has been licensing it to Arriflex since the mid-Eighties. In contrast to the Aaton FDT, Kruglak uses a linear, continuous system that looks like a miniature, variable-density optical soundtrack (Figure 6). The track is usually written just inside the sprocket holes, although Kruglak says Arriflex will be widening its reading slit to print across the sprockets and onto the outer edge. It is claimed that the inside-sprocket placement is more rugged in terms of laboratory handling and procedures.

First use of the system in theatrical 35mm with Arriflex 35 BL-4 cameras has been in Europe and the USSR. Retrofitting of cameras with time code modules is expected to be available in September. The system has been in use in 16mm productions (Figure 7) in the United States, providing time addresses for Arriflex cameras that have gone up in NASA space shuttles.

One of the chief benefits claimed of the Coherent system is the small cost of the readers used on editing tables and telecine machines. Kruglak says that the cost is expected to be "a few thousand dollars," although he notes that it's quite possible for the technically inclined to make their own.

"It is basically nothing more than an LED plus a little photocell and a little bit of electronics," he says. "Some of those parts you can get from Kodak [16mm] Pageant projector optical lens system."

A third system for recording time code on film is Datacode developed by Eastman Kodak. After a series of experiments at Universal Studios, it has been decided to halt development. Datacode employs a transparent magnetic coating over the camera negative applied during the time of manufacture. Time code is written and read by magnetic heads using the area outside of the sprocket holes.

We have yet to realize the full potential for time code on film to help eliminate manual work in post-production. After more than 15 years of committees and talk, technology is finally ready for filmmakers. The big question now is whether or not the infamous stationary inertia of the motion picture industry can open its mind to such radically different technology.

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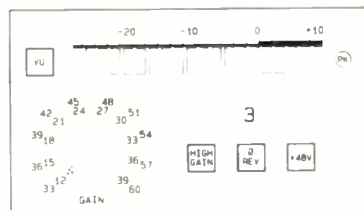
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Facility Spotlight: Red Zone Studios

By Denis Degher

Intending to provide a viable alternative to today's specialist studio, this new facility offers a variety of acoustic environments where different types of music can be recorded.

In today's age of specialization, the *project* studio has become a buzz word. This is a studio that fills a particular niche—be it MIDI, video post, overdubbing, tracking and so on—and tries to corner that specific market.

There is a need to specialize, and in some situations it may be the best solution for a studio owner. Unfortunately, the studio specialist may encounter the same problems as the medical specialist,

who can lose sight of the overall picture, and therefore the patient, while looking for a specific cause-and-effect relationship. In the studio business, the "patient" is the music creator.

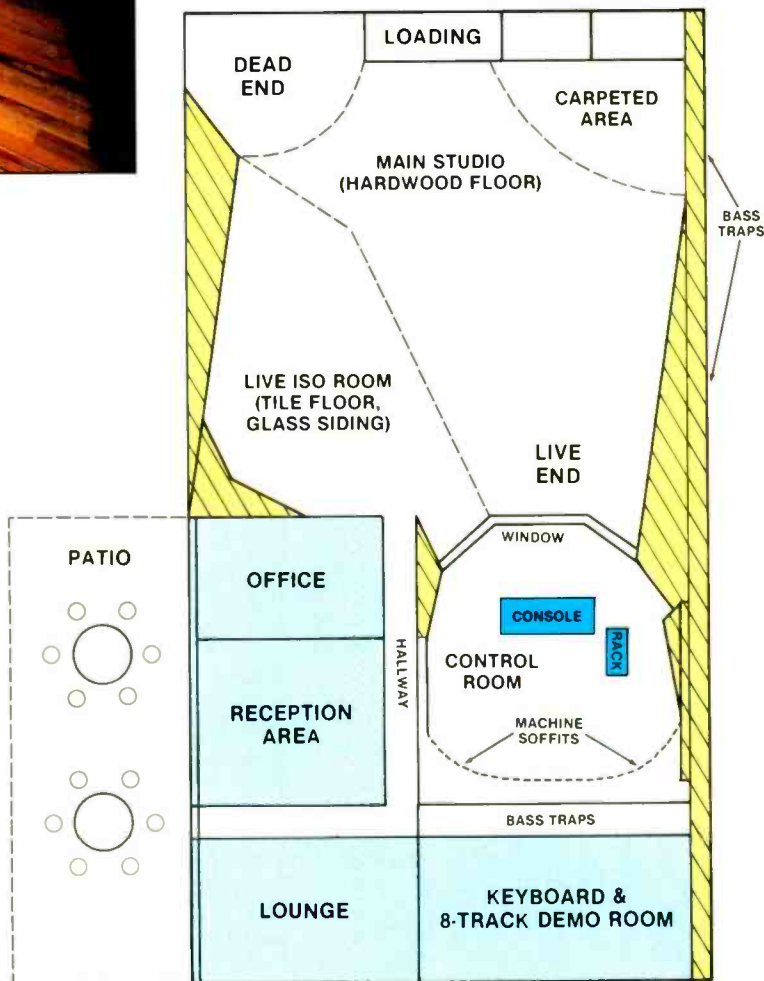
When my partner Frank Riesen and I decided to open Red Zone Studios, our basic concept was versatility. We knew we were going against the current trend, but we wanted a facility that was more in the tradition of the recording studio—a place where many different types of music could be recorded.

We wanted an environment where we

could track hard rock bands, handle a live recording of an orchestra or ensemble, or record electronic MIDI-controlled music in the control room, all within the same facility. To accomplish this dream, we needed to build, or find at an existing location, a facility that was very large and featured a variety of acoustical zones.

Our basic game plan was to build a completely new studio fulfilling our requirements. As sometimes occurs, however, propitious circumstances intervened. While discussing equipment pur-

Denis Degher is part owner in Red Zone Studios, Burbank, CA, and is a regular contributor to RE/P.



Floor plan of Red Zone Studios, Burbank, CA.

chases with Bob Hacken of Professional Audio Services and Supply, Burbank, CA, he informed me that the former site of Kendun Recorders Studio A, located directly next door to his showrooms, was presently available.

We took a look at the site and immediately decided that it suited our needs perfectly. Both the studio and control room were very large. With a modicum of revamping and modernization, I knew we could fulfill our desired sound goals and do it within a much faster time frame. Our course was set to revamp

and revitalize a well-known studio and adapt it to today's music recording needs.

Control room remodeling

The Tom Hidley-designed control room is very large, making it ideal for DI electronic music sessions. Even if we had built the control room from scratch, we couldn't have wanted a larger room. Having recorded electronic music in small, cramped control rooms, I knew the physical as well as psychological value of a spacious control room.

To help us complete the needed control room modifications, we contacted architect/acoustician Vincent Van Haaff. Securing his services—Van Haaff had just finished all of A&M's new studio renovations in Hollywood—was a stroke of luck. As part of the original crew that built Kendun Recorders, Vincent has a great affinity for the complex and took an active interest in the completion of the control room.

Originally designed for quad monitoring, his first task was to acoustically modify and treat the rear of the control

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Author Denis Degher (right) and his co-partner in the studio project, Frank Reisen. Red Zone occupies the building that formerly housed Kendun Recorders Studio A, Burbank.

room, including the rear monitor soffits, for *true* stereo mixing. We accomplished this by filling the soffits with fiberglass to a density similar to that of a Westlake monitor cabinet. We then sealed them with various layers of sheet rock and plywood, thereby essentially turning them into a bass trap.

After positive tests, we cosmetically treated the area and began our work on the front of the control room. Our main priority was refitting the main monitor soffits (which were originally designed to house the studio's very large Westlake cabinets) with a new Tannoy FSM system. The original soffits were of the modern, low-slung design, making costly modifications to the front wall unnecessary. As can be seen from the accompanying photographs, the lowered soffits allow for a comfortable listening position, enabling long sessions without ear fatigue.

Another advantage of the low monitor position is that the sound appears directly behind and over the console-mounted near-field monitors, making the angle and travel between large and small speakers negligible.

After determining the position of the console in the control room, Van Haaff and travel determining the position of the console in the control room, Van Haaff aimed and angled the main monitors. Using various acoustic techniques—coupled with his intimate knowledge of the control room—Van Haaff was satisfied after a day of computations and manipulations. We then played program material to check for stereo spread and panning. After extensive tests and further modifications, everything checked out to our satisfac-

tion and we buttoned down the monitors.

In its former incarnation, the control room featured a large video monitor mounted between the main audio monitors. Although a nice touch, the monitor cavity acoustically impaired the rigidity of the wall between the main monitors, and thereby degraded the room's low-frequency response. We therefore filled the cavity with fiberglass and sealed it with $\frac{7}{8}$ -inch plywood.

This modification had the noticeable effect of tightening and centering the low-frequency response. Satisfied with the aural results, we completed the cosmetics updating and prepared to install the remainder of the gear.

Equipment selection

As a start-up company, we had to be concerned with our bottom line and needed to purchase equipment that fit within the parameters of our needs and expectations. Consequently, at this stage in Red Zone's life, digital multitracks and computerized consoles were out of the question.

Our concept of being a versatile *recording* studio precluded the need for automated mixing capabilities, although the probability of adding it in the future is definitely there. Being located in Southern California means that renting specialty gear such as digital multitracks will be a common occurrence for customers who demand it. With the non-conformity within the digital domain, a wait-and-see attitude only seems prudent business for a start-up facility.

Besides selecting the location, choosing the console was perhaps our next biggest decision. Selecting one became a

question of which one excelled sonically, yet wouldn't cripple us financially.

As an independent engineer I knew what I sought in consoles: "musical" equalizers; flexible signal routing; numerous aux sends; module and function flips; flexible grouping; a very clean and "punchy" signal; a low noise floor; and reliability. As a studio owner, one more consideration had to be taken into account—that of popularity. Was the console considered to be "hip," and was its reputation gaining or waning?

After looking at all possibilities during the Los Angeles AES show last November, and then talking to some very busy and respected engineers about console sonics, I selected the Amek Angela.

The 36-input, 24-output in line console fit our sound and versatility requirements, and has a good reputation for durability. With six dedicated aux returns and the ability to route the monitor inputs into the stereo 2-mix bus, 78 line positions can be realized for multimachine sessions.

When my partner and I decided to open Red Zone Studios, our basic concept was versatility.

The console also features gain control on the line returns, thereby negating the use of direct boxes for many electronic instruments. With no vertical plane from meterbridge to floor, the console's low profile created minimal aural interference from the low-slung main monitors.

Our multitrack requirements were answered by the new 2-inch Otari MX-80 24-track pre-wired for 32 tracks. Having the ability to upgrade to 32 tracks in the future was a bonus, should the 32 track/2-inch format become popular. With Dolby SR noise reduction a possibility for us in the near future, the idea of an analog 32-track machine becomes very appealing.

Other handy features on the MX-80 include the provision of two tape bias setups for each speed, and a master bias control for all tracks. Dolby HX Pro is a standard feature, and helps to extend high-frequency headroom and reduce saturation by optimizing the bias for specific program material. The machine also offers a rehearse mode for punch-ins. For 2-track mastering we elected for Otari MTR-12s.

Our electronic music commitment is based on an 8-foot Kawai grand piano with MIDI Forte. With live recording and acoustic music again gaining popularity, we wanted our MIDI capability to be centered around an instrument. Using a

grand piano as the master MIDI keyboard can be very appealing to keyboard players originally trained on piano. With two separate channels, MIDI sustain pedal functions, and keyboard range control, one pass can yield a lot of great tracks.

Other electronic instruments we have available are an Oberheim Expander, two Ensoniq Mirage Samplers with an extensive sound library, and Akai S-612 sampler and a Sequential Circuits Prophet T8 with MIDI. We also have a Commodore Amiga with Soundscape sequencing and sampling software. Coupl-

ed with Deluxe Music software, the modular system allows music printing.

Recording area

Having gone through various renovations, the Hidley-designed studio area is one of the most innovative rooms I've had the opportunity to use, offering everything from very dead to very bright and "bangy" environments. The unusually shaped main studio features hardwood floors and measures 35'x39'x10'x50'. The studio has 14-foot ceilings that are covered with a multi-angled cloth

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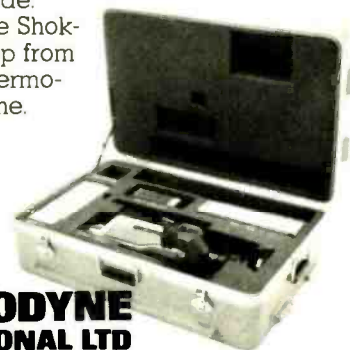
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To the left of the 36-input Amek Angela console are a pair of movable equipment racks that house a wide selection of signal processors. Located on top of the rack is an Amiga PC plus companion MIDI sequencing and music librarian software.



Across the control room, a second equipment soffit houses an Otari MTR-12 mastering deck, an Otari MX-5050 2-track and an Ultimate Support rack housing an Oberheim Xk and a Sequential Circuits Prophet T8.

drop ceiling that makes the room appear much lower than it actually is.

The end of the studio, located next to the control room, is quite live with the polished hardwood floors and window glass. Walking away from the control windows while clapping one's hands is an interesting aural experiment. As you proceed, the studio sounds considerably less reverberant until you reach the very back of the studio where you enter what has been dubbed the *dead zone*. The main studio appears, in fact, to offer distinctly different accoustical zones, making it possible to attain any combination of sounds.

Augmenting the main studio is a live isolation room that measures 24'x10'x 17'x32'. Originally dubbed the string room, the iso booth features red tile floors with mirrored walls and ceilings. It is separated from the main room by six sliding glass doors located on the long side next to the main studio, and two sliding glass doors on the narrower end. Although the iso room is exceptionally live and bright with no parallel surfaces, it is also extensively trapped.

In this age of live drum sounds, this room is ideal for the *very live*, "cracking" drum sound that is popular today. The room is also well suited for acoustical instruments, such as guitars, strings and horns. Because of the live ambience it is also becoming a popular vocal room.

The room's sliding doors play a great role in the acoustic nature of the main studio, producing drastic changes depending on whether the doors are opened or closed. A heavy velvet curtain can be drawn on the main studio side of the glass to dampen reverberance. The live room also makes an excellent live reverb chamber during mixdown, simply by feeding signal through the studio playback speaker and remicing it. For very large string sections or live orchestral work, all doors between the rooms can be opened to create an open recording environment.

At the other end of the technology spectrum is Studio B, our 8-track MIDI room. Obviously much smaller than Studio A, B is designed as a pre-production or writer's suite, with audio tie lines into the main control room.

Operating philosophy

Our philosophy at Red Zone is a service-oriented one. We wanted to offer an atmosphere of creative relaxation, where one can have all of the advantages of big city life without the inherent aggravation. Because some of today's new artists will become future stars, we want to develop long-term relationships with our clients. While interviewing pro-

spective employees, we looked for a positive, friendly and helpful attitude. Technical knowledge can be taught, but changing one's nature or personality is not so easy.

Running the business

As with any business venture, one of the most difficult things to pull together is the financing. The words "start-up company" may sound exciting and romantic on the financial page when a new company makes good, but they strike a note of terror to banks and leasing companies. Convincing lenders that you are going to survive in what is perhaps this country's most competitive studio market is no simple task.

After securing our building lease and initial equipment order with several large deposits, we set out to find a leasing company. Our reasons for using a leasing company instead of a bank were 2-fold: the lease payments can be written off as monthly expenses; and leasing companies are generally faster than banks. Unfortunately, unlike banks, leasing companies are not regulated by the government, so it's a case of buyer beware.

As with any business venture, one of the most difficult things to pull together is the financing.

We initially approached a leasing company and secured a contract that promised funding within one week, and we gave the company a good-faith deposit. After two months, the company still had not delivered. After intense negotiations, our funds were finally wired back into our account.

Others who have worked with this company were not quite so lucky, however. The good news is that there *are* honest and effective leasing companies. And we found one.

Before you enter into an agreement of any kind, be absolutely sure of the terms that are proposed. Check out the company with other clients, the Better Business Bureau, and law enforcement and consumer protection offices. Don't let your desires rush you. Read and *understand everything*, and have anything you don't fully understand explained, before you sign.

Now with our funding situation solved, Red Zone can get back to its original concept: creating *one* studio where many styles of great music can be recorded.

REP

All photographs by Elizabeth Annas/Photosensations

Control room equipment

Recording console: 36x24x36 Amek Angela.

Multitrack machine: Otari MX-80 24-track prewired for 32-track headblock and electronics

Mastering machines: Otari MT-12 and MX-5050B 2-tracks.

Monitoring: Tannoy FSM cabinets in wall-mounted soffits driven by Crest power amplifiers; Auratone T-6 and Yamaha NS10M on console for close-field monitor duties.

Special effects: Yamaha SPX-90 digital processor; Eventide SP-2016 digital reverb and special effects; Roland SRV-2000 digital effects;

Lexicon PCM-42 digital effects; Eventide H910 Harmonizer; two ADA 2.56i digital delays.

Signal processors: Two Valley People DynaMite noise gates; two Draumer DS-201 dual gates; two dbx model 160X compressor limiters; UREI LA-4 limiter; two UREI LN-1176 limiters.

Instruments: 8-foot Kawai grand with MIDI Forte mods; Sequential Circuits Prophet T-8; Oberheim Xpander; Korg EX-800; two Ensoniq Mirage sampling keyboards; Akai S-612 sampler; Moog Taurus II bass pedals.

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NAMM Summer Expo Replay

By Paul D. Lehrman

Spotlighting some of the more innovative MIDI-capable synthesizers, sequencers, effects and automation systems at the recent Chicago NAMM exposition.



Akai EW11000 MIDI-capable woodwind simulator.

Compared with the NAMM show in Anaheim last January, the Summer Expo held at Chicago's McCormick Place during the last weekend in June was some-

Paul D. Lehrman is a Boston-based free-lance writer, electronic musician, synthesist, producer and regular contributor to RE/P.

thing of a sedate affair. For a change, exhibitors took sound levels very seriously. The result was that for the first time in recent memory, you could have a reasonable conversation at a NAMM show.

There was some grumbling about this being the last summer show for some companies—one show a year was plenty, some considered—and indeed several major exhibitors already chose not to attend. Such rumors notwithstanding, June's Chicago Expo had much to offer.

Sound samplers

Samplers made the biggest noise. New models were unveiled in the form of keyboards and rackmounts, with and without drum machines or sequencers attached. After a long climb, in which it has seemed that each new bit was going to be twice as expensive as the last, 16-bit sampling technology has finally become the order of the day.

E-mu Systems showed a prototype of the Emulator Three, which is said to comprise a complete digital effects processor and post-production workstation in one package. The EIII features 31kHz and 44.1kHz stereo sampling with direct digital access for CD ROM or even WORM disk drives (when they become available) through a built-in SCSI port; 16-voice polyphony; an internal 40Mbyte hard disk drive; and up to 8Mbytes of RAM. Digidesign is developing software for the unit to run on an Apple Macintosh II, which promises to include modules for doing effects generation, sequencing and mixing. The unit will be available by the

end of the year, E-mu says.

Although the company will be discontinuing the Emulator II, it will continue to produce the E-Max, for which a 20Mbyte internal hard disk is now available. Also discontinued is the SP-12 sampling drum machine, to be replaced by the SP-1200, with built-in disk drive, 10 seconds of sample memory and improved software.

Sequential Circuits, which has not displayed at summer NAMM shows for several years, showed a new high-end sampler at a downtown Chicago hotel. The Prophet 3000 can sample in stereo at 32kHz, 44.1kHz and 48kHz, and has 2Mbytes of RAM, expandable to 8Mbytes. Eight-voice polyphony is standard, with another eight voices available through an optional expander module. Storage is handled by 3½-inch, 2Mbyte floppy drive, while a SCSI interface is provided for external hard-disk storage.

The unit has extensive built-in editing and sound-creation capabilities. It comes with a remote controller featuring an 8-line-by-40-character LCD display, which can be used to show sampled waveforms during editing or relative harmonics in an additive synthesis mode. Three envelopes and nine modulation sources are available per voice.

Even though Sequential plans to ship the Prophet 3000 in August, it showed none of the unit's operational software, but merely sampled sound from a CD and played it back, asking listeners to compare. One spokesman explained this was because SCI had encountered problems in the past with competitors "borrowing" its ideas before new products



Artisyn SX-01 Midisax wind controller.



Simmons SDX digital drum machine.



Akai/Linn ADR15 combination drum machine and MIDI sequencer.



Yamaha SPX90II digital multi-effects processor

were ready to ship.

Up at the high end, **Fairlight** was showing its expandable CMI Series III. The basic, entry-level unit includes two 20Mbyte Bernoulli drives, 4Mbytes of RAM and eight voices, while the largest format (the "Super System," which won't be ready until early next year) features 40 voices, 50Mbytes of RAM, 600Mbytes of hard-disk storage, and a 400Mbyte WORM drive. To make the system expandable, all of the CMI components now come in rack-mountable units. This configuration has the added bonus of making the control interfaces of all the models identical—your clients will never know how much you've really spent on the system!

Forat Electronics, a company that has taken over product support of old Linn Electronics products (which we'll get to below), introduced its own sampler, the F16, which provides eight voices of 16-bit sampling at a rate of up

to 60kHz. Samples can be replayed from audio triggers, MIDI velocity information or MIDI notes.

Each voice module contains 512Kbytes of RAM, and the number of modules is easily expanded for very long sampling times. A 730Kbyte floppy drive is included. Some editing functions are provided, although external software using the MIDI Sample Dump Standard is planned for more sophisticated editing.

Back in the 12-bit world, **Roland** showed two new sampling units: the S-550, which is a rackmount version of its S-50 equipped with extra memory; and the S-220, which is file-compatible with the S-10. In addition, two new software packages were shown for the S-50: a new operating system, which allows polyphonic, multitimbral operation through MIDI; and a sequencing program that uses the S-50's internal computer to drive its own voices or any MIDI instrument.

Korg also showed a rackmount sampler, the DSM-1, which is disk-compatible with the company's DSS-1. The DSM-1 features discrete outputs for each of its 16 voices, a 2Mbyte floppy drive plus built-in software for sample editing and additive synthesis.

Drum machines

There were a number of samplers on display at NAMM that were specifically designed as drum machines. One was the **Simmons SDX** system, which provides 16-bit sampling at 44.1kHz; each played sound can consist of up to nine samples. A built-in monochrome screen and track ball allows samples to be edited for length, threshold, loops, envelopes, filtering and pitch.

A software-based mixer is provided for control over the unit's 16 outputs, and 16 different "kits" can be stored in internal memory. In addition, a new type of drum pad has been developed which senses where it has been hit, and uses the information to control various parameters.

Another sampler-cum-drum machine is the result of the long-anticipated collaboration between **Akai** and **Roger Linn**. The ADR15 uses an "enhanced" 12-bit format at a 40kHz sampling rate, with 13 seconds of sample memory as a standard feature, expandable to 26 seconds. A total of 32 samples are available at a time, 16 of them playing simultaneously, and eight audio outputs are provided.

The unit features pads that are both velocity and pressure sensitive. When a pad is held down for repeated notes, the pressure translates into velocity. Three hi-hat samples with different decays can be used, and a slider allows you to fade between them in real time for a more realistic effect. A stereo drum mixer and an echo send with its own mixer are part of the unit's software, and an 8x40 character LCD display facilitates editing.

It wouldn't be a Linn design if the ADR15 didn't have a built-in sequencer, and this one has 60,000-note capacity, direct SMPTE time code and MIDI Time Code sync, four MIDI outputs and two inputs, and the usual looping, time-shifting, step editing and quantizing features.

Staying with the Linn connection, **Forat Electronics**, mentioned above, showed new software and hardware for the Linn 9000, including memory expansions, sampling cards, custom sound chip burning and time code interfacing, as well as a full-featured MIDI retrofit for the old LinnDrum.

There were also several non-sampling drum machines, including the first musical instrument from **Alesis**, the HR-16. Following the philosophy of the "most-bang-for-the-buck" exemplified by its line of inexpensive digital processors, the

company has crammed a huge number of features into the package, including 48 tunable 16-bit/47kHz samples (including 10 kicks and nine snares); velocity-sensitive pads; and software-based mixing with four outputs.

Although the folks doing the demo were a little over-enthusiastic about the source of the samples ("This is a custom-made, 13-inch 3-ply rosewood snare with a 2cm vent hole and a silk handkerchief attached to the bottom head with Brazilian rubber cement..."), the unit did sound good. The company has also released a companion 8-track standalone MIDI sequencer, the MMT-8.

Yamaha showed the PTX8, which is basically a rackmount version of its powerful RX5 without the sequencing capability but with 26 internal sounds and space for 64 user-defined ones. **Roland** showed another entry at the low end, the TR-626, which boasts 30 tunable sounds (although different tunings cannot be assigned to different MIDI notes), 96 patterns, eight audio outputs and external storage on a memory card.

Production automation and synchronization

Following the Japanese MIDI Stan-

dards Committee's long-awaited adoption of MIDI Time Code, devices that read and/or write the standard are finally beginning to appear. The PPS-1 ("Poor People's SMPTE") from **J.L. Cooper Electronics** converts time code to MIDI TC, and has the added bonus of generating "intelligent" tape sync, using a proprietary form of FSK that contains location information. The unit can then read back the sync data and translate it into standard MIDI clocks and pointers.

Opcode Systems' Timecode Machine is designed to convert time code to MIDI TC, with more features to be added. **Sonus** showed its MS120, which will read, write and regenerate time code in any of the four formats, convert it to MIDI TC, generate click tracks and lock a sequencer to it and act as a MIDI merge box.

Digidesign showed the Q-Sheet MIDI TC-based production and automation software, synced up to a Sequential Circuits Studio 440 and a Yamaha DMP-7 digital mixer, while **Opcode Systems** demonstrated its similar program, "Cue," which can now exchange tempo maps with a Roland SBX-80 Sync Box time code-to-MIDI convertor.

Other manufacturers, however, held back on implementing MIDI Time Code.



SSSound Sting EW2 MIDI-capable electronic woodwind controller.



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Notable among these is Yamaha, although one company spokesman admitted that MIDI Time Code is "where things are going." The Yamaha MSS-1, due out in September, converts time code to MIDI clocks and pointers, using an internal tempo map with up to 150 tempo events. Tempo changes can be programmed by hand, or the unit will record tempo changes being generated by another sequencer and then replay them.

The MSS-1 can also be programmed to send up to 2,000 other MIDI events—controller or program changes—at prescribed times, for mixing and processing chores. A tap tempo feature is included, and the unit's internal memory can be stored on cartridges or dumped through MIDI System Exclusive.

Tascam showed the new ES-50/51 synchronizer and transport controller. Both units can read and generate time

code using internal or external sync, and claim a synchronization accuracy of 50µs. Wideband circuitry allows time code to be read at 1/50 to 100 times normal speed, and the system can learn the protocol and ballistics for any tape transport—using either dc or FM servo control—in about 90 seconds. Each controller has four rely circuits with individual time code addresses, which can be set to operate in momentary or latching modes. Up to three slave units can be added.

At the Fostex booth, small but significant changes were in evidence. The model 4050 MIDI-to-time code converter and autolocator now offers a new feature whereby the unit's 10 memory banks can be chained into one performance. Interfaces are available to link the model 4030/4035 synchronizer/controller to Sony PCM-3324 digital 24-tracks, as well as Studer A810 mastering decks with

center time code tracks. The E16 1/2-inch 16-track deck is now available with a separate playback head with its own electronics, for easier alignment and for reliable verification on live shoots.

Japanese companies showing at NAMM are taking mixing automation very seriously. As well as the previously released Yamaha DMP-7 and Akai MPX-820, TOA showed a prototype of a digital mixing and processing system based around a Macintosh computer and the company's X2 4-channel digital processor. The software being shown was for designing speaker crossovers, with four bands adjustable as to frequency, filter type and delay. Other modules are also being developed.

Yamaha displayed prototype software being developed by Digital Music Systems for the DMP-7 and the Macintosh. The new software features graphics-based controller grouping, in which the controllers can be linked linearly or scaled to each other, and complete library functions.

Several manufacturers showed new MIDI-controllable processing devices. The Korg DRV-3000 reverb and effects processor features a 94dB dynamic range, 20kHz bandwidth and true stereo circuitry with four outputs. Two different effects can be used in parallel. A pitch shifting program allows MIDI note information to determine the degree of shift and the unit reads MIDI program changes.

Yamaha had three new devices on hand. The REV-5, which replaces the REV-7, features an additional digital processor for smoother sounding reverb, and multiple effects in parallel and/or series. A 3-band parametric digital EQ is available for each program. Bandwidth has been increased to 20kHz, and several SPX-90-style effects have been added. The SPX-90 itself has been upgraded, and the new SPX-90ii has four times as much memory, and a wider range for the sweep oscillator.

The REX-50 from Yamaha is an effects processor for the "lower end" performer or studio, with 30 presets and 60 user-definable memories. Programs are available for reverb, flanging, pitch change and, for the first time in a digital device, distortion. Pitch change is MIDI-note controllable, and effects can be switched with MIDI program changes.

ADA showed a MIDI retrofit that will give program-change control to its Pitchtraq pitch transposer and Digitizer-4 digital delay. Akai had two interesting devices on hand; the MB76 programmable "mix bay," which allows signals from seven audio inputs to be assigned, each with its own trim control, to any of six outputs, with 32 stored setups available

Continued on page 66

Need more information?

For additional information on the products mentioned in this article, circle the appropriate numbers on the Rapid Facts Card at the back of this issue.

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Eight Stand-alone Microphone Pre-Amplifiers

Part One: Technical Descriptions and Measurements

By Bob Hodas and Paul Stubblebine

In the beginning, man said, "Let there be recording" and he created mic pre-amps and they were good. Well, maybe OK would be more appropriate. And so man said let there be EQ, compressors, gates and auxiliary sends and thus consoles grew, and were praised. And, many years later, some men yearned for the good old days and re-invented the pre-amp, and said, "Let it be good." And it was.

With the advent of the console that was *everything* in line—with the possible exception of a candy machine—some members of the recording community felt that going as "direct-to-tape" as possible gave them a better signal than passing through a vast number of op-amps and circuits. Going straight from console pre-amp outputs to tape machine inputs at the patchbay avoids bus circuits, EQ and whatever else was not needed for the signal on that particular tape track. The technique worked well for getting a very clean signal on tape—as well as full, pure sound—leaving lots of flexibility for extra EQ in the mix. Tones were manipulated by microphone choice and placement during the recording process.

But this is not the only reason why

Bob Hodas is RE/P's evaluations and practices consulting editor, and Paul Stubblebine is a San Francisco-based recording and mastering engineer.

- Benchmark Media 4x4
- GML model 8300
- HART MP-501 PC card
- Innovative Audio UTMP-2
- Jensen Twin Servo 990
- Sontec MB-1A
- Studio Technologies Mic Pre Eminence
- Summit Audio TPA2

stand-alone mic preamps are becoming more popular. The age of sampling has made it necessary for many of us to record samples or sound effects in remote locations. We do not necessarily need multichannel, full-feature consoles to do our sampling, and many of us are learning that clean, quiet, non-equalized samples offer the greatest latitude for manipulation. At most, only two channels are needed for this type of operation; stereo pre-amps fit the bill perfectly.

There also seems to be a growing group of engineers doing stereo-pair recording, be it with spaced omnis or MS arrays, etc: for such sessions a straight-through, high-quality signal path is desirable. Ambience is also playing a larger part in modern music, and I know of several engineers who have moved stylistically from close-mic to the MS technique, coincident pairs, or spaced

omnis for recording horn, string and harmony sections. For such people a pair of high-quality, low-noise pre-amps is ideal.

Evaluation sessions

This article came about because we thought it would be interesting to audition some of these pre-amps for our own purposes. We discovered that there were, surprisingly, a number of stand-alone mic pre-amps currently in production, and a couple more in the planning stages. After finding out how many pre-amps were actually on the market, we thought that it would be a good idea to have a number of engineers benefit from our efforts to collect together the necessary equipment. We contacted 17 San Francisco Bay-area engineers, who expressed a genuine desire to participate in a listening test. The response was encouraging, and we proceeded with the program.

The units tested were the Benchmark 4x4 GML model 8300, Innovative Audio UTMP-2, Jensen Twin Servo 990 (manufactured for Jensen by Boulder Amplifiers), Sontec MB-1A, Studio Technologies Mic Pre Eminence, Summit Audio TPA2 and a custom-designed MP-501 card produced by Harvey A. Rubens Technology (HART). The latter is a free-standing PC card originally intended for installation in the high-voltage environment of Sony

MCI series 500 consoles. For our evaluation sessions, the card was built into a suitable enclosure and powered from a bipolar 32V PSU.

Apart from the MP-501, all pre-amps were production models supplied directly by their respective manufacturer. (An exception was the GML model 8300, which was supplied by a local owner. Apparently there was no stock on this pre-amp at the time of our test.)

We asked several other manufacturers to supply units, but either they had no stock or the units were not yet in production. Companies that reportedly will have pre-amps available in the near future include Cello, API, Hardy Company, Sound Research and Audio Engineering Associates.

The pre-amps fit into three basic groups. The Summit Audio TPA2 and Innovative Audio UTMP-2 are based tube technology. The Jensen Twin Servo 990 uses input and output transformers, while the balance of pre-amps are transformerless. Let's take a brief look at each

of the pre-amps.

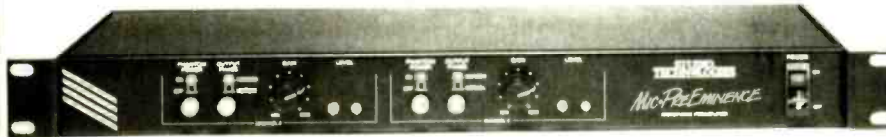
- The *Benchmark 4x4* is a 1U (single-space), rackmount unit containing four mic pre-amp channels. Each channel features a continuously variable gain control knob, as well as a 20dB pad and phantom power switch. There are no peak LEDs provided on the channels. The unit has detachable rackmount ears and uses an independent power supply. Access is via XLR-type connectors on the back panel. Outputs are balanced.

Benchmark offers a couple of options that were incorporated into our test model. One modification allows channels 1 and 3 and 2 and 4 to be summed to two XLR outputs on the back panel, thereby turning the 4x4 into a miniature mixer. A front-panel headphone jack with gain control pot may also be added to monitor these summed channels. Incidentally, the company plans to add a 2-channel pre-amp to the line; the pre-amp card is also available on an OEM basis.

- The *GML model 8300* occupies 1U and



Benchmark Media Systems 4x4 mic pre-amplifier.



Studio Technologies Mic Pre Eminence mic pre-amplifier.



Jensen Twin Servo 990 mic pre-amplifier, manufactured under license by Boulder Amplifiers.

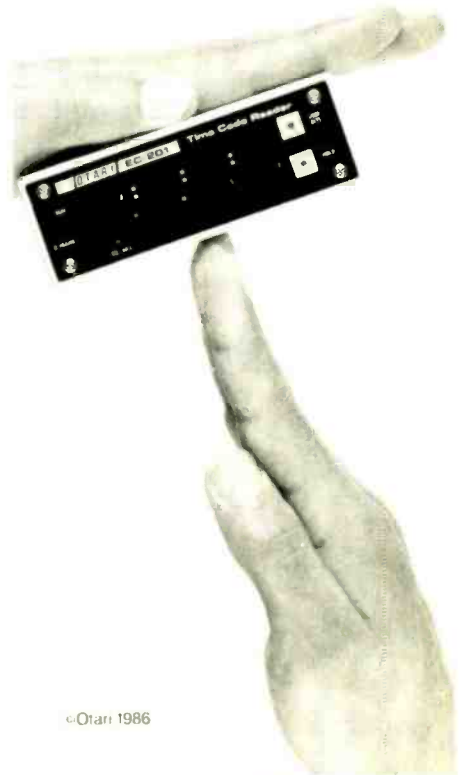
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Circle (33) on Rapid Facts Card

Table 1. General features of evaluated microphone pre-amplifiers.

Unit	Pad?	Polarity Switch?	Phantom Switch?	Gain Adjustment?	Peak LED?	Hot Pin-out	Power Supply
Benchmark Media 4x4	Yes	No	Yes	Variable	No	3	External
GML model 8300	No	Yes	Yes	Step	Yes	2	External
Hart MP-501	No	No	Yes	Variable	No	2	External
Innovative Audio UTMP-2	No	No	Yes	Step and Variable ³	No	3	Internal
Jensen Twin Servo 990	No	Yes	Yes	Variable	Yes	3	Internal
Sontec MB-1A	No	Yes ¹	No ²	Step	Yes	2	External
Studio Technologies Mic Pre Eminence	No	Yes	Yes	Variable	Yes	3	Internal
Summit Audio TPA2	Yes	No	Yes ⁴	Variable ³	No	3	Internal

Footnotes:
 1. Right-channel only.
 2. Selectable from inside unit only.
 3. Two-stage gain control.
 4. Both channels simultaneously.

Table 2. Specifications of evaluated microphone pre-amplifiers.

Unit	THD ¹ (%)	SMPTE ² (%)	Noise ² (dB)	Flat Response (Hz)	Down at 100Hz (dB)	Minimum gain (dB)	Maximum gain (dB)
Benchmark	≤0.01	0.01	-91	10-30k	0.9	14.5	72
GML	≤0.01	0.002	-89	10-100k	-	12.7	67
Hart	≤0.01	0.002	-94	10-100k	-	2.85	54
Innovative ³	≤0.1	>1.0	-87	10-20k	4.5	0.0	60
Jensen	≤0.01	0.002	-93	10-65k	0.6	15.0	59
Sontec	≤0.01	0.01	-91	10-100k	-	9.75	64
Studio Technology	≤0.005	0.002	-95	10-40k	0.8	14.0	65
Summit ³	0.5	>1.0	-80	10-10k	8.0	0.0	70

Footnotes:
 1. Load 50Ω; termination 600Ω.
 2. dBu, A-weighted; load 150Ω; termination 100kΩ.
 3. Best case.

is available as a 2- or 4-channel model; two units may be driven off the same independent power supply, allowing up to eight channels of mic pre-amplification. The front panel contains only the stepped gain pot and peak LED for each channel. The back panel holds a phantom power and polarity reversal switch, plus XLR-type inputs and unbalanced output connectors.

- The *Innovative Audio UTMP-2* is a 2U tube pre-amp that will handle mic- or instrument-level inputs. Each channel features a 3-position stepped input gain control (30/40/50dB) and a variable gain output pot. A switch allows the user to use either the front-panel phone jack for instrument-level inputs or the rear-panel XLR-type mic inputs.



Sontec MB-1A mic pre-amplifier.

Each channel features an LED peak indicator, but there is no provision for polarity reversal. Phantom power with

LED indicator is independently switchable on each channel. The unit contains a Jensen output transformer, with an output transformer being offered as an option. Outputs are unbalanced.

- The *Jensen Twin Servo 990* is a 2U unit that, with an expansion chassis, can be configured as a half rack or full rack model with a minimum of two channels and a maximum of 12 channels. It is also available as a table-top unit. I must admit that the *Twin Servo 990* was by far the most impressive piece of construction in the review batch. It was elegant, solid and all mechanical functions had a very good feel.

Each channel includes a peak-reading LED, variable gain pot, phantom power switch with LED, polarity reverse with

LED, and a microphone source impedance switch with LED that selects either 20Ω or 150Ω. Rear panel inputs and balanced outputs are via XLR-type connectors.

- The *Sontec MB-1A* is a 2-channel, table-top unit with a plastic casing. Each channel uses stepped gain pots and features signal and peak LEDs. A polarity reversal switch affects only the right channel, while an EQ switch activates an optional equalization circuit. Phantom power is selectable only by taking the unit apart and moving an internal connector. Rear-panel connectors are XLR-type and the power supply is independent. Outputs are unbalanced.

- The *Studio Technologies Mic Pre Eminence* occupies 1U of standard rack space and contains two channels. Each channel has an infinitely variable gain pot, signal and peak LEDs, polarity reversal switch and phantom power switch. Rear-panel inputs and outputs are via XLR-type connectors. A rear-panel switch selects balanced or unbalanced operation.

- The *Summit Audio TPA2* is 2U high and features dual-stage variable-gain control pots for each of its two channels. It will accept instrument-level inputs ac-

cessed by the front-panel phone jacks, and mic-level by rear-panel XLRs. Each channel features a switchable pad, phantom power is switched on simultaneously to both channels. There are no function or peak indicators on this unit. Outputs are balanced and a Jensen transformer is offered as an option.

Measurement results

Tables 1 and 2 provide comparison models. All measurements were performed with an Audio Precision System One test system connected to an IBM PC.

The pre-amps were all subject to the same test criteria. We conducted all measurements with 30dB of gain in the pre-amps; this value was selected as an average amount of gain one might use in a true recording situation. It may or may not be the most flattering for making measurements, but is certainly indicative of real life.

We chose to include frequency response information within Table 2—as opposed to providing graphs—because most of the units exhibited a flat response to at least 20kHz. The exceptions to this were the tube pre-amps, whose HF rolloff is not due to the use of tubes but rather to the dual-gain controls.

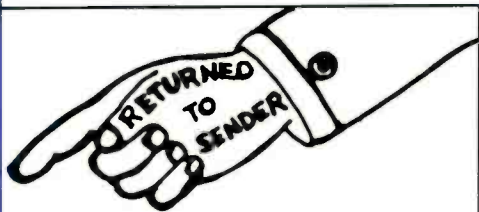


Summit Audio TPA2 tube mic pre-amplifier.



Innovative Audio UTMP-2 tube mic pre-amplifier.

As you will note from Figures 1 and 2, the response of the Summit Audio TPA2 and Innovative Audio UTMP-2 became less linear as input gain increased and output gain decreased. (This is a situation that can be likened to guitar-amp gain



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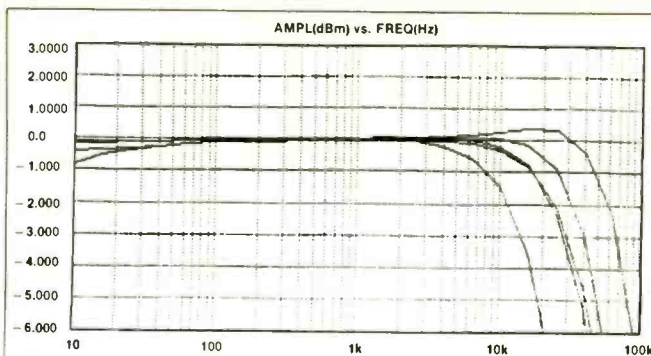


Figure 1. Frequency response of the Summit Audio TPA2 tube mic pre-amp at various gain ratios. Note that at elevated input gain the response begins to become less linear, and HF roll off begins at a progressively lower frequency.

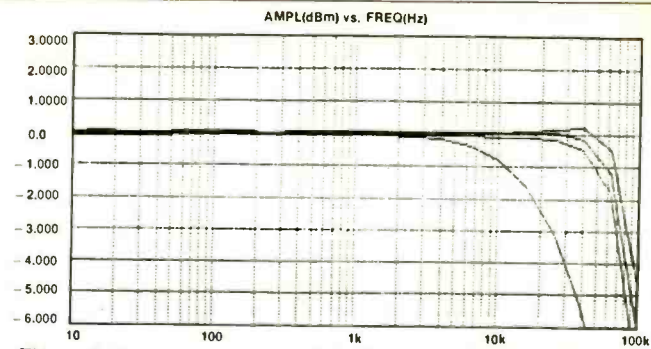


Figure 2. Frequency response of the Innovative Audio UTMP-2 tube mic pre-amp, which exhibits a similar gain-dependent frequency response as the unit measured in Figure 1.

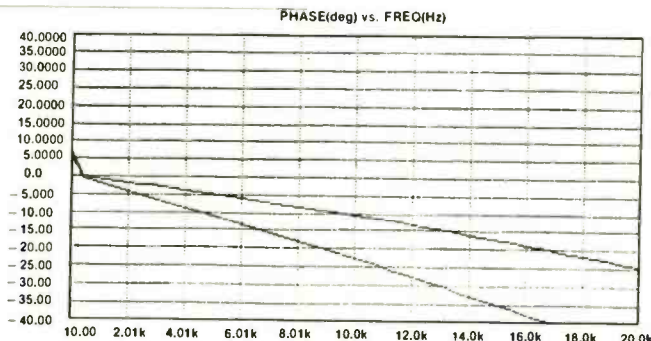


Figure 3. Linear phase response of the Summit Audio TPA2 tube mic pre-amp. The two traces were taken at different input gain settings. Note that the top trace for increased gain is curved, indicating a non-linear phase response with respect to frequency. The lower trace is the phase response for similar values of input and output gain. Note the extreme non-linearity at low frequencies.

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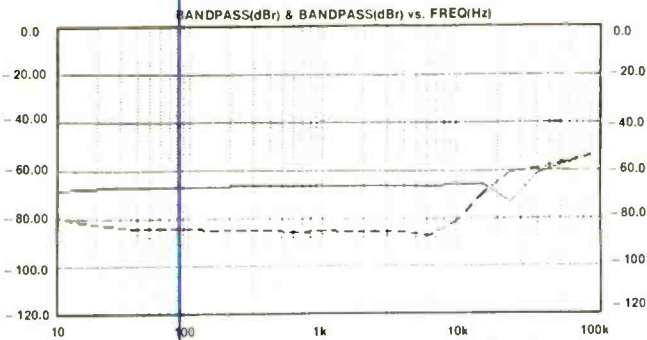


Figure 4. Common Mode Rejection Ratio for the Benchmark 4x4 microphone pre-amplifier. Note that, unless stated otherwise, in Figures 4 thru 11 the dashed line represents the CMRR of the right-hand channel, and the solid line the CMRR of the left-hand channel.

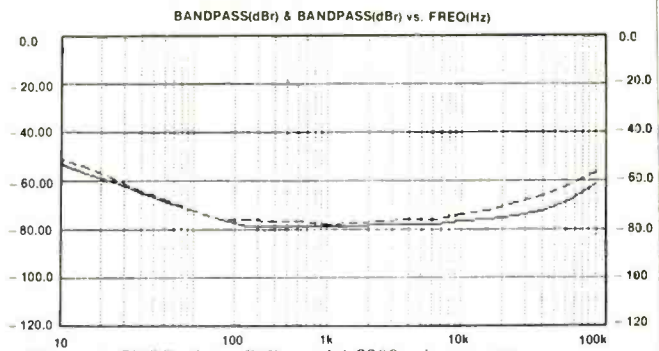


Figure 5. CMRR of the GML model 8300 mic pre-amp.

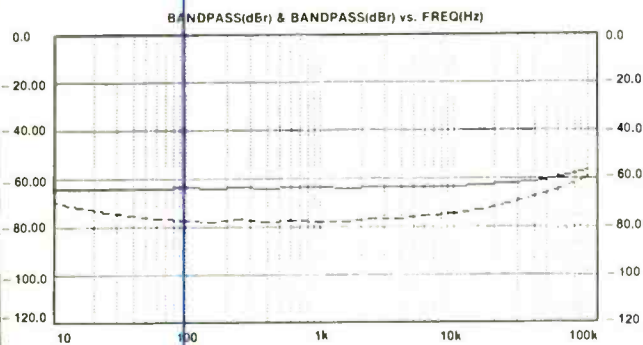


Figure 6. CMRR of the HART MP-501 PC card mic pre-amp.

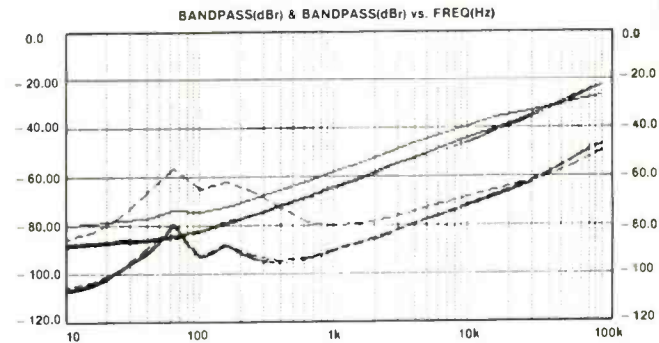


Figure 7. CMRR of the Innovative Audio UTMP-2 mic pre-amp. The two pairs of traces represent measurements taken at different gain settings.

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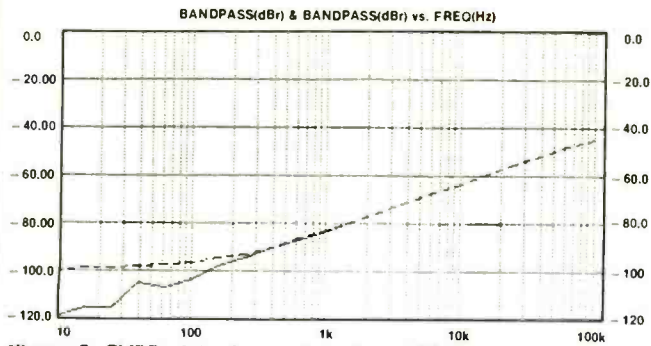


Figure 8. CMRR of the Jensen Twin Servo 990 mic pre-amp.

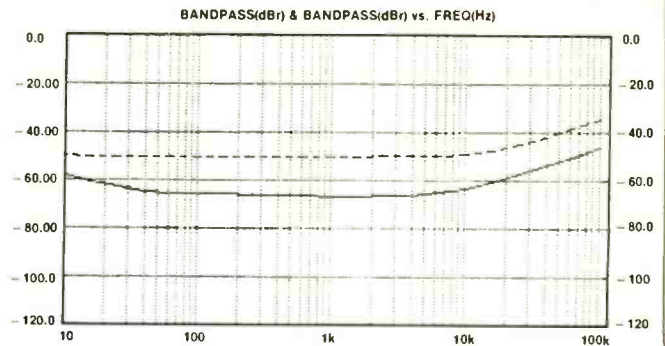


Figure 9. CMRR of the Sontec MB-1A mic pre-amp.

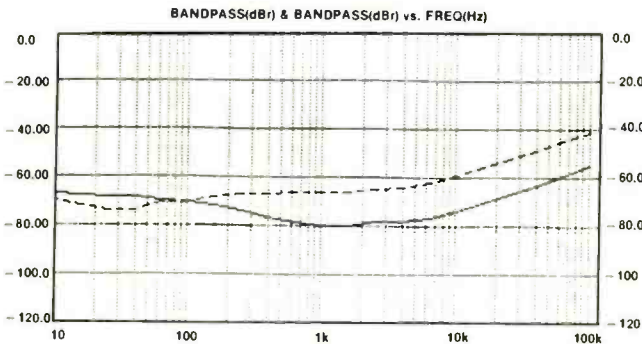


Figure 10. CMRR of the Studio Technologies Mic Pre Eminence mic pre-amp.

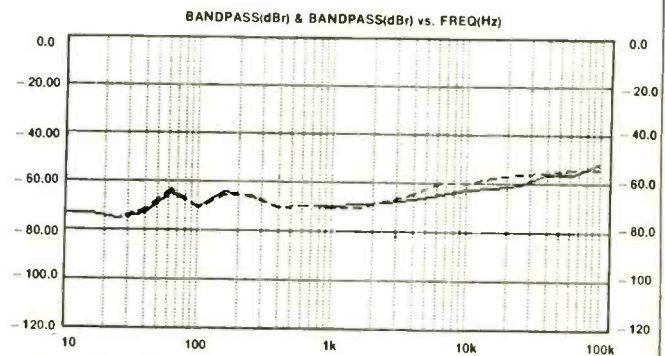


Figure 11. CMRR of the Summit Audio TPA1 mic pre-amp.



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staging for distortion purposes.) Best response is obtained by keeping the output gain at maximum and adjusting only the input gain.

The linear phase measurement represents pure delay; any curve or deviation from a straight line would indicate unequal delay over the frequency range—an undesirable feature. This measurement is most important to look for, regardless of the total number of degrees of shift. Because of time factors, however, group delay and deviation were not calculated; these factors may be even more revealing.

The graphs themselves are not that revealing, because everyone was fairly linear except at the lower frequencies. For this reason and because of space restrictions, they have not been included here; instead we will provide a verbal interpretation.

The Jensen Twin Servo 990 revealed the most linear phase response, exhibiting no non-linearity in the low-end. The next best group of pre-amps that exhibited less than four degrees of low-end shift included the GML model 8300, Sontec MB-1A, and Studio Technologies Mic Pre Eminence. The Summit Audio TPA2 and HART MP-501 PC card dis-

Need more information?

For further details on any of the eight microphone pre-amplifiers evaluated in this Hand-On review, circle the following numbers on the Rapid Facts Card located at the back of this month's issue:

Product	Number
Benchmark Media 4x4	143
GML Model 8300	144
HART Engineering MP-501	145
Innovative Audio UTMP-2	146
Jensen Twin Servo 990	147
Sontec MB-1A	148
Studio Technologies Mic Pre Eminence	149
Summit Audio TPA2	150

played about seven degrees of low-end shift, the Innovative Audio UTMP-2 about 10 degrees, and the Benchmark 4x4 about 13 degrees.

Turning to the tube pre-amps, phase measurements indicated similar results to the frequency response in that the

dual-gain stage relationship significantly worsened phase, as can be seen from Figure 3 for the Summit TPA2.

What was most interesting—or, should I say, surprising—were the Common Mode Rejection measurements. Figures 4 through 11 demonstrate how well the pre-amps reject signals that appear at both inputs (pins 2 and 3) in-phase at the same amplitude. (This unwanted signal could be RF or inductance into the mic line from some outside source.)

As can be seen from the CMRR data, only three of the units, the Jensen Twin Servo 990, GML model 8300, and Summit Audio TPA2, showed any signs of matching components in assembly, a fact that is demonstrated by the similarity in the curves between two channels. The rest of the manufacturers should take a lesson from these guys and pay attention to matching. If we buy a stereo pre-amp, it certainly should be a *priority* to have each channel sound the same, especially when doing stereo-pair recordings. **RED**

[Editors note: No photos were available of the HART MP-501 and the GML model 8300.]

Part 3 of this article, to be published in the October issue, will look at the results of listening tests conducted by Bay Area engineers.

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Continued from page 57

for recall by MIDI program changes; and the PEQ-6, which contains six channels of 7-band, fixed-frequency EQ with 32 setups accessed by MIDI.

The biggest surprise at the NAMM show was the simultaneous appearance of no less than six wind controllers for MIDI. Two of these were from Akai: the EV11000 and the EW11000.

The EV11000 is a new version of the Steinerphone synth controller for brass players. It features three "valve" keys, three trill keys, breath sensor for dynamics, lip sensors for glide, pitchbend and modulation controllers, touch sensitive keys and a thumb-roller system for a 7-octave range.

The EW11000 is a woodwind simulator, with 13 touch-sensitive keys, rollers and lip sensors for modulation control. Both instruments come with a special sound module, the EWV 2000, which contains two programmable monophonic voices with 64 memories.

Yamaha showed its WX7 controller, which has 14 height-adjustable keys, trill keys that allow half- or whole-tone trills on any note, a 7½-octave range and sen-

sors that respond to lower lip pressure for pitchbend effects. There is also a miniature pitchbend wheel. All other controls are onboard, either miniaturized or concealed, and the unit is powered by a battery pack that can be worn on a belt. A hold key allows a form of harmony playing, and the held and subsequent notes can be sent to two different MIDI channels.

Artisyn showed the SX-01 Midisax, which is specifically designed to feel like an alto saxophone, complete with crooked neck and mouthpiece ligature, although it hardly looks like one. Breath pressure can be routed separately to two different MIDI control parameters, as can two thumbwheels and an optional "bite" sensor for reed flexing (whose output can be sent to another MIDI channel for controlling a different device). There are 16 preset controller mappings and 10 user-definable ones, with all of the controls for defining and storing the settings on the unit. There's also an eye-level LED display to let the user know what's going on.

The Sting EW2, made in Japan by SSSound Co. Ltd., looks very much like a Lyricon, and seems to work very well, with a claimed response time of 2ms. Lip pressure can be mapped to pitchbend or

mod wheel or both, and wind pressure can be mapped to mod wheel, velocity or volume. A single cable, carrying both digital and analog information, and extendable up to 150 feet, connects the device with a rackmount interface.

Several pitch-to-MIDI converters were in evidence for singers and non-electronic instrumentalists. The most interesting had to be the Roland VP-70 voice and instrument processor. The VP-70 is both a pitch-to-MIDI converter and a pitch-shifter. The pitch tracking function includes adjustable pitchbend range (up to 24 semitones), octave transposition and key hold. The harmonizing function generates four voices from the incoming note, with a range of ±1 octave. The interval between the pitches is determined either by an internal setting (and there is memory for 128 such settings), or by incoming MIDI information.

The unit was being displayed at the NAMM gig by a gentleman who sang into a microphone and accompanied himself on the guitar. Except you didn't hear the guitar—instead you heard his voice singing the chords his fingers were playing. It should be lots of fun.

R-E-P

All photography by Paul Lehrman.



Tom Hannaford, Dixieland Productions, Atlanta, GA

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Studio Update

Northeast

Transcom Digital, (New York) post-production and production studio, now offers digital audio, video, film, sound and graphic design services.

The facility features a **Solid State Logic SL6000E** mixing console, **Sony 3324** multitrack recorder and a **CompuSonics DSP 2002**. 902 Broadway, New York, NY 10010; 212-529-1000.

Blank Tapes Recording Studios (New York) has added a **Solid State Logic SL 6056 E Series** console and **SSL Studio Computer** with Total Recall, 37 W. 20th St., New York, NY 10011; 212-255-5313.

Skyline Studios, (New York) has installed a second **Solid State Logic SL 4000 E Series** console at its studios. 36 W. 37th Street, New York, NY 10018; 212-594-7484.

Ear Force Sound (Greensburg, PA) has added **Christopher D. Armbrust** to their staff as senior engineer; **John E. Goughenour Jr.**, to lead engineer; and **Ronald A. Beltz** as engineer. 201 Beacon St., Greensburg, PA 15601; 412-834-4100.

Sync Sound (New York), a post-production facility for film and video, has purchased an **AMS AudioFile** digital system. 450 W. 56th St., New York, NY 10019; 212-246-5580.

Vogt Quality Recordings (Needham, MA) and its new Compact Disc label and production division, **VQR Digital**, has installed the **Sony** Compact Disc mastering system, including the **Sony 1630** digital processor, two **DMR 4000** digital master recorders, **DAE 1100A** digital editor and a **DTA 2000** digital tape ana-



Tom Brown, chief engineer and Hank Williams, studio manager of Master Mix Studios, Nashville, TN.

lyzer. The **Lexicon 480** digital effects system completes the digital mastering configuration. In-studio digital-direct-to-disc is still available with **Scully** automatic variable pitch lathe with the **Haeco SC-2** cutterhead and custom **Haeco** amplifiers via U-matic or Beta digital masters. P.O. Box 302, Needham, MA 02192; 617-444-8687.

Great Immediately Recording (New York) has just installed a complete rack of **Dolby SR's** for its **Ampex MM-1200** 24-track machine. 423 W. 22nd St., New York, NY 10011; 212-206-8561.

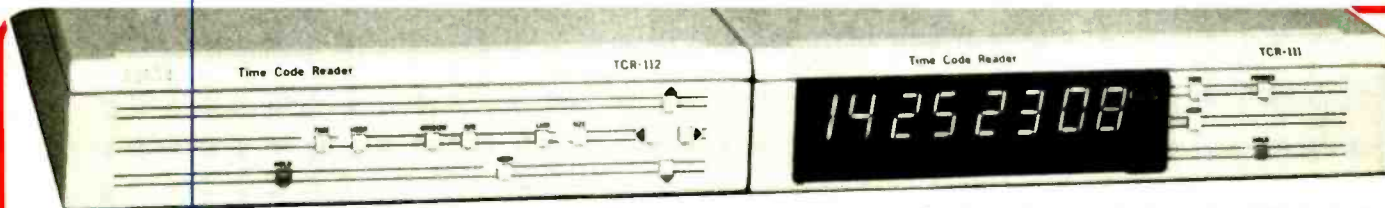
Southeast

Master Mix Studios (Nashville) announces the installation of a new **CAL-REC UA 8000 console**. 1808 Division St., Nashville, TN 37203; 615-321-5970.

Disc Mastering (Nashville) recently took delivery of a **Studer D820X**

2-channel **DASH** format digital recorder and a **Neve DTC-1** console for Compact Disc mastering. 27 Music Square East, Nashville, TN 37203; 617-254-8825.

The Mobile Control Room (MCR) (Nashville) is a master on-site audio facility that can interact with all levels of video and broadcast production. The MCR 22-foot trailer features a customized **Quad-8** console, a 19" **Proton** color monitor, **Nakamichi DMP-100** digital processor with a **Sony SL-2000 VCR**, **Lexicon PCM-60** digital reverb, **Valley People 610** limiter/compressor/expander, **MXR** digital delay, **ATI 4** output stereo distribution amplifier, **Electro-Voice Interface "A"** and **TOA 265-ME** monitor speakers. Custom 24 mic input mic splitter with **Jensen MBE-3** transformers at each input and isolated tech power also are featured. P.O. Box 150526 Nashville, TN 37215; 615-383-7015.



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Studio Update



Waring FDS Digital Mastering Lab features a second digital mastering room.

Midwest

Seagrape Recording Studios (Chicago) have recently added a **MCI JH40** synchronizer. The SMPTE-based dual multi-track system locks its two MCI multi-track machines together for a total of 40 tracks. An **Otari MTR-10** recorder also has been added. 5740 N. Western Ave., Chicago, IL 60659; 312-784-0773.

Southwest

Digital Services Recording (Houston, TX) announces the addition of a **Fairlight CMI III**. Other instruments include the **Emulator II+**, **Yamaha TX-816** and **DX-711FD**, **SP-12** drum machine and **Apple Macintosh Plus** computer. 1001 River Oaks Bank Tower, 2001 Kirby Drive, Houston, TX 77019; 713-520-0201.

Southern California

The **University of Southern California** (Los Angeles) has installed a 56-input **Solid State Logic SL4000 E-Series** console in a new recording facility specially created to service its graduate certificate and proposed BS in Music Recording programs. Linked to the console, which is equipped with Total Recall automation, are **Mitsubishi X-850** 32-track and **X-86** 2-track digital machines, **UREI** monitors, power amplifiers and limiters, and **JBL** monitors.

To provide a home for the new hardware, USC has commissioned the design and construction of a new control room/class room that will provide a critical listening area for up to 25 students. The new control room will be video linked to three recording areas. One room will be used to record USC's sym-

phony orchestra and film scoring sessions, while the other two rooms will be used for choral and mid-sized instrumental groups, big bands, jazz and smaller fusion ensembles. "**Beno**" **May** of Tom May & Associates, Northridge, CA, and **Dale Adams** of USC Architectural Services are supervising the design and construction of the new control room. *University of Southern California, School of Music, Los Angeles, CA, 90089-0851; 213-743-2627.*

Discovery Systems announces the opening of its Los Angeles office. The new office will provide service for CD-Audio mastering and replication services. 225 W. Broadway, Suite 508, Glendale, CA 91204; 818-500-7267.

Secret Sound L.A. (Woodland Hills, CA) have taken delivery of a **Studer A820** 24-track digital tape machine. The facility also has purchased the first of the new generation **Studer** CD player. 4836 Queen Victoria Road, Woodland Hills, CA 91364; 818-999-6160.

Baby'O Recorders (Hollywood) announces the promotion of **Mark Stebbins** to chief engineer. 6525 Sunset Blvd., Hollywood, CA 90028; 213-464-1330.

Waring FDS Digital Mastering Lab (Los Angeles) has announced the opening of its second digital mastering room. The room features a **Sony DAE-1100**, **Sony DMR-2000**, **DMR-4000**, **BVU-800** and **VO-5800**, **Sony PCM 701**, **PCM 1610/1630**, **Sony DASH 3202** and a **Mitsubishi X-80**.

Also featured is a **Sony MCI** 2-track. 4007 W. 6th St., Los Angeles, CA 90020; 213-383-2155.

West Side Sound (Los Angeles) has expanded to 24 tracks with full MIDI set-up. New equipment includes **Soundcraft 2400** 28x24x24 console, **Soundcraft 760 MkII** 24-track, **Otari 5050** 2-track, **Tannoy** monitors, **Lexicon PCM 70** and **60**, **SPX90**, **Drawmer** gates, **Teletronix** and **dbx** compressors and **Neumann** and **AKG** mics. Synths include **dx7II**fd, **Akai s900** sampler and a **super-jupiter mks80**. 2265 Manning Ave., Los Angeles, CA 90064; 213-470-3437.

Take One Recording Studios (Burbank, CA) has added a second room to the facility. The new room will feature a **Mitsubishi Wester** console with 40-inputs, **Tannoy FSM** monitors, video lock up and the **Mitsubishi X850** 32-track digital tape recorder. 619B S. Glenwood Place, Burbank, CA 91506; 818-841-8697.

Northern California

Live Oak Studio's (Berkeley, CA) production facility has upgraded to 24-track. Recent installations include the **MCI-JH24** recorder and the **Sound Workshop 34C** 32x28x24 console.

The Attic, a personal-production room, is a multitrack electronic music facility. 1442A Walnut St., Suite 364, Berkeley, CA 94709; 415-540-0177.

Northwest

Performance Audio (Salt Lake City) has taken delivery of a **Soundcraft 500** 32x12 monitor desk.

JBL speakers, **Carver** amplifiers and **Mogami** cable also are featured. 2358 S. Main, Salt Lake City, UT 84115; 801-466-3196.

England

Air Studios (London) is upgrading the facility with four **Mitsubishi X-850** digital multitracks and two **X-86** 2-track machines. London.

Australia

Metropolis Audio Pty. Ltd. (Victoria, Australia) has completed construction of a live acoustic environment in Studio One. 180 Bank St., P.O. Box 296, Melbourne 3205 Victoria, Australia. **REP**

Please send Studio Update to Dan Torchia, staff editor, Intertec Publishing, P.O. Box 12901, Overland Park, KS 66212.

New Products

Furman Sound VU-40 stereo system monitor

The VU-40 features two 20 LED color-coded bargraph meters, calibrated in decibel standard VU meters. Rear panel switches for each channel select either line level or power output of a power amp. The "zero" meter positions may be set to either -10dBv or 0dBv for line or from 25W to 400W to 8Ω or 50W to 800W at 4Ω for power.

Circle (151) on Rapid Facts Card

New England Digital digital retrieval system

The new system is designed for the Audio Workstation to augment the video/film/post-production and Winchester hard disk or tape storage system. Users can store more than 5½ hours of high fidelity digital material on each optical disk.

Circle (152) on Rapid Facts Card

DDA and Klark-Teknik in-line console

The DCM 232 features a DDA console design by David Dearden with computer automation designed by John Austin and Terry Clarke of Klark-Teknik. This is the first joint venture between DDA and Klark-Teknik.

The console configures all inputs with dual signal architecture. The 56-channel console allows for up to 112 inputs in mixdown, and 32 output buses are provided.

Switching functions have been removed from modules. Auxiliary muting, pre/post switching, EQ in/out, insert in/out, phase reverse, and mute bus selection are controlled from the central panel.

Each channel module also has its own individual set of LED indicators, which display channel status.

A channel setup can be copied to any or all channels and an 8-way sub-grouping function is also provided.

The DCM 232 uses split equalizer design, which allows either section of the equalizer to be assigned to channel or mix signal paths.

The console features a quoted 100dB dynamic range and a minimum of +22dB headroom at all stages.

Circle (153) on Rapid Facts Card

Valley Commanders compressor/expander module

Designed as an addition to the 800 series, the module's proprietary symmetrical release coupling circuitry allows

the commander to compress the audio signal for dynamic range control.

Circle (177) on Rapid Facts Card

Stramp USA CP-1 mixdown automation system

The computer-based MIDI system automates up to 64 channels modifying without mixing equipment.

The system features up to eight 8-channel hardware units, the software package and the MIDI interface.

Computer mixes can be changed, modified, refined and compared through the use of the editing routines in the REALCON software program. These mixes can be stored and reloaded for playback.

Circle (154) on Rapid Facts Card

Audio Control Industrial SA-3050A analyzer

The third-octave real time analyzer features peak hold on the spectrum display, averaging of six non-volatile memories, full screen digital SPL, and peak hold on the screen digital SPL.

The unit features a 30-band display with a 270 dot matrix grid, resolution controls of 1dB, 2dB, 3dB and 4dB/step, and fast, medium and slow decay integration.

Other standard features include six non-volatile memories with read, write freeze, and RTA-memory comparison functions.

Circle (155) on Rapid Facts Card

Digidesign FX Designer software

FX Designer features MIDI patching capabilities and allows control of all aspects of the Lexicon PCM70. The software features copying of a MIDI patch from one effect patch to another and a direct tie in with the Opcode Systems PCM 70 Librarian with filing capabilities.

Over 200 patches are included in the program.

Circle (156) on Rapid Facts Card

Magnifax 3800 series tape duplicators

The 3800 series features two models.

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New Products

which are both in three slave configurations. The 3801 uses a 1/2-inch master and the 3802 features a 1/4-inch master format.

Other features include motorized tape wipers, individual audio and bias adjustments for each head, synchronized bias amps, digital metering and extended headroom amps.

Circle (157) on Rapid Facts Card

ANT E413 multitrack noise reduction system

The 24-channel E413 unit features a quoted dynamic range of 118dB. The design is based on ANT's C4E card, a single channel compander card that features 70 operational amps and eight VCAs in transistor form on one side, and 350 SMDs on the other.

Cooling fans have been eliminated in the design and each card has remote control capability for record and playback. Connection to the tape recorder is provided allowing the E413 to be adapted to any multitrack recorder.

Circle (158) on Rapid Facts Card

Otaritech TC-50 center channel time code/FM processor

Otaritech, Otari's new product line, announces the TC-50, the first of the line to be introduced. It is designed to retrofit to Otari 2-tracks such as the BII, MKIII-2 or the MX-5050 and add center-track time code.

Also provided is the capability to decode stereo pilot tone signals for resolving tape speeds.

Otaritech also will offer the OT-1M upgrade package, MTR, which assures gapless/seamless record punch-in and punch-out transitions with any Otari MTR-10 or MTR-12 recorder.

Circle (159) on Rapid Facts Card

Skotel TCG-80-005 time code generator

The unit is designed for film to tape transfer and to assist in producing video programs or commercials originally produced on film. Features include 240Hz bi-phase film tack reference, 16mm and 35mm film footage count and 24 or 30 frame/second film rates.

Circle (160) on Rapid Facts Card

Yamaha expands MZ series of mics

Four new models, the MZ205Be and MZ204 drum microphones and the MZ203Be and MZ106S vocal microphones have been added to the MZ series.

The MZ205Be, is designed for pickup of higher-pitched drums and the MZ204 offers the deep response for pickup of lower pitched drums.

The MZ203Be vocal mic features a 3-point suspension system.

Circle (161) on Rapid Facts Card

Yamaha MC series of mixing consoles

The series features the MC802, MC1202, and MC1602 console. Inputs are electronically balanced, providing both Hi-Z phone jacks and Lo-Z XLR inputs for both line and mic level input.

A channel insert send and return jack allows connection of a signal processing device to any input. Each channel also includes an equalizer with 15dB of cut or boost over the hi, mid and low bands.

Circle (162) on Rapid Facts Card

Shure 849-LC and 869-LC condenser mics

The 869-LC is a ball-type microphone designed primarily for vocal applications and the 849-LC is designed for instrument micing and recording.

Features of the 849-LC and 869-LC include a recessed on/off switch and shock mounting. Both models may be powered by standard phantom power sources or by a 1.5 volt AA battery. The 869-LC also features a spherical wind and pop filter.

Circle (163) on Rapid Facts Card

Wheatstone SP-6 stereo production console

The SP-6 console features stereo line input channels with machine control and remote module status ports. The mic channels include tally and remote on/off ports, and control of multiple studio and control room mutes, interrupts, tally and talkback functions.

Other features include automatic and manual timer modes, clocks and full function tape remotes. Each input channel employs an equalizer circuit, four auxiliary send controls and stereo send capability.

Circle (164) on Rapid Facts Card

REP

Please send New Product releases to the REP editorial and production offices. Editor, 1850 N. Whitley, Suite 220, Hollywood, CA 90028 and Dan Torchia, staff editor, Intertec Publishing, P.O. Box 12901, Overland Park, KS 66212.

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Circle (2) on Rapid Facts Card

NEW Studer A820: Back to the Future



The future of multi-track mastering was commonly assumed to be 100% digital. But now Studer has built a multi-track for the future...by going back to thoroughly refine and update analog technology.

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Some other manufacturers apparently assumed analog could not get significantly better. With the arrival of the A820, that's now a questionable assumption. Call your nearest Studer office for detailed directions back to the future. Studer Revox America, Inc., 1425 Elm Hill Pike, Nashville, TN 37210, (615) 254-5651

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STUDER REVOX

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