ISSN0011-7145

THE SOUND ENGINEERING MAGAZINE

AUGUST 1979

$1.95

db

Let's Talk About AUTO·SET

Auto-Set is a complete self-contained automation programmer made by Harrison Systems. It is specifically designed to interface with all Harrison consoles ever built. Interface is quick and simple through the use of multipin connectors.

Auto-Set is a simple piece of equipment to operate. It can operate as simply and with as little facility as the other available automation systems. All that is required is the pressing of one button.

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WHY DOES AUTO-SET LOOK LIKE A COMPUTER?

Because it is a computer. Auto-Set contains two Z-80 microprocessors operating under firmware control to make your life easier and your final product better. There is no need for you to be a computer programmer as all programs are supplied, safely locked away in non-volatile memory.

HOW DOES AUTO-SET STORE DATA?

It can store data in two different ways because Auto-Set is in reality two separate automation programmers living in one box.

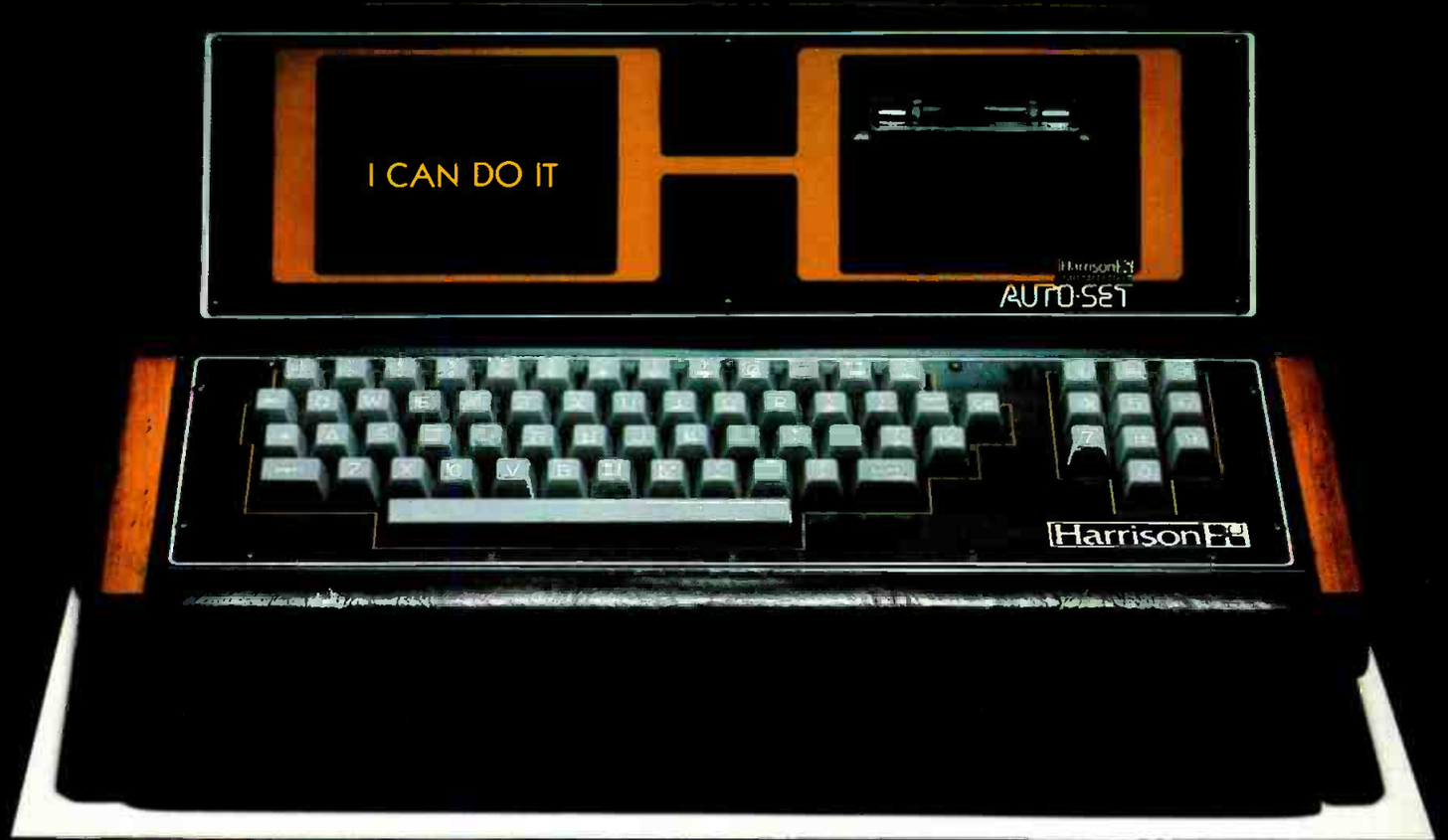
Normal Automation Data is stored on a standard studio tape machine. This is a very good way to store automation data as it is not necessary to purchase an expensive mass storage device and a time code system. Also, you never run out of data storage space as the data space increases with the length of material to be mixed.

Auto-Set allows the storage of four separate and independent mixes on one channel of a standard audio tape machine. Two channels of a machine must be allocated to automation for bouncing the data back and forth as a mix is built. Auto-Set allows you to independently select any one of the four mixes for each console fader. This selection process can be repeated 10 times for different parts of a mix and the selections recalled sequentially in a real time as the final mix is recorded.

Pre-Set Automation data is stored on special certified data cartridges which plug into the front of the Auto-Set. Preset automation is a totally different way of storing automation data. The previously described dynamic automation stores continuous changes in real time. Preset automation stores "snapshot" mixes only when you press a button, up to 630 of them on one data cartridge. Likewise, when you are ready to use the "snapshot" mixes they can be recalled one at a time with the simple push of a button.

The uses of preset automation are many and varied. It can be used for direct to disc recording using mixes taken in rehearsal and recalled by the mixer as needed. Also, it can be used in broadcast, live performance, theatre, or any other application where mixing is not from a prerecorded multitrack.

NOW, GO TALK TO THE OTHER GUYS.



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Coming Next Month

● Next month, we re-visit the world of broadcast audio, after concluding Douglas Dickey's two-part feature that begins in this month's issue. By way of contrast, Irv Joel will show us what goes into an automatic broadcast console. Then we'll visit WNCN-FM to see what a "hi-fi" station looks like. Plus a little something about tape cartridges—plus more, in next month's **db—The Sound Engineering Magazine.**

About The Cover

● As this month's feature articles point out, there's more than one way to build a "super console." But any way you get there, you'll still need some sort of data storage medium to keep track of things. To set the mood for August, Robert Wolsch takes a close look at some 3M data cartridges. For a somewhat more-conventional view of the same subject, see Dave Purple's feature story in this issue.

db

THE SOUND ENGINEERING MAGAZINE

AUGUST 1979 VOLUME 13, NUMBER 8

db is listed in **Current Contents: Engineering and Technology**

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db Calendar

SEPTEMBER

10-12 **Synergetic Audio Concepts Sound Engineering Seminars: Minneapolis**

17-19 St. Louis

For information on these three-day seminars contact: Synergetic Audio Concepts, P.O. Box 1134, Tustin, CA 92680. (714) 838-2288.

13-16 **Audio Expo 80,** Civic Center. Philadelphia, Pennsylvania

28 **Seventh Regional Convention and Equipment Show,** co-sponsored by Chapters One and Two of the Society of Broadcast Engineers. Syracuse Hilton Inn, Buckley Road. Syracuse. New York

OCTOBER

7-10 **National Radio Broadcasters Association's 6th Annual Conference & Exposition,** Washington Hilton. Washington, D.C.

Synergetic Audio Concepts Sound Engineering Seminars:

9-11 Syracuse

17-19 Boston

30-11/1 Philadelphia

For information on these three-day seminars contact: Synergetic Audio Concepts. P.O. Box 1134, Tustin. CA 92680. (714) 838-2288.

21-26 **121st Technical Conference Equipment Exhibit of the Society of Motion Picture and Television Engineers (SMPTE).** Century Plaza Hotel. Los Angeles, CA.

23 **1979 Sound Business Show.** Ambassador Hotel. Los Angeles, CA.

NOVEMBER

Synergetic Audio Concepts Sound Engineering Seminars:

6-8 Nashville

14-16 Orlando

For information on these three-day seminars contact: Synergetic Audio Concepts. P.O. Box 1134, Tustin. CA 92680. (714) 838-2288.

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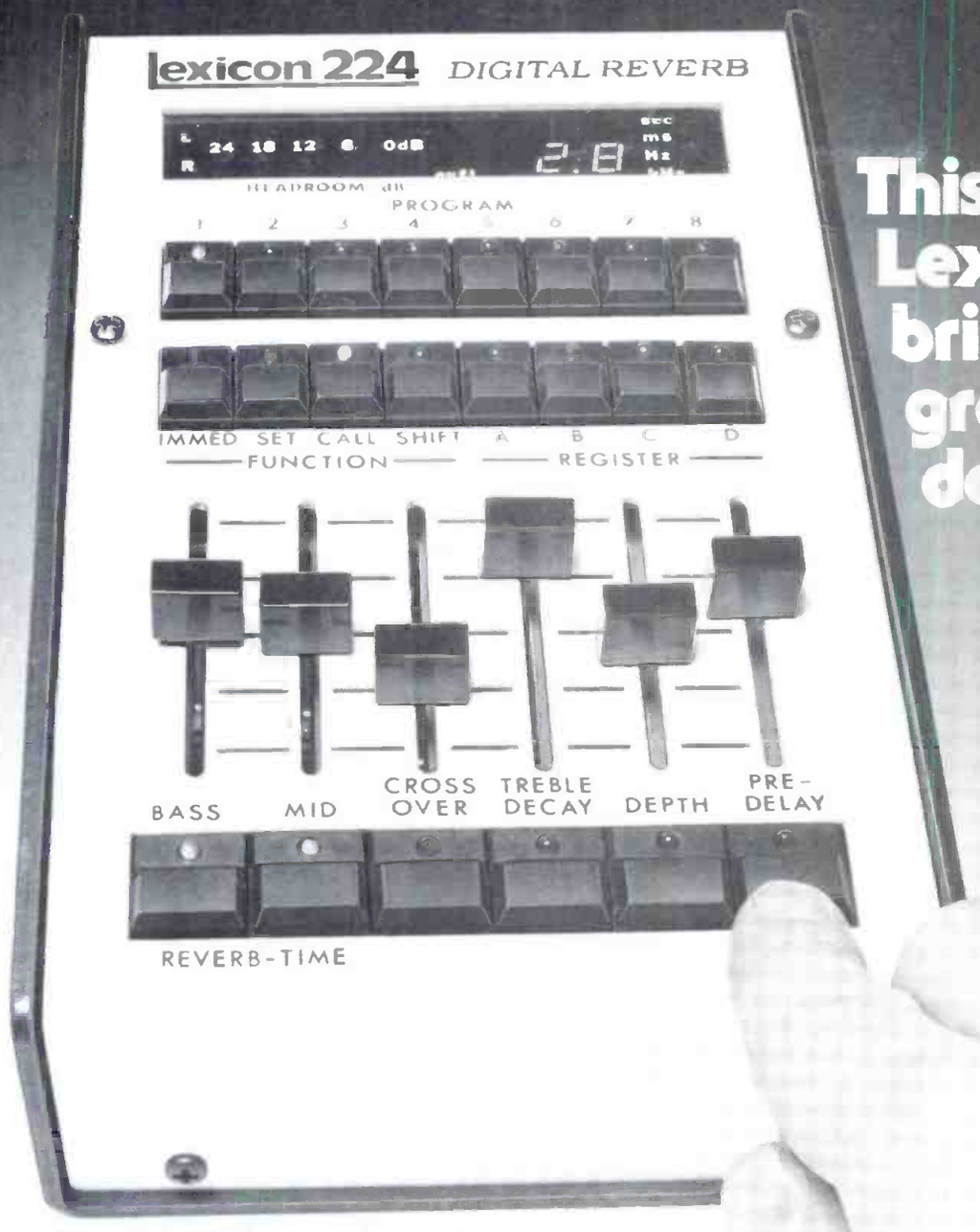

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db Letters

To The Editor:

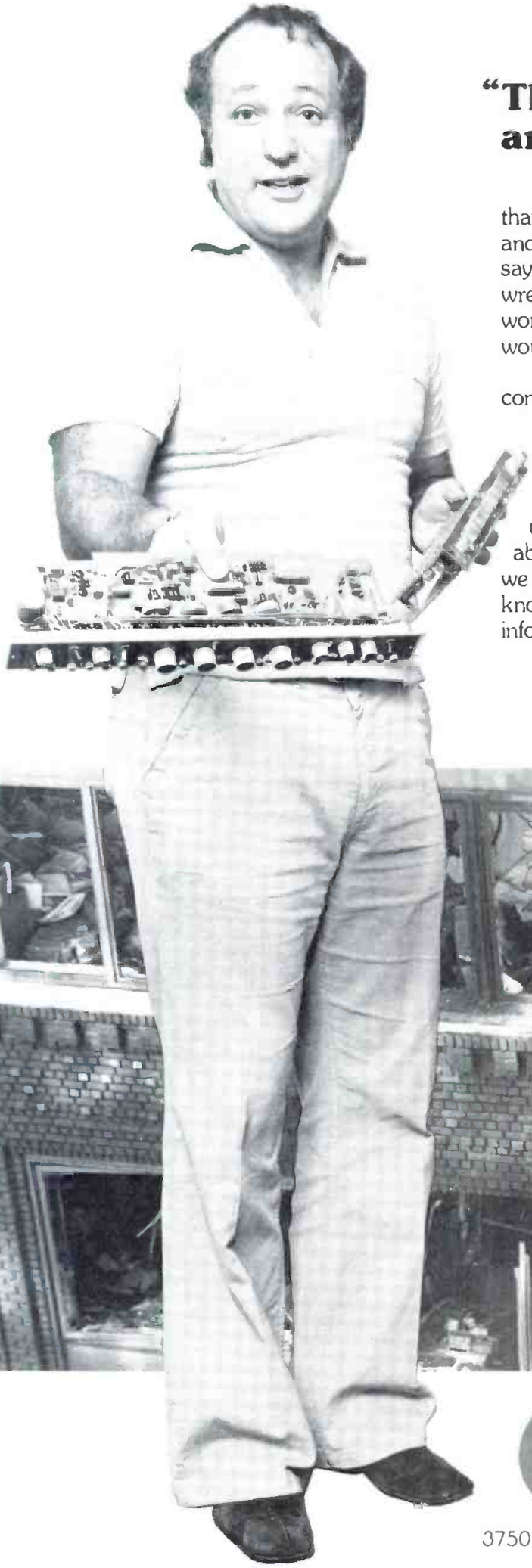
I think that the article "Distortion and How It Is Measured" by Larry Maguire in your April issue of **db** is excellent. However, further clarification is needed concerning the example of inaccuracies in distortion measurements in the third paragraph on page 53. Larry's article cites an example of a signal generator with 0.1 percent distortion and an analyzer with 0.1 percent residual distortion that are used to test an audio component. The article says that if the distortion analyzer indicates 0.0 percent distortion, that the distortion is in reality 0.2 percent because the ". . . 0.2 percent total distortion of the test equipment has been cancelled by the 0.2 percent distortion of the audio component, which was in opposite phase."

The article should have indicated that with the stated residual distortions in the generator and analyzer, the true distortion caused by the audio component *could* have been anything between zero and 0.2 percent, depending upon the phasing and amplitude of the various distortion products.

Assume, for example, that your best design efforts had paid off and the audio component being tested appeared transparent (negligible distortion). It is quite possible for the 0.1 percent residual distortions of both the generator and analyzer to cancel, giving the 0.0 percent distortion indication mentioned in the article.

If, as indicated in the article, the 0.1 percent distortion in the generator was in phase with the 0.1 percent residual distortion in the analyzer, then a direct connection between the generator and analyzer would indicate 0.2 percent distortion (if the distortion products are identical).

This type of error can usually be avoided by first making a measurement with the generator connected directly to the analyzer before commencing further tests. If the generator has a balanced or floating output, then the generator output connections or leads should be reversed and the measurement repeated. If these two mea-

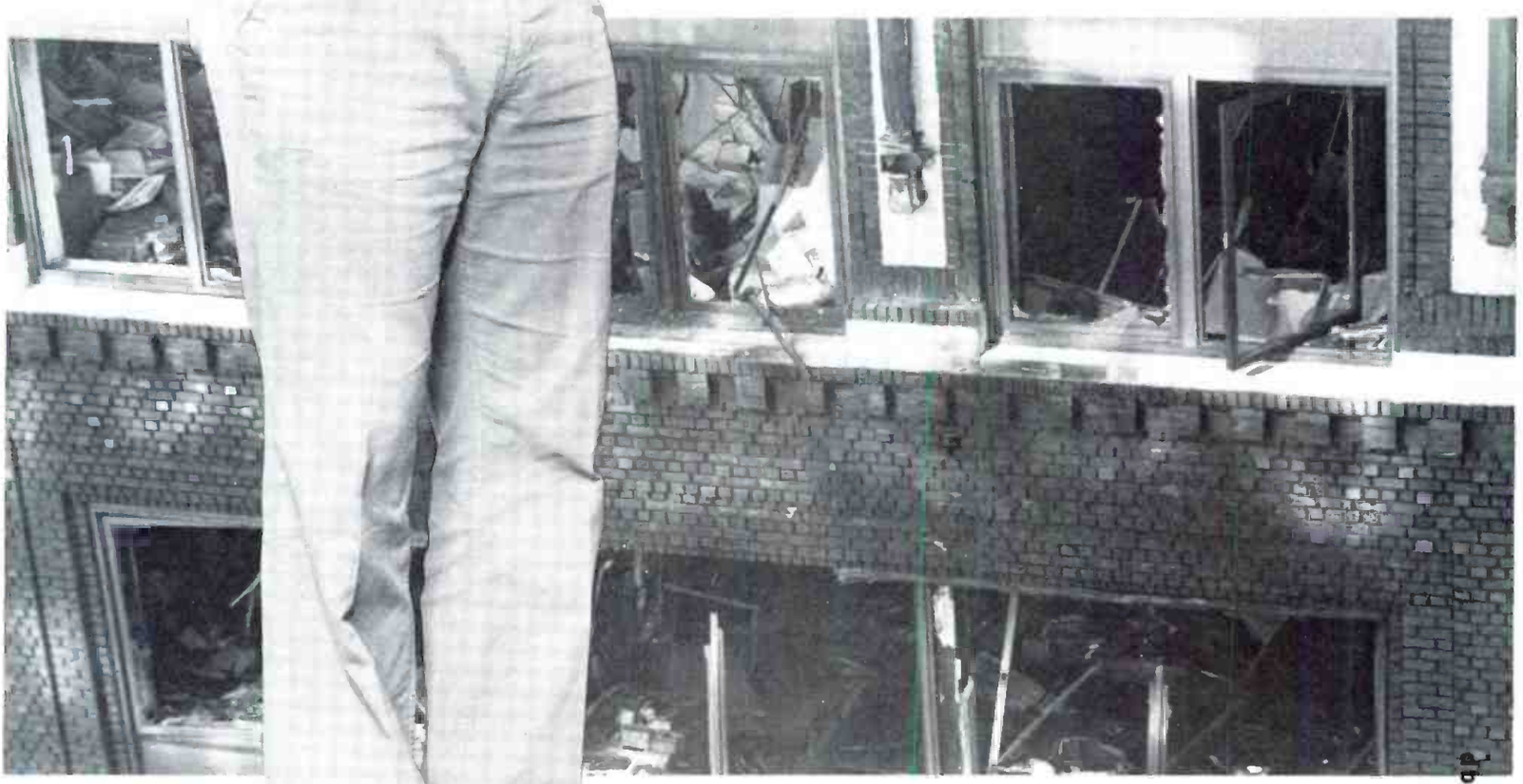


"These Auditronics 501 modules are all that survived the fire...

that totally destroyed our East 46th Street Superdupe and SDC Recording Studio complex last year," says Herb Gordon, studio owner. "Almost before the wreckage cooled and the insurance adjuster completed his work, I was already planning the equipment purchases we would need to get back into business."

"In all the years we had the Auditronics 501 consoles prior to the disaster, I don't remember anything going bad. Our purposes were well served by our old 501s, so I went back to Auditronics the week after the fire to order three more of the latest model 501s."

Herb Gordon and over 300 other satisfied users of Auditronics mixing consoles have learned much about Auditronics performance, quality and reliability since we delivered our first 501 back in 1971. If you would like to know what they know, call or write for complete product information and the name of your nearest Auditronics dealer.



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surements differ it indicates that distortion cancellation (or hum cancellation) is occuring.

It is always a good practice to connect a scope (and wave analyzer if available) to the output of the distortion analyzer to determine the nature of the distortion (or noise) products.

The best solution to this uncertainty is to use audio measuring equipment with residual distortion at least an order of magnitude better (lower) than the performance of the audio component being tested.

ALBERT S. JARRATT, SR.
Federal Communications Commission
Washington. D.C.

Larry Maquire Replies:

Mr. Jarratt's remarks are for the most part correct, although they are somewhat beyond the scope of the original discussion. where the worst case was intentionally presented. It is worth pointing out that the error cannot be avoided, as Mr. Jarratt suggests merely by reversing test leads. This technique will determine only whether the oscillator and analyzer have cancelling distortion. However, unless the device under test has a linear phase vs. frequency characteristic, the harmonics will be shifted with respect to each other and errors of unknown magnitude will result when it is inserted in the system. The only way to dependably improve on the worst case analysis is to measure the phase characteristic of the device under test and the magnitude and phase of each distortion component of both the oscillator and analyzer.

TO THE EDITOR:

A disturbing trend has popped up in recent years regarding "re-inventing the wheel." and in particular I refer to a lack of adherence to long-accepted terminology. One important example of this is how Reverberation time. long described by T_{60}, is now being referred to as "RT_{60}." I believe this came about as a model number of a particular manufacturer's product. This in turn has led to the belief among studio people that RT_{60} stands for Reverb Time, and that it is somehow OK to refer to "RT Times" and "RT Timers" and "RT_{60} Meters"! Not only is it historically incorrect but the additional R. which has a valid. accepted use as Room Constant (which I hope doesn't mutate into "RC") next to the T_{60} would be considered Room Constant multiplied by Reverberation time. So please. Industry, T_{60}. OK?

ALAN FIERSTEIN
President. Acoustilog
New York. NY

TO THE EDITOR:

As I read over the "Directory of Test Equipment Manufacturers" in your April issue, I was, to say the least, dismayed to find that Radiometer Electronics was not listed. I am sure that this was an oversight on your part. however an injustice to both your readers and us.

So that we might be included in your next listing, I am enclosing descriptive literature on our Model BKF10 Automatic Distortion Analyzer and Models RE402 & RE403 Wow and Flutter measuring and analyzing equipment.

JEROLD R. BUSH
Vice President/General Manager

db Replies:

Sorry about that! Radiometer Electronics' address and telephone number is:

811 Sharon Drive
Cleveland. Ohio 44145
(216) 871-8900

Also omitted from the Directory:

VIZ Manufacturing Co.
335 East Price Street
Philadelphia, PA 19144
(215) 844-2626

and

TELEFUNKEN
Gotham Audio Corp. (Distributor)
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New York. New York 10014
(212) 741-7411

TO THE EDITOR:

In reading the article, "Digitizing Audio With Delta Modulation" in the April, 79 issue of **db**. one very obvious. but theoretically possible idea comes to mind. What is to prevent us from putting the delta modulated digital signal onto tape?

Assume the following: The delta modulated signal is sampled at a frequency that closely approximates the bias frequency of an analogue recorder. and let's assume that the recorder is capable of handling the various pulse widths of the delta modulated signal. Assuming that the level is correct. we could then lay down the delta modulated signal directly onto the tape, of course, bypassing the recorder's conventional analogue electronics and their associated frequency roll-

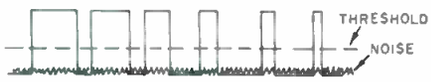

Figure 1.

offs. We now have a digital audio signal laid down on a storage medium.

Fine and dandy, but what happens when we play it back? We have a digital representation of an analogue signal, not a numerical grouping of pulses which stand for a specific analogue audio level. As can be seen in FIGURE 1, even though the pulses are on the tape, there is still the ever present tape noise. If we try to demodulate, and amplify the signal directly from the tape, we will also have the inherent noise with which to contend.

What if we insert a pulse detector into the audio path and establish a threshold level which would reject all signals and noise below that level? FIGURE 2 shows a block diagram of such a signal path. The detected pulses, and their corresponding widths are then used to regenerate new, clean pulses which are then demodulated into the analogue signal.

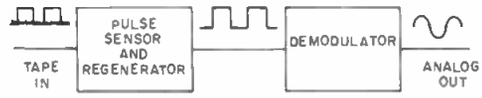

Figure 2.

The obvious advantage of this type of storage is that the original signal-to-noise ratio is still preserved since amplitude of the signal on the tape is no longer a factor. Wow and flutter may still be a problem on a poorly constructed deck, but most manufacturers have these figures way down to where they are practically undetectable. Distortion is limited to the ability of the tape and the electronics to accurately reproduce the width of the pulses, and with electronics able to slew in nanoseconds, distortion comes down to just how accurately the tape records the pulses.

Since there is no conversion of the analogue signal into a numerical value, the electronics in a delta modulated digital system are extremely simple, and far less expensive than other digital systems. Applications of the delta modulated digital system can extend into consumer tape decks, cassette recorders, video-audio disks, motion picture sound tracks, etc. In fact, in the Sony Pulse Width Modulation amplifier, model TA-N88B, the digital pulse width signal can be tapped (before it is amplified) and fed into a tape recorder (assuming the recorder will resolve the 500 kHz sampling frequency). Even radio, especially f.m., and to a certain extent TV could be broadcast this way.

Even though I have the knowledge, I do not have the means to check this theory out. I am quite certain, though, that somebody else is already working on what I have just described. I am sure that a lot of readers would be extremely interested in seeing an article on experiments done with this type of system.

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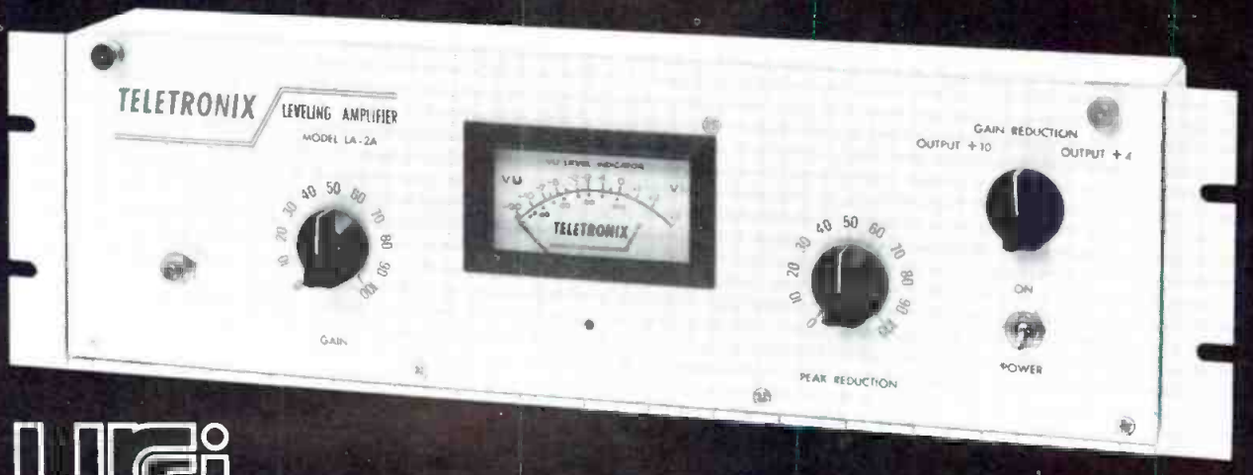

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PATRICK S. FINNEGAN

db Broadcast Sound

Multi-Track Tape Problems

● Although the full track audio tape system has been the workhorse of the broadcast industry for many years, the use of multi-track systems has been growing at a considerable rate over the past few years. Most generally these range from 2- to 4-track, and occasionally 8-track systems. Multi-track systems have inherent problems of their own which are not found in full track systems, and the application or use in the broadcast system itself can create additional problems. This month we would like to touch on a few of these problems.

GENERAL TYPES

We can divide broadcast audio tape equipment into two general types: the open reel recorders and the cartridge tape units. Although a variety of tape brands with differing electrical specifications are in use, the ¼-inch width is standard in all these systems. By far the most used type of machine is the open reel recorder, and for mono this is usually a full-track system. But next in order and running a very good race are 2-track machines. The majority of the 2-track machines are stereo units for producing or playing music tapes in stereo automation systems. There is still some use of the half-track unit for monaural which plays either direction on the tape. As a.m. broadcasting goes full stereo in the years ahead, the full-track mono tape recorder will no doubt fade out of the picture, as did transcription equipment a few years ago.

Cartridge tape systems carry a considerable work load in broadcasting, especially for pre-recorded commercial and other announcements. Some stations have their music on cartridges. Cartridge systems are all multi-track, whether they are mono or stereo. This system makes use of a separate control track on the tape for cueing and external switching purposes. And besides these normal functions of that track, there is an increasing use of the track to also carry data signals for logging or similar purposes. Although none of the information on this track is broadcast, as is the program tracks, the track must perform up to expectations or it can create havoc with the programming especially in automation systems.

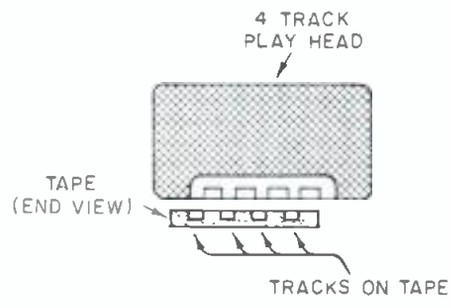
Although not as common and not a program machine, there is another type of audio tape recorder the broadcast engineer must maintain in a number of stations. This is the slow speed audio recorder used for program logging or verification. The most general type is an open reel, 4 to 8-track unit which uses ¼-inch wide audio tape and runs at a very, very slow tape speed. This machine has the usual multi-track problems, but the very slow speed creates additional problems.

CROSSTALK

Perhaps the first and most basic problem with any multi-track tape system is the potential for crosstalk between adjacent tracks of the tape on playback. This is essentially a mechanical alignment problem in that the pole pieces of the playback head are not directly over their respective tracks on the tape. A playback head may fall into the unoccupied space between two tracks or it may straddle two tracks.

Mechanical alignment of the record and playback heads in relationship to each other and the tape is very important in multi-track systems. The width of the tape should evenly cover the exposed pole pieces on both heads. Needless to say all machines in the broadcast system should be aligned to the same standard: both mechanically as well as electrically. Precision mechanical alignment gauges are avail-

Figure 1. If the play head lies between tracks there will be severe crosstalk.



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able for this purpose, and if you want the tapes to be compatible with outside tapes the electrical alignment should be to NAB standards with an NAB alignment tape.

Static mechanical alignment with gauges is one thing but the tape must stay in this same mechanical alignment while it is in motion. The tape guides have a direct bearing on this dynamic alignment as does the pinch roller. If the guides are not in proper position they will force the tape out of position as soon as it begins to move. And if the pinch roller is not in proper position in relationship to the drive shaft it will cause the tape to move away from or towards the deck according to the direction of misalignment. Proper pinch roller adjustment is more important in cartridge systems with their poorer guidance, but a poorly aligned pinch roller can affect open reel machines, even though they have better guidance.

SIGNAL LEVEL AND RESPONSE

Since the width of the tape has remained the same, the more tracks there are on the tape, the less space can be allocated to each track, and there must be space between adjacent tracks. The room for error becomes correspondingly less than in full track systems. With less tape area to work with and smaller head pole pieces for each track, signal amplitude becomes somewhat less and noise becomes more of a factor. With less iron in the head the very low audio frequency response becomes poorer. Proper electrical alignment and equalizer adjustment become more critical. Each track is an individual channel but if the information on two tracks have a relationship to each other, such as in stereo, the phasing between those channels also becomes a very important aspect. This phasing is related to the tape/head mechanical and electrical alignments, but the amplifier connections to the heads must also be properly polarized.

Head wear and oxide build-up are critical factors in the multi-track system. As the tape wears down the head, grooves will form. At the edges of the groove there is a more rapid build-up of oxide since the usual cleansing of the tape motion on a good head is not now as effective. This oxide will lift the edges of the tape from direct head contact, and both signal level and frequency response will suffer at that section of the tape. How serious the effect depends upon what is in that area. When many tracks are on a tape there will be a narrow track laid down right along the outer edge of the tape. Audio from such an affected track will be low and muffled. And if that track carries data or control tones there will be faulty logging or switching of the automation. A worn head requires cleaning more often than a good head, and must be replaced more often than would be the case in full-track operation.

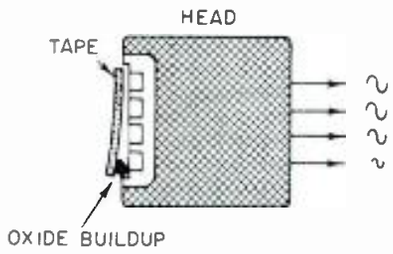

Figure 2. Oxide build-up on a worn head will lift the tape from the head and reduce the output from the affected track.

A tape with ruffled or damaged edges played on a non-worn head will produce the same or similar problems as a good tape on a worn head. In this case the tape edges are physically damaged. Such damaged tape cannot be recorded properly on those edges either. This type of damage is most often caused by bent tape reels or damaged tape guides. Such damaged reels should be discarded and damaged guides replaced.

CONTROL AND DATA

As mentioned earlier, the extra track on cartridge tape is a control track. Bursts of sine wave tone generated by oscillators in the record unit are recorded on the tape automatically when program is being recorded. In playback, these tones are amplified, separated by filters, and routed to proper logic circuits for cue and switching actions. Signal amplitude, noise and distortion enter into the activities. The recording bias and levels of each tone must be set properly when recording, and the various control function levels set properly in the playback machines. Whenever heads are replaced and realigned, always readjust the cue levels according to the NAB test tape levels for such signals, and also measure what levels are actually recorded on the tape. Check the track for noise and transient pulses as they can in themselves operate the control functions when least expected.

Besides the ordinary cue and switching signals applied to the control track, data may also be added. There are different types of data signals but they are applied to the control track as sine wave audio signals. One such system is

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The Shure SM81 cardioid condenser is a new breed of microphone. It is a truly high-performance studio instrument exceptionally well-suited to the critical requirements of professional recording, broadcast, motion picture recording, and highest quality sound reinforcement — and, in addition, is highly reliable for field use.

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- PRECISE CARDIOID polar pattern, uniform with frequency and symmetrical about axis, to provide maximum rejection and minimum coloration of off-axis sounds.
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- 120 dB DYNAMIC RANGE.
- ULTRA-LOW DISTORTION (right up to the clipping point!) over the entire audio spectrum for a wide range of load impedances. MAXIMUM SPL BEFORE CLIPPING: 135 dB; 145 dB with attenuator.
- WIDE RANGE SIMPLEX POWERING includes DIN 45 596 voltages of 12 and 48 Vdc.
- EXTREMELY LOW RF SUSCEPTIBILITY.
- SELECTABLE LOW FREQUENCY RESPONSE: Flat, 6 or 18 dB/octave rolloff.
- 10 dB CAPACITIVE ATTENUATOR accessible without disassembly and lockable.

Outstanding Ruggedness

Conventional condenser microphones have gained the reputation of being high quality, but often at the expense of mechanical and environmental ruggedness. This no longer need be the case. The SM81 transducer and electronics housing is of heavy-wall steel construction, and all internal components are rigidly supported. (Production line SM81's must be capable of withstanding at least six random drops from six feet onto a hardwood floor without significant performance degradation or structural damage.) It is reliable over a temperature range of −20° F to 165° F at relative humidities of 0 to 95%!

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SM81 Cardioid Condenser Microphone

SHURE®

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Manufacturers of high fidelity components, microphones, sound systems and related circuitry.

Circle 24 on Reader Service Card

a dual tone combination which decodes as numbers very similar to the telephone dialing system. Another is standard ASCII teleprinter code for clear text logging. This is basically a d.c. pulse code and cannot be recorded directly onto the tape very effectively, so it ordinarily modulates an audio carrier in some manner, such as FSK, before being applied to the track. Whenever such data is applied to the control track, operations of the track and associated circuitry become far more critical than would be the case otherwise. The proper signal level ratios of all these signals must be recorded properly, and just as importantly must be recovered from the tape in the same ratios and without noise and distortion added. The playback amplifier and control circuitry adjustments are very critical as are the filters in the playback amplifier. An FSK carrier for example, is somewhere in the area of 3-4 kHz. The cue filter in the playback should theoretically reject all other signals except the 1 kHz cue tone. But if that filter is sensitive to the 3rd harmonic of the cue tone, then it is also sensitive to the 3-4 kHz data signal and can cause the tape to stop, because of the data signal, only a few seconds after it began to play on the air. In this type of situation the signal levels become extremely critical and even though controlled properly, may not suffice. We may try the brute force method and modify the playback logic to be less sensitive to the data and it may work. But a couple dB drop in the recovered cue tone level, because of a dirty head or worn tape, can cause the playback logic to ignore the real stop tone and not stop at all!

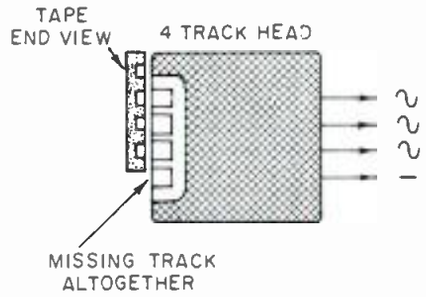
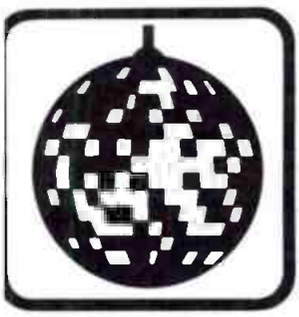
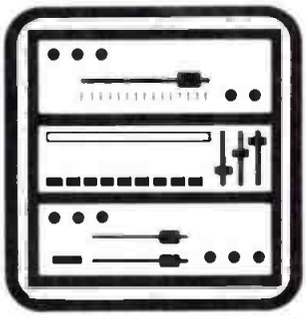


Figure 3. Improper mechanical alignment can cause the loss of a complete track.

SLOW SPEED LOGGERS

Of all the multi-track headaches, the slow speed loggers can be a real frustration, in terms of maintenance. Much of this is related to the very slow speed of the tape travel. In one machine it takes about 2 seconds for the information recorded by the record head to be picked up and played from the tape by the playback head. You can't just wiggle a wire or connector to find an erratic one—it takes about 2 seconds between the wiggle and the results to be heard! For the same reason adjustment of the head alignment is very difficult. You can't just rock the record head through adjustment and watch the results directly on the output as can be done in a normal speed machine. The adjustments here have to be in very small, well spaced steps.

Besides electrical alignment the mechanical alignment is very important in terms of compatibility. When used for logging purposes these tapes must be retained for a 2 year period. So if heads are replaced those original tapes must be played on the machine when needed. The new heads must be in almost perfect mechanical alignment with the original heads or crosstalk may be so bad as to make the playback unusable. When changing heads and after alignment, never consider the job complete until the machine can play back those original tapes in their entirety. That is, make sure all the tracks are being played. On an 8 track system it doesn't take too much misalignment to lose an edge track completely. ■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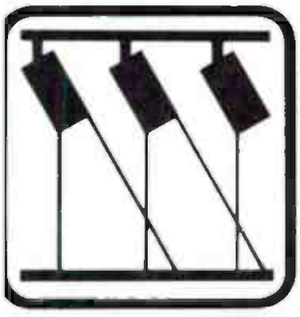


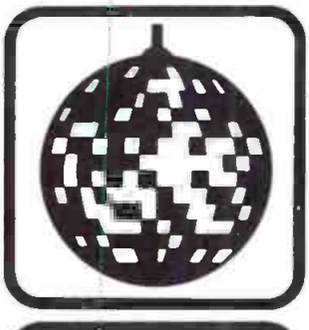

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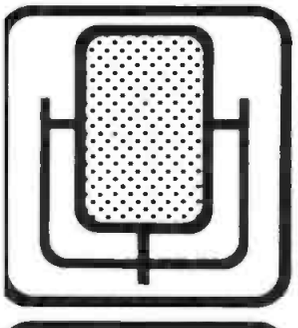

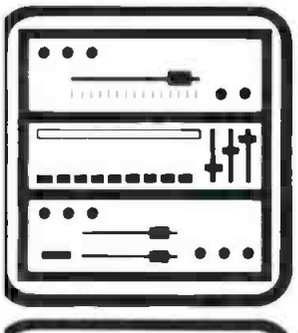
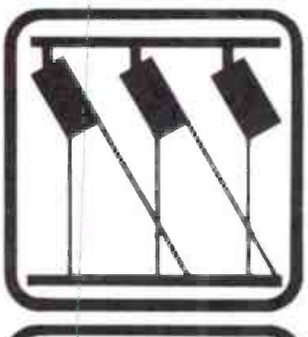
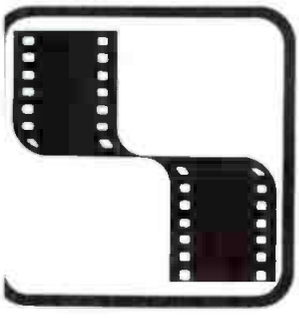


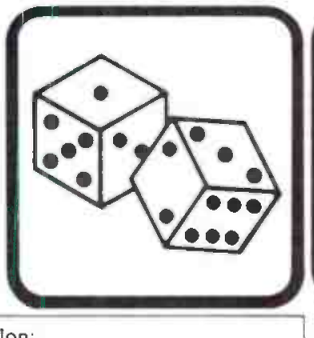


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db Editorial

SEVERAL YEARS AGO, one of the local professional recording industry societies held a meeting, at which the impact of technology on recorded sound was discussed. Most of the participants spent most of the evening damming the technology that was ruining the sound of music.

Various culprits were identified—notably, the multi-track tape recorder, and the sinister overdub session. And it's true enough—many a piece of music has been demolished by both. But who booked these sessions in the first place? Certainly not the tape recorder! Nope, the session was booked by the client: the one who discovered that at "Studio X," there was a new funny-box that was guaranteed to churn out hits. So the client began to depend more and more on gadgetry, and less and less on talent. Somewhere along the way, he forgot that the gadgets were intended to be tools—not replacements—for the creative artist.

By and by, if the artist didn't sing very well, he would reach for the reverb return pot. If the rhythm guitar couldn't get his licks together, there was the punch-in. And, when all else failed, why—"we'll fix it in the mix."

Fix *what* in the mix? Who needs to mix-fix a song that should be playable by any reasonably-alert high-school music student?

"Well, you see, this group really isn't together, so we have to double the lead vocals, bury the piano, and try to get a 'sound' on the rhythm. Later on, we'll sweeten it, and it won't be so bad." (And still later on, we'll blame the technology for this little horror show.)

You don't have to be a psychic to figure out what's wrong here. T'would seem the technology should be complaining about the artist, rather than vice-versa. But of course, the technology doesn't speak—it just sits there, waiting to be used, or abused, by anyone with the price of admission. And these days, that price is getting higher, and still higher.

Which brings us round to this month's subject—the "super console." No one's giving them away these days. In fact, prices seem to *start* at about $100,000. Since that's not exactly petty cash, we asked several prominent console manufacturers to describe their conception of what goes into the "ultimate music machine." As you'll see, there are differences of opinion about how to put a super-console together, and about how to use it once it's yours.

Which one is best for you? We'll cop out on answering that one, although Carl Yanchar's feature **Choosing a Console—Is a Picture Really Worth a Thousand Words?** may help. Or, you can follow Kendun Recorders' example, and get "one of each." (See the photos in Carl's story.)

Most discussions of super-consoles (and that includes this month's features) concentrate on the mix-down aspects of record production. It's obvious—to our authors at least—that recording remains an art rather than a science. And the prospect of "automated art" is too much for them (and for us) to bear. (Besides, its probably impossible.)

And maybe that brings us back to the meeting we mentioned earlier. Music is still created by artists. The artist may, or may not, be a technologist too. But the technologist who is not an artist won't get very far, despite all of his automated wizardry. If you attempt to turn the creative function over to the machine, you pretty-much deserve what you get. And it's really not the machine's fault. If it had half a brain, it certainly wouldn't ask you to do its job. So, why ask it to do yours? If you insist, then please don't blame technology for what you get. And, in the age of the "super console," you're apt to get an absolute disaster if you're not prepared to use the technology creatively.

We said earlier that technology doesn't speak—but even that's changing. Many of the latest boards have some sort of readout device—typically a CRT—to keep you informed of what it's up to. If you make an illegal entry, it "talks back," to let you know something's wrong. Depending on its "smarts," it may prompt you towards correcting your error.

Now wouldn't it be interesting if the technology eventually got tired of taking the rap, and started complaining about questionable artistry. For example, consider the following read-outs:

RECORD MODE DISABLED. TO CONTINUE, RETUNE GUITAR.

THIS HOOK USED 16 TIMES THIS MONTH. TRY SOMETHING ELSE.

NOT TOGETHER. GO TO REHEASE MODE.

SONG TOO DUMB. REDUCE TRACK ALLOCATION.

Then what would happen? Why, people would find it harder to blame technology when the fault lay elsewhere.

Well, it hasn't happened yet, but we're just getting started. Somewhere, someone is designing the "super console" of tomorrow. If that console tosses you out of the studio, don't say you haven't been warned.

J.M.W.

DAVE PURPLE

Automation: It's Evolution

With fully-digital consoles on the horizon, the future of console automation is boundless.

FIFTEEN OR SO YEARS AGO, "console automation" was a phrase which was not heard in recording studios. Most of our recording was done 2, 3 or 4 track, generally with all music and vocals performed and recorded simultaneously. If a reduction to mono or stereo was necessary, it was, hopefully, just a matter of setting the console faders at a given point, maybe adding a touch more EQ and/or echo, and rolling the tape machines. In those days, albums were "mixed in less than a day. A one-inch, 8-track machine was considered a device for making a duplicating master for the manufacture of car stereo-8 cartridges.

Then, the recording industry, in it's infinite wisdom, began dictating to console manufacturers the need for more and more tape channels, to allow "the flexibility necessary for today's creative recorded product." The manufacturers (not wanting to lose out on a brand new goldmine) readily complied with industry demands with great glee. The age of multi-track was upon us.

Recording techniques began to change. Things that previously had been recorded with echo were now cut "dry." Mixers began recording tracks at 0 VU, to take advantage of maximum signal-to-noise ratios on this new two-inch wonder which happened to be adding an additional 6 dB of tape hiss to their final mixed product. Mixing an album product, which once took a day (or two at most) became a nightmare of weeks, or even months.

WOULDN'T IT BE WONDERFUL, IF . . .

Although no one can ever be sure who was first, someone, sometime (either in a fit of rage, or in an incredible dream), must have said: "Gee, wouldn't it be wonderful if someone made a computer that would help me with this mess?" The seed for console automation had been planted. Manufacturers have now been talking about console automation since the early 1970's. One manufacturer even described a fully-digital console as early as 1972.

BUZZ WORDS, BALLY-HOOS, AND HYPES

I suppose this is why people in marketing departments get paid, but as automation technology progressed, so did the "buzz words," "hypes," and "bally-hoo" in the advertising of same: ("buzz words" are in bold face).

1. "This system will capture and fully decode **data** even when turning the reels of the tape machine by hand." (Is this a new way of mixing?)
2. "Worst-case system **access time**: 1.2 ms."
3. "Even after 20 **update cycles**, this **processor** is still faster than the human brain can **command** the human hand to move a fader."
4. "Nothing can go wrong."
5. "This system is **fool-proof.**"
6. "Nothing can go wrong."

How much does all of this mean to the *typical* end-user of an automation device? Probably, not much. If you are like me (one who is not accustomed to speaking the "1s and Øs" language of the binary digital format), you could probably care less about bi-phase encoding, parity bits, 24-bit words, baud rates, start-stop bits, etc. The most important things to know and believe about an automation system are that: *it works* and, how to utilize the system on a day-to-day basis. And that's the point of this article: specifically, an operator's-eye view of the implementation and operation of the Harrison Auto-Set.

THE HARRISON 864 AUTO-SET

The Harrison 864 Auto-Set seen in FIGURE 1 is a fully self-contained console automation programmer. Specifically designed to interface with all existing Harrison recording consoles, the Auto-Set may also be easily interfaced with any device which utilizes d.c. voltage control. Auto-Set is a twin microprocessor-based system, under firmware control, which resembles a small typewriter. (Firmware is a software operating system which usually resides in ROM memory.) Although the marriage of console and typewriter may at first intimidate some, the keyboard permits rapid access to the internal Auto-Set programs, and with minimal effort, this access may easily be accomplished. The inexperienced operator need not know the complete set of access instructions, as he can gain

Dave Purple is the sales manager for Harrison Systems, Inc., Nashville, Tennessee.

experience one step at a time. And as the operator gains this experience, he eventually learns to write more complex programs which, when implemented, allow him to perform some rather incredible tasks with ease and an amazing degree of simplicity.

DYNAMIC AND STATIC FORMATS

The Auto-Set is, in reality, two separate automation programmers housed in one package: dynamic and static automation formats. Standard dynamic automation (which most of us have just recently begun to know and love), is most-simply described as memorization of fader movement (or non-movement), and channel on/off, or mute, status. Dynamic automation involves data storage in real-time on tracks available on the multi-track master tape.

Unique to the Auto-Set is its ability to capture and store static, or "preset," automation data. In the preset mode, the Auto-Set takes a "snapshot" of the fader positions and mute status for up to 63 faders. The necessary data for the snapshot is internally generated and stored on a 3M DCD-100 data cartridge by the data cartridge tape drive built into the Auto-Set. The data cartridge is capable of storing 630 separate presets, or snapshots, of up to 63 faders. Data is recalled on command, as needed by the operator. Because data storage is outboard and independent of anything happening in real-time, the Auto-Set preset automation mode lends itself perfectly to such tasks as automation of a live performance, direct-to-disc recording, or a multi-track mixdown, when no data storage tracks are available for storage of standard real-time dynamic automation data (that is, when the producer has completely filled up your 24-track master.)

Auto-Set fader levels and mute on/off status are generated via a d.c. voltage at each individual fader on the console. Each channel's d.c. voltage is converted to a binary number, which the digital computer can understand. Conversion takes place in an analog-to-digital converter. As the fader is pushed up higher in (analog) level, its corresponding d.c. voltage goes down; therefore, the binary number which appears at the output of the A/D converter also decreases in size. The 8-bit binary number, or "word," yields 256 separate 0.5 dB analog steps. (Since each bit may be either a 1 or Ø, 8 bits = 2^8 = 256 possible combinations—Ed.) These 8-bit words for 63 faders are collected in a parallel (all at once) manner, re-converted to a serial format, and then transmitted in a priority-encoding format for storage in real-time on the multi-track tape.

GETTING STARTED

But enough of the technicalities. I mentioned earlier that it really isn't necessary for an operator to fully understand digital technology in order to make a console automation programmer fly. However, it is important that he be capable of taking full advantage of all the power which a programmer offers as a creative tool.

To better enable the operator to communicate with the computer, the Auto-Set programmer includes a CRT terminal. This may be seen in the upper left-hand corner of the programmer in FIGURE 1. This video readout permits the computer to "talk" to the operator. At any time during the session, the CRT may indicate the status of the automation system or prompt the operator through the necessary sequence of programmer key strokes.

The first item to appear on the CRT (FIGURE 2) is a "menu" of operator options. By pressing 1 or 2, the operator may choose between dynamic automation or preset (static) operation. Both are described below.

OPERATION: DYNAMIC AUTOMATION

To select the dynamic automation program requires one keystroke entry. The Auto-Set CRT will then display the Automation Operate Page seen in FIGURE 3. Before we continue, the unit wants answers to several questions, the first of which is "MAX CHANNEL." The unit comes up initialized for 61 channels (Group and Quad Masters account for number 62 and 63). But since the unit is a sequential scanning device, there really is no sense in wasting time scanning 61 faders if we are not using that many. Assuming we would only want to automate 32 channels, we merely press "C" (for channels), enter the number 32, and press CR (carriage return). The programmer will now scan faders 1 thru 32, the Group and Quad Master, and ignore faders 33 thru 61. In effect, we have doubled the scan rate of the programmer.

The next line deals with BAUD RATE for the Auto-Set. Baud Rate, simply, is the speed at which the computer generates its data. Normally the computer comes up at the high baud rate (9,600 baud). The high baud rate requires that a tape machine have a sync response of about 10 kHz. If desired, you may change the baud rate to medium (6,400 baud) or low (4,800 baud). Obviously, decreasing the baud rate will also decrease the speed of the programmer. You probably will never have to change the baud rate, but it is nice to be able to do this, should the need ever arise.

The next line on the operate page reads "MT RETURN -ABCD?" Although only two tape channels are required to perform updates, space for selection of four tape return channels has been provided. MT RETURN means "Multi-Track Return" and when implemented with a keyboard entry, becomes, in effect, an electronic patch cord

Figure 1. The main systems housing of Harrison Systems' Auto-Set. Above the keyboard are a five-inch CRT monitor and the data cartridge drive system.

Figure 1A. The 3M DC-100 data cartridge and its drive system.

system for selection of the tape track from which you wish to receive your "READ" data.

In discussing the next line down on the operate page (WRITE PASS-ABCD?), we begin to see one of the major differences between the Auto-Set and other console automation programmers. Other systems of this type store individual mix data on individual tape tracks. The Auto-Set is capable of storing *four* individual, totally different sets of mix data per tape track. In writing data, we must tell the computer which Write Pass (A, B, C or D) we wish to generate. On our first write pass, we would "Write A." For our first update, we would want to "Read A" (that is, what we had written on our first Write Pass), generate update data and "Write B." Of course, we would have to bounce data over to our second data track, but in writing "B," the new data set would be combined with the original "A" data, giving us both A and B on the same tape track. This is interesting, because now as we continue our update passes, the mix history follows us. Not until we do our fourth update do we erase the original data which we had first written.

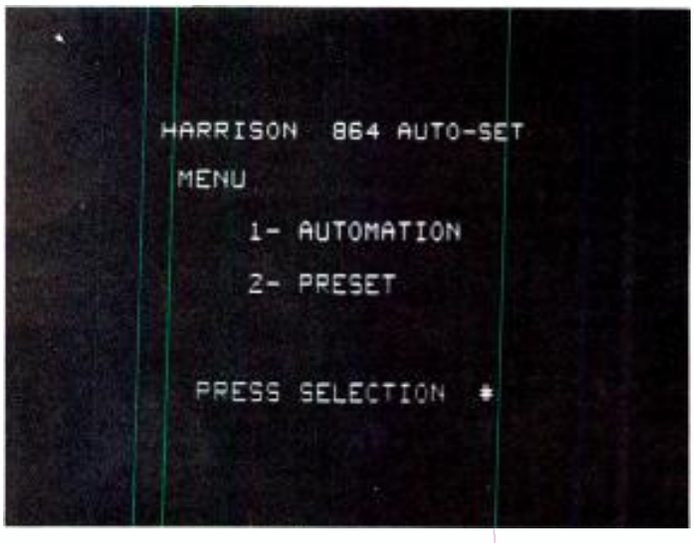
A console-mounted XY crossfader is provided for moving back and forth between any two data sets, and the CRT displays a cursor which follows movements of the crossfader. In FIGURE 3, the cursor is the asterisk seen next and the X in the upper left-hand corner. Transition between data sets may be accomplished in one of three ways: A manual ADVANCE, whereby the operator manually moves the crossfader from the X-to-Y or Y-to-X register. This manual advance causes a smooth segue of VCA d.c. voltages. A JUMP may be performed, which causes a rapid and sudden change of VCA levels. Also, a TIMED ADVANCE may be executed. The computer has an internal clock which may be programmed for a timed advance in 100 ms intervals for any duration between 0.1 second and 9.9 seconds.

It is possible to tell the Auto-Set to put the A read pass at the X end of the crossfader, and the B read pass at the Y end of the crossfader. A mix comparison of the A and B mixes can be accomplished simply by moving the crossfader back and forth between the X and Y ends. In FIGURE 3, X and Y have not yet been assigned to any read passes, as noted by the Øs next to X and Y at the lower right-hand side of the figure.

When starting an automated mixing session, my first suggestion would be to forget that you even have a console automation programmer. Practice your mix just as you would if the entire mix was to be done manually. Even though the Auto-Set offers extensive mix update facilities, my personal feeling is that you should try to get the mix as close to perfect as possible on the first mix pass. I've heard operators rave about the fact that it took them 26 updates to get their product to sound the way it did. I think that if more than four or five updates are needed to make the product right, you have done something wrong from the beginning. Start over again with a total "rewrite" rather than trying to repair your mix with update "bandages."

Still with it? Good, because there is even more power available with Auto-Set automation. Let's assume that our producer, who by now is almost ready to believe that console automation really works, comes to you and says, "Here's the situation: Everything we did in the intro the first time around was great. But, I really liked the way we mixed the rhythm section on the second pass up to letter 'A'; horns at letter 'C' were the best on the fourth pass; strings at letter 'D' and the coda are exactly what I wanted the third mix we did (these letters refer to score markings —not to read passes—Ed.) . . . and by the way, there are portions of one background vocal track that go flat in and out in different spots that I want to mute selectively without another mute pass. Is this possible?" (Usually spoken with sadistic grin.) No problem! Built into the Auto-Set is the ability to program new mix *situations* such as those just described by our friendly producer. There are ten

Figure 2. Auto-Set's "menu" page display.



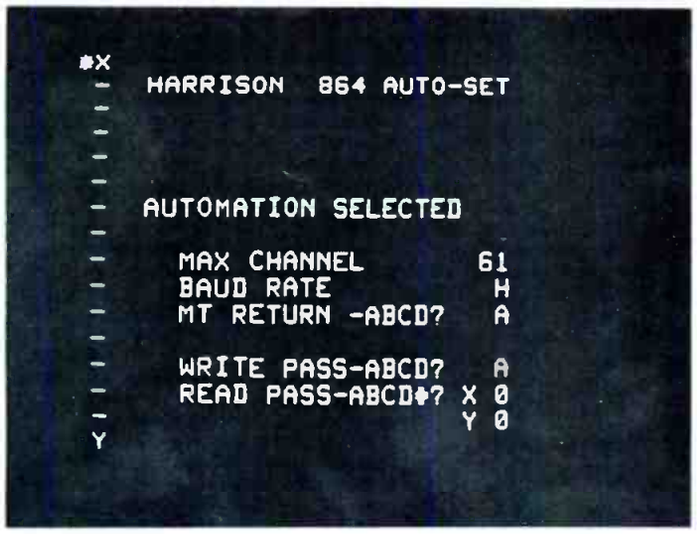


Figure 3. The automation-operate page.

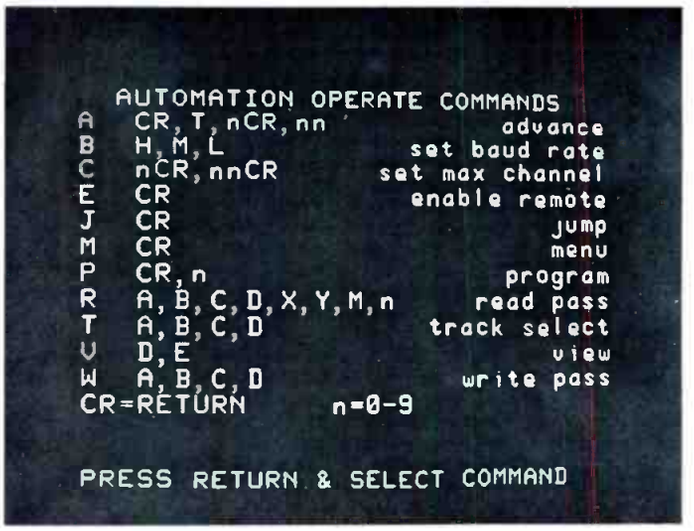


Figure 5. Looking for help? The Help! display lists various automation-operate commands.

internal situations which can be programmed. Touch P (for program) and CR and you now see the CRT display of the first programmable situation (situation Ø, which reads SØ in the lower right-hand corner of the screen in FIGURE 4). The numbers indicate which read pass has been assigned to each fader. A cursor may be set and the read pass for individual channels may be changed. For example, if the cursor is set next to channel 43, the existing situation (read pass A) may he changed to any other read pass. Thus, that horn mix on the fourth pass can he combined with the strings from the third pass, and soon. "M" may also he selected, thereby muting channels so desired. These programmable situations enable us to set up a cross-matrix for read passes that is 63 (channels) by 5 (A, B, C, D and mute read passes) by 10 (situations) deep. This programming capability literally enables the use of bits and pieces from each of the four mix passes for each fader in the console (3,150 different combinations).

HELP!

Just in case your train of thought may wander slightly during your learning curve, help is close at hand. The Auto-Set has four HELP! pages locked in non-volatile memory inside, one for each of the two operate and program pages. Just type "H" (for HELP!) and the CRT will then display operations possible, keystroke entries, and allowed completions for the given page with which you are working (FIGURE 5).

OPERATION: PRESET AUTOMATION

As discussed earlier, the Auto-Set is capable of console automation even when real-time data storage on the master tape is not available. This is accomplished with PRESET automation, where data is generated from console fader "snapshots" and stored on a data cartridge. The data cart will store 630 separate preset mixes for up to 63 console faders.

Entering the preset mode requires one keystroke. The tape drive will rewind the data cart to the top and begin reading the files from the cart while the CRT displays the preset operate page (FIGURE 6). We still have the XY crossfader functions available in the preset mode with the same provisions for advancing through data sets.

The data cart contains 63 files, each of which holds ten presets. To each file, we can assign up to a six-character alpha-numeric file name when the file is written. To read a file, hit "R," and the CRT displays all 63 files by number. Select the file you desire by number, enter the number, and press CR. The data cart will search for the file selected, and the CRT reverts to the normal operate display. When the proper file is read from the data cart, it is

Figure 4. The automation-program page.

Figure 6. The preset-operate page.

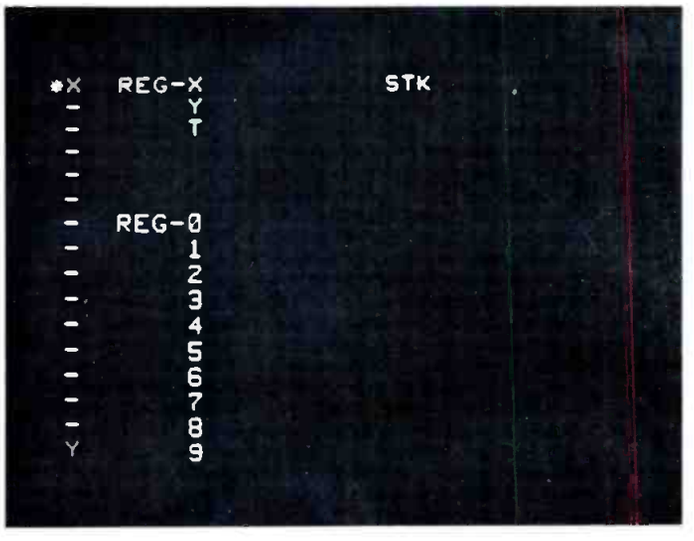


Figure 7. The preset-program page.

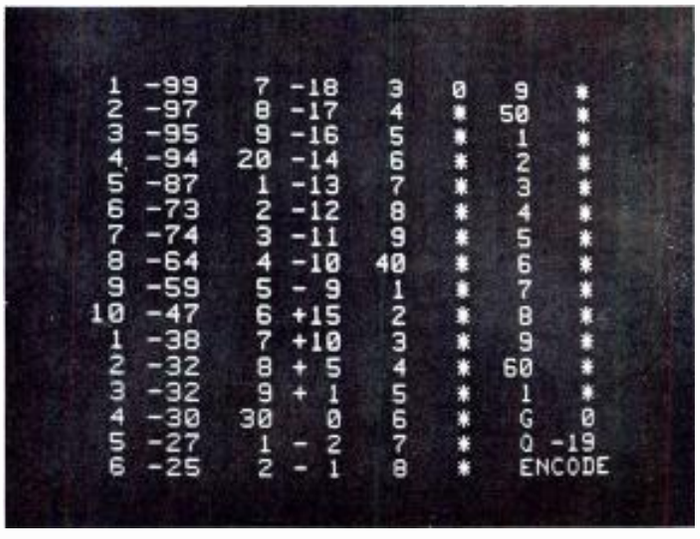
loaded into the "T" register, a transfer buffer. A JUMP command loads the "T" register into "STK." (stack) which is a 16 deep FIFO (first in, first out) stack. As soon as the transfer is complete, the next available file is read from the data cart and loaded into the "T" register for subsequent loading into the work stack.

Data may now be manipulated in the same manner as that for dynamic automation. Advances, jumps, and timed advances allow stepping through data sets almost as rapidly as one can press a button. Files from the data cart will continue being unloaded as long as files remain.

The stack marked "REG" is a ten-deep RAM holding register used either for rearranging data, or for holding data to be used at a later time. After a data set advances through the X or Y register, it is thrown away in a "digital wastebasket" and not used again unless re-read from the data cart. The 10-deep holding register is a viable method of recalling data sets when multiple usage is required.

Writing a file simply requires depressing "P" and "CR." The program page on the CRT is displayed, (FIGURE 4) and when "N" (Name) is pressed, the computer asks you to enter the name you wish to assign to the file (FIGURE 7). It is now that you may enter up to a six-character alpha-numeric name for the file. As soon as the sixth character (or a CR if less than six characters are used) is entered, the computer takes its first "snapshot." Pressing JUMP continues the snapshot process until the file of 10 is filled, at which point the file is written in the next available slot on the data cartridge. If less than 10 snapshots are required, press "W," enter the number of the next open file, hit CR, and the computer will store that file on the data cart.

Figure 8. A "view encode" display.



OTHER FEATURES

Often it becomes useful to be able to see the write (encode) levels going from the console to the programmer and the read (decode) levels coming back from the programmer to the console. This is easily done with Auto-Set. To look at the encode levels of the console when in the write mode, press "V" (view) and "E" (encode). The Auto-Set CRT will then display four column pairs of numbers, with console channel numbers on the left side of each column pair, and the *actual*, not relative, fader setting in dB on the right side of the column pair (FIGURE 8). As faders are moved, the levels displayed on the CRT will change accordingly. To view the decode level type "V" again and "D" this time and decode levels for the particular pass being read will come up on the display. Once in the "view" mode, you can pop back and forth between encode and decode to compare write and read levels.

There are a series of test programs and diagnostics built into ROM memory in the Auto-Set. From the menu page, hold down Control and Shift simultaneously and then press "T" to initiate the test program series. The CRT display will greet you with a pleasant "Hello, Roscoe" (there is a long story as to why). From there you may set up programs for azimuth and speed alignment of the data cartridge drive, A/D and D/A converter adjustments, and even erasure of the data cartridge when a bulk eraser is not close at hand.

Perhaps one of the nicest features of the Auto-Set is that it is a fully self-contained automation programmer. The only allied outboard equipment is a rack-mountable power supply. By unplugging six ribbon-type connectors you can easily move an Auto-Set from one studio to another, replug it into another console and, in effect, have a mobile automation system for about 15 minutes worth of changeover time.

WHERE DO WE GO FROM HERE?

The future of console automation is virtually endless. Had it not been for our outerspace programs, we would not be where we are today with automation. Soon, automation programming devices will be able to "talk" to outboard gear such as limiters, compressors, delay lines, flangers, etc. Programs will be written to allow parameter changes in these devices in real-time. Eventually, *everything* in a console will be automated (we could do this now, but the hardware costs would be *staggering*.) The console will become a fully-digital signal processing device, probably very compact and with no moving parts. Ultimately, as data storage technology improves, the studio recorder as we know it today will be phased-out (no pun intended), in favor of non-mechanical data-storage devices (can you imagine carrying your studio album masters home on four or five strange looking plug-in cards?)

That's about it, in a nutshell. It is very difficult to fully explain any console automation system, much less one as powerful as an Auto-Set, without a hands on, "show-and-tell" demonstration. I welcome questions from readers, both technical and operational.

As technology improves, it is possible that one of us could have a dream today which could be fulfilled in a few years, not unlike the person who, years ago, said "Gee, wouldn't it be nice . . . ." ■

CARL YANCHAR

Choosing a Console—Is a Picture Really Worth a Thousand Words?

Electronic specifications, reliability, price, financing, status, expandability, resale value, and interface requirements—all prime considerations, when choosing a new console.

DURING THE LAST TWENTY YEARS, the recording console has risen to prominence as the most important electronic link in the recording chain and, far too often, as the most important in the entire recording process. In years prior, recording was done primarily with broadcast consoles, not unlike many still in use today. This is not surprising, since in the beginning all recording was done direct-to-disc for later broadcast and syndication. With the advent of tape recording, the technology was expanded and modified to provide some special features such as, channel rather than program equalization—the tape recording medium allowed the experimentation time that was required. As the number of tracks multiplied, so did the demands and requirements of the console. As we entered the seventies, modular, mass-produced consoles were just in their infancy, and the only way to get the console you wanted, in terms of performance and features, was to have it custom-made, sometimes with sub-assemblies available from modular "stock" console people.

THE BROCHURE BATTLE

Soon the economics of 48-in/32-out consoles made custom, hand-wired consoles financially impractical, and so it has now evolved to choosing from the stock consoles available or a "battle of the four-colored brochures."

So, how do you read between the pictures? At first you would think that comparing specifications would make the

This Harrison Systems board is in Kendun's Studio 1 in beautiful downtown Burbank.

Studio 2 at Kendun has an Automated Processes console installed.

Solid State Logic's "super console"—recently installed in Kendun's new Studio D. This is the first SSL board to be placed in an American Studio.

choice easy, but as we will see later, this is not the case. Because a console should be (and usually is) a creative tool, it should be chosen on the basis of *value*, much like you would select any other piece of capital equipment that is important to your life or livelihood. Included in this concept of value are such considerations as features, electronic specifications, reliability, price, financing, status, expandability, resale value, and interface with other equipment and room acoustics.

ACOUSTIC DESIGN AND CONSOLE SELECTION

Starting at the end of the list, acoustic design and console selection are the two prime decisions a studio owner must make. Strange as it may seem, the two are not unrelated. Unless you mix with headphones, the product of any console must be interpreted by what comes out of the monitors and into your ears. It may be reassuring to some to know that their console is flat from d.c. to light with unmeasurable noise, distortion and phase shift—without ever being able to hear its effects. But since what goes in is acoustic and what comes out is also acoustic, should we really compare consoles on such subtleties as transient response if we are going to put them in rooms with acoustic parameters exceeded by even the least-expensive transistor radio?

There are, in fact, certain physical specifications for a recording and mixing console that do effect acoustic performance. First, it should offer minimal obstruction to the direct sound path, with optimal monitor height at five feet to the monitor bottom. This indicates that the console profile, as well as any outboard equipment or cabinets or racks in this direct path, be relatively low. Likewise, the bottom of the console, producer's desk or outboard equipment racks should be as acoustically open as possible, precluding any low-frequency build-up between console and control room front wall. The console, producer's desk, and outboard rack should be as rigid as possible, to avoid low-frequency resonance and just plain old rattling. More often than not however, these surfaces are constructed of thin metal or plywood, with ineffective bracing at best. The console, producer's desk and outboard rack should be physically, and therefore acoustically, as symmetrical as possible. Lack of this symmetry is by far the prime cause of non-uniform acoustic response at the listening position.

A Sierra Audio custom-built disc mastering console.

ELECTRONIC SPECIFICATIONS

Certainly, electronic specifications are important, also. With all the recent improvements in test equipment, you would assume that more-meaningful specifications would be available for comparison. However, there are several problems here. First of all, accompanying the proliferation of advanced test equipment, is the proliferation of ad-

第1スタジオ・コントロール・ルーム

Neve console at CBS/Sony Studios in Tokyo. The 32-in/24-out board is wired for 40 inputs and 32 outputs. (For more about the CBS/Sony Studio, see the March issue of db.

vanced testing methods. To make all this nice data (not information) meaningful, it must be gathered with comparable test parameters. Unfortunately, it's become a numbers game, as you can plainly see, both from the specifications published, and the voluminous articles in the various technical journals and trade magazines. It's really a matter of viewpoint; the seller is interested in absolutes and the buyer is concerned with relatives. To add to the dilemma, every console is a unique creation, with its ultimate performance determined not only by its number of inputs, but by what is connected to it externally, and by how this is done. For most state-of-the-art consoles, electronic interface plays a large role in total system performance. Unfortunately, measurements of your particular console in its final environment are not available before you place the order. So, to obtain meaningful technical performance data, the best you can do is compare sample modules from competing manufacturers and, using consistent and valid test procedures, test them yourself.

THE CONSOLE AS A STATUS SYMBOL

Practically speaking, one of the parameters that most influences value is "status," or client approval. How many consoles do you know of that were purchased because Fleetwood Mac promised to record their next album at a particular studio only if they installed Brand X console? Faced with such awesome realities, what studio owner wouldn't give an important client's recommendation prime consideration? Besides the client, there's also the competition. At times there is a certain competitive advantage to being different, if not better, than the competition, not the least of which is commanding a higher hourly rate to help finance the insanity.

Features and price are two components of value which usually go hand-in-hand. The object of this game is to get the most usable features for the least cost. The question is —what is necessary and what is luxury? Ideally, any function conceivable should be available at the push of a button, no patch cords involved. Consoles do exist today where this is nearly possible.

A procedure called "Dreaming and Eliminating" is probably the best method of arriving at affordable features, as well as putting things in their proper perspective. You start with the maximum and eliminate, based on price, the features which really are "not *that* necessary." Adding back to the cost, are the standard features that don't exist. Most studios need: 1) Monitoring for more than one two-track tape machines, cassette machines, quarter-track machines; 2) An output for an XY scope; 3) Monitor switching for at least three monitor systems; 4) Tape machine remotes; 5) Timer and real-time clocks; 6) Patch bay space for more mic inputs than the console has inputs, a second multi-track, and numerous outboard devices! 7) Provision for cue system monitoring at the console. Certainly these features and any of your own choosing can be made available from most manufacturers as options, but even for the most inexpensive console the cost of options can be sizeable.

The best console in the world is probably your least favorite the day it ceases to work. With their increasing complexity, a console's failure is inevitable. Failure, of course, can exhibit itself in several modes. If the console designer ever set foot in a studio, failure of any single module would not cause the entire console to self-destruct.

In other words, reliability is an important factor in determining value. At current studio rates the non-functioning console costs in the order of $100, $200 or even $300 per hour in lost studio time, plus any direct maintenance costs, not to mention negotiating over who pays the musicians. The best way to determine reliability is to talk to existing owners of the same console you are intending to purchase and trying to get *honest* information from them regarding reliability and the manufacturer's technical support. Also included under reliability is operability. A large percentage of console problems turn out to be operator errors. Obviously, better training is a part of the solution to this problem. But better information, such as a well written Operations Manual, is another. Most studios still deal with a large percentage of independent engineers, many unfamiliar with the equipment they're working with. If the lead vocal track never gets to tape—guess who's problem it ain't? Three hundred volts per micro-second slew rate, is not of much value here.

THE BOTTOM LINE

Not being an accountant, lawyer or I.R.S. agent, I am not really qualified to give advice on the best way to finance a console. But I do know that all of the above should be consulted—either by choice or eventual necessity. There are, in fact, many means of financing an expensive piece of capital equipment, such as a console, with after-tax consequences determined only with consideration of your financial goals and conditions. Don't overlook obtaining competent advice in this area, as it could have significant impact in your console selection decision-making. You may find that those extra features that you thought you couldn't afford, really won't cost you that much money after Uncle Sam contributes his part.

A POST-SCRIPT

A console can be outgrown as soon as you are ready. In many cases consoles are obsolete before they're even turned on. This is merely a tribute to the innovation of console manufacturers and the ingenuity of the engineers using them. It is not, contrary to popular belief, a plot: technology is just evolving very rapidly these days.

It is obvious to all but the most naive, that within the next decade the "start-of-the-art" console will be totally digital. There will be no outboard equipment as we know it; in fact, there probably won't even be a separately-definable console. What we will have is a digital recording system, capable of doing whatever needs to be done at the push of a button (or maybe with even just a thought). ■

Carl Yanchar is vice president of engineering for Sierra Audio Corporation.

DIANE WENDT

The MCI JH-556C Console

Freedom to concentrate creative abilities on the music is the end result of providing the engineer with complete control, through automation.

THERE ARE MANY recording/remixing consoles available today—custom as well as production models. Within this competitive market, manufacturers are constantly vieing to offer "a little bit more" for "a little bit less," in order to expand their share of the market. As a result, consoles have constantly improved over the years, as new technology has become available, and yet console prices have not increased as rapidly as console capability.

At MCI, our JH-556C "Super Console" was specifically developed to satisfy the requirements of the larger recording studio.

The JH-556C is the largest (24-plus-32, or 56 channel) stock production console available today. It is intended for use in 24-plus-24, or 24-plus-32 recording/remixing sessions. The size of a 56-channel desk, and the complexity of a 24-plus-32 track dual multi-track session (where two multi-track recorders are synchronized by time code) presents problems to the engineer and producer beyond those of an "average" session, yet the JH-556C not only allows the engineer and producer to remain in control, but it also allows them to concentrate on the music, not the mechanics, of the session.

This "useability" is achieved by improvements in three critical areas: layout, metering and automation. Left-to-right across the desk are 24 input/output (I/O) modules, master and group modules, and then 32 (or 24) additional I/Os, auxiliary modules and the patchbay. Directly above each I/O is the appropriate Plasma Display meter, which offers three modes of operation—VU, peak and d.c. The automation controls consist of sets of three buttons and three leds, which control and indicate the status of each channel, group and echo return. The simplicity of the JH-50 automation (only one hand is required, to push one button, to initiate any automation operation) helps the engineer to utilize the capabilities of the JH-556C console comfortably.

MAIN FRAME/MODULES

The main frame of the JH-556C will accept up to 67 modules, wired in the configuration shown in FIGURE 1 with a maximum of 56 I/O modules. The JH-556C, like the entire MCI line of consoles, is an in-line desk. All of the circuits needed for one complete microphone channel and one complete remix channel are contained in a single I/O module. The JH-556C is a split console main frame, designed for simultaneous use of 24- and 32-track recorders. It may also be programmed for 56-channel remix capabilities. The standard layout provides for:

- 24 I/O Modules (1-24)
- 2 Group Modules (Group 1, 2, 3 and 4 controls, and echo returns 1 and 2)
- 1 Communications Module
- 1 Master Module
- 1 Control Room Monitor Module
- 1 Studio Monitor Module (Automation Master controls included)
- 2 Group Modules (Group 5, 6, 7 and 8 controls, and echo returns 3 and 4)
- 32 I/O Modules (1-32)
- 3 Modules for options such as Spectra-Vue, equalizers, echo returns

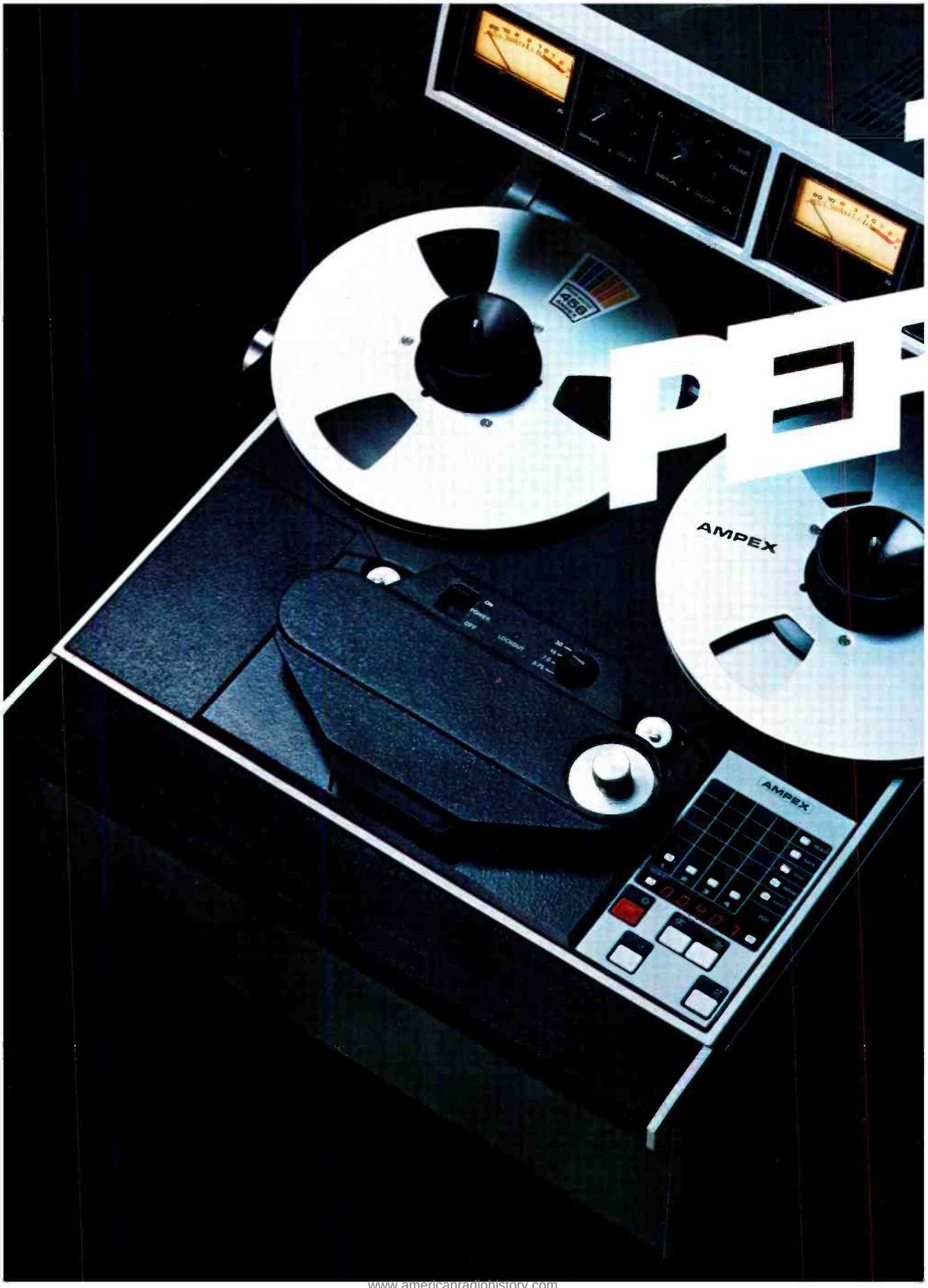
There is a phantom power switching on each I/O (with an optional phantom power supply) and differential-input amps on line inputs, in place of transformers. A minimum of 22 dB head room above nominal level of +4 dBv is available at all mixing points and equalizers in the console. The mike preamp circuit has 26 dB head room above its nominal level of 0 dBv.

OUTPUT BUSSES

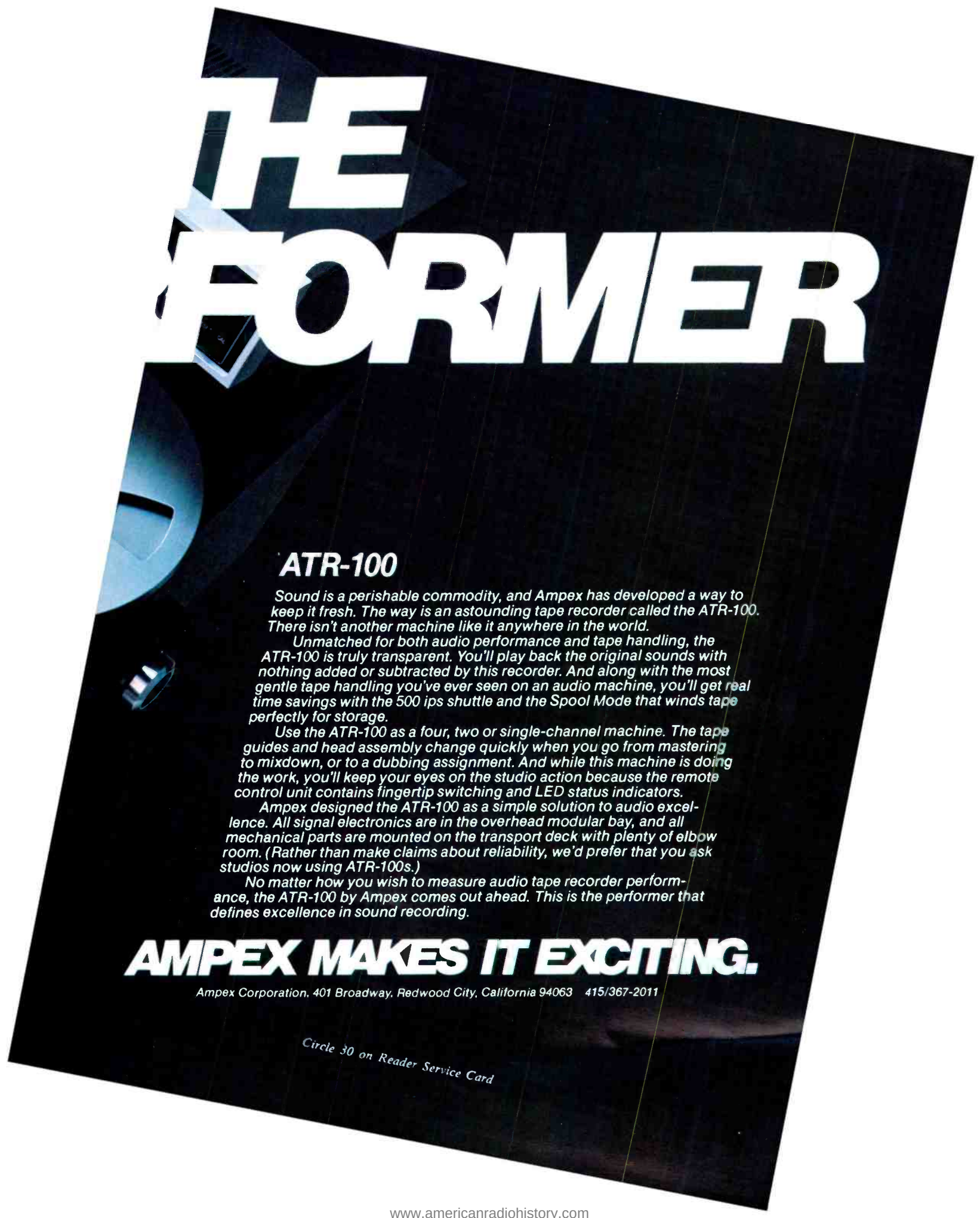
The 32 channel busses are configured to take advantage of the excellent common-mode rejection of modern opamps. The output of each I/O module is bussed as a differential voltage and summed with the output of other I/O modules by a differential active combining network

Up-close on the MCI JH-556C. The master and group modules run diagonally across the picture (lower left-to-upper right).

Diane Wendt is the editor of MCI News & Views.



THE PERFORMER



ATR-100

Sound is a perishable commodity, and Ampex has developed a way to keep it fresh. The way is an astounding tape recorder called the ATR-100. There isn't another machine like it anywhere in the world.

Unmatched for both audio performance and tape handling, the ATR-100 is truly transparent. You'll play back the original sounds with nothing added or subtracted by this recorder. And along with the most gentle tape handling you've ever seen on an audio machine, you'll get real time savings with the 500 ips shuttle and the Spool Mode that winds tape perfectly for storage.

Use the ATR-100 as a four, two or single-channel machine. The tape guides and head assembly change quickly when you go from mastering to mixdown, or to a dubbing assignment. And while this machine is doing the work, you'll keep your eyes on the studio action because the remote control unit contains fingertip switching and LED status indicators.

Ampex designed the ATR-100 as a simple solution to audio excellence. All signal electronics are in the overhead modular bay, and all mechanical parts are mounted on the transport deck with plenty of elbow room. (Rather than make claims about reliability, we'd prefer that you ask studios now using ATR-100s.)

No matter how you wish to measure audio tape recorder performance, the ATR-100 by Ampex comes out ahead. This is the performer that defines excellence in sound recording.

AMPEX MAKES IT EXCITING.

Ampex Corporation. 401 Broadway. Redwood City, California 94063 415/367-2011

Circle 30 on Reader Service Card

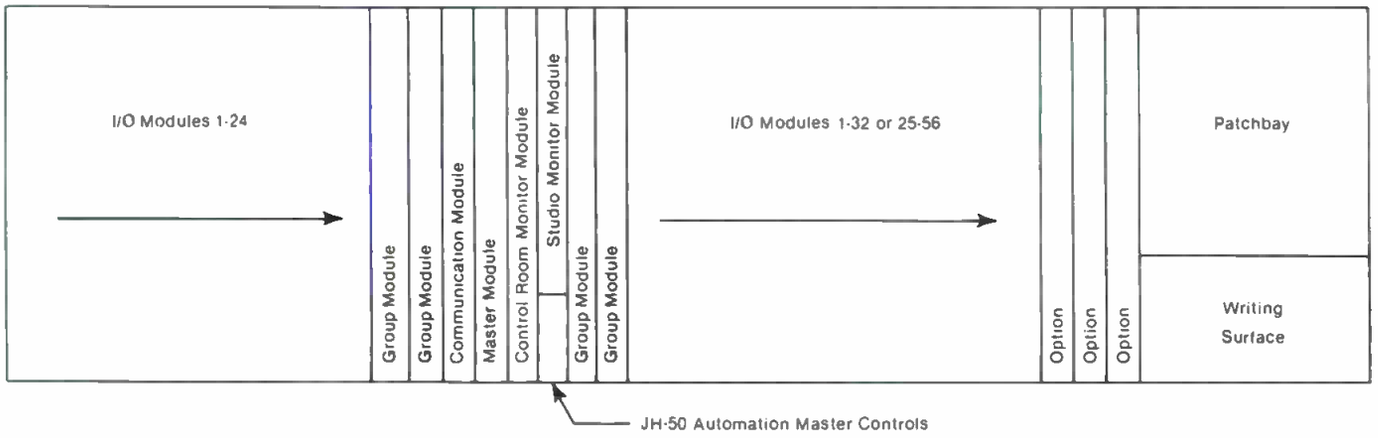


Figure 1. Module layout configuration for the JH-556C console.

Any rf, hum or noise pickup in the bus structure is seen as common-mode by the summing opamp and is therefore rejected. This design achieves a wide-band noise floor as much as 20 dB lower than many competitive consoles.

MUTES

Muting is accomplished by shorting the signal line to ground through a relay contact. This system keeps the noise floor at the lowest possible level by removing all of the preceding circuit noise as well as the audio signal. In addition to the Channel mute button, all Channel, Group and Master faders have a switch at the minimum gain end to activate the appropriate mute.

SEND SYSTEM

The JH-556C has six sends; four mono and one stereo. They are available as effect sends or cue system feeds.

CUE SYSTEM

Two stereo cue systems are provided. These systems may be fed from the stereo mix busses, the sends, the echo returns, from any external source, or from a combination of any of these sources. The six send busses may also be used as additional cue sends during taping sessions when echo or effect sends are not needed. This arrangement allows great flexibility in feeding a specific channel or group of channels into a cue line, to be summed with the complete stereo mix at a lower level if desired. Several of these special cue lines can be mixed at the same time, as well as the echo returns with panning into the cue lines.

EQUALIZATION

MCI's precision EQ circuits may be used either in the channel or in the monitor. They are carefully designed to musical scales, not to mathematical relationships. The switch positions define exact sound control points, which are easy to find and easy to recapture, or to reproduce on another module.

Instead of the usual variable-gain/fixed-"Q" circuits which change the tonal value as the amplitude is adjusted, MCI has designed variable-gain/variable "Q" circuits to maintain the tonal balance as the amplitude is adjusted. The band-pass of each circuit remains relatively constant, so that when the EQ is boosted or cut, the slope of the curve increases to emphasize only the sound desired. This eliminates the need for excessive EQ to achieve a desired effect. (For another approach to bandwidth and "Q" see Equalizers for Professional Recording Applications, by Robert Orban, in the May **db**—Ed.)

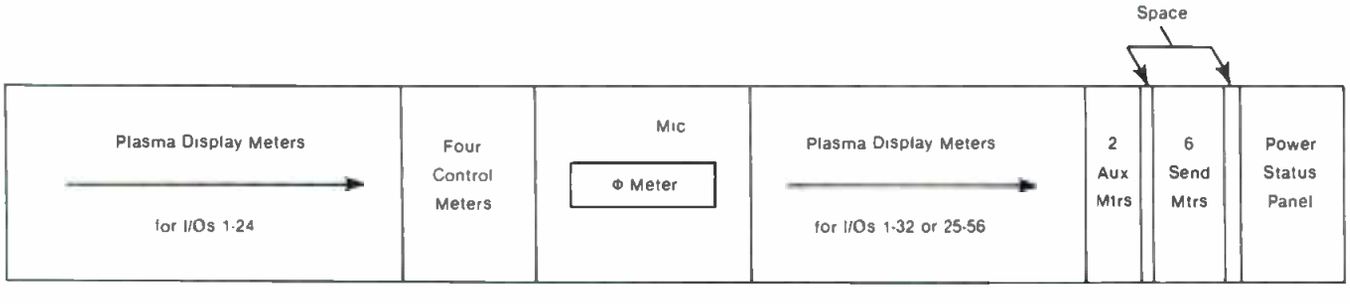
METER HOUSING

The JH-556C is equipped with MCI's patented Plasma Display meters with the layout as shown in FIGURE 2. The appropriate plasma display meter is located directly above the corresponding I/O module in this layout, which is certainly a great convenience for the engineer. From left to right the meter housing contains:

24 Plasma Display meters (channels 1-24)
4 Control Monitor Select meters
1 Microphone and Phase meter
32 Plasma Display meters (channels 1-32)
2 Aux meters (Plasma Display type)
6 Send meters (Plasma Display type)
Power and Status Panel

MCI's Plasma Display system uses a 100-segment neon glow tube to produce a lighted bar graph of the audio level. Cardinal scale markings on the graph are shown by brighter bars. This unique scale-marking system is produced by PROMs (programmable read-only memories), in

Figure 2. The layout configuration for MCI's Plasma Display meters.



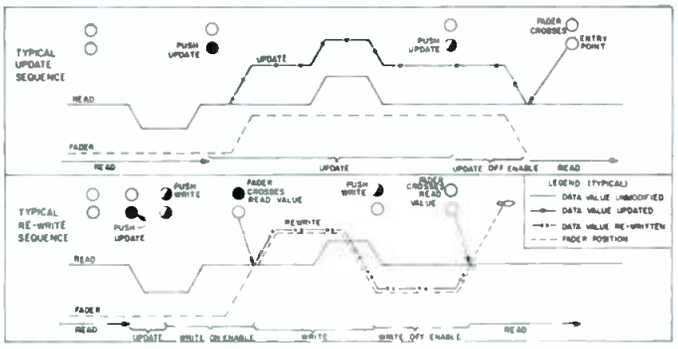


Figure 3. Some of the typical sequences during an automated mixdown.

a different pattern for each mode of operation—VU, peak, and d.c. Each bar graph is scanned (refreshed) over 40 times a second for a highly readable, flicker-free display. Lighted scale numbers appear beside each graph, providing quick identification of the scale being used and the values of the scale markings.

Since the scale is divided into 100 divisions, it is possible to achieve excellent resolution in the upper, expanded portion of the scales. Accuracy is better than ±0.2 dB from —10 dB to full scale. The dynamic range of the meter is: 50 dB—peak mode, and 36 dB—VU mode.

The VU mode scale has the familiar logarithmic markings. A two-pole active filter adjusts the ballistics to ASA standard, matching the popular Triplett VU meter.

The peak mode is displayed in dB on a linear scale. The rise time and the fall time are individually adjustable over a range wide enough so that the meter can be set to match any ballistic standard. Normally, the factory adjusts the ballistics to DIN standard 45406, which is: rise time: —20 dB to O dB in 10 ms, fall time: O dB to —20 dB in 1.5 sec. The range of adjustment possible is; rise time: 2 ms. to 20 ms., fall time 0.5 sec. to 3 sec.

The d.c. mode is used with automation to display the VCA control voltage on each channel. This type of display has been found to be a very important part of the working automation system. Its continuous display of the VCA control voltage on each channel (adjusted to include group and master faders) supplies fader level information which can be read at a glance.

Softly-glowing luminescent panels are a standard feature of the plasma display. They are both beautiful and useful. In addition to identifying the display mode, they light up to show the modes of operation being used elsewhere in the console.

AUTOMATION

The console uses MCI's JH-50 automation system which, despite its sophistication, has been designed with ease of operation in mind. You don't need a technical training course to learn how to use it. In fact, just three pushbuttons (write, update, mute) control the mode switching for the entire console. When a change is to be made within a subgroup of modules, an identical pushbutton set may be used instead. Similarly, each I/O module and each echo return has its own set of pushbuttons, to be used when only that one circuit requires changes.

STABLE AND TEMPORARY MODES

The JH-50 automation system uses four stable modes, and three temporary modes, as described below:

WRITE MODE—This is the first mode used in all mixdown operations, and is the primary mode for storing fader level and associated data. The write mode is used as a master function only to create the initial recorded data track. When the console is in the write mode; all controls work normally. Levels can be adjusted, or groups assigned, just as you would with a non-automated console.

UPDATE is used to modify a mix. The previous mix is played back, is modified by corrections made to the faders, and the new mix is recorded on a new data track. However, if substantial changes need to be made, it may be better to re-write, rather than update. The difference between update and re-write is that the update mode sums the new fader movement information with the information from the previously-written data track. Re-write ignores the previous data track, and writes only the new fader information onto the new data track.

MUTE WRITE performs the customary muting function. When using automation, the VCA and mute relay work together, to provide a musically-pleasing 40 ms. ramped "kill" down to the noise floor, followed by relay closure to ground. When the channel is un-muted, the relay contact release is followed by a VCA ramp back to the fader level. Muting instructions may be independently written during or after the actual mixdown.

READ is the "home" mode of the system. Any portion of the console which is in the read mode is completely controlled by the previously-recorded automation data. In this mode, the engineer cannot affect any of the automated functions. The read mode is entered when all three pushbuttons are released.

TEMPORARY MODES

The temporary modes are entered automatically, and provide a smooth transition between stable modes. For example, when the update button is released, the automatic

Figure 4. The JH-556C—MCI's latest "super console."

temporary mode is indicated by the flashing of the update led. During the transition, the last update value is compared with the original data track. If there's a difference, the update value automatically adjusts towards the read value. As soon as the values match, the system goes directly into the read mode, and the flashing update led goes off. The temporary mode can be bypassed by going directly into another stable mode, and pushing the appropriate button twice.

A very significant difference between the JH-50 system and other systems is that MCI's system operates in real time. No data-packing or priority system is used. The data rate is 9600 baud, and every automated function is scanned and updated each cycle. Each cycle lasts only 102 milliseconds, so it is impossible to overload the system, no matter how many controls are changed at once. Some other systems record changes only, which means that if you try to make too many changes at a time, you may overload the system and lose your data. Also, the number of updates is virtually limitless, in that only 1.6 ms. of delay is accumulated with each pass. The average human reaction time, between hearing and moving, is about 200 ms. You could make 125 update passes while working on a mix before the total accumulated delay reaches 200 ms.

AUTOMATION AT WORK

To get an idea how the JH-50 automation system can actually be used, we quote an article of *MCI News and Views* of July 1977, where-in Karl Richardson described some of his early experiences with the system during a typical mixdown session. Since then, many other techniques and forms of usage of the automation system have been practiced by Karl, as well as other engineers. Karl is senior engineer at Criteria Recording Studios, and a co-producer for the Bee Gees. These were his comments:

"The first "A" data track was written just as I would do a mix without automation. I tried to make all the movements and pick up all the cues required to get the best mix. When I made a mistake, I would rewind and punch-in on this first data track, just as though I were going to

splice my stereo mix. Always keep in mind that it is unwise to punch-in right on a fader movement because the data track will lose some bits of information. (A drop-out in data will play back as the same level you had before the lost bits.) I have punched-in successfully many times without any apparent degradation of data.

"Now that my "A" data track is complete, I get one of the true benefits of automated mixdown: I can listen!

"Often during a mixing session, I am adjusting levels, catching mutes, hearing the band members, and at the same time trying to hear the music. So, instead of recording the mix on a stereo tape, as I would have done previously, I can listen to the multi-track with the automation making all the moves. This speeds up the decision-making process.

"It is apparent that there are some changes to be made in this particular mix. So, I:

Transfer the "A" data to the "B" data track, with certain channels in the update mode.

Move the piano level in the intro and the second verse.

Change the guitar level on a complex lick.

Group all the vocals on a group fader, making them just a little louder throughout the song. (Since the group assignment is not stored as data, I can re-assign any channels to a group at any time—a tremendous advantage.)

"Listening a second time, I decide that I could ride the strings better throughout the tune, so I do a practice re-write. That is, I:

Read the "B" data track.

Put the string tracks directly into re-write, without using the automatic nulling (temporary) mode.

"Then I rehearse the string levels without recording data. As soon as I am satisfied, I record over the "A" data track, transferring "B" to "A" simultaneously with a re-write of the strings.

"On a more complex mix, it is wise not to try to make several complicated changes on a single update. Although you can punch-in on the data track, it sometimes takes less time to listen to the song up to the point of the changes, making successive data transfers each time.

"Some mixes require a building-up process, whereby I store just the rhythm section first, adding vocals, sweetening, etc. on later data transfers. I have done over twenty A-to-B, B-to-A transfers with punch-ins and tight mutes with no audible problems.

"Here are some additional recommendations:

"When you have two open tracks, write your first data pass on both tracks. This is so you can immediately punch-in and punch-out on the second track without destroying one copy of the original mix data.

"Obviously, it is nice to have as many data tracks as possible. If you have only one track open, use an SMPTE or other code generator or locking device to sync your multi-track to a spare four track recorder. This will give you unlimited data tracks." (See SMPTE Time Code Comes To Audio in the November, 1978 **db**—Ed).

CONCLUSION

Technically, the JH-556C console, with the JH-50 automation system, can do almost anything an engineer can dream up. What makes it a "super console" however, is the fact that the engineer can concentrate his creative abilities on the music—not the desk—while he's using it. Thanks to automation, the engineer (and MCI) can stay successful in a market that's growing more and more competitive every day. ■

DOUGLAS F. DICKEY

The Solid State Logic Studio System—Part I

Extending the potentials of control systems integration, the SSL "super console" incorporates an unusually-comprehensive computer automation system.

IN THE PAST FIFTEEN YEARS, there have been enormous advances, both in the quality and in the variety, of tools available to the recording artist. Studios around the world have increased their signal storage capacity from mono, to two, then four, and now, 24, 32 and even 48 tracks of discrete information. Digital technology has found widespread acceptance in studio applications such as machine synchronization, transport control, programmable mixing, and the creation of delays and other time-based signal processing effects. And now, the shift from analog to digital storage media is underway.

While these advances have vastly extended the creative horizons of the recording art, the "progress" which they have brought is not an unmixed blessing. Each new wave of more tools, toys, tracks and techniques has increased the strictly-mechanical, or non-creative, demands placed on the recording engineer. That most of this development has occurred on a piecemeal basis only aggravates the situation. For, as useful as each new device may be in its own right, far too little attention has been paid to the fact that ultimately, *all* of these devices must be combined into a *system* which can be operated comfortably by a human being.

As the growth of recording technology continues to accelerate, and producers and recording artists become more sophisticated, ingenious and demanding, the sad results of this failure to provide an overall "human engineering" context have become apparent. The electronic wizardry of the control room has grown increasingly uncontrollable. Despite the many marvelous achievements of the emerging technology, the current fragmented approach to recording systems design still relegates the recording engineer—and frequently the producer and artist—to a position subservient to the machine.

Douglas F. Dickey is president of Washington Musicworks, Washington, D.C.

A "TOTAL SYSTEMS" APPROACH

Although none of the diverse elements which comprise the recording chain function in isolation, the realities of commerce dictate that these devices are largely developed and manufactured independently. It falls to the console manufacturer to devise the means to link these discrete units into a functional whole.

Fortunately, the laws of physics and the hard work of numerous international standards committees simplify the problems of audio interface between these components. Unfortunately, less help has been forthcoming in the realm of human/machine interface. The traditional approach to console design has been almost exclusively concerned with signal routing, and to a limited extent, signal processing. For this reason, many "new" consoles are simply larger or more elaborate variations on standard themes. To be sure, a great many valuable refinements and a number of important breakthroughs have resulted from this evolutionary approach, but it has suffered from a limited understanding of the console's role in the recording chain.

The fundamental differences in the Solid State Logic System all stem from an expanded concept of the console's potential function in the control room. The key word here is "system." Rather than concentrate exclusively on improvements in audio interface and performance (machine/machine improvements), the SSL Research Group also focussed intently on the potentials of extended *control systems* integration (human/machine improvements).

The extent of the problem is such that even our most highly-skilled and talented artists must waste significant portions of their time and energy coping with totally uncreative, tedious and repetitive procedures. Worse, it is an all-too-common experience to encounter sessions in which one or more key members of the production team are so involved with mechanical maneuvers and manipulations that they have little opportunity for undivided, critical or contemplative *listening*. And that is a serious indictment of the State-Of-The-Art.

In 1974, the Solid State Logic Research Group, Oxford,

England, began its investigation of these problems. The impetus for the project came from the staff of the company's affiliated recording studio, who had felt that many of the restrictions imposed on them by available studio gear were unnecessary, and therefore, unacceptable. While Solid State Logic was not involved in the manufacture of audio equipment at the time, they were (and are) one of Britain's largest manufacturers of electronic control and memory systems. Over the next several years, an intensive day-to-day collaboration between studio engineers and the Research Group's design and production specialists shaped what is now known as the SL-4000 E Master Recording Console and Studio Computer System—commonly referred to as the "SSL."

The SSL "super console" incorporates an unusually-comprehensive computer automation system, as well as many features and functions not usually found in a recording console. In this article, we will explore the thought which went into the development of the Solid State Logic system, and discuss some of the ways in which it changes everyday studio operations.

In the initial design stages, the Solid State Logic System was conceived as a total control command center. This extension of concept was considered in every facet of the console's development, and opened the door to a number of simplifications in the multi-track production process. Working from this foundation, it has been possible to construct a total system which virtually eliminates many of the control room hassles once accepted as unavoidable by-products of the creative technology.

CONTROL SYNERGY

One area that suggested the need for immediate improvement was the synergistic relationship between the console and the multi-track tape machine. While these two units seldom function apart from each other, the extent of their interconnection has been limited to that of the audio interface, and the occasional remoting of transport controls—almost as an afterthought—into spare panel space on the console.

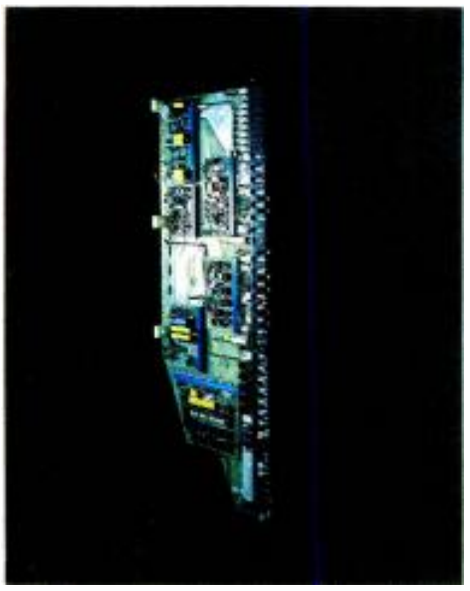
To increase operator convenience, most studios place the tape machine's remote electronics controls adjacent to the console, and some have gone to the extent of mounting the entire remote package into the producer's desk or some similar location.

Solid State Logic has taken this several steps farther, by thoroughly integrating all multi-track electronics controls into a sophisticated tandem switching system, which significantly reduces the time and effort required to set-up and operate the console and recorder for tracking dates, overdubs and drop-ins.

The logic switching network of the console has been extended to embrace control of the master states of the multi-track machine. Additionally, sets of common controls mounted in each Input/Output module configure that channel's monitoring, metering and foldback while simultaneously controlling the safe/ready and record functions for the associated channel of the multi-track. The nature of this system greatly reduces the possibilities of human error, automatically ruling out non-compatible statuses between devices. Of course, it is always possible to override the system's logic for exceptional situations, but now the recording engineer must be genuinely thoughtful, rather than merely careless, in order to overdub an entire string section out of sync.

MASTER CONTROL INTEGRATION

The SSL console has three Master Board States: Record,



An input/output (I/O) module.

Lurking behind the SSL console is a 16-bit minicomputer, with a dual floppy-disc drive system. The computer is typically supplied with 32K bytes of memory.

Replay and Mix—each determined by the selection of a single illuminated pushbutton. (Actually, several other master board states exist, but these three will illustrate our point.) Selection of any one of these buttons configures both the console and the multi-track to the task at hand.

Briefly, when Record is selected, all console channel inputs are switched to accept their mic preamp outputs, the multi-track machine switches to feed its sync line output to the console's monitor line input, and the multi-track's master safe/ready switch is set to ready.

When Replay is selected, all multi-track record functions are inhibited, and the multi-track switches to feed its repro line out to the console's monitor line inputs.

When Mix is selected, the multi-track record functions are again inhibited, and the repro line outs are fed to the console's line inputs. Other complementary functions also occur in each of the states, but these are the basic tandem switching functions between tape machine and console.

INDIVIDUAL CHANNEL ELECTRONICS INTEGRATION

Directly above the monitor fader on each console I/O module, are three illuminated pushbuttons labelled, Ready Tape, Ready Group, and Record. Either of the Ready buttons will switch its associated tape channel from safe to ready. The reason for two safe/ready buttons lies in the nature of their tandem console switching function, which will be discussed momentarily. The Record button's function is self-explanatory; it is simply that track's record button (simple, eh?). It is worth noting that these record buttons are momentary, electronically-latching switches, which enable positive dropping-in and dropping-out.

Several important advantages are gained by this control configuration. The physical position of the controls is con-

Looking past the master panel and alpha-numeric keyboard, the computer's visual display unit (VDU) is seen in the background.

The SSL master panel, shown here in its raised position, contains echo return controls, quad output compressor, plus monitoring and cue controls.

venient, and relates clearly to the other systems controllers, with which they interact. By placing more than one switching function under each control, and providing a logical lamping scheme, instinctive operation of the total system is aided. The engineer is informed in a single gaze of the exact status of both tape machine and console, and can modify that status with a single straightforward action. The benefits of this tandem system go beyond the obvious goal of killing two birds with one stone, however. The real synergy becomes apparent when we examine the more-complex studio procedures, such as that old favorite technique known as the drop-in, also known by the more aggressive term, punching.

SUPERCUE™ DROP-IN CIRCUITRY

Drop-Ins require great concentration if they are to be done well, yet a number of irrelevant and distracting mechanical tasks surround the event. The main problem involves foldback and monitoring.

Up to the point of the actual drop-in, the artist frequently needs to hear both the previous performance (tape sync out) and the new performance (console group out) which is going to he dropped-in. This dual monitoring is necessary in order to permit the artist to synchronize pitch, phrasing and subtle nuances so that the drop-in doesn't sound like a drop-in.

Even in the simplest situation, the engineer is left with a lot to juggle, while repeatedly performing a critical task such as dropping-in between a breath and a hard consonant, then dropping-out on a dotted sixteenth note! Needless to say, an error in timing could destroy an important performance, and a beautiful relationship.

It is here that the Supercue function comes to the rescue. SSL's tandem controls enable the appropriate foldback to be selected and switched while dropping in with a single button. The system is hassle-free, and works like this.

During initial tracking dates, the engineer selects Ready Group plus Record, and the tracks are recorded while the group output feeds the monitors, foldbacks and meters at all times.

During drop-ins, if the artist wishes to hear both the tape and group outputs, while the producer wants to hear only the tape output, the engineer selects Ready Tape. The tape out will be fed to the control room monitors at all times, but the artist's foldback will receive group-plus-tape, prior to the drop-in, and tape after the drop-in.

If both the producer and artist want to hear group-plus-tape prior to the drop-in, the engineer selects Ready Group *and* Ready Tape. This combination also provides appropriate monitoring for situations in which the artist is performing in the control room.

COMPUTER TECHNOLOGY IN THE CONTROL ROOM

Another potentially-liberating force in the control room is computer technology. While attempts to "automate" aspects of audio production pre-date commercial magnetic recording, it is just in the last six years that programmable studio devices have attained popularity.

The Solid State Logic Studio Computer System is not just a mixdown memory, but a powerful integrated computer designed for a number of tasks in the recording studio. As well as mixes, it stores a variety of information, such as reel catalogue numbers, song titles, cut points, track lists, comments on various takes, etc.; and uses this information to control the multi-track tape. In addition, the computer displays all reel information, mix bar graphs, and drop-in countdowns. It will guide the operator through many complex procedures, and provides printed copies of

session information to reduce the record-keeping paperwork which the engineer frequently gets lost in.

Before examining these many features, it is helpful to understand something of the system's physical architecture. The computer chassis houses the computer, a dual floppy disc drive unit, and special interface cards, called controllers. There is a controller for the tape machine; an SMPTE controller which generates and reads SMPTE time code; Fader Input/VCA Output controllers which employ 10-bit converters to read and write up to 150 analog inputs and outputs with a resolution of 0.1 dB, plus separate mute signals; a keyboard and visual display unit controller; a clock/calendar controller to automatically date and time sessions; a controller for the fader status switches and indicators; and a controller for the computer printer.

A video monitor visual display unit (VDU) is built into the meter penthouse of the console. A set of command keys and a miniature alpha-numeric keyboard are built into the master control panel of the computer just below the VDU. These units facilitate communication to and from the computer. A hard copy (paper) printer built into the producer's desk provides printout of track lists, timings and other useful information. The only additional controls are single pushbuttons, one located on each fader panel.

DATA STORAGE SYSTEMS

The three contenders for data storage in the studio are the multi-track tape itself, data cartridges, and floppy discs. For a comprehensive system, audio tape is simply not a suitable storage medium. A number of factors conspire in the studio to produce an unfortunately high error rate.

Crosstalk problems in the multi-track head stack prohibit the digital data from being recorded at anywhere near full medium saturation. The wear and tear which the master tape undergoes inevitably takes its toll on the magnetic oxide, resulting in drop-outs. While these may not be noticeable on audio signals, they can have catastrophic effects on a digital signal. Digital signals also require far more critical mechanical alignment of the recorder, and are extremely susceptible to the minute particles of dust, smoke and grit which love to bond themselves to stray fingerprints, etc. on the master tape.

To minimize these problems, one can introduce elaborate error-correction methods. This reduces the maximum data rate, which is not very high in any case, being limited by the frequency response of the medium. One can then work around this problem by introducing some form of priority detection and encoding technique; however, when many functions are changing at once, such techniques result in a reduction of the refresh rate of the console, introducing variable lags in control response.

Additionally, the number of mixes that can be stored on the master multi-track tape is limited, and a minimum of two tracks is required for even simple mix updates. It should also be remembered that the Solid State Logic System handles many other functions, other than console control signals; with the audio tape already hard pressed just to handle these, it is incredibly difficult to add substantial additional information.

The data cartridge could be employed, but it has one serious drawback in common with audio tape. Both of these are essentially serial media. Retrieval time is therefore very slow, often tens of seconds or even minutes. This is quite intolerable, given the range of features offered by the SSL system. If the producer wants to modify the cue list or take a quick look at the track list, it isn't very cool to have to wait 30 seconds while the cartridge shuttles to that information.

There are of course advantages and disadvantages to each system, and it would take a lengthy technical article on information theory as it applies to practical studio application to cover all points. Suffice it to say that the floppy disc medium, with its excellent accuracy and swiftness of data retrieval (less than 1 second to any data—typically 30 ms), is best suited for the multitude of functions performed by the Solid State Logic Studio Computer System. The ability to store unlimited mixes while tying up only one track of the master tape, the ability to store and instantly retrieve other-than-mix information, the disc's low cost, lower error rate, ready availability and ease of handling (several fit right into the two-inch tape box) all add up to convenience, performance and reliability.

COMPUTER LANGUAGE

Now, let's take a look at what the system does, and how it does it. Basic to Solid State Logic's approach to console automation is the perception that not a whole hell of a lot of recording engineers and producers want to be computer programmers. Many do not care to know *anything* about the subject, feeling that it is irrelevant to our real work, the production of music.

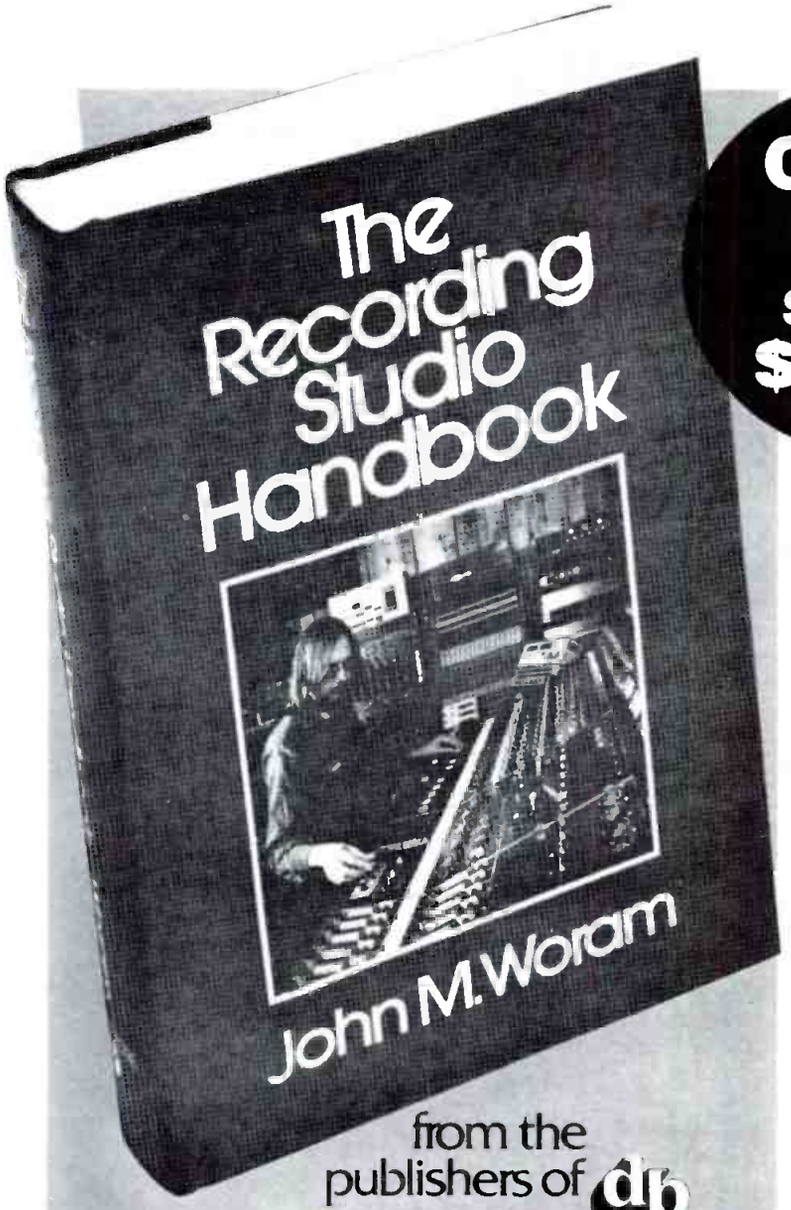
Some "automation" systems deal with this problem by providing only a small part of the potential benefits of computer technology, so that there is correspondingly little new to learn. Alternatively, some systems exploit more of the potential benefits, and place correspondingly-greater learning demand on the production team. Apparently, it is felt that once the producer and engineer catch on to the wonderful advantages afforded by the system, they will be willing to learn complex new languages and techniques in order to avail themselves of the benefits.

SSL does not endorse either of these approaches. Instead, it feels that what is really needed is an unobtrusive system which is easy to use, and at the same time genuinely helpful, so that its presence in the studio will be instantly and warmly welcomed by both staff and visiting engineers, producers and artists who may, at first, be unfamiliar with its use.

A simple conceptual twist was applied to the SSL system specification to accomplish this goal. Rather than limiting the system's overall capabilities so that it would be easy for the engineer to understand the computer, a far-more-sophisticated system was devised which makes it easy for the computer to understand the engineer! Because the computer itself is so much more complex, it is paradoxically much simpler to use.

A fundamental issue is the question of language. The SSL computer is essentially commanded in simple, straightforward English. Commands are constructed in simple sentence form; it is immediately obvious what they mean, even without explanation. The computer is able to interpret commands in an intelligent fashion, so that although there are many functions, they are all easy to obtain. Once you have learned the basic command structure for a few functions, you can learn the rest the first time you try them. Unlike most computer systems, the SSL computer is not rigid in its requirements. Usually, any of several logically-worded commands will produce the desired result. This means that you do not have to memorize exact sequences, or continually refer to the instruction manual. For less-common functions, the system provides prompting through the required procedures.

One key to this operational ease is the use of an extensive field of dedicated command keys. A "dedicated" command key replaces one or more individual words which would otherwise be needed to write an instruction to the computer. The SSL computer command center includes



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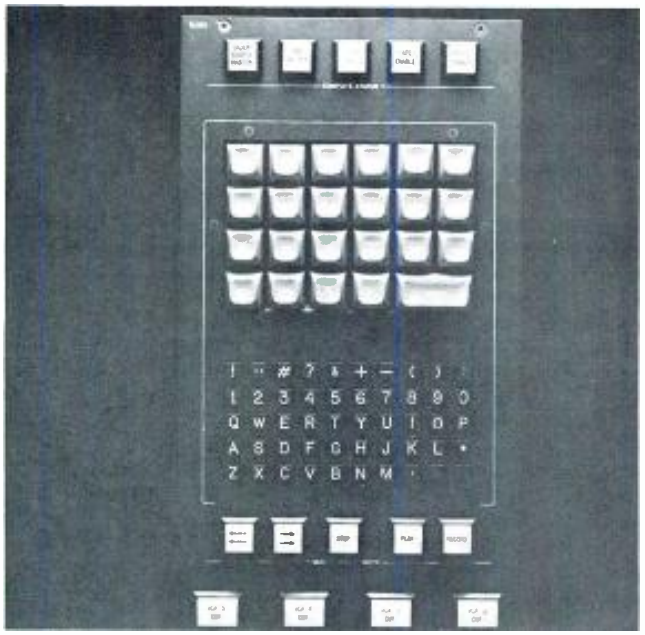

Figure 1. The computer command center. Above the alpha-numeric keyboard are the five master control buttons, plus 23 dedicated command keys. Immediately below are the multi-track remote controls.

23 dedicated keys, five master control buttons, and a complete miniature alpha-numeric keyboard for entering information and commands. Use of the system is best illustrated by examples.

THE ULTRALOCATOR™ FUNCTIONS

In the course of a typical multi-track album production, each of several multi-track master tapes may be shuttled back and forth thousands of times between dozens of cue points. Obviously, some form of assistance in controlling the tape machine and keeping track of these cue positions is desirable, and many microprocessor-based systems are available for that task.

The SSL Computer's Ultralocator™ stores an unlimited number of tape locations in a two-tiered heirarchy. The first tier is called "Title"; the second "Cue."

TITLE COMMANDS

When the initial tracking session takes place, the engineer puts the two-inch tape on the multi-track, and a floppy disc into the right-hand disc drive unit of the computer. This disc, called the "Reel Disc," stays with that two-inch reel throughout the production.

Prior to the first take, the engineer must select the "Transport Enable" button. This gives the computer the authority to control the tape transport. Next, the engineer enters the command line (NAME) (TITLE) Shake It Up (FROM) (HERE). Words in parenthesis are dedicated keys. Other words are typed-in via the alpha-numeric keyboard.

The engineer then starts the tape with the manual Play and Record buttons below the keyboard, and as the count-in to the song begins, the engineer presses (EXECUTE). The CRT will display the name of the song, the start time of the title, and a "?" in the end time column.

At the end of the take, the engineer stops the tape, enters (END) (TITLE) (EXECUTE) and the new tape position is automatically set as the end of the title. To play back the title, the engineer need enter only the commands (PLAY) (TITLE) followed by (EXECUTE).

The next title is similarly labelled, and so on. Note that actual names, rather than numbers, are used. Thus, a useful and complete list of titles, along with their start and stop times is automatically generated. This list is much more meaningful than a list of numbers from 1 through 100, particularly when you are working on a project over the course of weeks or months.

By the time we have finished all of the rhythm tracks on one reel, we will have built up quite a data base. Entering the command (LIST) (TITLE) will give us a display like this:

TITLE	FROM	TO
Shake It Up	0:15	2:30
Colin's Blues	2:40	4:19
Love Song (Take 1)	4:40	7:52
Love Song (Take 2)	7:59	11:16
Crawl Away	11:30	14:08

The command flexibility of the system can now be seen. If we enter (PLAY) (TITLE) S, the computer will instruct the transport to roll to the beginning of "Shake It Up," play the entire song, and then stop. We could command the same sequence by entering (PLAY) (FROM) 0:15 (TO) 2:30. In addition, once either of these sentences has been typed, we could repeat the sequence simply by entering (PLAY) (TITLE), or even just (PLAY).

Unless a command line contradicts them, the computer makes a number of assumptions. For example, it assumes you intend the same target positions as you did last time, the same title as last time, the same cue as last time and so on. Thus, though you can construct complex command sentences, you very rarely need to do this. Most of the time, you need type only one or two keys.

You may also have noticed that the computer accepts abbreviations. In the example above (PLAY) (TITLE) S, it recognized S as "Shake It Up." Had you entered (PLAY) (TITLE) C, the computer would have displayed the message:

This could be Title C:
Colin's Blues
Craw Away
Please Give Enough to Distinguish

Entering CR (EXECUTE) would tell it that you wanted to play Crawl Away, which it would then do.

Titles can be deleted from the Ultralocator memory at any time. Thus, once you decide that you want to overdub only on Love Song Take 2, you may instruct the computer to (DELETE) (TITLE) L S T 2 (EXECUTE). Once this has been done, any time you command (PLAY) Love Song, it will automatically play the correct take.

If you just want the computer to roll the tape to a particular location and stop, a (GO TO) command is used instead of a (PLAY) command. If you want to play the same segment repeatedly, a (CYCLE) command is used. Many other variations are possible. The important point is that all of these variations are simple common-sense English instructions which mean the same thing to the computer as they do to the human operator. The following list of commands illustrates this point.

(GO TO) (TITLE) S
(CYCLE) (TITLE)
(GO TO) 2:37
(PLAY) (FROM) 2:37 (TO) 3:05
(PLAY) (FROM) (HERE) (TO) 4:06
(CYCLE) (FROM) (HERE) (TO) (END) (TITLE)

■

(To be continued)

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■

IRWIN J. DIEHL

Convention Report:

63rd Audio Engineering Society Convention

A highlight of the more than 170 exhibits and over 70 technical papers—as they relate to the practicing engineer/technician.

AUDIO ENGINEERING SOCIETY conventions have become a prime source of information, if not revelation, regarding engineering developments in the audio-recording industries. The number of convention exhibitors and technical papers presented, as well as the sheer number of attendees, grows at an ever-increasing rate.

Our review of this past May's AES Convention in Los Angeles focuses upon those papers and exhibits most relevant to the practicing engineer/technician involved in recording, broadcast, sound reinforcement or related fields.

This report is only a highlight of the more than 170 exhibits and over 70 technical papers presented at the 63rd convention.

THE PAPERS

The papers reviewed here are available from the Audio Engineering Society, Inc., 60 East 42nd Street, New York, NY 10017. When ordering, please include the preprint number. The cost is $1.50 per copy for AES members ($2.00 per copy for non-members).

AN ABOVE THRESHOLD COMPRESSOR WITH ONE CONTROL (Preprint 1505), by Leslie B. Tyler of dbx, could be recommended as a good short course on compressor fundamentals.

Tyler presents an approach for deriving a compressor's gain characteristics, as dictated by present-day studio production methods.

A comparison of linear (constant compression ratio regardless of input level), above-threshold compressors (limiters), and the multiple-threshold devices lends insight into the numerous gain and gain-reduction schemes required of today's well-designed compressor.

The dbx model 163, Over-Easy™ compressor is offered as an embodiment of the design criteria outlined in the paper. A single straight-line control allows adjustment of gain, compression threshold and compression ratio, while the inter-relationship of these parameters is fixed in the design of the device.

Numerous graphs illustrate the "Over-Easy" character of the compressor's gain law, which affords an *easy* or smooth transition along the entire input-to-output transfer curve.

A FRESH APPROACH TO AUDIO CONSOLE DESIGN (Preprint 1503), Brown, Armstrong, Treby, Willis (Neve Electronics International Ltd.).

Irwin J. Diehl, founder of the Institute of Audio Research, New York, is currently employed as an independent consultant. Mr. Diehl has published previously in db, *as well as in* Billboard.

The dbx model 163 Compressor/Limiter features a single control to adjust compression.

Our industry is awaiting with eagerness (and perhaps trepidation) the realization of the all-digital audio production facility. Product planning at Neve is proceeding toward this goal, as is the case with possibly half-a-dozen other audio systems manufacturers. (For example, see our other feature stories—Ed.)

The application of Neve's studies to date is the subject of the authors' paper. The new 8108 series of Neve consoles is introduced as the technological culmination of currently-available resources.

The authors derive theoretical noise and distortion figures for non-inverting and inverting mix-bus configurations, to support the design approach utilized in this console series. While very-detailed descriptions of the system functions and operating features take up a good portion of the paper, there is also a more-theoretical discussion, which provides a good comparison of electronic-versus-transformer input designs.

An important additional feature described in the paper is the centralized analog switching system, with assign and switching-systems-control managed via a built-in microcomputer.

HIGH-PERFORMANCE DELTA MODULATION IS BOTH SIMPLE AND ECONOMICAL (Preprint 1500),

Neve's 8108 console series.

by Richard E. DeFreitas & Peter W. Mitchel of DeltaLab Research, compares the popular PCM digital encoding techniques to delta modulation digital encoding techniques.

The authors carefully evaluate advantages and disadvantages of both as employed in the current breed of digital audio devices. Applications are cited which could weigh in favor of either, depending on overall requirements. The delta modulation scheme for digital encoding would seem to warrant consideration where the advantages of digital audio are desired, but with a cost-effectiveness significantly better than PCM designs.

The cost-effectiveness of delta modulation is derived from the simplicity of circuitry needed to accommodate full-band audio. A simple comparator, a flip-flop and an RC integrator are all that are needed for encoding.

The particular scheme of delta modulation which would appear most suitable for professional audio is described as *Adaptive Delta Modulation* (ADM). Straight delta modulation suffers from noise limitations. Therefore, additional circuitry is added in the adaptive model to vary the comparator's reference voltage as a function of the waveform slope of the input signal.

The ADM technique is compared to that of PCM in terms of sample frequency and bit rate. The paper points out that, at a 50,000 words-per-second sample rate (employed in a 16-bit PCM system), the bit rate is equal to the product of sample rate and number of bits employed. In this example the bit rate would be 800,000 bits-per-second. Since ADM does not employ bit strings as such, but is rather a *one*-bit-word system, an ADM system with a comparable bit rate of 800,000 bits-per-second would also have the same sample rate. The problems associated with anti-aliasing and multi-pole anti-aliasing filters are thus alleviated since the sample/bit rate may be adjusted upwards (re. the 50 kHz PCM sample frequency) without necessitating super-wide-band systems.

This paper offers an encouraging alternative to the multi-thousand dollar PCM systems, since—according to the authors—the ADM described in the paper may be manufactured for about $3.00.

The paper includes several references and a number of graphs and circuit block diagrams. (And for more on delta modulation, see Digitizing Audio with Delta Modulation in the April **db**—Ed).

THE LEDE CONCEPT FOR THE CONTROL OF ACOUSTIC & PSYCHOACOUSTIC PARAMETERS IN RECORDING CONTROL ROOMS (Preprint 1502), by Don Davis, offers a glimpse of a new acoustical design approach which has drawn recent support from a small but growing number of theorists and practitioners alike.

The Live-End, Dead-End theory is one which essentially dictates that the front walls (behind and to the sides of monitor loudspeakers) are to be as acoustically-dead, or absorptive, as possible. The opposite rear walls must be as acoustically-live, or reflective, as possible.

The theory behind this approach, which is opposite to "conventional" control room design, is built upon the psychoacoustic effects of early reflections, as researched by H. Haas.

The mixer's position in the LEDE control room is generally equidistant between front and rear walls, on an imaginary line separating the live-end and dead-end halves of the room. The position—though variable—is critical, since the reflected sound from the rear must be properly "timed" in order to psychoacoustically vanish.

The purpose of the absorptive front walls is to prevent early reflections from combining with the direct sound field from the loudspeakers, causing comb filter effects.

It is claimed that such a properly-treated control room will: a) minimize coloration by room modes, b) allow a fair evaluation of monitor loudspeakers, studio acoustics and/or musical performance, c) psychoacoustically create a larger, more "open," control room than is actually the case.

The concise paper is rather brief in explaining details of the LEDE concept, but makes-up for the brevity with an extensive list of relevant references. The paper also includes a number of photos, graphs and architectural drawings to support the text.

MULTI-MICROPHONE PICKUP OF SOLITARY ACOUSTICAL INSTRUMENTS FOR SINGLE-CHANNEL TRANSMISSSION (Preprint 1491), by Wieslaw V. R. Woszczyck, is an ambitious effort to analyze and utilize the large number of factors which affect the timbral quality of recorded instrumental sounds. This is one of the few published guides on microphone placement or "technique" which is based upon scientific data and measurement, rather than upon subjective preferences.

The paper is a documentation of studies and experiments conducted by the author at McGill University, intended to discover the optimum number and location of multiple microphones applied to solitary acoustical instruments. The results are evaluated by comparison to natural, concert hall type of instrumental sounds.

The author has done an effective job of extracting pertinent information from numerous sources of acoustical data for musical instruments. An extensive bibliography references these sources. The data, consisting primarily of spectral and sound-radiating characteristics of various instruments, is used to support the author's contention that neither distant microphone pickup in a reverberant field, nor single, close microphone pickup in the near field can

capture both presence and full spectral content of a musical instrument. The proper placement of multiple microphones (which need not be identical), is suggested as a means for most "naturally" achieving a desired timbral quality.

Tests conducted with stringed and woodwind instruments were analyzed by means of both listening and instrumentation analyses. These tests seemed to bear results in favor of multiple microphones placed close to the instrument. The tests are supported by photographs of microphone placements and more than two dozen graphs of the instruments' spectra as provided by multiple microphones positioned near the instruments.

The preprint offers a great deal of practical information and many good references. We could recommend it to the student as well as practitioner of multiple microphone technique as a good *logical* guide to microphone placement.

FERROFLUIDS AS A MEANS OF CONTROLLING WOOFER DESIGN PARAMETERS (Preprint 1477), by L. Melillo and K. Raj of Ferrofluidics Corp., describes a number of factors important to the design of woofer systems.

The control of the woofer's mechanical damping coefficient is sought as a means for controlling system bass response. The authors cite the use of ferrofluids as a means for adjusting driver damping in order to fit the driver to a given enclosure volume. An additional benefit of heat sinking and improved power handling as a result of ferrofluid utilization is reported.

The paper offers theoretical discussions of mechanical damping of loudspeakers deriving relationships between ferrofluid viscosity and driver damping. Selections of ferrofluid magnetization levels are suggested as a means of controlling splashing of the fluids.

Overall, ferrofluids are offered as a method of controlling mechanical damping, thereby extending bass response and, under some conditions, providing as much as a two-to-one increase in power handling capability.

A number of references and many graphs and illustrations support the concepts presented in this paper. (For additional information, see The Use of Ferrofluid In Moving-Coil Loudspeakers in the February **db**—Ed.)

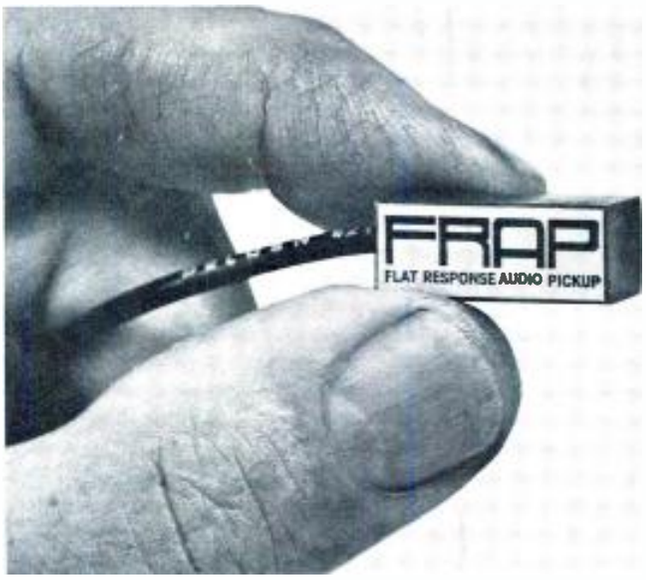
A SATELLITE COMMUNICATIONS, BROADCAST QUALITY AMPLITUDE COMPANDER (Preprint 1481), by Daniel B. Talbot, considers the factors relevant to an accurate frequency response and adequate signal-to-noise ratio through an f.m. satellite broadcast network.

The performance characteristics of an audio compander, which could fit a 120 dB dynamic range into a channel having 40 dB signal-to-noise ratio, is described. The manner and amount of pre-emphasis that could be employed in such an application is explored and an equation for deriving level-path pre-emphasis is given.

Desiring to achieve performance as described, utilizing off-the-shelf components, a computer program was developed to model and analyze compander performance.

The paper includes a listing of the Super-Fortran program as well as several references, a graph and circuit block drawings of the compressor/expander.

AMPLIFYING A SYMPHONY ORCHESTRA WITH PICKUPS (Preprint 1510), by Arnie Lazarus of Frap, reports on the methods and techniques employed in amplifying acoustic (string and woodwind) instruments using Frap transducers.



FRAP three-dimensional pickup.

During a 1977 Emerson Lake & Palmer concert tour, these direct pickups were used to amplify a 65-piece orchestra, so as to integrate the acoustical instruments with the electronic instruments, in a 72,000-watt sound system.

The goals in applying direct pickups where more traditionally, microphones are employed, were: an undistorted sound without feedback; natural (not "electronic") sound; stereo imaging of instruments, and a fast sound check.

Accellerometric, or vibration sensing, and pressure sensing types of pickups were employed. The selection of pickup positions on the various instruments and the mounting of the pickups is explained and illustrated with a number of photos and drawings.

The Technics 4-channel digital audio recording system.

The Space Station SST-282 delay/reverb processor by URSA MAJOR.

THE EXHIBITS

TECHNICS revealed a new digital audio recorder that uses quarter-inch tape, operates at 15 in/sec, and permits conventional, as well as electronic, editing.

The model exhibited in Los Angeles was a four track format—the product of a Technics research effort that would permit from 24 to 32 tracks on one-inch tape.

Features of this PCM recorder are: thin-film magnetic heads which permit narrower track widths and higher track densities, shortened, 1200 Hz data block rate (plus a separate analog signal track to facilitate tape-cut editing), and up to 60 minutes of recording time on a ten-inch reel.

A sample of the specifications are: sampling rate 50.4 kHz; quantization 16 bits linear; frequency response 20 Hz to 20 kHz ± 0.5 dB; dynamic range more than 90 dB; distortion less than 0.05 per cent. Technics, One Panasonic Way, Secaucus, N.J. 07094.

LEXICON exhibited its line of delay devices, as well as the newest addition to the product line—the model 224 digital reverberation system.

The 224 is built around a proprietary computer-based system designed with sufficient speed to achieve high-quality reverberation. A pushbutton remote panel allows complete control over all significant reverberation parameters.

The unit accommodates one or two inputs and can provide up to four outputs. Reverberation times are controlled by the operator in three frequency bands. Decay times may be adjusted from between 0.6 seconds to 70 seconds. A depth control is provided to adjust the apparent position of reverberation pickups in the hall. Lexicon, 60 Turner Street, Waltham, MA 02154.

WHITE INSTRUMENTS demo'ed its Series 4300 active equalizers offering 1/6-octave resolution in the low frequency range. Twenty-eight 1/6-octave controls provide precise control in the critical range of 40 Hz to 894 Hz. The 4300's equalizer bands from 1000 Hz to 16 kHz are provided in 1/3-octaves. All bands feature a 10 dB range of boost or cut.

White's new model 200 microprocessor-controlled real-time analysis system was also introduced. This new analyzer comes equipped with 1/3-octave filters and may be fitted, as an option, with an additional set of 1/6-octave filters in the range of 40 Hz to 1120 Hz. The model 200 features eight non-volatile memories which may be used in part to: store a "snap-shot" of the amplitude of every filter in both peak and average modes; to store an accumulation of maximum amplitudes of each filter; or used to

load a reference curve to be displayed alone or in connection with measured data. White Instruments, Inc., P.O. Box 698, Austin, TX 78767.

The URSA MAJOR exhibit featured that firm's Space Station SST-282 delay/reverb processor. First introduced in Belgium at the 62nd convention, the Space Station offers a reverberation mode with adjustable: initial delay; early reflection pattern; and separate decay times for high and low frequencies.

The multi-tap delay features eight audition delay taps, a choice of 16 audition delay programs and a built-in mixer. The delay programs are used to simulate early reflections and reverberation build-ups in rooms of four sizes. Other programs produce non-recursive comb filters; delay clusters to yield doubling, slap and echo effects, and space repeats that allow repeating the input signal as well as moving it in space left-to-right. Ursa Major, Box 18, Belmont, MA 02178.

SONY INDUSTRIES emphasized its commitments to PCM digital audio and the U-matic video cassette tape recording format.

The PCM-1600 2-channel digital audio system is designed to combine with a video-cassette deck, such as the BVU-200A, forming a professional two-track audio recorder. The unit utilizes a 16-bit linear code to achieve better than 90 dB of dynamic range with +0.05, —1 dB frequency response, 20 Hz to 20 kHz; and a thd of less than 0.05 per cent. The sampling frequency is 44.056 kHz.

The PCM-1600 features balanced line inputs/outputs terminating in XLR connectors. Additional features are: an 8 ohm monitor output; video input/output; composite sync output; as well as two *digital inputs* and one *digital output* at TTL levels.

An important new component added to the Sony digital product line is the DEC-1000, Digital Editing Controller. This system provides a ready means for electronically editing the digital audio tracks as recorded on the video-cassette format.

The DEC-1000 editing controller features: a variable speed control as well as fixed half-speed operation; an editing accuracy to within 90.8 microseconds (4 words); and an editing rehearsal mode. Editing requires two recorders, as in video editing operations, and is aided by a Sony cross-fade feature to minimize edit noises. The cross-fade duration is operator variable from 1 to 100 milliseconds in seven steps.

Two most prominent features of the editing controller are the search dial and the "memory" feature. The system memory stores an approximate edit point designated by the operator with 6-seconds of program in and around that vicinity. The search dial allows locating the exact edit point, in a manner similar to that used in analog tape-cut editing, by rocking the tape back and forth. A digital tape counter always displays tape position in hours, minutes, seconds and milliseconds. A SMPTE time code generator/reader is incorporated to further facilitate editing or allow synchronization to other production elements. Sony Industries, 230 West Hill Place, Crocker Industrial Park, Brisbane, CA 94005.

SONTEC introduced the Compudisk CD-80 lathe control system which can be retrofitted to all Neumann lathes (AM 131, AM 32, VMS 70 and VMS 66) as well as the Scully lathes. The Compudisk system operates on a principal of land control rather than pitch control. A unique computation algorithm allows for optimum groove fitting under all conditions. The system is equipped with a real-time memory which stores the exact detail of the location of the groove wall over a complete one-revolution cycle. Through this means, grooves are interlaced to the maximum extent practicable.

The sophisticated Sontec control system performs its own self-diagnostic checks and generates an error message should a malfunction occur, as well as indicating its probable location. Sontec Electronics, 10120 Marble Court, Cockeysville, MD 21030. ■

Sony's digital editor—the DEC-1000.

MARTIN DICKSTEIN

A Bit More on the 16mm Film Projector

• In the April issue, this corner mentioned that it was important to predetermine the type of film projector that should be used in any set-up. Several reasons were included on why one type (the old manual loading type, for example) has advantages over the newer auto-loaders. Even the later type, the auto/manual or slot loaders, were also included. Shortly after the magazine came out, a letter was received from EPIE (Educational Products Information Exchange) Institute.

To quote part of the letter: ". . . *you may want to note the advantages of the 16mm slot/channel load type projector. Initial loading of film in the projector is relatively easy, and rethreading in mid-reel (after a break in the film) is a much easier task than with the automatic or manual load machines. In EPIE's testing of seven different projectors we also discovered that the slot/channel load projectors were gentler with damaged film than the auto or manual load projectors.*"

Along with the letter came several "EPIE Equipment Reports," fairly thick magazine-size booklets, and a bunch of "EPIEgram: Equipment" papers. Our thanks to Robert A. Farian, technical associate, Equipment Evaluation, for his letter, all the printed material, and the permission to quote from them.

FUNCTION OF EPIE

Before getting into the reports, it might be well to mention what EPIE is so that you have an idea of the operation of the organization and the validity of the information the printed material contains. "*EPIE Institute gathers and disseminates descriptive and analytic information—along with empirical information of performance and effects on learners—about instructional materials, equipment, and systems. Chartered provisionally in 1967 and permanently in 1975 by the Regents of the University of the State of New York, EPIE is a not-for-profit, consumer-supported agency. It has no commercial sponsorships; its income is derived from memberships, from subscriptions to its semimonthly newsletters (EPIEgram: Equipment and EPIEgram: Materials), from fees for conducting Workshops . . . and from occasional non-restrictive grants from private foundations and public agencies . . . all EPIE's services are designed to facilitate the making of informed, rational purchasing decisions about educational products.*"

This quote is from the inside front cover of an EPIE Equipment Report. It also suggests that: "*For membership or any other information about EPIE, write to EPIE Institute, 475 Riverside Drive, New York, N.Y. 10027.*" We strongly recommend that you follow this suggestion. They can surely be of help to you, and maybe you can help them, too.

The Report we received contains, among other information, general notes on test results, practical tests on 16mm projector lamp life and lamp performance curves, evaluation reports on seven projectors that were tested, and a tabulated summary of data in the reports. The concluding Appendix includes a report on a survey of user problems with 16mm projectors. That's quite a lot of information in 48 pages.

MEMBERSHIP SURVEY

We will not disclose the results of their tests, but the book does start with an "Introduction" in which they discuss a recent survey of their members on ownership of, satisfaction with, and needs from 16mm motion picture projectors. The members who replied, 167 of them, collectively owned or controlled more than 12,000 projectors. The members made recommendations for improvements and the discussion lists 10 points in order of frequency mentioned. The list is given, starting with quieter operation and ending with longer lamp life. There follows a discussion of each of the items on the list

including the need for fast forward and reverse while threaded, and mention is made that only the slot/channel load projectors do this conveniently—however, not all of them. A big plus for this type of unit.

To give you an idea of the sort of material published in the "EPIEgram: Equipment," let's take a look at one which had information on 16mm projectors. EPIE provides a member service whereby they can learn of problems and equipment through member "feedback." Projectors of all kinds are used in schools, and usually the equipment is operated by non-professional (in the audio-visual field) people such as teachers, students, etc. Problems arise that the professional AV person may not run into, but should be aware of and not take for granted. EPIE recently asked a consultant to draw up a list of precautionary basics. The list contains some safety tips which even the seasoned AV operator might find of some interest.

SOME SAFETY TIPS

The report starts with the caution that although audio-visual equipment is designed and manufactured for safe use by untrained people, there might be some hazards, especially as the equipment gets older and is used in less than ideal conditions. For example, the suggestion is made that power cords should be removed from the wall sockets before attempting any lamp change, servicing, cleaning, or adjusting. Specific mention is made that power cords should be removed by pulling on the plug, not the cord itself. (Too often the AV expert, so called, forgets or doesn't think, and gets lazy, and one time in the not-to-distant future the projector doesn't work—bad plug on the power cord.) There is also the suggestion that checking the temperature, by touch, both near the plug and at the equipment will sometimes give a prewarning of future trouble.

Another section of the report discusses the "suicide plug," the 3-to-2 adapter and the 3-wire plug from which the ground pin has been removed. With the adapter, the user is cautioned to be sure the "green wire" or "spade" should always be attached to the screw on the duplex outlet. In the case where a 3-prong plug has been surgically made into a 2-prong plug, a proper plug with the grounding pin intact should be used as the replacement. In addition, there is a precaution not to move a projector while in use, or while the lamp and lamp-house are still hot. The reason—glass lenses, enclosures, and filaments of hot lamps are very fragile and any vibration or jarring can cause damage. The suggestion is also made not to replace lamps until lamp-house and defective lamp have cooled. (Those with burnt finger-tips probably have learned this lesson the hard way.)

In one section, it is recommended that during a cleaning or a repair of the interior of the projector, the lenses should be checked for cracks or other damage. Another caution offered is that overhead and opaque projectors have projection lenses that can act as burning glasses if left in sunlight. This type of unit should not be left near a window where sunlight comes in anytime during the sunny daylight hours. And, of course, remove paper or transparencies from these units when not in use.

Another precaution given, something most AV people are only too familiar with, concerns the propping of a projector on a book, box (match book, for instance) or other improvised prop, in order to level the screen image. The suggestion is offered that a series of wooden blocks be made which will offer a sounder tilting device for use anywhere—especially at the front of a balcony where the projector may be tilted downward.

PROJECTOR OVERHAUL

This particular "gram" also includes a list of a dozen checks that should be made, in this case, on a filmstrip projector. It clearly states that this is only a draft of an equipment maintenance checklist, only a beginning in trying to formulate a more inclusive and complete list in the near future. The list includes such items as checking operation of elevator leg(s) to be sure rubber/plastic feet are in position; clean the lens system with lens tissue or blower brush, being sure to handle a heat filter with gloves or a covering cloth; brush-clean transport mechanism and clear aperture of foreign particles, and clean any built-in rear screen with a soft cloth slightly dampened in water.

Other "grams" offer technical information and suggestions on other devices, and test results, as well as up-dates on tests and previously published material. With the correspondence from the many members, EPIE is able to keep up with what is happening in the field, and through EPIE's correspondence by letter and phone with manufacturers, they are best able to receive the latest information on new equipment, as well as answers to various questions and problems.

All we can do is suggest that you correspond with EPIE. The rest is up to you. ■

NORMAN S. CROWHURST

Finding Direction

• Frequent correspondence from readers asks for advice about finding a direction for themselves in this business. The March column prompted some more responses of this kind, of which the following represents them all so well, that I will use it as a starting point for this column:

Dear Mr. Crowhurst:

I just finished reading your article in the March issue of db *on 'The Role of Computerization.' I relate to what you call "Programmed Responders" a great deal and found your article very stimulating and interesting.*

You see, I'm nineteen years old with a grade eight education and at present a sound man for a rock and roll band. I never got along in school, although my grades were good. Now that I've been away from that scene for a few years, I can't help but feel I'm better off.

When I left school I was a programmed responder to a point. Over the past three years though, I feel I've started thinking again and am very happy to have my brain back.

Every spare moment I get I study electronics and find it very exciting. I have a problem with my situation and that is, no direction. I read anything I can get my hands on, from recording studios, to computers, to avionics and marine radar.

I feel I have to focus my learning if I ever want to make any progress. I I don't know which area would be the most prosperous and beneficial. I've also been told that with the electronic industry growing so fast that what you learn today will be outdated and obsolete in two years.

Well, I'm not looking for a programmed response, just some sound advice. So if you think you can point me in a practical direction in this business, please do, because I can't see where it's going or how to catch up with it.

Thanks very much.

"CATCH-UP" SEMINARS

It is precisely because of this phenomenon that people "in this business" spend so much time attending seminars, to "catch-up." I've been to my quota of these seminars, and what happens there is fairly typical. It will start with a well-programmed slide-tape presentation, in which I'm sitting there, waiting for them to get down to the meat. I look around, and others are looking somewhat dazed, or bored. Maybe I ask a question or two, before the first coffee break.

Then, at the coffee break, I find myself the center of attraction, rather than the guy "out front" who behaves as if he knows all the answers. Why is this, I ask myself, as I respond to the questions? I didn't know any more than these people asking questions did, before I came in. Did I learn something so much faster than anyone else, or what?

In conversation, I become quite sure that many of these people are at least as smart as I am, so that is not why they look to me. Perhaps the best quick explanation, is that I showed I knew the right questions to ask. That fellow out front *seems* to know it all, but really he's programmed, just as much as the slide-tape presentation he is there to present. And the other participants seem to sense that.

I have always felt that a good data sheet would do far more good than these time-consuming seminars. But the introducers of new devices, or new systems, seem to find seminars necessary. Why? Because of the "catch-up" problem to which my correspondent referred. A simple data sheet would serve, only if you have the solid foundation in electronics that enables you to know "the right questions to ask."

SYSTEMS DESIGN

My own career in systems started back in the '30s, before it was known as systems design. Although the company's "bread-and-butter" business was in audio, we had a bigger, less-known industrial side, that developed systems for virtually anything. And a company that does that has work crunches, where you hire more engineers to cope, then for a moment, not enough work to keep the new engineers busy, then another crunch, and so on. How do you cope with that?

Well, what I did, was to prepare documentation that systematized systems design. Its primary purpose was to spread the work load, but it also improved our engineering efficiency, and made us unbeatable to the competition—for the time being, anyway. When I left that company, I left that behind—after all, I'd been paid for it.

I mention that, in the present context, because I learned that the docu-

mentation I prepared back there in the '30s, somehow found its way into secret government files, and was pulled out in the late '60s—thirty years later—to be given a "work-out" with much newer technology. If *everything* is out of date, or obsolete, in about two years, then this 30-year old documentation should be, right? That was evidently what they wanted to find out—without my knowledge.

The project was to redesign a system that had proved defective, or inadequate. The project team was assigned to produce, not one, but two, replacement systems: one using the method spelled out in "my documentation" the other using current technological method. I didn't learn about this until 12 years later. After all it was classified.

The result? The system designed, using the 30-year old method, with parts that were not even dreamed of back there, worked perfectly, first time; the other one was no better than the "dud" one it was to replace. I cite this here to show that principles and basic methods do not change, only the superficialities — commonly called hardware and software. So how does this apply to my correspondent's question?

DEVELOPING A PROBLEM

Can I even tell you the answer? It seems to me it is a difference in how one thinks about the whole thing. In those seminars, as in modern technological education, engineers are taught to think in terms of what the hardware and software does. A sort of block-building system, putting together little boxes that do things. When a new system comes along, it's a whole new set of blocks to play with, and they have to "start over."

What we learned in engineering school was a basic problem-solving strategy. So instead of going to a seminar with the question in mind, "What do all the blocks in this new system do?" my mind follows the problem, or a whole set of problems, with the question, "How can this new set of blocks fit into the process of solving these problems?" As an aid to that, I may associate it with other systems, so I can produce an instant, mental "relative merit" chart, as we go along.

To people with programmed instruction background, working out any kind of relative merit comparison can only come after working with each system for a length of time, and sort of observing what happens. Being able to *predict* relative merit—while it is still only a theory—sitting there listening to a seminar, is a virtual impossibility for them. But they have probably had enough experience to recognize when someone else asks the right questions in this area.

BASIC PRINCIPLES

Now, unless I miss my guess, you—the reader who is just reading this page—are thinking of some specific examples from your own past experience, to which this applies. Your experience will probably not be the same as anyone else's who reads this column. But the principle embodied in it will be the same as all of them. Do you get the point? Can you extract what it is all of us need to learn, to get away from the programmed learning mode, into basics that make you so much more effective, in a changing world?

If, instead of sitting here in front of a typewriter, typing something that you are now reading, I was conducting a class, of which you were one participant, you could be taking turns in telling your various experiences, and we could be fitting them together to see how they differ, and what it is about them that fits the principle we are evolving.

Really, that is going about it backward, in a sense. But when you lose your way, sometimes you have to retrace your steps, which means you go backwards over the path by which you got lost. Had you learned principle, and how to apply principle to practice, none of this would be necessary. But now you've got to retrace your way back to that path.

THEORY AND PRACTICE

None of this is so very far removed from the isolation between theory and practice, that has been the central feature of this column over the years. Once you realize that theory and practice are different views of the same world, and thus cannot contradict one another, when rightly understood, you are ready to start "putting it together."

It all went wrong when teachers, at every level, started teaching theory as fact. If evolution teaches we came from monkeys, then we came from monkeys—"everyone" knows that! Or an emitter follower is an impedance changer and cannot distort, because the textbook says so (if you don't read the fine print—conditions), so that distortion must come from somewhere *else*.

Before someone tells me I can't prove that there is a God, let me correct him: what I cannot do, is prove to him that there is a God. I don't have to, because I have already proved it to myself. If he wants to, he can prove it for himself, the same way I did for myself. If he doesn't want to, why should I bother? I know I am alive, but I don't know he is!

We are different from man-made machines, in being able to engage in creative, or constructive thought. Perhaps the person who says I can't prove to him that there is a God is one of the millions today who have been programmed like machines, and thus does not, whether he can or not, engage in creative thought. But as a closing thought, let me ask those who admire that kind of so-called "education:"

You follow B. F. Skinner and others who see education as a form of programming to prepare everyone for his role in society: who designated men like B. F. Skinner with that superior intellect to enable them to designate a role for me? Better yet, how come the creative method I spelled out in the '30s still works in the '60s and '70s, when everything else gets outdated in about 2 years?

I dare you something: try reading the Bible, with an open mind—that is, to see what those who wrote were saying, not to prove it's wrong before you see what it says—and then tell me, if you find something that is not still as true today, in principle, as the day it was written. That's a serious dare: my address is P.O. Box 63, Dallas, OR 97338. ■

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WANTED

WANTED: USED RECORDING equipment of any kind. Expanding studio will pay cash. **Dennis Reed, Box 50022, Palo Alto, CA 94303 (415) 494-9344.**

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db People/Places/Happenings

● Chicago's full time stereo fine arts station, **WFMT,** will be the first radio station in the country to be relayed by satellite from coast to coast, in stereo. 24 hours a day. Using **RCA's** Satcom I Satellite, **United Video, Inc.,** Tulsa, Oklahoma is carrying WFMT on an experimental basis, and plans to implement service to cable systems, pending approval by the Federal Communications Commission. WFMT's signal will be transmitted by microwave from Chicago, along with **WGN-TV** (Channel 9, Chicago), to the RCA satellite uplink station at Lake Geneva, Wisconsin, and then to Satcom I orbiting the earth 22,300 miles above the equator. Transponder 3, leased by United Video from RCA, will be used to receive and retransmit the signals back to earth. Through agreements with United Video, cable systems with earth stations subscribing to WGN will be able to pick up the satellite signals and distribute WFMT by cable to their subscribers.

● Named vice president, marketing for **BGW Systems,** Hawthorne, CA, **Peter Horsman** will be working with BGW's professional and consumer product lines of amplifiers, pre-amplifiers, and electronic crossover networks. Mr. Horsman comes to BGW Systems from **James B. Lansing Sound, Inc.,** where he was professional division manager.

● **MCI, Inc.,** Fort Lauderdale, FL, and **EMI, Ltd.,** Middlesex, England have entered into a licensing agreement under which MCI will manufacture digital tape recording equipment, based on technology developed by EMI. The first machine to be developed under the agreement, the MCI JH-220, is a two channel stereo tape recorder—production models of which are expected by the end of this year. Additional projects under the licensing agreement will include future development of an editing system, as well as multi-channel digital tape recorders—prototypes of both these projects will be completed by MCI by the end of this year, and will go into production early in 1980.

● In an effort to provide more prompt and efficient service in both sales and engineering, **B. Morgan Martin** has been appointed as regional technical manager for **Rupert Neve Incorporated.** Based in Los Angeles, Mr. Martin will handle all console commissioning, service, and technical inquiries in Neve's West Coast operation; and can be reached at 6255 Sunset Blvd., Suite 609, Hollywood, CA 90028 (203) 465-0135. Previously, Mr. Martin was manager of technical operations for Metromedia Television's **Metrotape West** in Los Angeles.

● Representatives from top recording studios in the country convened in Ft. Lauderdale, Florida to hold open forums on the state-of-the-art; and to establish guidelines for the newly formed **Society of Professional Audio Recording Studios (SPARS).** The organization will be dedicated to achieving excellence in the craft, suggesting professional standards, and providing a forum for statements on technical matters affecting the industry. **Joseph Tarsia,** president of Sigma Sound Studios, has been elected interim chairman until official elections are held in November. Regional members of the board are **Bob Liftin,** New York; **Chris Stone,** Los Angeles; **Mac Emmerman,** Miami; and **Glenn Snoddy,** Nashville. Founding companies of the Society are: **A&R Recording Studios,** New York; **Atlantic Studios,** New York; **Criteria Recording Company,** Miami; **Filmways Heider Recording,** Hollywood; **Group IV Recording Studios,** Hollywood; **House of Music,** New Jersey; **Howard M. Schwartz Recording Inc.,** New York; **Kendun Recorders Inc.,** Burbank; **Larrabee Sound,** Hollywood; **Media Sound,** New York; **Record Plant,** Los Angeles; **Regent Sound Studios,** New York; **Sigma Sound Studios,** Phila.; **Soundmixers Inc.,** New York; **Studio 55,** Los Angeles; **Woodland Sound Studios,** Nashville. Membership is open to any recording studio agreeing to follow and maintain the standards postulated by SPARS. Applications should be directed to Kent R. Duncan, Kendun Recorders, 619 South Glenwood Place, Burbank, CA 91506 (213) 843-8115.

● Appointed manager of distribution products for **Switchcraft, Inc.,** Chicago, **Dean E. Cochran** will be responsible for managing the distributor business, development of new distributor programs, and the upgrading of existing distribution patterns. Mr. Cochran was formerly vice president of national accounts for **Mura Corp.**

● **Shure Brothers Inc.,** Evanston, IL, has announced the promotion of **Joseph J. Kaleba** to the position of vice president of manufacturing. An eighteen-year veteran of Shure, Mr. Kaleba will assume responsibility for all Shure manufacturing operations in Evanston, Arlington Heights, IL, and Phoenix, AZ. Prior to this promotion, Mr. Kaleba was assistant vice president of manufacturing.

● Bolstering the marketing staff of **Ampex Corporation's** audio-video systems division, three new sales engineers have been added. **Bob Natwick,** video sales engineer for the northeast region, will be responsible for Pennsylvania and New York state. **Earl Higgins** was named video sales engineer for the southeast region, covering Alabama, western Tennessee, and the metropolitan Atlanta area; and **Tony Dean** was named the audio sales engineer for the northeast region.

● **Susan Stolov** has been named the 1979 recipient of the Julian N. Trivers Internship Program award funded by the **National Radio Broadcasters Association.** A broadcast journalism/political science student at **Syracuse University's** SI Newhouse School of Public Communications, Ms. Stolov produces weekly programs for the campus radio and cable television stations.

● Taking inspiration from western hotels, hunting lodges, and mountain retreats; **Susan Wilson Design Associates** have just completed remodeling **Amigo Recording Studios** for **Warner Brothers Records, Inc.** The design of the studio provides 2200 square feet for a recording studio, control booth, artists' lounge area, circulation, and restroom facilities. The Amigo studio is located at: 11114 Cumpston Street, North Hollywood, CA 91601.

BETTER

That's right. This *NEW* series of 32 commercial sound products is better in every conceivable aspect than comparable Altec, Dukane, J.B.L., TOA, or McMartin engineered equipment. Write EDCOR at 16782 Hale Ave., Irvine, California 92714. (150 watt modular input, modular output mixer amplifier shown).