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THE SOUND ENGINEERING MAGAZINE

SEPTEMBER 1979
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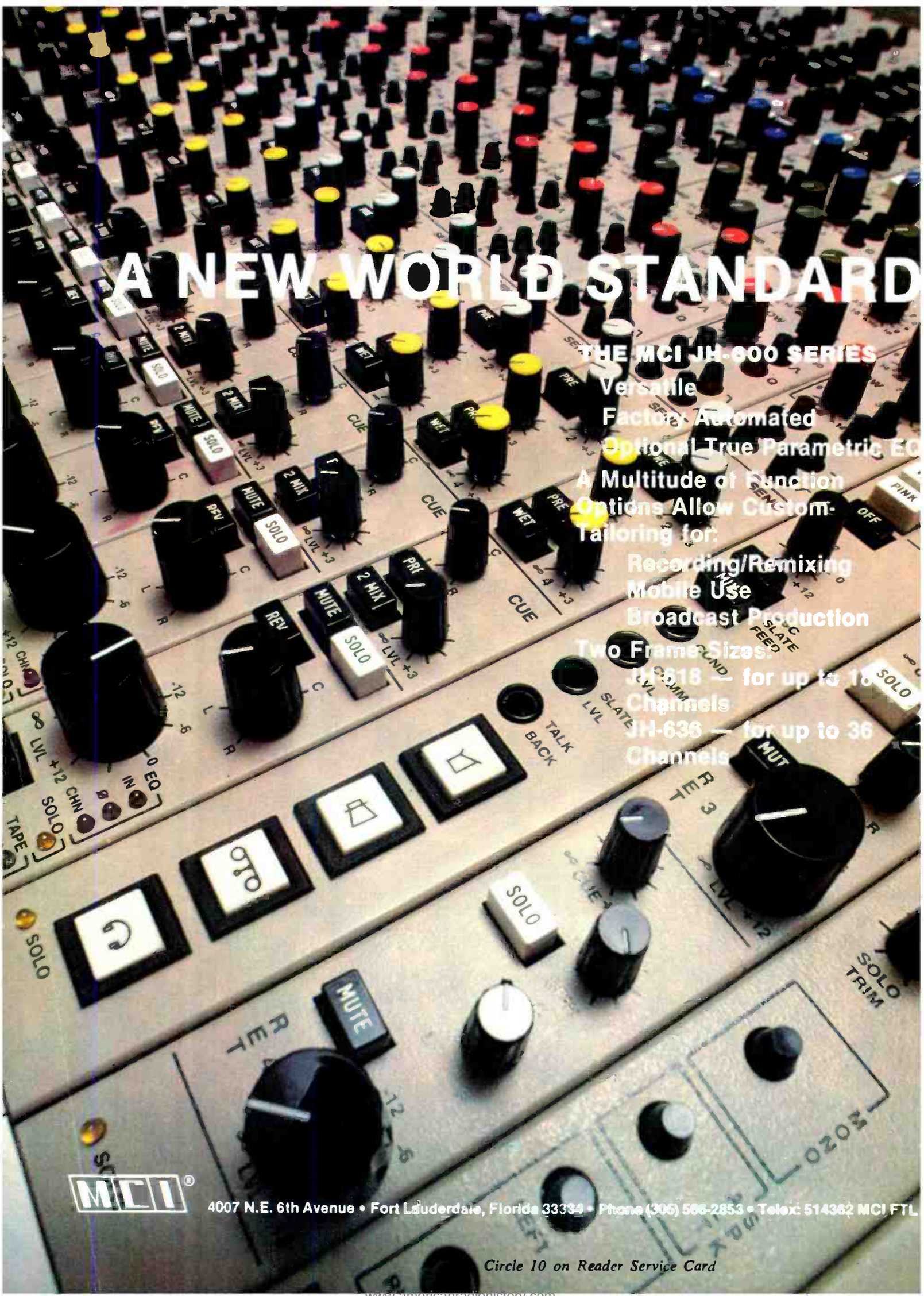
db



Coming Next Month

• If your business is pro audio, you already look to **db** (We hope!) for news about the latest developments in the field. Next month we supplement our regular coverage with some features on the business side of audio.

Thinking about some new equipment? Should you rent it, lease it, or buy it? Would you like to maximize studio income by minimizing down-time? Find out about this and more in next month's **db—The Sound Engineering Magazine.**

About The Cover

• The console is a Neve 8108 computer controlled console. The color print was sent in with John Borwick's article on the recently concluded APRS in London. His story begins on page 50.

THE SOUND ENGINEERING MAGAZINE

SEPTEMBER 1979 VOLUME 13, NUMBER 9

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

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db Calendar

SEPTEMBER

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

2-4 **Institute of High Fidelity Audio Conference.** New York Statler Hotel, New York. The conference is open to manufacturers, dealers and sales representatives.

7-11 **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.

21-27 **121st Technical Conference Equipment Exhibit of the Society of Motion Picture and Television Engineers (SMPTE)**, Century Plaza Hotel, Los Angeles, CA.

27 **1979 Sound Business Show**, Ambassados Hotel, Los Angeles, CA.

NOVEMBER

2-5 **N.Y. AES Convention**, Waldorf-Astoria Hotel, N.Y.C.

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.

db Letters

To The Editor:

Reference "A.M. Proof-of-Performance" by Patrick S. Finnegan, **db**, May 1979. I have no arguments with his article.

I do, however, have one suggestion concerning his presentation. "Noise and Carrier Shift" seem to be relegated to a lower position on the Proof totem. My suggestion is that noise be given priority and measured *first*.

While there is no guarantee, once the noise is reduced to the lowest possible, and acceptable, level the rest will usually (at least more easily and more likely) fall into place.

After the noise is reduced, *then* run the fidelity (frequency response) and distortion tests.

Carrier shift is pretty much an animal and world unto itself. I've found most of the problems due to *lousy* (emphasis mine) power supply design, especially the use of skimpy power supply components. Marginal power company transformers are a possibility but unusual. More likely is inadequate wire size from the switch fuse box to the transmitter.

In over thirty five years of broadcast engineering, mostly as Chief (except for my first year and a brief stint as Assistant Chief of a larger operation some years later) I've found the above procedure the best sequence.

Get that noise down first!

L. Edwin Rybak
Chief Engineer
WALL, Middletown, NY

Patrick S. Finnegan Replies:

Ed Rybak is correct when he stresses the importance of a high noise factor affecting most of the other measurements. I did mention this several times in the article, although without the same amount of stress.

I did not intend to imply that Noise and Carrier Shift measurements were of lesser importance than the other measurements. The sequence given is the same sequence in which I make the measurements. Since there are fewer Noise and CS measurements, it only seems logical to catch them as the sequence proceeds along.

I am with Ed, however, in that I do make a noise measurement *first thing*,

Index of Advertisers

fact:
some veteran products do their job so well, for so long, for so little money it's a pleasure to reintroduce them.

Through the years, Shure microphone mixers have gained the reputation of being the practical, efficient, economical way to increase the flexibility of public address, sound reinforcement, and paging systems, as well as tape recorders using multiple sound sources. In fact, they are used in almost twice as many studios as the next most popular brand . . . with good reason.

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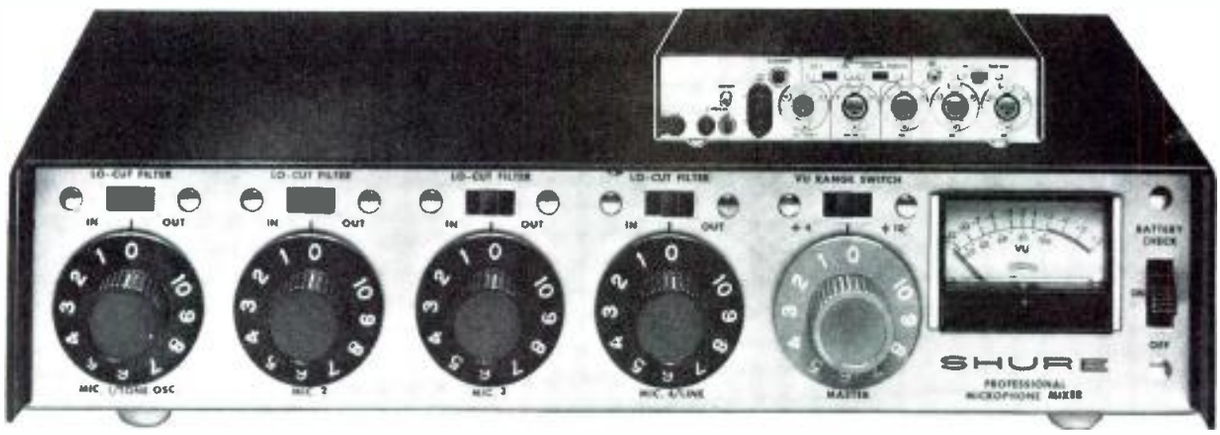

The Thoroughly Professional M67 Mixer . . .

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but this is in a few preliminary spot measurements I make of the system. If these spot measurements indicate that things can expect to check out normal, *then* I dive into making the actual proof measurements. Actually, I had covered this aspect in a column quite some time ago "Preparing for Proof" and didn't want to expand the present article to a greater length than it was, by including some of that previous material.

TO THE EDITOR:

I have read your magazine for a number of years. Now that the journals devoted to broadcasting have gone to pot, why not devote more space to cover a.m.-f.m. radio, since you have a great deal of interest built up with your "unknown engineer." (See "Letters to the Editor," in the September, 1978 issue of db—Ed.) I prefer articles like those by Pat Finnegan, with the practical approach. Theory is necessary, but how much will you remember if you don't put it to use? I have been in electronics for a long time, and broadcasting since 1950. I don't think I would have time to write a feature article on a regular basis, but I would certainly like to contribute.

Most of us are working at one of the smaller stations. And I think it safe to say that most of us are practical engineers, who know Ohm's law and all the rest, and are able to sort out the mis-information from the way it really is. On the other hand, we are not the super-geniuses that many lay people think we are. The need for exchange of ideas and practical problem solving seems to be greater today than it ever was.

For quite some time now, the equipment manufacturers (some of them at least) have neglected to pull all the important information about their gear in their manuals. And if you have a problem and call the factory, most likely you will be talking to someone who doesn't know as much about it as you do. Sound familiar?

The fellow who said he had just discovered limiters for f.m. should waste no time in buying one. We had the old RCA BA-6A on f.m. and could barely be heard 15 miles away with 3-kW E.R.P. Now with a high grade receiver the sports announcer cues his commercial 100 miles away. The operator pushes the cart button, of course.

We chose the Discriminate Audio Processor by Dorough. Basically, what the "DAP" does is defeat the pre-emphasis on the exciter input. It does this by splitting the audio spectrum into 3-channels, pre-emphasis, compressing, de-emphasis, peak clipping and then on to a line amplifier. All operated by a regulated common power source. All this sounds a little weird or "far out" at first, but after you read the manual and work with it a while, it all becomes quite clear. We retained our f.m. sound and added "the sparkle" that is being exposed today. And the sound is very transparent—not to mention doubling our average modulation peaks without overshoot.

There is one word of caution. If you are a purist, you will need to turn the input to the clipper down considerably and the line output control up to compensate for the reduced level into the peak clipper board. This is a very simple adjustment which is best done while listening with your critical ear. If you're a hard rocker, the factory adjustment is quite adequate and will make your station sound louder than the aforementioned adjustment. You are encouraged to fiddle with the level adjustments to each channel, until you get the sound you want. Should you have a problem, Mike Dorough is as near as your telephone—and quite helpful and concerned about your signal. The overall results with the f.m. "DAP" are fantastic and well worth the price.

If the engineer at some other station is doing something that makes his signal come pounding in at a longer distance, don't be so quick to criticize. Maybe he has a clean, crisp, clear sound along with that extra punch. If so, you most certainly will want to know just how he is accomplishing it, so you can start pounding in too. Each engineer likes to think that his station is the best. And each engineer has to make his own judgement as to how much compression, how much clipping will be permitted—consistent with a crisp, clear sound. Obviously, if too much processing is used, distortion sets in along with a poor signal-to-noise ratio. Perhaps even audible pumping, swishing, etc. My own preference is this: It must sound crisp and clear first, and above all. This, in itself, goes a long way towards loudness. Then if we can just get more of this on the air, with a uniform consistency, we have a stronger signal as well. Some compromise between the various conflicting factors must be effected to achieve what one will consider optimum. But with today's equipment, this is not hard to do at all.

I have a lot more to pass along. A remote antenna current meter device for a.m. radio which I have found to be quite helpful and many other little gems. But, for now, I have to get back to work.

GENE VINSON
Chief Engineer
WJDB Radio

(continued)

A funny thing happened on the way to the States

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TO THE EDITOR:

After reading some of the recent articles on unidirectional microphones with a cardioid directional pattern I find that the origin and early history of these microphones is almost completely neglected. The unidirectional microphone with a cardioid directional pattern was developed five decades ago. On May 1, 1931 I delivered a paper entitled "A Unidirectional Ribbon Microphone" at the Meeting of the Acoustical Society of America in Cleveland OH. The abstract of my paper is published in the *Journal of the Acoustical Society of America* Vol. 3, No. 3, p. 315, 1932. The unidirectional microphone, as described in this abstract, consisted of the combination of velocity and pressure ribbon microphones. The abstract gives the response as $R = R_0(1 + \cos\theta)$, R = response of the microphone, R_0 = response of the pressure section and the velocity section for $\theta = 0$, and θ = angle between the direction of the incident sound and the axis of the microphone. This is a cardioid directional pattern. The microphone was commercialized in late 1933 as the RCA 77A Unidirectional Microphone. The microphone was an instant success. The RCA 77A was followed by the RCA 77B, 77C, 77D and 77DX. A composite sketch of the RCA 77 Series of unidirectional microphones with a cardioid pattern is shown in FIGURE 1.

Figure 1.

The RCA 77 Series of microphones dominated the high quality unidirectional market in radio. sound motion pictures, disc recording, sound reinforcement and television for more than two decades. The success is attested to by the use of the RCA 77 Unidirectional Microphone as a symbol on the letterhead by one of the three major broadcasting networks over this period.

From the standpoint of performance. with regard to smoothness of response, frequency range and directivity the RCA 77 Series of unidirectional microphones are equal or superior to most of the present counterparts.

HARRY F. OLSON
Princton, NJ

(continued)

SUBSCRIPTION RATE CHANGE

Effective with the September, 1979 issue of **db—The Sound Engineering Magazine** there will be a change in the subscription price. The new rates will be:

Domestic	
1 year	$ 9.00
2 years	$16.00
3 years	$22.00
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1 year	$10.00
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3 years	$23.00
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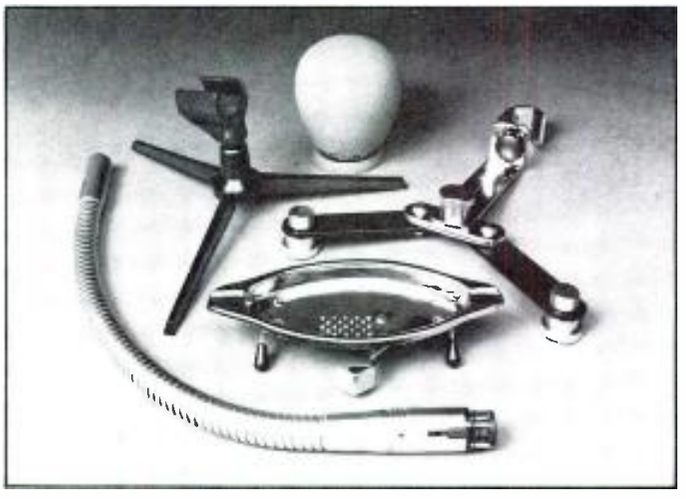

seven M 713. One of our unsurpassed studio condenser mics. Modular system; accepts different transducer capsules and power supplies. Gold-vapored mylar diaphragm for high transient response. Mu-metal shield. Temperature and humidity stable.

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To The Editor:

As there have been some rumours according to which the impedance matching circuit in Ortofon's cutting amplifier is taken for a simple output transformer, I should like to have the following contribution published under "db Letters" in **db Magazine:**

Impedance Matching in Cutting Amplifiers

To avoid any misunderstanding about the functioning of the autotransformer used in Ortofon's cutting amplifier this will briefly be described in this letter, and at the same time it will be demonstrated that the autotransformer is made on the basis of an air coil, for which reason it is not an ordinary output transformer. It constitutes part of an impedance matching network, which is necessary if the maximum output of the amplifier to the cutterhead is required in the complete audible range. The part of the impedance matching network necessary to explain the function is shown in Figure 1. At low frequencies C can be considered a disconnection, for which reason the transformer is ineffective. The current passes freely between "in" and "out," only loaded by the copper impedance of the coil. At high frequencies C shortcircuits one coil end, giving zero, and the circuit has the effect of an autotransformer. The breakdown frequency is placed so that the transformer becomes active at the frequency where the impedance from the self-induction in the cutterhead drive coil exceeds the d.c.-resistance of the drive coil. If the impedance matching mentioned is not arranged for, and a rise in the output signal therefore not obtained when the impedance in the drive system increases, the full effect of the amplifier will not be available where it is needed. Thus, it can be mentioned that in Ortofon's cutterhead 500 Watts are absorbed by cuttings at 30 cm/s at 20 kHz, but only 50 Watts at 8 kHz. It is therefore important that the amplifier has impedance matching for the cutterhead.

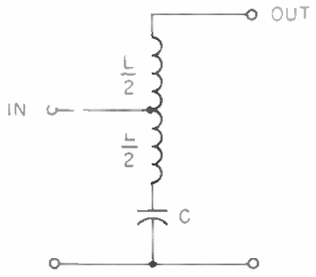

Figure 1.

Finally, it must be mentioned that impedance matching must of course be provided for by means of an air coil, which has been taken into account, as distortion would otherwise be introduced.

F. Nygaard
Ortofon Manufacturing A/S
Copenhagen

To The Editor:

With reference to Barney B. Rogan's inquiry on page 2 of the June **db** regarding the "singing flame," I believe the answer is actually William Duddell's "musical arc" or "singing arc." Many consider this to be the first electronic musical instrument (the first oscillator). Duddell began experiments around 1899. There is an excellent article, "Music in Electric Arcs," in the New York *Times*, April 28, 1907, page 7. I would also highly recommend "The Evolution of Electronic Musical Instruments in the United States," by Thomas Lamar Rhea (Ann Arbor: University Microfilms, 1972). The extensive bibliography would provide more references to Duddell's invention.

Reynold Weidenaar
New York, NY

To The Editor:

A paper entitled "Electro-Thermal Loudspeaker," presented by Robert T. Dicke, April 18, at the 1969 Engineering Exposition, should help **db** reader Barney B. Rogan find more material on the Singing Flame, as he requested in the **db** Letters of June, 1979.

I would also suggest the *Scientific American*, in which there was an article on this matter, sometime in the past ten years.

The 1800's work of Becquerel should also be of interest to Mr. Rogan.

R. A. Greiner
Professor of Electrical Engineering
University of Wisconsin

To The Editor:

Regarding the "Letters to the Editor," June, 1979 wherein Barney Rogan is inquiring about the "Singing Flame."

Please refer Mr. Rogan to the *AES Journal* 17, No. 3, June 1969, page 312.

R. G. Kent
President
Audio Applications
Chepachet, Rhode Island

■

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PATRICK S. FINNEGAN

db Broadcast Sound

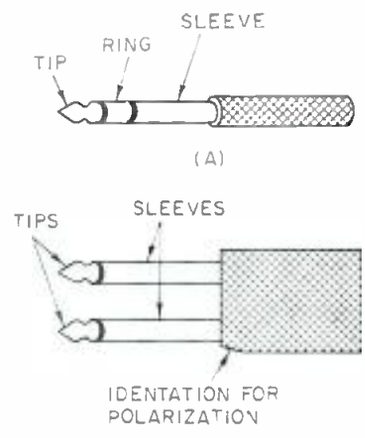
The Audio Patch Bay

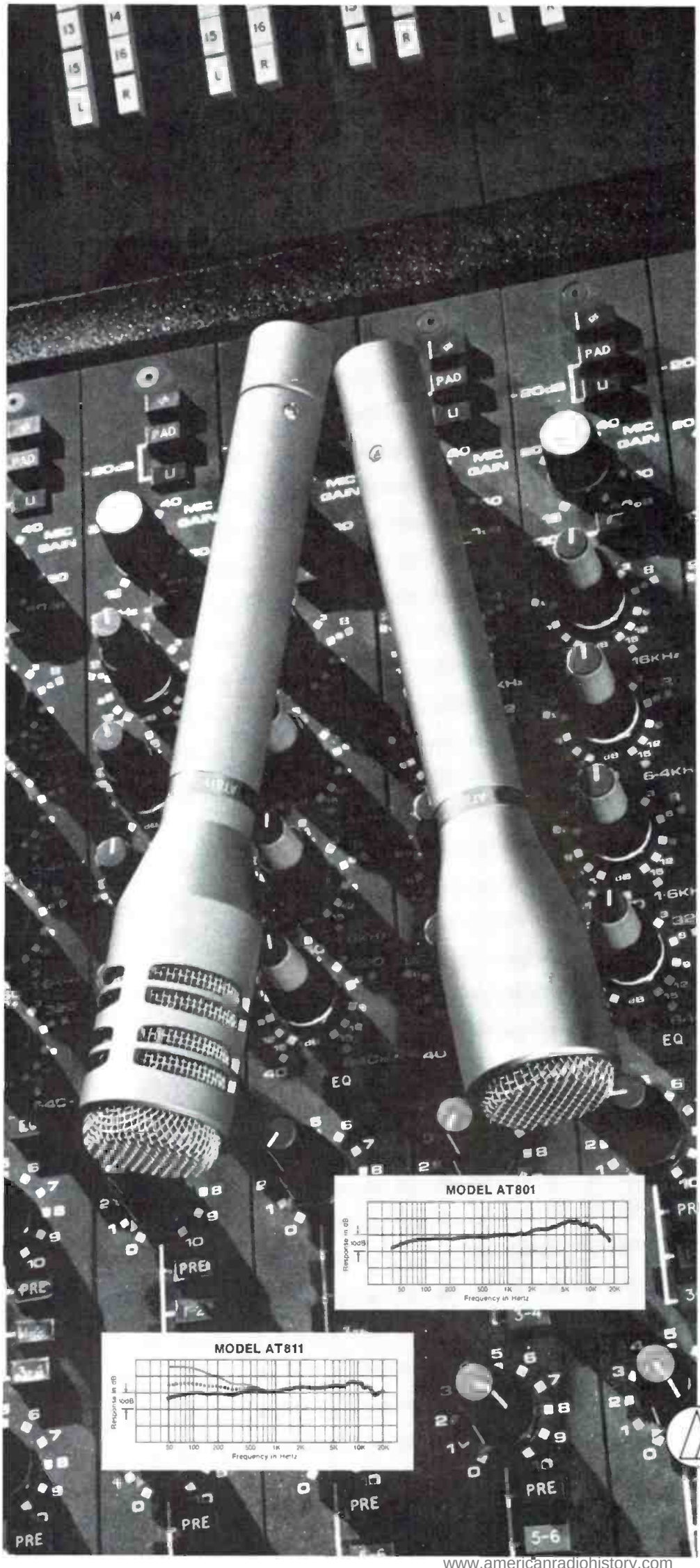
• There is a tendency to think of audio quality only in terms of microphones, amplifiers, speakers, and similar audio units. The broadcast studio system however, contains many elements which contribute directly or indirectly to the quality, as well as the convenience in producing the audio end product. One such element is the audio patch bay or jack field. Although the patch bay does not contribute directly to audio quality by itself, it certainly does contribute to the ease and convenience of producing the audio product. Since jacks are employed in the majority of the audio pathways of the studio, they can detract from audio quality by introducing noise, distortion, and rfi into the audio; and in some cases where faults develop in the jacks or patch cords. Because of its important position within the system, the patch bay should be installed with care, and given a certain amount of maintenance from time to time. The convenience of the patch bay to operations, testing, and trouble shooting problems far outweighs any problems it may introduce, by itself, into the system. This month we will discuss some of the aspects of the audio patch bay and its installation.

GROUNDING

The jack field is ordinarily installed in a steel cabinet rack along with other audio units. Even though the majority of the studio audio circuits route in and out of these jack fields, each rack should be considered as a single unit or component of the system, and be properly grounded by a single ground strap to the station's common building ground. A continuous 4-inch wide copper strap should extend from the top of the rack to the bottom, and on to the building ground in the trench or duct below. The strap should be electrically bonded to the building ground, and in the rack it should be bolted to the rack at several places, making bare metal-to-metal contact at each point. This strap becomes the main ground for that rack.

Figure 1. The two common types of patch plugs. (A) is the single system plug, and (B) is the double system plug.





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Every jack in the jack field should be bonded to a common ground which connects to the main rack ground strap. Most jacks are so constructed that the ground terminal of each jack protrudes at a slight angle from the audio terminals, and are slotted so that a bare #14 gauge copper wire can lay across the ground lugs of all the jacks in a horizontal row. This ground wire should be soldered to the rack ground strap and the ground lug of each jack in the row. Good grounding at the jacks is very important. A poor ground at this point can easily result in the introduction of hum, noise, crosstalk, and rfi into the audio signal.

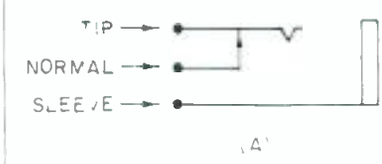
WIRING

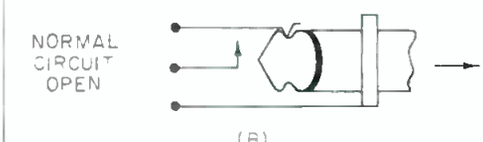
By considering the entire rack as a single unit of the system, it is only logical to wire it internally as a single unit, and then interface this unit to the rest of the system by audio terminal blocks at the base of the rack. All the jacks in the jack field should be wired permanently to one side of the interface terminal blocks: the opposite of the blocks for external connections to the system. Although wiring the rack in this manner does entail considerably more work initially, it provides long term benefits. If the blocks were omitted and the external wiring brought directly to the jacks, as changes are made over a period of time the jack field would soon become a mess—terminals broken off, excess solder shorts, and loss of shield grounds control is possible. By the use of terminal blocks however, changes can be made on only one side of the blocks (which are designed for changes), and the internal wiring is not disturbed.

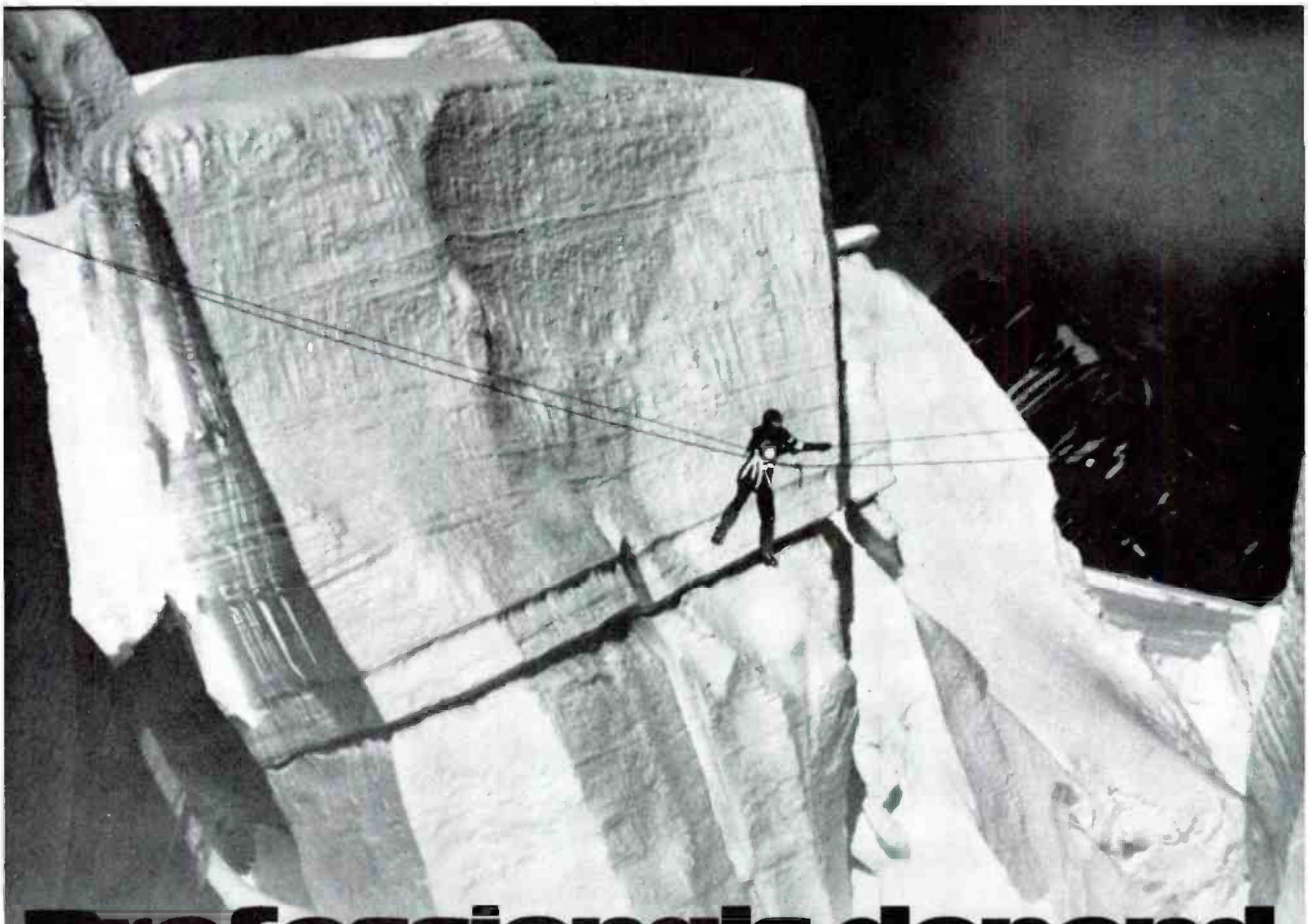
Fixed internal wiring from the jacks to the interface blocks allows for more direct control of shield grounds. All the wiring between the jacks and terminal blocks should have the shields grounded at the terminal blocks only. The shield should be cut off at the jack end of the cable since the jack frames themselves are already grounded by their own ground bus. By leaving the cable shield open at one end and grounding at the opposite end, an incomplete path is presented so that ground currents cannot circulate in the shields.

The wire cables used for this internal wiring should be of the type which has a very high percentage of shield coverage to prevent crosstalk. This is permanent wiring, so use the best. But shielding alone cannot prevent crosstalk in all cases, especially here where a variety of widely different amplitude signal levels will be present, for example, microphone levels, speaker levels, and so forth. We can assist the shielding by separation of the different signal levels at the blocks and jacks, and separating the individual cables according to signal level and grouping into larger cables. Do this for the main amplitude levels, such as low level, mid-level, high level and speaker levels. Another common practice is to separate the a.c. power circuits from the audio circuits by running the audio cabling up the right side of the rack, and the power up the left side of the rack.

Figure 2. The circuit comes in on the tip and goes out on the normal. A patch plug as in (B) will open the normal and pick up the tips.







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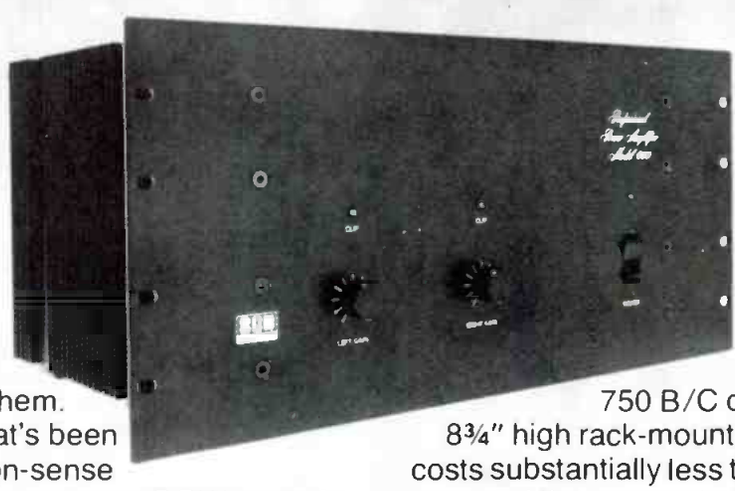
BGW has earned a reputation for building superbly engineered products ... massive heat sinks, large safe operating area, redundant output stages, welded steel modular construction are all synonymous with a BGW product.



We are now proud to introduce a new cost-effective 175 watt per channel power amplifier... the Model 600. It's a quality basic power amp, built around our super reliable 750 B/C output modules. It's in a big 8¾" high rack-mount package so it runs cool and costs substantially less than a 750C. It's a quality BGW amp and the answer to the professional who wants BGW on a budget.

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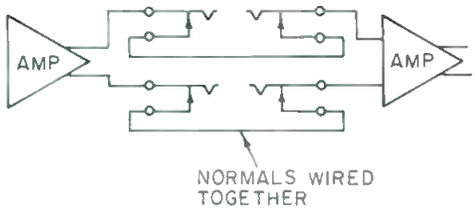


Figure 3. Two pairs of jacks offer greater flexibility. Wire the normals together.

SOME TERMS

There are many different jacks and plugs that can be used for an audio patch bay, but the two most common types are the single and double jack styles—the double jack the most common. In the single system, both sides of the audio and the shield are carried through the single jack, and the patch plug is a single prong device. This is similar to the old telephone switchboard plugs and jacks. The three points of contact on the plug are called the *tip, ring, sleeve:* the tip and ring are the audio, the sleeve the shield. In the double jack system, each jack carries only one side of the audio, so two separate jacks are required, and correspondingly the patch plug has two prongs. The contact points on this plug are the *tips,* and *sleeves:* the tips for audio, the sleeves for shield.

Jacks may be obtained with a variety of contacts and configurations. These are similar to lever switches without the lever key. The jacks are operated by the insertion or removal of the plug. The most common contact arrangement for the broadcast patch bays are a single make-break contact in each jack. The circuit is completed through the jack (make) contacts. When a plug is inserted, the circuit opens (breaks) and the plug tips pick up the tips connection of the jack and routes the circuit out onto the patch cord. The outgoing part of the jack circuit which is broken is called the *normals.* Both the tips and normals connection appear at the rear of the jacks so wiring can be connected to the jacks. When wiring, the important thing to remember is that the patch plug picks up the tips of the jack when it is inserted, so the circuitry should be wired with this in mind. For greater flexibility—especially for important circuits use a double set of jacks. Connect the normals of each set to the normals of the other set. Wire the incoming circuit to the tips of the first set, and the outgoing circuit to the tips of the second set. In this manner, the incoming signal can be routed over a patch cord to some other circuit on a temporary basis, or a signal can be introduced to the outgoing circuit at that point with a patch cord. This provides considerable flexibility in temporary patch-ups, testing and trouble-shooting problems, and temporarily "borrowing" an amplifier for some other use without actually physically moving the amplifier to another location.

Still greater flexibility can be accomplished for some circuits by wiring additional jacks in parallel to the original circuit jack. In this instance, the tips of the jacks are wired together. Jacks so wired are called *multiples* of the original jack, and they allow for monitoring or measuring the circuit without breaking the circuit. Care must be taken so as to use a bridging device such as a test instrument or earphones so as not to load the circuit.

INTERNAL DEVICES

A number of devices can be mounted in the rack and wired directly to the jack field only, for example: equalizers, transformers, and so forth. These provide additional convenience for operations, testing and maintenance.

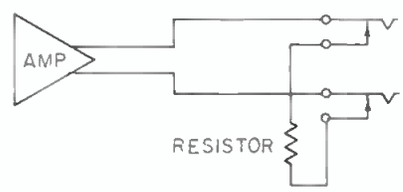
Devices can also be mounted on the jacks themselves, for example, a short or a terminating resistor. The resistor will provide a quick termination by patching it to the circuit desired (and for testing, a short is often desired). For most utility, use the resistor value most commonly needed for this purpose, such as a 600 ohm unit. Whether wiring up a termination or a short, connect these to the tips of the jacks so they will be picked up by the patch plug.

In a similar, but different, application, a terminating resistor can be connected to the normals of a jack while the circuit is attached to the tips. This will keep the circuit terminated until a patch plug is inserted which will lift-off the termination. When the plug is removed, the circuit is automatically terminated again.

Transformers, rack mounted, with their input output windings connected to jacks are very handy. Both matching and bridging types should be used. When more than one transformer is so wired, the only precaution is that the operator not only select the proper transformer for the desired task, but that he also plug into the correct jacks for the input output windings of the same transformer!

Still another convenience for those stations with several patch bays separated by several feet or in another room, is the use of *tie lines* between racks. These are nothing more than one or more audio circuits that appear only on jacks in the respective bays. These lines avoid the use of very long patch cords, or temporary cabling stretched out across the room.

Figure 4. Wire a resistor across the normals for self termination of the circuit in absence of a patch plug.



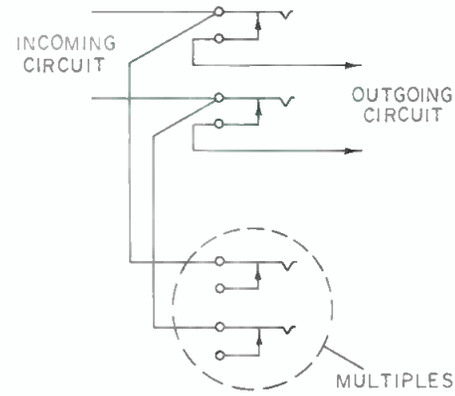


Figure 5. Multiples allow monitoring the circuit without opening the circuit.

PROBLEMS

A well installed patch bay will seldom present any major problems by itself. The problem areas that do arise will be the contact points in the jacks and patch cords. The contact points can oxidize over a very long period of time. This oxidation is a resistance that can open the circuit, or in a few cases, distort audio through the jack by capacitive coupling. The contacts are somewhat self-cleaning by the operation of patch plugs being inserted at different times. But a jack that has not been operated for years can oxidize. The contacts can be cleaned with a small burnishing tool, sometimes a piece of paper drawn through the contacts, and in many cases by simply inserting and pulling a patch plug out several times. The majority of the problems will show up in the patch cords; which see considerable use and are subject to flexing and other damage, as well as oxidation of its contacts. The plugs and the cords need regular maintenance, as well as polishing the plugs. The jacks should be cleaned of dust from time to time.

RECAP

Many elements make a contribution to the end audio product and its quality (or lack of it). The audio patch bay allows for considerable flexibility in operation, testing, and maintenance of the system. Since the jacks are used in most of the audio pathways, special care should be taken in the installation to prevent them from deteriorating the audio quality. Some maintenance is required from time to time, and the patch cords need regular attention. ■

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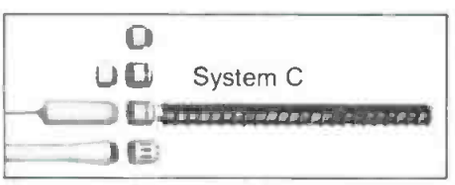

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NORMAN H. CROWHURST

db Theory & Practice

Economics of Audio, Etc.

• Recently a letter came that was typical of many I receive: some audio hobbyist wants to modify a system to do something different, and would like to pay me as a consultant to design the change for him. This is a very tricky type letter to answer, if one is to avoid offense. What he asks seems so simple, just redesign the system a little, so it does something extra.

From where he sits, he knows enough about audio (and this comment I am sure could apply in almost any field you would like to name, but ours is audio) to figure it out like this: the system he has cost, say $500. *If* some manufacturer had just had the forethought to design it "right"—how he now wants it—it would probably cost, at most another hundred bucks, $600. So if he pays me $100, and buys $100 worth of parts, he should have it, right?

DEVELOPMENT COSTS

Not quite. When he buys that system for $500, what he does not realize is what the prototype, that was designed by an engineer, cost. By the time the manufacturer had paid all development costs, it was at least $100,000. But nobody buys an individually-built prototype. The designer who worked on it was not thinking that was the only one he was going to build. Even though he might not have been a production engineer, the production engineer put in his two bits, to make sure the finished item could be made, in quantity, for a marketable price—the $500.

And the marketable price must include the development costs. That $100,000 gets spread over—the word is "amortized"—the production run so that, if they make 10,000 of this item, it amounts to only $10 each. If they make 100,000 of them, the development cost comes down to $1 each. By far the biggest cost now is made up of parts and labor. And even there, cost is cut: by buying in quantity, parts that might cost a total of $1000 for "one off," may come down to $250 in quantities to make them by the thousand.

The same thing happens in labor costs. In fact, it is mainly due to what happens in labor costs, that parts costs can come down so. One off must be virtually handmade. Quantities can have a production line set up for them. And many operations can be mechanized. If a special part is required, the machinery to make that part in quantity may cost thousands of dollars, but that too gets amortized, because it makes them in thousands.

Some say, refer him to the manufacturer. But the manufacturer cannot be bothered. If he gets enough inquiries, maybe he will initiate a development program that will come up with a new model to suit that demand, at the price my letter writer expects. But the manufacturer will not make "one off" just for that one inquirer—he can't afford to.

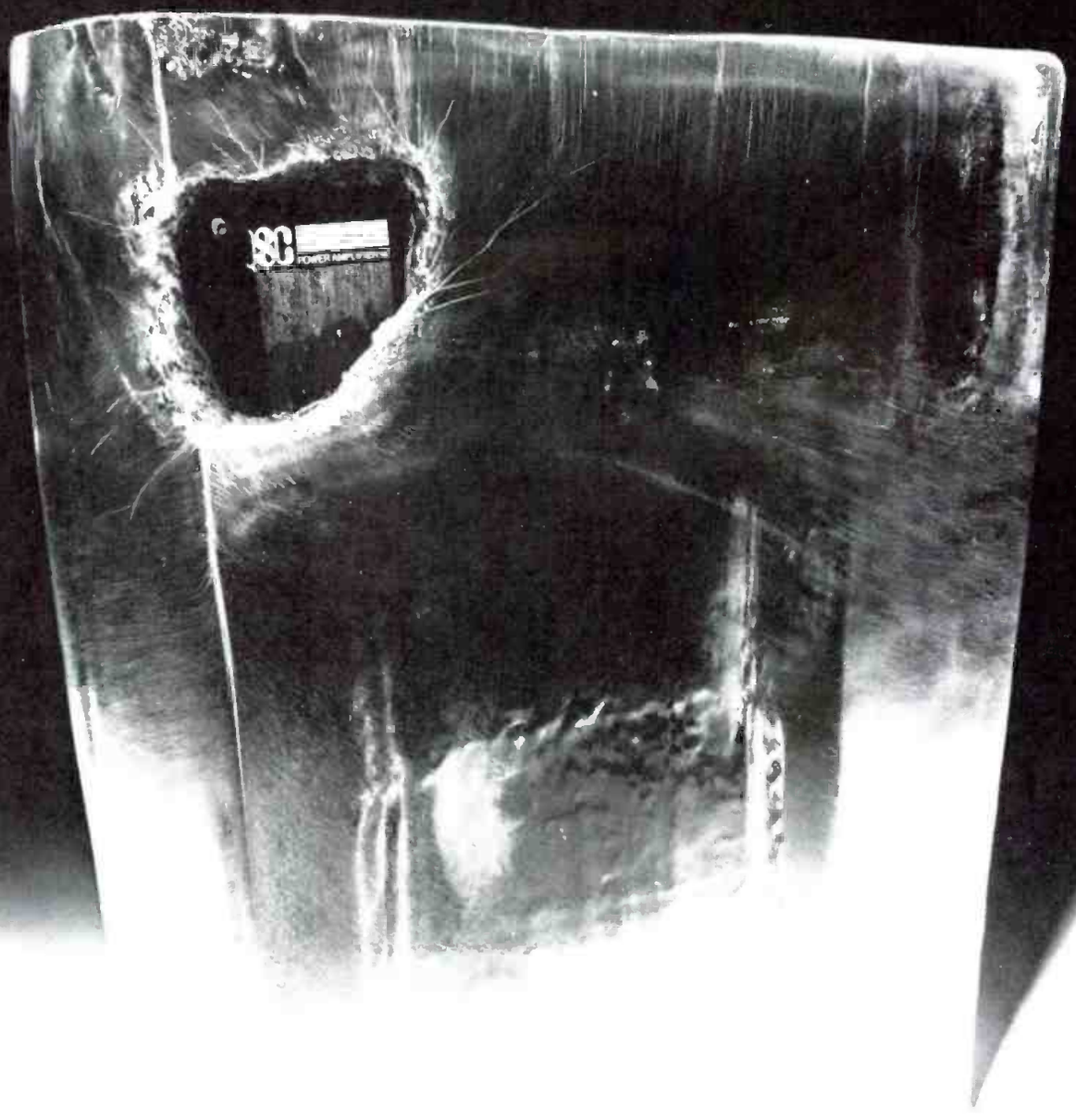
A similar inquiry comes my way about books, or articles, or magazines. Well, you are reading this in a magazine. Suppose this magazine answered all your questions, whatever they might be. If you happen to be a person who is reasonably typical of thousands of others, some magazine may be able to afford to do that. But if you are one who thinks quite individually—which in itself is good, we need more like that—a magazine that would do that for you would sell, maybe a dozen copies.

First, there are various printing processes, just as there are various processes for making printed circuits, or etched boards. For small runs—say around a hundred—one process will serve, because it has a small set-up cost, but the per-copy cost is higher. For large runs—in the thousands—another process is better, which may cost ten times as much to set up, but the per-copy cost brings the overall cost down.

Let us illustrate with some figures. Suppose the set-up cost is $100 and the per-copy cost is $1 each, for the small-run process. And suppose the set-up cost is $10,000, and the per-copy cost is 10¢ each, for the large-run process. We could be talking about anything: parts for some product, printing for a book, or magazine—anything. Let us make a comparison table, so you can see how it works out: *(See page 18)*

Of course, in a practical situation, that is oversimplified. In reality there would be more variables. But you should get the picture. And of course, there are often more than two choices. But consider the magazine—it has so many subscribers. Those subscribers, plus the advertising revenue, pay the salaries of the

PLAY IT COOL.



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1000	$1,200	$1.20	$10,100	$10.10
10,000	$10,200	$1.02	$11,000	$1.10
100,000	$100,200	$1.002	$20,000	20c
1,000,000	$1,000,200	$1.0002	$110,000	11c

editors and other staff members and, hopefully, pay the publisher for his trouble in producing such a publication.

ONE VS. MANY

Now, if you want a publication that tells you everything you want to know, as one particular individual, and it happens that very few other people want to know exactly the same things you do, then if such a publication started up, it would not sell many copies. Can you afford to pay the editor's salary and that of his staff, just for producing the magazine you want? No, you were thinking in terms of buying a magazine at a price comparable with others you see, that sell by the thousands, weren't you?

And the same with your special item of equipment. You know approximately what they cost, as produced and sold at the stores, so you figure on that basis, not realizing how that cost was arrived at. Audio technology benefits too from other electronic technology. That is why we are seeing an increasing amount of digital audio. There, the same principle has been compounded. Before transistors, Dr. Harry Olson, of RCA Labs, produced a synthesizer. We remember some of his demonstrations. And we also remember the responses of those present.

If you judged the performance of Harry's equipment against a Moog, or other modern production synthesizer, it just didn't "hack it." A lot of people listened long enough to realize that what Harry was playing "wasn't even music" and walked out. They were the unimaginative ones. Harry didn't have transistors. He worked with those old fashioned thermionic devices called tubes, that came in evacuated glass bottles. That is all there was — no transistors.

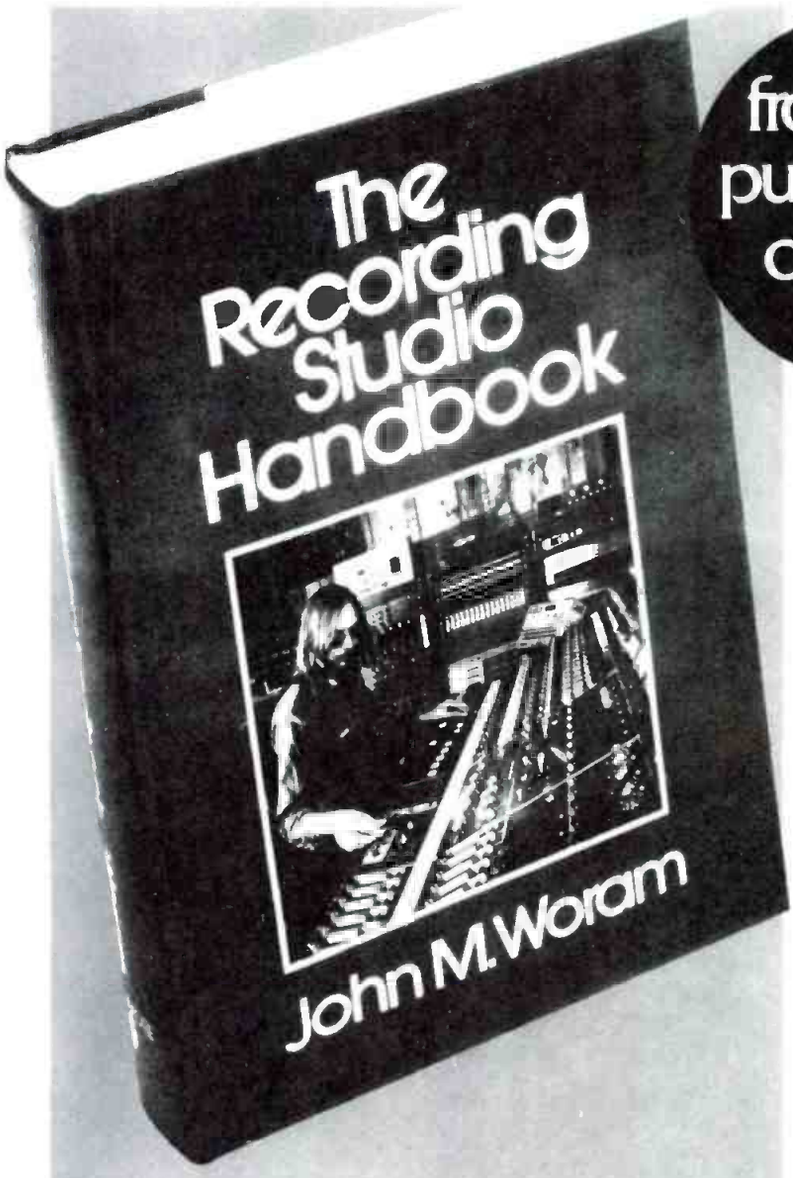
The cost of Harry's equipment was out of sight, but RCA made tubes, so they could let him have as many as he wanted. And he produced something that enabled imaginative people to look into the future, just as those who looked at the old Nipkow disc envisioned future television. Perhaps they thought tubes would get cheaper and smaller, perhaps they foresaw something like transistors.

When transistors first "came in" they weren't even competitive with tubes. They were a novelty, useful because they were so much smaller. Using transistors, Harry's equipment, which occupied a quite large laboratory, could be built (the electronics of it that is) in a volume about the size of one of Harry's tubes. But cost-wise there was not much difference, yet.

COST VS. QUANTITY

In the first place, tubes were as cheap as they were, because production techniques had been developed to cut costs. But compared to a transistor, there is a lot in a tube. So the solution to the cost factor in transistors lay in quantity. As soon as equipment, costing millions of dollars, had been built to produce transistors in multi-millions, their price came down.

But it did not end there. The next step was integrated circuits. In Harry's synthesizer, the biggest part of the cost was undoubtedly wiring all that stuff up. Etched circuits started a whole new ball game. Now parts were just dropped into place and mass-soldered. Production applied to wiring! But one thing seemed fairly obvious; tubes have got to have a vacuum around them, to work. Transistors, the working part of them, are extremely tiny.



from the publishers of db

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Most of the space in a piece of equipment using separate transistors, is taken up with connections, even though etched circuit "wiring" had scaled this down tremendously, compared with the old tube days. How small could transistors be made?

INTEGRATED CIRCUIT DESIGN

This thinking led to integrated circuit design. Now, on a single "chip" small enough to sit on your finger tip, you can have hundreds of semiconductors, along with all the wiring needed to build the system you want. All of Harry's synthesizer could be put in one of these chips! Not that anyone would want that and, of course, for that reason it would not be economic to do so. There is the secret.

Chips, or ics, must be made in some configuration, to serve some system purpose, that will sell in quantities enough to amortize the set-up cost. Once an ic has been set up, its cost is ridiculously low. You get all those hundreds of transistors, all "wired together" in the system you want, for a matter of a buck or so. Incredible, isn't it? But it is what the team of theory and practice can do theory to find a way to do it, practice to devise what to do, so it can be worthwhile, by virtue of adequate demand.

DIGITAL TECHNOLOGY

Two examples of such application from outside audio, show what can be done—pocket computers and digital watches. In Harry Olson's prime, such devices, if they could have been made at all, would have cost millions of dollars each. And then they would not have had the accuracy available today. But digital technology, using semiconductors in their simplest operation—either they conduct or they don't, as opposed to the so-called 'linear' mode—has made it possible to put the "guts" of a watch, or a computer, on a single chip.

When I was a boy, the most expensive part of a good watch, unless you had one set in diamonds or something, was the inside the works. Now, the most expensive part is the case. And digital watches have an accuracy that the old analog watches have a hard time equaling.

Old-fashioned audio was analog. It used so-called 'linear' devices to amplify complicated waveforms. Audio engineers were wedded to this idea. How else could you reproduce the larger and microscopically smaller nuances of an audio waveform, without distortion?

Makers of electronic organs were the first to see the possibilities of digital techniques. By using a master oscillator up in the megahertz, and a system of counters for each of the 12 notes in the musical octave, an organ could be more accurately tuned, right in its design, than even reasonably good tuners could achieve. Look at the cost saving right there. The tuning is on a chip, can be mass produced for a few bucks.

Organ makers went from there to digital waveform generators. Actually Moog and others were already there, from a slightly different approach, with synthesizers. But where did they get that from? Harry had the germ of the idea, long before that, but not the intricate parts to do it as well as it can be done today—or in the quantity in which it can be done today.

PROGRESS AT A SNAIL'S PACE

Why are we reluctant to progress? The earliest electric lamps used carbon arcs. They polluted as much as the gaslight they replaced. Thomas Edison came up with the incandescent filament lamp no pollution, permanent, or nearly so. Impossible, said the critics, until they saw it. Better than any form of light before it.

But still inefficient. A filament lamp converts less than 2% of its energy into light, more than 98% into heat. Now fluorescent are much more efficient: they get about 6% as light, only 94% as heat. An improvement, but would you really call that efficient? Harry's synthesizer took kilowatts to run. Today's can run on small batteries. As light generators, lasers are more efficient. We're getting there. But what takes so long? ■

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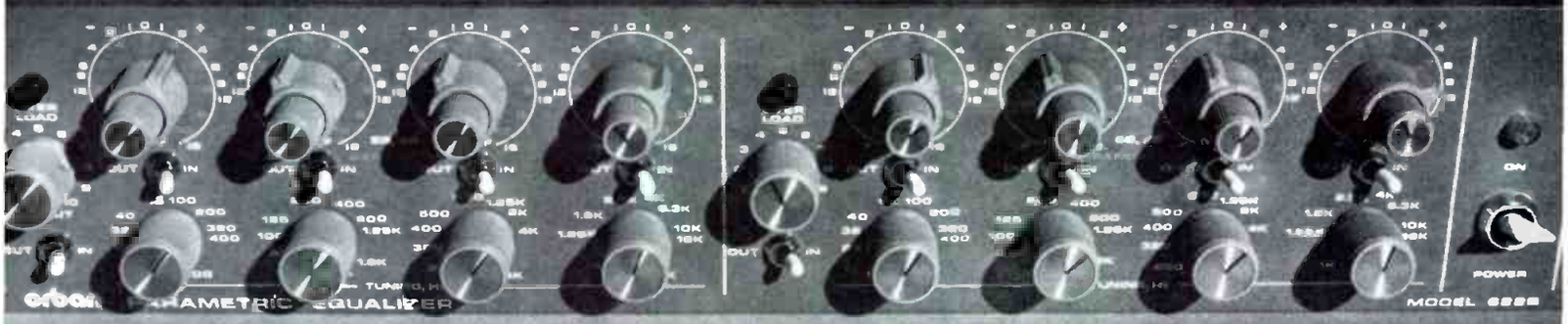

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MARTIN DICKSTEIN

db Sound With Images

The Business of AV

• Recently, I received in the mail a publication from an organization entitled Chicago Audiovisual Producers Association (CAPA). The publication, called *PROJECTION*, is a quarterly, and the issue sent was Vol. 1 Number 1, dated Oct., 1978. Included in this 6-page issue, were some interesting details—a few of which I figured might stir some thoughts in our readers.

The first page labeled CAPA as "The Organization that Speaks for Chicago's Audiovisual Industry." The opening paragraph makes an interesting point: "More audiovisual productions for business and education are created in Chicago than anywhere else in the world." Did you know that? Presumably, you might have thought (as I did) that New York was the leader.

The paper then went on to state that most of this production is done by their 40 members consisting of companies that specialize in business meetings, motion pictures, videotapes, slide and multi-media presentations, filmstrips, training programs, and any audiovisual medium which will help business people and educators better communicate to their audiences. While some of the firms work in one medium, many others work in all of them.

CAPA was formed in 1974, in an effort "to promote the highest professional and ethical standards" within the audiovisual industry. The organization limits its membership to the chief executive officers of Chicago-area production firms. All members are full-time producers whose principal source of income is custom audiovisual production. These companies have proven track records with major U.S. corporations, associations, school systems, and government agencies.

One important aspect, according to the group's president, David R. Hayes, is the education of the buyer. In view of the growth of the audiovisual industry in sophistication and complexity, Mr. Hayes says: "We attempt to inform the business world regarding audiovisual services" and "We try to show buyers how best to use their dollars, how best to match the media with the message and the audience."

CODE OF ETHICS

To this extent, CAPA has come up with a 12 point Code of Ethics of which, the following is a partial list:

CAPA members will in every instance honestly describe and represent their capabilities in every category of production. They will not falsely imply skills, facilities, or staff they do not have, nor shall they falsely cite clients served.

CAPA members will carry suitable amounts of liability insurance, so as to protect their clients and suppliers interests in the matters of property damage and personal injury.

When producing on location (schools, factories, supermarkets, offices, etc.) CAPA members' crews will conduct themselves in such a manner that they will always be welcomed back.

CAPA members will make every reasonable effort to meet deadlines which

have been agreed to between themselves and the client.

CAPA members shall respect their fellow members in order to gain respect for their industry and organization among clients. Members will seek business by emphasizing their own positive points rather than denegrating a fellow member or supplier. It is agreed that members– at all times—sell for their industry as well as for their own firms.

The other items omitted, here, deal with common-sense, business practices such as: making realistic bids instead of "bidding low" and hoping to negotiate for more later; operating their business in a fiscally sound manner and paying bills on time; agreeing that all materials are property of the client unless specifically stated in the contract, constant striving for a high degree of professionalism in all aspects of their work, and so on.

Although the organization is for producers of audiovisual presentations, certainly these concepts are valid in any phase of the audiovisual industry, whether the company sells or rents equipment, installs systems, produces the software, supplies the creative and artistic talent, furnishes the technical personnel, or organizes the whole program.

THE 35MM SLIDE

Then, publication Number 2 came in the mail and with it, some more startling facts: "In 1978 the 35mm slide the backbone of the audiovisual meetings became the largest audiovisual medium in the United States, surpassing both film and video.

"An estimated $1.29 billion was spent on slides, projection equipment and slide shows during the year; this represents an 11% growth over 1977. In comparison, $1.28 billion was spent on film in 1978, and $1.02 on video.

USE OF SLIDES IN MEETINGS

"....the majority of the 200 million slides and 400 million duplicates produced annually are used by business and industry. Most of these slides are used in meetings, marketing meetings, financial meetings, trade shows, association conventions, meetings to introduce products, to motivate employees or to build a company's image.

"Within the past year audiovisual producers based in Chicago produced hundreds of meetings ranging from $10,000 to more than $500,000.

"These meetings used every type of audiovisual medium: Broadway-type musicals, filmstrips and slides, slides to support speeches and audio tracks, or as part of sophisticated multiscreen productions.

The publication then went on at length to answer the question "Why meetings?", and to describe some of the projects produced by some of their members—finishing up with a page of advice to clients, of the members, on how to go about choosing an audiovisual production company. It started with: "Almost all audiovisual firms are small companies that produce custom-designed, handcrafted, one-of-a-kind products."

And then stated: "the audiovisual firms' clients, on the other hand, are usually large corporations that deal in a world of mass production and giant-sized operations.

"For the audiovisual firm and the corporation to create a successful meeting, they both must understand and respond to each other's requirements."

The rest of the article cautioned the client on what to look for in an audiovisual firm, which ones to ask for proposals, how to decide on one, some information about pricing and purchasing, what to look for in proposals, how to control expenditures, and so on.

I found *PROJECTION* truly a worthwhile bit of reading. The information they supplied to clients should help all audiovisual firms consider very seriously what their actions should be and how they should work with clients. The title of the last page was: "Working together: That's how the audiovisual firm and corporation create successful meetings." Makes good sense. ■

db New Products & Services

HEADPHONES

• Combining an open-air design with circumaural ear cushions, the HD-430 headphones weigh only seven ounces and offer a frequency response of 16 Hz to 20 kHz, with less than 0.5 per cent thd. Among the features incorporated in the HD-430 is the newly designed diaphragm (with a whirl-shaped, highly-flexible plastic surround) and a new magnet material, cobalt-samarium, one-third the weight of conventional materials. Separating the ear from direct contact with the inner foam pad, the circumaural ear cushions permit all the advantages of open-air listening—via rear openings. Impedance is 600 ohms per channel.
Mfr: Sennheiser Electronic Corporation
Price: $119.00
Circle 51 on Reader Service Card

BULK ERASER

• A hand-held bulk eraser, the Blank-IT, is capable of erasing magnetic tape of all formats—including audio tape of up to 1-inch, computer tape and both VHS and BETA video cassettes. Thermocouple protected, the unit shuts off when its duty cycle has been exceeded. After a short cooling period, it is operable again. The duty cycle for the unit is five minutes on, fifteen minutes off. The Blank-IT weighs 4.23 lbs., operates on 115 VAC, 50-60 Hz, and comes with a replaceable felt work surface to prevent scratching material under erasure.
Mfr: Fidelipac
Circle 52 on Reader Service Card

INTERCOM SYSTEM

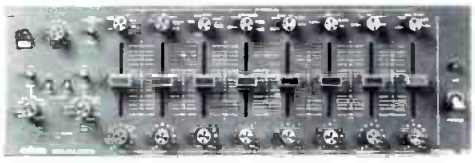
• The RS-202 4-channel remote intercom station consists of two discrete intercom channels and two listen-only program channels that are mixed in the remote station and fed to binaural phones. (Channel A intercom and program to one ear and channel B intercom and program to the other ear.) Two separate mic on/off switches along with listen volume controls allow for nine possible combinations of talk/listen functions. Channel A program is transformer isolated and balanced. Both program signals have individual volume controls and are at line level.
Mfr: Clear-Com
Circle 53 on Reader Service Card

MICROCOMPUTER SYSTEM

• Designed to provide capabilities that begin where the original TRS-80 approaches its upper limits, the TRS-80 Model II Microcomputer System provides more data storage, greater versatility and higher computing speed (the Model II is said to operate at twice the speed of the original TRS-80). The Model II can perform as a general purpose data processing machine, an intelligent terminal, or a word processor. In addition to either 32 or 64 thousand character (bytes) or internal Random Access Memory, the Model II has one built-in 8-inch floppy disc that stores an additional one-half million bytes, including the Disc Operating System. Furthermore, the Model II can be expanded to a four disc system for up to two-million bytes of storage. With a built-in 12-inch high resolution video monitor, the Model II displays 24 lines of 80 normal characters, or 40 expanded characters. The 76-key alpha-numeric keyboard (detachable and moveable) includes advanced functions such as Control, Escape, Caps, Hold, Repeat and two software-programmable Special Function keys. An enhanced Level III version of the TRS-80's Level II BASIC language and "TRSDOS" operating system are automatically loaded in memory when the machine is turned on. Built-in input output capabilities include two RS-232C channels, and one Centronics parallel port.

Mfr: Radio Shack
Price: $3,450.00 for the 32K 1-disc system
Circle 54 on Reader Service Card

EQUALIZER



• An 8-band, single channel, parametric equalizer, the model 672A offers the convenience of graphic-style EQ controls. EQ sections are reciprocal and have a ±16 dB range. Included in the unit are independent high- and low-pass filters which can be used to bandlimit—or by means of the auxiliary low-pass output, a two-way, 12 dB-per-octave continuously tunable crossover may be obtained. Tuning ranges include: 20-60 Hz; 40-150 Hz; 110-310 Hz; 230-750 Hz; 480-1900 Hz; 1.1-4.5 kHz; 2.8-9.0 kHz; and 5.9-21 kHz. Total harmonic distortion —less than 0.05 per cent, 20 to 20,000 Hz (+18 dBm).

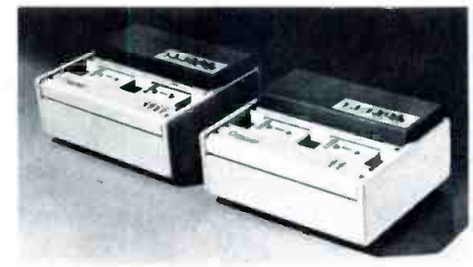
Mfr: Orban Associates, Inc.
Price: $499.00
Circle 55 on Reader Service Card

DIGITAL DELAY

• Designed for applications where only one delay is required, the DL-3 digital delay line features a 20 Hz to 15 kHz bandwidth at all delay lengths. with a dynamic range of 90 dB or better. Delays are available from 0 to 120 ms. in length. The DL-3 has one input and one output and a built-in bypass.

Mfr: DeltaLab Research, Inc.
Price: $775.00
Circle 59 on Reader Service Card

CASSETTE DUPLICATOR



• Two high speed cassette copiers, the model C-100 (mono) and C-400 (stereo), copy both sides of the cassette simultaneously at 30 inches per second—16 times real-time. (In other words, the unit duplicates a 30 minute cassette, both sides, in less than one minute.) Both units incorporate highly automated functions, designed for use by non-technical personnel. In addition each unit has an erase feature and automatic rewind of both master and copy (so that the cassette copies come off the machine ready to play). A warning light, designating "short copy," eliminates the embarassment of missed programming.

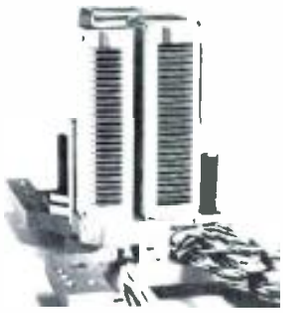
Mfr: Pentagon Industries, Inc.
Circle 60 on Reader Service Card

DISC PREVIEW SYSTEM

• An audio digital delay system, the ADD-1 provides the highest quality signal output during the final disc mastering step. The ADD-1 offers a delay of 16-bit digital or balanced-line analog input signals with a 90 dB dynamic range. Standard features allow the same setting to be used for any tape speed, and a push of a button on the led control panel to change delay times. Delay times can be preset, in addition to selectable delay times in 5 ms. increments. The system requires only two channels of audio, and eliminates the need for a special preview machine. The ADD-1 is totally compatible with normal and half-speed cutting. Featuring transformerless input/outputs, with both serial and parallel multiplexed digital input and output ports, the system can be used to preview direct-to-disc or digital recordings. The ADD-1 has an optional 100 kHz sampling rate with 2.56 sec. maximum delay to provide a wider-than-normal system frequency response.

Mfr: AMPEX Corporation
Circle 61 on Reader Service Card

REPLACEMENT HEAD



● This hot-pressed glass-bonded 24 track ferrite replacement head system is now available for the Mincom M79 recorder. The new ferrite heads, manufactured of hot pressed ferrite with glass bonded gaps, are available in all standard track formats from ¼-inch up to 2-inch tape.
Mfr: Saki Magnetics, Inc.
Circle 56 on Reader Service Card

TAPE MACHINE

● Easily convertible between 16 and 24 channels, the MTR-90 Mastercorder is a two-inch machine incorporating a fully symmetrical tape path and a pinch-roller-free direct-drive capstan. The capstan is controlled by a phase-locked closed-loop servo system, capable of providing full SMPTE synchronization with film chains and videotape recorders. Tape speeds are 30 and 15 in/sec. with a continuously variable ±20 per cent speed control and a digital readout indicating the percentage of over or under speed. Other features include: a constant tape tension system with full-time d.c. servo control of both the supply and take-up motors, noiseless and gapless punching-in and out with automatic monitor switching, selective reproduce on all channels, automatic motor driven head covers, and a 40 ohm floating output. Options include a remote control unit offering selective reproduce on all channels with led status indicators and a search-to-cue unit with a ten memory capacity.
Mfr: Otari Corporation
Circle 57 on Reader Service Card

EQUALIZER

● A dual channel equalizer, the model 1020 features ±12 dB boost/cut on each of 10 octave bands, with center frequencies at 31.5, 63, 125, 250, 500 Hz, 1, 2, 4, 8, 16 kHz. The unit offers balanced inputs and outputs, and hum-free, low noise artificial inductors. The 1020 weighs 8 lbs., and is standard 19" rack size.
Mfr: Neptune Electronics
Circle 58 on Reader Service Card

db Editorial

The Three R's—Radio, Recording, and Recession

A RECENT ARTICLE in the New York Times notes that *"the $4-billion-a-year record business, after 25 years of nearly uninterrupted growth, has suddenly run into a period of faltering sales, staff cuts and general demoralization."*

In this gas-less, energy-less, leader-less period that we seem to be enduring, perhaps this bit of news shouldn't be much of a surprise—except that, historically, the record business has been (and, *we think*, actually still is) more-or-less recession-proof. When people can't afford the other luxuries of life, the phonograph record remains a very good buy. And now that people are cutting down on their travelling, the record business *should* be experiencing an upturn, or at least maintaining a position of stability. With people staying at home more, they'll be playing (i.e., buying) more records, and listening to more radio.

And speaking of radios, September is broadcasting month here at **db**. Therefore, it probably won't surprise anyone that we've found a way to tie in the three R's of our title. After all, for most of us, our economy is directly related to radio and/or records.

Lately, we've been bombarding you with ominous warnings of the Federal Communication Commission's inquiry into changing the channel allocations for FM stations. As we've pointed out, the public notice of this fact was nearly buried, within an FCC Notice of Inquiry on the matter of quadraphonic broadcasting.

Ironically, the matter has little or nothing to do with quad, since a reduction in channel allocation will take more of an effect on that $4-billion mono/stereo business than it will on the comparatively insignificant quad scene.

At the risk of repeating ourselves, September seems to be a good time to remind our readers in the recording industry of what will happen if the broadcasting industry is forced into a re-allocation of channel spacing. We can expect an overall deterioration in broadcast quality, with—at best—marginal stereo performance. Also, be prepared to scrap your present receiver/tuner, since most of these (and top-of-the-line models in particular) will not function properly any more.

What has this got to do with the business of making records? Why, nothing at all, if "air-play" means nothing to you. But, is there anyone left in the recording studio who *still* doesn't understand the power of radio in selling records? If you've been reading **db** regularly, you know that recording technology is progressing at an incredible pace (Did someone say, "a little too incredible"? We'll talk about that some other time.)

Well now, wouldn't it be funny if we were all suddenly set back a quarter-century, compliments of the FCC? No, it wouldn't. But that's just what will happen if the NTIA (National Telecommunications and Information Administration) gets its way. The NTIA is a division of the U.S. Department of Commerce, and it has decided that the present 200 kHz FM channel allocation is excessive. In a splendid example of "double-speak," the NTIA says, *"We will comment only briefly on the issue of reduced channel spacings, while we examine in greater detail the question of whether increased distances between stations or reduced coverage due to interference will be required by quadraphonic transmissions, thus reducing the opportunity for increased diversity."* (NTIA Comments to the FCC, dated April 17, 1979.)

Perhaps the NTIA has devised its own dictionary, with a special definition of diversity. We always thought the word was synonymous with variety, didn't you? (Later on in the Comments, the NTIA contradicts itself on this matter. But then, they do that several times.)

The real clincher to their 70+ pages of comments is this little bit of cheerful news: *"We recognize... the real or potential disadvantages of (a) incurring the costs of changing station transmitter frequencies, (b) producing some increased interference on existing receivers, (c) reducing or even eliminating entirely certain applications of... SCA, (d) restricting or precluding the adoption of some systems of FM quadraphonic broadcasting, and (e) increasing the cost of FM receivers."*

Outside of these trifling considerations, no doubt their brilliant scheme works like a charm. Of course, it will go a long way towards demolishing our $4-billion industry, and should greatly diminish the value of most recording studio services. After all, why bother continuing the development of innovative recording techniques for ancient instruments (that is, for the new generation of receivers that will be required)?

Now that one branch of government is found meddling in the affairs of another, it seems to be time for all of us to start harassing our Congressmen and Senators.

Put it this way: Either *you* get Congress to put an end to this stupidity, or the stupidity will put an end to *you*. It's your choice.

Start writing. And don't forget to let us here at **db** know how you feel too. Write carefully—the job you save may be your own. ■

LEN FELDMAN

The Rebirth of Quality F.M. Radio

Rebuilt from top to bottom, WNCN stands as the beacon for quality f.m. sound. Will other stations "see" the light?

IF YOU ARE INVOLVED in f.m. radio —either "just listening" or actively working at it— you may share my disappointment in the quality of the audio signal you recover. What Major Edwin Armstrong (the inventor of f.m. radio) envisioned as a high-fidelity radio communications system has become, in most instances, a means for broadcasting music with a minimum of dynamic range, often highly-distorted and not much better than the low-fi sounds you hear from a hand-held a.m. radio.

While tuners and receivers keep getting better and better, it seems that broadcasters providing the programs are getting sloppier and sloppier. Of course, this does not have to be the case. As proof of that, I was recently invited to tour the facilities of radio station WNCN, in New York, a station whose 25-year history is extremely interesting.

A LITTLE HISTORY

WNCN was, and is once more, a classical music station. During its first twenty-plus years of existence, the station never made any money—not unusual for a classical music station. When quadraphonic sound seemed about ready to capture the fancy of the listening public, the management sold the station to Star Broadcasting Company. The new owners changed the format from classical stereo to rock-and-roll quad (using one of the matrix quad systems then being promoted) and, WNCN became WQIV.

The public outcry was unprecedented. People who had listened to WNCN during its classical (if impoverished) days suddenly realized that they had lost one of the few remaining classical music outlets in the New York area. Many thousands of signatures were applied to countless petitions, urging the station to restore the classical format. Whether the petitions prompted GAF Corporation's president, Dr. Jesse Werner, to purchase the station from Star Broadcasting Company (on June 6, 1976), or whether the WQIV format proved to be an economic loser, is academic. What is important is that GAF restored the station to its original format, and also restored the plant and studio, which were in a shambles. Studio equipment was antiquated. Microphones hung by threads. Turntables had more rumble than inexpensive home record players, and cartridges were eating up the chief asset of the station—more than 10,000 LP records.

REBUILDING THE STATION

GAF decided to rebuild the station from scratch, and make it the best-sounding f.m. station (regardless of format) on the air. They hired consultant Richard Sequerra to tell them what had to be done. Sequerra's name is probably familiar to db's audio buff readers. (That means everybody, doesn't it?—Ed.) He designed the legendary Marantz model 10B tuner, now regarded as a classic in the world of hi-fi. (Today, a working 10B tuner commands a higher price than its original price tag—*if* you can find one.) Later, Mr. Sequerra formed his own company and designed the Sequerra model 1 FM tuner, which now sells for more than $3000. It's still in limited production, although Sequerra has since left the firm to devote his talents to loudspeaker design.

New and spacious studio facilities were rented at 1180 Avenue of The Americas, in New York, and Sequerra developed a long-range overall plan to rebuild the station from the ground up. The initial phases of the job took seven months, and still there are many long-range plans that have not been completed. But WNCN now broadcasts an exemplary signal that other stations can aspire to.

I was recently invited to visit the facility and talked with Sequerra, and with Mr. Robert Richer, the station manager, to learn more about how an f.m. station can deliver the kind of reception that Armstrong, and other f.m. pioneers, envisioned.

The WNCN offices were as carefully planned as the electronic facilities of the studio. Subdued-but-adequate lighting comes from eye-level positions, rather than from the ceilings (a practical and effective attempt at energy conservation). Modular working areas, shown in FIGURE 1, seemed conducive to operating the business end of the station effectively without being claustrophobic in design.

RECORD LIBRARY & CARD CATALOG

Before getting around to technical matters, I was shown the station's vast record library, a portion of which may be seen in FIGURE 2. Since GAF acquired WNCN, more than 5,000 additional discs have been acquired from all over the world.

Len Feldman is the technical director of the Institute of High Fidelity. Mr. Feldman was chief engineer of Crosby Electronics, and worked with the late Murray Crosby on early stereo f.m. systems.

Figure 1. Modular working areas of the WNCN offices.

Figure 2. One section of WNCN's vast record library.

Catalogued in the order in which they are received, each record is also indexed in a card catalog that provides many facts, such as the record's playing time, composer, conductor, soloists (if any), instrument (in the case of solo-instrument recordings), and much more. The catalog files are shown in FIGURE 3.

Using this data, supplemented by his prodigious knowledge of classical music, music director David Dubal plans programming, some three months ahead of broadcast date. The unusually-long lead-time is necessary because WNCN now publishes a magazine called *Keynote*. The publication provides articles dealing with the performing arts, and includes an hour-by-hour program listing, as well as a comprehensive composer's index.

Mr. Dubal uses a huge planning board (shown in FIGURE 4) into which he slips the catalog cards he has selected. The cards are then sent to a computer facility where the "composer's index" is automatically compiled for the month. Once the month's programming has been compiled, there is little deviation permitted and announcers have virtually no leeway. In the future, the station hopes to input all its library data into the computer, which will free Mr. Dubal from having to remember such details as when a particular piece of music was previously aired.

THE ENGINEERING DEPARTMENT

Before visiting the control room and studios, I was shown the engineering and maintenance department. Having visited other f.m. stations in the past (most of which would shock the casual visitor because of their poor-quality monitoring and test equipment), I was delighted to find the very latest test equipment, some of which is shown in FIGURES 5 and 6. Included were precision audio oscillators, a spectrum analyzer, a vector scope, precision frequency counters, a variety of metering instruments, wow-and-flutter meters, and much more.

THE CONTROL ROOM

The first thing that caught my attention when we entered the studio/control room complex (the master control board can be seen in FIGURE 7) was a vaguely familiar-looking device that resembled a pair of turntables. Mr. Sequerra explained that it was the Keith Monks professional automatic record-cleaning system, and it was then that I learned that every single record played over the air is professionally cleaned just prior to being aired—*every* one, with no exceptions! Talk about getting noise-free reception, *that's* the way to get the signal launched with no strikes against it from the very beginning. The cleaning machine is shown in FIGURE 8. (Or, for a still closer look, see **Keith Monks Record Cleaning Machine**, in the October, 1978 **db**—Ed.)

Gazing beyond the control room's glass window, we recognized a familiar monitoring tuner nestled in between some conventional FCC-approved monitoring equipment. What we saw was—you guessed it—a specially-modified Sequerra tuner. WNCN is one of the few (if not the only) f.m. stations that

Figure 3. All records are indexed in a card catalog. The cards, within the catalog, contain various pieces of information about each record.

Figure 4. Programming planning board with cards from the catalog in place.

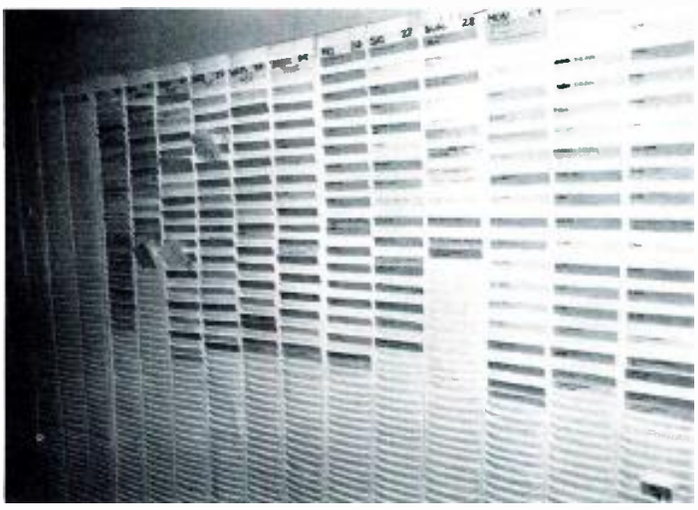


Figure 5. The testing facilities of WNCN's engineering and maintenance department. Chief Engineer Richard Koziol is at work.

monitors its broadcasts directly "off-the-air." Because the station's studios are so near the Empire State Building (where their signal is diplexed, along with countless others, onto the complex antenna system there), the receiving antenna for the Sequerra off-the-air monitor is actually directed toward the RCA Building (facing away from the transmitter) and a "bounced" signal is received.

Among the cluster of equipment shown in FIGURE 9 are dbx and Dolby noise reduction systems. However, neither system is broadcast over the air. Unlike other classical music stations (in New York, WQXR — Ed.) WNCN management has not resorted to Dolby f.m. broadcasting, maintaining that if proper broadcasting practices are followed, Dolby is really not necessary nor is it particularly beneficial. (For more on the subject, see our **Dolby F.M. Update** in the September 1978 **db** — Ed.)

Actually, when WNCN produces one of its many live, remote broadcasts, the program is dbx-encoded at the source and decoded prior to over-the-air playback. According to Sequerra, this helps reduce what little noise is generated in the telephone lines between the remote location and the studio. (*Little* noise? — Ed.) And speaking of those phone lines, the twin pairs of lines that WNCN leases from American Telephone & Telegraph are, according to the station's technical staff, of extremely high quality and are definitely not a limiting factor in the consistent 74 dB signal-to-noise ratio that the station is actually able to maintain. As for modulation practices, WNCN is able to provide a consistent dynamic range of 47 dB or more, while providing between 9 and 17 dB of headroom at all times. That's what can be accomplished when absolutely *no* compression or peak-limiting is introduced into the signal path. Compare those figures with typical f.m. operations around the country and you will realize just how impressive they are. Another network f.m. station (specializing in "top 40" rock and noisey D.J's) was recently checked out by WNCN's engineering department. The station was found to have an absolute dynamic range of no more than 6 dB! How's that for "loud Muzak"?

LIVE PROGRAMMING

I asked about future plans for live music programming (aside from the New York City Opera and Chamber Music Society concerts presently provided via remote pickups). To better answer the question, I was promptly escorted into a giant two-story shell of a room that may someday be completed as a performing and recording studio, in which small musical groups might perform live. The space has already been stripped

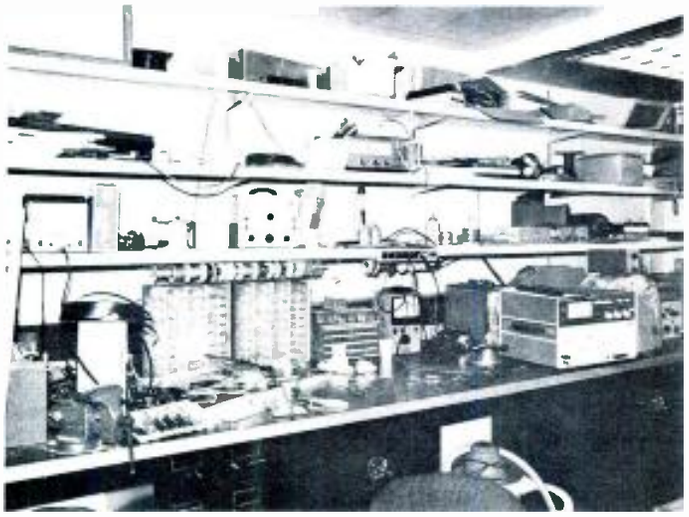


Figure 6. Another view of the test bench.

Figure 7. The WNCN control room.

down, to expose the beams and steel columns, so that future construction noise will not interfere with present live broadcast operations in the smaller studio (FIGURE 10) and in the announcer's control room.

Mr. Sequerra—as always, a "purist"—maintained that this tremendous studio will accommodate *only* a grand piano (well, *maybe* a small trio or quartet—perhaps even a quintet). The piano—positioned one-third of the distance away from any wall surfaces will enjoy perfect reproduction of its lowest keys, with no standing-wave or room-resonance problems.

When completed, the new studio will have oak flooring and be equipped with special rotatable polycylindrical-diffuser acoustic modules (designed by Sequerra, and ready to be fabricated). These will permit varying the reverberation time of the room from 0.5 seconds to 3 seconds at will. Calculations have already ascertained that the ideal audience in the room during a live performance will be no more than 12 persons. The new studio may double as a state-of-the-art recording studio, and even direct-to-disc recording is envisioned for it some day.

Meanwhile, such live broadcasting as is initiated at WNCN (other than the remote location pickups) comes from the completed, but much smaller, studio. To date, this studio has been used primarily for interview programs, although Mr. Richer informed me (out-of-earshot of the demanding Sequerra) that they had experimentally recorded a trio performance in the room and, according to chief engineer Richard Koziol (himself a highly-skilled recording engineer), the results were quite good. According to Mr. Sequerra, the special acoustic panels that he specified for this smaller studio would be helpful in many listening rooms as well.

A rundown of some of the hardware used in the station operation would gladden the heart of any audio engineer: Technic's SP-10 Mark II direct-drive turntables are mounted on McCurdy mounts for rumble-free disc playing. Micro-Seiki model MA-505 tone-arm housed carefully-aligned AKG P8 cartridges, among others, feeding Mark Levinson phono preamplifiers. Studio monitoring was via Bryston model 4B power amplifiers driving a pair of Pyramid loudspeaker systems. We also found a Nakamichi model 1000 cassette deck, plus all kinds of high-quality open-reel decks, including vintage Ampex 351s and newer Studer A-80s. Furthermore, these playback machines are continually upgraded and maintained. Rubber puck wheels are replaced when they dry out, heads are demagnetized, and raw tape is bulk-erased (using a video-tape bulk eraser) that makes it around 9 dB quieter than when first unpacked from the supplier.

Obviously, GAF is not pinching pennies in their desire to provide the "cleanest FM signal on the air." Spectrum analysis of the frequency baseband of WNCN's signal has revealed that

Figure 8. Each record is cleaned, prior to being aired, on the Keith Monks Record Cleaning Machine.

Figure 9. The control room equipment rack. Note the dbx and Dolby noise reduction systems, as well as the Sequerra tuner.

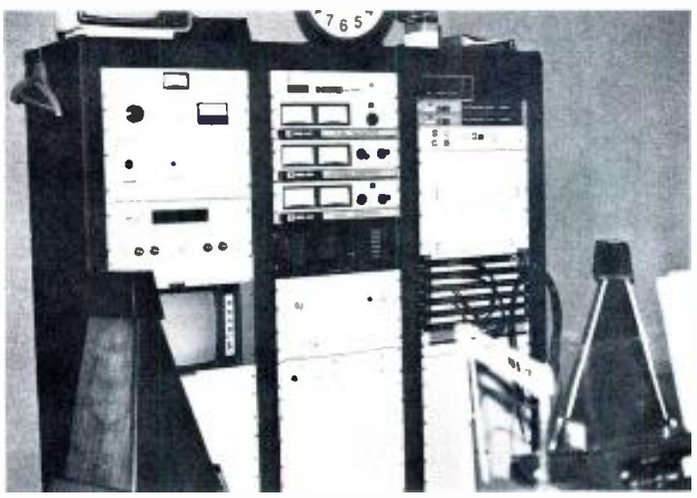


Figure 10. WNCN's production studio.

this signal is a "textbook example" of what the FCC specifies as an ideal f.m. transmission characteristic.

According to Sequerra, who has made similar measurements at radio stations around the country, the only other stations that can meet FCC proof-of-performance requirements in every detail are WGBH in Boston and WFMT in Chicago, both of which have excellent reputations in their listening areas. Sequerra feels that current broadcast specifications are unrealistic and outmoded, and that to maintain proper modulation, stations should be peaking at deviations fully 3.5 dB lower than most of them now employ (even though most use extreme signal compression and limiting).

No matter how good the signal that originates at the studio, the f.m. transmitter must be equal to the task of getting that signal "on-the-air." Sequerra also worked extensively on the two 10-kW transmitters WNCN has installed on the Empire State Building.

We also talked about the problems of f.m. reception in cars, and agreed that the only way to solve such problems is to use a "diversity" type of receiver. Such a receiver would have two separate front ends and antennas,one polarized vertically, the other horizontally. A high-speed logic and switching system would select the best, most multipath-free of the two signals available at any instant. The difficulty with such a system is not technical but economic, since it would be a very costly receiver to install in a car. Diversity receivers are often used in military communications, where loss, or garbling, of signals cannot be tolerated, even for a short time.

Before I left the station, I was shown some of the antiquated test and monitoring equipment that had been replaced since the station's rebirth in 1976. Unfortunately, some of this equipment is still being used in many f.m. stations around the country.

As in any audio signal path, a broadcast station's signal quality is only as good as the weakest link in the signal chain. WNCN has succeeded in upgrading its broadcast quality because enough people cared and were dedicated to excellence. From worrying about such details as the proper vertical-tracking angle of the stylus in the record groove, all the way to tweaking and linearizing the modulation capability of the two transmitters, the WNCN management continues to prove just how good f.m. can really be. If thousands of people could bring about the metamorphisis of one station in New York City, perhaps you, in your area, can demand better broadcast quality from your favorite f.m. station by pointing out what can be accomplished by the radio station that cares. You might also point out that as of this writing WNCN's operations are actually showing a *profit* for the first time since the station went on the air, over 25 years ago. And who knows, next time you turn on the radio, that super-sounding record you hear might be one of *yours*! ■

VLADIMIR NIKANOROV

A New Cartridge Tape Recorder

Various design and performance characteristics of the Series 99 Cartridge Tape Recorder provide a fresh, new approach to overcoming the age-old problem of phase conformity between channels.

EVER SINCE AUDIO was broadcast in stereo, phase conformity between channels has been regarded as one of the most important parameters of quality reproduction, mainly because the majority of listeners, according to statistics, stubbornly use mono table radios. Yet, the NAB cartridge (one of the most popular media of recording-reproduction) and cartridge tape recorders in general, are notoriously deficient in this area.

Quite recently, one of the major manufacturers of cartridge tape recorders, International Tapetronics Corporation, made an attempt to overcome this major problem with a new cartridge tape recorder, the ITC Series 99. According to ITC engineers, its design parameters were set forth after a detailed analysis of the problems associated with the NAB cartridge.

Vladimir Nikanorov is technical director at Bonneville Broadcast Consultants, Tenafly, New Jersey.

When Jack Jenkins and Tom Baily introduced the cartridge tape recorder in 1958, there was only one radio station in the United States experimenting with stereo transmission. Four years later, the three-track stereo cart recorder was born, out of necessity.

THE PROBLEMS OF PHASE SHIFT

There are three profound effects of poor phase performance. When the shift is introduced, the original stereo location becomes distorted. Also, there will be a loss of level due to azimuth error, and if the channels are electrically combined, cancellation results.

To ensure that all three objectionable results of phase shift are kept within reason, strict tolerances must be observed. For instance, to assure no noticeable distortion of the stereo image, the maximum time delay between channels must be less than 0.25 milli-seconds. At 7.5 in sec., an azimuth angle of only 10 minutes, 27.2 seconds will create a 1 dB drop at 15 kHz. (60 minutes = 1 degree.) To prevent audible cancellation of channels electrically combined to mono, the time delay should be no more than 6.25 milli-seconds.

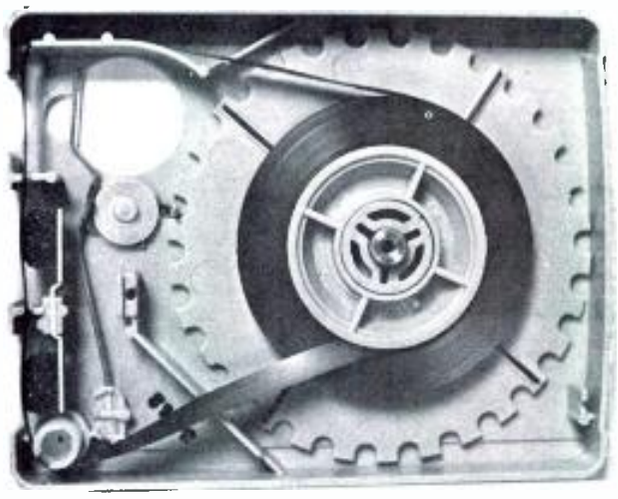


Figure 1. NAB Cartridge

Phase Shift	90	50 Hz–12 kHz
Frequency Response	+3.5/-2.5 dB +1.5 dB	40 Hz–15 kHz 230 Hz–10 kHz
Signal-to-Noise	-40 dB	Unweighted
Level difference between Channels	3 dB 2 dB	40 Hz–15 kHz 125 Hz–10 kHz
Crosstalk	-40 dB -6 dB/Octave +6 dB/Octave	300 Hz–7.5 kHz below 300 Hz above 7.5 kHz
Wow & Flutter	0.15%	Weighted
THD	4%	1 kHz, +8 dBV 80 Hz, +8 dBV

Table I IBA Code of Practice.

These are basic requirements that any cartridge or open-reel recorder should satisfy, while conforming to conventional technical data defined by NAB standards. Yet, there are fundamental reasons why the cartridge system exhibits marginal performance, in comparison with open-reel formats.

It is well-known that the mechanical properties of an endless-loop cartridge loaded with lubricated tape contribute a great deal to the instability of its performance, in general, and to phase instability in particular. The tape is pulled from the pack by the capstan/pinch roller, and is twisted as it is pulled from the pack. Several factors which actually define tape-to-head contact position are unpredictable. One is the relative elasticity of the twisted tape; another is the friction of the lubrication used in the backcoating of the pack; the third is the guidance system. Unlike open-reel recorders, the guides within an NAB cartridge are part of the shell. Usually, they are made of plastic, like the rest of the assembly, with wide tolerances. Naturally, these design deficiencies have influenced the performance of the cart recorder itself, which initially was intended to conform to reel-to-reel specifications. As a result, practical standards have been written to match the poor performance of the total cartridge system. For instance, the IBA (Independent Broadcast Authority) in the U.K. has introduced a Code of Practice based on the practical performance of cart machines, the great majority of which are manufactured in the United States. (TABLE I)

It would be a mistake to say that the cartridge alone is responsible for such non-professional performance. In fact, tolerances of the cartridge are complimented by typical cart recorder errors, and until recently, most system improvements were the result of advances in cartridge design. There were changes involving cartridge configuration and tape pack, adjustable corner guides and other features. For instance, 3M came up with a cartridge which requires an entirely different configuration of the tape recorder. However, despite these efforts, problems of instability, especially in the area of phase conformity, remain unresolved.

The idea of a tape recorder which would be able to compensate for deficiencies in the NAB cartridge is not particularly new. There have been a number of publications on the subject. However, an overall solution to the problem would require a major change, either in the construction of the cartridge, or, in the design of the recorder.

For ITC, one of the major cart recorder manufacturers, it was only natural to seek the solution in the area of cart recorder design. Their effort is represented in the Series 99 cartridge recorder.

Conventionality of the new cart recorder starts and ends with the fact that it is designed to record and play NAB cartridges. According to the published specs, the Series 99 conforms to professional open-reel standards for 7.5 in/sec.

Evaluation of the new recorder design took several directions. First was evaluation of the principles of operation. Next came measurements of conventional parameters which actually determine the specifications, with emphasis on the area of phase performance. Finally, the tape recorder was evaluated from a standpoint of long-term use in "real" broadcast situations.

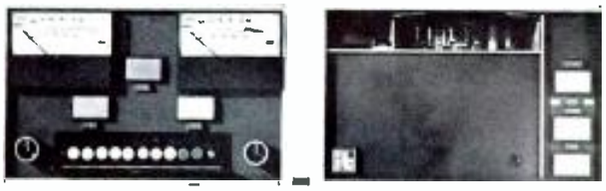
The Series 99 tape recorder looks quite different in comparison with others. FIGURE 2 shows its two modules. One contains most of the electronics, while the other houses the tape transport system, and a bulk eraser.

Procedurally, operation of the recorder is divided into the following phases.

1. Insert cartridge
2. Bulk-erase cartridge
3. Record 15.5 kHz tone
4. Record-head azimuth adjustment
5. Bulk-erase cartridge
6. Cartridge wind to the splice

One of the mentioned operations, the cartridge insertion, is actually manual. The others are automatic and are initiated by depressing a push button. The basic logic of the described series of operations is to physically adjust the record-head azimuth for whatever discrepancy the cartridge introduces. (Playback is adjusted manually with a test tape and serves as a reference.) The record head is driven by a servo motor which

Figure 2. The Series 99 System



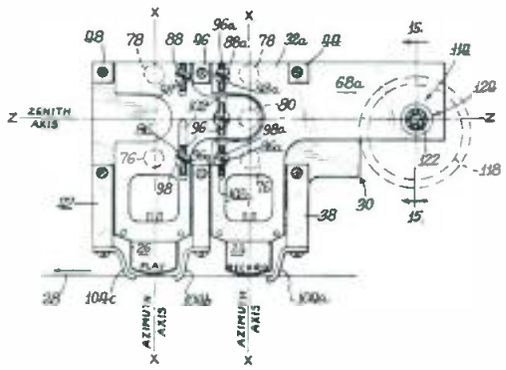


Figure 3. Patent sketch of the Series 99

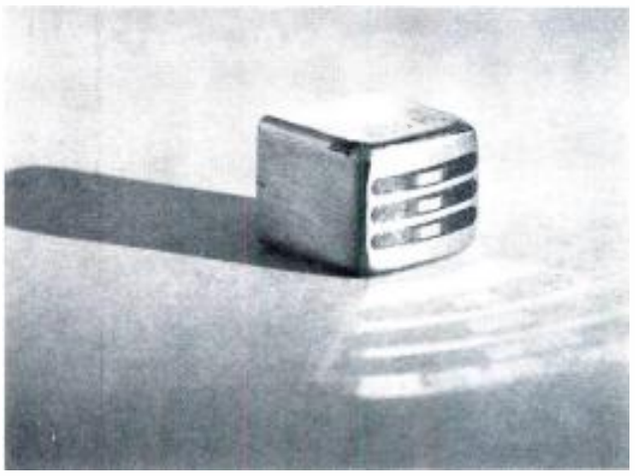


Figure 5. Series 99 playback head.

is controlled, in turn, by a microprocessor. FIGURE 3 shows the design of the head module which makes such an adjustment possible. It is designed as a center-pivot mechanism, with rotational axes on the exact vertical and horizontal line of the heads. Two mechanical planes—height and zenith—are independently adjustable and locking. Azimuth adjustment does not affect the height and zenith of the head. A built-in oscillator produces the 15.5 kHz tone which is recorded. A microprocessor detects the difference between recorded and reproduced 15.5 kHz signals. The difference occurs only if the gaps of the playback and record heads are not parallel. In this case, a voltage is produced and the motor is driven, adjusting the azimuth of the record head. The playback head is in close proximity to the record head so there will be no noticeable mechanical displacement as the recorded material is moved towards the playback head.

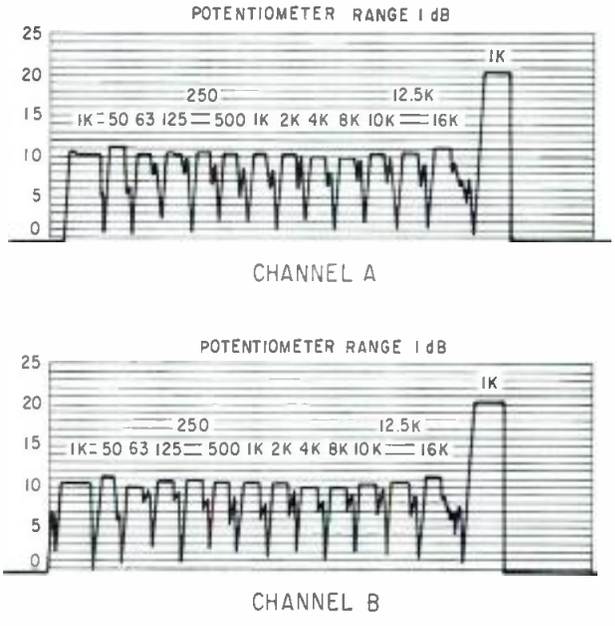
In order to get real benefit from such an advantage, all the other parameters somehow have to be better than those in the Code of Practice (TABLE I).

SIGNAL-TO-NOISE AND FREQUENCY RESPONSE

The first consideration is signal-to-noise and frequency response. In cart machines, these two parameters are interrelated. There are two factors which contribute to poor signal-to-noise and inferior frequency response in conventional cart machines. One is compact design—the playback head is in close proximity to two sources of magnetic fields: the capstan motor and latch solenoid. The Series 99 uses a d.c. brushless servo motor with a crystal referenced frequency, phase-locked servo control system. (FIGURE 4) The shaft of the motor is ceramic, which prevents the magnetic field from coupling through it. The field of the latch solenoid depends on the current flow through its coil. This current is at maximum only at the start. Then, the current is reduced as the solenoid plunger is locked. The combined forces of the locking mechanism and the reduced magnetic field provide an adequate and permanent hold. The other factor is a typical boost +3.5 dB at 128 Hz. The boost is an undesireable function of head design, which features quite a limited length of the laminated core, and short windows. The design is actually dictated by the mechanical construction of the NAB cartridge and requirements for low susceptibility of the playback head to hum. To compliment such a head, the typical cartridge playback amplifier has a low frequency roll-off starting at 80 Hz.

Figure 4. The d.c. brushless servo motor employed in the Series 99.

Figure 6. Playback frequency response of the Series 99.



	Channel A	Channel B
Signal-to-Noise Ratio	-54 dB	-53.5 dB
THD (Average %)	1.5	1.5
IMD (Average %)	2.9	2.9

Note: The average figures for thd and imd were calculated from measurements of 10 different tapes. (For the tape and cartridge manufacturers, see Table III.)

Table II Signal-to-noise ratio, thd, & imd of the Series 99.

FIGURE 5 shows a Series 99 playback head. It is an open-faced head. The shape of the face is cylindrical rather than hyperbolic. The windows through which the core protrudes are considerably longer than conventional windows, and the lamination is constructed of mu metal, with aluminum oxide particles for better durability.

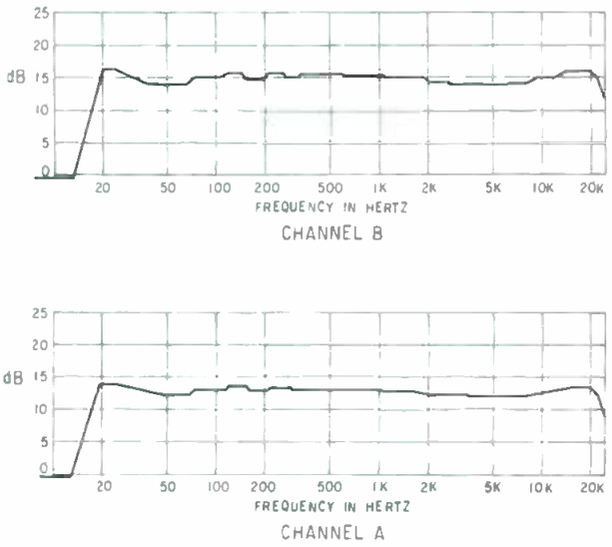
Playback frequency response (FIGURE 6) was measured with an STL test tape, recorded in accordance with 1975 NAB standards. It is flat within ±1 dB from 50 Hz to 15 kHz. Signal-to-noise ratio, referenced to +4 dBV output level at 600 ohms, is -54 dB. This remarkable improvement is made possible mainly through the reduction of the magnetic field in which the playback head is required to work, as mentioned above.

The Series 99 record amplifier is loaded by a low inductance record head and precisely adheres to the NAB curve. Frequency response of the record/playback is shown in FIGURE 7. Signal-to-noise ratio, thd, & imd are displayed in Table II. As can be seen, playback/record frequency response, thd, imd, & signal-to-noise are improved.

PHASE PERFORMANCE

In terms of phase response, it is known that the capstan-pinch roller combination is the most-critical mechanism in a tape machine. Changes in friction between capstan and tape and between tape and pinch roller will affect the stability of tape movement against the heads. This is particularly true with the slippery back-coated tape used in the NAB cartridge.

Figure 7. Frequency response of the record/playback of the Series 99.



J.S. Patent

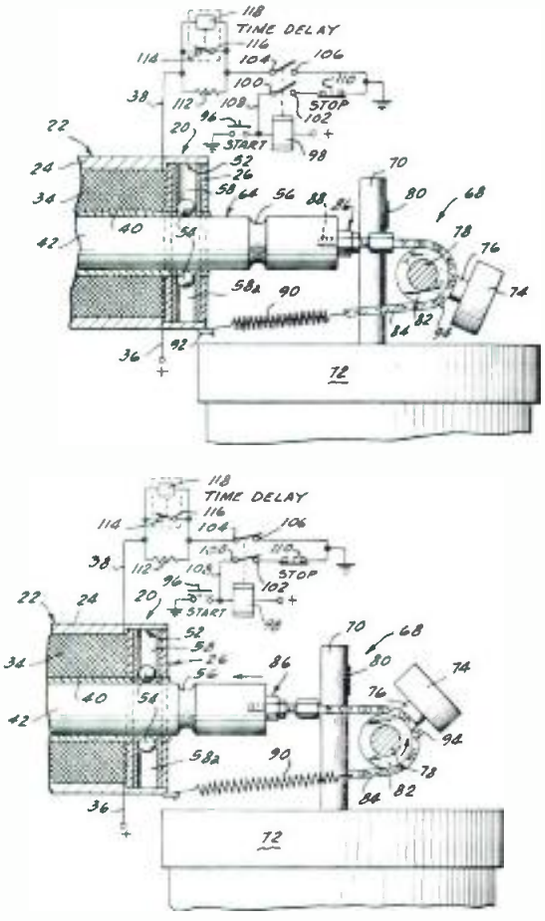


Figure 8. Latch solenoid patent sketches. Depicted are two stages of its movement.

Two factors have to be absolutely intact in order to maintain constant friction. One is the latch solenoid. Size-versus-pulling power of existing solenoids is such that the necessary current through the coil inevitably raises its temperature. The heated coil changes resistance, thus reducing the current and pulling power of the solenoid, which gradually changes the capstan-pinch roller pressure. FIGURE 8 shows the latching solenoid at two stages of its movement. The pinch roller is pulled by the chain-and-sprocket assembly and is pressed against the capstan. At that time, the two steel balls enter the matching groove and lock the plunger in position. Voltage on the coil drops from 50 vdc to 18 vdc, reducing the current and the magnetic field. When released, the field collapses fast, stops the tape quickly and produces less transient interference in the amplifiers.

Another item which contributes to phase performance is the pinch roller itself. Slight changes in size, due to changes in temperature, humidity or wear, result in varying phase performance. Popular polyurethane rollers are hygroscopic—that is, their dimensions are affected by change in humidity. The Series 99 pinch roller uses a new "525-K" rubber compound in its pinch roller. It is not affected by temperature and humidity, and claims to be immune to most cleaning agents. Less tape skewing, and subsequently better phase performance, is the net result of the new compound.

The locking solenoid and 525-K pinch roller are coupled by a ceramic motor spindle. The spindle has been used on previous ITC models to reduce thermal dissipation which is, in fact, not terribly important in the Series 99. (Although the porous nature of the ceramic surface is good for driving tapes, it does attract tape particles, and may present problems, if not cleaned regularly.)

Another feature which plays a significant role in phase performance is the cartridge-positioning mechanism. Unlike commonly-used mechanisms, the Series 99 applies downward pressure on the outer load-bearing rails of the cartridge, and also against the left-hand side, to seat it properly against the right hand cartridge guide.

CARTRIDGE PERFORMANCE TESTS

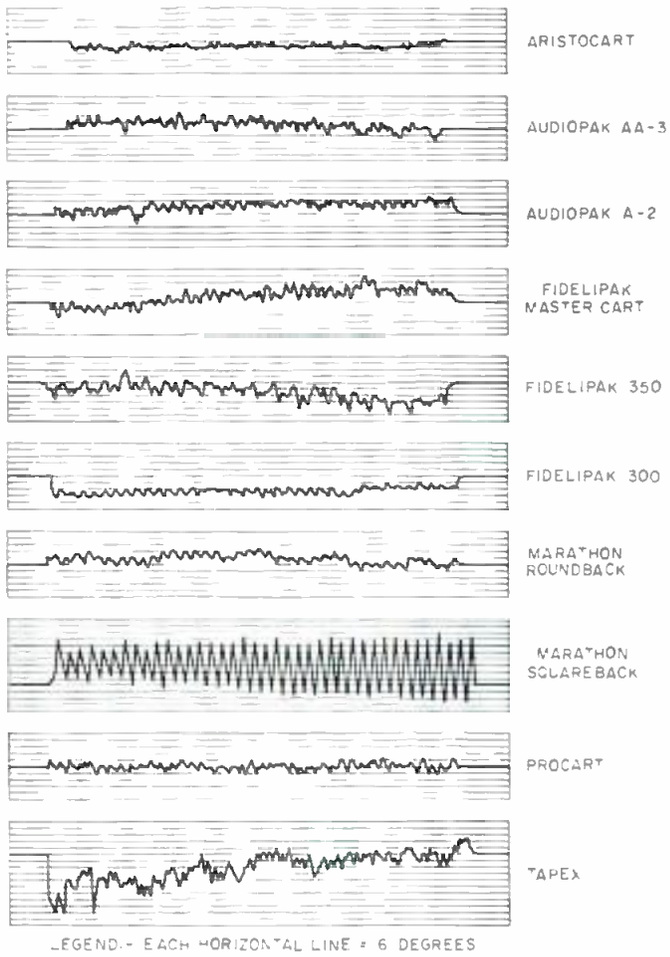
To get an idea of actual phase performance, a number of the most commonly-used cartridges were tested in the Series 99.

A Wavetek model 740 Phase Meter, connected to a Brüel & Kjaer chart recorder, was used for the evaluation. The Series 99 internal oscillator was used as a tone source. The oscillator is capable of producing all the standard frequencies from 50 Hz through 15.5 kHz. 10 kHz was chosen as a test frequency from a practical standpoint. This frequency is recommended for bias and record/reproduce EQ adjustments. Also, it makes more sense to use 10 kHz than 15.5 kHz, since the f.m. broadcast bandwidth is 16 kHz. Test results are given in FIGURE 9 and TABLE III.

The first and the most significant result that can be seen from the data is that—compared with The Code of Practice—the phase error between channels is greatly reduced. In fact, it is comparable or better than what is commonly seen on open-reel 7.5 in/sec. machines (The Code of Practice figure for open-reel machines is 60 degrees).

Some of the carts show better performance than others. This tells us several things: One, there are differences in cart performance for which the cart recorder cannot compensate. According to the charts, the nature of the differences is fast phase fluctuation. Judging by the Aristocart chart, the machine

Figure 9. Cartridge phase performance results of commonly-used cartridges tested in the Series 99.



Cartridge Manufacturer	Maximum Phase error (degrees)	Maximum Wow & Flutter
Aristocart	-7	0.11
Audiopak AA3	+12	0.12
Audiopak A2	+13	0.13
Fidelipak Master Cart	+17	0.15
Fidelipak 350	-25	0.14
Fidelipak 300	-17	0.14
Marathon (Round Back)	+12	0.14
Marathon (Square Back)	-37	0.14
Procart	-47	0.14
Tapex	-45	0.16

Table III Cartridge Phase Performance.

is capable of recording and reproducing the NAB cartridge with ±3 degrees phase fluctuation. One of the worst cases is found in the Marathon "Square Back." This cartridge is designed to avoid any back tension related to the movement of the cartridge turntable. It is held back only by the friction of the lubricated tape in the pack – a factor which is highly unpredictable. It is further complicated by an unusually long pressure pad, which is made of foam. Despite the fact that the Marathon "Square Back" is not known in the industry for its particularly inferior design, it showed a number of unpredictable factors which could result in unacceptable phase performance.

The phase fluctuation shown on the charts could be compared with the wow and flutter figures in TABLE III. The best phase fluctuation and wow and flutter figures are demonstrated by Aristocrat. The worst ones are by Fidelipak Master Cart and Tapex.

The last part of the evaluation was to test the machine's performance under "real" broadcast situations. Although there are as many opinions of what should be a "real" situation as there are radio stations in the U.S., it was decided to use the Bonneville Broadcast Consultants automation system, and associated broadcast processing as a representative example of a high-quality system. A block diagram of the equipment used is shown in FIGURE 10.

Bonneville's microprocessor-controlled automation system accommodates 4 open-reel Otari playback decks, 2 Carousel-type cart machines based on ITC RP series, and 2 ITC triple-deck cartridge machines. The output of the program amplifier of the system comes to a switch which is located in the QC-Listening room. It switches the processing equipment and exciters mounted on a rack by the automation system.

A Series 99 playback system was connected in place of one of the triple deckers. It was activated once-per-hour, playing a three-minute cartridge over a period of eight hours.

The selection was recorded on the Series 99 recorder. Open-reel decks were fed with Bonneville's "Beautiful Music" format and the cart players played commercials recorded on ITC RP series cart recorders.

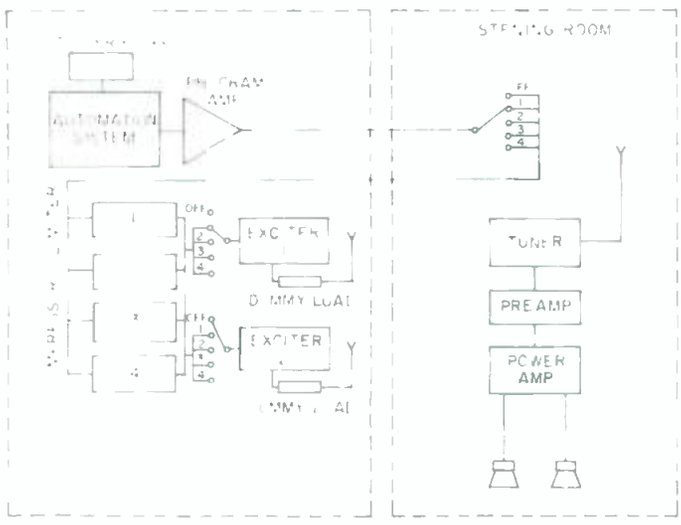
The automation system was switched to the combination of Harris MSP-90 limiter and Harris MS-15 exciter loaded by a dummy load. The exciter dissipates some 5 watts of f.m. signal on the air, providing tuners in the Bonneville offices and in the QC room with sufficient signal. Among the five people who were listening none were able to detect a difference between the selection played off the "Series 99" and the open-reel decks. At the end of the day, the playback unit did not display any significant temperature rise. The 10 kHz tone recorded on the same cartridge displayed about a four-degree departure from its original position when it was recorded in the morning.

CONCLUSIONS

The measurements and listening evaluation showed a clear superiority of the Series 99 over conventional cartridge tape recorders. There are several things, however, which cause some concern. First, the combination of an inferior mass-produced NAB cartridge, and a radio station's tight schedule, could minimize, and perhaps almost eliminate, the advantages of the Series 99. The new machine calls for a new level of production technique at radio stations. Second, its more-complex electronics will require more sophisticated (and expensive) maintenance procedures.

However, for the quality conscious radio station, the Series 99 system appears to offer a definite improvement in cartridge system performance. ■

Figure 10. Block diagram of equipment used to test the Series 99 under "real" broadcast situations.



IRVING JOEL

An Automatic Broadcast Console

A one button broadcast system—some design considerations behind the planning of the JL-412 Automatic Broadcast Console.

MOST CONSOLES AVAILABLE to the broadcaster are designed along basic outlines which were established many years ago. They fall into two general categories. In the first, a manufacturer will offer three or four models with the same basic features, the only variations being the number of inputs and a limited number of options. The other category offers a main frame, and selection of different modules. This approach allows much more flexibility and the ability to expand or change the configuration at a later date. Some models do incorporate remote machine switching, but this rarely includes audio switching.

Traditionally, 1.5-dB-per-step rotary faders are found in a large number of broadcast consoles, although some manufacturers have gone to linear conductive-plastic slide controls. To achieve flexibility and multiple input and output capability, many switches are incorporated in most models.

Physically, most of the consoles are designed with a very low profile. This was a requirement from the days of "traditional" broadcasting. With an engineer in the control room, and talent in the studio, good sight lines were necessary. But today, most operations are "combo," where one person functions as engineer, announcer, newsman, weather man and traffic reporter. The same person answers the phone, and is responsible for all log-keeping and other duties. Today, over 30 per cent of the stations are basically programmed by a syndication company. There are many that generate their own pre-recorded programs. Program formating has become quite sophisticated. It is recognized that timing and pacing consistency are extremely important. For this and other reasons many stations have found it feasible to convert to a fully-automated operation. As with any system, automation has its advantages and drawbacks. It is capable of achieving excellent program control and can be expanded to take care of logging and other clerical duties, but most of the systems are quite expensive and still require operator attention.

As a consultant to a major syndicator, the opportunity to visit many radio stations across the country and observe their operations has had a significant influence on the system which we have designed. We found that in order to provide consistency in programming, many of the controls and switches had to be wired out of existing circuits, or locked in position, to achieve a more reliable signal flow. Some of the controls were usually covered with clip boards, and many times the weather forecast wound up in front of the VU meters. The JL-412 was

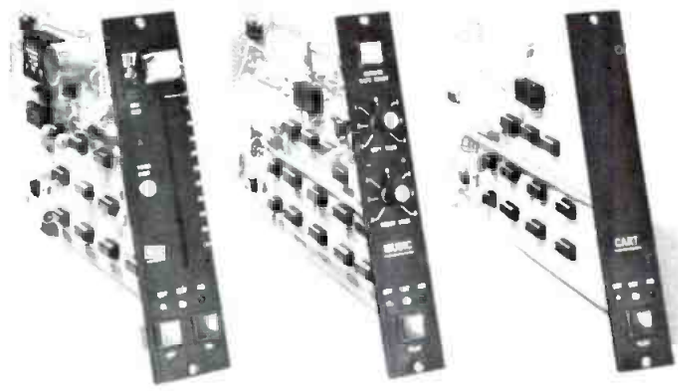
The JL-412 Automatic Broadcast Console

Irving Joel is president of Irving Joel & Associates, Teaneck, N.J.

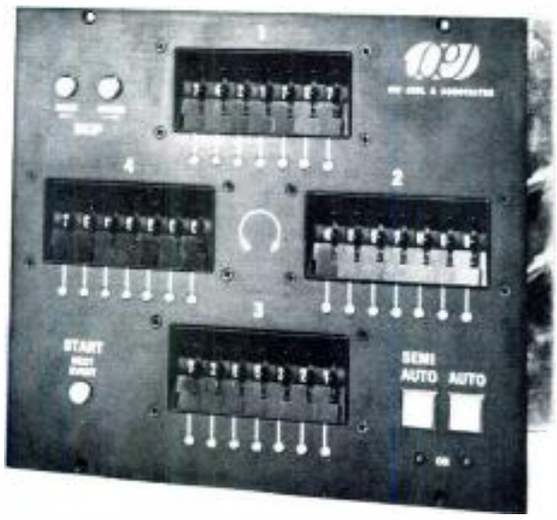
designed to overcome some of these and other problems, which had been pointed out by engineering and operating people as drawbacks in existing systems. They also suggested new features which they thought would help the operator.

At hand was the task of designing a system which would take some of the "gear shifting" functions away from the operator and assist him in getting the right signal on the air at the right time, with precise timing and smooth transitions. What we ended up with was a system that in most cases could be operated with one button. To develop a new concept, one of the first considerations is the size and shape which will best facilitate the job to be done. If laying clipboards and things on the console was a problem, why not raise the console up in the air and put the work surface underneath? But then we had to figure out how the operator could comfortably perform the operating control functions, and design the frame to accept the necessary equipment. Early in our planning stages, it was decided to use the mother-daughter board concept, which allows good flexibility, ease of maintenance and high reliability. By using this type of construction all connections from the console can be made on the rear of the mother board and all the modules are plugged into the front of the mother board. By using modular construction, each console can be configured to best suit the operations of a particular user and can be reconfigured at a later date if necessary. It also provides for ease in maintenance and trouble shooting. During the original design meetings, care was taken to insure that we weren't trying to pack ten pounds of electronics in a five pound box, and that access to all the modules would be convenient in a normal operating condition.

What stays in and what comes out? To better answer the question, we had to look closely at the functions to be performed. Our design goal was to build a system especially suited to air quarter hour/quarter hour pre-recorded music programming, with a live or pre-recorded voice. Now if we are going to have a live voice, then we better have a microphone and a microphone preamplifier. Any thing special here? YES! Voice levels are always a problem. Field tests have shown that announcers tend to be quite constant with their mic technique and individual voice levels, but that there is a fairly-wide variation in overall level from announcer-to-announcer. Since consistency is the key, we decided to use a thumb-wheel step attenuator with a wide enough latitude so that each announcer could be assigned the particular position which best suited his voice. We considered a simple on/off switch but our final design incorporates a conductive-plastic linear fader, which is used in the full-open position.



The microphone, music and cartridge module.



The sequencer module.

Since pre-recorded music is the predominant sound of the station, special care was taken to insure the highest quality and consistency of the musical program. Most pre-recorded tapes have a set-up tone at the beginning of the tape. In order to accurately reproduce the tape, trim pots are provided on each music module to precisely calibrate the system for each tape. Because the level on the tapes are preadjusted, input level controls were omitted.

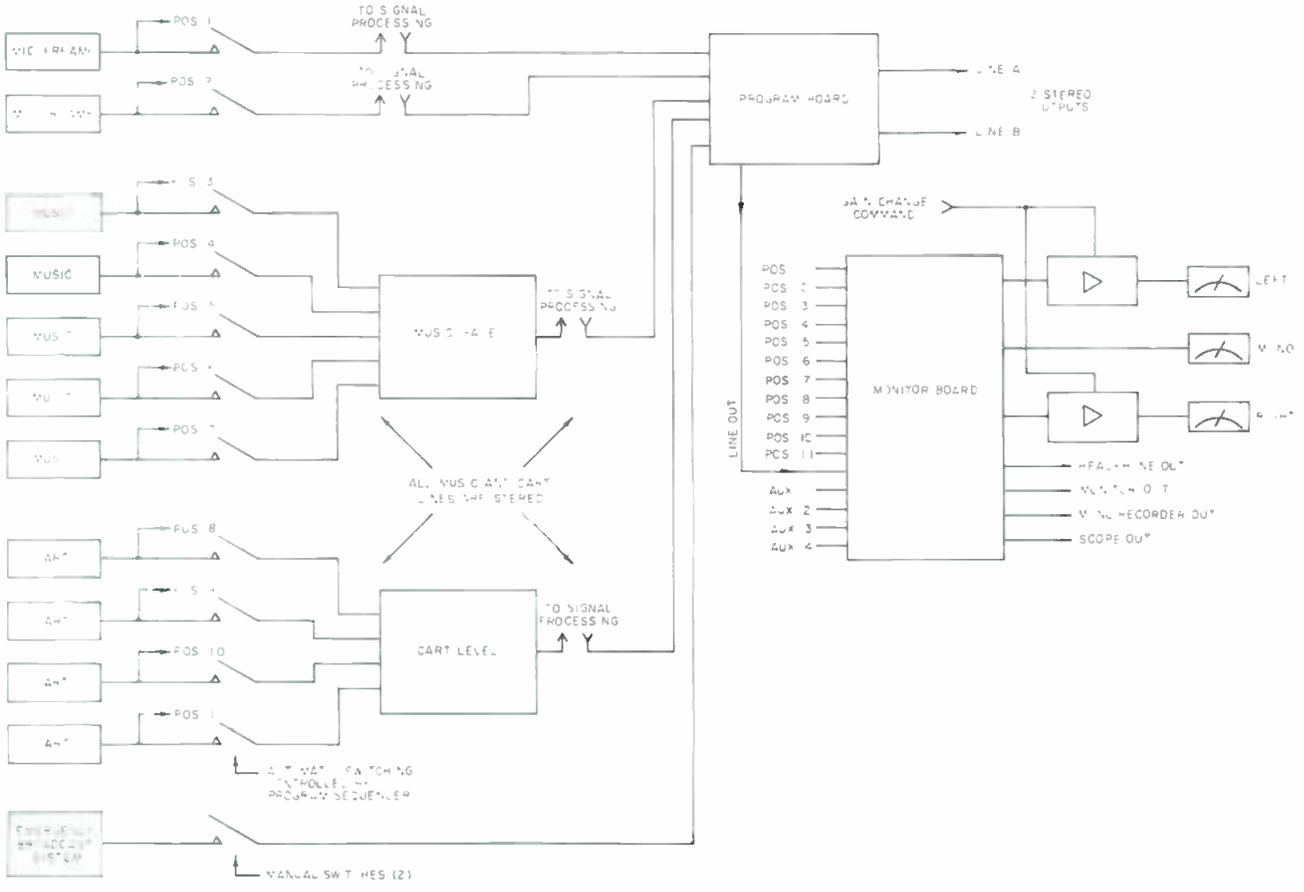
The system has been designed to allow only one source on air at any time, with the exception of the voice-over feature. Up to five music sources can be summed to one music fader. This fader is normally run in the full-open position, which actually switches it out of the circuit. If fades are necessary, then the full length of the fader is available, to produce a very smooth transition. There is also a 25 Hz cue tone at the beginning of each segment or program. To make use of this tone it was decided to incorporate a 25 Hz detector and use its output to precisely cue the tape and perform other logic functions. In the past, many of these detectors have been less than 100 per cent reliable, resulting in false triggering or missed cue tones. We are using an extremely stable phase lock loop which is not level sensitive and must see 25 Hz to operate, assuring a high degree of reliability. (For more on the subject, see **The Integrated Phase Locked Loop** in the December, 1978 **db**—Ed.)

In most stations, cartridges are used for commercials and other voice material. To maintain uniformity, it was decided to omit the level control on the cartridge module as well. There is a master cartridge module which does have a level control on it to accommodate voice-over operation and cartridge fades if necessary.

THE PROGRAM SEQUENCER

So far what we have described might be accomplished by heavily modifying an existing piece of equipment. However, the JL-412 also includes several features that are not easily accomplished by retro-fitting an older console. To assist the operator, we have included a control system which allows the console to be programmed one hour at a time, or, as twenty-eight sequential events. All of the eleven inputs can be accessed through a matrix made up of twenty-eight thumb wheel switches. To simplify operation, these are arranged in four groups of seven switches each, in a clockwise fashion. In this way the operator can easily see the sequence for the hour, quarter hour, half hour and so on. The sequencer is capable of being operated in two modes: semi-automatic, and automatic.

Figure 1. Audio block diagram of the JL-412 Automatic Broadcast Console.



SEMI-AUTOMATIC MODE

This mode allows the operator to advance the sequence on command. This automatically takes the previous event off-air, advances the sequencer to the next event, and puts a new source on-air.

In order for a source to go on-air, two conditions must be met. The source must be ready, which in the case of a tape source-means that it has played through the 25 Hz cue tone and has been automatically cued. A cartridge is presumed to be ready if it is properly seated in the machine. The source also has to be "next," as selected by the sequencer. If the source is *not* ready and is next, it will not go on-air, an alternate source will have to be chosen. This is most-easily and quickly accomplished by the operator finding a source that *is* ready and performing an insert by pressing a button on that module. An insert allows the operator to put any "ready" source on-air at anytime by pressing the play button on the appropriate input module. The sequence could also be altered by skipping the next event register ahead, or back, to a desired source.

Early in our design meetings, we looked at basic requirements of the system. A great deal of attention was paid to providing the operator with the feedback necessary to know what the console is doing at all times. Because of their high reliability and long-life expecteney, leds were used as function indicators. Different devices were tested, as well as different mounting schemes, to insure good visibility and position recognition. Switches and buttons were selected for long life, quiet operation and positive feel. Color coding was used to identify status and function. Each module has three led indicators on it; green indicates that the source is ready and yellow is used to indicate that that module will go on-air next. Yellow indicators are also used under each thumb wheel so the operator can easily identify the thumb wheel associated with the next event. Red leds are used to indicate that that particular module is on-air. In this way, the operator can see what the console is doing now and – most important –what event will go on the air next, and its ready status.

AUTOMATIC MODE

In the automatic mode, the sequencer is advanced to the next event by detecting the cue tone on a music source, or the secondary tone on a cartridge source, and interpreting it as an end-of-message signal. When this signal is received by the control module, the next event is automatically put on-air. In this way, automatic programming can be continued for as long as desired. During the planning stages many programming possibilities were investigated, including the use of a microprocessor. The thumb wheel display was chosen for its direct access by the operator and its ability to constantly display the entire program reliably.

MONITOR SECTION

The monitor section of the console allows the operator to look at and listen to any input, the output of the console or four auxillary sources, one of which would normally be a high-quality off-air monitor. By providing this flexibility the operator can make A-B comparisons between any input source, the signal leaving the console and the air signal. Three VU meters with special meter amplifiers are used to display left, mono sum, and right signals. Logic is provided to vary the gain of the left and right meter amplifiers. When a mono source is on-air, or selected as an input to the monitor, the gain of the left and right meter amplifiers can be automatically increased to compensate for input level differences between mono and stereo sources.

There is also a manual plus-10 dB switch which precisely increases the gain of the left and right meter amplifiers. This is very useful for looking at low level test tones.

In many installations, the console will be the prime source of on-air audio. To insure a clean feed to the transmitter, two identical sets of outputs can be provided, one of which is usually dedicated as the prime audio feed. The second output can feed a complete backup link to the transmitter, as well as being bridged to provide in-house monitoring.

To reduce the number of external items necessary in the control room we decided to provide a digital time-of-day clock, and a digital event timer, as part of the console. The event timer is automatically reset each time a new source goes on-air. Five-watt earphone amplifiers are included in the monitor section, but the loudspeaker amplifiers (which we feel should be no less than 50 watts-per-channel to adequately drive most control room loudspeaker systems) are not built into the console.

Other front-panel monitor functions include monitor level, headphone level, left and right louspeaker mute, and mono from both loudspeakers. The monitor is automatically muted when the microphone is on-air, or being previewed.

Early field tests proved our design concept to be workable. With very little training, operators became comfortable with the system. But no matter how many preliminary checks are made—and even though we actually operated the console with tape machines, cart machines and live microphones—it wasn't until we got into the field and actually hooked-up at the radio station did we discover that everything wasn't working the way we thought it should. Things were fine until we went from live microphone to music. At least that's what we wanted to do, but the system wouldn't budge yet it had all worked beautifully back at the shop. After a little head scratching, and a couple of quick measurements with an oscilloscope, we found that the in-house relay used to turn the telephone bell off and the air lights on, was feeding a 150-volt spike down the line. A well-placed diode cured that problem, and we have had a year of reliable on-air time to prove the system.

The JL-412 design project has been a team effort, with Josh Weisberg responsible for the PC board layout and supervision of the console construction, Howie Lieberman for the audio design, Keith Morris for the digital design and Hugh Russell for the mechanical design. ■

DOUGLAS F. DICKEY

The Solid State Logic Studio System—Part II

The SSL computer automation system—its features and operation.

PART I OF THIS FEATURE ARTICLE appeared in the August, 1979 issue of db. In part II, we will take a look at what the SSL system does, and how it does it. We begin here, with cue commands.

CUE COMMANDS

As mentioned last month, Ultralocator functions are heirarchal. Within each "title" segment, unlimited "Cue" positions may be named or numbered. A named cuepoint is generally most useful because the name will remind you what the cuepoint is about. For example, "Guitar Break," "Flat Note," "Drummer Falling Into Piano."

Generally, cues are named in two ways. One, you can write the Name Command line completely except for the Execute Key, such as: (NAME) (CUE) Bridge, then play the tape with the manual (non-computer) play button. When the tape reaches the cuepoint, press (EXECUTE) and the cue you have named will be labelled at that point.

Alternatively, the computer can assign sequential numbers to cuepoints. First, the tape is played under computer control [using a (PLAY) command line]. Then, at each desired point, simply press the (CUE) command key. The next available number will then be assigned to that point on the tape, up to a maximum of 999 per title. These numbers can then be given meaningful names very simply. The command (NAME) (CUE) 17 (NAME) GUITAR SOLO will assign cuepoint 17 the name Guitar Solo; and so forth.

The operator's vocabulary is very quickly increased simply through use of the system. With no training, you can now command the system to play the guitar break of "Shake It Up," or to cycle from the beginning of the song to the bridge, or any number of similar useful commands:

(PLAY) (TITLE) S (CUE) G

(CYCLE) (FROM) (TITLE) (TO) (CUE) G

It is true that a really hot-shot drug-free tape op with a sharp pencil and a fat notebook can perform these functions, but not with the speed and unerring accuracy of the SSL Studio Computer. The software is modified to accommodate the precise specifications of the machine it is controlling. The weight of the spools, distance to the target point, acceleration and braking characteristics are all factored into the Ultralocator "brain," which calculates the most efficient transport commands for each search.

DROP-IN COMMANDS

Another standard procedure which benefits greatly from computer assistance is the drop-in process. Solid State Logic's proprietary SuperCue™ circuitry, described previously, is also tied into the computer, enabling the drop-in to be rehearsed with no danger of erasure.

The computer must be given authority to initiate any record commands. This is done by activating the "Record Enable" button at the top of the control panel. During drop-in rehearsals, this button is left *off*. Then, the engineer plays the tape with the manual remote transport control, and enters the command DROP-IN. The CRT displays the words "Drop-In." When the drop-in point is reached, the engineer hits (EXECUTE) instead of the Record button. The CRT will display the phrase "Drop-In At 1:23:11, for example. This means the drop-in will occur at precisely 1 minute, 23 seconds and 11 frames. (EXECUTE) is hit again at the point where the machine is to leave the Record state. The CRT will display an additional phrase "Drop-In Out At 1:47:09."

Next, the computer is commanded to (PLAY) (DROP-IN). This command will cause the computer to pre-roll the tape to 10 seconds before the drop-in, play the tape, and display a countdown in seconds and frames to the drop-in point. At the drop-in point, the console's SuperCue circuitry will simulate the tape machine's switching to allow the drop-in's timing to be evaluated. The circuitry will simulate the drop-out switching at the programmed time, and stop the tape. Of course, if AutoCue has been selected, communication between the studio and control room will automatically open so the drop-in can be discussed.

If all is in order, the engineer activates the computer's Record Enahle switch, and commands (PLAY) (DROP-IN). The sequence will repeat, only this time the computer will cause the multi-track to drop-in and out of record at the programmed times.

Incidentally, the pre-roll time is only nominally set at

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

Figure 2. The system has been programmed for a drop-in that will begin at 2:44 and end at 2:46. It's now 2:38, with six seconds until drop-in.

10 seconds. If the artist wants to hear the entire verse prior to the drop-in, or some similar requirement, the command line (PLAY) (DROP-IN) (FROM) whatever cue or time, can be substituted the first time around. Subsequently, the programmed pre-roll will occur simply by commanding (PLAY) (DROP-IN).

DROP-IN SAFEGUARDS

As mentioned, the computer must be specifically armed with Record Enable authority before it can perform a drop-in sequence. As absolute accuracy is obviously required, the drop-in sequence will not function if SMPTE time code is not present. If the system loses code in mid-cycle, it emits an audible warning and the word SMPTE flashes and then disappears from the CRT screen. The SSL Studio Computer System performs all functions, including drop-ins, well within frame accuracy. One frame of NTSC SMPTE code is approximately 33 ms, which is faster than the bias switching and rise time cycle of all commercially available analog multi-tracks.

The operator can abort from any situation at any time simply by pressing the blank "escape" command key. Finally, unlike the other system data, all drop-in sequence programs are automatically deleted when the next function is commanded. They are never stored, preventing the computer from placing the machine in Record on some future session. With these safeguards, the possibility of a damaging error is substantially less than that existing when drop-ins are under human control.

Most importantly, the system eliminates much of the stress associated with the critical drop-in process. This frees everyone involved to concentrate fully on obtaining the best possible recorded performance, in a substantially relaxed atmosphere.

NAMING TRACK LISTS

Each Title has an independent track list which can be written, added to or changed at any time. The initial command sequence is simply (NAME) (TRACK). This command produces a blank form on the video monitor, with a flashing marker next to the number 1. A description of the track, up to 15 characters long, may now be typed for Track 1. Pressing (EXECUTE) completes the line and moves the marker to Track 2, where the process is continued.

Tracks are generally named as the session progresses. Thus, the engineer can fill in the lists while the artist and producer are working on other matters. Usually, several titles will be recorded on the initial date with approximately the same assignments. The need to write the same list repeatedly is eliminated by the ability to copy track lists.

When working on Title B, the track list from Title A can be directly transferred. This is accomplished by the command line (NAME) (TRACK) (FROM) (TITLE) A. The new track list, headed by the name of Title B, will appear on the screen and be stored on the reel disc automatically. The flashing marker will appear next to Track 1 in case any changes need to be entered.

It is also possible to add or change a specific track name without cycling the marker to its position. The command (NAME) (TRACK) 17, for example, will cause the track list for that title to be displayed, with the marker flashing by the designated track. Typing the new name and pressing (EXECUTE) will complete the process.

LIST AND PRINT COMMANDS

Any system data can be displayed or printed out at any time. Again, the command structure is quite simple. (LIST) (EXECUTE) will cause the list appearing in FIGURE 4 to appear on the screen. Changes to this list are entered by typing the first two letters of the category [EN for Engineer, SP for Speed, etc.] plus a space and the new information. (PRINT) (EXECUTE) will cause this information to be

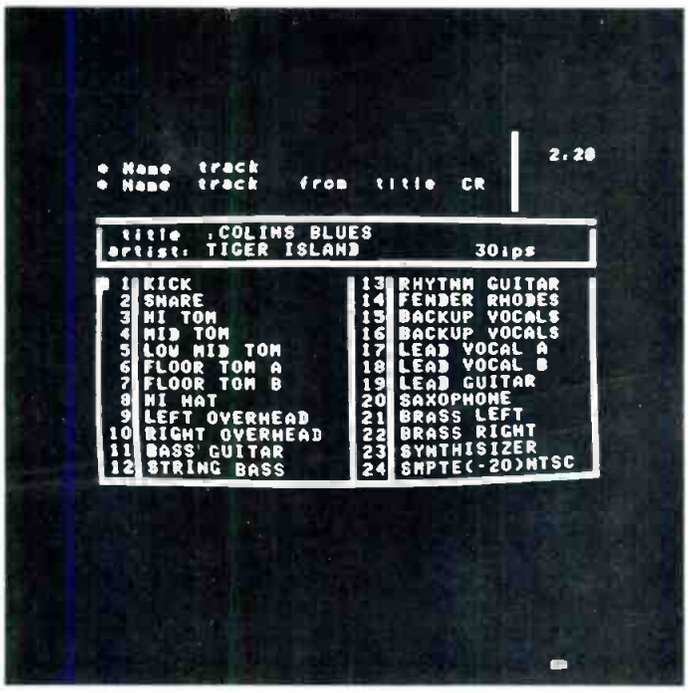


Figure 3. The track list for "Colin's Blues."

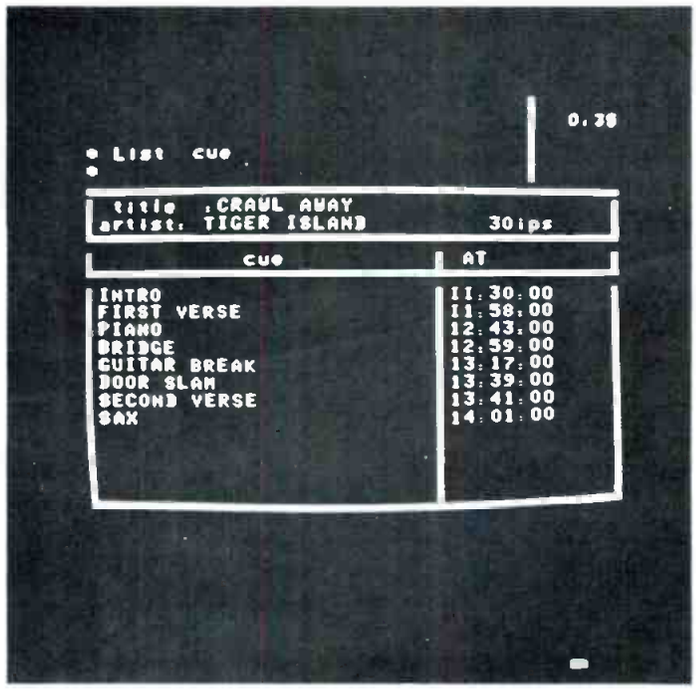


Figure 4. The VDU displays the cue list for the hit single, "Crawl Away."

printed out, for use as a tape box label, or for the producer's notebook.

(LIST) (TITLE), (LIST) (CUE), (LIST) (TRACK) will cause the requested information to appear. If several "pages" of such information exist, the computer will display the statement "There is more, do you want it?" Type Y [Yes] will cause the next page to be displayed, and so forth. Substitution of the command (PRINT) instead of (LIST) will cause the requested information to be printed.

SUMMARY OF NON-MIXING FUNCTIONS

As you have seen, the SSL Studio Computer readily performs many useful automation functions. This is a prime example of the payoff afforded by total systems integration. None of these functions could be obtained at reasonable cost with comparable sophistication if they were conceived as individual units. By combining them into a comprehensive system, the cost-versus-performance ratio is inverted in favor of the artist, engineer, and producer.

We hope that it is clear how easy the system is to instruct. Furthermore, the software has been written to encourage experimentation, leading the operator to useful discoveries in a very natural way. The reader is cautioned, though, that on occasion the computer loses patience. At a recent session, a visitor entered the command line (PLAY) (FROM) (HERE) (TO) Eternity. The computer retorted, "ETERNITY is not defined in this version, you filthy pervert!"

MIXING OPERATIONS

In the construction of the SSL mixing programs, there was a more-rigid criteria than existed for the non-mix functions just described. Many of those functions never existed before, so there was no tradition to adhere to. Mixing, on the other hand, is an established and delicate art. Solid State Logic felt it imperative to devise a system that was natural to work with, predictably and meticulously accurate in storing and recreating the engineer's moves, and as non-demanding of special attention as could possibly be achieved.

Fortunately, there was no need to cope with retrofit considerations into consoles of various manufacture. The system could be built specifically for the Solid State Logic desk, which gave us enormous design latitude. This freedom is reflected in the physical presentation of the system.

The only additional console control required for the mixing automation is a single pushbutton mounted at the top of each fader panel, to the left of the fader track. Two

leds, one red and one green, are the only additional indicators.

A single button is all that is needed because the SSL software controls its function at all times. This intelligent assistance eliminates the need for extra thought on the part of the engineer, who is busy enough during mixing anyway.

Mix operations are initiated by depressing the MIX ON/OFF button at the top of the computer command center. The legend "MIXING" then appears on the video monitor. Next, the transport is instructed to (GO TO) the desired location. When it arrives, the tape stops and the console emits a warbling sound to draw attention to the video monitor. There it says "You Can Adjust Fader Status Now. Press Execute to Continue."

Actually, no major adjustment is required. The computer has switched all faders to their Absolute mode, and indicated this by igniting all of the red leds on the fader panels. Should the engineer wish to establish a mix of the rhythm tracks only, he or she may depress the single push-buttons on all other channels. This will cause those faders to enter the Isolate mode, in which their positions are not written into the mix program. [Pushing the button again will restore the fader to Absolute status. The fader will not cycle into "Trim" status at this point, because the computer knows that there is no previously stored data to trim.]

Once this is done, the engineer presses (EXECUTE), and the tape starts to play. All movements of any faders in "Absolute" are written into this mix program, which is temporarily stored on the left-hand "scratch-pad," or system floppy disc. The video monitor also switches at this time to display a bar-graph of the instantaneous d.c. levels of the mix in progress.

When this first mix attempt is completed, the engineer enters the command (END). This stops the tape, and causes the computer to display the inquiry "Name?" If the mix was worthwhile, the engineer enters a Name, which causes the mix to be transferred to a file on the right-hand "Reel" disc, and locked in that file under the assigned name. When the mix is named, the name is entered into the Mix List file, along with its start and end points.

Mixes are assigned names rather than numbers for purposes of clarity. As the system can store large numbers of mixes and freely re-assemble them, it is useful to have mix labels which remind the user what they are. ["Brass too loud" means a lot more than "Mix 29," particularly at 3:00 a.m.!]

Updating a mix is also easy. Every time a mix is played back, it generates an automatic update. This is another advantage of using a separate store because this new mix will never displace an old one. There is therefore no need to distinguish between playing and updating a mix—simply type, for example, (PLAY) (MIX) [optional name]—that is; play the mix just created [or the one named].

The system now sets all the faders to UPDATE [green led] [except any which were isolated on the original mix —since on these, there is nothing to update. They are therefore set to WRITE]. There is no need to set the faders to any NULL point [either by a mark on the fader strip or by using nulling lights]. Instead, whichever positions the faders are in at the beginning are read by the computer and used as null points. As the playback progresses, the computer compares the current position of each fader with its position at the beginning. If the fader has moved, the difference is calculated and used to adjust the level of the old mix to a new value. These new values can be displayed on the video monitor as a bar-graph, so the actual position of the faders is irrelevant. If any levels are getting too close to the ceiling or the floor, this can be seen at once.

The combination of automatic set-up of the channel status and nulling level saves a great deal of time. In fact, most often, there is no need to do *anything* but ask for the mix to be played, adjust levels, and name the results. Of course, the setting up can be manually overridden. Using the fader pushbuttons, the statuses can be cycled to ISOLATE or WRITE. In this way, a mix can be built up, channel-by-channel.

Figure 5. The bar graph shows the d.c. levels generated by each fader.

Incidentally, notice how the same pushbutton has different functions at different times. For the new mix, it cycles between WRITE and ISOLATE. Since there is nothing to update, the computer locks out the UPDATE state. For the playback and update, all three states are possible, so the pushbutton cycles around all of them. During update, the pushbutton can be used for a further purpose—suppose you have a mix in which one (or more) channels need to be boosted briefly, then returned to *exactly* the original level. The matching of this level can be done automatically by the computer and is achieved very simply.

The boost is made; then, at the same time, shortly before returning to the original level, the button is pushed. Both leds now light up to show that a request has been made. They remain lit as the fader is moved back down, until the level exactly matches the original one. Then, they both go out, and the fader becomes isolated. Any further movement is ignored by the computer, and the result is an exact return to the original level.

It is not necessary to play all of a mix to update part of it. This can be done in either of two ways. A new mix can be made of just the relevant section. For example, (PLAY) (MIX) (FROM) X (TO) Y [where X and Y could be timecodes, or cuepoints, or anything else that describes them].

Then, this new mix can be added into the old one, without playing tape at all. The command line is straight-forward. Suppose you have a mix A from 5:00 to 10:00, and a mix B from 6:00 to 7:00 which is an update of that section of mix A. To insert the update, enter (JOIN) (MIX) B (TO) (MIX) A, or if you only want to insert part of mix B, (JOIN) (MIX) B (TO) A (FROM) 6:15 (TO) 6:30.

If there are any mis-matches of level, the computer will list them, and you have the option of having them smoothed over the period you select.

Alternatively, you can update the relevant section alone but output the rest of the original mix as well. This is done by a command line such as this:

(PLAY) (JOIN) (MIX) A (FROM) 6:00 (TO) 7:00

In this case, the tape will only play from 6:00 to 7:00, but the new mix will run the full length of mix A.

A song does not have to be mixed as one complete unit. Instead, different sections can be mixed independently, and later joined together to form the final mix. For example, you might want to mix the introduction, vocal section and instrumental break of a song separately. Each of these mixes can be given the appropriate name and refined separately. Then, the mixes are joined to produce a new complete mix. Joining is also useful simply to replace unsatisfactory sections of a mix.

There are two types of JOIN: Butt joining and insert joining. The command format for a butt join, which joins two mixes end-to-end, is as follows: (JOIN) (MIX) B (TO) (MIX) A. This command results in a new mix being created, which consists of mix A up to the start time of mix B, then mix B until its end time.

If the mixes overlap, you can select the point of joining, for example, (JOIN) (MIX) B (TO) (MIX) A (AT) 11:00. When an (AT) time is stated, the new mix stops at the end of mix B, even if mix A actually extends further.

An Insert Join inserts all or part of one mix into another. This is achieved by typing a sentence such as (JOIN) (MIX) B (TO) (MIX) A (FROM) 11:00 (TO) 12:00. Mix B, or a section of it, now replaces mix A in the output mix. Two examples of command lines for Insert Joins are shown here: (JOIN) (MIX) A (TO) (MIX) B. Mix INSERT A is inserted completely into mix B. To insert only part of a mix into another, enter (JOIN) (MIX) C (TO) (MIX) A (FROM) 11:00 (TO) 11:15. Mix C only replaces mix A between the stated times. It is not necessary to include both the FROM and TO times; if either of them is left out, the computer assumes the start or finish of the insert mix.

LEVEL MIS-MATCHES

It is possible for there to be level mis-matches at joins. If a mis-match occurs, it is detected by the computer and a message appears listing the mis-matches in dB and the channels on which they occur. You are then asked "Do You Want Them Smoothed Over?" If you answer "Yes," you are further asked for the number of seconds over which to do it. The mis-matches are then ramped out smoothly.

COMPUTER RECAP

This brief look is not intended as a complete explanation of the SSL Computer System's total capabilities: that would require a small book. It *is* intended to provide a glimpse of a different [and we feel improved] approach to the problem of providing full computer assistance with the least possible pain or re-learning. Due to space limitations, many features have been quickly sketched, and others not even mentioned. Perhaps we can cover these in a future article strictly about the computer.

The essential points are these: A multi-purpose system can achieve enormous sophistication at a relatively low cost-per-function. These functions can be accessed with easy-to-understand English commands. Programmable mixing can be almost as simple as manual mixing; and with extensive mix data manipulation capabilities, difficult mixes become easy, while impossible mixes become possible.

Finally, software-based systems allow their owners to keep up with the state-of-the-art inexpensively. More than six man-years of software programming have already gone into the Solid State Logic Computer System. The next two years worth of design goals are already scheduled. Many of these refinements and new features are the direct result of feedback from engineers and producers already using the system; and for the cost of a floppy disc, these users will be able to upgrade their systems to their own ideals as soon as the improved programs are completed.

It is also worth noting the growing number of studios which have at least one computer-conversant technician on their staff.

With the addition of an inexpensive programming unit, these studios will be able to add custom modifications to their own systems. For while the SSL system has been designed to provide numerous standard functions to the non-computer-oriented, it is nonetheless a powerful computer with vast unexplored studio potentials.

CONCLUSION

While this article is by no means a complete catalogue of the possibilities created by the Solid State Logic Master Recording Console and Studio Computer System, it hopefully sheds some light on the potential of total systems integration. The engineer is still confronted with an enormous number of controls, but tandem switching networks, software-defined switches, computer power and intelligent, thoughtful human engineering have been applied to reduce the mechanical burdens of managing the modern control room, freeing the production team to exploit the full benefits of the technology.

Someday we may just sit back and telepathically transmit thought-mixes directly to laser-beam etched tetrahedrons manufactured from synthetic crystals grown in stationery earth-orbit at L-4. Until that day, we can at least avail ourselves of innovative control synergies, such as those embodied by the Solid State Logic Studio System. ■

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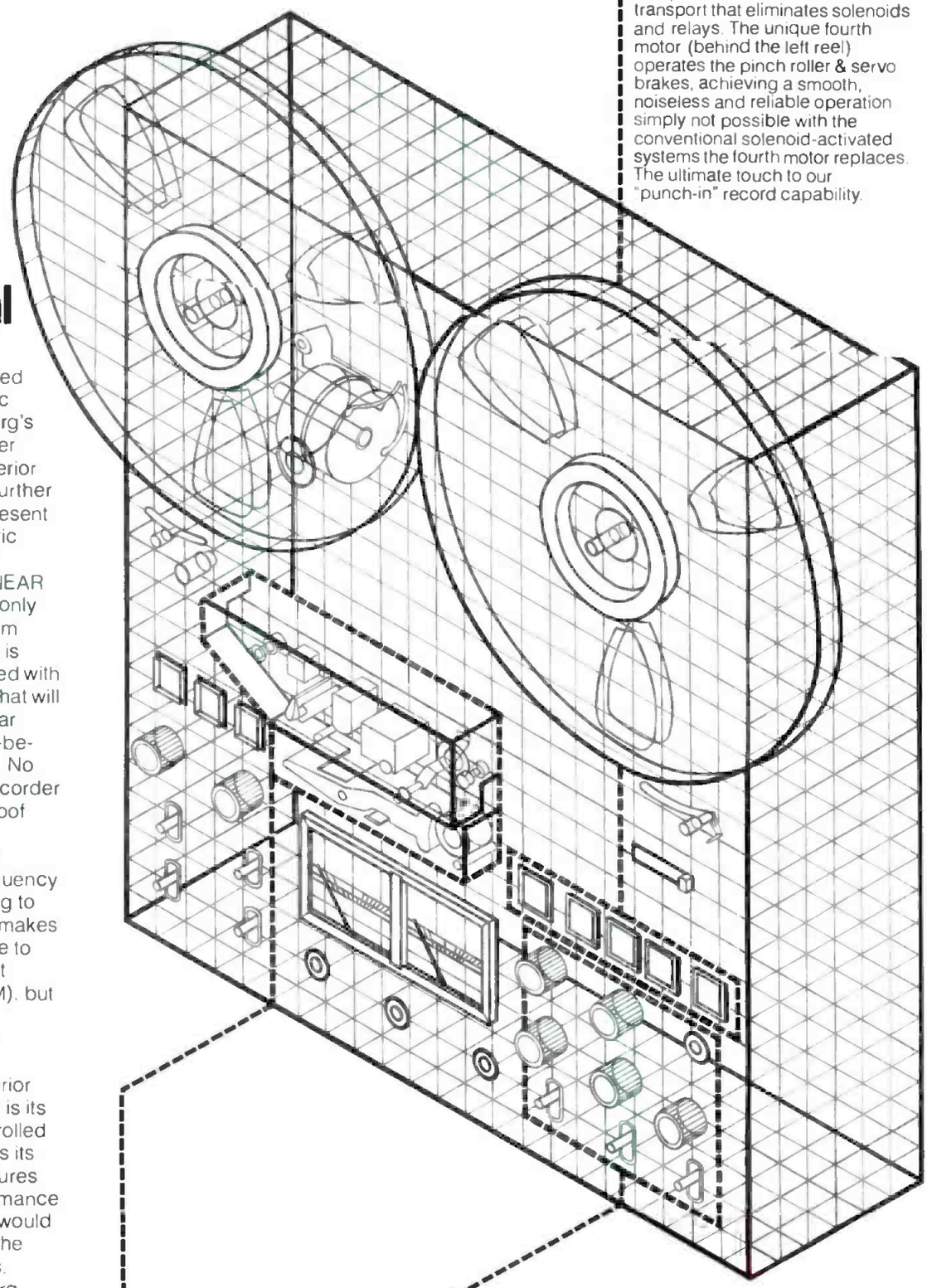

Tandberg's exclusive PROM computer-controlled four-motor transport that eliminates solenoids and relays. The unique fourth motor (behind the left reel) operates the pinch roller & servo brakes, achieving a smooth, noiseless and reliable operation simply not possible with the conventional solenoid-activated systems the fourth motor replaces. The ultimate touch to our "punch-in" record capability.

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JOHN BORWICK

Convention Report: APRS 1979

Highlights and impressions of the 12th Annual Association of Professional Recording Studios (APRS) Exhibition.

IN THE SEPTEMBER 1978 issue of **db** I reported on last year's Exhibition of Professional Recording Equipment organized by the Association of Professional Recording Studios. Now, footsore but buoyed up by the generally optimistic view of the state of the audio art voiced by most of the engineers I met, I can set down a few impressions of the 1979 Show—the 12th APRS annual exhibition.

The venue was again the resplendent Connaught Rooms in London—a mere 200 yards from Kingsway Hall, the most popular classical recording hall in the UK. More than 90 companies were exhibiting mixer desks, tape machines, microphones, speakers and sound processors of every description, to a total value calculated (not by me) to exceed $5,000,000. This has grown into a truly international focal point exhibition, with studio managers and similar potential customers from home and overseas converging on the representatives of virtually every important manufacturer supplying the recording and broadcasting industry. I will perhaps be excused if I seem to concentrate more on European products than Stateside ones. All the big USA names were at APRS, but their latest equipment had mostly been shown already at the AES Conventions and reported on then. (See **Convention Report: 63rd Audio Engineering Society Convention,** in the August, 1979 issue of **db**—Ed.)

DIGITAL RUMBLINGS

A unique example of British/American cooperation was cleverly stage-managed, with the news release embargoed until June 20th, the Show's opening day. This announced a licensing agreement between EMI Limited and MCI of Florida, "under which MCI will manufacture digital tape recording equipment based on technology developed by EMI." A prototype MCI JH-220 two-channel digital recorder, the first machine to be completed under the agreement, had been rushed to the Show. The buzz went round that a privileged few would be given a demonstration after the Show closed that evening, and sure enough a huge army of us—privileged and unprivileged—swarmed into the room above a pub across the road (well, can you think of a better place to launch a new recorder?) where the JH-220 replayed some of EMI's latest digital tape masters. As expected, the lack of modulation noise and any kind of wow and flutter was very impressive.

The amazing thing about digital tape mastering is the way it has run right through the recording world in the past few months. The PCM discs produced by Nippon Columbia (Denon) in Japan for more than two years have produced only a few yawns, but now every major record company seems to be taping digitally like mad. The EMI/MCI machine could not stay at APRS for the remaining two days because it was needed for sessions at the Abbey Road studios. I was at a Decca recording session in the same week (the USA's favorite tenor, Pavarotti, making a new arias album in digital) and their hook-up of three own-make digital recorders plus editing unit is in practically continuous use. (It had just been flown in from recording "Fidelio" complete in the USA, and I am invited to the editing sessions.)

Another British/USA combined effort in this field, of course, is the 3M digital recorder made to a BBC design. It seems that studios are placing orders for these machines apace. EMI/MCI are offering their 2-channel recorder this year, with a multichannel version and editing system by early 1980, and I saw on a recent visit to Sony in Japan a production line of their PCM-3224 24-channel digital recorders, every one already sold to local studios.

Other applications of digital techniques were everywhere at the APRS Exhibition. On the MCI stand itself was the

The "APRS 79" show occupied eight of the Connaught Rooms.

John Borwick is audio editor of the British publication The Gramophone.

A prototype of the MCI JH-220 two-channel stereo tape recorder—the first machine to be completed under the licensing agreement between MCI, Inc. and EMI Limited.

formidable AMS DMX 15-80 digital delay line (costing around $5,000) with an interesting harmonizer add-on. The EMT 444 uses only a 12-bit A/D converter but cleverly boosts low level signals at three thresholds to give 15-bit performance with a 115k bit static RAM register. What this boils down to is great operational versatility with, for instance, delay time from 1 to 255 ms in 1 ms steps and even scope for 0.5, 0.25 and 0.125 fractional incrementing, echoes at –0.5 to –60 dB with 4 Hz to 1 kHz repetition frequencies, phasing, etc. etc. An even newer EMT toy was their EMT 446 digital message storage unit, which can do so many memory jobs that my mere human memory has failed to store them all.

FINDING THE PLACE

The days of the junior tape-op perched on a stool, logging counter readings on the tape boxes and spinning back to replay Take 38 for the umpteenth time are numbered. Everybody now wants to sell you an auto-locate device. An interesting example, which again illustrated a degree of transatlantic cooperation, was the XT-24 Intelocator ($2,600) designed by Audio Kinetics (UK) with encouragement from the British branch of 3M, who handle its distribution everywhere except the USA (where the distributor is to be Quintek Inc.). This is described as "the first intelligent autolocator" and it actually finds the place on a tape quicker the second time round. It is programmed to learn the behaviour pattern of the particular recorder/tape, searching at high speed, then entering a momentary slow-down phase and finally speeding up again to decelerate firmly at the location point without overshoot. (For a little more on the XT-24, see **Report from Hamburg**, in the May, 1978 issue of **db**—Ed.) Audio Kinetics also showed the QLOCK 210 synchronizer ($11,600), a multi-microprocessor SMPTE generator/synchronizer for locate and lock operations on any pair of audio or audio/video machines. It makes 48-track operation child's play and offers single frame video ac-

Audio Kinetics XT-24 Intelocator—"the first intelligent autolocator."

curacy. It even has a "Cascade" feature allowing two QLOCK 210s to control three machines or accept commands from an external computer.

Studer has brought microprocessor ("μP") control of all local and remote function instructions to a fine art in their A800 multichannel recorders. The μP is used to define all functions, status indicators and displays by software only, giving flexible control from a rack-mounted panel which features electronic editing by exact timing of erase and record head switching.

Neve, famous for their NECAM computer assisted mix-down system, have not been resting on their laurels—or their motorized faders. Their latest 8108 Series of recording consoles take a huge leap into the technology of the 1980s. The version on show acted like a huge magnet to visitors. It had 56 input channels with fully parametric equalizers and 48-track outputs. Mother boards and modules dispensed with the traditional frame wiring looms, but the central assignment panel was the main point of interest. This had touch-sensitive assignment zones instead of buttons, with alpha-numeric "Channel to Track" and "Track from Channel" readouts, and "interrogation" keys beneath every fader to reassure the engineer. Complex assignment patterns could be stored in 4 integral memories for instant recall. Each channel strip had primary and secondary faders with the choice of VCA sub-grouping, NECAM or manual faders available. All along the raised back portion of the desk was an impressed array of new high resolution (200 segments) baragraph meters, switchable to VU or PPM characteristics.

Two other intriguing Neve novelties were the rack-mounted Multi-channel Monitor and Spectrum Analyzer. The former could display on a color video monitor up to 56 input levels in the form of baragraphs, VU or PPM, arranged in groups of 4 or 8, color coded (in up to 8 colors) to identify channels, reverb, etc. The electronically generated graticule was in 2 dB steps between –24 dB and +6 dB, with zero adjustable and the baragraph color changing to red above a preset overload

The QLOCK 210, a multi-microprocessor SMPTE generator/synchronizer by Audio Kinetics.

Lyrec of Denmark unveiled their ATC (Audio and Tape Controller) desk-top computer interfaced with the TR532 multi-track recorder and its Tape Position Controller to perform every cueing, recycling and locating operation yet devised. A standard PET 8K home computer on the Eventide stand was tied to a real-time analyzer board to give incredibly flexible third-octave displays. It could be programmed to compare old and new data and work with the Eventide harmonizers, 1745 M digital delay line and Omnipressor. This was one of a number height. The real-time Spectrum Analyzer covered 31 Hz to 16 kHz in third-octave bands and could be passed through a UHF modulator to drive a normal television receiver.

The Neve 8108 Series recording console. Note assignment panels in foreground, and baragraph meters along the raised panel.

of exhibits from which I found a demo recording cassette in my doggy-bag of memorabilia when I got home from the Show. Another was for the Ursa Major "Space Station," showing all its delay, echo and reverb effects. A two-sided disc was needed to extol and demonstrate all the tricks you can play on the Marshall Model 5002 Time Modulator—such as cyclic detuning of one track in Automatic Double Tracking "to keep the second track warm," as the demonstrator described it.

Special effects units, harmonizers and vocoders are difficult to show at a no-noise exhibition like the APRS, and so many stands were packed all day long with groups of visitors wearing headphones, being talked to by a demonstrator holding a microphone while he flanged, pitch-changed and chorus-effected on a prepared tape. Hence the new fashion of hand-out demo recordings.

ANALOG FIGHTS ON

Standard analog tape machines continue to improve and proliferate. The British companies of NEAL and Ferrograph, recently amalgamated, were doing good business both with the NEAL professional cassette decks in 3- and 4-track versions having sync tracks and built-in calibration oscillators, and the latest Ferrograph open-reel recorders. Stellavox of Switzerland are famous for their SP8 professional portable and AM148 mixer. These were joined by a new TD88 slim-line mains recorder designed for universal application, taking up to 14-inch diameter reels of ¼-inch tape, 10½-inch of ½-inch (up to 8 tracks) and 8-inch of 16mm perforated film stock. Their great Swiss rivals, Nagra were showing the even more popular Nagra IVS battery portable with its 10½-inch reel adaptor, and a very interesting mains recorder at present in data recording form—to be followed by an audio model next year.

Neumann and Ortofon continue to be the top names for disc-cutting equipment in Europe, and their exhibits at APRS 1979 confirmed this. Cybersonics made a brave show with their compact Disc Master 2002 lathe, attractive for smaller studios and direct-cut recording on location (basic price around $47,700). Audio & Design (Recording) have built their reputation on compandors, and particularly the "Scamp" modular series which now totals 13 different modules. They were even showing blank modules in various widths for user-built power supplies etc. The latest ADR product was the Express Limiter, with knobs for input, output, attack and release but momentary gold plated spring buttons for all other functions.

A clever ME 501 tape head tester ($13,000) that had me wondering why nobody had thought of it before was shown by Wollke of Germany. It uses a 95cm tape loop with a test head mount to suit all types including multi-track heads. Height and azimuth can be set accurately and bias voltage is infinitely variable in four ranges. The trick is that the magnetic properties of a short piece of tape are practically constant, and so checking heads against a standard head gives 100 per cent accuracy.

Cordless foldback systems have been around for a year or two, since their introduction independently by Sennheiser and Beyer Dynamic, and the infra-red transmission technique has done much to reduce the spider's web of headphone cables in studios and on stage. A new gadget called "Cuemix," introduced by Design Electronics of London carried the idea a little further. The box, mounted on a goose-neck stand and with a battery-powered amplifier, allows each musician to produce his own foldback mix of up to 5 signals, each on a different carrier frequency on an induction loop system, with stereo panning. This should cut out time consuming requests to the studio engineer for a different mix, or just more volume.

Cuemix receiver for foldback headphone control.

Loudspeakers were shown here and there, but APRS is an open-stand "quiet" exhibition, as I have said. So apart from inviting visitors to risk a hernia by lifting speakers to show how heavy they were—exhibitors had to push likely customers into a taxi-cab and run them across town to their showrooms if they wanted a working demonstration. The British company Vitavox make particularly heavy speakers, but are very proud of the fact that their popularity in Japan is now so great that a Vitavox Appreciation Society holds regular meetings in Tokyo.

As a postscript, I might just mention that one week before the APRS Show, the world (or that lucky part of it that attended the June meeting of the AES British Section) heard for the first time a new prototype electrostatic loudspeaker. The designer, Peter Walker, who developed the famous Quad Electrostatic 21 years ago and still has a waiting list of customers buying all he makes at his Acoustical Manufacturing Company factory, gave a stimulating talk. His new speaker uses concentric ring electrodes, with progressive delay to the outer rings in simulation of the wavefront at a plane in front of an idealized spherical wave source. A neat idea which produced very impressive sounds. Is this the speaker of the future? And will 1980 also see the commercial arrival of a home digital reproducing system based on the Philips Compact Disc, or the Sony, or...? Whatever happens at the consumer end, the studio boys will be ready for it, using all the professional gear that the APRS Exhibition unwraps for us annually. ■

The Express Limiter is the latest signal processing device manufactured by Audio & Design (Recording).

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REELS AND BOXES 5" and 7" large and small hubs; heavy duty white boxes. **W-M Sales, 1118 Dula Circle, Duncanville, Texas 75116. (214) 296-2773.**

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db People/Places/Happenings

• **Adam Yokoi, Scott Minakami** and **Jeffery Berkowitz** have all advanced to vice presidential positions at the **Panasonic Company,** Secaucus, NJ. In addition to his duties as vice president, Mr. Yokoi will retain the office of general manager, product planning and engineering division, with responsibilities for new product planning and development, product safety, government relations, quality assurance and technical training. Mr. Minakami, who will also retain his position as general manager, industrial sales division, will be formulating and implementing the merchandising plans for the company's industrial products. Mr. Berkowitz, in addition to his vice presidential duties, will continue in his position as general manager, Technics and home audio division, handling the formation and implementation of merchandising plans for the company's Technics line of high fidelity components and Panasonic home entertainment system.

• Responsible for new product development, handling unusual technical situations, liason with the European divisions and preparation of technical data, **David Rubenstein** has been appointed technical manager– audio/video tape for **Agfa-Gevaert, Inc.** A graduate of **Hofstra University,** Mr. Rubenstein has an extensive background in the audio and video fields with broadcasting and production organizations.

• **Polydax Speaker Corporation** of New York, a subsidiary of **Audax,** France, has named **Evan Struhl** to the position of chief engineer. Mr. Struhl was previously with **Miero-Acoustics Corporation.**

• **Kent R. Duncan,** founder and president of **Kendun Recorders** in Burbank, has acquired the operation of **Artisan Sound Recorders, Inc.,** Hollywood, CA. A pioneer in the field of independent disc mastering, Artisan Sound Recorders (previously owned and operated by **Robert MacLeod**) will undergo renovation of its two existing cutting rooms, by the **Sierra Audio Corporation of California.** In addition, a state-of-the-art mixdown room will be incorporated into the complex which will open October 1, 1979.

• **ESS Inc.** has announced the acquisition of **Dynaco.** ESS is known for its Heil air-motion transformer loudspeaker systems, while Dynaco garnered early recognition in the hi fi field through its component kits. Reorganization of Dynaco's marketing staff and manufacturing operation are currently underway, with ESS technicians and designers involved in refining and improving the better components in the existing Dynaco product line. In addition, Dynaco will be offering three new speaker systems and a state-of-the-art line of electronics separates.

• **Dan Katz** has been named producer/chief engineer of **The Audio Group (TAG),** a Los Angeles commerical production recording studio. Mr. Katz was formerly associate media coordinator for the **Jewish Federation-Council of Greater Los Angeles.**

• Appointed to the newly-created position of director of communications for **James B. Lansing Sound, Inc.,** Northridge, CA, **Curtis Pickelle** will direct the company's marketing services operations including supervision of national advertising and in-house production of dealer coop materials, sales literature and point-of-purchase aids. In addition, Mr. Pickelle will assume the responsibility for JBL's monthly rep and dealer newsletters, as well as the company newspaper, "The Troubadour." Prior to joining JBL, Mr. Pickelle served as the director of the marketing communications dept. at **Altec Lansing.**

• A fully operational prototype of the first broadcast quality speech compression module has been used to produce several commercials in the laboratories of **Integrated Sound Systems, Inc.,** Long Island City, New York. The module allows recorded sound to be played faster or slower than the rate at which it was originally recorded without loss of clarity or tone. Using variable speech control in broadcasting, a 30 second commercial can be shortened electronically to 20 seconds without producing the high-pitched "chipmunk" effect that normally occurs when a recording is played faster. Production models are expected to be available by the end of 1979.

• **Ampro Broadcasting,** Feasterville, PA, has purchased **Scully Recording Instruments,** formerly a division of **Dictaphone Corporation. Tom Creighton,** previously national sales manager at Scully has been named vice president of sales and marketing. **Bill Hamilton,** formerly Eastern regional manager, has been promoted to the position of national sales manager, and **Ed Zdobinski** has been appointed customer service manager for Scully products. Ampro/Scully has added facilities for the manufacturing and warehousing of Scully recorders, as well as housing executive offices, at 826 Newton Yardley Road, Newton, PA 18940. (215) 968-9000.

• In his new position as director of marketing at **McMartin Industries,** Omaha, Nebraska, **Charles E. Goodrich** will be planning marketing strategies, supervising new product introductions, and overseeing the advertising and corporate communications areas. Formerly director of engineering, Mr. Goodrich joined McMartin Industries in 1967. In addition, **Don Denver** has been named to the position of national sales administrator, a post which involves liason between the Eastern and Western sales managers, the district managers, and McMartin customers. Prior to joining McMartin Industries, Mr. Denver held an administrative and engineering post at **KECK** in Lincoln, Nebraska.

• **Gary Rilling** has been appointed to the position of national sales manager for both the Professional and Musical Sound product lines of **Altec Lansing,** Anaheim, CA. Mr. Rilling will be primarily responsible for the marketing of Altec Lansing's Industrial Professional and Musical Sound Products; while managing a network of Altec-employed district managers throughout the United States. Previously, Mr. Rilling served Altec Lansing as district manager for the Mid-Atlantic region from New York to Washington, D.C.

• The **Rauland-Borg Corporation,** Chicago, Illinois, has announced the appointment of **George Owen, Jr.** to the post of vice president of engineering. Mr. Owen's entire professional career of some twenty-nine years has been with **Motorola,** where he served most recently as product manager.

INTERNATIONAL ENTERTAINMENT EXPOSITION

DECEMBER 11-14, 1979

LAS VEGAS CONVENTION CENTER

Seminars for Professional Sound and Lighting Personnel

The Sound System:
Tuesday, 9:45-11:15 A.M.
Bob Heil, Bob Heil Sound
Glen Meyer, Electro-Voice
Tom Walter, Community Light and Sound
Cerwin Vega (to be announced)
Michael Pettersen, Shure Brothers
Don Pearson, Ultrasound
Ray Kimber, RKB Industrial Inc.
Others to be announced

Lighting, Engineering the Perfect System:
Wednesday, 9:45-11:15 A.M.
Jim Moody, Sundance Lighting
Bill McManus, McManus EnterPrises
Bob See, See Factor
Chip Monck, Moderator
Rich Bay, P&B Lighting
Ted Van Bemmel, Vanco Stage Light
T.J. McHose, FM Productions

The Lighting Worlds—Cross Breeding:
Tuesday, 4:30-6:00 P.M.
Imero Fiorentino, IFA Associates
Lee Watson, Lighting Dimensions Editor
Marty Aronstein
Jim Moody, Sundance Lighting
Tom Folsom, Crews Folsom Assoc.
Brian Edwards, Wavelength

The Special Event—THE KISS SHOW:
Wednesday, 4:30-6:00 P.M.
Bill McManus and Associates

Sound Reinforcement—State of the Art:
Wednesday, 11:30-1:00 P.M.
Steve Neal, FM Productions
Jack Maxum, Showco
Stan Miller, Stanall Sound
Chip Monck, Moderator
Northwest Sound (to be announced)

The Rock Tour:
Thursday, 9:45-11:15 A.M.
Robin McGruder, Showco
Larry Hitchcock, FM Productions
Elliot Krowe, See Factor
Chip Monck, Moderator
Northwest Sound (to be announced)

Lighting Equipment Marketing:
Tuesday, 11:30-1:00 P.M.
Bob Schiller, Strand Century
Joe Bates, Litelab
Dr. Joel Rubin, Kleigl Brothers
Tom Pincu, Berkey Colortran
Marge Roman, Olesen
Larry Cada, Diversitronics
Bob Benson, Skirpan

Running a Successful Stage Equipment/Supply House:
Thursday, 9:45-11:15 A.M.
Jack Ransom, Metro Lites
Grand Stage Lighting (to be announced)
Barbara Brennan, Cinema Services
Jan Musson, Musson Lighting
Marge Roman, Olesen
Lee Watson, Lighting Dimensions, Editor-Moderator

Laser Technology:
Thursday, 11:30-1:00 P.M.
Ivan Dryer, Laser Images
Bart Johnson, Laser Displays
Carl Schulthess, Spectra Physics
Dick Sandhaus, Science Faction
Brian Castelle, Bur. of Radiological Health

Safety Standards:
Thursday, 4:30-6:00 P.M.
Dr. R.W. Davidson, Alexander and Alexander
Others to be announced

Running a Successful Sound Installation and Service Operation:
Thursday, 11:30-1:00 P.M.
James Elliot, Audio Unlimited
Charlie Moore, Dimension Five Sound
Larry Jaffe, DBX-Moderator
Barry Brownell, Brownell Sound
Claire Ford, Ford Audio

Performers Speak Out—Leading Performers Discuss the Live Show:
Friday, 11:30-1:00 P.M.
Chip Monck—Moderator
Speakers to be Announced

Plus more to be announced.

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