RECORDING ENGINEER/PRODUCER

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PRODUCING AUDIO FOR • TAPE • RECORDS • FILM • LIVE PERFORMANCE • VIDEO & BROADCAST

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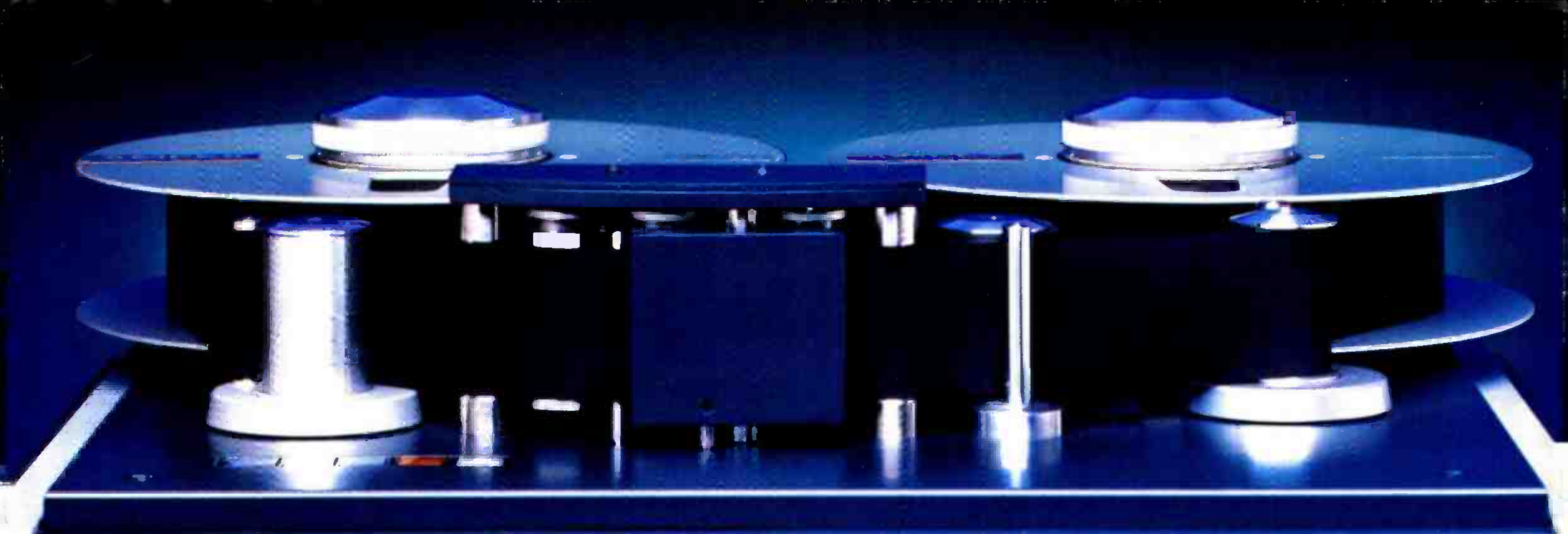

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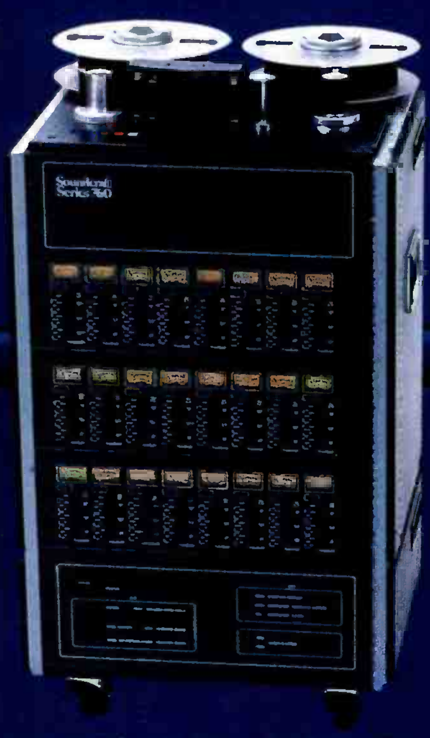

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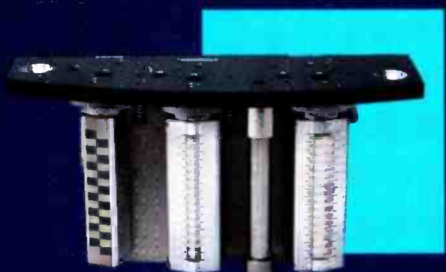

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- **Film**
- **Live Performance**
- **Video and Broadcast**

— the magazine produced to relate recording **ART** . . . to recording **SCIENCE** . . . to recording **EQUIPMENT**.

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Volume 16 — Number 1

— Contents —

— the Cover —
This month's cover shows the new Audio Sweetening/Mix to Picture post-production facility at Century III Teleproductions, Boston. An article describing the room's design and construction begins on page 150.


VANNELLI ON AMEK
"Getting the sound you hear in your head takes time, patience and a little sweat. Having the *right* console at your fingertips makes life a little easier."
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THE YEAR OF 24-TRACK ANALOG

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1985:
THE YEAR OF 32-TRACK DIGITAL

News Letters Views

CONCERT SOUND TRAINING

from: **John Allen**
Lakewood, CA

Having just read David Scheirman's article on Modular Sound Reinforcement, based in Trenton, NJ, [October 1984 issue], I'm impressed with this company's diversification of system designs and industry application.

I live in Los Angeles, which is a far cry from New Jersey. The reason I state this is my interest in the desire I have in finding an apprentice program offered locally, similar to that of Modular's engineering program.

I hope that *R-e/p* can suggest a similar apprentice/working program here in the L.A. basin. I am currently pursuing my B.S. degree in broadcast engineering, and would like to merge quality audio with video coverage.

Many thanks for a wonderful publication.

David Scheirman replies:

You're right, John: Los Angeles is a long ways from New Jersey. As far as I know, Modular's apprentice program is unique. However, you are fortunate to live in a geographical region that is literally crawling with individuals who are active in the sound reinforcement field.

Take the time to attend a few live-performance events in your area, and attempt to discover what individual(s) and firm(s) are handling the live audio. There may not be an apprentice program available in your area *per se*, but the odds are that an aggressive search on your part for a company that is seeking ambitious, hardworking trainees just may result in a part-time job offer.

I invite responses from Los Angeles area readers who may know of a work-study program for individuals interested in the type of career which John Allen has targeted. □□□

REGARDING AUTOMATION

from: **Doug Dickey, vice-president**
Solid State Logic Ltd.
Oxford, England

I enjoyed your December 1984 double-feature on "Recent Advances in Servo-Controlled Console Automation Systems." Mr. Degher and Mr. Martin write impressively, and the improvements they describe are no doubt welcome news to many in the industry. Along with our congratulations on their achievements, we would like to offer some clarifications.

Both articles refer to "valid complaints with most VCA/tape-based automation," or "problems with most VCA-based systems." Because Solid State Logic is the only manufacturer of a VCA mixing system to be mentioned by name in either article, your readers may gain the mistaken impression that the SSL system suffers from the negative characteristics which both authors attribute to "most" systems using voltage control technology. We owe it to our clients and your readers to correct any such misunderstanding.

Concerning the reference to VCA "problems" including "signal degradation, noise, drift, distortion and grouping inaccuracy," we agree that certain VCAs in poorly designed circuits exhibit one or more of these problems to an objectionable degree. In fact, we acknowledge that the legacy of some earlier VCA-based systems has sometimes made it difficult for the SSL to get a fair hearing. However, when a skeptical engineer, artist or producer *does* take the time to audition music through an SSL and to become familiar with the system, they seldom come away unimpressed.

The myth that VCA automation inherently yields results that are either "unnatural" or "unmusical" can be laid to rest by simply listening to any of the exemplary recordings produced by some of the industry's most respected engineers and producers using SSL systems exclusively. The typical *R-e/p* reader who listens seriously to recorded music will probably find many of these recordings in their personal collections.

The old stories die hard, and after so many years it gets a little tiresome reading the "valid complaints" about VCAs that could just as honestly be made about many other components in a poorly designed signal path, including transformers, op-amps, capacitors and even meters! The fact is that gifted professionals will *always* differ in their opinion of what equipment combinations best achieve their concept of the "ultimate" sound. This is just one of those things which makes life interesting and serves to define recording as an art.

On to some of the more specific statements in the two articles. For the record, the SSL system does not exhibit any cumulative update error. As Mr. Degher points out, this is a problem for systems using the audio master for data storage. The SSL Studio Computer is a disk-based system, and maintains frame accuracy throughout unlimited updates and manipulations of the mix data.

I would also like to address Mr. Martin's theory that "In VCA-based systems there are really two faders in each channel" (one "real" and one "imaginary"). This analogy manages to simultaneously oversimplify and complicate the issue, which isn't easy to do. Nor is it particularly helpful to anyone trying to understand the real operational differences between the two approaches.

The concept of an "imaginary" fader is completely foreign to the operating philosophy of an SSL. Stated simply, the SSL system employs the fader to control a VCA, offering two basic modes of operation. When the channel is in "Absolute" mode, the computer records the exact levels and changes made by the mixer. When the channel is in "Trim" mode, the engineer uses the fader to trim the previously stored levels.

When a channel is in "Trim," the computer compares the instantaneous values of its stored mix against the physical movements of the fader, and automatically generates an "update" mix which is identical to the stored mix until a fader is moved. When the operator moves a fader, the system adds those adjustments to the stored mix.

In practice, the procedure is both natural and highly useful. Unlike some other systems, there are no read, write or update switches in the SSL system, and there is no need to use dedicated LEDs, meters or video bargraphs to match the "real" fader with the one my friend Morgan Martin imagines.

There *is* a single "status" button on each SSL fader panel, which serves multiple functions defined by the system's software. For example, it may be used to request special functions such as track joining during mix edits, and to summon Total Recall displays for controls other than faders and cuts. It also allows the engineer to request special statuses, such as "Preview Absolute," which enables the mixer to listen to a level change on the monitors prior to a cue, and then to effect a precise level jump in the mix data exactly on that cue, eliminating the needing to rewind the tape after identifying the desired relationships.

Solid State Logic has a policy of continually investigating all technologies which might be helpful to our clients. We have researched the features of servo-motors, and have decided that the benefits offered by voltage-control technology outweigh these. Others are free to disagree with our conclusions. It is clearly beneficial for the industry to have a choice — but the real issues can be presented without inventing and then attacking phantoms.

We have always trusted the professional community to make its own judgements based on the overall merits of each approach. If SSL may be allowed its own brief commercial message, I would say only that we have had

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STING

Tony Mitchell talks to

Tracking Sting down to his lunchtime retreat within the maze of Shepperton Studios, where he's currently filming 'The Bride', proved to be less difficult than I'd feared.

I simply followed the long trail of glitter which led to the control room of the recording studio where he's doubling as producer for a new band called 'A Bigger Splash'.

My initial belief that Sting had succumbed to an early seventies fashion kick was dispensed only when I learnt that he'd just been filming a glitter party scene for the movie – a romantic version of the Frankenstein story said to be more in keeping with Mary Shelly's original story than the Karloffian video nasties we're all familiar with.

And there he was, lounging against the control room wall clutching a Fender bass and looking only slightly 18th century in ruffle-necked shirt, brocade waistcoat and riding breeches. His assistant Danny was despatched to make some tea.

Sting bought his Synclavier just six months ago, and like a good novel he's hardly been able to put it down since. His enthusiasm for it is impressive – a combination of reverence for its technical achievement and childlike amazement at the creative possibilities he's still discovering. Ask him to sum up its potential impact on music making and he'll come up with a modest comment such as:

"It's as radical and important an invention as the piano was centuries ago." And so, on to the first question.

MITCHELL: "What was it that first made you think this might be the machine for you?"

STING: "It was a sort of dream of mine, when I first started to actually write music down on a stave – you know it's impossible to read after half an hour – that it would be great if everything you played on a keyboard immediately transmitted into notation. I was sure one day someone would invent it. And one day I was looking at some roadie's magazine in America – and there it was! The Synclavier did it.

I was totally over the moon and it was only then that I got to find out about all the rest of the functions of this amazing machine. And it's great fun, it really is.

Now its kind of taken over my life. It takes a lot of technical application which I think is fun, because it's about learning something totally new, but there's no way you can use it the day you get it."

> *"It's as radical and important an invention as the piano was centuries ago"*

MITCHELL: Between hurriedly gulped mouthfuls of salad – he went on to explain that he'd become very disillusioned with synthesisers and synthesiser bands "because they all sound exactly the same," and because he was inspired to write music by the sound an instrument makes, he was always trying new instruments, and that the synthesiser element of the Synclavier had "this wonderful range of warm, organic, rich sound which makes me want to play with it."

And with the computer, he says, you have a system which allows you "to compose beyond the limitations of your physical skill – in fact beyond the limitations of anyone's physical skill."

STING: "Another thing is that I've never worked with an orchestra before it would be a very expensive experiment for anyone to hire an orchestra for the day to see what happened. But with the Synclavier I have an orchestra at my fingertips."

MITCHELL: If that sounds a mite indulgent, then don't think Sting isn't aware of it. He knows devices like the Synclavier are often branded as rich men's toys but the integral facilities and the constant updating process initiated by the Synclavier's designers convinced him that it would be a very sound investment.

STING: "It's almost the responsibility of those with enough bucks to invest in this kind of thing. It's like, the only people who could afford orchestras in the days of Mozart and Beethoven were the crown princes of Europe. And us rockstars 'ave taken over from that. I see myself as a kind of Medici of the Arts in the 1980's – know what I mean?" (ha-ha)

> *"One interesting feature of the Synclavier is that it translates tempo to frame time"*

MITCHELL: "Has your experience with the Synclavier turned you on to computing generally?"

STING: "No – I'm not really into home economics!"

MITCHELL: "You don't feel the need to have a machine that'll address a lot of envelopes for you?"

STING: "No, I've got Danny to do that!"

MITCHELL: "Can you use the computer for anything else?"

STING: "Well there's a floppy disc floating around somewhere, so you could do your accounts on it, in between scoring something."

MITCHELL: Stifling an inclination to say what a good idea it was to have a musical instrument that can tell you how much you've got left in the bank after you've paid for it, I moved on instead to raise with Sting one of the criticisms which is sometimes voiced against the Synclavier – its restriction, on the digital sampling side, to monophonic sampling.

STING: "Yes, monophonic sampling. That might be a temporary disadvantage but polyphonic sampling is only a short time away. In the meantime, if you want, er, a chord of milk bottles breaking or something, you can do it with a tape machine."

MITCHELL: "One application of the Synclavier that's bound to appeal to a man with tandem careers as an actor and musician is in the creation of film scores. Had that opportunity presented itself yet?"

STING: "Well I've been asked to do the music for this film. And one interesting feature of the Synclavier is that it translates tempo to frame time. You could have written a piece of music that lasted 30 seconds to fit a scene exactly, then the director says he's gonna cut a bit or add a bit to it, and you're stumped. What do you do? You either cut a bit off the music or re-record it. But with the Synclavier you just punch the relevant keys and the music is translated through frame time into the right length, either shortening minutely each note or lengthening it. And that is... outrageous!

I also like the idea that I can play something on the keyboard, record it on the memory recorder then translate it to screen editor so it comes up as computer language, and then you can perfect it. Using the integral recorder is so quick, you can try out things with different voices so quickly. And once polyphonic sampling comes in, you won't need a studio at all, you'll just need a Synclavier. You'll be able to make a record without using tape."

MITCHELL: Not surprisingly, Sting has no qualms at all about using the device on stage with the Police – he thinks it will be great fun. But I wondered if he'd embarked on a sampling programme, perhaps walking around Hampstead or jogging to the studio each morning, to equip himself with new and unique sounds for that purpose.

STING: "I haven't had that much time, to be honest. I'm quite interested in things I haven't got around the house, like timpani, cymbals or a snare drum. You can just hire them for the day, mess around with them and you've got the full range of what they can do at your fingertips."

MITCHELL: "Do you have the Synclavier in a music room at home?"

STING: "No, I have it in my bedroom. As I crawl out of bed in the morning, I turn it on, I plonk away, and if I hit a good chord, I carry on, and if I don't . . . I have breakfast.

Actually we haven't talked about the resynthesis angle, which is quite new. It basically records a sound and it comes out as a spectral display, a wave form. You can increase the intensity of it and copy it. You can do as many as 54 sections of that wave, so resynthesis is actually very, very close . . . and as a learning device, it's a wonderful way of finding out how sound is constructed. I haven't written any music lately, I've just been doing spectral displays!"

MITCHELL: "But getting back to your disillusionment with synthesiser music because it 'all sounds the same', don't you think there's evidence that exactly the same thing is happening even with these sophisticated sampling devices? Isn't everyone using them to make the same kind of records at the moment?"

"You can compose beyond the limitations of your physical skill - in fact beyond anyone's physical skill"

STING: "That's really where you have to bring back the human element. When the electric guitar was invented you had the same sort of thing – Oh God, everybody's going to sound the same. So you wait for the Bert Weedons to come along and show the way. You can't replace human beings. It's just gonna be different.

At the moment anybody can do it. Everybody thinks they can make a David Bowie record. It's time something new happened and that's why someone is going to have to take it somewhere else."

MITCHELL: "That's got a lot to do with current record industry attitudes – they 'sign up some haircuts and get a producer in to do the rest' syndrome."

STING: "Listen, I was signed up on the strength of my haircut. I mean, let's call a spade a spade."

MITCHELL: "I know, but you had other qualities as well."

STING: "Tight trousers."

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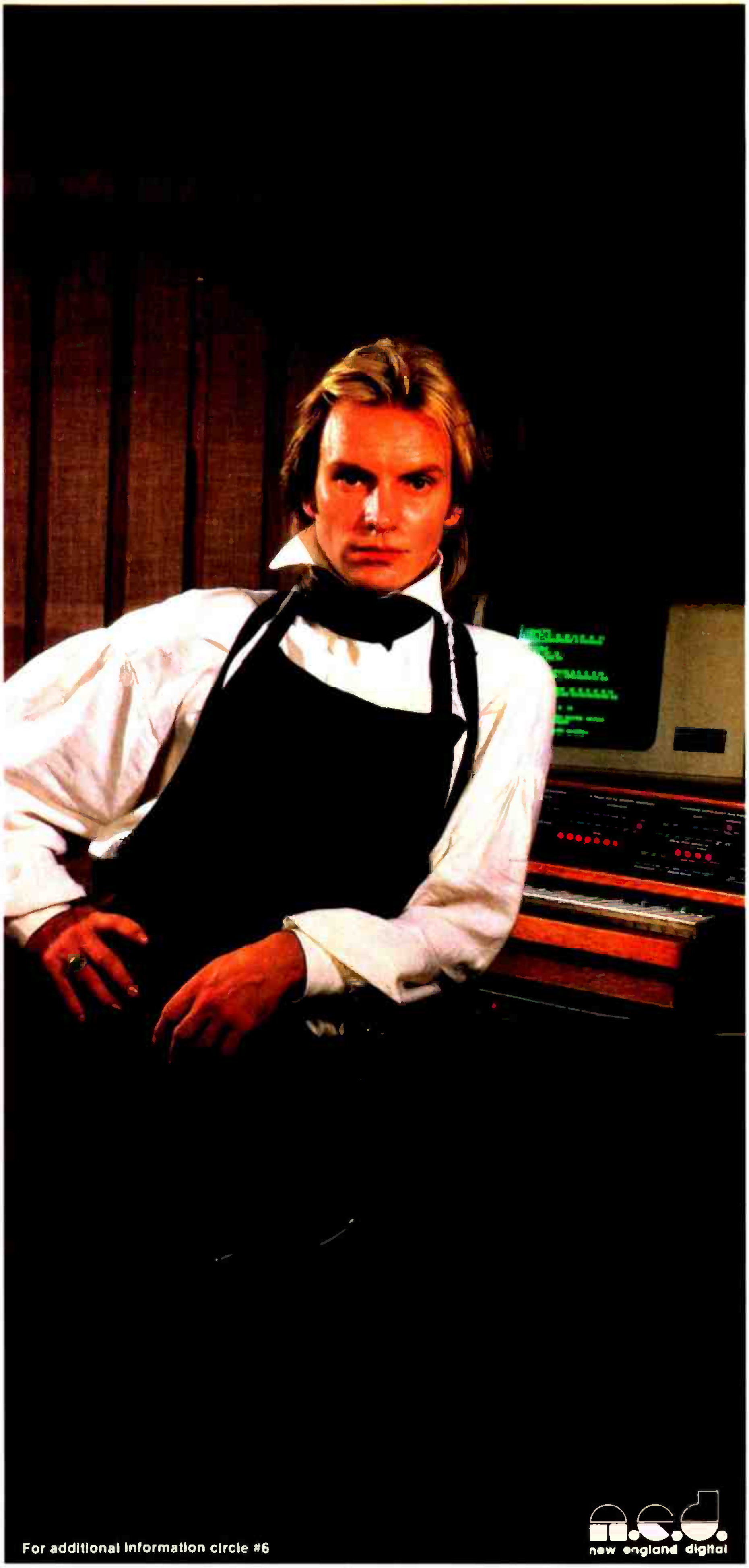

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— continued from page 8 . . .

no problem in placing SSL systems in hundreds of facilities around the world on the basis of merit.

One final point: Mr. Degher's article states that the Massenburg system can "read data recorded on the NECAM or Solid State Logic automation systems." This statement requires some elaboration and some caution.

As I understand it, GML has accessd the portion of the SSL disk where we store fader and mute data, and has developed a scheme for "translating" that data into a form which their system can use. I am told that this feature was developed at one of their client's request.

While touting the cause of compatibility between mix systems of various manufacture, it is unfortunate that GML have chosed to proceed on this matter without so much as a word to SSL, much less any technical discussions between their programmers and ours. It is instructive to note that GML's announced data transfer process is a one-way system, and that data generated by their system remains incompatible with the SSL and other systems currently in use.

George Massenburg assures me that his process works, and I take him at his word. However, SSL is not prepared to recommend a transfer process that we know nothing about, and we cannot speculate as to the value of raw SSL data operating in any system other than that for which it was intended. □□□

Morgan Martin, regional manager, Rupert Neve, Incorporated, replies:

I am pleased that Doug enjoyed my article on NECAM 96. I, in turn, found his letter to *R-e/p* interesting, and wish to respond to those comments which relate to my article.

When NECAM was developed nine years ago, our intent was to avoid VCAs and their attendant problems. True, VCAs have come a long way since then — our own VCA circuits will attest to this — but VCAs do not match the sonic performance of the NECAM conductive-plastic track.

As to my analogy concerning Real and Imaginary faders, I stand by it. To explain a bit, first consider the "Absolute" and "Trim" modes referred to by Doug. Operation in these modes is identical to the "Normal" and "Relative" modes introduced on NECAM faders at their inception nine years ago. They are both useful modes, as SSL must have realized when they designed their system later on. However, the NECAM moving-fader system is significantly more user friendly in several ways that Doug didn't mention.

Now to explain Real and Imaginary faders, let's look at Relative. On NECAM when you make a relative change, you do it by moving the fader to produce the offset desired. Subsequent moves by the fader under NECAM control reflect this offset. And since there is only one fader, the Real one, you always know how much room you've got to the top or bottom of the scale. Neat and simple.

In VCA non-moving fader systems, the situation is different. If you want to make a relative change, you move your Real fader to produce a change in voltage (not a change in the audio level directly). This voltage is combined with that coming from the computer's Imaginary fader to finally tell the VCA exactly what the level is to be. Note that you don't see this actual level on the Real fader scale — all you know is how much you have moved your fader by. Not so neat, I'd say.

Now, to be fair, there are any number of approaches that manufacturers of VCA non-moving fader systems have devised to take care of this and other problems inherent in VCA non-moving fader systems. But all of them fall short of the simplicity and power of the approach taken with the NECAM moving-fader automation.

By the way, for those who prefer centralized group master *without VCAs*, remember that this option is possible with NECAM 96.

And if I might also be allowed a commercial message, I would like to point out that Neve has literally thousands of consoles in facilities worldwide, with hundreds of NECAM systems in use.

On to other matters. I would like to make a couple of comments regarding my article on NECAM 96. The new Neve flat-belt fader was said to travel from

. . . continued on page 16 —

Making Life Easier

To give the artist and engineer complete freedom to explore these new potentials, SSL invented Total Recall™. At the end of each session, Total Recall scans every knob and button on all Input/Output modules. Then, in less time than most people take to find a pen that works, it creates a permanent and portable record of these settings on floppy disk. Which means that you can stroll into any SSL Total Recall control room anywhere in the world and recreate last week's monitor and cue mix, or last year's incredibly complicated but not quite final version.

Control accuracy is within a quarter of a dB! Best of all, Solid State Logic has accomplished this without affecting the audio path. Providing a dynamic range and bandwidth that comfortably exceed the performance of the best 16 bit digital converters and recorders.

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Total Recall is just one aspect of the SL 4000 E Series Master Studio System, an integrated range of hardware and software components designed to make even the most elaborate productions more humanly manageable. Practical innovations such as the SSL Studio Computer provide the world's most versatile mixing automation. The SSL Integral Synchroniser and Master Transport Selector offer computer-assisted control of up to five audio or video transports in perfect lock.

Other system elements include events control, programmable equalisation, and a variety of mainframe and metering options to suit many different requirements and budgets.

Whatever your initial specification, all SSL systems are designed so that economical upgrades can be performed on site as your business grows and diversifies. This policy is supported by continuous software development that enables SSL studios to keep pace with an increasingly inventive clientele.

We can build an SL 4000 E Series Master Studio System for your control room in about three months. We'll be happy to assist with your technical and financial planning. We'll provide expert help with installation and training. And we'll back you up with prompt parts support and worldwide field service.

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top to bottom in about 500 milliseconds, at its fastest; actually, the figure is about *150* milliseconds. Also, the Editor's knife cut one important point. Mixing with NECAM 96 can be simple as 1, 2, 3. #1: Put up a tape; #2: Put in a disk; #3: Push the Start key. You're off! Mixing with NECAM 96 is totally transparent. You don't have to use labels, cue and the like. You just mix.

Second, regarding the excellent article on the GML system, I have three comments. First, NECAM allows use of fader and mute groups without timecode or disks or, for that matter, tape. Unlike GML, with NECAM you can have any number of groups, not just eight. Second, I would caution NECAM I and II users to take care in attempting to use their disks on the GML system. GML developed their conversion program without the knowledge and help of Neve. I have discussed this matter after the fact with GML, and must report that the conversion produces poor results. Third, NECAM 96 will operate both in the Slave and Master modes. In Master mode, the deck may be controlled by NECAM, or any other control (tape remote, the deck's own controls, etc.) at any time. In Slave, NECAM doesn't attempt to control the deck. Thus we provide either mode to the operator.

Also, it is with pleasure that I have read the interview [in the same issue] with Bob Clearmountain. One small correction I would suggest involves a comment Bob made about it being impossible to trim levels on the NECAM as you can with a VCA automation system. As I noted above, all NECAM faders, past and present, allow just the kinds of "relative ride" that Bob spoke of. All you do it flip the fader switch to "R" for Relative, and make the relative adjustment as you like.

Thanks for this opportunity to comment. As always, I'm happy to discuss this fun subject with anyone who cares to call.

George Massenburg, GML, Inc., replies:

Allow me to address Doug Dickey first as a designer. Then I have a few thoughts as either a recording engineer or an artist, depending on one's aesthetics, or sales, or both.

I agree with Doug; it is tiresome if not boring to hear of more complaints about VCAs. Rather than add to the endless VCA dialog, GML shall continue to respond to specific user needs and requests. It appears that SSL is of this same mind, as they have been looking at servo-motors for some time now. I leave the reader to speculate on exactly how they will be put to use.

GML has an open posture on its many developments. We have demonstrated our system to a great many people, including Mr. Dickey himself. Our systems accurately and regularly convert SSL data, assuming that one enters the SSL group assignments; SSL has neglected to provide for recalling this in their software. It is important to note that our various conversion software tools were written solely in response to customer demand. SSL conversion was intended as a tool rather than a provocation.

After much research on various SSL consoles and many mixes, we have found that SSL stores fader data in 8-bit format, dB-linear (we store 10-bit dB-linear). We convert this faithfully and would be more than happy to show anyone our research data.

When we are requested to write software to convert from GML format to SSL format, we will do so expeditiously.

Now, as an engineer.

I can only add that I have made recordings on a great many consoles in the 21 odd years I have been chasing down the chimera. I honestly have no ultimate conclusions as to what electronics guarantee a great recording, and I don't trust that anyone else does, either. GML or SSL automation has made the time I spend in the control room far more productive. Just this summer I did the Philip Bailey/Phil Collins record on SSL and GML consoles; I moved SSL data to a GML console to mix two tunes. It works. Record's doing fine, thank you very much.

I think that it would be helpful to expand this concept — to move between consoles and facilities in the course of doing a record.

I can only hope that I can interest various parties to respond to my recent request for participation in an industry-wide automation seminar to be held at the Anaheim AES Convention in May. I have recently the assurances that SSL shall themselves participate. ■■■

Editor's note: The following letter arrived at *R-e/p* after the previous correspondence had been received. Its author is not connected in any way with either Neve or GML.

SERVO-FADER AUTOMATION

from: **Michael J. Frenke**
Independent recording engineer
Los Angeles, CA

The December 1984 article on the GML Automation System, by Denis Degher was, in my opinion, a bit too brief compared to the article on Neve NECAM 96. It was knowledgeable reading, no doubt, yet short due to space considerations. Morgan Martin's article on the NECAM 96 system was also very knowledgeable, and extremely descriptive; from reading the article I actually feel that I "know" the NECAM 96 System, even though I've never used it. In an attempt at balancing the amount of information provided in both pieces, I feel justified in wanting to make (non-technical) creative types, like myself, aware of the advantages of the GML Automation System.

As an engineer, I have experienced all types of systems (VCA-based, tape-based, moving fader, etc.) courtesy of MCI, SSL and GML; no slouches by any standards. But the amazing speed and accuracy of the GML system [installed at Conway Recorders, Hollywood] absolutely floored me. Considering the fact that I have really had no prior "computer" experience besides console automation systems, I got around that old Motorola like I really knew what I was doing. The fellows in R&D at GML did an amazing job of interfacing man with machine.

The system's main configuration is a "read mode" — pure genius from a simplicity point of view. The console (and individual channels) can be shuttled back and forth between several configurations, which allows for easy updates, punches, groups, etc. The speed of the "punch" is really mind blowing. All information is kept on-board the console, and constantly updated; no need to keep 50 mixes, because information data is kept in memory until you decide to actually go to hard disk.

The system is also idiot-proof — there is no way to wipe someone else's information because each session/client/song is clued from your personal name and password.

I've never felt so strongly about a product to write to a publication or manufacturer, but for the engineer who has yet to experience the GML Automation System, "you're not going to believe it." My special thanx to Buddy Brundo at Conway for the experience, and GML for the "expertise." ■■■

Photographic Credits and Corrections

On the opening page to "Sound for the Olympics," published in the December 1984 issue, we omitted the credit for the color photograph by Greg Morton; also, photography for the GML article was by Kathy Cotter. More seriously, we inadvertently identified the right-hand photograph on the opening page to "Nashville in the Eighties," as being Treasure Isle Studio. In fact, the photograph shows the interior of **Music Mill** — *Editor.*

QUANTEC ROOM SIMULATOR

from: **Bob Hodas**
Sausalito, CA

I feel that I must make an addition to my Quantec Room Simulator review, published in the December 1984 issue of *R-e/p*. Marshall Electronic is no longer the QRS distributor in the U.S. [Distribution is now being handled by Europa Technologies, Venice, CA. Telephone (213) 392-4985 — *Editor.*] As such, Marshall's upgrades and band selection of several analog and digital components for precision, matching, and sonic purity will no longer be performed as described in paragraph 3 of my article. The QC-3 Software Control Program for the Apple II will still be available direct from Marshall; versions for the Apple Macintosh and IBM PC are currently being developed. ■■■

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For additional information circle #11

Some thoughts on building a better piano.

"We let machines do what machines do best, and people do what they do best. It's as simple as that. And it's made all the difference in the world."

Joe Kennedy, Quality Control Manager

Joe Kennedy's story is the Baldwin piano story. Joe joined us in 1940 at our Cincinnati plant. Today he's personally responsible for every Baldwin that comes off the line at our newest facility in Trumann, Arkansas.

And just as Joe's role has evolved, so has the way we do things at Baldwin. Nothing represents that more than our state-of-the-art plant in Trumann. This is where we feel we've finally achieved that elusive balance between man and machine.

Under aggressive new ownership, Baldwin has transformed the quality of its products from what has traditionally been termed "very good," to distinctly superior. Not only has the product improved in tonal quality, but it has also been refined and redesigned in terms of cabinet appearance, interior cosmetics, workmanship, raw materials and engineering.

In other words, *there's been a complete rededication to the Baldwin heritage of quality*, and a commitment to an even greater future.

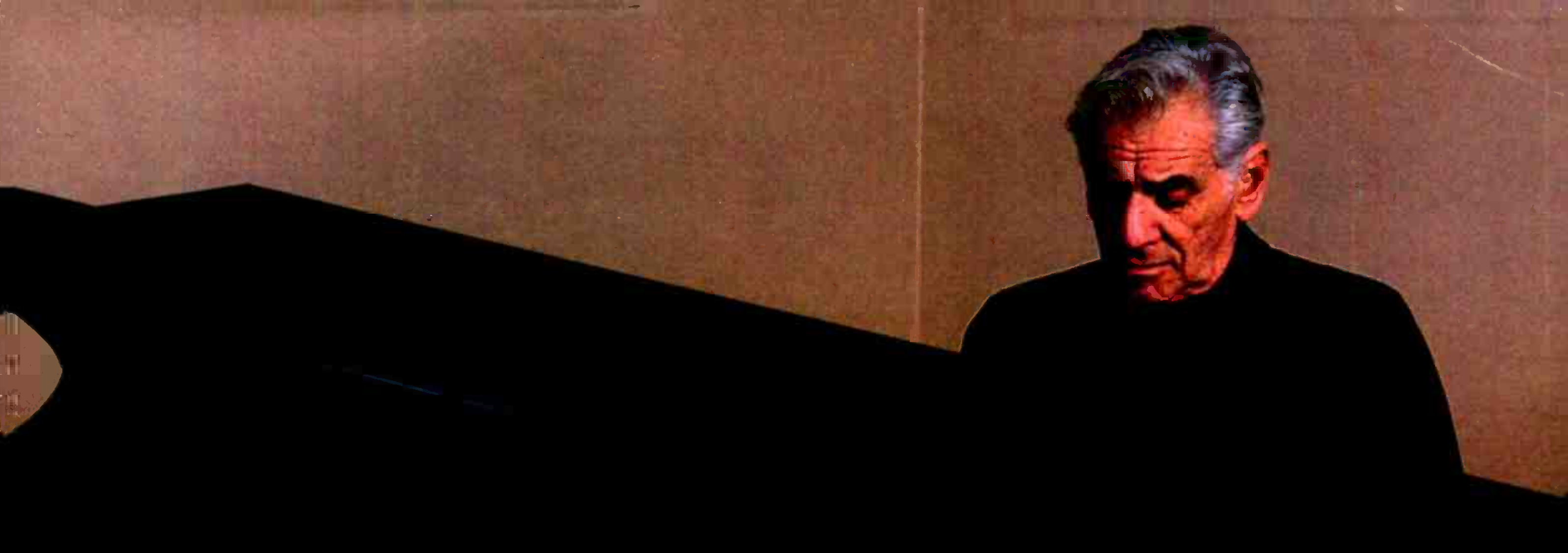
Representative of this new spirit is the major investment we've made in automated technology. This automation gives us the tooling that allows precision duplication of parts, so that each critical relationship is aligned, measured and set exactly to ensure quality that is unequaled in the industry.

The immediate benefit of this investment is the cost savings we're able to pass on to our customers. But there's another advantage automation gives us. It allows our people to do what people do best. To regulate key action or hand-finish fine wood cabinets, while the machines do what machines do best.

With nine factories, more than 1.4 million square feet of plant space, and over 500 dealers, Baldwin is America's single largest piano manufacturer.

But after all is said and done with technology and statistics, the Baldwin story is one of people. People like Joe Kennedy who have devoted their careers to making what D.H. Baldwin dreamed of 123 years ago: "The best pianos at the best price."

So Joe Kennedy is right. Machines and people working together *are* making all the difference in the world.



"I learned 'Petrushka' on a Baldwin, and composed 'West Side Story' on a Baldwin. I have a Baldwin in my studio and play one just about everywhere I go. I guess you could say Baldwin is my piano."

Leonard Bernstein, Composer, Conductor, and Pianist

The choice of an instrument by a distinguished pianist, composer or conductor is a very personal decision. It's a choice based on professional experience, and as such represents the highest tribute an artist can pay to an instrument. This is not a commercial endorsement, it's a reflection of individual taste, because no performer in the world can afford to entrust his or her career to an instrument that doesn't meet their highest personal standards.

That's why Baldwin takes so much pride in being *the expressed and continued preference of such a celebrated group of renowned musicians.* It is the single greatest testimonial that can be made in support of the principles that guide all of us at Baldwin to build the very finest pianos possible.

What is especially gratifying is that the entire spectrum of the international music community is represented by the classical, jazz and popular pianists, conductors, composers, educators, vocalists and orchestras who request Baldwin for performance, practice and pleasure.

It's a remarkable list that's as rich in variety as it is in talent. To name only a few: *Aaron Copland, Jorge Bolet, Dave Brubeck, Billy Joel, Zubin Mehta, Marian McPartland, André Previn, Liberace, George Shearing, Ronnie Milsap, Burt Bacharach* and *John Williams.*

But our story doesn't end on the stage. Because these artists are continually involved in the critical evaluations of our designs — being asked to judge tone quality, touch, repetition, volume of sound and other crucial factors. This regular judging by artists makes a substantial contribution to the improvement of every Baldwin made, whether it be for the Metropolitan Opera or Mister Rogers' Neighborhood.

"After a week of Pink Floyd in here, only the Baldwin is still in tune."

Rick Hart, Engineer, Producers One, Hollywood

In a recording studio only one thing is asked of an instrument. It's sound. Irrespective of brand names, that's the bottom line. But it has to be reliable sound.

If you have to tune an instrument during a session, you're losing time. And in the recording business, that means money.

That's why Baldwin has the reputation it does with both engineers and session players. Baldwin hangs in there long after the others have folded and gone home.

What makes Baldwin so stable and tunable day after day, year after year? One reason is the 41-ply pinblock in our grands. It's the pinblock that's ultimately responsible for tuning stability. So the more laminations, the more stable the piano. By comparison, a popular competitive grand piano has a 5-ply pinblock. So just how long will *our* pinblock last? We honestly can't tell you since, to our knowledge, we've *never* had a 41-ply pinblock fail.

More reasons for our growing studio reputation are our patented strings and treble termination bars which give Baldwins such rich bass notes and clear highs. We use only the finest maple hardwood for the inner rim so the sound is reflected back onto the soundboard for better volume and sustain qualities.

Not surprisingly, those are also some of the same reasons Baldwins sometimes cost a little more. But when you get down to it, what do you really want in your studio? A delicate instrument that has to be pampered, or one that can't wait for Pink Floyd to come back next week?

"If over the years I've learned one thing about pianos, it's that the original purchase price rarely bears any relationship to the final cost."

Tommie Pardue, Music Education Consultant, Memphis City Schools

As funding for music education becomes an increasingly critical factor in school budgets, lower-priced pianos naturally become more attractive. But as is true of so many other major long-term investments, the original purchase price can be somewhat deceptive. What must be considered is the *long-term value.*

The reason Baldwins may initially cost more than other pianos is because better quality materials go into them. And in the long run they're a wiser investment because of their lower maintenance costs and longer playing life.

That's why so many teachers feel confident in recommending Baldwin to students and parents alike. They know that the tone, touch and long-term value of a Baldwin make it the piano without equal.

The reason Baldwin is committed to the music educators of America is because we believe the future of music is in their hands. Together with a national advisory board, we've created a *goal-oriented* syllabus to assist teachers. We sponsor the Baldwin Music Education Centers, and teachers seminars. We also co-sponsor with the Music Teachers National Association the Baldwin Junior Keyboard Achievement Awards. And look for the Baldwin-sponsored specials on the Arts & Entertainment Network this year.

They're our way of completing the circle that begins at the Baldwin plants. Which in turn has made the Baldwin Hamilton piano *the most popular school piano in America for the past 40 years.* And as we strive to make an even better piano, we're confident that we'll continue to be the preferred piano for the next forty as well.

"When we first introduced the SD-10, a critic wrote, 'If Beethoven had had a piano like this, the course of music would have been radically altered.' It's that kind of talk that keeps me going back to the drawing board."

Harold Conklin, Chief Piano R&D Engineer, Baldwin

For all the professional recognition that Harold's received over the years, fortunately, none of it's gone to his head. He's still hard at work trying to design the perfect piano.

Because at Baldwin we believe that perfect piano tone is an ideal we share with everyone who designs, builds, plays and services pianos. That's why tone has always been the preeminent focus of our research efforts.

An example of this philosophy is how we improved the traditional string. What we came up with was a unique way to synchronize the string's longitudinal and flexural modes. The result is the SynchroTone® string. And it's just one of the many exclusive design breakthroughs developed for our grands that's now found in every piano we build.

Another area of intense research and development is our *electronic keyboard program.* While others are dropping out of this particular market, Baldwin is actually increasing its commitment. An example of our stake in the field is the fact that Baldwin is one of the few American keyboard manufacturers with its own computer-aided design and manufacturing (CAD/CAM) capabilities.

It is in this challenging environment of invention and experimentation that people like Harold Conklin thrive. They have only one objective: *to create the very finest instruments possible.*

Because at Baldwin, good isn't good enough anymore. It *must* be the best. After all, that's what drawing boards are for.

"I don't know what they're doing at Baldwin, but we've never seen instruments of finer quality from anyone."

Bill Washburn, Washburn Musicland, Phoenix

If you've read this far, you probably know a little about what we're doing at Baldwin.

We're building instruments without equal.

And we're offering our dealers and their customers *America's most complete single-brand keyboard line.* Everything from portable electronic keyboards, home organs, and classical church organs to spinets, consoles, full-size uprights, grands and concert grands.

All with one name. Baldwin. All with one purpose: to be the best instruments with the best value. Period.

Because we want everyone, no matter what their budget, to have the kind of quality instruments that Baldwin makes. And we're doing everything we can to make that happen.

Our comprehensive customer financing program is the only one of its kind. It allows our customers to realize the dream of having a Baldwin, whether it's for their home, their school, or their studio.

Baldwin also has an unparalleled service system. We include a service policy with our warranty on every new piano. It pays a qualified technician $40 toward the cost of inspecting, adjusting and tuning your piano. And we have our own fleet of trucks which speeds deliveries and service.

It all adds up to what we're doing for you at Baldwin. We're building better pianos and organs.

This is the Baldwin story. Who we are, what we're doing, and where we're going. We believe that with the right tools and the right people, you can do anything. Including making the finest instruments possible. If you have any thoughts or comments you'd like to share with us, or would like to find out more about what Baldwin can do for you, please write to either of us personally: Dick Harrison, Chairman, or Harold Smith, President, Baldwin Piano & Organ Company, 1801 Gilbert Ave., Cincinnati, OH 45202

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EXPOSING AUDIO MYTHOLOGY

Laying to Rest Some of the Pro-Audio Industry's More Obvious "Old Wives' Tales"

by John H. Roberts

Since this column marks two full years of "Audio Mythology," I thought it appropriate to look back and tie up a few loose ends. In Part 2 we will delve into yet another digital topic. Perhaps we should rename this column "Digital Mythology"? (Just kidding, guys.) Now onto the re-cap.

• *April 1983*: It seems like yesterday, and I still recall my trepidation at committing my views to print. It's difficult to pretend you didn't say something when there are thousands of copies of *R-e/p* around to haunt you. Fortunately, the only thing I have to change about my first column is my bio. I resigned from Phoenix Audio over a year ago, and now am only president of one company, Phoenix Systems. I can still be reached at (203) 643-4484, but encourage open letters to the column so we can share the conversation. I don't have much to add on the subject of Absolute Polarity, other than to note that I did not get a single response from anyone claiming to hear it. Therefore I stick to my original suggestion that you watch polarity for archival purposes, but don't worry too much about your car speaker systems.

• In *June '83* we described the basic limitations of PCM digital, and mentioned Companded Delta-Modulation in passing. Since then, CDM has been making waves in Satellite links, and high-quality two-track recording.

It is not a coincidence that two major players in CDM — dbx and Dolby — are both firmly rooted in the companding noise-reduction market. While, for a given bit rate, Delta Modulation generally will have high-frequency performance superior to PCM, the latter contains more raw information. As information translates to dynamic range, DM is a natural for companding.

The dbx Model 700, which has been well described here (See: reference #1) and elsewhere, beats 16-bit PCM's dynamic range of 96 dB by a healthy margin. The ruggedness of the digital medium for storage or transmission makes tracking errors much less of a problem than with typical companded systems. Hybrid digital/analog techniques can even improve the transient response. Delta Modulation may not be a knock-out over PCM, but it is definitely a horse race.

• In *August '83* we talked about connector polarity, distortion boxes and, in passing, the Patent System. I am pleased to note steady progress towards universal acceptance of pin #2 hot. If I could make a suggestion, let's stop calling pin #3 hot the "American" standard, because it certainly isn't any longer.

On the subject of patents, I repeat my call to take full advantage of the system. By law, the inventor must disclose the "preferred embodiment" (best approach). By sending $1.00 and the magic number to the Patent and Trademark Office in Washington, DC 20231, you get valuable information. And, while you cannot directly copy the technique (read the Claims carefully), often you will gain insight and knowledge about a given problem. Rather than re-inventing the wheel, you can stand on the shoulders of those who worked on a problem before you.

• *October '83*: On the subject of speaker wire, I would like to expand upon my original conclusion. While I still feel the dominant mechanism is wire resistance interacting with a speaker's non-flat impedance, there is another mechanism in multiway systems using passive crossovers. The wire resistance can throw off the crossover point(s), resulting in various sonic consequences; another good reason to keep those wire runs as short and low-Z as possible.

• In *December '83*, against good advice, I wrote of VU meters and things dB. Although I wasn't struck by lightning, I don't know that I accomplished universal enlightenment. At last count there were about a half-dozen variations on definitions for dBm, most linking zero dB to a voltage reference of 0.775 volts. Since none of these are strictly correct, I don't care which one you use. But please try to mention at least *once* within a given article or spec sheet exactly what you are defining *your* dB to mean. When specifying characteristics like maximum input or outputs, parameters that are typically voltage-limited, why not use good old volts? The dB variant of your choice could be use in parallel for convenience without any ambiguity.

Regarding the rather acrid sniping in the Letters' Column over the subject of dB, I would like to make one observation: System designers often begin and end with acoustic power. Any technique of describing intermediate equipment gains that result in correct answers is of obvious merit. However, equipment designers are not dealing with a black box connected to known input and output impedances. To the circuit designer, the black box is the entire universe, and its output is a simple function of the input voltage. I've seen the decibel used to describe such voltage transfer functions (gains) in papers going way back. There is no historical high ground regarding usage.

I believe that the circuit designer should accommodate the system designer, and provide whatever input/output specification makes the job easier. Likewise, the system designer should not deny the circuit designer use of the decibel for dealing with the voltage gains inside those black boxes. Let's not lose sight of the real issue: accurate transfer of information. Use internally whatever system works for you — just take care that everyone reading a spec sheet or article can figure out what you mean.

• In *February '84* we discussed how capacitors can be non-ideal, and why they make so many different kinds.

• In *April '84* we compiled a listing of how manufacturers were wiring their XLRs. I notice that at least one of the pin #3 hot companies has gone under; hopefully the new startups will take the opportunity to do it right (pin #2 hot!) Any companies that have changed from #3 to #2 hot since publication of the survey are welcome to drop me a line. I will either mention it in the column, or we can run it as a "Letter to the Editor."

In April we also looked at how negative feedback is instrumental in getting accurate performance out of audio circuits.

• In *June '84* we discussed amplifier slew rate, and how it alone was not adequate to characterize how well an amplifier would perform.

• In *August '84* I got back into talking about digital, and described "Gibb's" phenomenon. While I noted that the prering of some digital or time-correct filters was actually the ideal response, it is worth noting that the perfect-looking waveforms from CD players when playing back squarewave test disks are still not accounting for the time delay in the A/D anti-aliasing filters. Thus perfect playback of a test disk should show some additional lead.

• In *October '84* I noted that digital isn't forever, but it is for a long time. I also talked about what DC servo amps do and do not do.

• *December '84* brought a discussion, but, alas, not an answer to the age-old question: "Is it better to turn it off or leave it on?" We also looked at impedance and interfacing.

• In *February '85* we recapped the preceding two years' columns, and then started talking about . . .

Hiding Behind the LSB

It is common when analysing the performance of a digital system to stop looking beyond the Least Significant Bit, assuming that the LSB is the theoretical noise floor and distortion limit. This is true enough for low-bit systems, but, as the quantization floor of a high-bit system drops into the input signal's

. . . continued overleaf —

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For additional information circle #13

noise, a phenomenon known as "dithering" can occur (See: reference #2). High-frequency noise riding on top of a low-level signal causes the LSB to toggle rapidly. Rather than turning low-level signals into a squarewave at the quantization limit, the LSB is duty-cycle modulated. The output anti-image filter will integrate and recover the original signal at levels well below the LSB (See reference #2 for more information about the requirements for effective dithering.)

Properly dithered, the LSB is no longer a limit to a digital system's accuracy or linearity — the limitation now is the tracking between bits. PCM's ability to compress so much information into so few bits (16 bits = 65,536 quantums) is accomplished by weighing each bit twice as heavily as its neighbor. By doubling across a 16-bit word you get a MSB (Most Significant Bit) that is 32,768 times greater than the LSB. If that ratio is off by one part in 32,000, you may as well be using a 15-bit system.

A/D convertors will not work properly if these ratios do not track better than 1 LSB. D/A convertors have no such restraint, however, and it is not uncommon to see 14-bit D/As with 12-bit linearity, or 12-bit D/As with 14-bit linearity. Unlike dynamic range, which will be a simple function of the number of bits, linearity — and thus distortion — has no such theoretical linkage. The D/A's non-linearity will add to that of the A/D (less than 1 LSB) to determine total system non-linearity.

A circuit trick available to closed-loop designs involves the use of a very fast D/A for both output and input circuits. (A/Ds have a D/A inside.) A first-order cancellation of non-linearity will occur. Unfortunately, this technique is pushing the speed limit of present circuitry, and any digital code so generated will only benefit from playback on that one system.

What Does It All Mean?

This is a tough one to characterize. The distortion mechanism is almost exactly opposite typical analog non-linearity, being insignificant for large signals and peaking near the quantization floor for a properly dithered signal. While I expect a severe non-linearity to be audible, I'm not even sure what to listen for. Non-linearity in analog circuitry is nice and predictable; non-linearity in a D/A will occur at one or more bit transitions, and be expressed whenever that bit switches. Note: Measuring sinewave THD at -20 dB or -30 dB will *not* tell the whole story, since such a signal will not exercise every bit transition. A test more like the good old SMPTE IM distortion — but with, say, 1 kHz instead of the 7 kHz riding on top of the larger 60 Hz waveform — would be more informative since it could involve all major bit transitions.

In conclusion, a given bit rate does not predict or limit signal purity. A system can be more or less (usually less) linear than its quantization limit. While I expect consumer digital gear to be more variable, it is worth considering that all 16-bit systems are not created equal.

Reading for extra credit.

1. "The dbx Model 700 Digital Audio Processor; Design Parameters and Systems Implementation," by Robert W. Adams; *R-e/p* October 1982, page 150.
2. J. Vanderkooy and S.P. Lipshitz, "Resolution Below the Least Significant Bit in Digital Systems with Dither," *J. Audio Eng. Soc.*, Vol. 32; pp. 106-113, March 1984.

INDUSTRY FORUM

Professional Audio Discussion Topics

SUGGESTED AGENDA FOR A FORUM ON AUDIO POST-PRODUCTION MACHINE CONTROL STANDARDS

from Steve Krampf, VP Sales and Marketing, Otari Corporation

Editor's Note: From time to time there emerges from within the pro-audio industry certain topics of discussion that have far-reaching implications for the recording studio, audio-for-video, film post production, and related areas. One such topic is the development of a "standardized" interface for connecting audio and videotape transports to automated consoles and editing systems, to enable centralized control of all audio post production and editing functions, plus standard remote and locate capabilities. While several organizations are developing interface and connection protocol to achieve such machine control — including the Society of Motion Picture and Television Engineers (SMPTE), the European Broadcasting Union (EBU), and synchronizer manufacturers — many studio engineers are still unclear what features these interfaces will provide, and how they can be implemented. To address the problem of ensuring that any proposed standard will satisfy the current *and* future needs of our industry, Otari has prepared the following proposal for a Forum that, hopefully, can answer the multitude of questions relating to machine-control standardization.

The theme of Otari's involvement in this forum, and indeed a major focus in terms of product development, is to promote cost-effective interfaces which will lead to an *expansion* of the audio post-production marketplace. Our industry can indeed champion the cause of audio for all levels of picture production, not just features and expensive commercials.

If our industry can forge cost effective, supportable interfaces, then more production people could justify the investment in Audio Post Production, i.e., audio for film and video. Our feeling is that APP is still in its infancy. Currently, the interface of all necessary equipment requires a considerable amount of engineering talent. Because of this, only a small percentage of potential facilities can financially justify, or have access to, the required engineering expertise. Even if a production company can make meaningful income projections, they find it difficult to project a meaningful capital and expense budget under present circumstances.

Those facilities with in-house design engineering will always have the edge. But we believe, as noted before, the APP market is much deeper than we have all been experiencing. There are other levels of customers — such as those in the expansive industrial market — that have, in general, not included serious audio post production in their systems or productions.

Plug-compatible interfaces (known as PCM in the computer industry) can be cost effective; this level of standardization can provide the boost and incentive necessary for the remaining facilities to invest.

Proposed Questions

SMPTE/EBU Serial Interface Standard:

- Does the "evolving SMPTE/EBU serial interface standard" meet the present and future needs of the APP industry in terms of:

  1 — expense per node;
  2 — speed of data transfer;
  3 — other considerations the industry thinks are important.

- Other SMPTE/EBU interface questions:

  a) What is the status of the SMPTE/EBU protocol, in your opinion?
  b) At what machine-command level will you be compatible?

  1. Addressing the *transport* (basic) functions of the machine e.g., FF, Play, etc. as a serial remote.

  or

  2. Addressing the "intelligent"

. . . *continued overleaf* —

functions of the machine. For example: *Go To* a timecode address; or *Chase* the master.

c) Do you believe that there should be a certification process to allow a manufacturer to claim a certain level of "SMPTE/EBU Compatibility?"

d) Do you expect an audio machine to either *self-resolve*, or *emulate a video machine*?

e) Are you advocating a star or buss transmission system?

f) Do you incorporate a time line i.e., time-of-day clock? Why yes or Why no!

g) Is timecode information distributed? If so, is it raw SMPTE/EBU LTC, SMPTE/EBU VITC, or timecode data?

h) What is your sub-frame resolution? e.g., 1/80th, 1/100th?

i) How do you handle multitrack channel addressing? Can additional insert edits (punch-ins) be programmed or manually entered after a given number are originally assigned?

RS-232 Questions:

- Which products utilize RS-232 interfaces? Are they supported, and is their protocol compatible with:
  1. Adams-Smith
  2. Audio Kinetics
  3. BTX
  4. CMX
  5. Any others
- Do these RS-232 protocols serve a valuable need today? In the future?

Suggestions for the Form of Answers:

- Basic Architecture: Provide suggested block diagrams for the interface of each manufacturer's (machine, controller inc. console mfg's., synchronizing) products in the different type of systems that they envision selling to.
- List of different interfaces available for each product. In a given case, a manufacturer's *proprietary* interface might be acceptable, if it is more cost effective or yields enhanced performance. We feel, however, that they should also offer industry-standard interfaces. List of existing machines with existing interfaces.

Timecode Questions:

- How should the industry handle different time bases such as film and video (EBU and NTSC)?
- How do we treat DF and NDF, even if they are in the same time base? As we discussed in our last correspondence ["Letters" page, December 1984 issue], hopefully they will be.
- How should VITC timecode be integrated into systems? Presently there is no widespread editing use for VITC, because very few audio and video editor/synchronizers contain a VITC input. Is Longitudinal Timecode conversion a good idea? Is a VITC data interface a better idea? Should that interface be parallel or serial?

System Upgrade and Flexibility:

- What are the manufacturer's function life cycle, for machines, for editors, etc? Is there a different life cycle [depreciation schedule] for electronic hardware intensive, electronic software intensive, and electro-mechanical equipment?

■■■

News

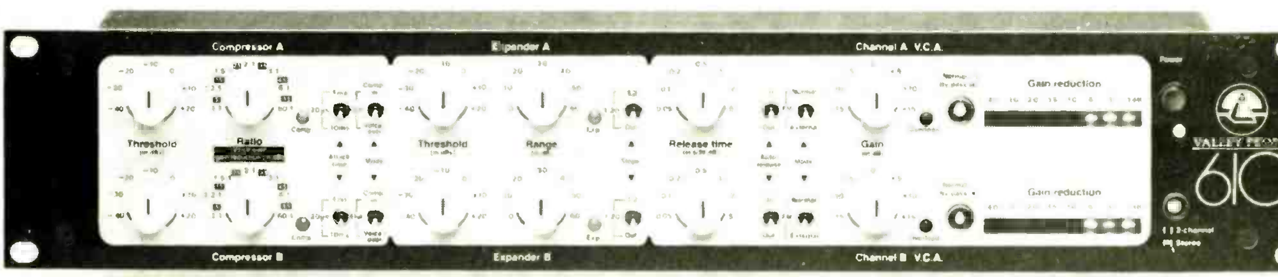
INTERLOCKED MITSUBISHI X-800 MULTITRACKS USED ON "WE ARE THE WORLD" BENEFIT SESSIONS

At press time, the final remix of the *USA for Africa* "super-session" benefit song, "We Are The World," which features a literal who's who from the recording industry, was taking place at Lion Share Studios, Los Angeles, with two Mitsubishi X-800 digital 32-tracks running in SMPTE timecode interlock. Producing the session is Quincy Jones, with engineers Humberto Gatica and John Guess, plus assistants Larry Fergusson and Khaliq Glover. We understand that the final decision regarding the mastering format — either Mitsubishi X-80 digital, or analog two-track — has yet to be resolved.

Royalties from record sales will be donated to charity; it is also stated that everyone involved on the project is donating their time and services free of charge.

Basic tracks for the song, co-written

. . News *continues on page 36* —



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INDUSTRY INVENTIVENESS IN THE EIGHTIES

Professional Music Analysis: An Evaluation Program for Engineers, Producers, and Musicians Wanting to Break into the Music Business

by James Riordan

Everyone working in the Music Business knows how hard it is to try to get your tapes heard. There are thousands of people trying to break into the industry, but very few openings. Major record labels receive better than a hundred tapes a week from artists trying to land a deal. Unfortunately, many of these people have no chance at all. A high proportion of them are so far off the mark that they're wasting everybody's time, especially their own.

The main problem is that, by and large, there is no way for outsiders to intelligently evaluate their chances at commercial success, or to learn what they must do to improve those odds. In reality, the only people qualified to provide them that information are being smothered by demo tapes that they cannot possibly find time to listen to, much less comment on. So the tapes keep pouring into A&R departments, and sometimes the potential superstar, along with the strictly amateurs, get lost in the shuffle.

This has always been the problem in just about every area of our industry. Those who wanted a career in the music business had to be prepared to "pay their dues," and "keep knocking on doors" until they got a break. But getting that break very often meant having one person who was in position of power to take the time to sincerely evaluate what you had to offer. Once a real industry pro gave you some direction on how to better your act — in terms of performance, technique, and production — things might start to become easier. Sooner or later, after enough direction and "'connections" with people who could help you, something might possibly work out.

The same scenario is essentially true for a career as a recording engineer, session musician, songwriter, or the next singing sensation; there simply aren't very many ways you can go. The chief drawback to obtaining solid career advice is that the people who knew enough to give it were far too busy doing it to takeout time to instruct you. If you set your sights on becoming a top session engineer, for example, you have to get experience anywhere you can, and then wait for a chance to work with a big name who could show you the inside tips and shortcuts necessary for career-success. No matter what field you were interested in, there seems to be no way to really find out your potential without putting in years of dues.

Recently some well established music professionals joined forces to provide one possible solution to this dilemma. The goal of the new company, known as Professional Music Analysis (or PMA), is to provide professional input for a variety of career areas. Essentially, the new service works as follows: A tape is submitted for evaluation in any one of 12 categories, and then passed to a leading professional in that category for evaluation. The pro listens to the tape and gives his input by using an evaluation format established by PMA. The applicant then receives a copy of the evaluation, plus a personalized summary written by the evaluator. A photo and biography of the evaluator is also included in the evaluation kit.

. . . continued overleaf —

Picture courtesy Atlantic Studios, N.Y.

The key to the program is the evaluation format which, according to PMA founder Boyd Hunt, took over a year to develop. (Hunt also founded what is described as the world's largest musician referral service, Professional Musicians Referral, based in Minneapolis.) The evaluation format breaks down each major career category into several subcategories, each of which is then further defined and divided into problem areas, suggestions, and tips for improvement. Each subcategory receives an individual rating, and an overall rating is also given to each submission.

An example of the procedure might be the category of "DRUMS." Drummers who submit tapes are evaluated by top drummers, such as Ed Shaunessy of the Tonight Show Band, or Andre Fischer (formerly of Rufus). Drums are broken down into several subcategories, including Technical Ability, Continuity With Bass, Tempo, Rhythm Parts, Tonal Quality, and so on. Each of these is further divided into elements such as Dexterity, Tuning and Clarity, for example under the Technical Ability subcategory. Vocalists are rated under categories such as Articulation, Pitch, Phrasing, and so on.

The available categories include Vocals, Guitar, Drum, Bass, Keyboard (and Synthesizers), Horns & Reeds, Strings, Percussion, Original Songs, Groups, Engineers, and Producers. The engineering category was said to be one of the hardest to compile, and requires an extensive information sheet completed by the applicant. Although, material within all other categories may be submitted only on cassettes, engineers, on the other hand, are encouraged to submit reel-to-reel ¼-inch, 15 ips tapes with a full set of alignment tones.

The engineering category provides input on such subcategories as Microphone Selection and Placement, Physical Room Setup, Tape Information, Track Breakdown, Signal Procesing, and more. The engineering evaluation format was created by Joel Fein, an Emmy winner and Oscar nominee who formerly served as chief engineer at The Village Recorder, Los Angeles.

Other names associated with PMA include Grammy Award-winning saxophonist Ernie Watts, former Fleetwood Mac guitarist and solo artist, Bob Welch, songwriter/-producer Dennis Lambert, bassist Tim Bogert (of the Vanilla Fudge, and Beck, Bogert & Appice), Grammy Award-winning record producer Bob Monaco, songwriter John Madera, and vocalist Sylvia Shemwell. In addition, several leading record label and publishing executives from Capitol, Motown, and many other companies are also PMA Evaluators. According to Hunt, every evaluator is a well established music industry professional.

PMA stresses that they are not in the business of discovering new talent, and they do not function as an employment agency. Because of the contact between those seeking new talent and those that possess it, there is bound to be some eventual success in these areas, but that is not the thrust of the organization, Hunt stresses.

The people behind PMA feel that they provide a much needed service for all levels of talent in the music industry. According to Hunt, the company's long-range success will be based on repeat business with client following the advice of their evaluator and resubmitting another tape for further input.

PMA provides direct access to some of the top professionals working in the industry — a major achievement in itself — but the really exciting aspect of this new company, in this writer's opinion at least, is that someone is taking a step to bring some long overdue structure to the music industry.

For years I have been hearing negative comment after negative comment about the problem of too many demo tapes being submitted to record companies and publishers. (And have seen the entire corner of a room stacked halfway to the ceiling with unauditioned tapes.) At last an organization is attempting to remedy this problem in possibly the only way that it can be solved: through the formation of a profitable enterprise providing a real service, which represents a good example of Industry Inventiveness in the Eighties. ■■■

News

— *continued from page 30 . . .*

by Lionel Richie and Michael Jackson, were recorded on X-800 digital multitrack at Lion Share on January 22, with producer/conductor Jones assisted by Tom Bahler, and Gatica at the console. Musicians on the date included keyboardists Michael Boddicker and Greg Phillinganes; synthesizer players John Barnes and Michael Melvoin; bassist Lewis Johnson; and drummer John Robinson. Also on the session were Toto's Steve Porcaro and David Paich, Paulinho daCosta, Ian Underwood, Michael Omartian, and Marcus Ryles. Guide vocals on the date were by Richie and Jackson.

A gathering of 45 recording artists who came together on January 28 at A&M Studios, Los Angeles to record vocal overdubs for "We Are The World."

Six days later, following the *American Music Awards* ceremonies, several dozen vocalists and musicians gathered at A&M Studios, Hollywood, to record lead and backing vocals. Prior to the overdub date, Gatica and Guess had bounced over several guide/cue tracks to a second X-800 multitrack, which was taken to A&M for the "super-session." Vocalists recorded on the overdubs included Harry Belafonte, Dan Aykroyd, Lindsey Buckingham, Kim Carnes, Ray Charles, Bob Dylan, Sheila E, Daryl Hall and John Oates, James Ingram, Jackie Jackson, LaToya Jackson, Marlon, Michael, Randy and Tito Jackson, Al Jarreau, Billy Joel, Quincy Jones, Cyndi Lauper, Huey Lewis and the News, Kenny Loggins, Bette Midler, Willie Nelson, Jeffrey Osborne, Steve Perry, The Pointer Sisters, Lionel Richie, Smokey Robinson, Kenny Rogers, Diana Ross, Paul Simon, Bruce Springsteen, Tina Turner, Dionne Warwick, and Stevie Wonder.

Returning to Lion Share with the digital multitrack tapes, Jones, Gatica, and Guess have begun the process of remixing the various tapes. Gatica says that the 25 overdubbed vocal tracks were combined and recorded on 16 open tracks on the master. During the track-bouncing stages, the engineer also points out that various drum tracks also were spun in from an interlocked analog

. . . **MORE NEWS** *on page 196* —

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There has been a growing trend in recent years for engineers to have aspirations towards co-production and producer status. But, as many have found to their cost, numerous additional skills are required on a session, beyond the technical aspects. One engineer-producer that has met with great success since making that upward move is **Joe Chiccarelli.** Having begun his career in Boston, he relocated to the West Coast in the late Seventies. Formative early sessions with Frank Zappa (*Sheik Yerbouti* and *Joe's Garage*), Poco (*Legend* and *Inanammorata*), Juice Newton (*Juice* and *Dirty Looks*), and Oingo Boingo (three soundtrack albums) lead to recent assignments with Glenn Frey, Romeo Void, Van Stevenson, The Bangles, and Stan Ridgeway. Throughout his transition from engineer to producer, Joe Chiccarelli has maintained a close contact with the "essence" of producing records: to put the artist at ease, consequently establishing a cohesive identity on a track. *R-e/p* caught up with the engineer/producer at Sound Castle Studios, Los Angeles, during a break in sessions with Robert Tepper.

Joe Chiccarelli

Interviewed by Mel Lambert

Photography at Sound Castle Studios, Los Angeles, by Kathy Cotter

R-e/p (Mel Lambert): How did the L.A. recording scene in the early Seventies compare with the world you were used to in Boston?

Joe Chiccarelli: Very different. Having engineered my own sessions in Boston for a lot of years, I kind of became a big fish in a small pond. Having moved to L.A., all of a sudden you're getting coffee for people! Which can be bit of a blow to your ego!

My first session at Cherokee was a demo for an unsigned band whose name I can't remember, but the drummer was Craig Krumpf. He was an incredible studio drummer from L.A.; I've since used him on a lot of sessions. And the singer was Steve Perry, now with Journey! I kind of went: "If this is the caliber of muscianship out here, this is the *major* leagues." Not to say there isn't any great music in Boston, but out here it was unbelievable.

R-e/p (Mel Lambert): Apart from a couple of major studios, Boston is usually considered more of a demo city, because of the music colleges, and the large number of musicians living there.

Joe Chiccarelli: Exactly. Boston's a learning ground. I was back there some months ago, and there's a lot of small record labels producing New Music. There's a lot more studios nowadays.

It's not being overshadowed by New York so much these days. When I worked there, it was very frustrating because anybody who got a record deal would move to New York to produce their record. I was pretty lucky, though, because I got to work with some really good people back there. I did some work with John Payne, who was a jazz artist and played with Van Morrison. I did an album with James Montgomery, and a lot of demos with Tom Scholz, from the band Boston. There was a lot of talent back there.

R-e/p (Mel Lambert): How did the sessions with Frank Zappa come about?

Joe Chiccarelli: Working at Cherokee, I was pretty much the low man on the totem pole, in terms of all the other second engineers there. Frank booked in, and he has something of a reputation of maybe being a little bit "demanding." So they booked me on the session, figuring that since I was easy-going, which I am, I would be able to cope with the session. I can't remember the exact circumstances, but Frank had just come back from a tour in Europe. His engineer, Davey Moore, was tied up in Europe, and wouldn't be here on time. As a result, Frank initially booked in just to listen to some live tapes. When he got into the studio he was pretty pleased with them, and immediately wanted to start overdubbing.

So here I was: I had been at Cherokee for maybe six months, had done a few little engineering dates, and here was Zappa really anxious to work on his

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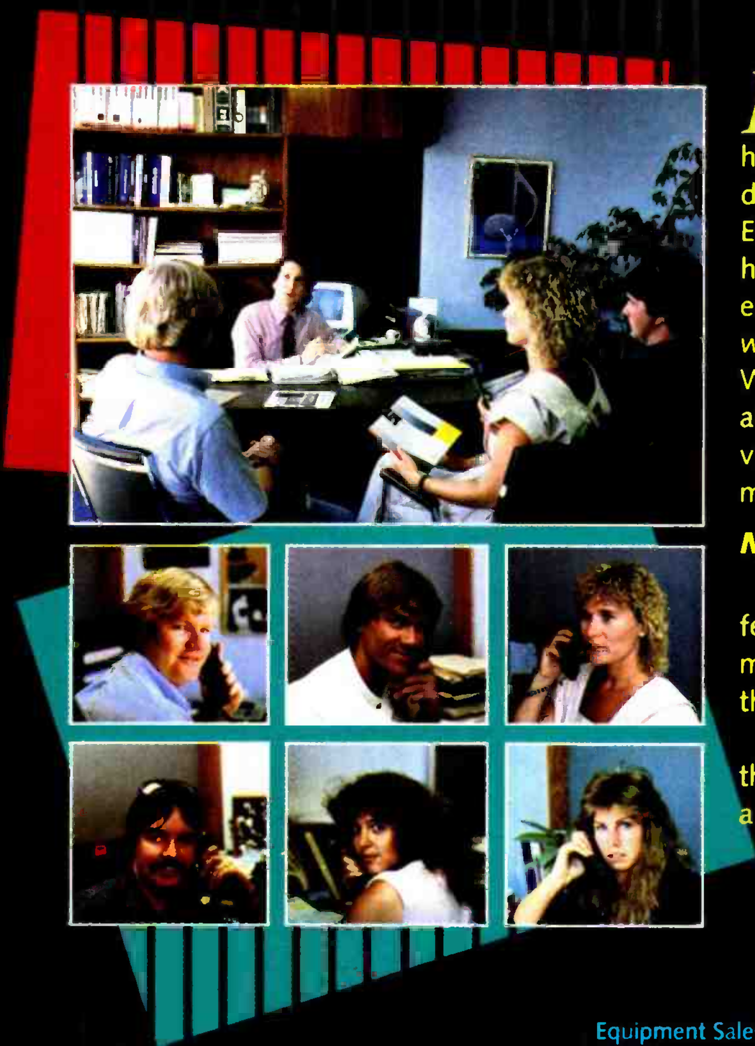

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"I'm sure that I have a 'sound' that is probably continuous through a lot of records that I've made. But I really hope that I'm 'invisible' in terms of the final sound of the record, and that it really sounds like the *artist*."

material. I was, of course, *very* nervous. Anyway, we started doing some guitar overdubs and some vocals, and it all seemed to work out really well. We got along great.

R-e/p: Zappa is pretty technical, isn't he? He seems capable of finding his way around the equipment, and now owns a Sony PCM-3324 digital multitrack. Do you think that, because you had engineered full sessions in Boston — even though you were now seconding at Cherokee — you were able to get into the mood more quickly?

JC: Probably the fact that at that time I wasn't a very strong force in the studio — I didn't have a lot of preconceived ideas as to how I thought Zappa should sound, or how I thought his record should be produced — made it very easy for me to fit into the situation with him. He could direct me in the way he wanted the record to sound. Frank always has a certain sound in mind, and really wants to hear that. However, he does give you a lot of freedom. All artists have their own "sound," and you really cannot take that out of them. You can enhance or strengthen their weaknesses, and play up the things that make them, say, Frank Zappa, or whoever it might be, but artists do have certain things that they look for in sound.

R-e/p: Given that you had your technical chops down, I presume that there were some tricks you could pull on the tracks, without masking his characteristic sound.

JC: Absolutely. I must have done maybe five, six albums with Frank, and he gave me an awful lot of space. He's got a reputation, like I said, of being very, very tough but, with me, he was great. There were a couple of times he would have a specific sound in mind — maybe to use this mike, gate this track, or just compress the heck out of it — but overall he would give me a lot of room.

And the great thing is that he would never really want things to sound "traditional;" he would *always* want them to sound just a little bit off the wall. He'd always say, "Make it sound stupid," and things like that. He was experimenting with sounds and textures.

At the same time you had to give him a lot of freedom, because a lot of time during those sessions — talk about flying by the seat of your pants — he would come up with some really outrageous idea. Everybody would look at each other, going "This is *never* going to work; this is really out there!" But, as with any other artist, you've got to give them their freedom and just go with it, because that's where the genius comes from. Nine of 10 times Frank would all of a sudden take a tune that was just real straight sounding, turn it inside out, and produce a masterpiece.

In that respect, it was real interesting to work with him. For exmple, when he'd start punching-in on the kick drum track I was getting very nervous — you've got to have pretty quick fingers, and a quick sense of tempo. There was one tune in particular that just had a straight-ahead beat to it, with kick and snare. At one point he wanted to punch right into the kick-drum track, and overdub this huge 40-inch parade drum as the one and three. Then to put some other sound as the two and four, and turn the whole mood of the tune inside out for whatever portion of the song it was.

R-e/p: Many people are interested in making the transition from engineer, to co-producer or producer. When you are engineering, how did you relate, on a creative level, to the producer?

JC: The great thing about being an engineer was that I got to work with a lot of different producers, and learned a lot of things from every one of them. Each one has their strengths and their weaknesses, and you try to fill in the spots. Maybe one guy isn't sure at picking a tempo, or which take to go with —you reinforce those things. Other producers may not be sure what they want in terms of the sound; they know when it fits the artist and when it doesn't, but are unsure what specific kind of guitar sound, for example, they want to go for.

R-e/p: So you might offer them ideas about different textures and sound colors?

JC: Yes. You really have to listen to the song, and have some sensitivity for the material and the artist. For a particular song, or a vibe that a producer is trying to create, I might hear a flanged guitar, or maybe the traditional Les Paul/Marshall sound — whatever it might be.

R-e/p: If we're looking at categories, many would consider that there are engineers who are non-technical but have a strong musical background, and those who have no musical background, but are strong in the technical areas. Maybe the "ideal" combination is somebody that's competent in both areas.

JC: I think I was kind of lucky in that respect, because I played bass and guitar in a bunch of bands. In some ways I wish I had more formal training but, at the same time, I think that I had just enough to the point where I'm not so picky in terms of, for example, what specific notes a player may play. I'll give him the freedom to play what he *feels*, and then know when it's right or when it's wrong. Some of the producers I've worked with have had a lot of formal musical training, yet sometimes they get caught up in that, and tend to want to hear *every* note precisely the way it should be. Sometimes you can lose the emotion in music that way.

R-e/p: You are known for being aware of the delicate balance between "mechanical" production techniques, and the "live" feeling in recording. Given the wide spectrum of musical styles you have encountered — from Juice Newton to Zappa — how do you set yourself up for a particular session.

JC: I'll do some homework, maybe listen to past records. I'd certainly sit down with the artist and find out what they want to do; what they liked and didn't like about their last record; what they want to try to do with this record. Even if I'm engineering a project, I'll always try to get to a rehearsal, or see the band live. But I'll really try to just find out who the artist is, and what they want to do on the project.

Going to see a band live or in a rehearsal is real important. Like I say, even if I'm engineering on the dates, it's *very* difficult for me to just walk into the session cold. More so now, because it's *very* important to create an "aura" for the particular artist or band, and give them a particular sound signature.

R-e/p: How do you adapt your engineering approach to different musical styles? Poco, for example, strikes me as having a highly produced, polished, "tight" sound, Juice Newton has more of a lush, gentle vocal texture. Red Ryder and Van Stephenson, on the other hand, have a more rough style. Obviously, today's engineer has so many tools at his or her disposal, that the options are practically limitless. How do you keep in mind *an artist's overall "sound signature?"*

JC: In the case of Juice, both [producer] Richard Landis and Juice and Charlie Calello, her arranger, really have a great vision in their minds of what she should sound like. So the parameters are really set, and my job was to bring things out, and to know the extremes.

R-e/p: Juice Newton's recordings have a lot of air around them. I presume that you laid off the compression and limiting, plus extremes of EQ, because her voice is reasonably subtle.

JC: Yes. It's interesting working with different singers, because you find that you have to change the arrangements, the instrumentation, your EQ and how you tune a snare drum, to fit with the voice. With Rita Coolidge, who has a really deep, husky voice, you always had to be careful where, in terms of pitch, you put the snare drum, because her vocal has a real chesty quality — a lot of that 200-cycle material. Sometimes you'd get the snare drum sounding huge, and everything was wonderful, but when you put the voice there they would be in the same register. You had to move pitches and EQs around just to make space for her.

R-e/p: Do you say to the arranger, for example, that a particular instrumentation, or a pattern playing behind a vocal isn't going to work, for technical reasons?

JC: Absolutely. Usually it comes out in the process of: "Why isn't it happening? What's wrong?" In a very diplomatic way you say, "I'm having a hard time getting the guitar sound to fit in with this track because of the drum part that's being played," or maybe the key that it's in. "Maybe if we tried a Stratocaster instead of a Les Paul" — which obviously has a thinner sound, and takes up less room — "that might work." Or maybe that the part could just be emptied out a little bit. I find most people take that input all the time; my advice may not necessarily be right, but they'll certainly give it a try!

Like I said, one thing I really learned from the Zappa sessions is that you've got to give people the room to be who they are. The minute that you don't, it's not music. I don't know if I want to put down Phil Spector, because his records were brilliant, but they're *his* records. For me, there's something that is lost in them; something that's not *real* about those "Wall of Sound" records.

R-e/p: In recent years we've seen a trend towards a more "naturalistic" approach to recording. Maybe because people are hearing more live music, albums should bear a stronger resemblance to how the band sounds in live performance? Poco, for example, has been consistently popular during the five years you've been working with them, from The Legend *in '78 to* Inanammorata *in 1983. Did their production techniques change during the period you worked with them, or were they after a consistency in sound?*

JC: When I started working with them, Poco had made a *lot* of records. The first record I worked on was something like their 13th or 14th album, so they had been with several different producers, and certainly worked in dozens of studios. Because the band had gone through a lot of changes they really had a lot of knowledge of studio production, and were very sure of where their limits were, and where they could stretch to on either side of the spectrum.

The first album I did, *Legend*, represented a big step for them, in terms of a more "produced" sound, and it was really, really successful for them. On the next record, *Under the Gun*, they wanted to incorporate that more "textured" sound but, at the same time, to put a little bit more edge on it. I think they felt that maybe they had lost some of the aggressive edge they had in previous records.

R-e/p: Was that combination of texture and edge a factor you could bring out during the recording?

JC: Yes. I tended to be just a little less clean in the approach, and to leave less room in the mix. In fact, when we started mixing *Under the Gun*, I unconsciously went back to the old sound, but it wasn't right. We ended up remixing the first few things that we'd done because they were too open and lacked any punch and attack. All that tighter sort of sound that comes at you more in a wall was gone — I had gone back to a layered approach, and it just wasn't right. It wasn't what the band wanted to hear. So I tightened up the sound to give it more edge in the mix.

R-e/p: Let's move on to your co-production duties on the Robert Williams *and* Willie Phoenix *album sessions, plus your work with Oingo Boingo. How did that transition come about? Was it a natural progression; something you'd always wanted to do?*

JC: I think that maybe I'm a creative junkie [laughter]. Although the role of an engineer is not limiting as such, you want to try more things, and expand yourself. As an engineer, in some situations you don't have as much overall input in terms of the album's big picture. I wanted to be more creative, and to have a little bit more input on the album's production.

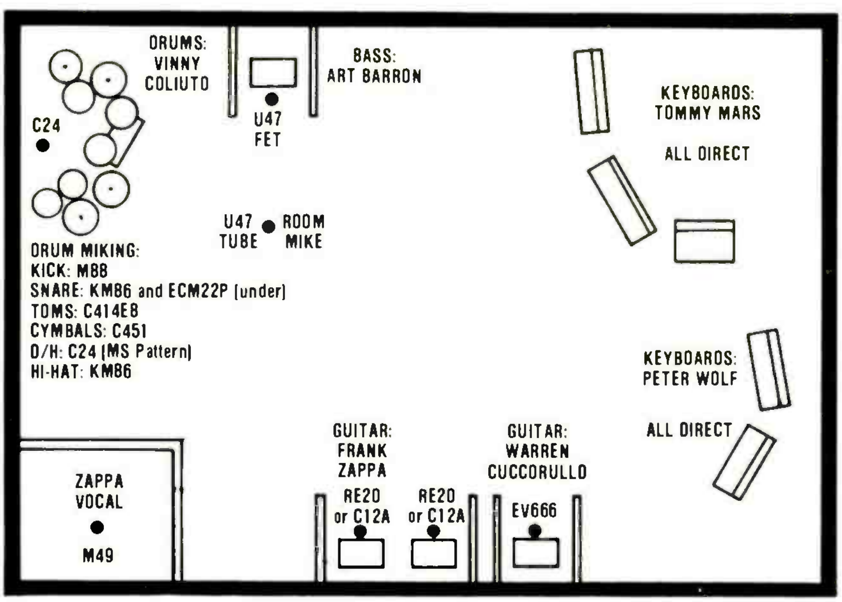
R-e/p: And also more responsibility on the session, because a lot more decisions come down to you?

JC: I enjoyed the responsibility; the pressure doesn't really bother me. I like the challenge of trying to come in under budget — which I don't always do — and, being a pretty organized person, I enjoy the organizational aspects of producing. But making the transition from engineer to producer, which I'm still in the midst of, is very difficult from a lot of angles. When you're an engineer, you've got to maintain your living by engineering for specific producers or record companies. Some producers might feel threatened that you're all of a sudden maybe going to be their competition, so that's a little tricky.

R-e/p: An attitude of "Here's a guy who's looking to take over my role," and set himself up in competition?

JC: Exactly. But I certainly would

MICROPHONE AND ROOM LAYOUT FOR FRANK ZAPPA SESSIONS AT VILLAGE RECORDER, LOS ANGELES, FEBRUARY 1980. ENGINEER: JOE CHICCARELLI

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ple, that you forget what the overall record should be like. You can get real bogged down on a smaller detail as an engineer, because it's such a precise art. But I still enjoy engineering, and it's still a challenge. I wouldn't give it up until I was definitely bored with it.

R-e/p: How do you overcome that focused viewpoint when you are both producing and engineering? Do you need to keep alternating between the two roles, to remain objective?

JC: Sometimes it feels like that, but lately I've been trying to do more and more pre-production, because in a rehearsal room I don't have to think as an engineer at all. I can get the band to a point where it feels good, and all the parts seem to be working. Then when you get in the studio you can concentrate on the sounds. You always have to work out some arrangement details, because the minute you hear the song over the monitors, things that sounded great in rehearsal might not work in the studio. There's only so much room on the VU meters, and that's it.

When I get into the studio, I spend the first few hours getting the correct sounds for the tracks, so that I can relax a bit about the engineering end of it, and sit back and go, "Yeah, that's a great take." Or, "No, this bass part needs to be changed." That sort of thing.

Control-Room Playback (*left to right*): Joe Chiccarelli, Robert Tepper, studio drummer Myron Grombacher, and bass player Tim Landers (*seated*).

R-e/p: Some of your recent sessions — ones that you're now producing — have been with some relatively unknown artists, including Gabriel Kane, Robert Tepper, and Stan Ridgeway. How did you get involved with those projects?

JC: Gabriel Kane is a new artist for the Scotti Brothers label, who I co-produced with Craig Leon. Craig has produced The Ramones, Talking Heads, and a lot of different artists. The Stan Ridgeway sessions came about, basically, through my involvement with Oingo Boingo. Stan used to be the lead singer in Wall of Voodoo, and Wall of Voodoo and Boingo both are on IRS Records. I had been a Voodoo fan, and I had produced some tracks for Boingo, so it just kind of happened.

We talked for a while about working together; in fact at one point Wall of Voodoo called me to engineer one of their records but, because of scheduling reasons, it never happened. The link had been made, however, and Stan asked me to produce his solo album.

R-e/p: You also remixed Romeo Void's Instincts, *Bangles'* All Over the Place, *and the new Glenn Frey "Sexy Girl" single. Do you prefer to be involved with a project — either as an engineer or producer — throughout the entire process, from recording basics to mixdown?*

JC: No, not necessarily, because I really like mixing. Tracking and mixing are certainly the most fun parts of making a record. It's really nice to come in at the end of a project. Not being involved with it for two months gives you more objectivity; again, you don't get bogged down on some particular melody line that you've heard for months, and are addicted to. It's easier for you to come up with something that is exciting and works musically.

And it's fun to work on somebody's else's tracks, because you certainly learn a lot from what they did on an engineering and production level. There's a million aproaches to making a record, and it's great to learn from somebody else. Sometimes the tracks that I mix, but don't record, come out sounding better, because they are a combination of more people's efforts and energies. The record is shaped into something that's maybe a little bit more unique than it might have been if it were just one or two people's vision.

I find that in mixing you can get more creative, with lots of different delays and textures and all. Sometimes when you work on your own tracks, you have preconceived ideas: "I'm going to mix it

RUNNING THE SESSION . . . continued —

and effects that you have at your disposal nowadays, it's really easy to use all the tricks in the book, and obscure the artist.

"Maybe I've just been very fortunate in that most of the artists I've worked with have had a really *clear* picture of what they are; it certainly makes my job as producer a lot easier. I like to get as much of the picture of what the record's going to sound like in the basic tracks. The earlier in production that you can put as many elements to give you the direction of where the record is going — be it songwriting demo, pre-production/rehearsal, or the basic track stage — the more it sounds *real*, and less mechanical.

"There have been times that I'll put a track down, the whole thing feels great, and everybody's jumping up and down, but you know in your heart that the guitar or bass sound isn't right. You try to go back and change it, but you start losing that energy and it sounds wrong. You've got to be able to say, 'Well, I've got to live with that bass-drum or guitar sound because the entire track works as a whole.' "

Do you find that your rapport with an artist continues to develop during the course of a session? That things change once you do get into the studio environment?

"You've got to really trust your instincts, and if your instincts say, 'Do another take,' you've got to do it. If everybody's getting bored, but you feel that the good take is just around the corner, you've *got* to be the taskmaster and just say, 'Come on guys, listen to it, it's not that far off, give it a couple more shots.' Sometimes you feel like beating people up, but you've got to follow what you believe in. I try to give people lots of space, but sometimes it's a matter of knowing when to be tough. Hopefully you can do it just before things get out of hand, just so you can turn the situation around and everything still flows smoothly.

"Making a record is about capturing that excitement and energy, and magnifying it. I don't believe it's a photograph. A record is *your* interpretation; you've got to take a look at it, and then amplify it even more to make it bigger than life.

"When a songwriter first comes up with a concept, he takes the idea and expands upon it. Then, when you bring the keyboard player into the picture, he takes a look at it, kind of pulls it apart, expands it and tries to widen the picture. Every person along the way, down to the mastering engineer, does that." □□□

"You've got to give people the room to be who they are. The minute you don't, it's not music. I don't want to put down Phil Spector, because his records were brilliant. But they're *his* records. For me, there's something that is lost in them; something that's not *real* about those 'Wall of Sound' records."

Joe Chiccarelli

so I'm going to put this delay on this, and that delay on that"; you close your mind off to the alternatives.

R-e/p: Three of the four Oingo Boingo sessions you produced were for sound-track albums: Fast Times at Ridgemont High, The Last American Virgin, *and* Surf II. *Do you find there's any difference in production techniques for film soundtracks, as opposed to traditional record dates?*

JC: The only thing I find different is that you go more for the energy and the attitude; you don't get as exacting in terms of sounds. Instead, you go for something that feels great, and seems to work with the picture.

R-e/p: Did you have any idea what the movies were going to be about when the songs were being recorded?

JC: In a couple of cases, I had seen the films and picked up a general attitude, in terms of maybe how off-the-wall to make it sound, or maybe how tame. But, in terms of the engineering or production approach, there wasn't much difference.

R-e/p: On the technical side, what facilities do you look for in a recording studio?

JC: Obviously it depends if you're mixing or tracking, but the *first* thing I check out are the monitors. Although I'll refer to a pair of Yamaha NS-10M monitors, I like to use the big monitors from time to time. It's essential that the room monitors are *accurate* and, more so, that you can go between the small ones and the big ones and the sound doesn't change. They're never going to sound exactly the same, but I look for a consistency in sound.

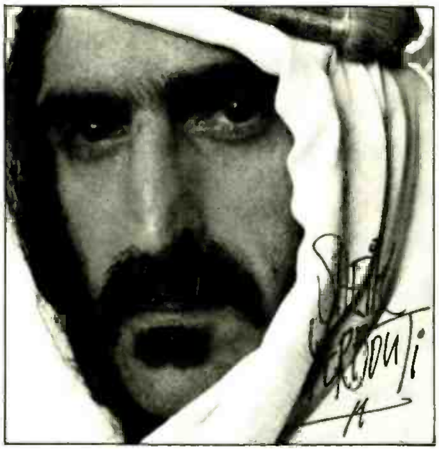
I find that the big room monitors come in handy on a tracking date, when it's real hard to bring in a handful of musicians and play back the track on a small speaker after they've been out in the studio banging away for hours at high volume levels! You want them to come in and feel that the track is bigger than life.

Obviously the extremes in sound are going to be exaggerated when you go upstairs to the large monitors, but they've *got* to be close enough that when you can make an adjustment on a small speaker, and then bring it upstairs, it doesn't sound like: "I just made the vocal ridiculously loud, now what do I do?"

I also use a pair of close-field Hemisphere speakers, which are built by a guy by the name of Jay Lewis. They tend to be almost the opposite of the Yamahas, in that they don't have a lot of upper mid-range, and they have a lower-mid/upper-bottom bump, where the Yamahas are a little recessed in that area. By going between all three sets of speakers you can really get your bass sound in shape. It has to work on *all* of them for me to be happy with the sound.

R-e/p: What West Coast studios do you prefer to work in?

JC: I'm a real Neve/Studer fan, so I tend to work in those type of rooms. Capitol has got two really good-sounding rooms. Sound Castle, where I'm working now, is great, and I work a lot at Conway Recording. There's something



about the bottom-end response of the Neve console that I like, and Studer multitracks always sound clean.

In the process of selecting a studio, I look for a room that sounds real live; a room in which instruments sound good, but which has controllable acoustics. I don't like drum booths — it's been a *long* time since I've put baffles and that sort of thing around a drum kit [laughter].

R-e/p: Do you prefer to track in one studio, overdub in another, and remix in another, or to use the same facility throughout a project?

JC: It depends on the project. Sometimes it's fun to go to another studio, because you hear things that you might have done differently in the first studio.

In a way, every piece of equipment tends to taint the sound — positive or negative — and it's interesting that different consoles give you a different sort of sound. Sometimes it's great to track on an older Neve 8068 and mix on a newer Neve 8128, or an SSL. Or to track on an API and mix on a Neve — that's a great combination to me.

Some consoles are "fatter-sounding," and some are "airier-sounding." If you record and mix on the same console, you end up with maybe *double* air or *double* fatness. You also have a tendency, because certain frequencies sound good on certain consoles, to use that same frequency twice; you might end up with a smaller sounding record than if you were on another console.

For a rock 'n' roll band, I'll go to a studio that's got a real big ambient room, and maybe a console that's a little dirtier-sounding — maybe an older, all-transformer desk. And if the date was something more techno-pop, I'd go to an SSL rom, because the SL-4000 has the gates, the compressors and parametrics on every channel. You can take a drum machine and really turn it into something else!

For mixing, I love Neve NECAM automation; that's my favorite. And the Massenburg [GML] system [recently installed at Conway Recorders] is very good. It does get a little complicated — maybe I haven't had enough experience on the GML yet — but the automation can do an *unbelievable* amount of things at lightning speed. A moving-fader system is great, because if you want to change something, you just grab the fader. Better still, when you play it back, you can see the fader move. With some of the VCA-based systems, I guess I never trust them because you cannot see the fader changes.

R-e/p: Do you try and track as many instruments as possible while recording the basics, to capture the band's energy, even though you may replace them later?

JC: Yes, I think it's important to get a picture of what the end product is going to be. It's real hard for me — and I think it's the same for most artists — to envision where the record is going if they just hear a bass and drums, or whatever. It's also difficult at times to get dynamics happening — to maybe make a chorus feel like it's sped up a little bit, or make the verses come down a little tighter — without hearing some of the other instruments. Even though, as you say, you might replace them to get maybe a better sound.

But sometimes you can get a take with the whole band out there and the whole thing feels great, and maybe you don't like the synthesizer sound, or whatever. You replace it with a better sound, but where the notes are hitting, how the sound fits in, and the energy that the player put it, just doesn't work as well as the first take. You have to trust your instincts.

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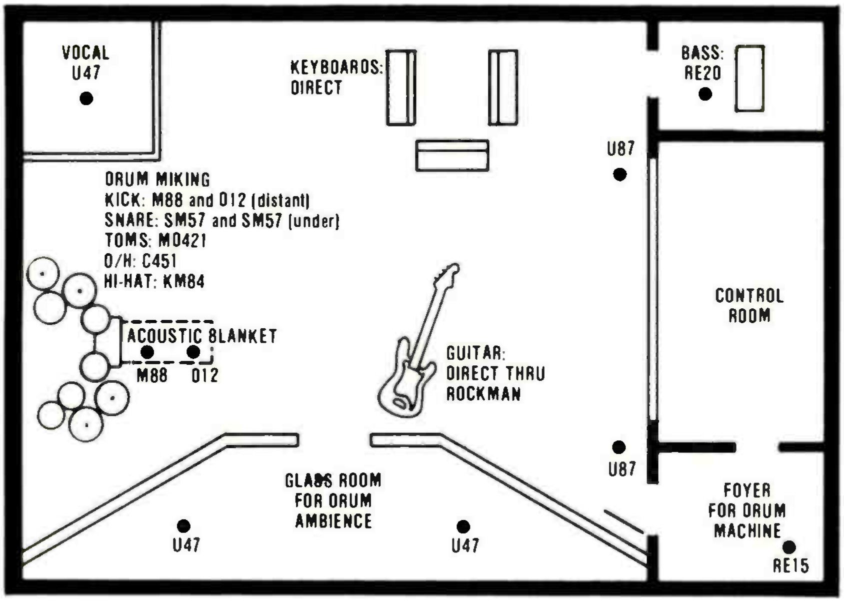

MICROPHONE AND ROOM LAYOUT FOR ROBERT TEPPER BASIC TRACKS AT SOUND CASTLE STUDIOS, LOS ANGELES JANUARY 1985. ENGINEER/PRODUCER: JOE CHICCARELLI.

Detail of Chiccarelli's two-mike technique for kick drum, with a blanket to prevent leakage.

Joe Chiccarelli

R-e/p: What about instrument miking and recording techniques. Let's start with the drums.

JC: I find that, in general, my miking changes a lot. There's a few mikes that I always go back to, but from project to project, and even within a project, I get bored using the same microphone every time. Maybe for a particular album, if I'm thinking of a fatter snare sound, I'll try a Neumann KM86 instead of a Shure SM57. I'll mike top and bottom. Sometimes I don't use the bottom mike, but I like to have it here just in case I need it.

R-e/p: And for kick drum?

JC: My standard miking — and again it depends on the project and the sound I'm after — is either a Beyer M88 or a Sennheiser MD421. The Beyer has got just the right amount of punch, and is not as "slappy" as a 421. It's also got a little warmer bottom end. But if the kick is really dull or not punchy enough, I tend to use the 421. I'll mount the M88 or 421 inside the kick, maybe five or six inches in from the outside rim, pointing at the skin. I also use another mike, an AKG D12, that's mounted three or four feet back from the kick. That way, you get more impact from the front mike, while the back one captures more of the 30 or 50 cycle low-frequency. I build a kind of tunnel over the back mike and the kick to keep out the sound of the cymbals, although sometimes I find I don't need it. I always end up gating that ambient mike, the one that's a little further back, because you get so much cymbal splash. But because it's a D12, it doesn't pick up a lot of top end anyway. I usually roll off the ambient mike, and add a little bit of 30- to 60-Hz boost, to get a little bit more "puff."

I'll also put up a couple of Neumann U87s for room mikes — or, every once in a while, I'll use Neumann M49s — that are close to the back wall, facing the drums. Sometimes I put them close to the floor, if the room doesn't have a lot of bottom end. But I'll usually just put a stereo pair of 87s and then some others around the room.

Here at Sound Castle, I have an Electro-Voice RE-15 out in the hall that picks up some real trashy ambience. In the isolation booth — which is nice and live — I have two U47s that give a real bright ambient sound. For different tunes, you can have a longer room sound or a shorter room sound.

To prevent the cymbals from bleeding too much into the room-mike tracks, I'm keying the room mikes from the kick, snare, or tom tracks. By keying from a spare cue buss, I'm able to regulate what drum I want to use to open up the room mike. You've got to be gentle with the gating attack; you can't just sort of open and close it like a brick wall because it would sound *very* odd.

R-e/p: What mikes do you use on toms?

JC: Lately, I've been using 421s, because they have a nice edge to them. For the Zappa sessions, I used AKG C414s pretty much on all the toms. I've used 57s every once in a while on toms that are a little dull, and when I want to get some more crack or snap to them. There was a period of time when I liked Neumann KM84s on toms, but sometimes if you have a lot of condensors up on a drum set, it just gets real fizzly, and you end up with phase problems, because the top-end response is so good, and the polar patterns are wider than on a dynamic. Also, the cymbals were getting everywhere, and I had phase cancellation across the kit. Even if you move them around, condensors still give you that fizzly high end, so I go back to dynamics.

On snare, it really depends on the song and the sound I'm after. For ballads or songs where I want to get a fatter snare sound, I'll use either a KM86 or MD441, which are hypercardiods, and help get rid of some high-hat leakage. But, in general, I'll use a 57 on top, because it's just nice and cracky, and on the bottom either a 57, 441 or KM86.

High-hat is usually covered with a KM84; a couple of times I've used a Beyer M160 ribbon. If the high-hat is a little dull, or I can't get in at the right angle, I'll use maybe a C451 or something like that.

Choice of overheads also depends on the sound. For a lot of really tough rock 'n' roll dates, sometimes a pair of M49s sound great — they add a thickness to the cymbals that really seem to work.

R-e/p: And on guitar backline amps?

JC: I usually put in a handful of mikes, including a 57, Neumann U67, maybe an M49, U87, U47, and others. I'll mount four, five, or more mikes in front of the amp, at different angles and distances. Then I'll put an 87 back about three to five feet, and another 87, or maybe a 414, way back in the room — I'll try different distances and different combinations. Depending on the sound, I like to squish guitars a lot, especially for rock 'n'roll —I like to get them where the meter doesn't move and just sits right up there!

For compression I like to use old 175Bs — which is really a tube version of the UREI LN-1176; Sound Castle has several of them. They just have a nice distortion to them; there's something about the way the 175 breaks up that sounds great on guitar. I'll also use a dbx 160 on guitar. Or, for things like muted guitar parts, or something you want to be a little cleaner, a little tighter sounding, I'll hook in a 160X.

Lately, I've also been compressing drums more than usual. Rather than leaving the kick and snare unaffected in terms of dynamics, I'll compress them a little bit. Again, I find it just brings up the sound in your face, and gives them more power.

When I listen to a record, the *first* thing that I hear is the snare drum. To me, that determines how energetic the track is, how big it is, and how bright it is. These days you really have to have a happening snare-drum sound, or the track sounds wimpy. If the snare drum isn't tough, or at least unique sounding, I feel like I've been sold short on the song.

Vocals, for me, are the most difficult things to mike, because every voice is different, and you can't use just one mike on every single voice. I try out a different vocal mike every day while we're tracking. You'll find that there'll be three or four mikes that work with that particular voice; one mike might work great for the ballads, another for hard rockers, another when the singer is in the upper register, one when he's lower, and so on.

Sometimes I've actually used a combination of mikes for vocals. In fact, for Paul Cotton from Poco we used an Electro-Voice RE-20, and he sounds great. But, for me, there wasn't enough top end on it, so we put an AKG C451 or C452 right on top of the RE-20 — it kind of looked like a space shuttle — and "time aligned" it by moving the 451 back and forth until it sounded right. By just blending in a little bit of the 451 with the 20, we got the air that made Paul's voice sound great.

R-e/p: Have you had an experience with digital recording?

JC: I've mixed a lot of things to JVC VP-900, Sony PCM-1610, and the Mitsubishi X-80. Of them, my favorite has been the JVC; I found that to be a little warmer sounding on the bottom end. Actually I did some Soundstream recording, and the bottom-end response on that system is unbelievable. The Mitsubishi sounded great on tunes that were a little airier, more atmospheric, but on the songs that demanded a little bit more punch and tightness to the bottom end, I thought the bottom end kind of "loosened up," and wasn't as tight as that well-known analog compression sound.

R-e/p: Do you change your mixing techniques for digital?

JC: When you start mixing digital, you become more aware of the flaws in your mixes. Also the dynamics aren't

Joe Chiccarelli

squished as much by the tape as they are with analog. So I found that I've gotten a little more finicky in terms of digital mixes. Maybe in terms of making the top end a little sweeter sounding because, in a lot of cases, with digital it comes back sounding a little edgier.

But with analog, it's amazing how much top end goes away in a matter of hours, and more so after a few plays across a head — no matter how demagnatized it is. You cut a track yesterday, listen to it today, and it's not going to sound as bright.

R-e/p: When you track to analog, do you mix a little brighter, knowing that the top end will gradually fade?

JC: No, I don't find that I compensate, because right now it's something we have to live with. I've been using Agfa PEM468 for a lot of years, and I love the way the top end holds on 468 compared to other tapes. I like the sound of half-inch masters: The bottom end is more solid, and the top has a little bit more air than quarter-inch. A lot of people still like the sound of quarter-inch for rock 'n' roll. But, to me, it's just a little too *small* sounding, and compressed.

R-e/p: Which albums did you mix to digital?

PC: Let me think. The last Poco album, *Inanammorata*, we did on the Mitsubishi X-80. Ray Manzarek's *Carmina Burana* we mixed to Sony PCM-1610 and that, because of the dynamics, sounded great on digital. It has these huge, bottom-end synthesizer notes and kettle drums. With analog you would be made aware of the flutter and noise in the quieter passages.

R-e/p: Are there any particular items of outboard or processing hardware that you've been using recently that have impressed you?

JC: The new dbx 160 and 160X are really nice compressors. I use the Drawmer gates a lot, because they have a keying feature that's frequency selective. You can make it open up on the bottom end or on the top end, depending on how you adjust the filters on the key input. I just bought an AMS DMX15-80 digital delay that I love. Number one, it's *so* clean sounding.

R-e/p: Did you use the AMS to sample sounds?

JC: Yes, I use the sampling feature a lot. Many drum machines, because of their eight-bit sampling, lack of top end. When you add HF boost on the console, it just doesn't sound right. So I've done a lot of things by using a drum machine to trigger a snare or kick drum sound that we have put into the AMS. That sounds really good, because you can get pretty close to a real tough-sounding snare drum that we've sampled into the DMX.

The AMS RMX-16 digital reverb is also great; in particular some of the ambience programs and the reverse feature. I've also been using the MXR 01A digital reverb, and there are a couple of special effects programs that are fantastic. I was really impressed with the Number 9 effects program, which is like a reverse echo, but has a weird character to it.

R-e/p: What's the memory capacity of your AMS DDL?

JC: 3.2 seconds.

R-e/p: Are you thinking of increasing that capacity?

JC: To 25 seconds? Maybe at some point, but I find that if you want to fly in a whole chorus of background vocals, for example, it's just as convenient to dump it to half-inch tape and back

again. Plus the fact that *then* it's a challenge to get it in time! [Laughter]

R-e/p: You don't lay it over to a two-track with center-channel timecode and then synchronize it with the multitrack?

JC: Actually, that's a great idea. I never thought of trying that. One of the new Studer A810s, Otari, or MCI two-tracks with timecode would be useful.

With two multitracks it's fun doing that. If you have a slave full of background vocals, you can easily program in an offset as you dump them back into the master tape. Like moving what's in the first chorus to the second chorus.

R-e/p: Do you do many 46-track sessions?

JC: More and more lately. It seems that with all the drum machines and synthesizers, plus wanting to record effects on tracks, and tracking more things in stereo, I can't get away from it.

It also saves wear and tear on your master when you're overdubbing. If I can, I love to record my basics on 16-track, and then sync up a 24-track for the overdubs. The 16-track sounds great, because the extra track width gives you a tougher and quieter sound. Sometimes it's impractical to go 16, because a lot of studios don't have a 16-track, or the headstack is hidden away somewhere and they've forgotten where it was!

R-e/p: With the growing emphasis on synthesizers and drum machines, are you getting more into the using sync tracks, sequencers and things like that?

JC: Pretty much on every track I'm doing now — unless it's a "feel" thing, where it can't be done to a click — I'll find the tempo and print a sync tone from the LinnDrum, plus a cowbell or whatever it is for timing purposes. Provided that the take is in time with the click track, that gives you the ability to add sequencers, or trigger things from a Linn trigger track. Or if you run a whole different drum texture for one verse of the bridge, you can easily punch in a LinnDrum with some weird sounds, or some sampled sounds from the AMS. I'm trying to get hold of the new Roland SBX-80 Sync Box, which allows everything to be triggered from SMPTE timecode. That would allow you to roll the tape to the middle of the song and just punch right in on it, as opposed to having to run from the start of the tune all the time. I would like to be able to reference the drum machines and sequencers to the SMPTE track. That way, if I put in a drum pattern in at such and such a timecode number, you just go to that location and it's all in sync in a matter of a couple of beats. The unit fires off trigger codes and MIDI pulses when the location arrives from the tape.

R-e/p: There seems to be an emphasis being placed these days on subtle "textures" and "colors" in a production. Are you a particularly visual producer? Do you have a complete "picture" of the finished product before you start a project?

JC: After getting input from the artists, I try and visualize how I want the finished record to sound — how big I want it; how deep I want it; what kind of textures and instruments I want to use. But it always changes. Things that you have in your head to begin with, all of a sudden don't work, or you decide you want to take a left turn. Or maybe your picture was a little *too* restrictive, and you want to get a little wider with it by putting in unusual percussion sounds, or whatever it might be. But I do try to get a picture of who the artist is, and what he wants to sound like, and build the record around that.

But with all the technology available in a modern recording studio — and the ability, with drum machines, digital delay and reverb, to come up with a wide variety of textures — it is more important than ever to maintain a very clear focus on *who* the artist is, and *what* they want to sound like. That's the challenge of the creative producer. ■■■


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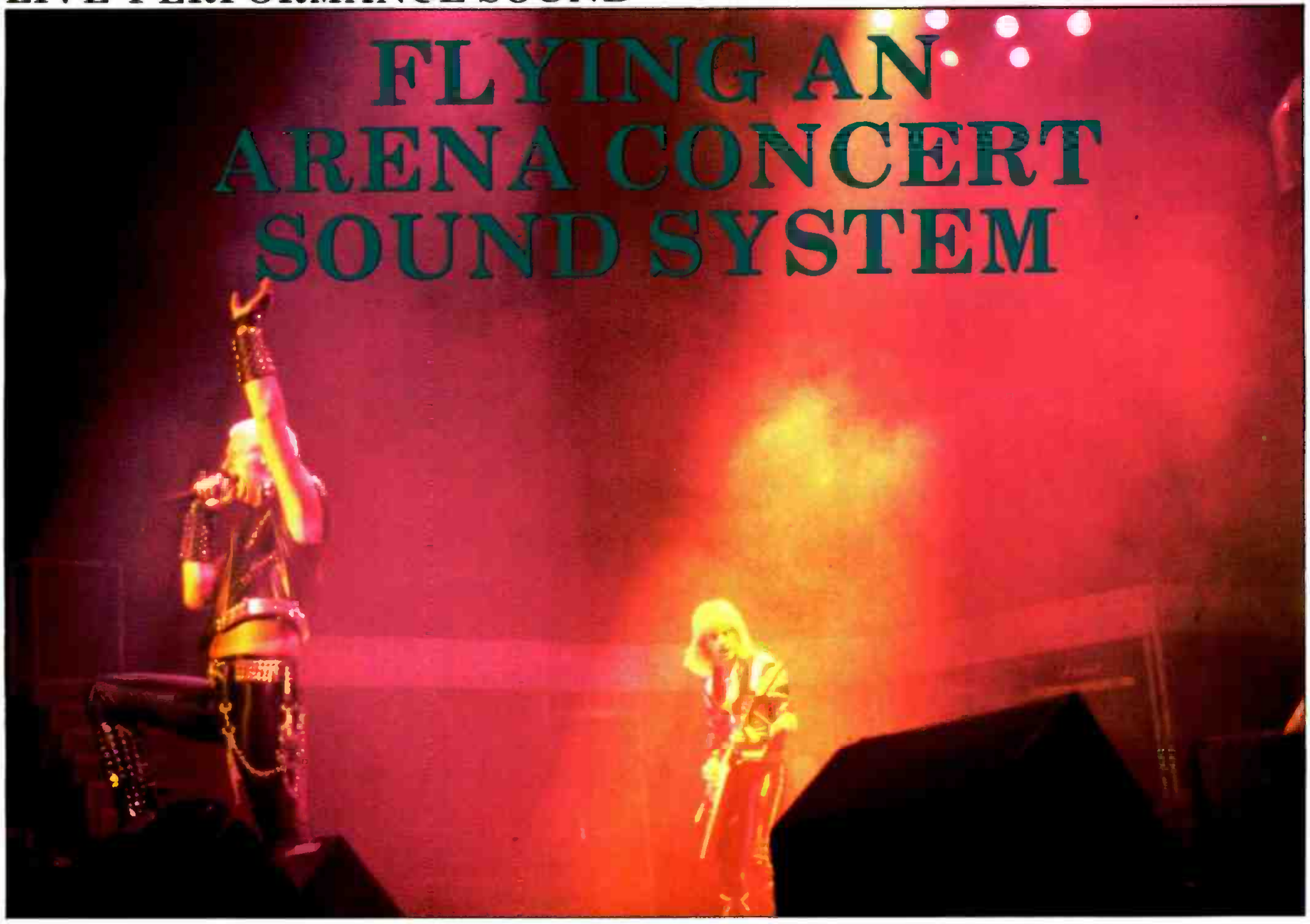
FLYING AN ARENA CONCERT SOUND SYSTEM



All photography by David Scheirman

The TASCO Harwell System on Tour with Judas Priest at the Long Beach Arena

by David Scheirman

In the early Seventies, several pioneering concert-sound companies, including Clair Bros., Showco and Stanal Sound, developed loudspeaker systems intended for use in a "flying" configuration. As large, multi-purpose convention centers and sports arenas became acceptable venues for live-performance musical events, the larger sound reinforcement systems being brought into play required specific hardware arrangements to enable the loudspeakers to hang, or fly, from the venue's ceiling.

Innovators in the field, such as Bill McManus (McManus Enterprises, Philadelphia), developed methods for using chain-motor hoists to suspend heavy lighting trusses above performance stages. Early flying sound systems used similar techniques and hardware to suspend existing loudspeaker enclosures.

Today, many concert sound systems are deploying loudspeaker arrays that have been engineered specifically with hanging requirements in mind. And, as hanging-system technology has progressed, it has become apparent to firms involved in the field that correctly designing the array itself is perhaps of greater importance than the make-up of an individual speaker cabinet. For example, a loudspeaker enclosure that works well as a stand-alone device may not provide optimum results when used as part of a poorly designed flying system array.

One concert-sound system specifically engineered for use in a flying configuration is the Tasco Harwell System, and named after the English engineering firm that developed it. Tasco is a British-based, full-service entertainment production company that also maintains systems in the United States, with office and warehouse space in the Los Angeles area. The Tasco Harwell System has been deployed for recent, major arena tours by such artists as Duran Duran, Diana Ross, and Judas Priest. In the Spring of 1984, this writer made a trip to the Long Beach Arena in Long Beach, California, to observe the system in action as Judas Priest played two sold-out nights at the 13,000-seat venue.

System Design Concepts

"In Britain, there are probably only a half-dozen venues the size of the

FLYING ARENA SOUND SYSTEMS

Long Beach Arena in the entire country," notes Tasco crew chief Tony Blanc (*pictured left*).

"Also, the typical tractor-trailer truck is of smaller dimensions. PA gear, as a general rule, tends to be smaller than it is here in the United States due to such factors. However, there is a great need in the North American market for arena flying systems. Tasco needed systems like this one to service its clients who were touring this country, including acts like Judas Priest. The Harwell design group developed this system in England, and it has seen extensive use all over the European continent. Since bringing the first Harwell system over to this country, we have been working it practically non-stop."

Blanc feels that several features of the Harwell speaker system make it particularly innovative. "In my opinion, there are only four designers or companies that are actually working on the development of anything that is really *new* in concert speaker systems — [a design] that is not just a rehashing of the same old ideas that have been around for over 30 years. In Britain, you have Dave Martin of Martin Audio, the Turbosound Rentals people, and the Harwell group. Here in the U.S., John Meyer is developing some interesting new systems. To me, everything else just looks and sound about the same."

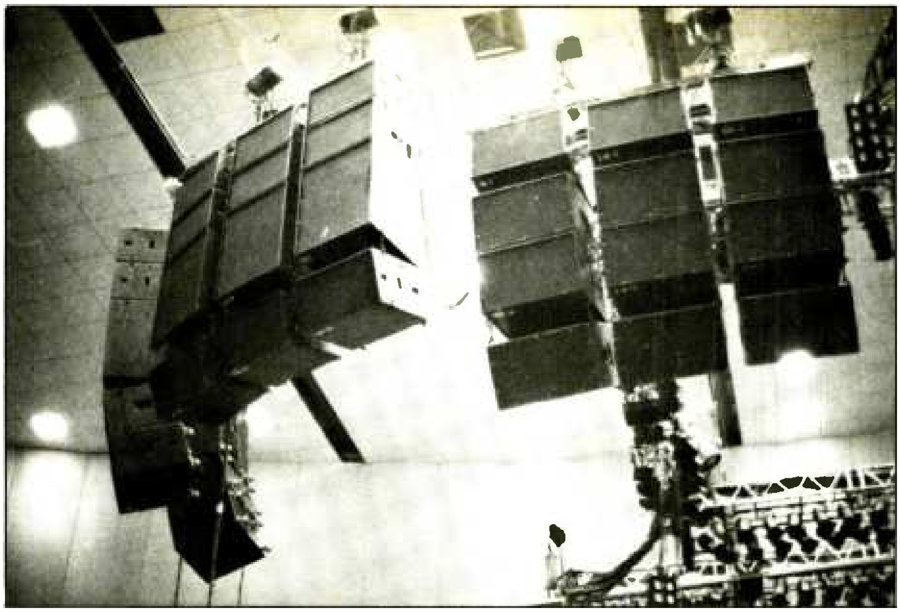


Figure 1: The Tasco Harwell system comprises three different types of enclosures (bass, low-mid and high-mid) that are combined in multiples of various configurations to form a massive flying system array. Shown here is half of the overhead system (stage right).

Tasco's Harwell system comprises multiples of three different types of loudspeaker enclosures. "The packaging of a road system today is very important," Blanc explains. "If you put all of your components in one box, that's like putting all of your eggs in one basket, so to speak. The system will not be nearly so versatile, in terms of being able to direct different parts of the speaker system to different acoustical areas. On the other hand, having a half-dozen different types of boxes, in all sizes and shapes, makes for a very difficult truck pack, as well as being time consuming. We feel that this system is a good compromise between the two."

Blanc also notes that the system is in a state of design evolution, and a newer version is in the works that features smaller cabinets.

Loudspeaker Cabinets

The Tasco Harwell System as supplied to Judas Priest contained three different types of cabinets. The boxes were designed with identical width and depth dimensions (40 by 40 inches). The largest box is the bass cabinet, measuring 48 inches high. Half-sized mid-bass and high-mid

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Figure 2: Component layout of Harwell System loudspeaker cabinets.

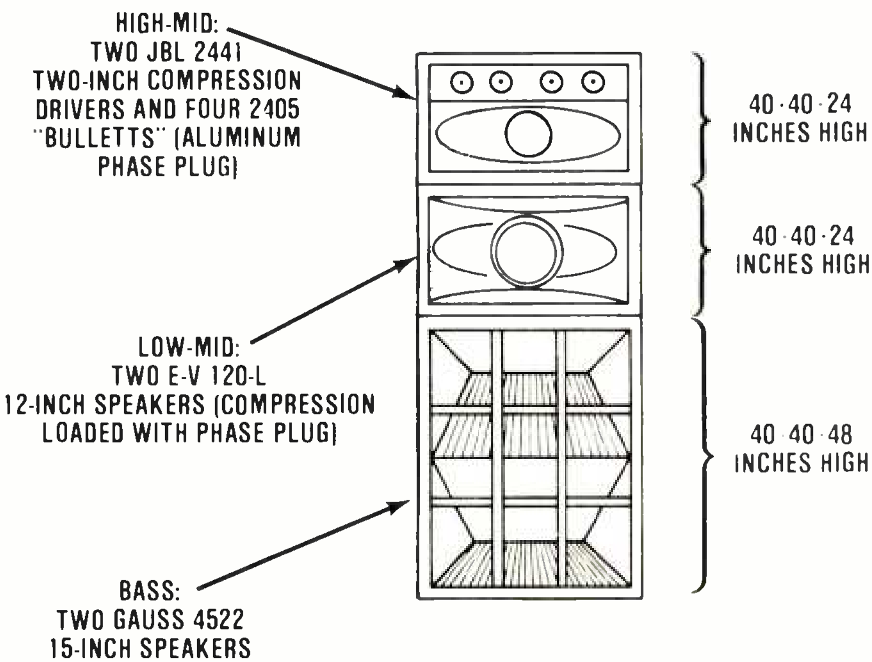
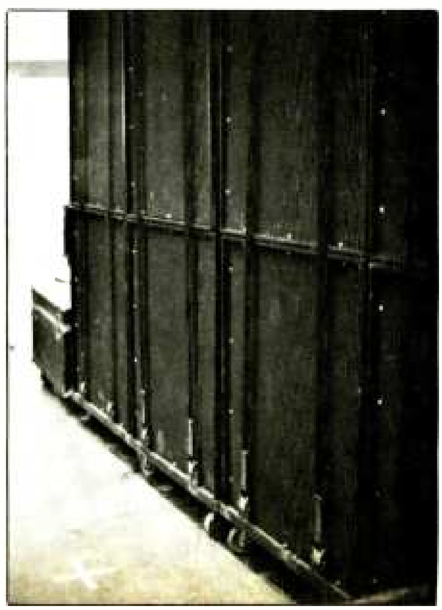


Figure 3: Tasco bass cabinets, each measuring 40 by 40 by 48 inches high, were left on travel dollies and stacked at floor level.



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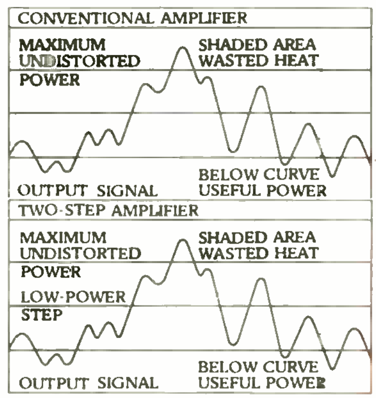

QSC Quality Service Commitment

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FLYING ARENA SOUND SYSTEMS

cabinets complete the package, and each measures 24 inches high. The bass cabinet (housing two, 15-inch speakers) weighs approximately 310 pounds (Figures 1 and 2).

Loudspeakers used to handle the bass-frequency band are Gauss Model 4582s. These units are housed in a folded-horn enclosure that features extensive bracing to maintain the cabinet's structural integrity at extremely high sound-pressure levels, and are fed a signal of 20 to 200 Hz. (Figure 3).

The low-mid cabinet contains two Electro-Voice Model 120L 12-inch speakers housed in a unique enclosure that features a large foam-filled phase plug. The speakers are compression-loaded into a fiberglass horn that provides exponential flares in both the vertical and horizontal planes. These cabinets handle the 200 Hz to 1.2 kHz bandwidth.

The high-mid cabinets each contain two JBL Model 2441 two-inch compression drivers loaded via a four-way manifold throat onto a custom fiberglass horn. An aluminum phase plug also is employed. The midrange frequency bandwidth is 1.25 to 6.3 kHz, high frequencies being presented with four JBL 2405 HF radiators housed in the same cabinet and operating in the 6.3 to 18 kHz band.

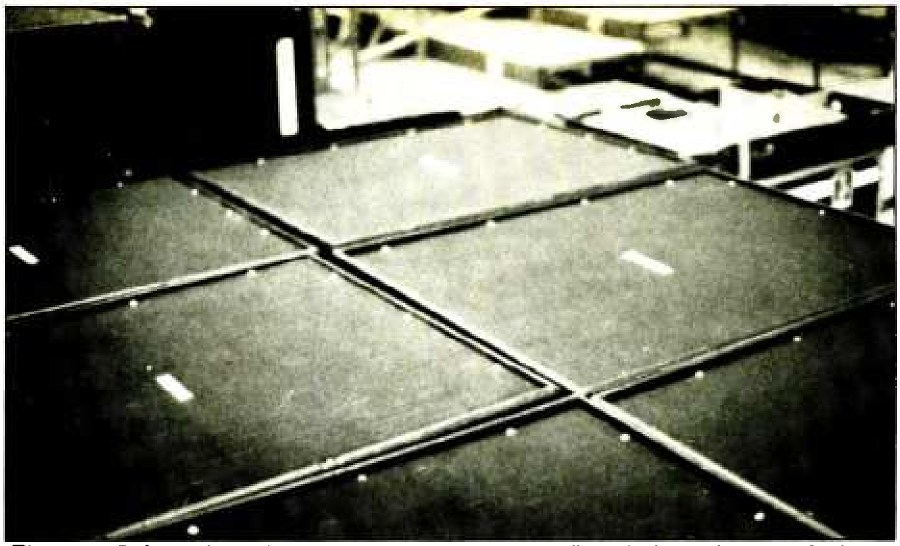


Figure 4: Subwoofer cabinets were arrayed on the floor behind the sound wings, pointing straight up. A narrow-bandpass signal of 20 to 80 Hz was passed through them.

For Judas Priest, Tasco supplied 30 bass cabinets, 32 low-mid cabinets, and 28 high-mid cabinets. Additionally, 12 bass cabinets were supplied for use as subwoofers. Positioned on the floor behind the sound wings, six per side, these bins were fed a narrow-bandpass signal of 20 to 80 Hz (Figure 4).

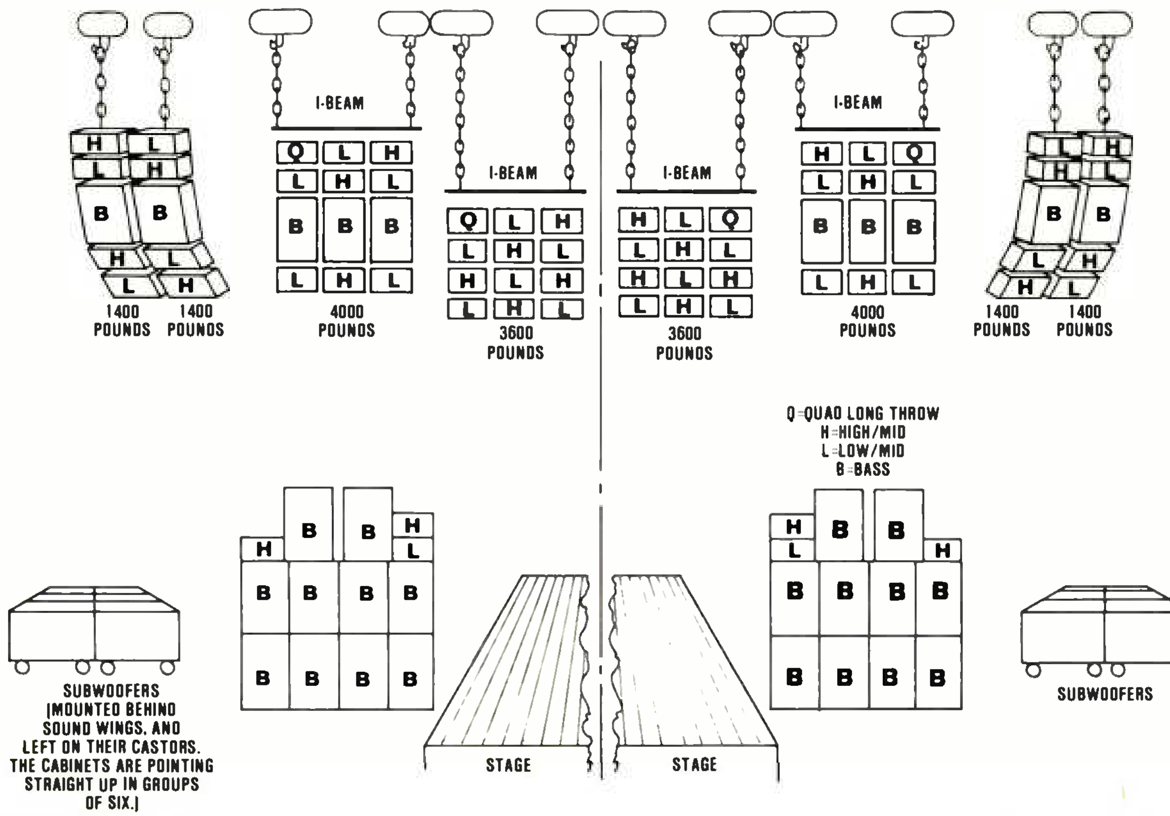
For long-throw high-frequency coverage, four cabinets were also supplied, each housing four JBL Model 2441 drivers mounted on long-throw fiberglass horns. These boxes were positioned in the hanging array so as to point towards the parts of the arena that were farthest from the loudspeaker system.

All told, the loudspeaker system contained a total of 84 15-inch speakers, 64 12-inch speakers, 72 two-inch compression drivers, and 112 high-frequency "bullets." The majority of the bass enclosures stayed on the ground (10 in the air, and 32 down, including the subwoofers), while a majority of the low-mid and high-mid cabinets were flown (Figure 5).

Flying the Sound System

Rigging requirements for the sound system in the Long Beach Arena total 12 individual hanging points. Each "point" comprised a spot in mid-air to which a single chain-motor hoist

Figure 5: Block Diagram of complete hanging and stage system for Judas Priest at the Long Beach Arena.



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could be attached. Double-bridled half-inch steel cable was used to support each point.

"In England, there are very strict laws concerning the hardware and methods that may be used to suspend anything above the general public," Blanc notes. "Half-inch steel is mandatory, and the nylon span-set straps that are commonly used over here in the U.S. have traditionally not been allowed."

Two-ton, CM-Lodestar chain-motor hoists lifted the weight of the loudspeaker system off the ground and into the air above the performance area. Rigid I-beams attached to two motors supported four sets of speaker enclosures, with each set weighing up to 4,000 pounds, including hanging hardware. Additionally, four sets of enclosures were single-hung (one motor), with each set weighing approximately 1,400 pounds, including cable.

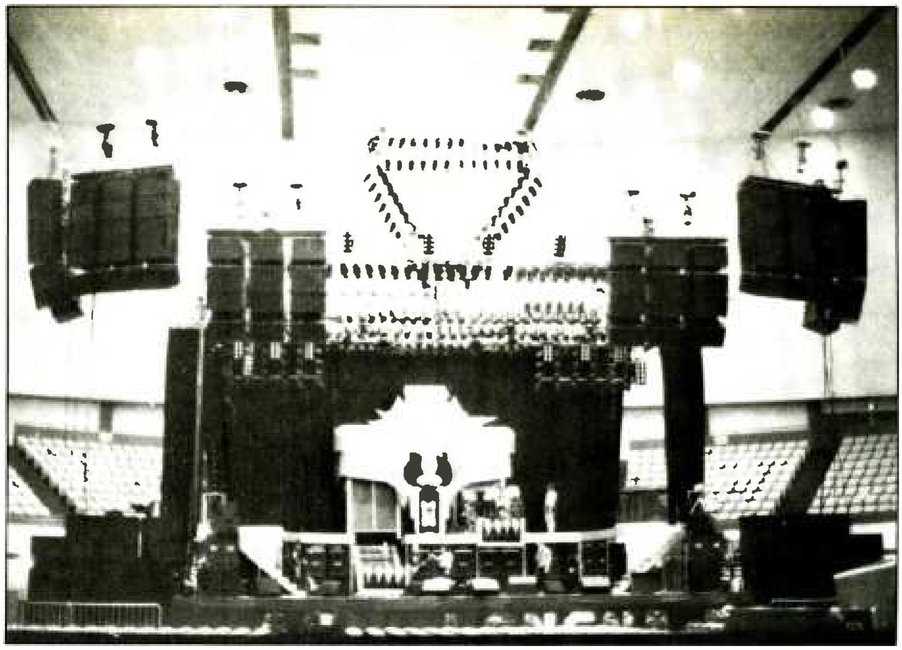

Figure 6: For Judas Priest, Tasco supplied a total of 42 bass cabinets, 32 low-mid cabinets, 28 high-mid cabinets. Ten bass bins were stacked per side on the floor, with an additional six cabinets per side used as subwoofers.

HANGING SOUND SYSTEMS
Keeping the Hardware in the Air

Many a concert-goer has enjoyed the sound of shows presented with hanging arena systems. Since the early Seventies, sound-system rental firms have been faced with the challenge (and responsibility) of hoisting thousands of pounds worth of loudspeaker cabinets into the air above the performance area.

Chances are that few audience members actually stop to consider exactly how that ponderous black mass of sound and lighting equipment was suspended in mid-air, who put it there, and just what keeps the gear in place until after the performance.

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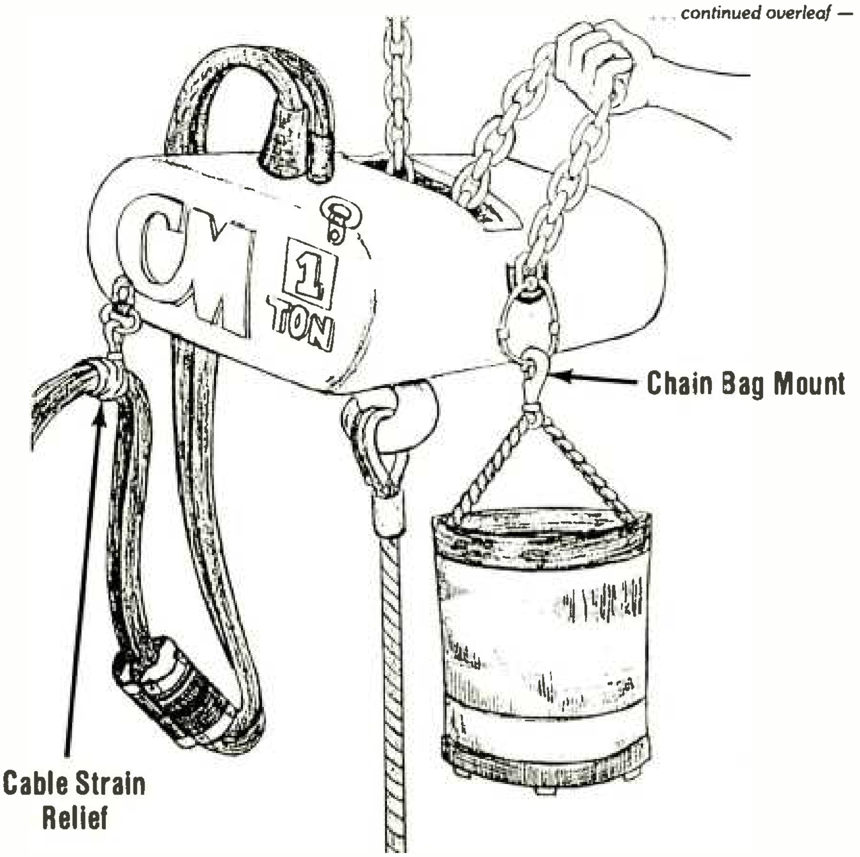


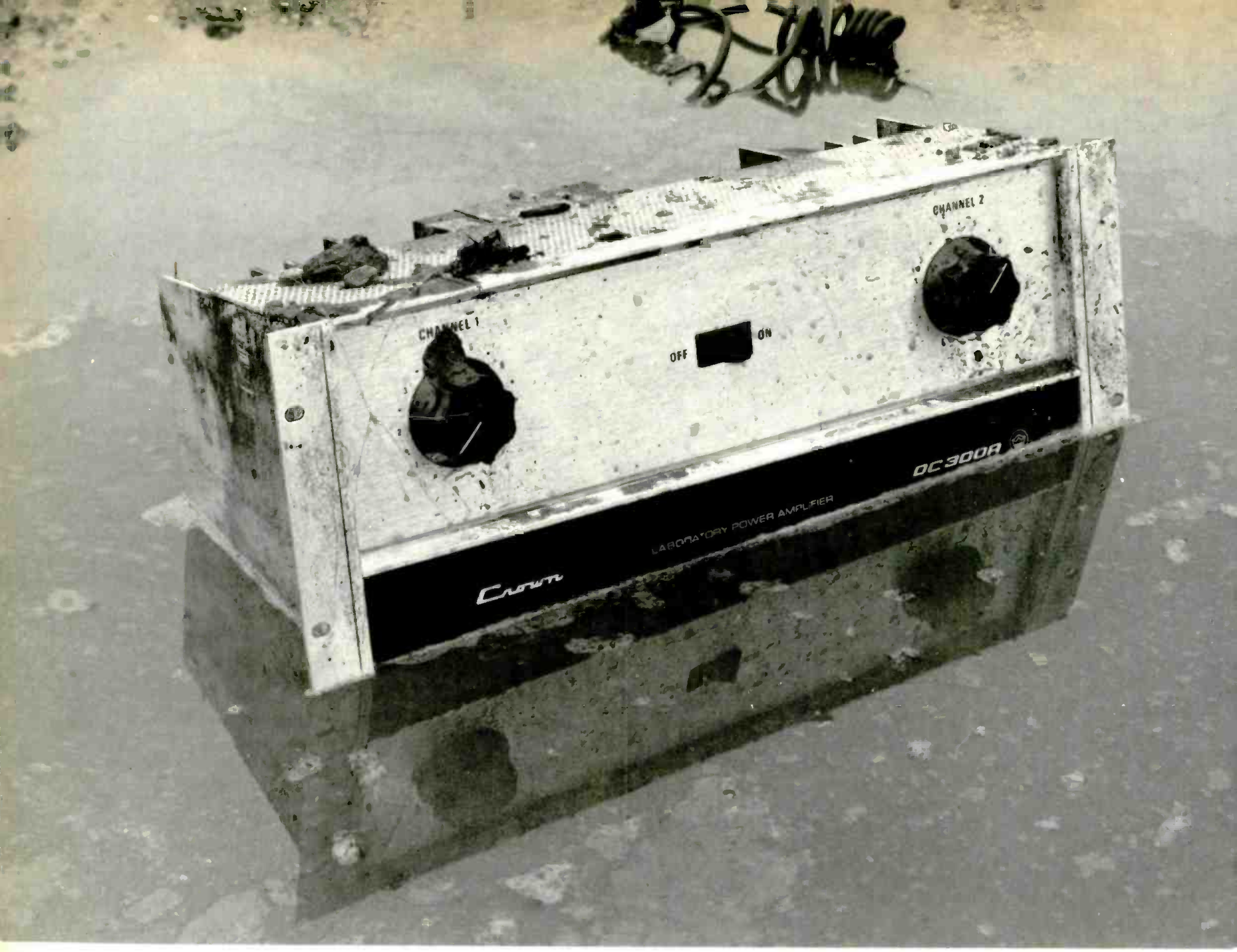
A total of 12 motors and four I-beams kept a total of 68 cabinets, weighing approximately 20,800 pounds, suspended for the duration of the performance event (Figure 6).

"The system configuration that you see in use here is what has worked out to be the best-sounding and most practical method of using this system with Judas Priest," explains Gordon "Gungy" Patterson (*pictured right*). "I have experimented with using it in a variety of different setups; it has worked out best to leave most of the system's low end down on the deck, and to get the high-frequency components up in the air. Since the speaker system is laid out in a modular fashion, I am able to pick and choose just how much of each frequency bands I want to stay down, and how much to go up in the air.

The system, as you see it here, is set up in pretty much this same configuration in every venue that we play. That gives me a very consistent audio system to work with for presenting the band's music. The variable, of course, is the particular acoustics of each hall."

Amplification and System Front End

BGW and Crown amplifiers powered the Judas Priest system, bass cabinets being driven by BGW Model 750-B and 750-C units, bridged mono, with four, 15-inch speakers connected to



In the early evening of Sept. 17, 1973, Jay Barth was at the wheel of a 22 ft. utility truck that was loaded with sound equipment. Just south of Benton Harbor, MI an oncoming car crossed the center-line; fortunately Jay steered clear of the impending head-on collision. Unfortunately, a soft shoulder caused the truck to roll two and one half times. Exit several Crown DC-300A's through the metal roof of the truck's cargo area.

The airborne 300A's finally came to rest — scattered about in a muddy field, where they remained partially submerged for four and a half hours.

Jay miraculously escaped injury; the amplifiers apparently had not.

Unbelievably, after a short time under a blow-dryer all the amps worked perfectly and are still going strong.

The rest — and the truck, is history.

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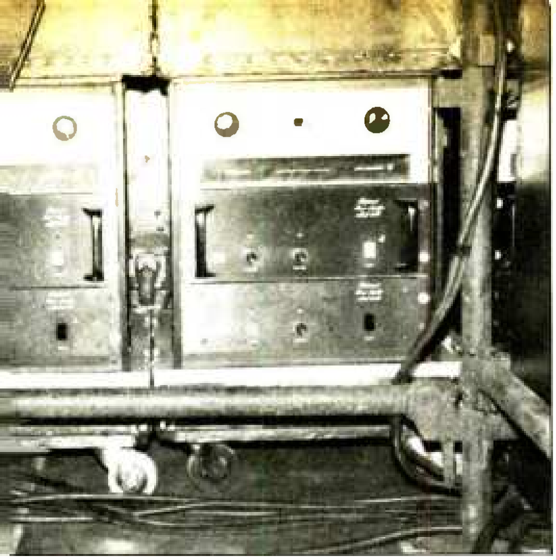

Figure 7: Representative power amplifier racks housed BGW-750s and Crown DC-300A models.

each single amplifier. The bass cabinets used as subwoofers were driven by Model 750-As run in stereo mode, with one cabinet driven by each amp side. BGW Model 250s powered the low-mid cabinets in bridged mono, with each amplifier driving four loudspeakers.

Crown DC-300As powered the high-mid boxes, two cabinets containing four, two-inch compression drivers being driven by each channel. In addition, eight high-frequency "bullets" were connected to other channels. The long-throw horn cabinets were powered by BGW Model 250 amplifiers with four drivers per amp channel (Figure 7).

"The power section of this system is undergoing changes right now," explained Tasco's Tony Blanc. "The company recently bought out all of the amplifier racks that were used by an East Coast sound company; that is where all of these older DC-300As came from."

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HANGING SOUND SYSTEMS . . . continued

The universally accepted method of hanging today's sound systems from the ceiling beams of arenas makes use of electric chain hoists, such as those manufactured under the name C/M Lodestar. A typical arena tour travels in a self-contained fashion with one or two dozen motors, along with steel shackles, bridles in assorted lengths, and nylon Span-sets (endless-loop, high-capacity load-bearing straps) for use in lifting and suspending the sound and lighting systems. Typically, this hardware is overseen by high-climbing, strong-backed individuals known as "riggers."

Most major touring sound companies maintain their own stock of rigging gear for use with arena sound systems. Chain motors are also a common sight at large outdoor festival sites, where they are used to construct "elevators" for assembling the towering stacks of loudspeaker cabinets.

Rigging as a Service Industry

Oftentimes, sound (and lighting) firms will find the rental of rigging gear to be more cost-effective than actual purchase, an economic fact that has led to the development of specialized rigging companies, formed to serve the touring entertainment industry.

Stage Rigging, Inc. (P.O. Box 95, San Carlos, CA 94070) has grown from humble beginnings (six hoists) in 1977 to now being one of the nation's largest specialty rigging firms for touring shows. Under the direction of Rocky Paulson, the firm's working stock of more than 150 chain motor hoists has recently serviced such arena tours as Huey Lewis and the News, Barry Manilow and Neil Diamond.

"Sound companies often come to us because they do not want the responsibility of owning, maintaining, and inspecting rigging gear," explains Paulson. "Sound engineers would usually rather be digging into an amplifier than an electric motor!"

C/M Lodestar hoists are commercially available with rated capacities ranging from ⅛-ton [250 pounds] to two tons. One-ton and two-ton motors are by far the most popular sizes for hanging sound gear. A hoist with a one-ton rating is able to dead-lift 2,000 pounds of speaker cabinets, steel shackles and cabling into the air, and hold it there indefinitely (at least until the concert is over!).

"The load rating is a bit misleading," notes Paulson. "There is actually a 5-to-1 safety factor built into every part of a C/M hoist. A clutch prevents the unit from actually lifting more than 125% of its rated capacity, but it can actually bear a standing load five times that weight before you would see hoist failure or broken parts. But it is very important to not exceed the rated load capacity of a given hoist."

Most users of rigging systems rely on 220V AC models of the chain-motor hoists. However, occasionally a sound company will opt for the 110V version. Paulson feels that the 220V is the best choice: "Each 110V unit draws up to 15 amps when operating," he points out. "But, when you have initial start-up of the motor, you have three times the current draw, and that takes a pretty heavy-duty power-distribution system to run on 110 volts."

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HANGING SOUND SYSTEMS . . . continued

Strapping Up the Sound System

Once upon a time, component sound systems were piled and strapped onto cumbersome platforms or "baskets" that had hook-up points for attaching steel cables and the chain motor. Today, most touring arena sound systems are set up with simple I-Beam or bar hardware. Nylon straps with safety clips are often used to assemble modular speaker cabinets in vertical columns. Each cabinet will typically have integral stress-rated hanging points.

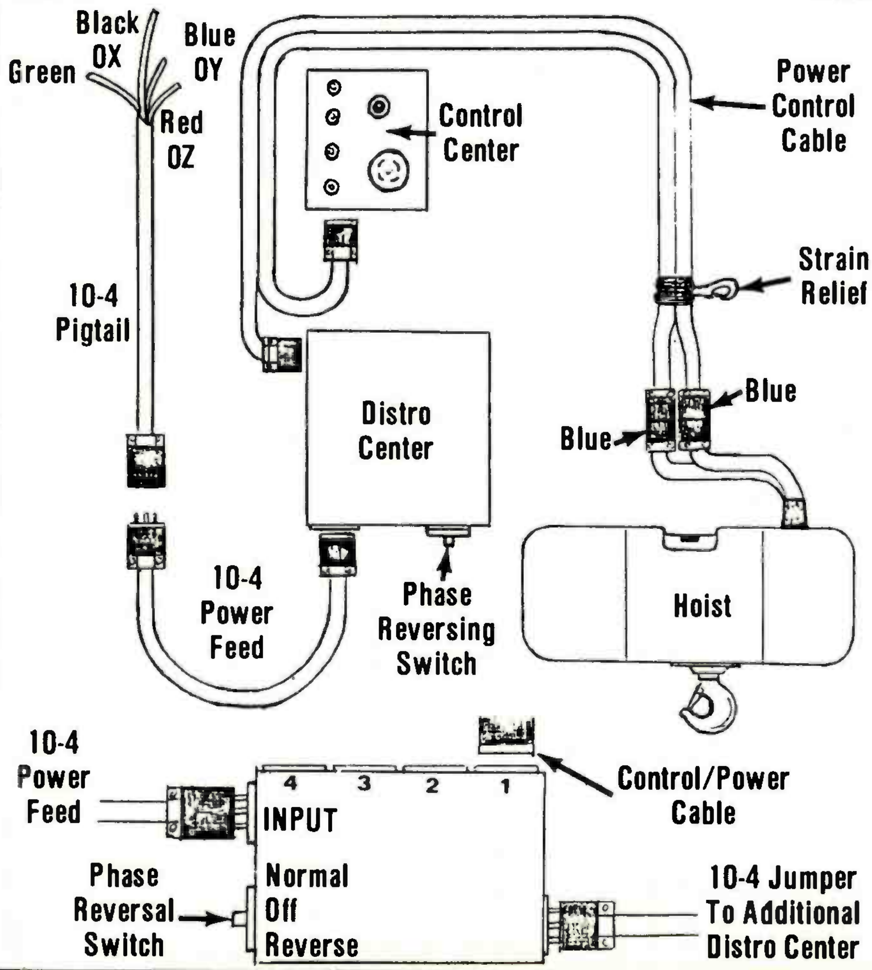
"There are now some excellent synthetic materials for use as rigging gear, including nylon," explains Paulson. "We have available to us a wide range of hooks and fitting made from many different synthetic materials — far more items than are supplied in steel. However, steel cable is still a very big part of what goes into putting a sound system into the air."

Even with laboratory-tested hardware and conservatively rated load capacities, proper operation and maintenance techniques are a critical part of using chain motors to suspend sound systems above performers and audiences. As with a chain, any technical system is only as strong as its weakest link.

"Constant, regularly scheduled maintenance is important," advises Paulson. "Many sound companies may own motors and rigging gear, but it can be expensive to keep somebody on full-time to stay on top of it all the time and assume responsibility. Lives can depend on the proper or improper use of rigging hardware."

Stage Rigging, Inc. maintains expensive liability insurance coverage, as do other reputable rigging firms and sound companies involved in hanging systems. Fortunately for the touring entertainment industry, rigging accidents are few and far between. "Ultimately, the overall responsibility rests with the act, and the show's promoter," says Paulson. "We like to think that touring acts use our services because they know they are getting well-maintained rigging gear set up by knowledgeable, competent technicians. It helps to minimize their risk."

Hanging an arena sound system is a bit more complicated than hanging a picture or chandelier. Before attempting a first-time involvement in these highly specialized techniques, sound company personnel would do well to thoroughly research the subject. □□□



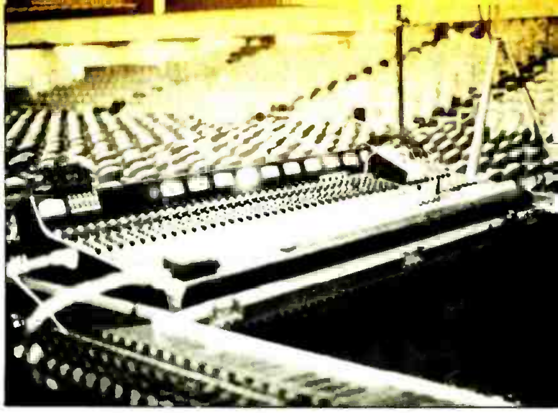


Figure 8: Two Midas 32-input consoles provided a total of 64 input channels. The console at left was used as a submixer for drum channels, and to service the opening act.

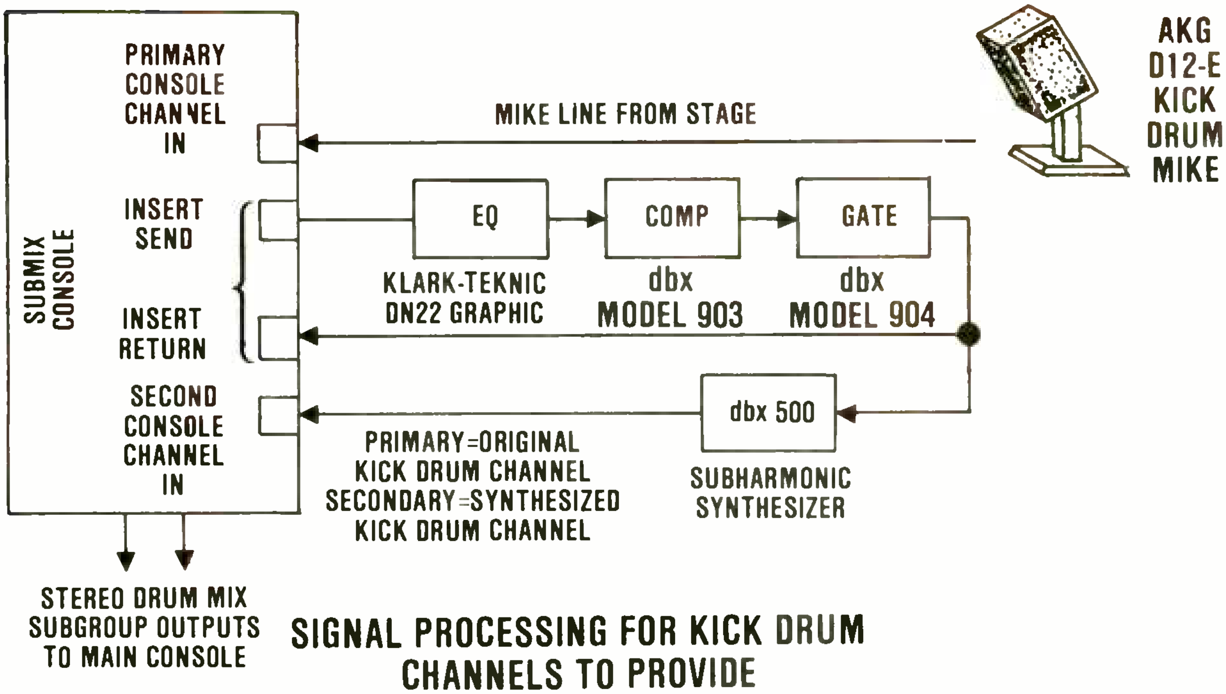
A Midas 32-by-8 main house console equipped with Pro-4 modules and auxiliary effects busses was supplemented with a Midas 32/8 submixer to provide a total of 64 inputs (Figure 8).

"The desk that I am using as a submixer has Pro-5 modules," notes Patterson. "The basic difference there is the parametric mid-band. I am taking three stereo pairs from it into the main board: two pairs give me the tom-tom and overhead drum mike submixes; then a stereo pair from the master of the submix board is injected into the main board."

Main left and right output signals from the Midas console pass through Klark-Teknik Model DN27A third-octave graphic equalizers, and then passed through dbx Model 165 Over-easy compressor/limiters. Model SFX-MP electronic crossovers are manufactured by Tasco London. "These crossovers were put together and modified specifically for the Harwell System," explains Patterson. "They can be used either as a mono, six-way crossover, or as three mono, two-ways. Obviously, they work well for multiple-mix stage monitoring applications as well as for the main drive crossover in the house system."

Additional dbx Model 162 stereo compressor-limiters were provided for use on the midrange and high-frequency bands of the crossover outputs. A Klark-Teknik Model DN60 real-time analyzer provided graphic visualization of the system's frequency response in the room. The system's main drive components were housed in one of three matching electronics racks (Figure 7).

Additional signal-processing and special effects devices included a Roland SRE-555 Chorus Echo, a Lexicon Prime Time, Eventide Harmonizer, and a Lexicon Model 224X digital reverb. An additional rack housing dbx 900 Series compressors and noise gates was used for channel-insert applications on vocal, drum and



SIGNAL PROCESSING FOR KICK DRUM CHANNELS TO PROVIDE EXTENDED LOW-FREQUENCY IMPACT

instrument microphones.

Heavy-Metal Kick Drum

To give Judas Priest's heavy-metal fans the full benefit of the subsonic impact of a bass drum unleashed on an unsuspecting arena, Patterson uses an interesting set of processing devices, as shown in the accompanying block diagram. The channel-inserted chain of devices includes a Klark-Teknik Model DN22 graphic equalizer, a dbx Model 903 compressor module, and a 904 noise gate module. The processed signal is then passed through a dbx Subharmonic Synthesizer ("Boom Box"), returned to the submix console channel, and eventually finds its way into the main console.

"The really deep, heavy bass drum

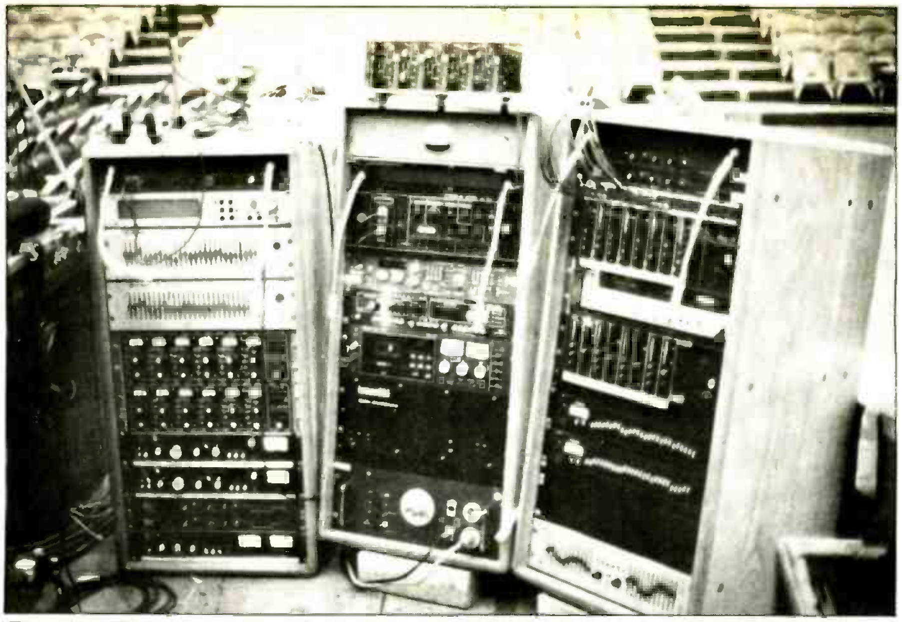


Figure 9: Three matching electronics racks housed a variety of signal processing gear. The main drive rack, at left, contains a Klark-Teknik DN60 real-time analyzer and DN27A graphic equalizers, two dbx 165 compressor-limiters, and Tasco SFX6-MP electronic crossovers.

sound is an important part of Judas Priest's concert sound," notes Patterson. "Just putting the mike up there by itself doesn't really get me the desired effect without low-end feedback. It may seem like a lot of trouble just for the sound of the kick, but it works."

Microphones and Inputs

"With the high stage-level sound that we are dealing with, the noise gates are important, particularly on the drum microphones," explains Patterson. "This show would be much more difficult to mix without them."

Two bass drums, the rack and floor toms, and a snare top and bottom mike are all gated with dbx Model 904s. A double-neck bass (four- and eight-string) took up two inputs, and other devices included Moog Taurus bass pedals and stereo guitar cabinet microphones, as well as special echo effects guitar pedalboard. Nady wireless units were used for the electric bass and guitars.

"I keep half of the submix console —16 inputs — open for use by the opening act on the show," comments Patterson. "That way, their soundman can set up his mix on that desk, and it doesn't affect what I have set up on my inputs. It works out well for both of us."

The crew chief finds that setting up the house mix position in an arena such as the one in Long Beach requires some forethought. "If you put yourself too far back in the reverberant field, you will have a tough time mixing the show," he offers. "However, being too close to the stage will actually put you underneath the throw of the flying system, so you don't get an accurate idea of what is happening behind you in the room. I use some gear down on the deck, at stage level, but I like to be able to hear the entire system, not just *part* of it. Generally, about a hundred feet back in an arena this size will work well for me."

Stage Monitoring

Stage monitor sound for Judas Priest is handled by Tasco's Martin Rowe. An active splitter system feeds a Midas stage console equipped with Pro-4 EQ modules. Six Klark-Teknik graphic equalizers and six Yamaha Q1027 equalizers provides graphic

Figure 10: The monitor mix position, overseen by Martin Rowe. A 12-mix Midas stage console is augmented with Yamaha Q1027 and Klark-Teknik DN27A graphic equalizers.

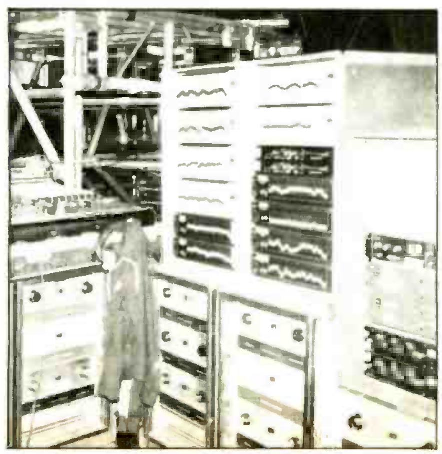


Figure 11: A variety of floor slant monitor speakers were provided, featuring JBL, Cetec-Gauss and Electro-Voice components.

controls for each of the 12 outputs (Figure 10).

A variety of slant floor monitors were provided, with components by Cetec-Gauss, Electro-Voice and JBL. One pair of slants targeted for the guitarist featured Electro-Voice EVM Series 12-inch loudspeakers, the same as used in the guitar amplifier cabinets (Figure 11).

The massive side-fill monitor stacks included Martin bass bins, Community Light & Sound mid-bass flares, and JBL horns and lens units (Figure 12).

Subjective Comments

While walking throughout the seating areas of this cavernous venue, I noticed a fairly high reverberance factor when the room was empty. 120 dB bursts of pink noise used to adjust the system's crossover levels were quite irritating in the vacant hall. Playback music was difficult to listen to.

Figure 12: Massive sidefill monitor stacks included Martin bass bins, Community Light & Sound mid-bass flares, and JBL horns and lenses.

However, with live program material in the sold-out arena at showtime, it was a different story. At the console and throughout the floor seating area, full-frequency sound was presented with a very clear mix. The subwoofer section enhanced the performance with a tremendous boost in the lower register.

In the uppermost regions of the arena, the system had a very warm characteristic, with mid-bass frequencies overshadowing the higher frequencies. This situation was reversed when one listened to the system on-axis with the high-frequency cabinet arrays from the seating areas level with the hanging components.

Variances in predominant frequencies were seemingly due to the physical construction of the arena itself, as opposed to this particular system and its method of deployment. A centrally located single point-source cluster would have its obvious advantage in such a setting. However, flying concert-sound systems such as this one have made possible the presentation of high-level contemporary musical events, with a maximum number of seats available to the public due to the clearer lines of sight than if the same amount of loudspeaker enclosures were stacked on the stage. ■■■

PERFORMANCE ASSESSMENTS OF STUDIO MICROPHONES

Additional Stereo Evaluations of 15 Models at the University of Iowa School of Music: Part Two

by Lowell Cross, Professor of Music and Director of Recording Studios

The first part of this article, published in the December 1984 issue, concluded with a description of stereo monitoring conditions prevailing throughout our listening sessions, which made use of JBL 4310 and 4320 monitor loudspeakers, as well as AKG K-141 stereo headphones.

Before proceeding with the actual evaluations of the various stereo techniques and the individual microphones themselves, I would like to acknowledge again the presence of subjective factors in this process. One example is the problem of separating *stereo* qualities from *microphone* qualities. For this reason, our 20 participating auditioners in this round of evaluations were asked to concentrate upon their perceptions of the qualities of the individual microphones themselves. The evaluations of the stereo techniques that now follow are those of the author.

Stereo Techniques

The real winner here was the surround-sound representation of the musical performance in the recital hall, recorded in the single-point Calrec Soundfield technique. The Ambisonic sound image as reproduced over four discrete, matched channels was spectacular in effect, yet completely natural. This mode of reproduction via four loudspeakers was accomplished by matrixing the X, W, and Y horizontal B-Format signals within the Neve console, through proper selections of channel assignment, level, and signal polarity. In the absence of a B-Format surround-sound decoder, it is possible to use the following arrangement with a conventional console to convert the three-channel horizontal B-Format signals into appropriate loudspeaker feeds, in a process that is very similar to deriving left and right signals from MS recordings.

Having selected four appropriate bus outputs to represent the four-channel surround-sound sends — for example, #1 as left-front, #2 as right-front, #3 as left-rear, and #4 as right-rear — the B-Format W signal, representing an omnidirectional microphone, is connected to an input channel and routed to all four busses, as shown in the accompanying diagram. The X signal, representing a forward-facing figure-of-eight, is connected to two console inputs, one of which is routed equally to bus #1 and #2, and the other to bus #3 and #4.

To preserve directionality, the phase of the input to the second "X-channel" must be reversed relative to the first; the same concept can be achieved via a phase-reverse patch cord if the console does not offer phase-reversal switching. In the same way the Y signal, representing a side-facing figure-of-eight, is connected to two more inputs — with phase reversal on the second channel — and routed equally to busses #1 and #3, and #2 and #4. By balancing the channel faders carrying the X and Y signals to be 6 dB down in level relative to the W or omni signal, a faithful reproduction of an Ambisonic horizontal sound field can be achieved.

The remaining, two-channel choices, listed in order of my personal preference, simply could not be expected to approach the aural excitement of Ambisonic surround sound. I hope that the availability of the Soundfield system, despite its considerable expense, will motivate a resurgence of interest in Ambisonic recording and reproduction among audio engineers and the general public.

. . . continued overleaf —

Photography by James J. March, Linda Bourassa and Steven Sergeant

SYNCON series B

Shopping for a Desk these days can be a chancy business. Limited budgets all too often mean limited equipment, limited performance – and limited potential for future expansion. It's a story that many smaller studios know to their cost.

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MICROPHONE ASSESSMENTS

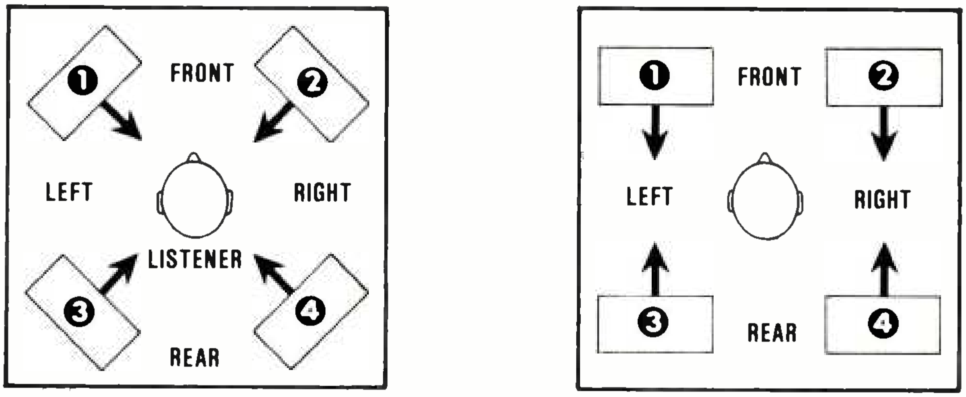
1. Slightly spaced or near-coincident cardioid pair, including ORTF.

It is my firm belief that if one sets out to make a stereophonic recording, clarity of musical texture, precise stereo imaging (including depth perspective), and spaciousness (even ambient "warmth") are paramount considerations, while monaural compatibility assumes a position of only secondary interest. If one wants to make a monaural recording, let him (or her) do so; my own preferences are for true stereophonic reproduction (or if available, surround sound), providing that the music itself remains the end of all such means. There is no doubt, of course, that stereo recordings destined for disk mastering, broadcasting, etc., should avoid any exaggerated random phase differences between channels: Not only is monaural compatibility seriously degraded, but also (and more importantly) the stereo qualities of the recording are severely compromised.

Since we hear with two ears that are slightly spaced, it follows that accurately recorded time-difference cues between channels can greatly enhance stereo perspective. My experience has been that a carefully chosen near-coincident microphone technique will satisfy all of the goals of proper stereo reproduction more successfully than the strictly coincident ones, which project the stereo illusion primarily from intensity cues. (The obvious exceptions are the coincident Ambisonic and "perisonic" techniques, which offer very convincing illusions of depth perspective.) In summary, a near-coincident recording appears to have more depth than one made with a coincident technique; a coincident recording appears to be two-dimensional by comparison (that is, existing on a line between the two loudspeakers rather than occupying the space between and behind the loudspeakers). These perceptions were evident throughout all of the listening tests, no matter which near-coincident pairs were being auditioned at a given moment.

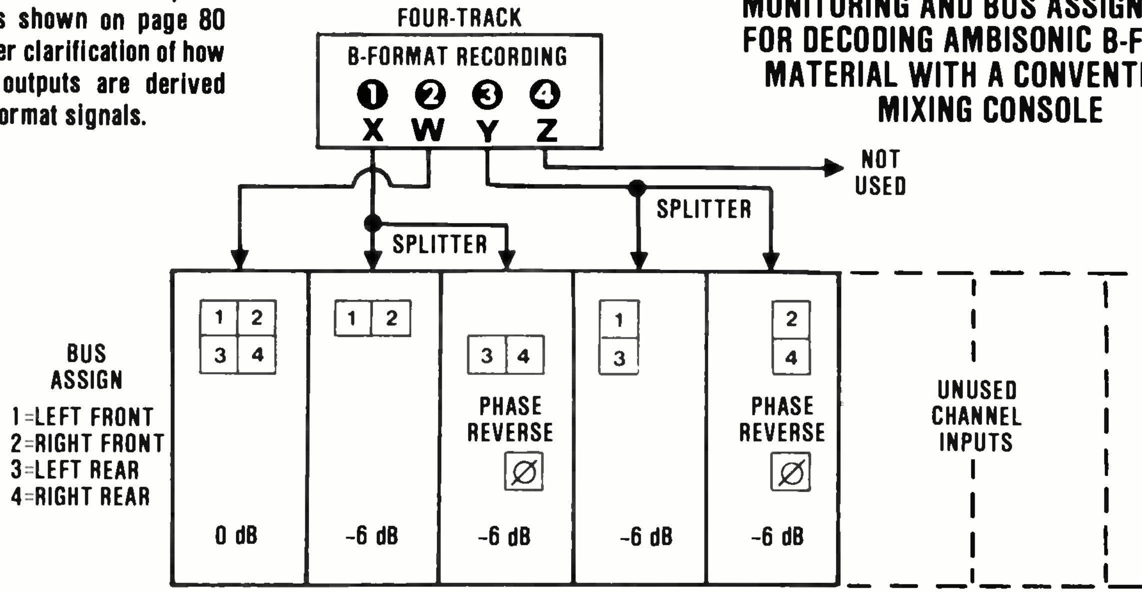
An RTW Model 1260 stereo correlation meter, with memory, was used to monitor the phase conditions between channels in the various stereo techniques. (Readings in the range of 0 to +1 indicate "proper" stereophony; excessive out-of-phase information is present when readings move consistently into the -1 to 0 range.) The worst-case reading for our near-coincident pairs was -0.1, remaining in the 0 to +0.9 range most of the time (ORTF range: -0.1 to +0.8).[21] These are absolutely correct readings for proper two-channel stereo.

All of my criteria for good stereo reproduction were satisfied by the near-coincident microphone placements. Furthermore, near-coincidence



ALTERNATE MONITORING LAYOUTS FOR AMBISONIC REPLAY

In addition, refer to the polar diagrams shown on page 80 for further clarification of how monitor outputs are derived from B-Format signals.



MONITORING AND BUS ASSIGNMENTS FOR DECODING AMBISONIC B-FORMAT MATERIAL WITH A CONVENTIONAL MIXING CONSOLE

LF: $W + \frac{1}{2}X + \frac{1}{2}Y$
RF: $W + \frac{1}{2}X - \frac{1}{2}Y$
LR: $W - \frac{1}{2}X + \frac{1}{2}Y$
RR: $W - \frac{1}{2}X - \frac{1}{2}Y$

(0.5 factor equals 6 dB; ± factor equals 180° phase reversal)

The unique Calrec Soundfield Microphone MK IV

THE FEATURES

- ☐ Totally steerable both vertically and horizontally in **POST PRODUCTION**, off tape.
- ☐ Variable stereo capsule angle and polar pattern in **POST PRODUCTION**, off tape.
- ☐ Variable zoom, both forwards and backwards, in **POST PRODUCTION**, off tape.
- ☐ The only truly coincident stereo microphone in the world.
- ☐ Very low noise performance for the digital era.
- ☐ Separate outputs for Ambisonic surround sound.
- ☐ Level frequency response both on- and off-axis.
- ☐ The most accurate polar patterns in the world.
- ☐ Maximum input for less than 0.5% THD 140 dB SPL.

THE FACTS

The spherical three-dimensional pick-up of the Soundfield Microphone is such that the phase errors introduced by the capsule spacing in normal microphones are effectively eliminated and the resulting stereo output of the control unit has virtually perfect image placement at all frequencies. The differing frequency responses of the pressure and gradient components of the signal are also corrected, thus giving an equally flat response to both on- and off-axis sounds. These two facts make it possible, for the first time ever, not only to generate exactly signals envisaged by A. D. Blumlein when he first proposed the M/S system, but to extend them into three dimensions.

This spherical representation of the original soundfield allows a stereo signal to be extracted pointing in any direction and of any first order polar diagram. The angle between the two microphones may be varied between 0° (mono) and 180° and the apparent proximity to the original sound sources may also be adjusted.

The control unit also provides a four-channel output signal, known as "B format," which exactly represents the first order characteristics of the soundfield. Recordings stored in "B format" allow the POST SESSION use of all the aforementioned controls. The advantage of being able to set such critical parameters as image width, direction of point and tilt, polar patterns and distance — all in the peace and quiet of the dubbing studio — cannot be over-emphasised. "B format" is also the professional signal format for Ambisonic surround sound and may be encoded directly to domestic transmission and consumption formats.

For further information, please contact your local dealer, national distributor or:-

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Hebden Bridge, West Yorkshire, England.
Telephone: 0422 842159. Telex: 51311 Relays G.

U.S.A. & CANADA
Audio and Design, Inc., P.O. Box 786, Bremerton, WA 98310
Telephone: 206275 5009 Telex: 152426 ADR USA.

For additional information circle #51

is quite flexible, easy to set up, and generally less expensive than techniques requiring costly stereo microphones. Two Neumann KM84s or two AKG C460Bs, which can also be used for coincident XY stereo, panpot stereo, etc., are far less expensive than either one SM69fet or one C34.

2. Binaural dummy head (Kunstkopf).

I preferred the stereo image produced by the re-engineered KU81 dummy head to that of the coincident methods, for the very same reasons as those just given above. Here is another near-coincident system, using omnidirectional pressure transducers that take on certain directional properties owing to the head construction. In fact, the KU81 may indeed approach an "ideal" stereo technique — but it has at least three significant drawbacks: expense (approximately $2,500); lack of flexibility for other uses of the microphone transducers; and a real problem with physical appearance. It is unlikely that *Kunstkopstereofonie* will ever find much favor in "live" recording, especially of stage events where the "head" can be seen by the audience and/or the performers. The range registered on the correlation meter for the KU81 was -0.1 to +0.8; a quite acceptable figure for good stereo.

3. Coincident MS, matrixed to left and right channels.

With good equipment, proper room acoustics and microphone placement, and an understanding on the part of engineers of matrix techniques, MS recording can yield predictable and generally satisfying results. One investigator refers to a "lack of intimacy" in MS recordings.[22] I prefer MS to "normal" coincident XY because the phenomenon of "center buildup" is minimized. The tendency for nonmatrixed XY recordings to emphasize the center of the stereo sound field, owing to the overlap of the cardioid patterns, becomes very noticeable when the two channels are summed to a single monaural channel. In-phase mixing of the left and right channels of recordings which began with *M* (middle) and *S* (side) components yields only the *M* component back again; *S* drops out and there is no center over-emphasis remaining, only an excellent mono resultant.

The MS technique is an especially attractive method for use with solo instruments and voices in multichannel recording; much more spaciousness is heard "around" the soloists than ever available from panned mono channels. Stereo microphones such as the AKG C34 and C422, Neumann SM69fet and USM69, and Schoeps CMTS501U are very flexible: each has multiple polar patterns per capsule; the capability for rotating one of the capsules through 180 degrees or more; and applications to both the XY and MS techniques. These microphones are expensive, however, and a successful MS technique is invariably dependent upon good acoustical surroundings. The correlation meter range for our MS recordings (matrixed to left and right) was 0 to +1; an appropriate figure for stereo and indicative of the high degree of monaural compatibility available from this technique.[23]

Photo detail of mike arrays for second evaluation session (*above*), and third session (*top*).

4 and 5: Coincident cardioid XY and coincident bidirectional XY (Blumlein).

Coincident cardioid XY and bidirectional XY rate about even in this ordering. Cardioid XY microphones exhibit the "center buildup" problem just mentioned; the Blumlein method minimizes this effect but picks up more reverberation. The latter is therefore more dependent upon good room sound. While the cardioid XY technique may be "safer" in many reverberant environments, we should not forget that the figure-of-eight patterns of microphones required for the Blumlein technique are usually more frequency-independent than their cardioid counterparts. Correlation meter ranges were within +0.4 to +1 for both XY techniques, indicating the presence of more in-phase information between channels than found in any of the other methods.

6 and 7: Widely-spaced omnidirectional or boundary layer microphones.

Likewise, these two techniques rate about even, since they both present real problems of stereo imaging and monaural compatibility, at least in the opinion of this author. (See also Bruce Bartlett's remarks quoted in Part One, published in the December issue; reference #10.) *Closely*-spaced omnis and PZMs offer little, if any, stereo separation or imaging. However, it is possible, by the ingenious use of large boundary surfaces, to emulate the "classical" coincident and nearcoincident techniques with PZMs.[24] But one must really grapple with the problems of stereo recording using omnidirectional microphones. The use of a three- or even five-microphone, equally-spaced omni technique, in which a good center-fill or monaural-only pickup is present, is one approach.

The three-microphone method (left, center, right) was in widespread use in the late Fifties and early Sixties, when record companies were selling mono and stereo versions of the same release; three-channel half-inch mastering recorders were built especially for this purpose by Ampex and other manufacturers. Nevertheless, an equal or nearly-equal mix of widely spaced microphones of *any* configuration, to achieve a two-channel result, will still present phase-cancellation problems.

As noted earlier, Bruel and Kjaer is committed exclusively to the manufacturing of recording. The booklets that accompany the 4000 Series contain application notes and drawings that relate only to monaural (single-microphone) recording and broadcasting — no mention of stereo can be found.

Consequently, I decided to contact the person in charge of the 4000 Series B&K, Henning Moller, to seek his views on the proper stereo technique for 4007s under the conditions prevailing for this evaluation. On two occasions, he suggested that I try a separation of "four to five *meters*" [italics mine].[25] He also advised that I should listen to the Denon phonodisk No. OF-7013-ND (a digitally-mastered recording of Beethoven's *Symphony*

. . . continued overleaf —

For additional information circle #52

MICROPHONE ASSESSMENTS

No. 5, performed by Staatskapelle Berlin, Otmar Suitner, conductor), since the overall stereo pickup in this recording was from widely-spaced B&K 4000-Series omni microphones. (B&K has sent notices about this recording to persons on its mailing list.)

I have auditioned this disk and can confirm that it is a well-engineered recording. But I should report that a photograph of the recording session, reproduced in the program notes, reveals several "spot" microphones (U87s?) distributed among the various instrumental choirs of the orchestra. Therefore, the recording does not enable the listener to listen to widely-spaced B&K omni microphones in isolation, or as the sole contributing factors to the stereophony of the recording.

A four- to five-meter (13- to 16-foot) separation proved to be unacceptably wide for our recording, so the 4007 microphones were moved to within 2.8m (9 feet) of each other. Even this distance yielded a -0.7 to +0.7 range of readings on the phase correlation meter, indicating the presence of more random-phase components than are generally considered tolerable for accurate stereo reproduction.

Bruce Bartlett of Crown International suggested six feet as a "typical" spacing for floor-mounted PZMs.[26] Again, the distance was found to be too wide, so the separation was adjusted to four feet, resulting in a -0.4 to +0.6 range of correlation readings. The stereo imaging obtained from either the 4007 pair or the PZM pair can best be described as "amorphous."

At this point, I would like to return the favor to Mr. Moller (and his colleagues at B&K) by offering some thoughtful advice of my own: The company should consider marketing its truly excellent pressure transducers in a dummy head, so that wide spacing is not required for stereo applications. Equalization characteristics for free-field, diffuse-field, or "in-between" uses could even be made switchable. While I am aware that the installation of two 4000-Series transducers in an artificial head might compromise their phase and frequency response characteristics, at least a *choice* would be available to those of us who are concerned with the coherence of the stereo image.

TABLE 3: STEREO MICROPHONE TECHNIQUES UTILIZED DURING CURRENT EVALUATIONS

Stereo Technique*	Microphones	Evaluation Number (2) = May 9, 1984 (3) = June 13, 1984
Coincident MS, Matrixed to Left and Right Channels	AKG C34	(2)
	Calrec MkIV Soundfield**	(2)
	Neumann SM69fet	(2)
	Neumann SM69 (tube)	(3)
Coincident Cardioid XY	AKG 460B	(2)
	Calrec CM2050C	(2)
	Calrec MkIV Soundfield**	(2)
	Neumann KMF4	(2)
	Schoeps CMC54U	(2)
	Milab XY-82	(3)
Coincident Figure-of-Eight (Bidirectional) or Blumlein XY	Calrec MkIV Soundfield**	(2)
Slightly Spaced (1 ft., 30cm or less) Cardioid Pairs, approx. 45° toed-out angle	AKG C414EB	(2)
	Neumann TLM170	(2) *and* (3)
	Milab DC-96B	(3)
	Milab LC-25	(3)
ORTF	Schoeps MSTC54	(2)
Widely-spaced (9 feet/2.8m) Omnidirectional Pair	Bruel & Kjaer 4007	(2)
Widely-Spaced (4 feet/1.2m) PZM pair, on floor	Crown PZM-31S	(3)
Binaural Dummy Head	Neumann KU81	(2)
Single-point Ambisonic Surround-sound	Calrec MkIV Soundfield**	(2)

*Note: All microphones were located approximately 7 to 8 feet (2 to 2.5m) above the floor of the stage on a line about 4 feet (1.2m) downstage from the singer, *except* the B&K 4007s and the PZM units. The 4007s were on 6 feet (1.8m) stands, on the same line as the cluster of microphones, spaced 9 feet or 2.8m [!] apart. The PZMs were on the floor, on a line approximately 3 feet (90cm) from the singer, spaced 4 feet or 1.2m apart. See photographs of the recording sessions, and Figures 1 and 2.

**Note: In addition to recording the *stereo* output from the Soundfield system onto the Studer/telcom 15 ips 24-track tape during the May 9 evaluation, its B-Format outputs were simultaneously recorded by a four-channel Ampex ATR-100, operating with Dolby A-Type noise reduction systems (15 ips, Ampex 456 Grand Master half-inch tape). The Ampex/Dolby B-Format recording provides the recovery of all four techniques (**) shown above (Matrixed MS, Cardioid XY, Blumlein XY, and Single-point Ambisonic), as well as others beyond the scope of this project. I am indebted to Steven Sergeant, our audio engineer, for his assistance with the Soundfield B-Format recording. □□□

Microphone Evaluations

These tests were conducted in a manner that offered no deliberate advantages or disadvantages to any of the microphones. There was no intentional sequential ordering during the playback of the multichannel tape; individual models were simply announced as "no. 1," "no. 2," etc., with as much skipping around and returning to requested units as time would permit.

Following are composite ratings of the microphones, averaged from questionnaires received from the 20 participants in our second and third experiments. Listeners were asked to rank the units from 1 (least favored) to 10 (most favored); it was possible to give the same score to two or more microphones.

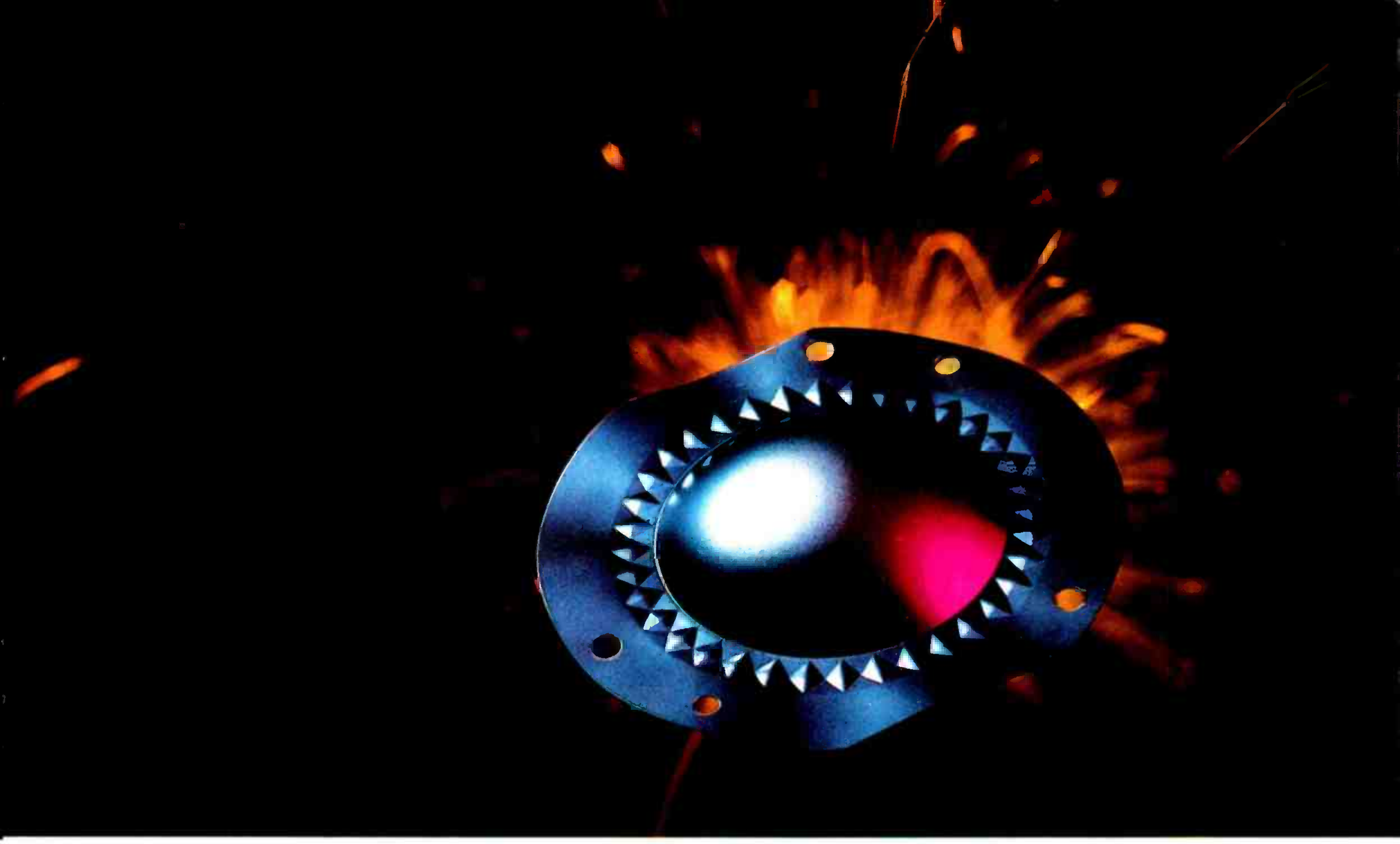
EVALUATION #2, MAY 9, 1984

Calrec MkIV Soundfield 8.02
Neumann SM69fet 7.62
Neumann TLM170 ....7.62
Schoeps MSTC54 (ORTF).......7.58
Neumann KU81 ....7.56
Schoeps CMC54U 7.34
AKG C34 ....7.22
Neumann KMF4 ....6.80
AKG C414EB 6.75
AKG C460B 6.59
Bruel & Kjaer 4007 ....6.43
Calrec CM2050C 6.04

EVALUATION #3, JUNE 13, 1984

Neumann SM69 (tube)..........7.56
Neumann TLM170 ....7.31
Milab XY-82..................7.12
Milab LC-25..................6.91
Milab DC-96B 6.86
Crown PZM-31S 6.57

By comparison, the composite



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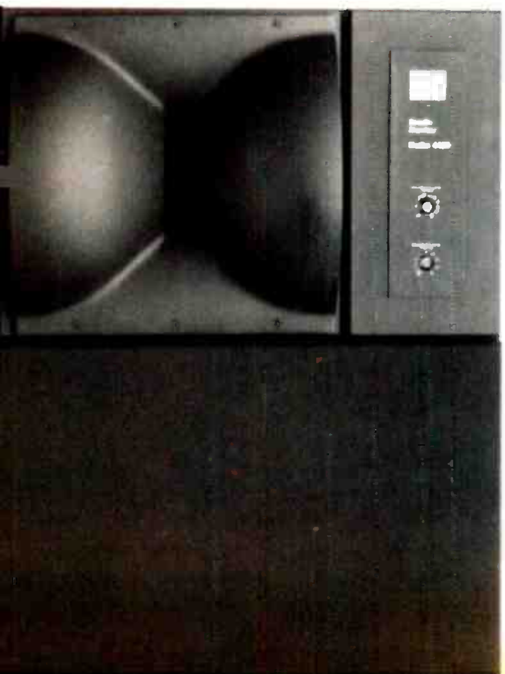

The result? A diaphragm that delivers and sustains a power and purity to high frequency response never before approached in the industry.

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For additional information circle #53

scores for the "control" microphones (SM69fet and TLM170s) were respectively 8.00 and 7.44 in the December 1983 evaluation.

As before, one must proceed with caution in interpreting the outcome, since these orderings are the results of listening tests, during which subjective factors could, and probably did, influence the responses of the participants. For example, the Schoeps microphones were preferred by most of our auditioners in the ORTF configuration (over the XY setup), even though *exactly* the same capsule types (Mk 4) were used in both cases. We asked the listeners to concentrate on their perceptions of microphone quality over stereo quality, but clearly the latter exerted a strong influence. (This preference for ORTF over cardioid XY is a minor, yet interesting, revelation.) In this same context, there is the very strong possibility that the B&K 4007s and Crown PZMs would have found more favor if each had been represented by a single microphone, placed closer to the singer, recorded in mono, and assigned to both monitoring channels during playback.

Table 4 lists the composite rankings if the results from all three evaluations, in which a total of over 50 listeners participated, are interpolated.

TABLE 4: SUBJECTIVE RANKINGS OF ALL MICROPHONES FROM THREE EVALUATION SESSIONS

Rank	Microphones	Stereo Technique	Composite Score	Evaluation Number
1.	Calrec MkIV Soundfield	Stereo output, similar to matrixed MS	8.02	2
2-3.	Neumann SM69fet	MS, matrixed	7.62	2
	Neumann TLM170	Slightly-spaced cardioid	7.62**	2
4.	Schoeps MSTC54	ORTF: 110°, 17cm cardioid	7.58	2
5-6.	Neumann SM69 tube	MS, matrixed	7.56	3
	Neumann KU81	Binaural dummy head	7.56**	2
7-8.	Neumann U89	Slightly-spaced cardioid	7.38	1
	Neumann U47 tube	Slightly-spaced cardioid	7.38**	1
9.	Schoeps CMC54U	XY coincident cardioid	7.34	2
10.	AKG C34	MS, matrixed	7.22	2
11.	Neumann U87	Slightly-spaced cardioid	7.13	1
12.	Milab XY-82	XY coincident cardioid	7.12	3
13.	Milab LC-25	Slightly-spaced cardioid	6.91	3
14.	Milab DC-96B	Slightly-spaced cardioid	6.86	3
15.	Neumann KMF4	XY coincident cardioid	6.80	2
16.	AKG C414EB	Slightly-spaced cardioid	6.75	2
17.	Neumann KM84	Slightly-spaced cardioid	6.72	1
18.	AKG C460B	XY Coincident cardioid	6.59	2
19.	Crown PZM-31S	PZms, on floor (4 feet)	6.57	3
20.	AKG C452EB	Slightly-spaced cardioid	6.54	1
21.	Neumann KM86	Slightly-spaced cardioid	6.50	1
22.	Bruel & Kjaer 4007	Widely-spaced omnis (9 ft.)	6.43	2
23.	Shure SM81	Slightly-spaced cardioid	6.38	1
24.	Calrec CM2050C	XY coincident cardioid	6.04	2
25.	Electro-Voice RE15	Slightly-spaced cardioid	5.75	1
26.	RCA 77-DX	Slightly-spaced cardioid	5.38	1
27.	Sennheiser MKH405	Slightly-spaced cardioid	4.27	1

*(1) = Dec. 14, 1983; (2) = May 9, 1984; (3) = June 13, 1984.
**Indicates a Tie

CONCLUSION: A Personal Evaluation of 27 Microphones

For the second time in less than a year, I find myself advancing a set of personal opinions about a group of microphones. And again, I admit to the presence of subjective elements in this process, certainly including my own ears and modes of listening, my previous auditory experiences, and the manner in which I relate to our Control Studio A equipment.

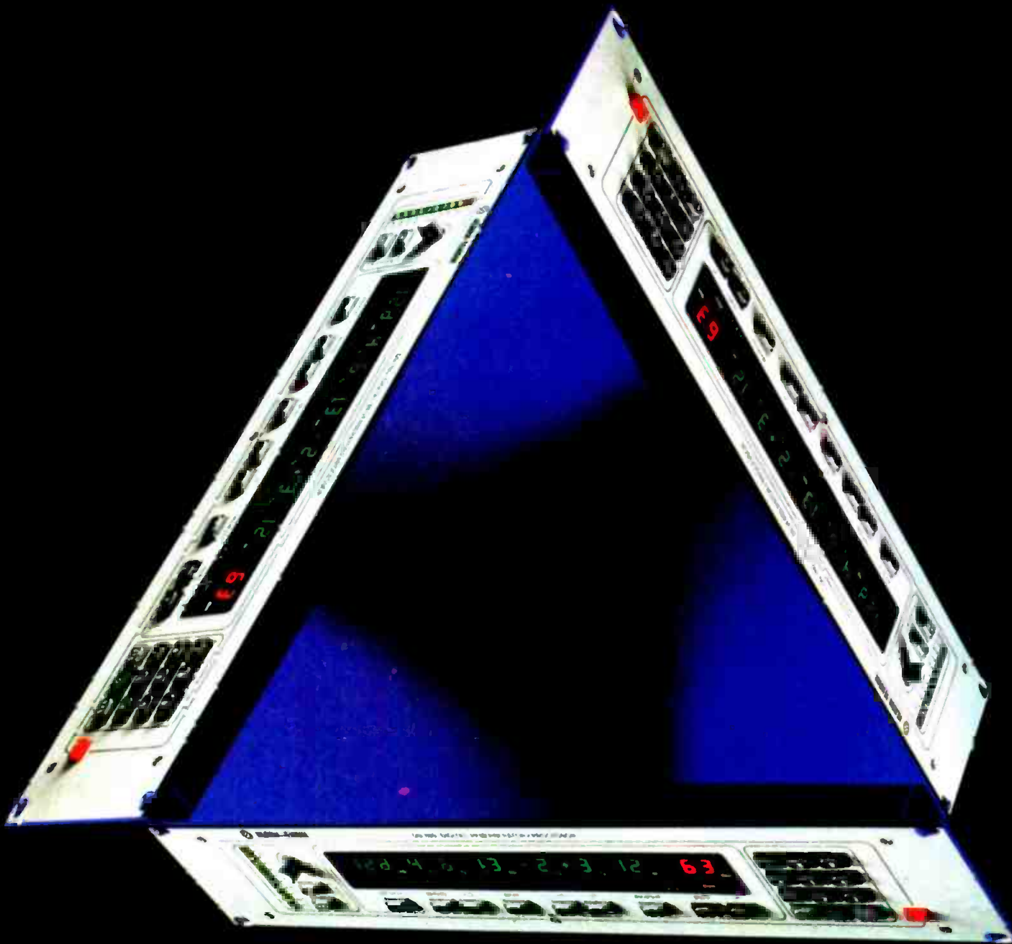
The following microphones all deserve highest praise: they are superior examples of today's "state of the art" microphone technology (9+ or 9, note ordering). *Neumann TLM170 and U89* (still my personal "favorites" in this application, in terms of sheer opulence of sound and elusive "musicality" — discussed in the earlier article), *Schoeps MSTC54* and *CMC54U, Calrec MkIV Soundfield, Neumann KU81*, and *Bruel & Kjaer 4007*.

All but the B&K Model 4007 have some kind of directional capability; the last named may therefore have fewer applications. It should be reported, however, that we made excellent use of a single Model 4007 to record solo flugelhorn in a recent multitrack jazz session; at the same time a Model 4006 (larger diameter capsule) was used on piano. As solo microphones, the B&K microphones come into their own. Both were brought into extremely close proximity to the musical instruments with absolutely no overload or frequency-emphasis problems. I regard the addition of *one* of the 4000-Series microphones to our collection as an attractive prospect; I do not anticipate buying two for stereo.

Schoeps microphones are eminently deserving of their reputation for maintaining the highest standards. Uniformity of response (with only a trace of "condenser-sound" brightness), very high level signal outputs, and transformerless design characterize these elegantly conceived units.

The Calrec Soundfield system can hardly be faulted, either in brilliance of design concept or quality of sound. It does have a bit of the "bright" quality of many condenser units (more than a TLM170, for example). Calrec should make available a balanced-

A small universe



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output option for the control unit, either with or without transformers; and more thought could have gone into designing a less imposing exterior appearance for the microphone itself. These observations cannot possibly detract from the significant development that the Soundfield system exemplifies in the historical evolution of microphones. Now all that is needed is for the price to come down.

I became quite fond of "Fritz II," the Neumann KU81, whose ears have remarkable hearing acuity. Stereo headphone listening *and* loudspeaker reproduction are indeed compatible. The overall effect is an exceptionally ingratiating aural "view" of the musical performance as perceived through Fritz' ears. This system has dispelled my previous bias against the use of pressure transducers for stereo recording.

The next category of microphones is only slightly less favored than those just discussed. Rating between 8+ and 8- in this scheme, all of the following are recommended as excellent pieces of equipment: *Neumann U87, AKG C34, Neumann SM69fet and SM69 tube, AKG C414EB, Neumann KM84 and KM86, AKG C460B, Milab XY-82, LC-25, and DC-96B*, and

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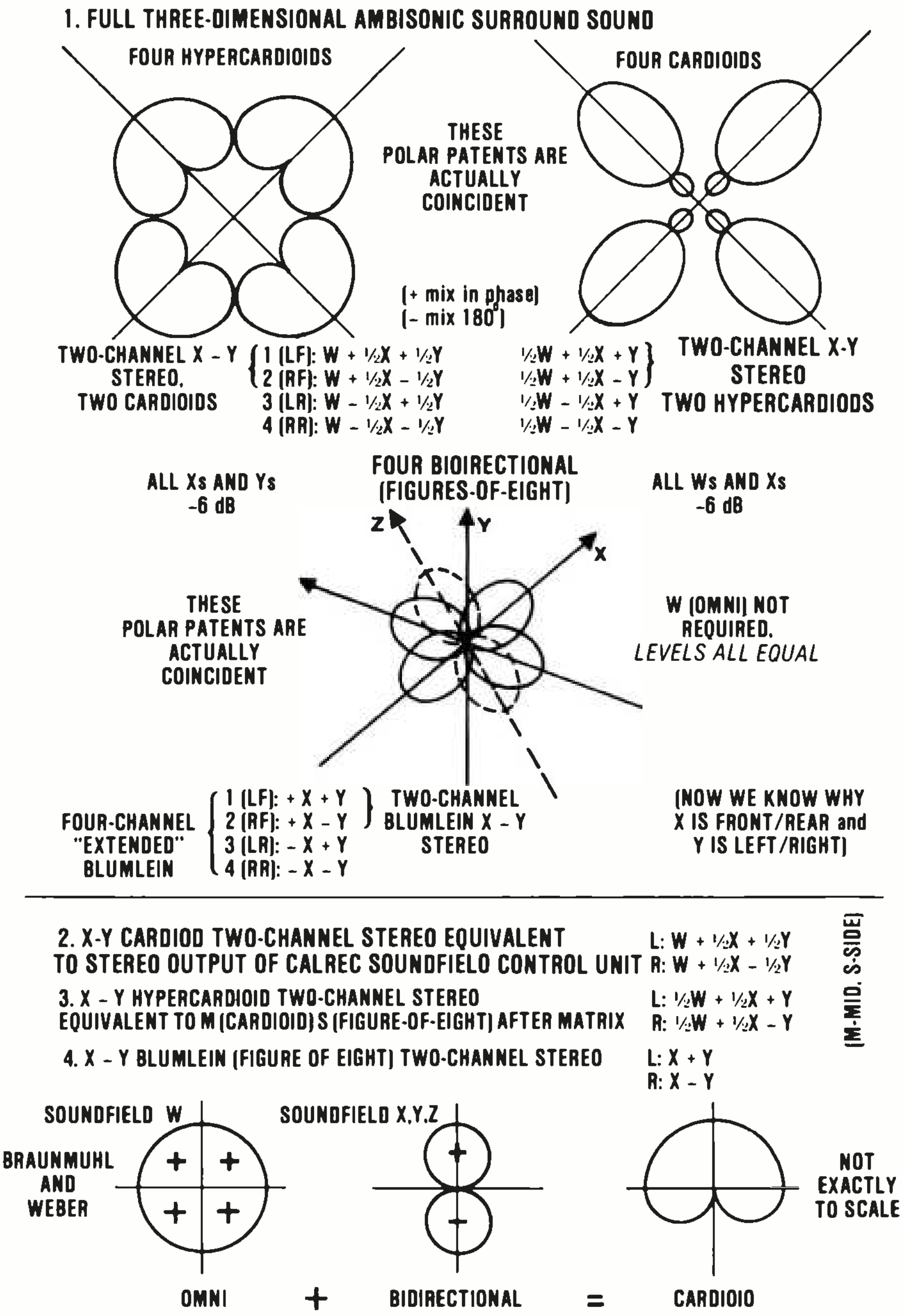

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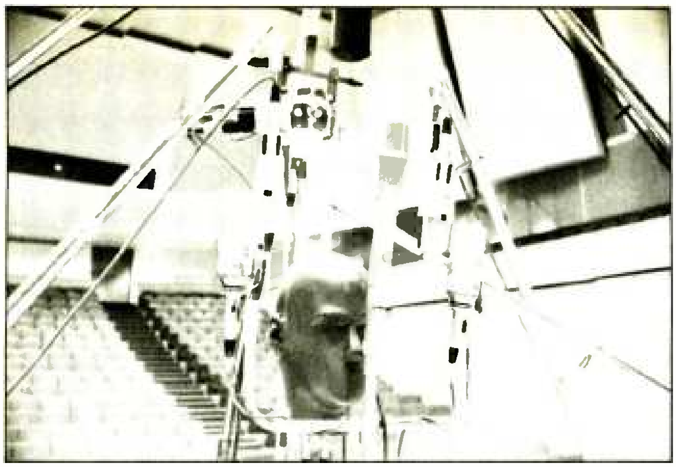


Close-up detail of various microphones utilized in the second evaluation session, May 9, 1984 (*left*), and the third evaluation session, June 13, 1984 (*right*). Annotated drawings of the mike arrays can be found on pages 100 and 102 of the December 1984 issue.

the *Neumann U47 tube.*

With the exception of the SM69 tube model, all of the Neumanns were discussed in the previous article. Since Carol Meyer's voice could not have overdriven the electronics in either of the SM69 versions, I doubt if anyone would have been able to distinguish between the two; I know that I could not. According to Gotham's Russell Hamm, writing in a *JAES* article, "The basic cause of the difference in tube and transistor sound is the weighting of the harmonic distortion components in the amplifier's overload region."[27] Under the conditions of these experiments, we essentially could hear "through" the very transparent electronics of either SM69 system — tube *or* transistor — back to the properties of the capsules themselves as they behaved in the reverberant environment of the concert hall. The transducer design of both SM69 systems results in a "condenser sound," with characteristic emphasis in the 8 to 10 kHz portion of the spectrum. I repeat my admonition to Neumann to start building "SM89" stereo microphones using the TLM170/U89 capsules.

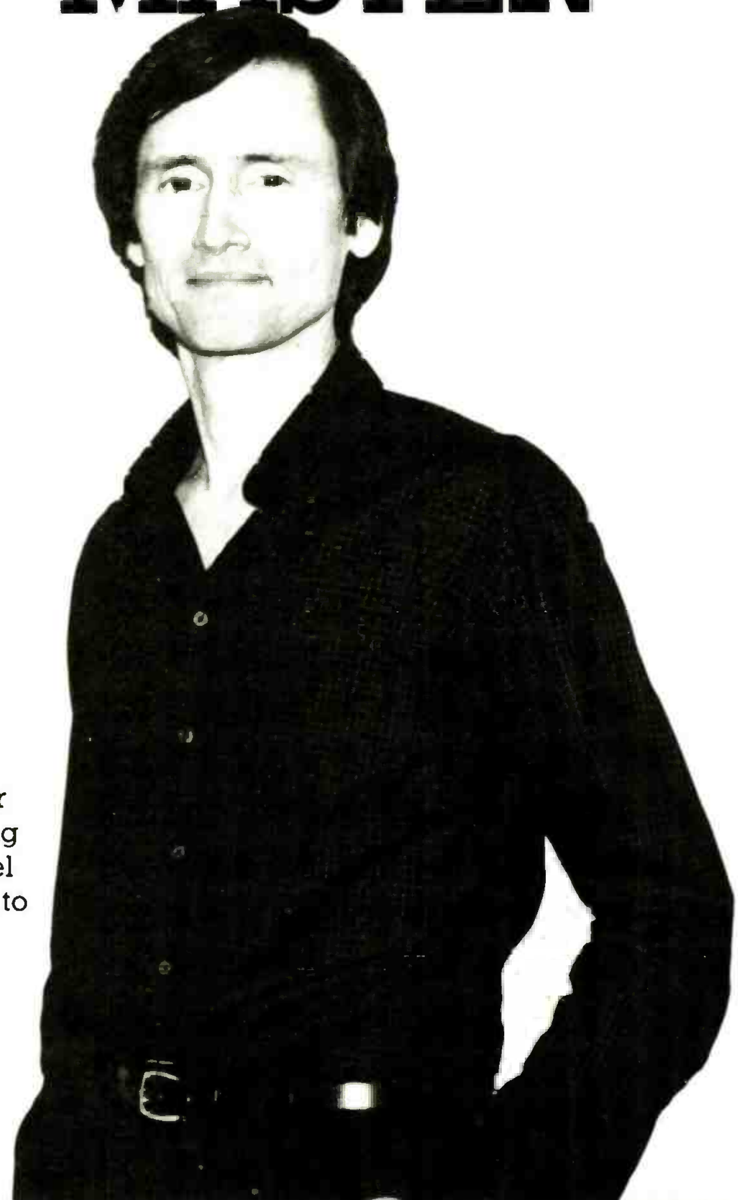
The AKG C34 is a very impressive microphone. While possessing "crispness," its tendency toward high-frequency emphasis is well under control. In fact, all of the AKG models in the second experiment appear to have undergone something of a metamorphosis in comparison to previous generations of AKGs, known for their brightness. The C414EB and C460B have only a slight high-frequency emphasis; both have audibly quieter electronics than the C450 Series. Current pricing of the AKG line is quite competitive; serious potential purchasers take note.

The three Swedish Milab microphones all made a very positive impression. Like Volvos and Saabs, they are just designed "differently" from average, everyday products and accordingly, they may be able to establish a loyal following. Each of the three is somewhat brighter than a TLM170; the LC-25 is characterized by slightly more upper midrange emphasis (3 to 8 kHz) than either the DC-96B or the XY-82. Milab's direct competitor to the TLM170, the VIP-50, will soon be imported into North America — it is a transformerless studio microphone with five polar patterns and specifications which would indicate a very wide dynamic range. I look forward to the opportunity to compare the VIP-50 and the TLM170 to each other.

The Neumann U47 tube microphone (see previous article) is in a special category all of its own, and deserves to be mentioned again if for no other reason than to acknowledge its position of honor in microphone history.

The foregoing group of Neumann,

The MASTER



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Schoeps, Calrec, Bruel & Kjaer, AKG, and Milab microphones were a pleasure to use; I considered it a privilege to become acquainted, or re-acquainted, with them all. Each one can make a significant contribution to the art and science of recording when used in a manner that is in accordance with its design criteria.

The *Neumann KMF4* achieves a 7 in this ranking. Granting that it was introduced primarily to fill the need for a concealable, miniature cardioid unit, it lacks the low-frequency performance of the more impressive studio microphones; it simply did not do justice to the lower registers of the Steinway Model D grand piano.

For different reasons, the *Calrec CM2050C* and *Crown PZM-31S* each receive a 6–. The CM2050C emphasizes the upper mid-range portion of the spectrum more than the other studio models under discussion. It also exhibits a bit of "roughness" in this region (2 to 10 kHz), detracting somewhat from the enjoyment of the vocal quality and the piano sound. PZMs have many useful applications, especially those associated with concealment when used on a surface, but floor-position stereo recording of a vocalist with piano accompaniment is not one of them. Unless the engineer is willing to go to the trouble of dealing with large movable boundary constructions, PZMs are unlikely to be chosen for stereo use over pairs of studio microphones or MS/XY stereo models. Furthermore, the sonic properties of the PZM-31S units, while certainly acceptable, have a bit of "edge" in the sound that becomes evident when they are directly compared to the finer studio microphones already mentioned.

Others rating a 6 or 6– are the *AKG C452EB* and *Shure SM81*, followed by the *Electro-Voice RE15* (5–), *Sennheiser MKH405* (3), and *RCA 77-DX* (3–); all have been discussed in the previous article. However, the 77-DX, like the U47 tube, deserves special mention because of its long and honorable career in the audio industry.

* * *

This personal form of evaluation has involved opinions: mine, and those of about 50 other collaborators. Some previously held views were reinforced, and certain prejudices were dispelled. If opinions can be changed through direct, tangible forms of experience, then one can learn in the process. These actual, practical recording sessions gave those of us involved a chance to develop *educated* opinions about selecting microphones, even if the application is limited to recording a soprano with piano accompaniment, in stereo, in a reverberant environment.

We did learn something: about microphones — and significantly, about our own recording capabilities. ■■■

Notes and References
For a listing of references #1 through #20, see the December 1984 issue of *R-e/p.*

21. For additional information about stereo correlation meters, see John Eargle, *Sound Recording*, 2nd ed., pages 70-73. New York, Van Nostrand Reinhold, 1980.

22. " . . . we note that the general agreement of the audience to distinguish the lack of intimacy of the MS system and to select the ORTF system as a best overall compromise is surprisingly high." Ceoen, *op. cit.*, page 26.

23. It should be noted that Soundfield recordings — representing a logical extension of MS sum-and-difference matrixing techniques — also mix correctly to mono with no objectionable "center buildup" effects.

24. See Lamm and Lehmann, *op. cit.* Unfortunately, time constraints prevented the loan of Mike Lamm's L²MicArray for use with the PZMs in this experiment.

25. Telephone conversations, April 2 & 3, 1984, Henning Moller and the author.

26. Bruce Bartlett, note to the author, June 7, 1984.

27. Russell O. Hamm, "Tubes vs. transistors — is there an audible difference?" *Journal of the Audio Engineering Society*, Vol. 21, No. 4 (May 1973), page 272.

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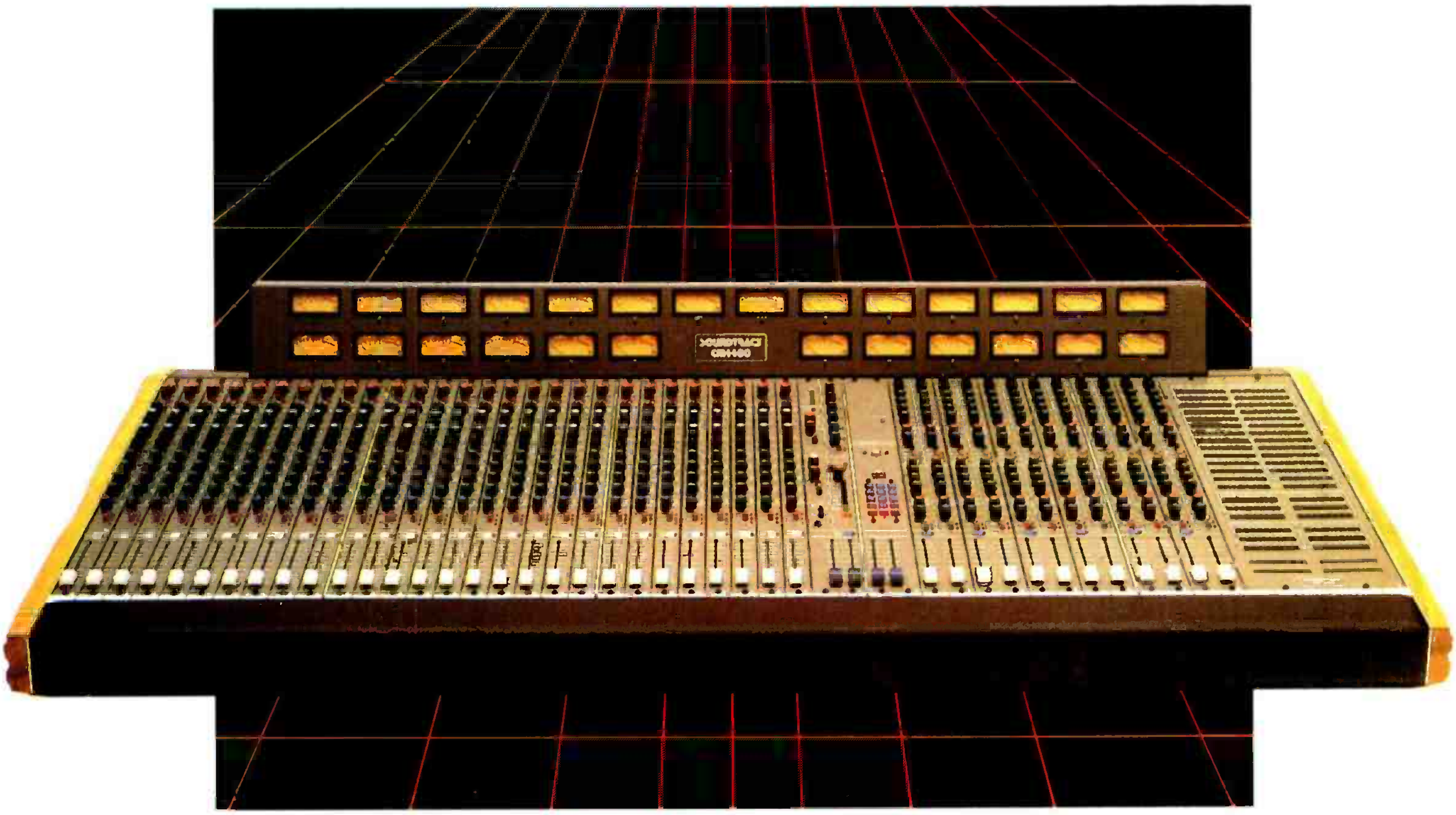

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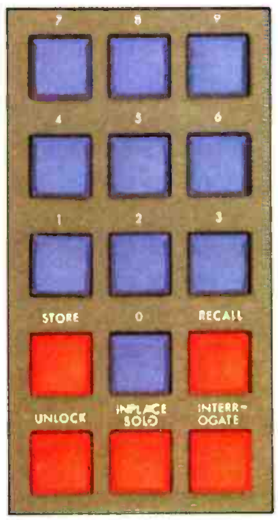

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For additional information circle #60

DISK MASTERING: THE MISUNDERSTOOD LINK

by Bernie Grundman, with Denis Degher

Possibly one of of the most difficult procedures to fathom in modern recording is what happens in the disk mastering suite, and beyond. Since many knowledgeable producers and engineers do not fully understand the disk cutting process, they take their mixes to a *trusted* mastering engineer, more or less hoping that it transfers well to this different medium, and that their work can even be enhanced by the engineer's skill. No longer are we dealing strictly with magnetic waveforms or digitized bytes on tape, but with an electro-mechanical process; a process in which pure electronic energy must be converted into electro-mechanical energy.

A good disk cutting engineer can be extremely valuable to the producer or mixer, because he is capable of offering and objective ear on a project to which the producer may have become too close. A mastering engineer, by the nature of his work, is involved in many different projects within a short period of time. The better cutting engineers have the possibility for developing true objectivity regarding *any* particular poject. He has access to the newest sounds, the latest trends, and a practical overview of the state-of-the-art in production techniques. This background and experience naturally provide a benchmark for the cutting engineer to use when he hears a tape for the first time. When a master is first played for him the mastering engineer and producer or mixer can make various judgments about the recorded material in light of what they feel will enhance the product and bring it to its full potential.

Is the disk mastering process as completely incomprehensible as many experienced industry people perceive it? Not necessarily — the basic mechanics of a modern album are based on a very simple principle. Picture it this way: The cutting head mounted on a disk cutting lathe is basically two miniature speakers, with a chisel-shaped stylus connected to the coils instead of a loudspeaker cone. When cutting a disk, we are reproducing in miniature the precise movement that the replay speaker will duplicate in a larger way. For example, if we have X Hertz on the disk in a particular part of the program materrial, X Hertz will then be transferred to the speaker cone, and that sound will be transferred into pure acoustic energy in the form of sound waves into the atmosphere.

Although the tenets are basically simple, designing a high-quality stereo cutting system with a wide frequency response and low distortion can be very difficult.

Transfer Losses

The transfer of taped program material to analog disk can have a dramatic effect on the final outcome of the product. The philosophy and problem-solving ability of the mastering engineer can either perpetuate the ebullience of the mixdown sessions, or generate newfound anxiety. During the first mastering session, a reference disk should be cut that the client can take to several familiar listening environments for evaluation. In this way it is possible to determine how the material transfers to lacquer, and how certain inherent characteristics and/or losses affect the project.

The physical process of disk cutting will introduce certain inherent losses that are intrinsic to the medium. Number one is a generational loss, and number two the loss of high-frequency content as the diameter decreases towards the center of the record. Even though all parts of the record are traveling at the same number of revolutions per minute, because of the reduced "writing velocity" at smaller diameters, there will be a loss of high-frequency material above approximately 10 kHz. The third major loss is an increase in distortion as the circumference decreases towards the inner grooves. Because the same amount of information in the groove must be packed into an ever-decreasing diameter, there will be greater signal density than in a comparable area at the outside of the record. Finally, there will be a loss of stereo separation, particularly above

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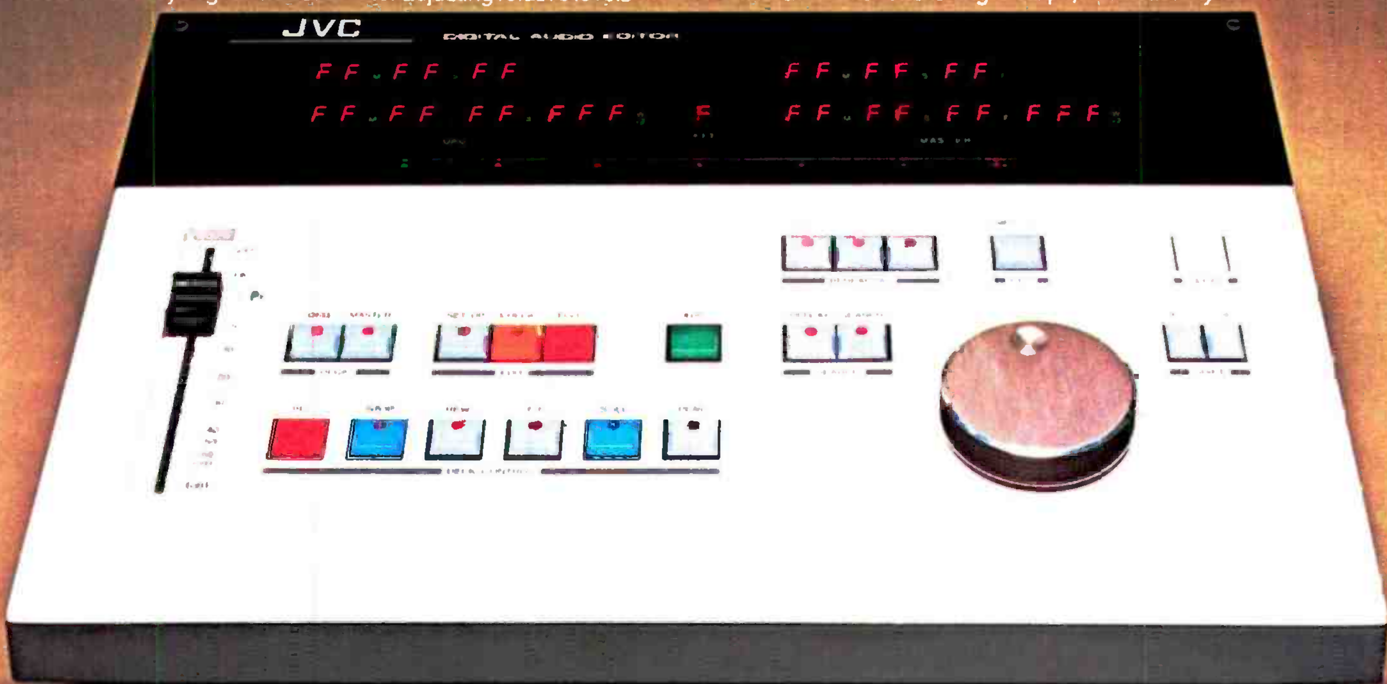

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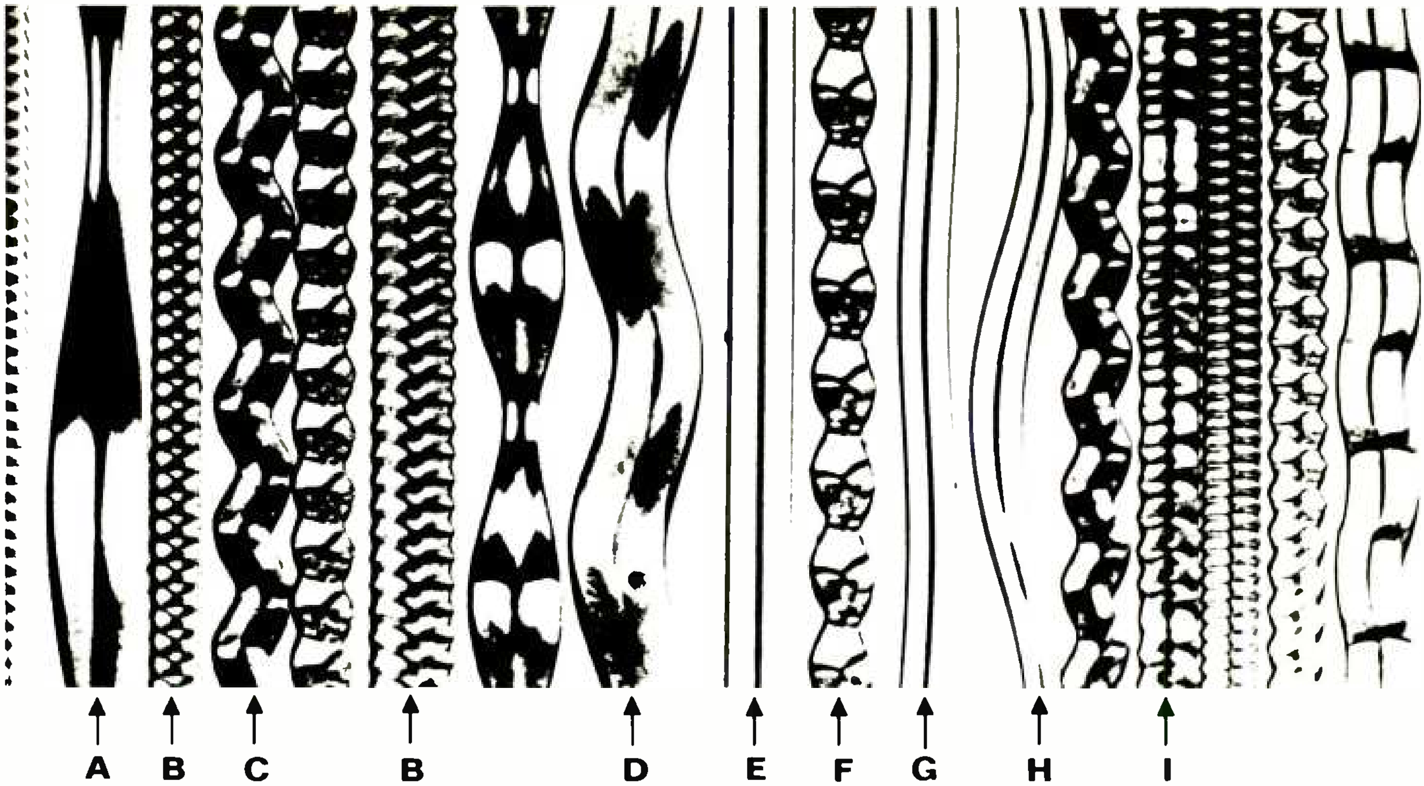


TYPICAL GROOVE APPEARANCE ON LP RECORDS

KEY: A = Out-of-phase low-frequency material; B = Two examples of high-frequency material; C = In-phase mid-frequency material; D = In-phase low-frequency material; E = No signal; F = Out-of-phase mid-frequency material; G = Low-level signal; H = Ultra-low-frequency material; I = Typical complex groove with program material.

The figure reproduced above was taken, with permission, from *Basic Disc Mastering*, by Larry Boden, and published by Swordsman Press, Inc., Woodland Hills, CA. Photography by Dave Hernandez.

15 kHz.

Any inherent losses or weaknesses transferred to the cutting lacquer will be magnified during the actual pressing process, and several others will be added. Transient losses caused by the manufacturing stage result in reduced high-end attack on most instruments, manifesting itself as a lack of clarity. Another disturbing and little-understood phenomenon is the loss of echoes and/or ambience, a general shrinkage of sound, and the introduction of boundaries to the sound. Again, this latter effect can be attributed partially to a loss of transients causing reduced definition.

Another reason for this loss is the inability of the playback stylus to reproduce some of the finer delineations in the groove. As can be seen from the accompanying photographs, higher frequencies are represented by finer groove excursions, while the lower frequencies, having a longer wavelength, are manifested by larger sweeping excursions.

Lastly, an increase in noise caused by vinyl composition and assorted other reasons can take place during the manufacturing process. (The causes for manufacturing degradation will be examined later.)

Role of the Cutting Engineer

The mastering engineer's philosophy can play a vital part in the preparation of the final product. As with the selection of the recording engineer for a particular project, a concerted effort should be made to define audio terms so that technical adjustments can be made to realize the client's aural goals. It is not always easy for a mastering engineer to know what the producer or artist is looking for, because the client usually has an aesthetic idea about how he (or she) wants the record to come across to the listener. Music can be examined in two ways: technically, and subjectively or aesthetically. If a mastering engineer is unfamiliar with the artist or project, he will usually strive for technical perfection from the purist's point of view — that is, to make the record as clean, clear, and "large" as possible. By doing so, however, the chosen approach might adversely affect the client's aural interpretation of how the final product should sound. As a result, the mastering engineer will have to back-track and go for the more "subjective" sound, although this orientation might go against the grain from a technical point of view.

Perhaps the client wants the disk to sound very "hot," a decision which in itself will introduce a certain amount of distortion. Maybe this is what the client is looking for — a hotter, more aggressive sound — and distortion *per se* is not necessarily bad. To some people, distortion can be a desired sound, and fill a certain psycho-acoustic need. Increasing the level can also give the record a more compressed or processed sound, akin to electronic limiting or compression. Certain instruments and voices lend themselves very well to signal processing, and there is certainly room in the marketplace for a wide variety of different kinds of sounds.

Since the disk-mastering suite is the last controlled link in the recording chain, it is the cutting engineer's ultimate challenge to try and realize the final degree of sound that the client is striving for to fulfill his or her artistic statement.

Preparation for the Mastering Process

Certain positive and definitive steps

should be taken before a producer or engineer enters a cutting studio. Master two-track alignment tones for the entire album project are imperative, and particularly important when moving from one remix studio to another. Another very important factor when moving between studios is correct azimuth adjustment. If certain tracks are mixed at a different studio and the azimuth is not checked, it is impossible for the mastering engineer to optimize the sound of those tracks using azimuth tones aligned to another facility. So long as we're assured of having accurate tones and azimuth settings, the main difference in subjective sound from studio to mastering suite can be attributed to the control-room monitoring.

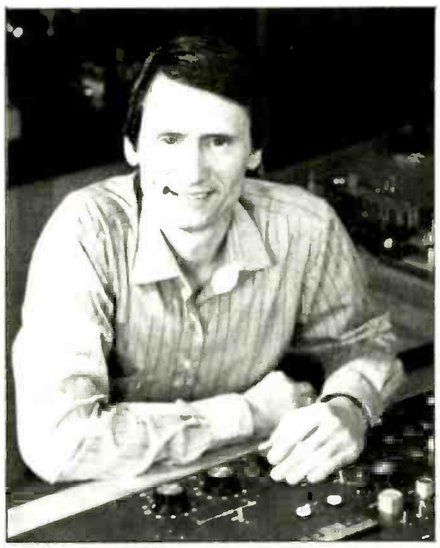
Problems can also arise when various producers and engineers are working on a project. Since there can be many variations of format, recording level, tape speed, and noise-reduction alignment, whenever possible these factors should be agreed upon in advance. Otherwise it may become necessary to transfer the entire project down one generation, so that the album sides can be cut continuously.

Certain negative factors should be avoided before the mastering stage.

The custom-designed disk mastering console at Bernie Grundman Mastering is built entirely from discrete components. Tape replay is from a rebuilt Scully deck; playback from Sony PCM-1610 and Mitsubishi X-80 two-track is also available.

One of the greatest causes of an otherwise good sounding tape transferring to disk badly is the problem of sibilance. Certain sounds — including "esses" — contain a very intense, short burst of energy; the more top end that is added to a vocal track, the more pronounced this phenomenon becomes. Since recording tape can handle an "S" sound better than vinyl, it may not be apparent in the recording studio. The problem of sibilance is a function of the replay stylus, not the inability of the lathe to cut the sound.

De-essing or sibilance correction should be performed during the mixdown session, and applied to the specific tracks requiring it. When radical de-essing of the composite mix is performed during the disk-mastering stage, it also affects all other high-frequency elements in the mix, and



— the Author —

Formerly the cutting engineer for A&M Records, Los Angeles, **Bernie Grundman** now heads his own mastering facility based in Hollywood. His numerous credits include Michael Jackson's *Thriller*, Prince's *Purple Rain*, The Jacksons' *Victory*, Steely Dan's *Aja*, Supertramp's *Breakfast in America*, Herb Alpert's *Rise*, Donna Summer's *She Works Hard for the Money*, and the majority of Windham Hill releases.

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can create a drop-out effect or pumping sound that can have an overall negative impact on the final product. Phase used to be a problematic factor in the past, but has been eradicated almost completely through the use of new computerized cutting systems that allow the lathe to cut even grossly out-of-phase material.

Manufacturing Degradations

Because of the mass-production techniques used to manufacture a modern album, there is great room for quality variation. After the master lacquer has been cut, the album goes through four more generations before an album arrive in the record store. From the master lacquer, a reverse "metal master" is made, followed by a metal "mother" and a metal "stamper" from which the vinyl is pressed. After the master lacquer has been cut, each step of the process requires a reverse image of the step before it, until the final stamper that is the image of the master lacquer. With so many generations involved, it is pretty easy to see why the final product may not sound quite as good as the master two-track tape.

Other variables in the manufacturing process can also add subtle changes to the final record, including too many mothers being taken from the metal master; too many stampers off the mother; and too many records off the stamper. Because of these limitations in the manufacturing process, it becomes necessary to remaster after approximately 75,000 records sold. (You can imagine how many master lacquers were cut for Michael Jackson's *Thriller* album.)

One of the two tandem Scully lathes, refitted with phase-locked motors, Sontec Compudisk computer control, and customized cutter-head suspension. The cutting head combines a Haeco magnet assembly with Westrex drive coils.

There are other variations that can occur in manufacturing; some pressing plants just seem to sound better than their corporate counterparts. Variations in the vinyl blend also can have its effect. Is the record company still sending the master tape for mastering, or are they sending a degraded tape copy? All these factors can enter into the mass production of the modern analog album, and can create quality control problems during various stages of a record's lifespan.

State-of-the-Art Disk Mastering

There is a great deal of variation beween one disk-cutting system and another. While there may not be much room for "hot-rodding" tape machines, the same cannot be said for the cutting system. State-of-the-art, computer-assisted lathes make it possible to cut disks that are virtually *too* hot to play back on any home system. Once the depth and width of groove are programmed into the computer, the lathe will usually maintain these parameters regardless of level, although sometimes at the expense of replay time per side. In this respect, today's cutting systems have far outstripped the systems of yesterday. If all steps are done correctly, many would argue that the analog disk still represents the highest quality audio available to the consumer.

COMPACT DISC MASTERING

Today, many disk mastering studios are becoming involved in the production of submasters for cassette duplication and Compact Disc release. Oftentimes the submasters now being employed for cassette duplication are digital tape copies, to help reduce the inherent noise in audio cassettes. For the most part cassette duplication submasters are straightforward copies of the master tape used for album manufacturing, with little EQ or level change. Submasters for Compact Disc manufacturing, on the other hand, must be dealt with differently, since they have special requirements.

The CD submaster must be made with continuous program and uninterrupted SMPTE timecode, with at least two minutes of timecode before and after the program material. The majority of CD manufacturing plants specify that the CD submaster should be Sony PCM-1610/U-Matic format, although certain plants will accept Mitsubishi X-80 and/or JVC VP-900 formats. Program level must be between +18 and +20 dB, approaching the headroom limitations of the PCM processor. Reference tones are not necessary because production of the CD submasters involves a digital-to-digital transfer.

Since off-tape monitoring normally cannot be performed during the re-recording stage, the CD submaster must later be scrutinized for drop-outs or any other sounds that could be misinterpreted as part of the original music source. If any sounds or noise of this nature are detected, they must be documented on a comment sheet, so that the CD manufacturing plant is aware of it and production can proceed as planned.

Each song selection must be timed individually, and the start and end time documented accurately. When the digital master reaches the CD plant, the access points will be entered into the PQ Subcode CODER, which automatically encodes the timings at the beginning of the CD side before the start of the program material. This PQ Subcode data enables a CD player to instantaneously locate song selections on command. □□□

Compact Disc: The Wave of the Future

While the Compact Disc is certainly garnering a great deal of attention, is it really better than the analog disk? We've all heard about the poor quality of digital tapes that have been sent to the CD mastering facilities. However, Grundman Mastering has cut analog albums from the same digital two-track that was later digitally copied and sent to the Compact Disc plant for mastering and production. To our surprise, quality-oriented experts at the label were decidedly in favor of the album's quality over that of the CD. It has taken the mastering industry many years to define and remedy the problems of the analog disk, and we are only in the infancy of the Compact Disc.

Many questions remain unanswered with the CD, as well as with digital in general. According to the manufac-

. . . continued

turers, there can be no loss in digital-to-digital transfers, but several experts have testified that this is not the case, and that there can be a definite generation loss during D-to-D transfers. Whether or not there is a quality loss during the transfer from digital copy to glass master at the CD plant remains an unanswered question. And, assuming that the PCM digital signal has made it onto the CD, is a relatively inexpensive consumer playback system capable of retrieving the information? These questions will have to be answered if the CD is to be named champion in the ensuing battle for aural supremacy.

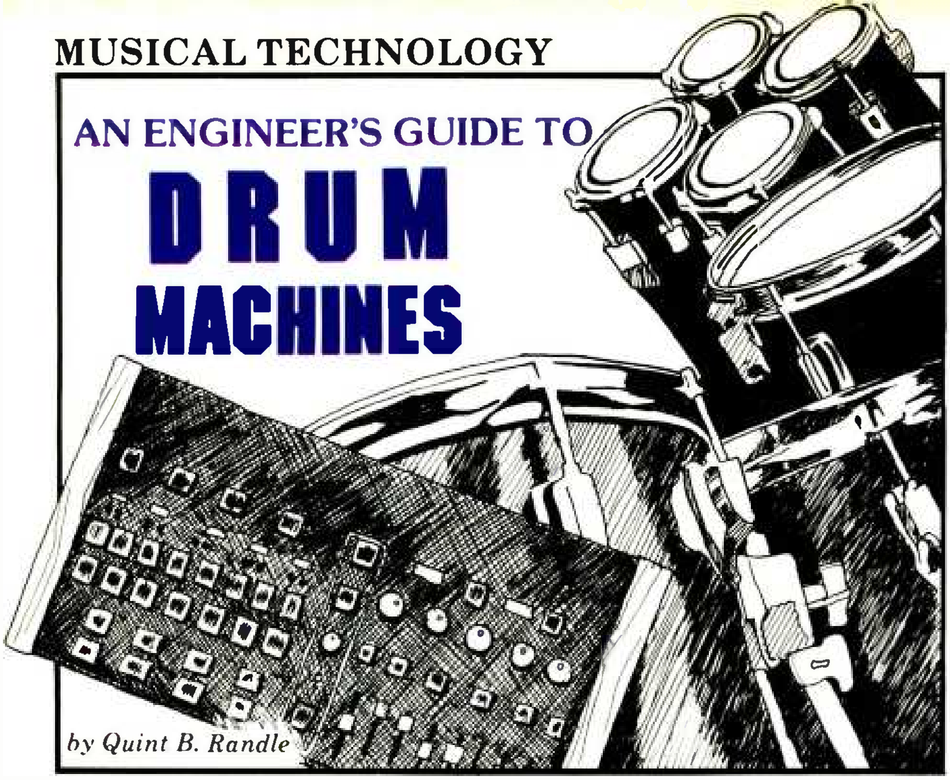
While all the facts are not in regarding the question of audio quality, the Compact Disc does offer some distinct advantages over the analog disk. The CD is quiet; it does not wear out; it provides continuous music for up to 60 minutes; players can be automated; CDs are of consistent quality throughout the entire manufacturing process. With digital you are more or less dealing with a known commodity; good or bad, the digital information is designed to stay the same from plant to manufacturing plant. Sound changes or losses are immediate — you either notice it or not. With analog, however, there is great potential for manufacturing inconsistencies.

Who will be the winner of this high stakes competition? For sure, the manufacturer will be a winner because the Compact Disc is a marketing person's dream. Imagine every owner of a standard turntable forsaking the old medium, purchasing a Compact Disc player, and then rebuilding their record library with CDs. The chances are that the consumer will also be a winner at some point in the future, when all the idiosyncracies of the CD production and reproduction systems are fine-tuned to their maximum. In the meantime, the traditional analog disk will remain a viable medium for many years to come, offering high-quality audio at a very affordable price. ■■■

DIRECT-TO-DISK SESSION FROM THREE LOCKED MULTITRACKS

At press time we heard that Bernie Grundman Mastering was recently involved in a direct-to-disk session for a 12-inch single release of Supertramp's song "Cannonball," using three synchronized 24-tracks at Ocean Way Studios. Having set up a mix on the GML Moving Fader automation system fitted to Ocean Way's custom console, the stereo output was carried via permanent tie lines to Grundman Mastering for the cut. Bernie Grundman is also setting up with Allen Sides, president of Ocean Way, a new custom label, Ora, for direct-to-disk productions. □□□

AN ENGINEER'S GUIDE TO DRUM MACHINES



by Quint B. Randle

The last few years have brought a near rainfall of electronic drums and drum machines to the recording industry. Everyone from Mattel to the manufacturers of more traditional drum kits, like Tama and Gretsch, have produced electronic drums of one form or another. Despite the fact that drummers at first thought these intimidating little machines were going to put them in the unemployment line, many have now become experts at using them to compliment the sound of acoustic drums. And who better to program a complex pattern into a drum machine than a drummer?

Early on in the development of electronic drumming, the sounds of the various drums were synthesized. Today, however, virtually every serious rhythm machine uses digital PCM technology to capture and store the sound of a real drum or other percussion instrument on a read-only memory (ROM) chip. Modern drum machines are, in essence, ultra-specialized computers capable of reproducing sampled drum sounds that additionally can be sweetened synthetically. While writers, producers and artists are using drum machines in songwriting and pre-production processes, many are finding such devices to be an integral part of the 24-track product. Drum machines are not replacing drummers, but are saving time, hassles and money as well as helping to create a drum sound which, in many instances, is hard to beat.

As the sampling and programming qualities of such devices improved — along with the ability of engineers and producers to manipulate them — it has become increasingly more difficult to distinguish between a live drummer playing acoustic drums and the so-called "fake" sounds of a drum machine. Long gone are the days of the generic synthesized drum sound, classically recorded on Linda Ronstadt's "Poor, Poor Pitiful Me," and many a disco tune.

For the purposes of this series of articles, electronic drumming will be divided into two main categories: drum machines, and electronic drums. Drum machines, as previously explained, are computers that execute programmed rhythm patterns of real drums captured in ROM. Electronic drums, on the other hand, producer synthesized or sampled sounds in response to the striking of a contact pad. A third category can be created by combining categories one and two through the use of sequences, triggers, mixing and other techniques.

Part One of this series deals specifically with drum machines; Part Two, to be published in a future issue of *R-e/p*, will examine electronic drums and interfacing.

While, other than sound preference, there are not always a whole lot of differences from an engineer's standpoint, each product has its useful and lacking idiosyncracies. Programming is a totally different ball game and should be left up to a talented drummer. While the following comments on different drums machines are arranged by manufacturer, most of the techniques discussed can be applied to all devices listed here.

ROLAND TR-909

The main difference between the TR-909 and its immediate predecessor, the TR-808, is that the 909 is truly tap programmable through the use of its tap buttons, and the cymbal sounds are digitally sampled. Patching the unit to the console is fairly basic, says **Frank Heller**, chief engineer at Unique Recording in New York, and who recently engineered albums by Melba Moore, James Brown, Freez, and LaToya Jackson. His credits on the Jackson session also included drum programming.

"I go right into the mixer," Heller explains, "I patch through direct boxes straight into the microphone inputs. There's enough signal level [-10 dBv] on the individual outputs to put them right into the mike pre-returns on our MCI console."

If the situation calls for it, Heller says he'll use the TR-909's two stereo mix outputs. The problem with this scheme, however, is that the user is forced to use Roland's concept of what stereo should be: one tom is panned left, another right, with bass and snare drums in the middle, etc. Since the unit has no internal pan controls, you're "pretty much stuck with it," Heller says.

Moving on to producing sounds from the unit, Heller is quick to point out that, invariably, it is not the engineer's job to make sound selections, although sometimes you can become producer by proxy. But, one of the basic things he does with the 909 is limiting. "If there isn't enough attack on the drums, EQ doesn't necessarily bring out snappiness in a lot of cases." Sometimes, he says, a fair amount of limiting will add a certain kind of "snap" to the drum. The initial attack passing through a limiter set to react relatively slowly adds more bite, creating a "*Ppunnch*" rather than a "*Boooom.*" Heller either puts the limited signal directly to tape on the same track as the 909, or prints it on a separate track that will be mixed in later. In addition to basic limiting, Heller finds it practical to gate every output channel from the 909.

While the various manufacturers don't always point to the clocking and triggering capabilities with other companies' products, Heller says that in most cases the electronics can communicate if the signal is gated correctly. "I don't think there's one [drum machine] on the market made right now that allows there's an incredible integrity of cross-communication between different drum machines. There are, however, devices from Garfield Electronics, various J.L. Cooper Electronics products, and MARC's MX1, that address themselves to the particular problems of re-triggering."

The 909's cymbal sounds are most pleasing to Heller. (It should be pointed, however, that the cymbal sounds are sampled recordings, while

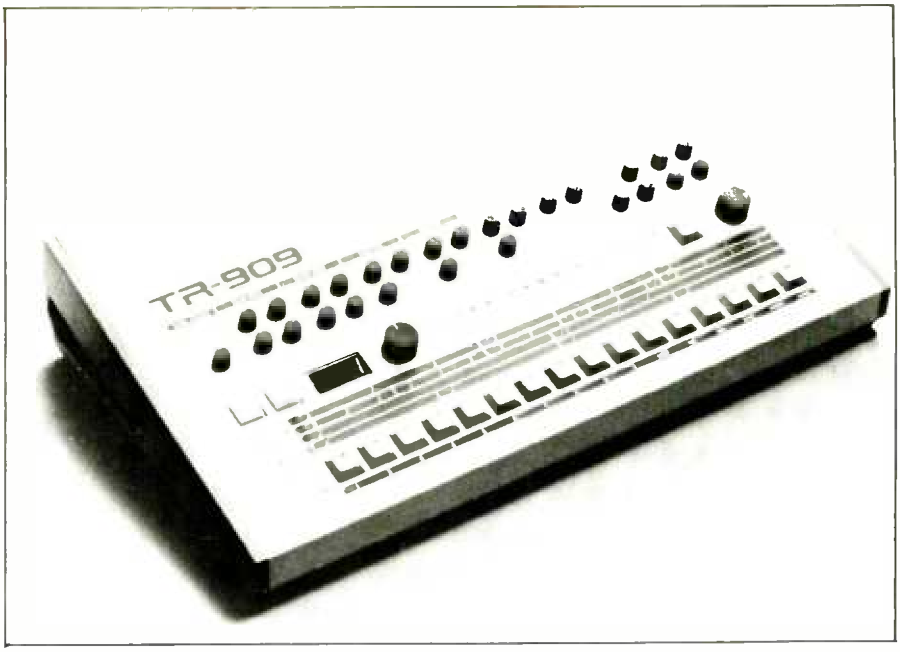
For additional information circle #68

the remaining sounds are synthesized within the drum machine). Heller usually combines the best sounds of several different drum machines to create the perfect set. In such situations, the Roland can produce a timing discrepancy because of the unit's processing speed. According to Heller, "If you hooked up three machines [Roland, Linn, and Oberheim] at the same time, you'd find that the Roland would be running behind."

To solve such lag-time problems, Heller's favorite gadget is the Roland SBX-80 Sync Box. First, he stripes the multitrack tape with SMPTE timecode and a ¼-note click for more than the length of the song. These two signals are then fed into the SBX-80, which internally records the location of every click against the corresponding timecode. The unit can then output everything from MIDI clock pulses and Roland Sync Code, to 24, 48, 96 and 120 per ¼-note pulses.

"The SBX-80 is a brilliant piece of gear for drum machines," Heller confides. "Now that you have a song relating to a number code, you can interface any drum machine with this unit, and move it forward or backwards into time. If you stop at any given point on the tape, there will be a SMPTE number on the SBX-80 readout. You can then call up an offset function [to take care of processing time lag] and simply retard or advance the triggering codes by a few bits."

Roland has just released the TR-707 drum machine, which contains both digitally sampled drum sounds and cymbals; further details of this new unit are included in an accompanying sidebar.



— ROLAND TR-909 —

LINN ELECTRONICS LINNDRUM

"It's very flexible, easy to work with and, number 1, always shows up on time," is how veteran engineer-producer **Bruce Swedien** describes the LinnDrum. Swedien has worked with the machine extensively on the latest releases from Michael Jackson, Sergio Mendez, and Missing Persons.

Many studio engineers and producers consider that the original LM-1, LinnDrum, and new Linn9000 (reviewed elsewehere in this issue), which also features a multitrack digital MIDI keyboard recorder, were the first "serious" digital drum machines, and now are probably the most widely used in studios around the world. The company boasts users ranging from techno-poppers like Prince and The Cars, to country boys like Willie Nelson and Ronnie Milsap.

To hook up the LinnDrum, Swedien says that generally he uses the transformerless microphone inputs on the Harrison board at Westlake, the Los Angeles studio where he does a lot of his session work. Although such an interconnect scheme works well, "In some cases I'll run straight into the tape machine, usually via a direct box to isolate it for hum or noise," the engineer says.

Depending on the year of manufacture, the replacement chips being used, and so on, virtually every LinnDrum machine has a different sound. So, Swedien says, there's no set approach for getting the best sound. He uses his "less is best" philosophy, and does no limiting and as little equalizing as necessary on the LinnDrum's output, which seems to say something for the general quality of the unit's drum sounds.

According to Swedien, the most effective way to take care of any lag-time problems when connecting other machines, synthesizers, and sequencers, to the LinnDrum is as follows: Lay down a quarter-note click in real time from a rhythm source, such as a Garfield Electronics Dr. Click, in a time base that is compatible with the LinnDrum. Then route the click through a 100 millisecond delay, and use the delayed signal to clock the Linn and lay down its sync code. Now

THE PERFECT DRUM MACHINE

Although several drum machines are equipped with just about every desirable feature, to many users the "perfect" electronic drum machine has not yet been marketed. And while less expensive units do not offer everything an engineerproducer/artist may need, some of them do contain features not found on the more expenseive models. If *R·e/p* was to attempt to create the "perfect drum machine," shown below are some of the primary features it might include:

- 32 Internal sounds: four bass-drum sounds, ranging from a "Linn-type" to a heavy-metal sound; six different snare sounds; four separate tom outputs (high, middle, low, and floor); open (loose and tight) and closed high-hat (the latter with light and heavy sounds); side stick; three ride cymbals; three splash cymbals; two cowbells; clave; tambourine; two congas; and a shaker.
- 15 Programming buttons/pads.
- 16 Individual outputs.
- Stereo and mono mixed outputs.
- User-programmable panning (preferably with pots or faders.)
- Output mixing with faders.
- All drums capable of being tuned.
- Replacement cards/chips should be Drumulator- and/or Linn-compatible.
- Clock output/input: 24, 48, 96 pulses per quarter-note.
- Six separate trigger inputs for external triggering of internal sounds, with high procesing speed to prevent time delays.
- Software and interfaces available for Apple and IBM on-screen drum programming.

We would be interested in feedback from users of drum machines regarding other features that they look for in electronic percussion. □□□

any delayed outputs from other devices, caused by slower microprocessor speeds, will be in sync with the Linn.

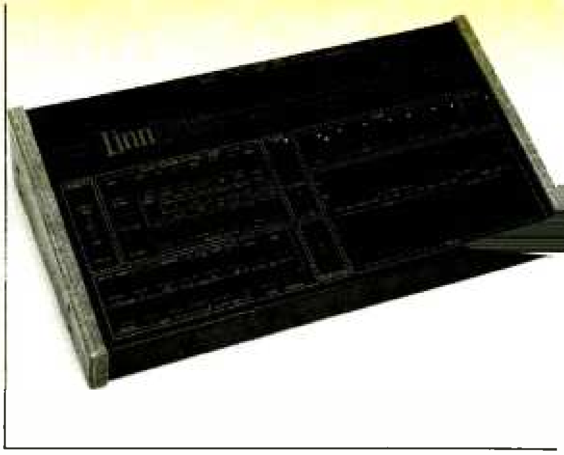
"If you went ahead and just played the LinnDrum and hoped that everything would fly, you'd get into all sorts of trouble, like [having to] flip the tape over, etc. [Flipping the tape over to enable the lagging signal to become an early trigger signal is considered by most engineers to be an archaic technique.] All those things I have done, but it's such a pain in the neck. The 100 millisecond window allows me some leeway; if the [slaved unit's] microprocessor has a little bit more lag in its response than the LinnDrum, I can then shorten that delay and make it play in time with it."

On Missing Persons' album *Rhyme and Reason*, Swedien mixed the LinnDrum's bass with drummer Terry Bozzio's Simmons electronic kick-drum sound. The weight of the Linn kick, combined with the snap, presence and mid-range of the Simmons kick, is what ended up on the record. The individual bass drum output from the LinnDrum was used to trigger the kick sound of the Simmons, he explains.

The Jacksons' tune, "State of Shock," features several LinnDrum sounds, along with some hand claps. To achieve a monstrous snare sound, Swedien gated the Linn snare to open and close the return from an EMT 251, so that the added reverb lasts for the *exact* duration of the snare hits. "It gets really big, and then instantly small. To add a little space to that, and to keep it from getting too drastic, I just put in a little normal reverb sound."

Another little trick that, on occasion, has worked for Swedien is to feed the LinnDrum snare sound through a speaker, place it face down onto a real snare drum, and then mike it normally. "You get a little rattle and acoustical effect," he explains.

The engineer has only one minor complaint about the LinnDrum: "I wish the kick drum was tunable," he offers. Well, in direct response to this



— LINNDRUM —

TECHNICAL REFERENCE TABLE FOR SELECTED DRUM MACHINES

MODEL	Int. Sounds	Outputs	Stereo Outs/ User Panning	Level Mixing	Individual Tuning	Plug-In Sounds	MIDI	Clock Input	Clock Output	Computer Interface	Note/Song Capacity	Recent Session Examples
TR909*	11	11	Yes/No*	Pots	Kick, snare toms, ride, crash	Yes	Yes	Roland compatible slaving only. MIDI	MIDI	No	896 Measures (about 10 saves)	High-hat, cymbals, some snare on LaToya Jackson's album *Heart Don't Lie*
Linn-Drum	15	15	Yes/Yes*	Faders	Snare, toms, congas	Yes	No	Anything MIDI	48 MIDI	No	99 songs 250 steps per song 99 segments	Al Jarreau *High Crime*; Hall and Oates *Big Bam Boom*
Linn 9000	18	18	Yes/Yes	Faders	All	Yes	Yes	Anything MIDI	48 MIDI	"To Come Soon"	50 songs 50 sequences "multitrack recorder"	"Too New"
DMX	24	8	Yes/No	Faders	All	Yes	No	96	96	No	100 songs 256 steps per song 200 sequences	Steve Perry's single "Foolish Heart"
DX	18	6	Yes/No	Faders	All	Yes	Yes	24 48 96 MIDI	24 48 96 MIDI	No	50 songs 256 steps per song 100 sequences	"Too New"
Drumulator	12	9	No	Program	No	Yes	Yes	24 MIDI	24 48 96 MIDI	Apple II/IIc, Mac	64 songs 99 steps per song 36 segments	"China Town" from Chaka Khan's album *I Feel For You*
RX11	29	12	Yes/Yes	Program	No	No	Yes	24 48 96 MIDI	24 MIDI	"Not Yet"	10 songs 255 steps per song 100 segments (patterns)	"Too new"
RX15	15	None	Yes/Yes	Program	No	No	Yes	24 48 96 MIDI	24 MIDI	"Not Yet"	10 songs 255 steps per song 100 segments (patterns)	"Too New"
Drumtracks	13	6	No	Program	All	Yes	Yes	24 48 MIDI	24 48 96 MIDI	No	About 10 songs 99 steps per song 99 segments	"Too New"

LATE NEWS: Oberheim has unveiled the DX Stretch, which provides an additional four rows of three drum voice; Sequential Circuits now offers the TOM digital drum machine — for full details, see the "New Products" item elsewhere in this issue.

* Also see accompanying sidebar description of the new Roland TR707 with full digitally sampled sound.

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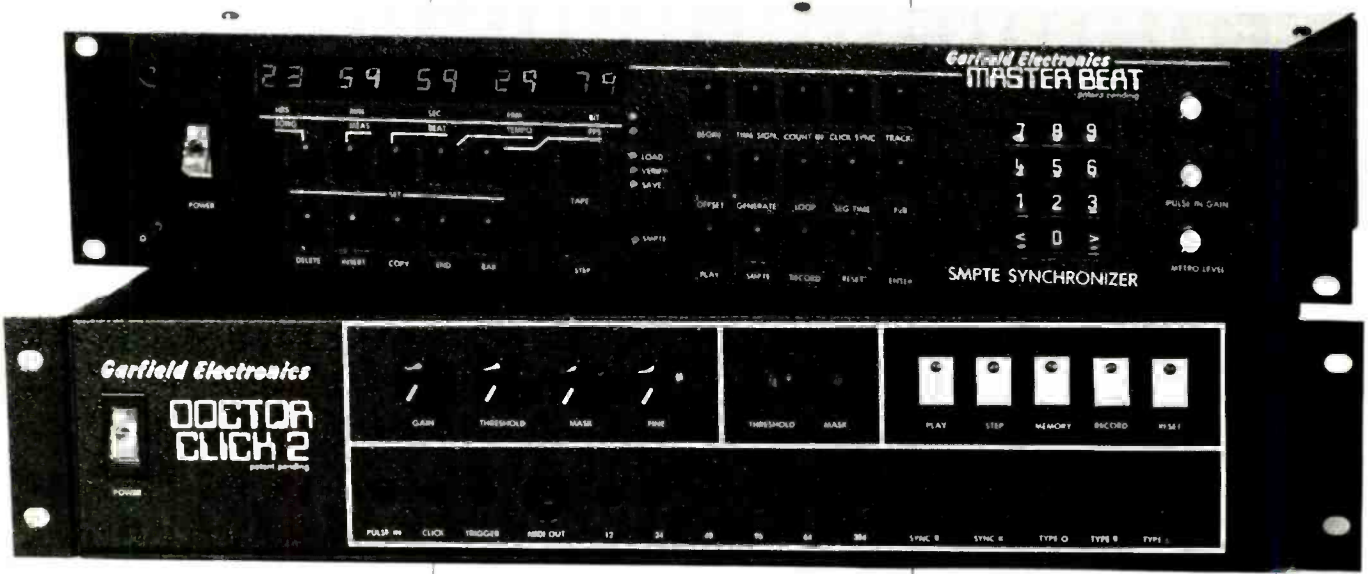

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- *Simultaneous* production of *all* sync formats: 6 fixed clocks, 1 variable arpeggiator/step sequencer clock, the 5 sync-to-tape codes, 2 DIN sync formats, and MIDI. Click and trigger outputs also.
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For additional information circle #69

question, a technician at Linn says that, by going inside the machine, the entire array of sounds are capable of being tuned up or down in pitch. While the company doesn't advertise this fact, in theory it is possible to tune the bass and other drum sounds.

For additional information circle #201

E-MU SYSTEMS DRUMULATOR

— E-MU SYSTEMS DRUMULATOR —

One of the more cost-effective machines on the market, the E-mu Systems Drumulator has several features offered by more expensive units. One feature not found in the Drumulator, however, is composite stereo outputs. According to **George Johnsen**, chief engineer at EFX Systems, Burbank, the mono combined output isn't particularly useful. Instead, he utilizes the individual drum-sound outputs because the mono-mix out just doesn't do the trick, even when sweetening. (In fairness, the facility's unit was purchased several years ago, and was one of the first available models.)

EFX recently finished recording a film soundtrack for *First Strike*, a promotional video for the movie *Dune*, along with a commercial for Dole Pineapple.

Johnsen says that, again, patching the machine is fairly basic; there is enough output level available to connect directly to a console patch point. "To clean up some noisy channels we connect the output through a gate before it passes into the console," he points out. The bass drum channel, in particular, produces some digital noise after the sound itself has ended.

A factory modification available for the older models reduces the output noise, and the problem has been alleviated in newer models. There is still some crosstalk or bleeding of the various drum sounds into other individual outputs, Johnsen points out. "For instance, if you had an entire drum set programmed on the Drumulator, but were only outputting the kick sound, you would still hear some snare and other things on that output."

In terms of processing, the engineer uses a lot of digital reverb to build a "room" around the kit. "The Quantec Room Simulator works real well to get the Drumulator sounding like a real drum set. I don't like it when people use different echoes on each drum, because they do not sound like they are in the same room. Once the Drumulator has been 'set up' in a particular room, you can then add more reverb to individual drums — perhaps the snare — to get a classic sound."

Johnsen feels that, compared to other units, the Drumulator has the best rhythm programming capabilities. "The only drawback to the machine is there's no tuning. The best way to tune the Drumulator is to go ahead and print a sync code on tape, and record the sounds for which you are not going to change the pitch. Then go and varispeed the recorder to move the sync speed up and down, so you can change the pitch of a drum. The unit will track the off-speed sync code, and lay the new tracks right in tempo.

"For example, to achieve the sound of 'Thunder Toms,' just speed the machine up a bit when recording the toms. Using a VSO during the record process provides more control; you can pitch the sound by semi-tones, half-tones, and standard musical intervals."

Like other drum machines, the Drumulator is able to act as master while driving, via the unit's clock out-

For additional information circle #70

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For additional information circle #116

— OBERHEIM DMX —

— OBERHEIM DX —

put, other slave drum machines. Johnsen likes the Drumulator's programming features, kick and snare sound, and then triggers the toms on a LinnDrum slave. "A lot of people are coming into the studio with material they've composed on a Drumulator, and they're looking for 'that' kick sound." He also points out that the range of Drumulator replacement chips made by Digidrum sounds good.

Overcoming sync time anomolies that can be encountered when triggering different drum machines or synthesizers is a big problem, says Johnsen, but easily solved. "We have been experimenting with using SMPTE and sync codes printed on a separate piece of tape, and simply offsetting the sync signal forward, using a timecode synchronizer."

For additional information circle #202

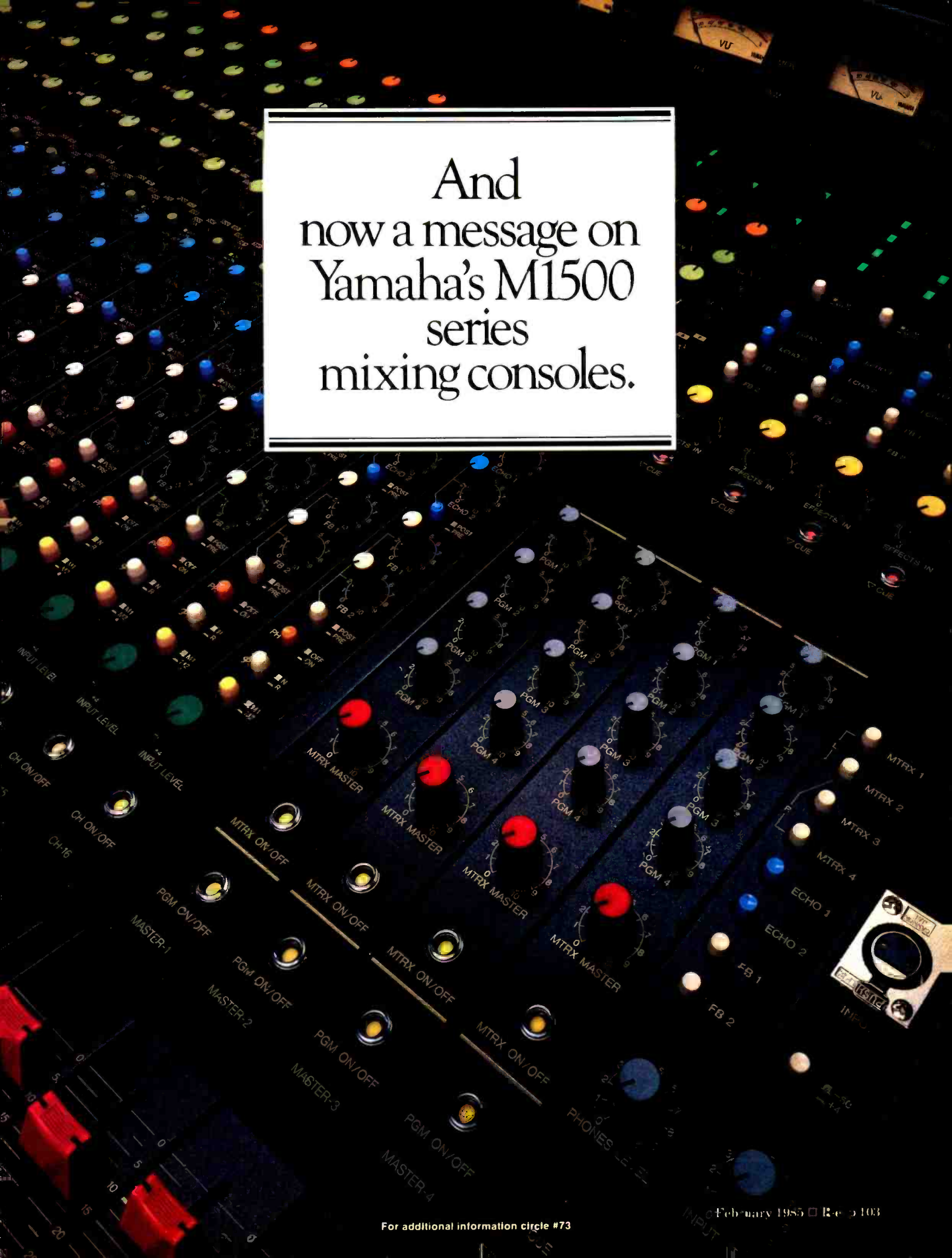
OBERHEIM DMX and DX

Of Oberheim Electronics' two drum machines, the DMX incorporates additional features, such as an alpha/numeric display, and more precise, longer sounds, than those available on the DX.

Niko Bolas, staff engineer at Val Garay's Record One Studio, Sherman Oaks, California, recently engineered albums for Toto, Don Henley, Steve Perry, and Karla DeVito. The engineer explains that he doesn't use direct boxes when patching the DMX, because "they tend to soften the sound a little bit, and we have [Valley People] TransAmps in the front-end of our board [a custom API]. The slew rate can't be beat through these transformerless inputs." Alternatively, the DMX outputs can be patched into the EQ-in, avoiding the Transamps. "But I usually go into the mike pre so I can have some control over the level," he explains.

While some engineers find a mixed-mono output to be not very useful, Bolas sometimes prefers to use the DMX's built-in mixer, along with the individual mono outputs. "It sort of

Niko Bolas serves as session engineer at Record One Studio, and recently engineered sessions for Toto, Don Henley and Steve Perry.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now a message on
Yamaha's M1500
series
mixing consoles.
For additional information circle #73

M1516A

GENERAL SPECIFICATIONS

FREQUENCY RESPONSE +0, −3dB, 20Hz to 20kHz; +0, −0.5dB, 30Hz to 15kHz.

TOTAL HARMONIC DISTORTION (THD)*
Less than 0.5% @ + 10dB, 20Hz to 20kHz. Less than 0.1% @ + 20dB, 50Hz to 20kHz.

HUM AND NOISE* (20Hz to 20kHz, 150Ω source, Input Selector set at "−60")
- −128dBm Equivalent Input Noise (EIN);
- −95dB residual output noise with all Faders down.
- −73dB PROGRAM OUT (77dB S/N); Master Fader at nominal level & all Input Faders down.
- −64dB PROGRAM OUT (68dB S/N); Master Fader and one Input Fader at nominal level.
- −73dB MATRIX OUT; Matrix Mix and Master controls at maximum, one PGM Master Fader at nominal level, and all Input Faders down.
- −64dB MATRIX OUT (68dB S/N); Matrix Mix and Master controls at maximum, one PGM Master Fader and one Input Fader at nominal level.
- −70dB FB or ECHO OUT; Master level control at nominal level and all FB or ECHO mix controls at minimum level. (Pre/Post Sw. @ PRE.)
- −64dB FB or ECHO OUT (68dB S/N); Master level control and one FB or ECHO mix control at nominal level. (Pre/Post Sw. @ PRE.)

MAXIMUM VOLTAGE GAIN (Input Selectors set at "−60" where applicable)
PROGRAM & MATRIX 84dB; Channel In to the corresponding output. EFFECTS 20dB; Effects In to PGM Out.
FB & ECHO 94dB; Channel In to FB/ECHO Out. SUB IN 10dB; Sub In to PGM Out.

EQUALIZATION (±15dB maximum)
LOW: 50, 100, 200, 350, 500Hz, shelving. HIGH MID: 1.2, 2, 3.5, 5.7kHz, peaking.
LOW MID: 250, 350, 500, 700, 1000Hz, peaking. HIGH: 10kHz, shelving.

HIGH PASS FILTER 18dB/octave rolloff below 80Hz.

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DIMENSIONS/WEIGHT M1516A 34" W x 36 1/2" D x 14 1/2" H 147 lbs. M1524 55 3/4" W x 36 3/4" D x 14 1/2" H 213 lbs.
M1532 55 3/4" W x 36 3/4" D x 14 1/2" H 231 lbs.

*Measured with a 6dB/octave filter @ 12.47kHz; equivalent to a 20kHz filter with infinite dB/octave attenuation.

The specs shown are for the 16-channel M1516A console. When you need the same outstanding performance but more channels, there's the 24-channel M1524 and the 32-channel M1532. All three mixers have remote rack-mounted power supplies and are ideal for just about any fixed or portable sound reinforcement or broadcast application.

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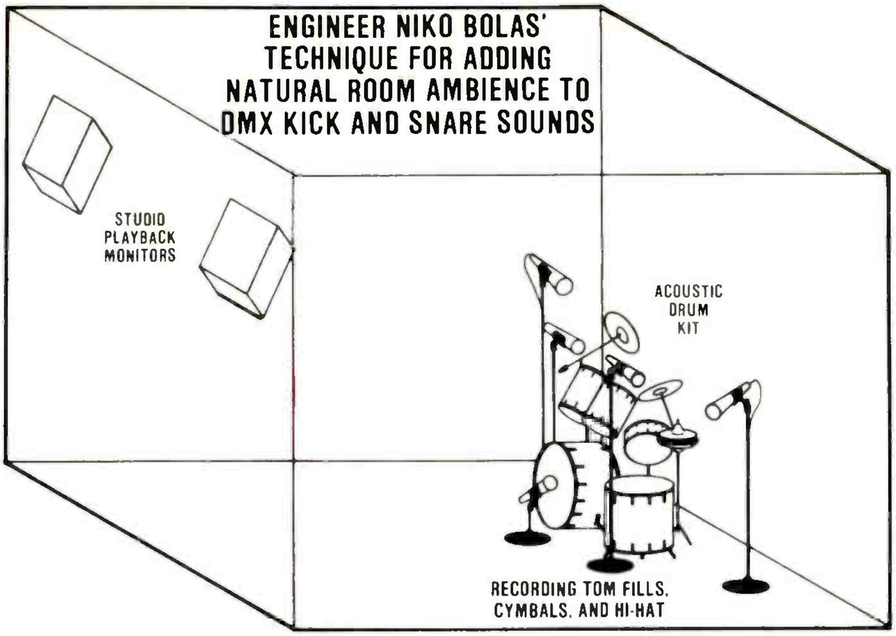
YAMAHA

For additional information circle #207



meshes the sound together, and then I put an old tube limiter over the whole thing to make it a rhythm sound, rather than a drum set."

Like several other drum machines, the DMX features user-replaceable sound cards, a feature that can be important for building a specific sound. "Generally the pre-packaged snare just doesn't *sound* like a drum," Bolas offers. "But if you fire a DMX tom tom, side-stick and snare, along with a Linn and Simmons snare — all on the same beat — and then mix it all together, you can build an *incredible* snare sound.

Response-time problems can be avoided by laying sync codes onto tape first, and overdubbing everything off the code. "All you really want to do, no matter how you do it — and the means isn't really the question — is to get everything in time with everything else. How you do it comes down to whatever is easiest, with whatever gizmos you've got."

Continuing with his "patch-it-in-'til-it-works" philosophy, Bolas says he gets the best ideas from listening to an artist's demos. Sometimes, he offers, cheaper ways to do things sound better.

On Don Henley's album, *Building the Perfect Beast*, the song "Driving With Your Eyes Closed" includes an eighth-note that runs throughout the whole tune. "What that is," continues Bolas, "is the snare at that tempo from a drum machine, Y'd with the output of a [Yamaha] DX7 that had a matchbook stuck in the actual key of the song's basic primary note. Both of the outputs went into [an MXR] noise gate.

"Every time the snare drum was fired, it would open up the noise gate and you'd get that along with the note off the DX7. The snare drum was the only thing strong enough to key the MXR noise gate, so it would immediately close.

"That's stuff you can do in your bedroom, yet it sounded better than anything I've tried to do with the fancy sequencers!"

Combining the DMX with the sound of a real drummer creates "kind of a weird cross-breeding between a machine and a human," Bolas says. "On the Karla DeVito album, I sent the sound of the DMX kick and snare through monitors out into the studio room, where drummer Craig Krampf was playing live tom fills, cymbals, and high-hat."

By opening up a room mike placed above the kit, and playing back the DMX kick and snare sound — "which would be blasting through the monitors" — it's possible to achieve more ambience. "You wind up with a little bit more of a human sound than a very non-ambient drum signal," he confides.

For additional information circle #203

YAMAHA RX11 and RX15

Session musician **Rick Marotta** was the drummer who created and played that Syndrum sound on "Poor, Poor, Pitiful Me," mentioned earlier in this article. "It still haunts me," he says with a chuckle. In addition to his acoustic Yamaha and electronic Simmons kits, Marotta now uses a Yamaha RX11 as a player/engineer/producer. The unit's smaller brother, the RX15, is similar, yet lacks individual drum sound outputs and has 15 drum sounds instead of the 29 available on the RX11.

While the RX11's 29 internal drum

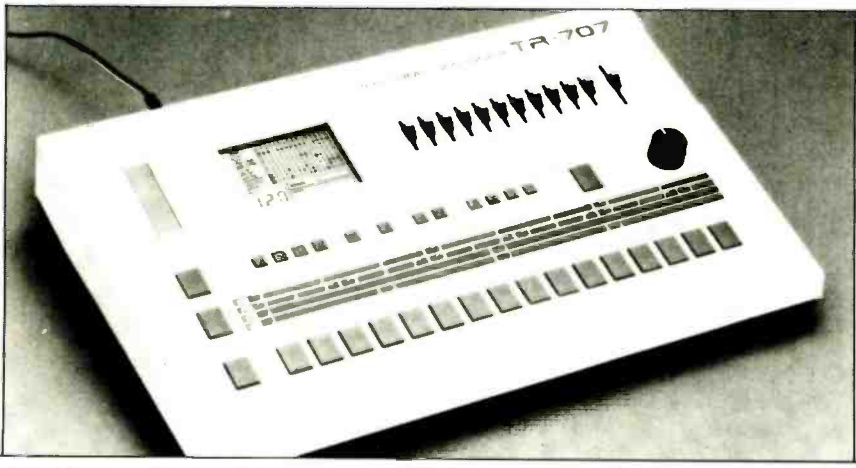
STOP PRESS: ROLAND DEBUTS TR-707 DRUM MACHINE WITH SAMPLED SOUNDS

The new TR-707 Rhythm Composer offers 13 digitally recorded sounds. Each drum sound has its own audio output, in addition to master stereo outputs. Up to 64 individual patterns can be created and used to write complete rhythm tracks with a total memory capacity of 998 measures. Once completed, the pattern and track data can be stored externally in two formats: audio tape, or Roland's M-64C Memory Cartridge, which serves to double the instrument's immediately available memory.

An LCD provides a beat-by-beat readout for graphic editing. Individual drum scoring, accents, measure numbers, track numbers, tempo, and operating modes also are all displayed.

The TR-707 is equipped with MIDI, Sync 24, programmable Trigger Out, and full Tape Sync facilities. Complete dynamics can be programmed with any dynamic MIDI controller or sequence controller. □□□

For additional information circle #204



Drummer/producer Rick Marotta has wide experience with digital drum machines.

sounds put it ahead of other drum machines, it does suffer from the drawback that other voices cannot be added through the use of additional chips, cards, etc. However, according to Marotta, the sounds loaded into the RX11 compare favorably with those available from other drum machines.

"Before I got my RX11," the engineer explains, "I played an RX15 through an R1000 [Yamaha reverb unit] side by side with two different machines. And, for my money, it was amazing; the sound of the bass drum and snare drum blew the others away."

When processed through high-end gear, such as AMS and Lexicon delays and reverbs, "the RX11 sounds as good to me as anything I've heard on anybody's record." Marotta says.

After patching the unit to a console's mike pre-amp input, the only general equalization required "is perhaps a little bottom end to further beef-up the kick or snare. Sometimes

— YAMAHA RX11 —

— YAMAHA RX15 —

if you add a little reverb, you'll have to EQ out some of that tone. But, basically, that's all you really have to do."

While the RX11 may be lacking when it comes to changing its internal sounds, the unit's MIDI capabilities potentially make up for the deficit. "One good thing about MIDI," Marotta continues, "is that it enables you to interface the unit with, let's say, a DX7, or any other MIDI-equipped keyboard. The RX11 or 15 then become sequencers of musical notes, instead of drum sounds." The number of separate notes that can be triggered externally is limited to 12, the unit's number of individual outputs.

For additional information circle #205

SEQUENTIAL CIRCUITS DRUMTRACKS

— SEQUENTIAL DRUMTRACKS —

About a year ago, **Tony Rockliff**, engineer/writer/producer at Classic Sound in Hollywood, sold the studio's existing drum machine and picked up a Drumtracks. "The snare and cymbal sounds are very good," he confides. "Although the bass drum may not sound as good as other drum machines, adding a lot of 5k on the Drumtracks' kick will dramatically improve the sound."

Rockliff has the Drumtracks patched into a Hybrid Arts 16-track polyphonic MIDI sequencer and, in some instances, does not record the Drumtracks' output to 24-track tape. Instead, the Drumtracks, as well as other MIDI keyboards, are sequenced in real time by the Hybrid Arts. During mixdown their outputs are fed into a 20-channel Biamp submixer, and then into the studio's Soundcraft console. This "direct-to-master" technique adds clarity to the drum sounds, he says, by eliminating the multitrack tape stage.

Classic also owns one of the new Simmons SDS-1 electronic drum units. On several occasions, Rockliff has used the output from the Drumtracks, both directly from the machine and from tape, to trigger the Simmons. This technique, the engineer says, "adds depth to the Drumtracks' snare sound. Or, I'll put the snare through an old Pultec tube equalizer to get a very, very fat sound."

In a recent session, during which the producer wanted a heavy-sounding kick, Rockliff came up with the follow patch: "The Drumtracks' bass drum was set up to trigger the Simmons kick sound which, in turn, was set up to trigger a Linn bass drum. These signals were then sent out into some PA monitors and miked from a distance. The combination of the direct outputs and the PA monitors was pretty astounding," he recalls.

One potential complaint, Rockliff says, is that "the toms come out of the same output. There is no way to pan two toms, or however many toms you are creating, unless you do it one by one while overdubbing.

Another function lacking on the Drumtracks is a trigger input, a function that is useful "when you want to 'repair' a real snare drum, for instance, by adding a snare sound from the drum machine. Without a trigger input, it's impossible to get a drum machine to be fired from a snare track."

For additional information circle #206

In Part Two, to be published in a future issue, we'll move on to consider electronic drums and percussion.

■■■

STUDIO UPDATE — FACILITIES EQUIPMENT PEOPLE

Northeast:

☐ **MASTERDISK** (New York City) has purchased a **Mitsubishi X-80** digital two-track. Until now the facility was able to satisfy customer demands for digital equipment through **Audioforce**, a rental agency for Mitsubishi digital products on the East Coast. "We simply reached the point where it made sense for us to buy one,' says mastering engineer **Bob Ludwig**, who explains that nearly 25% of the studio's "major mastering projects" involve digital masters. Ludwig believes that this trend supports the notion that studios the world over are phasing out analog recording in favor of digital techniques. Clienteie will also be able to take advantage of some of the X-80's latest accessories, chief among them a variable pitch playback ability. Using a **DEC-VCO** unit manufactured by **Digtal Entertainment Corporation**, Mistubishi's U.S. marketing outlet, a mastering engineer can alter the pitch of a master tape. The DEC-VCO, developed to provide an interface for post-production synchronization with the **BTX**, **Audio-Kinetics**, and **Adams-Smith** systems, in this mode actually replaces the master clock of the recorder with a variable signal, thus affecting the playback speed. This facility is also claimed as the first to utilize a new autolocator, a software controlled device that operates under TACH or SMPTE timecodes, and provides a number of search-to points and autolocation, simplifying the repeated mastering of album sides. *16 West 61st Street, New York, NY 10023 (212) 541-5022.*

☐ **NUTMEG RECORDING** (New York City) has renovated and expanded its facility. Studio A is now remodelled with an enlarged control room, plus the addition of a video interlock system comprising a **Sony 5850** ¾-inch video machine, a **BTX Softouch/Cypher** system, a **Lexicon model 200** digital reverb, and a **Tascam 8516** 16-track recorder for interlock. The B room is said to be geared for high-tech sound processing by utilizing a **Tascam 58** eight-track recorder, a **Lexicon Prime Time** digital delay, an **Eventide Harmonizer,** along with an **Oberheim OB8** music system (including the **DSX Sequencer** and **DMX** digital drum machine), which will be used throughout the studio in general. *45 West 46th Street, New York, NY 10036 (212) 921-8005.*

☐ **TRACKMASTER AUDIO** (Buffalo, New York) has added to Studio A an **EMT 140** tube plate reverb and **WBI** stereo plate reverb to its exisiting equipment array, which includes a fully-automated, 32 channel **Auditronics 532** console, 24-track **MCI** recorder with auto locator, **MCI JH-110B** two-track and **Scully 280** two-track recorders for master work. The monitoring system includes customized speakers made from **UREI** and **JBL** components. Studio B and C, described as broadcast rooms, noew house two **Otari MX5050 Mk III** eight-track recorders, and two **Yamaha R1000** digital reverbs, which are linked to a customized 18-channel **MCI JH-416** and a 16-channel **Audio-Technica** console, respectively. *One Franklin Park North, Buffalo, NY 14202 (716) 886-6300.*

☐ **KAJEM RECORDING** (Philadelphia) has installed a **Solid State Logic 4000E** computerized console. According to **Kurt Shore,** studio manager, the facility is the first studio in the region to offer such a 48-channel SSL system. Additionally, Kajem is redesigning the control room with acoustical design by **Acoutilog's Al Fierstein,** New York. The SSL will be supported by **Studer** recorders, and Kajem's assortment of special effects gear which includes an **EMT 251, Sony DRE 2000,** digital delay lines by **Eventide, Lexicon, Deltalab, Ursa Major,** and **MXR.** Outboard equipment comprises **Pultec, API, Audioarts, Valley People Gain Brain,** and **Kepex** products. *1400 Mill Creek Road, Gladwyne, PA 19035 (215) 877-2513.*

☐ **BEARSVILLE STUDIOS** (New York City) has renovated its 24- and 48-track Studio A, featuring a rare **Neve 8088** 40-mike/80-line desk, formerly utilized at **Ramport Studios,** England. Console modifications include 80 line inputs for mixing, EQ on either monitor or fader, fader clip, pre- and post-auxiliary sends on either fader, two stereo mix busses, and 48-track recording. The studio features a **Studer A800** 24-track recorder, **UREI 813A** monitors, and many pieces of outboard gear including **AMS** delay, **Lexicon 224X** with Larc, and **Drawmer** noise gates. In addition, owner **Albert Grossman** announced the appointment of **Steven Bramberg** as studio manager. *P.O. Box 135 Speare Road, Bearsville, New York, 12409 (914) 679-8900.*

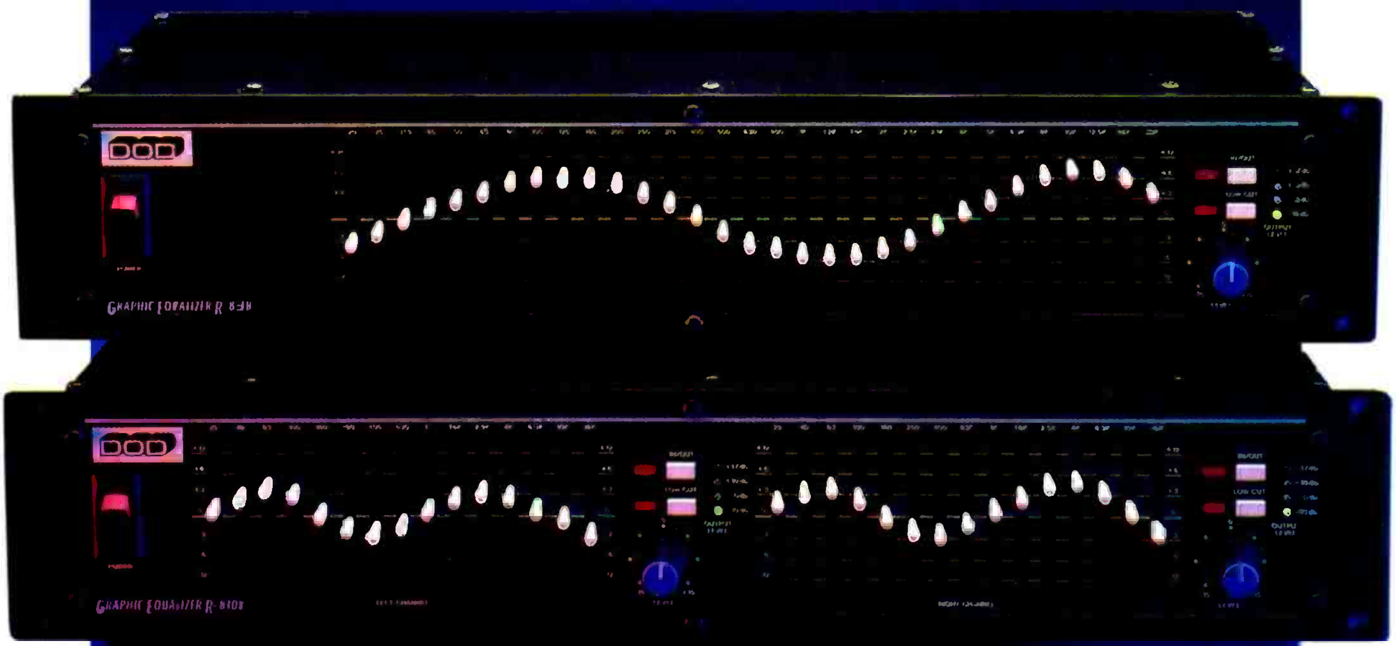
☐ **JOHN HILL MUSIC** (New York City) is expanding its headquarters with the addition of a new 16-track recording studio opening in mid-March. Among the equipment to be offered in the new studio will be: a **Sound Workshop** console, **Auto-Tech** 16-track two-inch recorder, **Lexicon PCM 42** digital delay and **Model 200** reverb, **Eventide H-910 Harmonizer, Orban 622, Electro-Harmonix** sampler, **Valley People** Series 800 gates, and **Yamaha NS-10M** monitors.The electronic music list to be housed in the studio includes a **Prophet V, Yamaha DX7** and **PF15, Korg Poly 800, Hohner** electric pianos, **Yamaha RX-11** drum machine, **LinnDrum 9000** drum machine, and **Roland MSQ 700** sequencer. *116 East 37th Street, New York, NY 10016 (212) 683-2273.*

☐ **BREWSTER SOUND & VIDEO** (Lynchburg, Vermont), formerly known as Turnkey Audio Visual, recently installed an **Otari MX-5050 MkIII** eight-track with eight channels of **dbx** noise reduction. The facility also purchased an **Otari MX-5050 MkIII** two-track, an **Otari Autolocator** for both tape machines, a pair of **Shure SM-81** microphones, an **Yamaha DX-7** synthesizer, and several additional pieces of audio/video gear. According to **J. Troy Jones**, studio manager, "The new equipment will give the studio and our clients a new flexibiltiy in all types of recording, such as demos, sound tracks, A/V narration, voiceovers, audio-for-video, and master recordings." The facility also opened studio B, which Jones says is "our dubbing room and helps us free up Control Room A from video and audio dubbing. We can now do video format transfers or video sweetening while at the same time doing high speed or real time audio dubs in the same room, and not conflict with any work being done in room A." The new A/V equipment purchase includes two **Kodak** 16mm and 35mm projectors, a **Sharp** 19-inch monitor, a **Sony** ½-inch VCR, and a 12-foot by 12-foot **Daylite** projection screen. *2532 Langhorn Road, Lynchburg, VA 24501 (804) 528-4448.*

PDS — new PPG digital synthesizer

☐ **PHILADELPHIA DIGITAL SOUND** (Philadelphia) is reported to be the area's first computer music and sound synthesis studio to acquire a new **PPG Computer Music System** and a variety of analog and digital synthesizers, including a **Yamaha DX7, Oberheim Expander,** and **Prophet** synth. Says co-owner **John Hodian,** "Using a combination of MIDI interfacing, variable clock devices, and SMPTE synchronizers, we can lock up the computers and music system, and all the synthesizers, drum computers and sequencers. We can create huge, complex, multi-timbral pieces using all of our different synthesizers, and by-pass our multitrack facilities by going direct to digital master." According to engineer/producer **Michael Aharon** (pictured left with Hodian) the PPG allows the studio to achieve virtually any sound or any type of synthesis: digital, analog, additive, sound sampling, or FM. *210 Church Street, Philadelphia, PA 19106 (215) 922-1340.*

TRUE EQUALITY



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For additional information circle #78

STUDIO UPDATE
FACILITIES EQUIPMENT PEOPLE

□ **QUADRASONIC SOUND SYSTEMS** (New York City) has taken delivery of a 40-input **Solid State Logic 4000E** console and computer, which will be linked to its existing **Studer A-80** 24-track, **Studer A80** two-track, and **Ampex 440** and **350** two-track recorders. Outboard gear additions include an **Orban 662** parametric EQ, an **AMS** digital delay, an **AMS** digital reverb, and a **Lexicon PCM-42** digital delay processor. In addition, the facility also purchased three sets of keyboards: a **Steinway** grand and baby grand piano, and a fully programmable, eight card **Yamaha DX7**; a **Garfield Electronics Dr. Click** was also added. *723 7th Avenue, New York, NY (212) 730-1035.*

Midwest:

□ **STUDIOMEDIA** (Evanston, Illinois) has added a customized, 32-input **Trident Series 80 console, Threshold S-100** power amplifiers, and a **Lexicon 200** digital reverb. Studio B expands from eight to 16 tracks with the acquisition of a **Tascam MS-16** one-inch machine, a **Trident Series 65** console, **UREI 811** monitors, and a**Lexicon PCM 60** digital reverb. *1030 Davis Street, Evanston, IL 60201 (312) 864-4460.*

□ **SWEETWATER SOUND** (Fort Wayne, Indiana), has added to its synthesizer array with the purchase of a **Roland MKS-30**, which utilizes 16 MIDI channels, a **Yamaha KX5**, and a **Yamaha RX-11** digital rhythm programmer. In addition, Sweetwater Sound has added **Craig Voekler** to regular staff personnel.*2350 Getz Road, Fort Wayne, IN 46804 (219) 432-8176.*

□ **JOR-DAN, INC.** (Wheaton, Illinois), previously known as a cassette duplication facility and media studio, has extended its capabilities with the inauguration of Studio A. Designed by **George Augspurger** and **John Edwards** the new room is said to feature the first **Studer A80 VU MkIV** 24-track delivered in the U.S., along with a vintage, all-discrete 28-input **Neve 8058** console modified under the personal supervision of Rupert Neve, with Neve compressors, 31102 EQs, 16 added subgroups, and four wild-grouping faders accessible through the patchbay. Other equipment in the studio includes **Studer A810** and **B67** two-track recorders, **UREI** monitors, **Hafler** amplifiers, plus reverberation and signal processing by **EMT, Lexicon,** and **Aphex**, among others. The facility also offers a large stock of micophones by **Neumann, Sennheiser, AKG,** and **Beyer**; instruments include keyboards from **Yamaha**, and drums from **Sonar.** *1100 Wheaton Oaks Court, Wheaton IL 60187-1043 (312) 65-1919.*

□ **STREETERVILLE STUDIOS** (Chicago) has expanded its audio-for-video post-production capabilities. According to chief engineer **Steve Kusiceil**, the remix room at the five-studio complex opened in late February with a new **SSL 6000E** Series Stereo Video System, which will be equipped with SSL's Integral Synchronizer and Master Transport Selector. "We will be using the system to control a **Sony BVU-800** VTR, twin **MCI** 24-track recorders, and an **Otari MTR-10** four-track, and an **MCI** one-inch audio layback machine," explains Kusiceil. "The SSL display shows what every machine is doing, so we'll be able to keep them in a separate area adjacent to the control room." At Steeterville, the SSL will allow producers to mix as many as 64 sources into standard or "split" stereo and mono formats. The split formats permit separate stereo music, effects, and dialog submixes to be created at the same time as the main stereo or mono program mix. The system is also said to be one of the first in the world to be fitted with SSL's programmable EQ, providing two channels of dynamic parametric EQ and panning automation for dialog, EFX matching and special effects. *161 East Grand Avenue, Chicago, IL 60611 (312) 644-1666.*

Southeast:

□ **HIDDEN MEANING STUDIOS** (Warner Robins, Georgia) has installed a new **Tascam M520** console, **Tascam 85-16B** 16-track recorder, an **Eventide Harmonizer, an AKG C414** mike, a **Yamaha** digital reverb, **JBL 4312** monitors, a set of **Simmons** electronic drums, and a **Roland** guitar preamp. The control room also have been remodeled. *1134 Watson Blvd., Wanter Robins, GA (912) 923-5507.*

□ **FLOOD ZONE STUDIOS** (Richmond, Virginia) has expanded from an eight- to a 16-track facility. Installed into a newly acquired 16-foot, control-room trailer, the added equipment includes a **Soundcraft 1600** Producer Series console (24 × 8× 24), a **Studer A80VU MkIV** 16-track, **Studer B-67** and **Revox A-77** two-track machines, **Hafler** powered **JBL 4430, 4411,** and **Auratone 5C** monitors, **Lexicon** digital reverb and delay lines, **MXR** digital delay line, **U.S. Audio Gatex,** and compressor/limiters by **UREI** and **dbx**, according to owners/engineers **Bruce Olsen** and **Steve Payne.** *P.O. Box 7105, Richmond, VA 23221 (804) 644-0935/224-8993.*

□ **STRAWBERRY JAM** (West Columbia, South Carolina) has upgraded to an automated 24-track facility with the installation of a new **MCI JH-24** multitrack, supplied by **Studioworks** of Charlotte, North Carolina. The new equipment has been intergrated into a new control room designed by **Tom Irby** of **Studio Supply Company**, Nashville. *3964 Apian Way, West Columbia, SC 29169 (803) 356-4540.*

□ **NEW RIVER STUDIOS** (Fort Lauderdale, Florida) claims to be the first studio in Florida to take delivery of an **AMS RMX16** digital reverb. Other equipment purchases include several **Valley People Kepex IIs** and **Gain Brain IIs**, a **UREI 1176 LN**, and and a pair of **Schoeps M221B** tube microphones. *408 South Andrews Avenue, Fort Lauderdale, FL 33301 (305) 524-4000.*

South Central:

□ **SUMET-BERNET SOUND STUDIOS** (Dallas) has completed a total renovation of its Studio A's headphone/foldback system, with redesigned headphone stations containing selectable XLR or quarter-inch headphone jacks, switchable stereo/mono function, and individual high-power level controls. Headphone amplification is provided by **Crown.** A pair of **JBL 4311** speakers on roll-around stands were added for live orchestra/chorus foldback. Also added to Studio A, to compliment it four natural chambers, are an **AKG BX20, EMT** plate, **MICMIX XL515,** and the new, four-program **Lexicon Model 200** digital reverb. *7027 Twin Hills Avenue, Dallas, TX 75231 (214) 691-0001.*

□ **MUSIC RESOURCES** (Franklin, Tennessee) is the name of a newly opened recording facility that is reported to be Nashville's only recording studio geared exclusively for computerized and electronic music production. Designed and built by owner/musician **Steve Schaffer**, the studio features a **Synclavier** computer synthesizer system along with a **Tascam 85-16B** multitrack machine, which utilizes **dbx** noise reduction, a **Tascam M-15-B** console, a dual keyboard **Sequential Circuits** Prophet 10, a **Lexicon Model 200** digital reverb, and a **Sony PCM-F1** digital two-track recorder for mixdowns. The facility also offers a digital effects library which Schaffer has named "Real Sounds," consisting of musical notes and figures stored on floppy disk ready to load and


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play on a computer connected keyboard, with a 25 kHz bandwidth. The library also is said to cut costs and offers a great variety of sounds, in addition to giving producers enormous flexibility for replacing or repairing tracks by enabling quick and easy synchronization in video or film work. *3524 West End Avenue, Nashville, TN 37205 (605) 269-5296.*

MUSIC RESOURCES — new synth studio

☐ **THE CASTLE RECORDING STUDIO** (Franklin, Tennessee) now offers a keyboard-synthesizer room for audio pre- and post-production. Complimenting the existing **Yamaha DX-7** and **DX-1** is a new **Fairlight CMI Series IIX** digital synthesizer with, a **Jupiter 8** synthesizer, **Simmons** drums and **SDS 6** sequencer, **Oberheim DMX** drum machine, **Roland TR808** drum machine, and a **Yamaha RM1608** console linked to a **Studer A80 MkIII**. According to owner **Joseph Nuyens**, the room, designed and built by staff personnel, was established mainly for production and publishing of demos in conjuction with **Castle Productions and Five Tower**, a division of Castle Publishing. "The new synthesizer room makes demos easier for the artist," Nuyens says, "and it gives our clients an alternative to the main studio, which lately is pretty booked." In addition to equipment upgrades, new staff acquisitions include **Chuck Ainly** as chief engineer, **Keith Odle** as second enginner, **Giles Reaves** as synthesizer programmer/-engineer, and **Tommy Dirsey** as synthesizer programmer/arranger. *Old Hillsboro Road, Route 11, Franklin, TN 37064 (615) 791-0810.*

☐ **HOYT & WALKER PRODUCTIONS** (Little Rock, Arkansas) has opened a new eight-track facility designed specifically for broadcast production. The equipment list includes a custom **Neotek** Series I console, **Otari MX5050** eight-track recorder with **dbx**, autolocate and remote, an **Otari MTR-10**, and **Valley People Gain Brain** and **Kepex** processors, **Orban** compressor and de-esser, **EXR EXIV** psychoacoustic enhancer, **Eventide 949** Harmonizer, **Symetrix** telephone patch, **Fostex** and **Auratone** monitors powered by **QSC** and **NAD** amps, **Lexicon Model 200** digital reverb, and microphones which include **Neumann U-87**s and **EV RE20**s. The facility, which was designed by **Steve Blake** of **PRS**, Boston, measures as follows: control room — 18 by 20 feet; and studio — 15 by 20 feet. According to business manager **Jeff Hoty**, the "Live-end/Dead-end"-type studio design is said to be the first in Little Rock. "Comments from local broadcast producers have been astonishing," Hoyt notes. "Their commercials now have a cleanliness and a 'bite' that really make them stand out on the air." *3422 Old Cantrell Road, Little Rock AK 72202 (501) 661-1765.*

H&W — broadcast eight-track studio

☐ **SOUND STAGE RECORDING STUDIOS** (Nashville) has installed two **Mitsubishi X-800** 32-channel digital recorders, according to president **Ron Kerr**. Says MCA-Nashville president **Jimmy Bowen**, who has worked at the Sound Stage's second room, The Backstage, for a number of years, "I had the opportunity to use both the Mistubishi and the Sony multitracks while on the West Coast, and have recently recorded a couple of projects on the 3M DMS. I found the operation of the X-800 transport to be more efficient than the 3M, and much closer to the feel of an analog recorder. Our decision to commit to Mitsubishi was also based on its full 32-track capability and sound quality." he added. "We plan to use the X-800s on all projects from now on, and I am encouraging other producers that MCA works with to do the same." The first Mistubishi was delivered in early January, and began its recording chores almost immediately at **Emerald Studios**, on a project with vocalist Reba McIntre. The sessions are being produced by Bowen with the assistance of engineer **Ron Treat**. The second unit arrived shortly thereafter, and was also put to service on Bellamy Brothers projects at **The Castle**, Nashville. After the completion of these projects, one of the decks will return to The Sound Stage for in-house sessions, while the other will float for a while among Bowen's Nashville-area projects. The acquisition also means that Sound Stage will be able to join New York's **Clinton Studios** and San Francisco area's **Fantasy Studios** in offering 64-track digital recording. *10 Music Circle South, Nashville, TN 37203 (615) 256-2676.*

☐ **WORD OF FAITH TELEVISION MINISTRIES** (Dallas) has installed a 40-input **SSL 6000E** Series Stereo Video System, reportedly the first in Texas according to president **Robert Tilton**. The facility, utilizing its own satellite facilities to distribute closed-circuit programming to over 1,200 churches in the U.S., says that the system is capable of handling up to 80 simultaneous inputs, and is equipped with SSL's Studio Computer Integral Synchronizer, and Master Transport Selector. This configuration claims to provide dynamic dynamic mixing automation as well as complete machine control over the facility's complement of **Otari MTR 90** and **MTR 12C** audio and **Ampex VPR-3** video transports. The facility's SSL 6000E is also fitted with Total Recall to facilitate changeover between live- and post-production projects. *P.O. Box 819000 Dallas, TX 75381.*

Northern California:

☐ **THE BANQUET STUDIOS** (Santa Rosa) recently added a **UREI 521** electronic crossover and additional power amps by **Nikko** and **SAE** to bi-amp its **JBL 4430** monitors to provide a tighter-end response and additional headroom (250W/side low end and 125W/side top end). The facility has also mounted an additional pair of **JBL L-112**s in a closer field array for a third monitoring perspective. *540 B East Todd Road, Santa Rosa, CA 95407.*

☐ **BEAR WEST STUDIOS** (San Francisco) has remodeled and upgraded its 24-track and 16-track recording facilities with a redesigned control room and monitor system. Studio A features a new 28-input **Sound Workshop Series 34** console to accompany the **MCI** 24-track recorder, and **MCI** and **Ampex** two-track mastering recorders. New outboard gear includes a **Lexicon 224XL** digital reverb and **PCM-41** digital delay. Studio B, a 16-track room, has been outfitted with a **Tascam M-520** console and a one-inch **Tascam** recorder. In addition to remodeling the building and studio interiors, Bear West has added an upstairs lounge. Also recently added to the staff as assistant manager is **Fran Feldstein**, who was formerly night manager at Record Plant Studios in New York, and The Automat, San Francsico. *915 Howard Street, San Francisco, CA 94103 (415) 453-2125.*

Australia:

☐ **AUSTRALIA BROADCASTING CORPORATION** (Sydney, Australia) has placed an order for 46 **Otari MTR-10** two-tracks for use in its production areas in Melbourne and Adelaide. Australian distributor **Klarion Enterprises** reports that some of the Otari MTR-10s will also be used in the ABC's new **Radio Australia** complex in the Melbourne suburb of Burwood. In addition,

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the ABC has also recently purchased over 170 **Otari MX-5050** full-track recorders for use in both radio and television facilities throughout Australia. *Sydney, Australia.*

Quebec.

☐ **McLEAN HANNAH STUDIOS LTD.** (Hamilton) has added an new **Tascam 85-16B** 16-track tape machine to its existing eight-track studio. The machine will be linked to a **Synchronous Technologies SMPL System**, a timecode-based computer editing system said to be the first mating of such equipment in Canada. Company owners **Dan McLean** and **Paul Hannah** say that the studios produce radio and television commercials, industrial presentations, album and seven-inch projects, and audio-for-video works. *154 Sanford Avenue, Hamilton, Ontoario, Canada L8L 5Z5 (416) 526-0690.*


AMBER — new Mistubishi digital X-800

☐ **AMBER STUDIOS** (Toronto, Canada) has added a **Mistubishi X-800** 32-channel recorder, reportedly the first digital multitrack at a Canadian facility. "The trend in Canada has been, for more successful acts, to do substantial recording and mixing in New York or Los Angeles, because of limited availability of advanced recording equipment in Canada," explains owner **George Semkiw**, (pictured here with Tore Nordahl and George Eschweiler of GERR Electro, representatives of DEC in Toronto). "Now that Amber has the latest in digital audio recording systems, I am sure we are about to reverse that trend. All of the major Canadian studios have been sitting back with a 'wait-and-see-attitude' about digital audio recording system. But before committing to digital recorders, we considered installing a newer and bigger analog console, but after thorough technical and operational evaluation, I concluded that the Mistubishi digital recorders would provide us with more versatility, much improved sound performance, 32 tracks and much greater competitve edge in the marketplace." The facility's monitor design is by **George Augspurger**, containing **JBL** components in a three-way system with extended low-frequency response. *Toronto, Canada (416) 868-0528.*

United Kingdom:

☐ **STEWART COPELAND** (England), drummer **The Police**, recently purchased a **Harrison MR-4** console for his personal-use 24-track studio. His decision to purchase a Harrsion board was said to have been made after experience with a number of Harrison mixing consoles in America during recent tours, and most recently, at **Surrey Sound Studios**, which were the first commercial studio in the U.K. to re-equip with the MR-4.

To be included in future issues of "Studio Update," send details of equipment, services, and personnel changes to *R-e/p*, P.O. Box 2449, Hollywood, CA 90078.

A CRITIQUE OF AUDIO COMPACT CASSETTE HOUSINGS

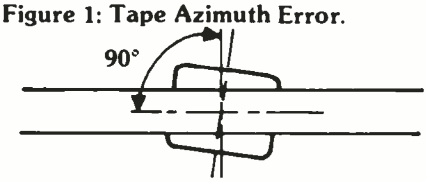
Five Important Parameters to be Considered When Selecting Materials for Tape Duplication

by Robert L. Piselli

It is common knowledge that the use of audio Compact Cassettes has grown in recent years, beyond the popularity of the LP record. The cassette's portability, compactness, and ease of use are a few of the reasons that account for this tremendous growth; audio cassette tape has come a long way since the early days of the Sixties when Philips first introduced the Compact Cassette format in Europe. Enormous sums of money have been spent on research and development by various tape manufacturers — literally millions of dollars have been poured into tape formulas, oxides, and binder systems that attach the dispersion to its carrier, commonly referred to as the polyester film. Today's tape formulas of chromium dioxide, and cobalt-doped ferrics by far exceed the quality performance originally intended by its inventor. Even high-speed tape duplicating systems used by the major record labels are now capable of recording high frequencies up to 18 to 20 kHz. In terms of the audio quality delivered to the consumer, it is fair to say that the majority of improvements have resulted from enhancements in both tape and professional high-speed duplicating systems.

But what about the cassette housing itself? After all, the tape, advanced recording techniques, and the plastic cassette housing are integral parts of the finished product. If a high-quality tape is used in a poor-quality cassette housing, it will yield much the same results as a poor-quality tape in a high-quality housing. This article will consider some of the factors affecting the performance of a plastic-molded audio cassette housing.

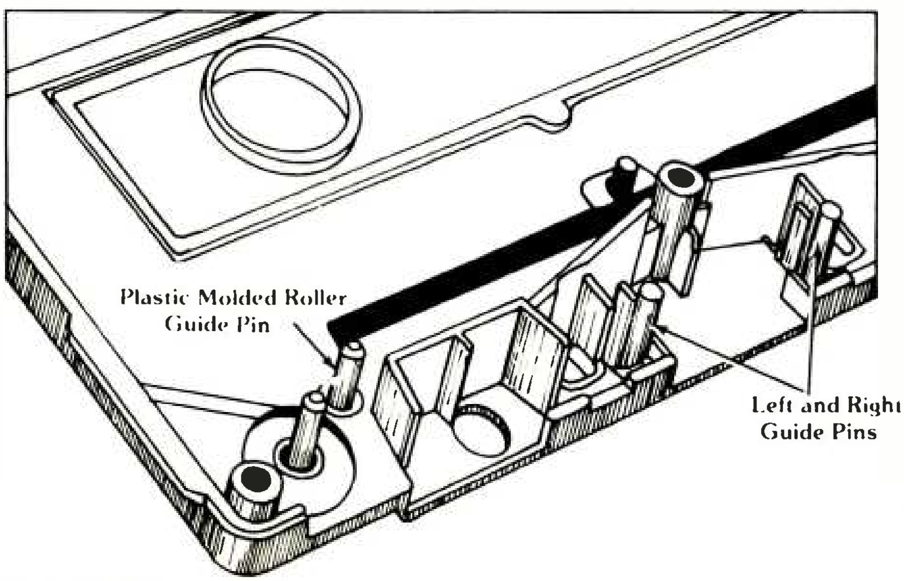
Figure 1: Tape Azimuth Error.



—the Author—

Bob Piselli is the southeastern regional sales manager of IPS, Inc., a supplier of both audio and video cassettes. Prior to joining IPS, Piselli served for seven years as national sales manager of professional products at BASF.

Figure 2: Locations of guide pins in a Compact Cassette housing.



Azimuth/Tape Alignment

Azimuth and tape alignmemt are, by far, the most important parameters in the reproduction of high-frequency information; a useful definition of azimuth would be "the physical position of the tape head gap in relation to the magnetic lines of force on the tape." Ideally, the reproduce head gap should be set at 90 degrees to the magnetic print on the tape (Figure 1). Important elements influencing tape azimuth alignment in the cassette housing are the left and right guide pins, as illustrated in Figure 2. If these plastic pins are not molded perpendicular to the horizontal plane of the cassette housing, the tape, as it travels across the head, will skew or literally move across the head at an angle other than 90 degrees (Figure 3). Depending upon the amount of skewing, the loss of high-frequency output can be as much as 12 dB. (Tests have shown that an azimuth error of 0.2 degrees will result in a 12 dB drop at 15 kHz.)

To dramatize this effect, use an open reel-to-reel tape recorder and record a 15 kHz signal. While monitoring the playback output on a VU meter, place your forefinger against the oxide surface of the tape at a 90-degree vertical axis to the tape path.

Figure 3: Tape Skew resulting from misaligned guide pins.

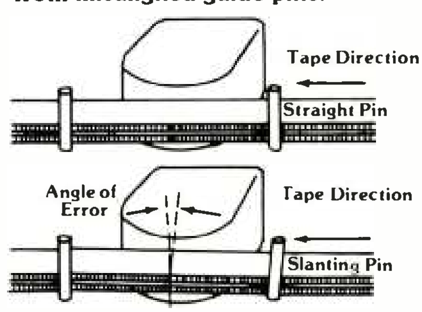
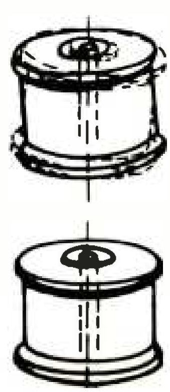


Figure 4: Roller Guide "Wobble."



The Professional Cassette Decks

Tandberg's new Series TCD 900 is a superior and cost-efficient alternative to the (unprofessional) practice of using inferior home tape decks for Professional applications. These new Professional Cassette Decks offer unparalleled sound capability, advanced mechanical and electronic design, plus extraordinary control flexibility based on an 8-Bit microprocessor with 32K of EPROM memory. These are the type of quality products for which Tandberg is well known, and are designed and built in keeping with the company's more than 50-year reputation for quality, performance and long-term owner loyalty.

TCD 910
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TCD 910 is designed to replace both reel-to-reel and cartridge machines in many applications, and is capable of producing tapes at sound and silence levels beyond that required by broadcast and studio requirements. Combined with its extremely accurate real time counter and sophisticated autolocator functions, this machine is truly a multipurpose cassette recorder.

Features include:

- High precision, rugged 4 motor tape transport with direct load, instant access cassette positioning.
- Discrete three head system with built-in record azimuth adjustment is combined with Tandberg's proprietory Active Phase Correction Circuitry, exclusive discrete, wide band electronics, plus the highly regarded Actilinear II and Dynec systems. In addition, the latest generation Dolby B and C noise reduction processors are utilized.
- All audio circuitry uses high spec polypropylene capacitors and metal film resistors.
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- Optional RS 232 computer interface, infrared wireless and hard wire remote with facer start.
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TCD 911
Cassette Playback Deck

The TCD 911 offers the same quality of construction and design as the TCD 910. Its special features include:

- Playback pitch control.
- External playback azimuth control, for optimum performance from *any* pre-recorded cassette.

In a multi-deck studio situation, the TCD 910, combined with the TCD 911, makes for the ideal match of performance, reliability and price.

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For additional information circle #208

Apply a small amount of pressure against the tape and move your finger around that vertical axis. Note the level change on the VU meter. This same principle applies to the guide pins in the cassette housing.

Roller Guides

Of equal importance in controlling azimuth and tape alignment are the roller guides. Most manufacturers of cassette housings utilize steel pins for the roller-guide axis. Such steel pins are inserted by pressure fit into one-half of the housing, and the roller guides then inserted onto the appropriate pins. Until recently, it has been determined that the runout, or diameter difference, between the roller guide and the steel pin have caused serious azimuth errors. Referring to Figure 4, the roller guide, due to these differences in the axis runout, creates a "wobble" effect of the guide itself. The molding tolerance of a steel pin roller guide is very difficult to maintain because of its small size opening. Recently, it has been determined that molding a roller guide with a larger opening, and supporting it with a plastic pin, will reduce azimuth errors considerably. Bear in mind that this plastic pin is now molded into the cassette housing, just like the guide pins, so that perpendicularity of the axis is assured, and the roller guide itself has a larger axis hold reducing molding dimensional variances (Figure 5).

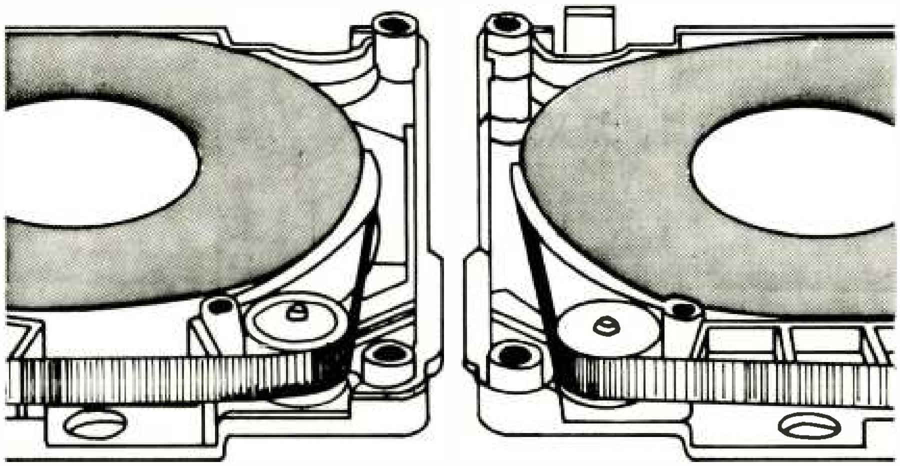

Figure 5: Differences in appearance between steel axis pin and roller (*left*) and plastic molded roller guide and axis pin (*right*).

A very simple test to determine runout of a cassette's roller guide, is to fast forward or rewind the cassette. The chattering noise heard during high-speed winding is related, in most cases, to the roller guide wobbling on its axis. This effect, in turn, will contribute to aximuth errors during normal play speed.

Pressure Pad

Another component affecting the performance of a cassette is the pressure pad which, because it exerts force and ensures intimate head-to-tape contact, is essential for good high-frequency response. The most important function of the pressure pad is that there be sufficient play in its movement to allow it to self-seat. By seating correctly, an evenly distributed pressure is applied as the pad contacts the wide variety of head shapes in use today. Because of the various head configurations used on many different cassette players, a large pressure pad is desirable, to more evenly distribute this pressure over a larger area.

Cassette Moulding and Liner

Of course, the guide pins, roller guides, and pressure pad all contribute to improved azimuth alignment and tape guidance. But of greater importance are the two molded halves of the cassette housing itself. The type of plastic used must withstand temperatures of up to 80 to 85°C, without warping, for a period of not less than 24 hours. (After all, the temperature in an automobile, with the windows closed and in the heat of summer, can reach as high as 160°F.) If warpage of the cassette occurs, misalignment will result due to the deformed contact points between the housing and the recorder player itself. Thus, the head-to-tape contact becomes misaligned, not to mention jamming of the tape, which will occur during play, fast-forward, or rewind modes.

The last item of importance that ensures proper operation of the cassette is the liner, also referred to as the "slip sheet." The liner is a thin piece of plastic film, or paper, inserted into each half of the cassette housing, with the supply and take-up hubs sandwiched between each liner. The liner's function is to exert a slight pressure to the edges of the tape during the play, rewind or fast forward modes; this edge pressure guides the tape onto the hub, resulting in a smooth, even wind.

Some of the most commonly used liners are: Polyolifin; paper impregnated with graphite; PVC coated with graphite; and uncoated PVC (clear).

It is debatable as to which type of liner is most effective; however, the configuration of the liner, not the material, has proved to be of greater importance. Rather than having a flat, smooth surface, the liner is crimped in two opposite directions along its length. If viewed from a cross section, the liner would appear to look like the letter "Z"; thus, the name "Z liner" is given. The crimping gives the liner a spring effect, creating a gentle, more efficient guiding action onto the edges of the tape, resulting in a superior wind.

The criteria listed above are just a few of the important developments made in recent years to provide the end user with an efficient, smooth operating cassette housing.

As molding techniques improve and the materials used refined, the quality of the cassette performance will continue. The record industry, together with the manufacturers of blank tapes, have made significant progress toward the development of audio cassettes. Teams of engineers meet on a regular basis to discuss new ideas and how to implement them into the manufacturing process. Without this consciencious effort on the part of the manufacturer, there is little doubt that the audio cassette would not be as popular as it is today. ■■■

THE FINAL STAGE

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Key to Services:
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Northeast:

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CRYSTAL CITY TAPE DUPLICATORS, INC.
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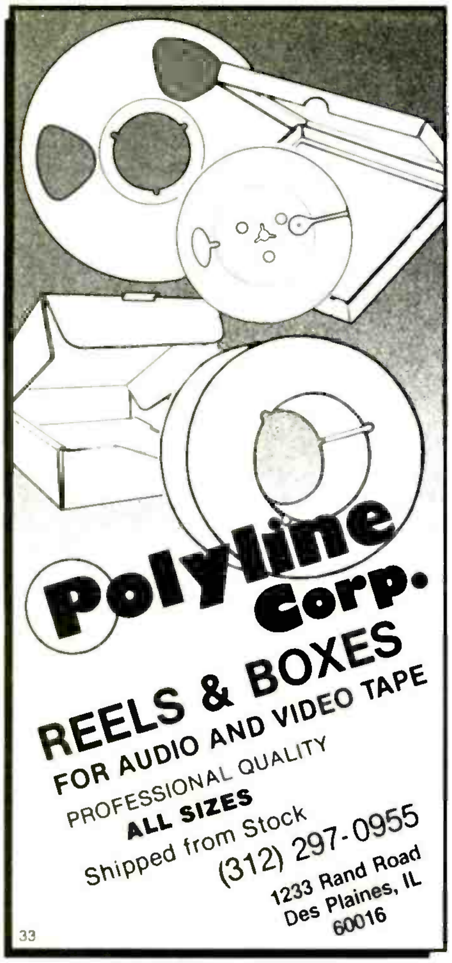

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ON THE STUDIO TRAIL

Mel Lambert reports on the Neve DSP Digital Consoles at CTS Music Center and Tape One, London

During a recent trip to England, I had the opportunity to visit and talk with the users of Neve Digital Sound Processing (DSP) consoles installed at CTS Music Center, Wembley, north west London, and Tape One, an independent disk and CD mastering facility based in central London. Many *R-e/p* readers already will be aware that digital technology is, without doubt, a primary wave of the future. Currently, our control rooms are considered incomplete without the ubiquitous array of digital delays, sampling devices, reverbs and pitch shifters — to say nothing of the virtual arsenal of digital synthesizers, drum machines and computer-controlled sequencers and composition software. And, now that digital multitracks and mastering machines are being used routinely throughout the record, film, and audio-for-video industries, many studio owners are preparing for the inevitable final element in an all-digital studio: the digital console.

Although the majority of engineers and producers would consider some form of "computer literacy" to be essential for an understanding of today's recording and synthesizer technology, there is no denying that the internal workings and operational flexibility of an all-digital console can be somewhat intimidating at first encounter. Also, it can take a long time to develop the hardware and software necessary to at first mimic — so that we can at least operate the board in a way with which we are familiar — and then offer all the additional features that could never be made available on an analog console. To say nothing of having to debug the system so that reliable and consistent operation can be taken for granted.

Early in November last year, the DSP at CTS Music Center Studios finally found a permanent home in the facility's new Eastlake Audio-designed Studio A. As many of you may already be aware, following the original DSP delivery at CTS last summer, one or two software problems necessitated the console's temporary removal to a nearby room, so that the final debugging process could be carried out without disrupting sessions. According to CTS managing director Peter Harris, by running buffered mike lines to the DSP's input racks studio staff were able to make full use of the temporary downtime by familiarizing themselves with the console's myriad features, while the last few system bugs were resolved.

Following return of the DSP to Studio A control room, the console saw its first commercial digital session with senior engineer Dick Lewzey recording and mixing Maurice Jarre's orchestral score for the film *The Bride*, with the Royal Philharmonic Orchestra. Other sessions slated for the first few months of this year included the recording of a Carl Davis soundtrack for the film *King David*, and Paul Stillwell recording a music album.

The DSP console at CTS is set up for tracking to and remixing from a Studer A800 analog 24-track or Sony PCM-3324 digital 24-track, and is configured to accommodate up to 48 input channels and 32 group outputs. To enable first-time users of the DSP to feel more comfortable in making the transition from analog to digital operation, the console is laid out ergonomically in a "split" configuration, with an additional bank of 32 faders mounted to the left of the central mixing position. At normal system setup (more on this later) the DSP's internal software — remember that routing and bus assignment "switch" and "fader" in the DSP is microprocessor controlled — the board is configured with 48 mike/line inputs routing to 32 output/multitrack *every* busses, these latter outputs appearing as the inputs to the monitor bank.

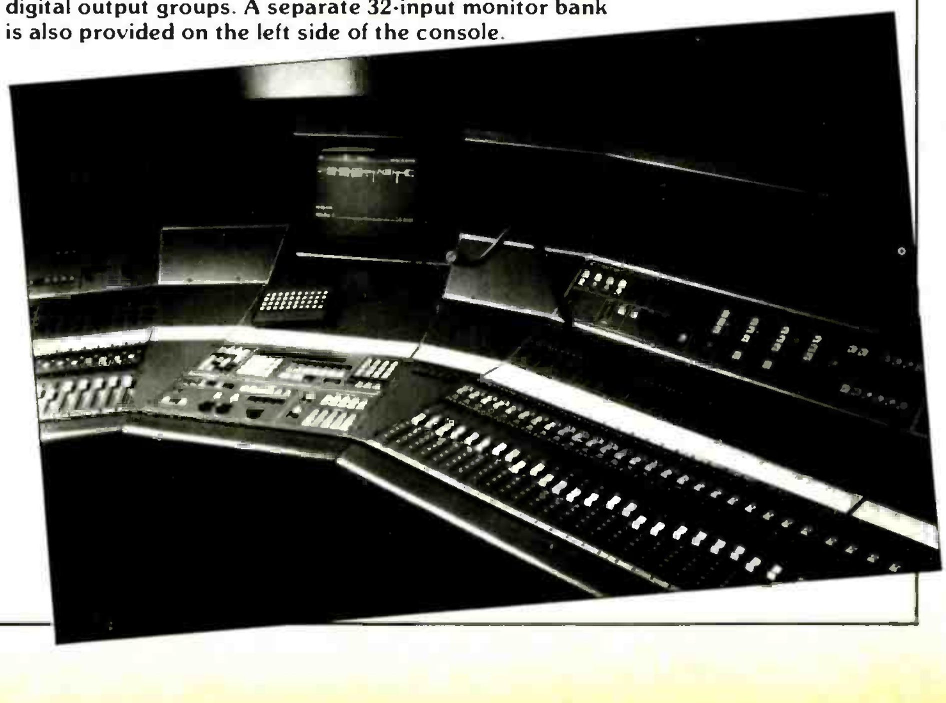
As can be seen from the accompanying photographs, physically the DSP comprises a console "control surface," video display, plus an ASCII keyboard for labelling various functions, and effecting data storage and recall. A single fiber-optic cable connects the control-room hardware with a bank of six, 8-foot high racks that house the various analog mike/line pre-amps, digital signal processing, analog-to-digital and D-to-A converters. As well as "conventional" analog inputs and outputs, the DSP will accept 16-bit/48 kHz digital inputs and outputs, and is designed for direct connection via an AES/EBU interface to the facility's PCM-3324 multitrack. (A Studer Sampling Frequency Converter also is on hand for direct digital connection to a Sony PCM-1610 processor, which operates at a sample frequency of 44.1 kHz.)

Since every function of the digital console, including fader levels, cue/effects assign and level, EQ parameters, routing, etc, is under direct software control, the DSP also features "Total Reset," which enables the entire control-surface configuration to be stored on floppy disk. In addition to the console's servo-controlled faders, there are future plans to add expanded NECAM automation to provide real-time storage and recall of EQ, aux sends, pans, and compressor-limiter dynamics.

Each input channel features a servo-driven fader, plus a four-segment LED window that enables each strip to be labelled with an input designation from the ASCII control keyboard. (The same labels appear on remote microphone input panels in the studio, enabling unambiguous assignments to be made during session setup.) Similar LED windows can also be toggled to display the groups and monitors to which a particular input fader is routed internally.

A separate pair of equalization and dynamic adjustment sections can be assigned to individual inputs, outputs, monitor channels or auxiliary busses. These controls allow precise changes to be made, and then assigned to a selected section of the signal path. The two EQ sections — to enable a pair of channels to be adjusted simultaneously —provide a high- and lowpass shelving filter sweepable from 25 to 280 Hz, and 2.0 to 15.7 kHz, respectively. A four-band parametric

The DSP at CTS Music Center is configured to handle 48 input and 32 analog/digital output groups. A separate 32-input monitor bank is also provided on the left side of the console.



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A810-TC shown with Studer TLS4000 modular synchronizing system.

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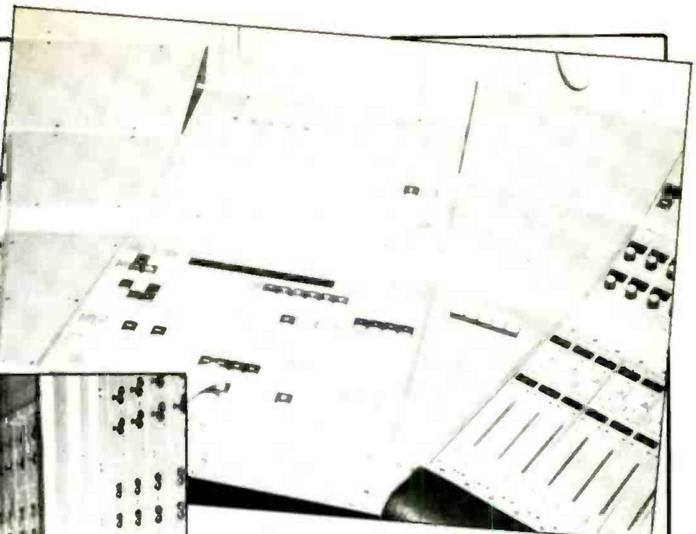


The DSP at CTS — seen *above* in its temporary home, prior to its recent return to Studio A's control room — enables all functions to be controlled from a central VDU/keyboard (*above left*). Shown right are digital processing racks.

EQ bank supplies up to 18 dB of cut and boost at center frequencies between 30 and 600 Hz (low), 100 Hz and 1.2 kHz (low-mid), 600 Hz and 12 kHz (high-mid), plus 1 and 15 kHz (high). Since EQ control is effected digitally, the cut/boost and center-frequency controls are covered by 21 discrete steps, rather than continuously; bandwidth "Q" is also variable in four steps from 0.5 to 2. The dynamics section features a pair of limiters, compressors, expanders and noise gates; each section is provided with full control of threshold, attack and release times, plus compression ratio.

A total of 48 simultaneous EQ, dynamics and filter settings can be assigned to various sections of the console, and inserted into the digital signal path in any order — one configuration might be a dynamics module, followed by the filter section, and then the four-band equalizer. In addition, a delay module can be insterted between the mike/line gain control (-26 to +80 dBu, in two-dB steps) and the EQ, filter or dynamics section. A total delay of 2.4 seconds is available per 24 channels of signal processing, and can be allocated in 20-microsecond steps up to 400 microseconds, 100-microsecond steps up to 2.5 milliseconds, and in one-millisecond steps thereafter. (The distributed delay also can be inserted as the first function block in a group return.) One application for the assignable delay would be to phase/time compensate a multimike array, to ensure indentical arrival times.

A central bank of eight auxiliary send masters can be assigned under software control to the appropriate foldback or effect bus outputs, routing to individual aux busses being controlled from a pair of assignable controls per input channel

A second series of assignable pushbutton controls provide the digital equivalent of Solo in Place, Pre-Fade Listen, and a function called SOFT (standing for "softkey") that can be set up from the central control panel to effect filter in/out, mute, auto-fade, insert in/out, and numerous other functions. Two additional controls located in the center of the input-fader bank are also software-definable switches; on the CTS console, the upper control can be switched between input gain, stereo width, delay, and the odd-numbered auxiliary sends, while the lower knob switches between pan, monitor level, fade, and even-numbered sends.

Level monitoring is provided by 32, 100-segment bargraph meters, whose response can be switched between VU and peak-reading ballistics. A four-element LED window below each meter can be used to identify group output/multitrack assignments. Because of the enhanced dynamic range offered by digital technology, the metering range extends from -36 to +24 dBu.

And just when you might have thought that the DSP's features were becoming a touch complicated, a central color video display enables the console routing and current status to be interogated from a central position. By way of an example, the VDU will display the routing assignments *to* and *from* the various groups, as well as the configuration of the six internal subgroups. The insertion order and signal flow through the dynamics, filter and EQ sections within each input, monitor and output strip can also be displayed.

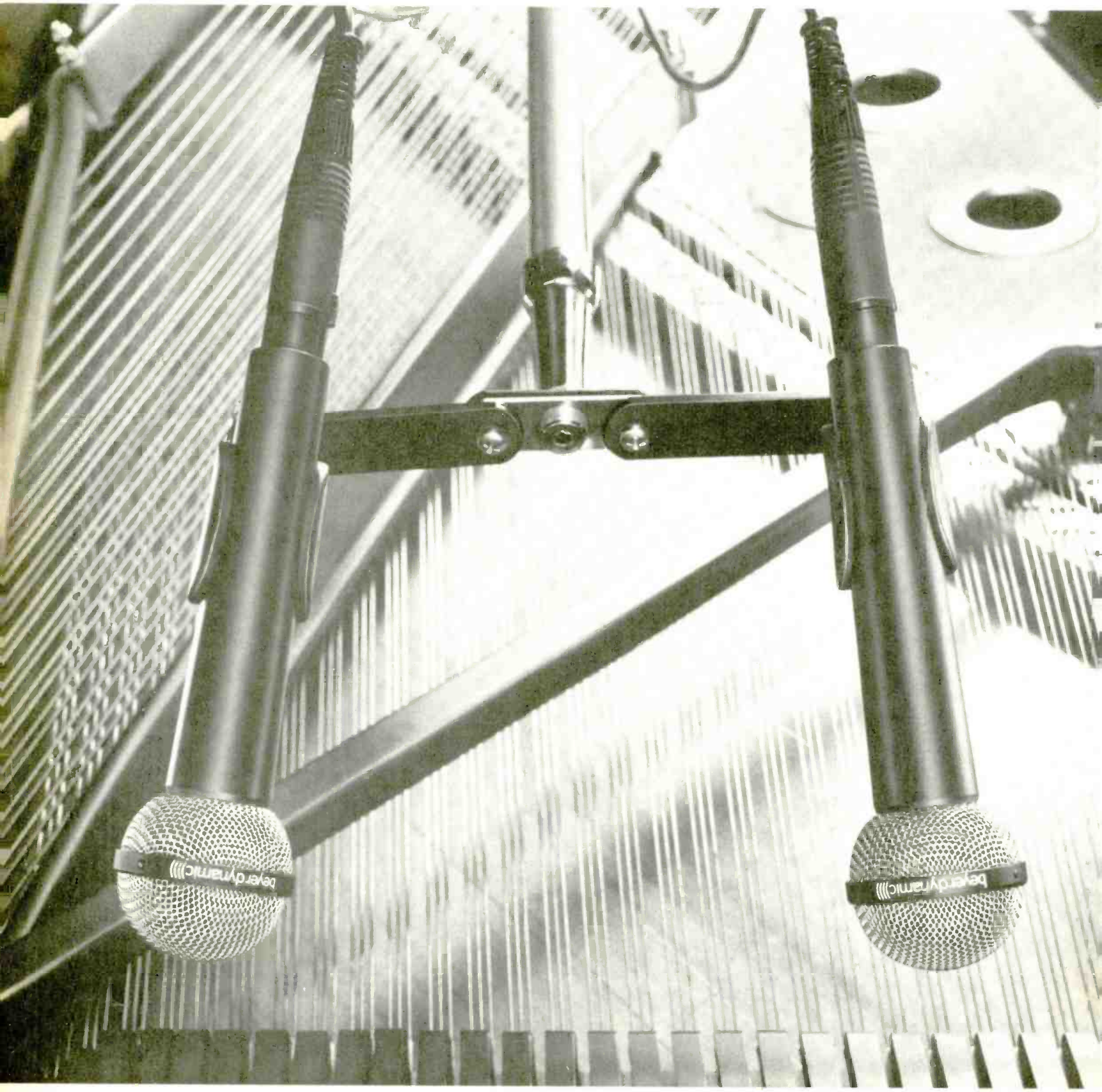
To simplify the setting up of commonly encountered console configurations, various input/output routing and assignment topologies have been programmed into the CTS digital console. For example, Mode #1, "Laydown," sets up the console as a 48-input/32-bus tracking board, with full EQ, dynamics, and filter sections inserted into the first 40 input channels, with eight spare inputs to be re-assigned as required. Mode #2, "Multitrack Mixdown," resets the board to accept 24 inputs from a multitrack, plus 20 spares, and provides a stereo group master and three "spare" output groups (possibly for simultaneous three-stripe film mixes). For added convenience, the 48 channel strips are laid out left to right as follows: stereo master (one fader controlling both left and right levels), three spares output groups, 24 multitrack returns, and the 20 unassigned inputs. According to various engineers I spoke with at CTS, this particular console configuration enables the off-tape track levels to be monitored on the bargraphs located directly above the appropriate channel fader.

Mode #4, "Minimum Hardware System" (#3 had yet to be defined at the time of my visit) is intended to provide a basic setup in the rare event of a partial hardware failure, and configures the DSP with full EQ and dynamics on 24 inputs routing to 16 outputs. Mode #5 retains the current system configuration in non-volatile CMOS memory, and enables the console to be returned quickly to the format utilized on a previous day's session. Upon initial power-up of the console, a set of built-in diagnostic utilities checks the status of all input and output processing hardware, and reports any problems.

With such operational flexibility available to a recording engineer, Peter Harris readily concedes that it will be a while before the staff at CTS Music Center can take full advantage of the DSP's capabilities. However, having gained sufficient day-to-day experience with the console system, staff and visiting session engineers will be able to develop new production techniques based on the practically limitless circuit topographies of a truly reprogrammable control surface. As Harris points out, "Effectively, each operator can customize the desk his own way for his preferred method of working, whether in music recording or film work. It is this flexibility which allows the DSP to offer more sophisticated capabilities in the same space as a conventional console."

And if the new music-recording DSP at CTS Music Center might be described as possibly offering the ultimate in console technology, the smaller Compact Disc Mastering DSP installed at Tape One is no less

BEYER RIBBON MICROPHONES AND



THE DYNAMIC DECISION

THE DIGITAL RECORDING PROCESS

Digital technology holds forth the promise of theoretical perfection in the art of recording.

The intrinsic accuracy of the digital system means any recorded "event" can be captured in its totality, exactly as it happened.

Naturally, the ultimate success of digital hinges on the integrity of the engineer and the recording process. But it also depends on the correct choice and placement of microphones, quite possibly the most critical element in the recording chain. This can make the difference between recording any generic instrument and a particular instrument played by a specific musician at a certain point in time.

The exactitude of digital recording presents the recordist with a new set of problems, however. The sonic potential of total accuracy throughout the extended frequency range results in a faithful, almost unforgiving, recording with no "masks" or the noise caused by normal analog deterioration. As digital recording evolves, it places more exacting demands on microphones.

***Ribbon microphones are a natural match for digital because they are sensitive and definitively accurate.** The warm, natural sound characteristic of a ribbon mic acts as the ideal "humanizing" element to enhance the technically perfect sound of digital.*

Beyer ribbon mics become an even more logical component of digital recording due to an exceptional transient response capable of capturing all of the nuances and dynamic shifts that distinguish a particular performance without the self-generated noise and strident sound generally attributed to condenser mics.

***Beyer is committed to the concept of ribbon microphones.** We manufacture a full range of ribbon mics for every vocal and musical instrument application.*

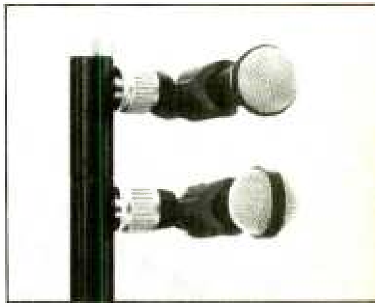
The range of Beyer ribbon microphones. From left to right: M 500, M 160, M 260, M 130

The Beyer M 260 typifies the smoothness and accuracy of a ribbon and can be used in stereo pairs for a "live" ambient recording situation to record brass and stringed instruments with what musicians listening to a playback of their performance have termed "frightening" accuracy.

Because of its essential double-ribbon element design, the Beyer M 160 has the frequency response and sensitive, transparent sound characteristic of ribbons. This allows it to faithfully capture the sound of stringed instruments and piano, both of which have traditionally presented a challenge to the engineer bent on accurate reproduction. Axis markers on the mic indicate the direction of maximum and minimum pickup. This allows the M 160 to be used as a focused "camera lens" vis a vis the source for maximum control over the sound field and noise rejection.

Epitomizing the warm, detailed sound of ribbon mics, the Beyer M 500 can enhance a vocal performance and capture the fast transients of "plucked" stringed instruments and embouchure brass. Its diminutive, durable ribbon element can also withstand extremely high sound pressure levels.

The Beyer M 130's bi-directional pattern enables the engineer to derive maximum ambience along with clean, uncolored noise suppression. Two M 130s correctly positioned in relationship to each other and the source can be used as part of the Mid-Side miking technique. The outputs from the array can be separated and "phase-combined" via a matrix of transformers to enable the most honest spatial and perceptual stereo imaging — sound the way we hear it with both ears in relationship to the source.



***Given the high price of critical hardware used in digital recording, the relative price of microphones is nominal.** Realizing that microphones are the critical sound "source point," no professional can allow himself the luxury of superficial judgements in this area. Especially when one considers the value of ongoing experimentation with miking techniques. For this reason, we invite you to acquaint yourselves with the possibilities of employing Beyer ribbon technology to enhance the acknowledged "perfection" of digital recording technology.*

Beyer Dynamic, Inc.,
5-05 Burns Avenue, Hicksville,
New York 11801

For additional information circle #101

beyerdynamic)))

impressive in its features. (The CD console was scheduled to be joined in late January by a companion DSP Disk Mastering Console; I hope to provide a full description of the facility's latest addition in a forthcoming issue.)

Designed to enable level, EQ and dynamics changes to be made entirely in the digital domain, Tape One's DSP is configured with a pair of stereo inputs and outputs that accept BNC composite-video signals from a companion Sony PCM-1610 digital processor. The sampling frequency currently is set at 44.1 kHz, although later units from Neve will be designed for 48 kHz operation. According to the facility's managing director, Bill Foster, the EQ section is identical to that provided on the CTS console, except that the cut/boost range is restricted to 10 dB to enable more precise adjustment. "We are currently evaluating the standard DSP EQ section to see if we need extended high-frequency ranges for mastering," he says. "We may change the settings via software at a later date." Also provided is a standard DSP lowpass/highpass filter section, plus switchable de- and pre-emphasis on the inputs and outputs, respectively. The console also features the standard dynamics section; in addition, the distributed delay can be set up in a feed-forward limiting mode, if required.

At the time of my visit, Foster was also contemplating the purchase of a second DSP CD mastering console, designed for direct connection via an AES/EBU interface to a Studer D820 digital two-track operating at a sampling frequency of 48 kHz; a Studer SFC would be made available for use with 44.1 kHz PCM-1610 processors. Under further consideration was a purpose-built F-1 Editing Suite, possibly based around an HHB/Amek CLUE system. "Lots of smaller studios in England are mastering to F-1, and then transferring to 1610 format for editing," he offers. "Our planned facility will allow transfers to and from various formats to be made digitally, and without loss is audio quality."

The Tape One DSP Compact Disc Mastering console (shown *left* in close-up) enables EQ and dynamics changes to be made entirely in the digital domain.

To put the recent digital-hardware investment at Tape One into a wider perspective, Foster relates that since installing the Compact Disc DSP, his facility has been mastering in excess of 10 CD releases per week, and that over 5% of the world's CD titles have been mastered at Tape One.

Both Peter Harris at CTS Musc Center and Bill Foster at Tape One would appear to have an inside track — from the recording and mastering ends of the production spectrum — on the growing impact that digital technology is having on our industry. ■■■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All photography courtesy of NBC. Photographers: Herb Ball, Gary Null, and Frank Carroll

AUDIO POST PRODUCTION FOR NBC ENTERTAINMENT PROMOS

by Roman Olearczuk

In this information age, the television viewer is being presented with a vast array of program choices, and there exists fierce competition for his or her leisure viewing time. Down through the years, television networks have discovered that, in order to inform the public about upcoming programs, announcements over title cards or end credits are not enough to grab the viewer's attention. Today the promo has evolved into a slick, information-packed, short video that utilizes many of the same effects found in other visual media.

To complement these exciting video images, promo audio has adopted several techniques found in the recording and film industries. This article will attempt to provide an overview of how various promotional spots are created at NBC, Burbank. Specific details of the process, ranging from off-line editing, through audio sweetening and mixing, hopefully will provide a useful insight into the diverse elements of audio production of NBC entertainment promos.

Audio Transfers and Scheduling

A promotional spot is scheduled for production by the Network On-Air Promotion department approximately two weeks in advance of its air date, when the Network Promotion Schedule arrives from the Media Planning department. At that time each promo producer receives his or her assignment, and then proceeds with an editor to view the show episode or movie to be promoted in one of four viewing rooms.

At NBC Burbank each viewing room is equipped with similar equipment: A Sony BVU-200 ¾-inch U-Matic video-cassette with remote control; a Sony CVM-1900 19-inch color monitor; and an audio monitoring system.

It goes without saying that before a show can be viewed it has to be transferred to ¾-inch U-Matic and one-inch videotape for promotional production. The On-Air Promotion Acquisitions department is responsible for obtaining and scheduling the viewing dubs. Show reels arrive daily from various production companies, in both 35mm film and one-inch video formats, and


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R E De F

DICK DEFREITAS — (Engineer, Musician)
President DeltaLab

DeltaLab

PRODUCING AUDIO FOR NBC ENTERTAINMENT PROMOS

can range in content from safety video masters to unsweetened film dailies and video iso shots. Depending upon the original format, dubs are fed by either the Telecine or Video Tape department. Each set of ¾- and one-inch video dubs includes matching SMPTE timecode and duplicate ID numbers. Upon completion, these show dubs are made available for promotional use.

The promo producer and his editor first watch the show in its entirety, after which the video is scanned through different segments and notes taken by the producer. These remarks would include timecode numbers where dialog and eye-catching visuals occur, as well as the initial concept of a promo script. When the viewing is completed, the promo producer books off-line video editing time.

Off-Line Editing: Audio Considerations

Off-line video editing is the starting point for the cutting of all NBC promos, with four edit rooms having been designated for the off-line production of such spots. Each room includes a CMX 340 editing system equipped with a five-by-one video/audio switching interface; a Conrac 13-inch text monitor; three Sony BVU-800 ¾-inch videocassette recorders with CMX I^2 interfaces; one Sony CVM-1900 19-inch color monitor for record preview; three Sony CVM-1750 17-inch color monitors for program viewing; an RCA MTC-551 timecode generator; a Yamaha M508 eight-in/two-out console; two Marantz MS-10 speakers, a Shure 561 dynamic microphone; plus an audio monitoring panel and video and audio patch bays.

During off-line editing, the spot's promo producer can develop his ideas with the editor at a significantly lower cost than in the on-line editing bay. The producer uses notes taken

Author Roman Olearczuk at the Quad-Eight/Westrex console in NBC's PPS-2 post-production facility.

from the viewing session to put together a concept for the promos. He then provides the editor with timecode locations needed to locate the different video segments, and previews the pieces in different sequences until an aesthetically pleasing spot emerges.

The editor builds the spot by recording onto a rough (work) videocassette loaded in one of the Sony BVU-800 U-Matic recorders. The ¾-inch rough cassette contains previously recorded timecode on audio track #2 for location address, and black-burst video to ensure clean edits. The promo video and audio is assembled from the same show cassettes used in viewing, and which are played back on the two remaining Sony BVU-800 VCRs.

Audio from the pair of playback video decks is routed to two audio channels of the mixing board. At present, the off-line editing rooms are configured for use with only track #1 audio from the VCR, track #2 audio being dedicated for SMPTE timecode. In the near future, software updates will allow the time-code to be placed on a third audio track — or address track — enabling the editor to place audio edits across both audio tracks. With the present system, however, the editor must provide notations under each audio edit for correct track placement in the on-line bay.

By using the console faders and some minor equalization, the editor adjusts the audio from the playback machines and then routes the signals to the CMX switcher, which automatically routes the audio to follow the video as the record edits are being assembled. The output of the switcher is brought back through another console fader for additional level adjustment, and assignment to track #1 audio of the record VCR.

An Edit Decision List, or EDL, is generated for the final sequence of video and audio edits, and provides the online editing bay with detailed editing information for compiling the

. . . continued on page 140 —

MULTITRACK ASSIGNMENTS USED IN NBC's PPS-2 POST-PRODUCTION FACILITY

It should be noted that the track layout below differs from that shown right, since tapes generated in PPS-2 seldom leave the room, and therefore can be considered to conform to an "internal" format:

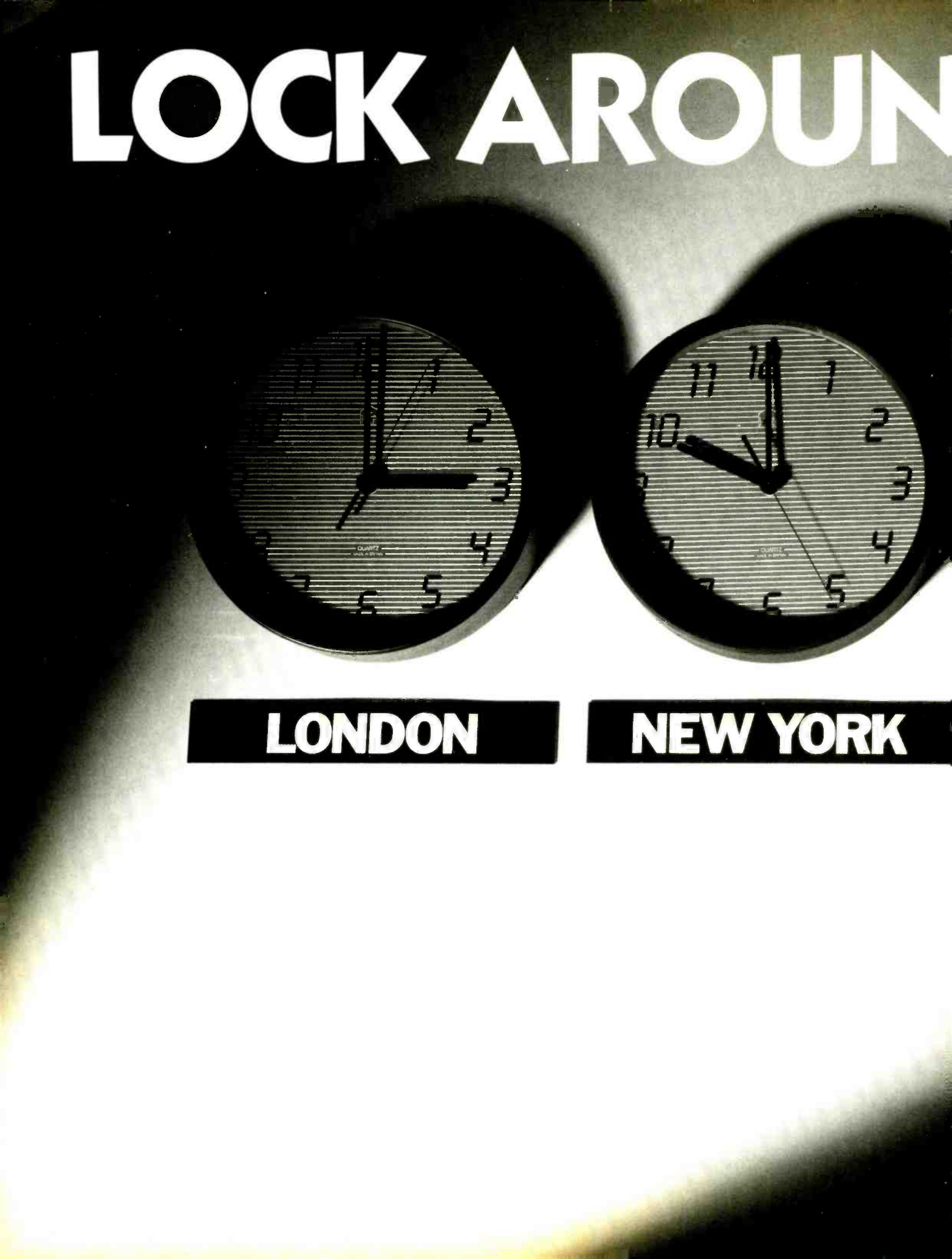
1 = Sound Effects	13 = Music Track A
2 = Sound Effects	14 = Alternate Music Track
3 = Sound Effects	15 = Music Track B
4 = Sound Effects	16 = *SPARE*
5 = Sound Effects	17 = *SPARE**
6 = Sound Effects	18 = FINAL MONO MIX
7 = Announcer	19 = *SPARE**
8 = Alternate Announcer	20 = *SPARE*
9 = Alternate Announcer	21 = MONO MIX-MINUS* Without Announcer and "windows"
10 = Alternate SOT	22 = *SPARE*
11 = VCR Layover #1 (SOTs + Main Dialog)	23 = *SPARE*
12 = VCR Layover #2 (SFX + BG Dialog)	24 = SMPTE Timecode

*For a "Three-Stripe" mix of music/effects/dialog, Track #17 would hold the Announcer Submix, #19 the SOT/Effects Submix, and #21 the Music Submix.

Shown below is the more "standardized" multitrack layout adopted by the majority of Hollywood studios, including NBC's PPS-1 post-production facility:

1 = *SPARE*	13 = Sound Effects
2 = DIALOG SUBMIX	14 = Sound Effects
3 = MUSIC SUBMIX	15 = Sound Effects
4 = EFFECTS SUBMIX	16 = *SPARE*
5 = *SPARE*	17 = Music Track A
6 = FINAL MONO MIX	18 = Music Track B
7 = FINAL MONO MIX (backup)	19 = *SPARE*
8 = *SPARE*	20 = Alternate SOT/ Production Sound
9 = *SPARE*	21 = Main/SOT Production Sound
10 = Sound Effects	22 = Alternate SOT/ Production Sound
11 = Sound Effects	23 = *SPARE*
12 = Sound Effects	24 = SMPTE Timecode

Obviously, alternative track assignments will be found at other post houses, particularly those mixing for stereo release.


LOCK AROUN
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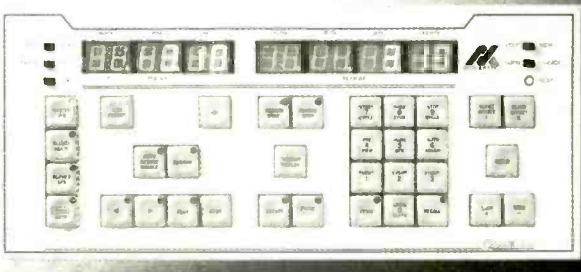
D THE CLOCK.

LOS ANGELES TOKYO



The ability to synchronise video and audio recorders is an increasingly vital facility required in studios all over the world. As much as three-quarters of today's audio recordings involve a visual aspect, and recording is more international than ever before. Basic tracks in New York, string and brass overdubs in London, dubbing in Los Angeles . . . modern international productions need an international standard for machine synchronisation, and there's really only one: Q-Lock by Audio Kinetics.

It's the same all over the world. The simple, uncluttered controls. The custom interfaces that suit your machines. The remarkable software capability. The integrated system with built-in expansion possibilities. It all adds up to accuracy and ease of use, and that means speed and creative flexibility. It means Q-Lock. If you're looking for an international standard, you've found it.

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Telex 230 194 781.

AUDIO KINETICS

— continued from page 137...

PRODUCING AUDIO FOR NBC ENTERTAINMENT PROMOS

promo from the show's master one-inch video reels onto a one-inch video submaster. The off-line editor also provides notes along with the EDL to aid the on-line editor in identifying video segments and audio dialog. Notations are made as to whether the audio edit is assigned to audio track #1 or #2 of the final submaster. On the one-inch video masters, track #1 audio is generally reserved for up-front dialog, with track #2 being used primarily for sound effects, music, and background dialog.

Once the video EDL for the working version of the promo has been stored on a floppy disk, a viewing version of the spot is generated for management approval. A dub, without timecode, of the producer's work videocassette, along with a reading of the script, is recorded onto the daily viewing cassette. (It should be noted that this cassette contains all the spots completed in the off-line edit rooms for that day.)

The promo producer reads his script as he watches the completed promo. Audio is routed from a pre-amp through a remaining console fader, and level and minor equalization adjustments made prior to assignment of announce to audio track #1 of the viewing cassette. If music edits are generated during off-line editing, this audio can be mixed along with the S.O.Ts (Sound On Tape) on to audio track #2, to present the full concept of the promo. Once approved, a series of promos, based on this rough cassette version, is scheduled for on-line editing.

On-Line Editing: Audio Considerations

The next step in promo production is the on-line video editing process. NBC's Edit 8 is a dedicated facility

PPS editor Jerry O'Neill cues a sequence of sound-effects carts during a sweetening session for "V — The Series."

Announcer Danny Dark (*left*) reading a promo script, while producer Steve Domier checks the timing against video script cues.

staffed by a four-person production team consisting of producer, video editor, an assistant editor, and Chyron character-generator operator. Two production teams, working successive shifts, are required to complete the large volume of promos requested daily.

The on-line edit bay houses a Grass Valley 1600-3K video switcher with memory; an Image Video routing switcher; a Quantel 5000 Plus digital effects generator; a CMX 340X editing system with a Conrac text monitor; five Sony BVH-1100 one-inch C-Format VTRs with CMX I^2 interfaces; five Conrac 13-inch color monitors for playback program viewing and two 19-inch color monitors for assembled and record preset viewing; a Quantum 12-in/four-out console; two dbx Model 160 limi-ters; a URIE Model 535 dual graphic equalizer; two Broadcast Electronics Model 3200-PS NAB cartridge machines; two JBL Model 4311B speakers; and an audio control panel.

The room producer details the production sequence to the technical staff on a priority basis. The editor loads the off-line EDL for a particular spot into the CMX 340X system and, by scanning through the EDL, can inform the assistant editor which master video reels are needed for assembly; up to four different one-inch playback reels can be accommodated at any given moment on four C-Format VTRs. The promo under assembly is recorded onto a predesignated submaster reel threaded on the remaining VTR, and used to generate all the "day-of-the-week" versions of the same promo — these spots all have the same video and audio as the submaster, except for different titles and "windows." The titles, generated on a Chyron Model 4100 MGM-EX system, visually announce the show's name, air time, and air day. The "windows" are pre-built video inserts with stars from the NBC series shows, announcing short on-camera tag lines such as "Tuesday!", "Tonight!", or "Let's All Be There!". (The inserts are referred to as "windows," because the overlaid border around the star's cameo resembles a person looking through a window.)

These three elements — the promo submaster, titles, and "windows" — are combined to make up each unique master promo. The increased production efficiency, with the elimination of redundant edits, far outweighs the additional generation loss in dubbing video and audio from the submaster.

Both audio tracks from the four playback VTRs are routed to eight of the Quantum console's line-input channels, program busses #1 and #2 being the designated feeds to the record VTR's audio tracks #1 and #2. During recording, the on-line editor has a choice of either manually assigning the audio via the program busses or, preferable, by switching the audio automatically via the Grass Valley audio switcher.

Input to the audio switcher is CMX-selectable between audio track #1 or #2 to follow video from any playback VTR; the switcher also allows the editor to program fixed-rate crossfades between any two audio sources. In addition, while shuttling to a particular timecode location, audio from the playback VTRs is muted by the switcher logic until each machine goes on-line, a feature that frees the editor from constantly having to disengage bus assignment switches during cueing.

The editor assembles the audio edits according to the spot's EDL, and adjusts audio levels so that the discrete segments are recorded at full level. Upfront dialog is placed on audio track #1 of the master VTR, while effects and music edits are assigned to track #2; each individual track is monitored by feeding the audio to a separate JBL 4311B loudspeaker. If the EDL contains overlapping audio edits, the editor has to place these pieces back and forth across both audio tracks in a "checkerboard" fashion. Generally, any

On-line editor Richard Russel adjusting incoming audio levels from one-inch video source reels, prior to EDL assembly of a promo.

PRODUCING AUDIO FOR NBC ENTERTAINMENT PROMOS

tight audio editing, equalization, processing, or limiting of these segments is dealt with later in the audio sweetening room. After the spot is assembled, the editor records a 1 kHz reference tone at zero VU (+4 dB) over the video slate, appearing 10 seconds before the promo begins.

Audio Sweetening of Promos

After the promotional spots have been assembled in the on-line video edit bay, they are brought into post-production sound room 2 (known as PPS-2) for audio sweetening. The facility accommodates a three-person, dual-shift production team consisting of a producer, an audio mixer, and an editor. [An article describing the design, construction and equipping of NBC's latest post-production sound room, PPS-1, can be found in the December 1984 issue of *R-e/p — Editor.*]

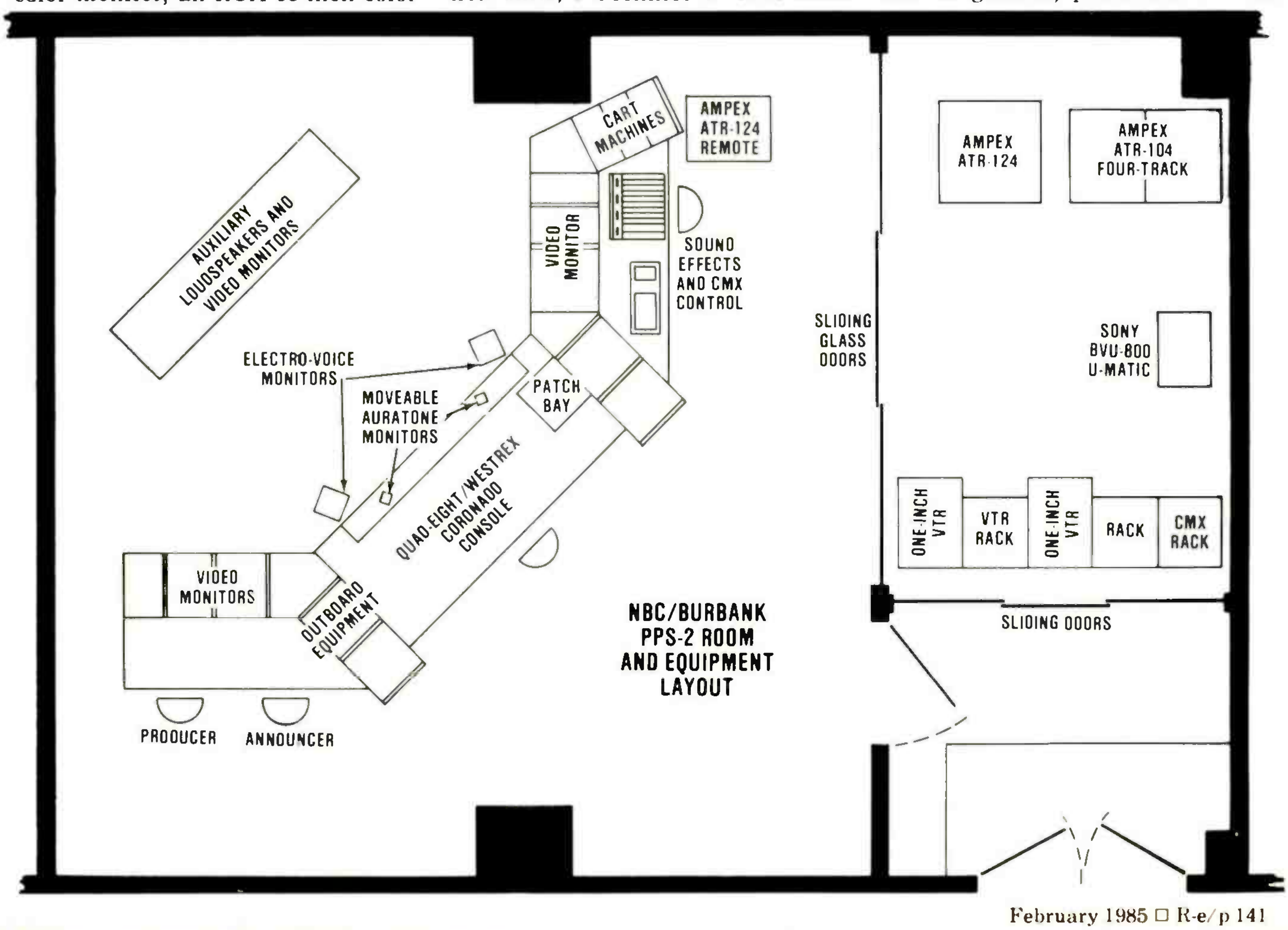
The producer and announcer work closely in a defined production work space located on the left-hand side of the room, equipped with a writing table, a Panasonic CT-110 11-inch color monitor, an RCA 19-inch color receiver, and AKG C414-P48 microphone, an AKG K-141 headset, two Auratone Sound Cubes, communication and headset controls, and Sonex acoustic foam paneling. The work space also has an unusual design feature — since there is no announce booth, the announcer is recorded right there in the control room. To prevent acoustic feedback the loudspeakers are muted during recording, and the narration tracks carefully monitored during playback; close-miking techniques help ensure minimal background noise pickup. In operational terms, the inconvenience of not being able to monitor the announcer during recording is far outweighed by the speed of communication between the announcer and the production team.

The audio console located in the center of the room is 28-input Quad-Eight/Westrex Coronado equipped with 24 group outputs. Monitoring is handled primarily by Auratone Sound Cubes and Electro-Voice Sentry 100A loudspeakers. Outboard equipment includes a Quad-Eight Model 5 reverberation unit; a UREI Model 537 graphic equalizer and Model 565 filter set; a Orban Model 526A desser; and Eventide H949 Harmonizer; EXR IV Exciter; Dolby CAT43A film processor; two dbx Model 165A limiters; two MICMIX Dynafex D-2B noise reduction units; a Technics SL-1600 MkII direct-drive Turntable equipped with a RTS pre-amp and KLH TNE 7000A Transient Noise Eliminator; and two Sony PVM-1900 color monitors.

On the right side of the room, the editor controls all tape transports, plus NAB sound-effect cartridge machines. Equipment located in this area includes a CMX 340X editing system with a Conrac text monitor; a Panasonic 11-inch color monitor; seven ITC 3D playback and two WP playback/record cart machines; a Yamaha M508 eight-in/two-out auxiliary console for off-cart sound effects pre-mixing; a Tascam 122B cassette deck; an Ampex ATR-102 with VSO; two Auratones; a video monitor switcher; and communication controls.

The CMX 340X editing system allows the editor to synchronize up to six different audio and videotape machines, and also to start eight audio carts under SMPTE timecode address. The remotecontrolled machines comprise an Ampex ATR-124 24-track; two Sony BVH-1100A one-inch VTRs; two Ampex ATR-104 four-tracks; and a Sony BVU-200 ¾-inch VCR. All remoted transports are housed in a soundproofed machine room directly behind the editor.

The daily production schedule revolves around the promo routine ship list. In general, production is com-



NBC/BURBANK PPS-2 ROOM AND EQUIPMENT LAYOUT

PRODUCING AUDIO FOR NBC ENTERTAINMENT PROMOS

pleted reel by reel on an availability basis from Edit 8, immediate attention always being given, however, to spots scheduled to air the same day. For example, there have been occasions when a promo has aired within minutes of completion. Although this situation can result in many tense and pressured moments, it becomes the perfect opportunity to instantly evaluate the audio being transmitted to millions of households on a nearby office television!

Shown on this page is a sample production rundown of the work undertaken in PPS-2. The routine lists the video reel number and the number of promotional spots scheduled for completion, each promo being numbered and titled. The duration, version, show description, and individual air date is also provided, along with the spot producer's initials. As an example, promo #27 on video reel #756K is titled as a "COSBY/TIES" spot. The N:30A symbol signifies a *n*etwork promo of 30-second duration, jointly promoting "The Cosby Show" and "Family Ties" episodes, airing on October 25, 1984. A station promo of this same spot, if requested, would have an "S" prefix, such as; S:30A. (Station spots closely resemble network promos, except that they are locally customized by the NBC network affiliates. Versions of these promos can vary from complete video and audio inserts added locally, to just a few lines of an additional station announce over the premixed audio tracks.) Coded suffixes, like the "A" in N:30*A* example above, denote different announce versions, and include: "A" — day of the week; "B" — tonight, "Q" — next; "T" — next (day of the week); "X" — exact date of show air, and "Y" — tomorrow.

As will be explained later, such information allows the mixer to uniquely set up the audio tracks to accommodate multiple audio versions of the same video spot.

NBC Memorandum

To: THOSE CONCERNED

From: BRAD SCOTT

Date: FRIDAY, OCTOBER 19, 1984

Subject: BURBANK PRODUCTION RUNDOWN
EDIT 8 ON LINE FRIDAY
DANNY DARK: PPS FRIDAY
PROJECTED
COMPLETION: 10/19 12:00a FEED/SHIP

TIME CODE		NET A&P PROD. #756K BB CVT: 10/19/84		PROMO AIR	SHOW AIR
	PROMO #25 N20:A	COSBY/TIES	BC	10/21	10/25
	PROMO #26 N:20B	COSBY/TIES	BC	10/25	10/25
	PROMO #27 N:30A	COSBY/TIES	BC	10/21	10/25
	PROMO #28 N:10A	THE CITY KILLER	EJ	10/22	10/28
	PROMO #29 N:10B	THE CITY KILLER	EJ	10/28	10/28
	PROMO #30 N:20T	SHATTERED VOWS	RS	10/22	10/29
	PROMO #31 N:20A	SHATTERED VOWS	RS	10/29	10/29

Typical Network Promo Production Rundown Schedule

Multitrack Audio Assembly

The first stage in the audio sweetening process is layover of audio tracks #1 and #2 from the videotapes onto tracks #11 and #12 of an Ampex ATR-124 multitrack. Also, regenerated SMPTE DF timecode (via the CMX I^2 machine interface) is patched from a Sony BVH-1100A one-inch C-Format VTR to track #24. Initial layover levels are set with the promo slate tone that was recorded earlier in the online edit bay. As the audio layover is monitored, the mix engineer re-adjusts and relays the tracks for any large variations in level, equalization, and program content.

The script (a copy of which also is shown here) is consulted for direction in separating the elements onto various audio tracks. The basic technique here is to place all the "up-front," message-carrying SOTs onto track #11. (Note: SOTs are dialog bits from the show used to play off the announcer's copy.) Background dialog and show sound effects go to track #12. Any sound inserts that have unwanted "blips" from on-line editing are cleaned up with either one- or two-frame edit trims, or with quick console fades and mutes during re-layover.

MICMIX Dynafex noise reduction is used on all these layovers to reduce tape noise. Also, a Dolby Cat 43A Film Processor, a single-ended, four-band noise reduction processor, is inserted to reduce background noises, such as hum, wind, and traffic, from production dialog.

On all these layovers, audio starts 15 frames into the actual video spot

On-air promotional script for a 20-second spot promoting both "The Cosby Show" and "Family Ties."

NBC ON-AIR PROMOTION

Length N:20A/B
Spot I.D. COSBY/TIES
Airdate 10/25/84
Producer BC

Audio	Video	Length	Sound EFX	Chyron
ANNC: THURSDAY/TONIGHT! ON THE COSBY SHOW! COS IS FIGHTING FOR CONTROL ESPECIALLY WITH A HEADSTRONG DAUGHTER!!			fight efx	
sot: "You go or you die!"	-Cosby			
ANNC: THE COSBY SHOW! THEN ON FAMILY TIES! WHEN MALLORY'S JOB IS ON THE LINE, WILL ALEX SAVE IT?			store efx	
sot: "You can be really cunning... wanna be!"	-Mallory			
sot: "Stop trying to butter me up!"	-Alex			
ANNC: FAMILY TIES!				
sot: "Thursday/tonight!"	-Alex (window)			
			Music	
			Show themes	

Date Script Submitted:________

```
TITLE: PPS MUSIC EDITS FOR "V/SERIES"  N:30 PREMIERE 01  VER 01  BIBB/JH
ISC  SUPER EDIT  MODEL 31  SN 818  NBC BURBANK
DROP FRAME CODE

001 060    A C        01:03:42:13  01:03:50:13  10:00:00:25  10:00:08:25
IN: 25 FRAMES AFTER FIRST VIDEO
OT: CROSSFADE WITH EDIT 002 AT CYMBAL CRASH ON DIANA'S I.D.

002 060    A C        01:05:17:26  01:05:28:05  10:00:06:11  10:00:16:20
IN: CROSSFADE WITH OUT OF EDIT 001 (SEE NOTE ABOVE)
OT: FADE ON SHOT OF ALIEN EYES
* ------------

003 060    A C        01:03:28:24  01:03:36:06  10:00:13:25  10:00:21:07
LOOP BEGINS
IN: BEGIN TO FADE ON SHOT OF ALIEN EYES

004 060    A C        01:03:27:09  01:03:34:22  10:00:21:07  10:00:28:20
LOOP CONTINUES
OT: NATURAL, OR FADE IF TRACKS IN NEXT EDITS OVERRIDE
* ------------

005 060    A C        01:06:58:29  01:07:04:16  10:00:22:23  10:00:28:08
FANFAIR OVER HISS
IN: NATURAL
OT: NATURAL
* ------------

006 060    A C        01:06:10:07  01:06:13:02  10:00:26:20  10:00:29:15
STINGER FOR CLOSING TITLE
IN: NATURAL
OT: NATURAL
* ------------
```

Music Edit Decision List for "V — The Series" promo (title-card version).

and ends 15 frames early, such blank areas being necessary for switching purposes during transmission (i.e., relay closures, etc.) to prevent audible glitches whenever the spots are programmed for air. (Audio blanks also are inserted during other stages of sweetening.)

Next, the number of versions of the same video spot are considered. Prime-time series promos with a duration of 20 seconds or longer will usually have "windows" at the end of each spot. As explained earlier, these "windows" are specially-built video inserts, with electronic graphics, that have a series star announcing when his or her show will be aired. For example, the Thursday night "COSBY/TIES" 20-second promo is frequently routined in three different announce versions (A, B, and Q) with three corresponding "window" tags. A "Thursday" announce at the top would have a matching "Thursday" window; the same is true for the "Tonight" and "Next" announces. Even though it represents three different on-air video promos, this set would be treated as a single spot from an audio point of view, the main body of the spot being exactly the same. Audio for each alternate "window" is laid onto discrete tracks, at the same time reference corresponding to the original window. Only during playback do the unique versions surface. (This technique will be explained in greater detail later.)

Following these layovers are the music transfers, the script again being consulted for music direction. A library of series show themes recorded in mono is maintained on half-inch, four-track reels, the themes having been transferred from original music scores or, at worst case, from the shows' sweetened air masters. The editor keeps detailed notes of which re-occurring musical cues work for the different length versions. Edit overlaps of two to four frames are used for music transitions in block promos, such as *A Team/Riptide/Remington Steele*, where each of the three themes is placed on a separate track.

In addition, a library of Network Production Music is also available. As new disks arrive, transfers are made to half-inch four-track tape and ¾-inch VCR. [The promo producers use these latter videocassettes to pick music cues for their spots during off-line editing.] Once determined, the cues are noted on the script or stored on an EDL floppy disk. With an EDL the post-production editor can call up complex music edit notes on the spot for assembly onto the multitrack tape.

Audio Edit Decision List

An example of detailed music edit notes is shown on this page. The promo, titled *"V" The Series (Title Card Version)*, was conceived to have a Fifties "B-Movie" trailer look. Specially designed graphic titles pop up over the visuals, promoting the action, romance, and mystery of the new television series. Six music edits were called for, as shown on the EDL.

The two rightmost columns of SMPTE timecode numbers included on the EDL indicate the record-in and -out times of a particular edit. (It should be noted that these times refer to the SMPTE timecode address for the case of the video located on the

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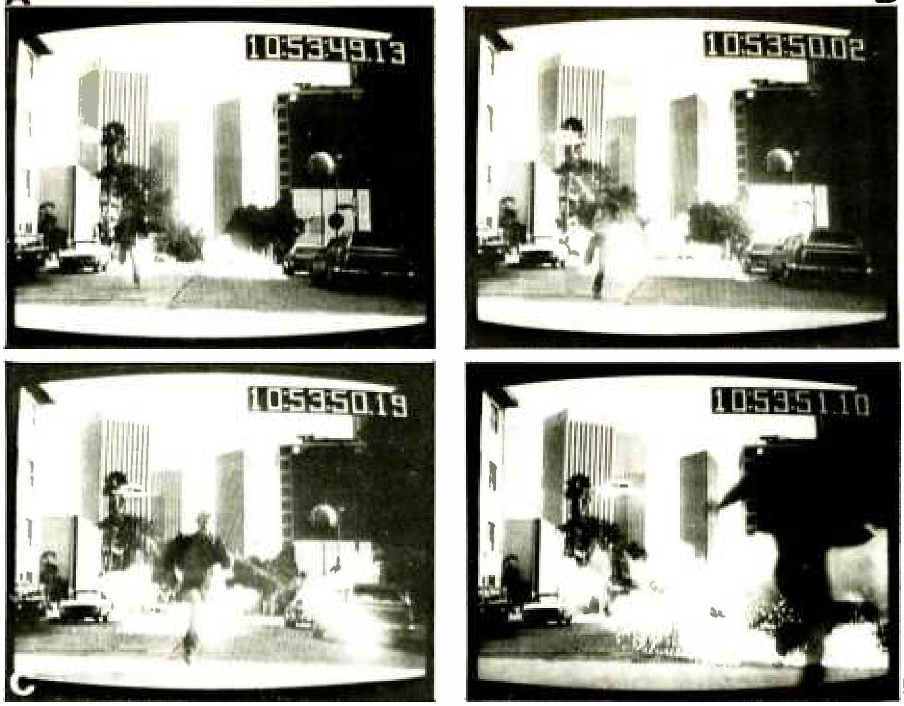
rough cassette. On the final one-inch video master, for this EDL to make any sense, timecode numbers must be updated for all the edits on the list shown here. To do this, the editor simply has to enter the new record-in offset number and the CMX will automatically update all the edit numbers. For the sake of this example, the reader should assume that the record-in/out numbers already correlate to the final spot numbers.)

For the first music edit, record-in starts at 10:00:00;25 and record-out stops at 10:00:08;25. The length of this edit is the difference between these two timecode numbers: eight seconds. The remaining set of numbers provide start and stop times of the playback music reel, the difference between these two sets of numbers also being eight seconds. In all cases, the durations of the playback edits and record edits should be equal.

The second edit has a duration of 10 seconds and nine frames. However, the record-in time of 10:00:06;11 occurs *before* the record-out time of 10:00:08;25 of the previous edit. This is the area where the off-line editor wants an audio crossfade to occur, as noted in the comments directly below edit #001.

To preserve audio tracks, these edits were transferred onto tracks #13 and #15 in a "checkerboard" fashion. For example, edit #1 is placed on track #13, while edit #2 would go to #15; the next sequential edit would then go back to #13. However, before edit #3 is laid in, edit #5 on track #15 must first be recorded. The engineer has to hear this edit in order to fade the loop of edits #3 and #4 (being built on track #13) under edit #5. Edits #5 and #6 are laid in with naturally occurring fades on both the in- and out-points. In actual practice these last two edits would be smoothed out, if needed, with additional fader moves.

A B



C D

E

This sequence of photos, A through E, shows a chase scene from "V — The Original" that was used in numerous promotional spots. Marc Singer, playing the resistance fighter Mike Donovan, runs away from an attacking alien Visitor spaceship. Five overlapping laser sound effects carts were added in sweetening to match the video. Also, a large explosion cart was used near the end of the sequence when one of the lasers hits a parked car. During the mix, each effect was mixed at increasing levels as the chase progressed. In photo A, at timecode 10:53:49;13, Donovan and the alien spaceship start to cross from left to right; photo B at 10:53:50;02 marks the first laser firing; photo C at 10:53:50;19 continues the chase motion; photo D at 10:53:51;10 marks a fire splatter; and photo E at 10:53:51;23 marks the point of the final laser zap and an explosion.

TIMECODE CLARIFICATION

In the past, there has been some ambiguity regarding the correct way of writing timecode locations, the traditional technique being to use colons to separate hours, minutes, seconds, and frames — for example: 10:34:53:11, to designate a timecode location of 10 hours, 34 minutes, 53 seconds and 11 frames. Confusion can arise, however, when only the minutes, seconds and frames data is provided — and, since few audio reels run longer than one hour, the leading digits are seldom necessary. To uniquely identify a timecode location, *R-e/p* has adopted the format of using a semicolon to mark the transition from seconds to frames, so that even a shortened-form timecode number can be written unambiguously. In the example quoted above, the new format would read: 10:34:53;11. We would be interested in receiving comments from *R-e/p* readers regarding the proposed timecode format — *Editor*.

Sound Effects and Announce

After the music layover, the producer, mixer, and editor jointly decide whether the production effects are usable. Sound effects that are noisy, poorly recorded, or mixed with music, are erased from the audio tape and rebuilt with library ¼-inch sound-effects cartridges. Additional sound effects (as requested on the script, or needed by visual action in the spot) are layed down on tracks #1 through #6.

With the CMX system, the editor runs the videotape at one-sixth play speed, to mark the record-in and -out times on an audio track for a particular effect to match video. The record-in SMPTE timecode locations derived from the marking session are also used to program the CMX General Purpose Interface to trigger the start of an audio cart. A 10- to 15-frame offset is added to the start times to allow for the cart to cue up to speed.

Action videos, such as car chases, crashes, shootouts, and comedy promos with missing laughs, need numerous sound-effects carts to make the sound believable. Shown on this page is an example of a three-second, five-frame video sequence from a promo to *V-The Original*, which required six sound carts to make it work. The editor can preset any combination of up to eight carts to trigger-in a programmed sequence of events, such carts being routed via a small Yamaha mixer to the main Quad-Eight console.

At this point the mix engineer determines the track assignment, level, and equalization for the effect. If audio tracks are readily available, each effect would be assigned to one track at a time. In a more complex action spot, however, to save tracks a combination of sound carts are premixed onto one track. MICMIX Dynafex noise reduction also is used on all of the effect layovers to keep cumulative tape noise under control. If required, audio fades are built in, while an Eventide 949 Harmonizer can be used to change the pitch of a sound to match video, as necessary. For example, a water hose cart can be used to give the illusion of the sound of a fire hose when set at a lower pitch. Or a fast moving conveyor belt can be made to sound slower through the Harmonizer. Echo and recirculating delays are also used to extend any short-duration effects.

Ideally, the announcer is scheduled to record his lines *after* the promos have been prepared with music and effects tracks. The announcer is recorded dry with essentially no equalization and no limiting, track #7 through #9 being reserved for the different announce versions. A complete reading of the script is recorded on track #7, while different announce opens and closes would be placed on the remaining tracks. For the alternate versions, the announcer reads the top copy until the first SOT sequence occurs. For the bottom tag, a punch-in recording is made during this last SOT and the announcer finishes the promo with the alternate closing copy. Later, the mix engineer can pick up the announce sections to assemble the correct versions during the mix and layback by switching tracks during the time the SOTs are replaying. On days when 28 promos are scheduled for announce, this technique not only saves time but also helps to preserve the quality of the announcer's voice!

Multitrack Remix and Layback

The final stages in sweetening are

. . . continued overleaf —

VIDEO MUSIC EDITING

A Contrast In Styles: Conversations with editors James Howard and Duke Kullman, and producer Lew Goldstein

Video editors James Howard and Duke Kullman have both been involved with on-going projects that require intricate editing of videos and music. For the past year, James Howard has been editing *V* promos, for the two original mini-series and the weekly series premiere, which include specially scored music that is edited to fit the video images. Duke Kullman is the editor responsible for the series of *Miami Vice* promos that feature multitudes of flashy images cut quickly to the beat of contemporary songs. The style of these spots is reminiscent of the rock videos currently being shown on MTV, and other cable music outlets.

Both editors achieve highly visual and sonically exciting spots, yet the editing styles are truly opposite to one another. James Howard cuts music to fit the video, while Duke Kullman cuts video to fit the music.

"V — The Mini-Series": Cutting Audio to Match Video

"Special music was scored for the promotion of the science fiction mini-series *V*," Howard recalls, "what we now call *V — The Original*. For the sequel, *V — The Final Battle*, we needed additional music based on *The Original* theme. Bob Bibb, the promo producer for *V*'s on-air promotion, went back to the composer with the assignment to open up the existing theme by giving it a fuller, more orchestrated sound. The resulting package gave us three major themes, and a group of 'stingers' and 'fanfares.' 'Stingers' and 'fanfares' are short bursts of music that have a finality to them: just three or four notes pulled from the master themes. Also, since we were going to do lots of editing to conform the music to an existing video cut, we especially required the tempo and pitch to be consistent. The completed music mix was delivered to NBC on half-inch, four-track format, and dubbed to ¾-inch video cassette for off-line editing. Both copies shared the identical drop-frame timecode reference.

"I indexed the music using the timecode numbers displayed as video characters on the ¾-inch cassette. The exact in and out points for each major theme, stinger, and fanfare were located. The major themes were broken down into shorter sub-themes, making it possible to go in at specific points and pluck out shorter elements. The indexing resulted in 28 segments pulled from the five-minute score." Examples of Howard's timecode sequences are shown below:

Cut	Code In (H:M:S;F)	Duration (S;F)
"ROBIN'S THEME"	01:07:28;10	10;08
"LOVE THEME" (with soft trumpets)	01:02:14;15	23;00
"MUSIC BOX"	01:07:14;00	08;05

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VIDEO MUSIC EDITING . . . continued —

"STINGER 2" (sharp kick — ominous tag)	01:06:10;07	09;00
"DOWNBEAT GONG"	01:01:08;13	03;05

"Names were assigned to the cuts to help in writing edit notes," Howard continues, "and in communicating with Bob and PPS, although the machines prefer the numbers.

"The off-line editing process began with a video cut of a 10-, 20-, or 30-second promo containing dialog, sound effects (if they were clean on the film print) and, in some cases, a scratch announce track. Existing audio tracks were important, because a transition from one music cut to another would be less obvious, if it occurred under a laser gun sound effect or dialog.

"Bob and I worked with the video and music in an ISC off-line 'cuts only' editing bay. I was able to move tapes on the record and playback machines frame by frame to achieve the perfect video-to-music and music-to-music match. The weakness of the system was the availability of only two audio record channels. Often a new recording wiped a previously laid track, and listening to a completed off-line 'mix' required some imagination.

"A single edit consisted of the 'in' and 'out' timecodes for the desired music cut, as referenced to the promo's video record 'in' time." Below is an example edit:

Play In	Play Out	Record In	Duration
01:01:08;13	01:01:11;18	10:00:02;15	00:00:03;05

As can be seen, the music desired was loaded on the playback reel at a timecode of one hour, one minute, eight seconds, and 13 frames; this is the Play In time. The start of the promo video on the record cassette was at 10 hours even. The music edit began two seconds and 15 frames from first video or at 10:00:02;15; this is the Record In number. The edit Duration for the video and music equalled three seconds and five frames. The Duration time also equals the difference between the Play Out and Play In times.

Each music edit was followed by notes for the mix engineer, which included the descriptive name for the cut ('Downbeat Gong'), a reference to the video it played with ('at shot of mothership over neighborhood'), and instructions if a crossfade or other transition into and out of the edit was necessary. The mixer, using these notes as a guide, would then determine points and durations of crossfades and the balance between SOTs (Sounds On Tape), voiceover, effects and music.

"Typically, a 20-second promo contained four to six music edits," Howard continues. "This data was delivered to PPS [one of NBC Burbank's post-production studios] on CMX-Format eight-inch floppy disks called Edit Decision Lists, along with printouts of the numbers and notes. Once the EDLs were loaded into the on-line computer, the PPS editor guided the CMX Auto Assembly program in building the selected music cuts at specific points on a 24-track machine. The mixer would determine track assignments aided by the mixing and layback, the promos being mixed according to the routine needs. To ensure synchronism with the various multitrack elements, a mono mix (minus any "windows") is recorded on track #18 in Sel-sync mode. In addition, a mono mix-minus (final mix minus announcer and "windows") is simultaneously recorded on track #21. In multiple version spots, the promo with the closest air date is mixed onto track #18. The remaining versions of the same promo are put together in layback with the mix-minus, announce pieces, and "windows."

Promo producer Bob Bibb (*left*) pointing out the video cut point to editor James Howard, where a music edit would occur during a recent "V" promo off-line editing session.

All of the sound elements that comprise the final mix are sent through a dbx Model 165A limiter (set to 6:1 compression ratio) for some added "punch" prior to being layed down. The mix-minus track, however, is *not* limited. Group assigns are used to regulate the level into the Model 165A, so that limiting occurs around the limiter's threshold. The output of the limiter is then adjusted to provide an average zero-VU recording level on the mix track. Grouping also allows the mixer the flexibility to raise or lower entire mix elements as required.

The final mix is monitored on an Auratone Sound Cube speaker, and checked on the producer's RCA domesticstyle TV receiver. Ideally, everything should be heard in the mix, but the main focus is the message. When audio levels for the announce and SOT are balanced and properly equalized, music and effects can be brought up. Effects that play in the clear, or are low frequency, are mixed at maximum levels. An EXR Exciter IV processor is utilized to improve clarity of the announce and production audio tracks. In addition, quite often the mix requires dynamic fader movements — if necessary the trailing ends of the announcer's words and SOTs are boosted with quick 3 to 5 dB moves. This technique yields an understandable message, without the reduction of the music and effects levels.

The audio layback process to the edited one-inch videotape is straightforward. Each promo is adjusted for average zero-VU record levels to both audio tracks on the one-inch VTR, the same master mono mix being fed to track #1 and track #2 for backup. If required, "window" tags are added to the mixed tracks during layback. Alternate announce versions, as mentioned earlier, are assembled with pieces of different announce takes across several tracks. Great care is taken in level matching and equalization to achieve a smooth sounding spot. For these alternate versions, the non-limited mixminus, announce, and "windows" are routed through the limiter, its output —unlike the mix mode — being fed directly to the one-inch VTR during layback. After this procedure is complete, all spots are checked against each slate reference tone. If incorrect, the tone is immediately replaced.

These techniques work quite well for short spots of up to one-minute in duration. For longer mixes, such as two-minute movie fills and 10-minute affiliate convention presentations, the mix has to be broken up into three individual submixes of music, SOTs plus effects, and announce. This mix, analogous to the film industry's three-stripe dub, allows the engineer to break down a complex mix into smaller submixes, and to easily accommodate the inevitable script changes.

The music submix is recorded first, onto track #21. Even though all tracks are being monitored in Sel-sync mode, and at zero setting on each input fader, only the music edits are adjusted for levels, crossfades, and equalization. When the music submix

VIDEO MUSIC EDITING . . . continued —

notes.

"The promo, *V: 30-second Title Card Version*, is a good example of how the catalog of score segments was used to match the video pacing. Designed to have the look of a Fifties movie trailer, the promo contained 'bigger than life' clips that served as background for the bold title graphics. The music had to change texture as the video transitioned through three phases; in other words, when video and titling said 'ALL THE ACTION,' the music track had to have high energy. We chose a 'soft' music bed as the two main characters embraced, and the title card read 'ALL THE ROMANCE.' 'ALL THE SUSPENSE' required using the stinger called 'Downbeat Gong' as the title swung into view, and the sharp kick of 'Stinger 2' when Diana placed her weapon to Donovan's neck. Twelve music edits were used for the *Title Card* promo."

"Miami Vice": Cutting Video to Match Music

"Lew Goldstein [on-air promo producer] and I are somewhat limited in the music we can choose for our spots," says editor Duke Kullman, "because we have to use music that's in the show. Before we start editing a song, we look for something that has a real driving rhythm — enough so that when we do cut to that rhythm, it accentuates the video as much as possible. It can't be too slow, although if we cut on downbeats or intermediate beats, we can use slower pieces of music. In our shorter promos we can't really use vocals; 10 seconds just isn't enough time to establish them. So for the shorter promos, we primarily look for instrumentals with a strong rhythm. This usually turns out to be the intro into a song, part of the bridge, or instrumental later in the music.

"For the premiere promo of *Miami Vice*, we used the Phil Collins' piece 'In the Air Tonight' as the music. The 30-second version had five music edits in it. We knew we wanted a strong opening, and the dynamic drum fill from the middle of the song fit the bill. We had an idea of what video we wanted to use, but we weren't sure if it would work.

"I must say that once we had the music edits done, the opening shots from the climactic gun battle in the show fell in relatively easily. Often times when we're trying

VIDEO MUSIC EDITING . . . continued —

to put motions or gunshots on certain beats, a number of times we have to move the video around by a few frames before the video and music elements complement each other to our satisfaction.

"The chances of getting a drumbeat and getting shots out of a show with gunflashes going off in the same rhythm as the music is slim — they're never going to match up perfectly, but after some minor adjustment, this one did. Sometimes you get lucky.

"For the next music edit we chose the music with the title lyrics as the body of the promo. The third edit was taken from some lyrics later in the song that we knew we had video for. We wanted to end the promo with the lyric 'When I Remember,' so that became the fourth edit.

"But musically those two adjoining edits didn't really work: They were from completely different areas of the song, and contrasted too much; one was loud, and the other was soft. To cover those two pieces of music, we found a descending guitar slide and put it over the top. These edits were done on a multitrack tape and mixed together, which covered up the music contrast on that edit.

"As for the video, we were looking for a visual that would work over that long bending guitar, and luckily we found it. A guy had been shot and propped up against a wall. Just as the main characters, Sonny Crockett and Ricardo Tubbs, locate their man, the body falls over towards the camera. This shot worked perfectly with the descending guitar slide.

"Sometimes we spend hours and hours — more hours than I even like thinking about — doing the music, and sometimes it all falls in place quickly. Even a few edits can take a long time to sound right musically, depending on how drastically you change a song. It's not hard to cut something to a beat, or to cut off a phrase, or move a phrase to a different place in the music. What *is* hard is to make the edited song sound decent musically! I don't want to offend the artists and/or record companies who are kind enough to let us use their music and/or property.

"When we get the music for a show, one of the first things we do is find out the time between the beats. We'll lay over all the music onto a ¾-inch videocassette with SMPTE timecode. Then, I'll mark in and out times, using a CMX editing system, to the syncopated beat on the machine as music is rolling by. Most songs are 4/4, so a quarter-note is one beat. Instead of counting from one quarter-note to the next, I'll count between the eighth-notes, which are twice as fast as the quarter-beats, to see how many frames there are between these notes. Usually that'll remain constant throughout the piece.

"For most compositions that we use, it seems to turn out to be somewhere between eight and 10 frames [260 to 330 milliseconds] between the eighth-notes. If I only counted between quarter-notes, we wouldn't have enough flexibility in putting video to the music. Once I find what the magic number is, we'll try to utilize that as a reference to place the video shots all the way through the spot; it makes things considerably easier.

"For the *Hit List* episode we used a number of shots that were eight to 10 frames long. In most other types of editing, 10-frame shots are just too quick, but once a

Editor Duke Kullman (*right*) and promo producer Lew Goldstein discussing concepts and ideas for "Miami Vice" promo during off-line editing session.

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is satisfactorily completed, the engineer then begins to record the SOTs and effects submix on track #19. During this stage, the announce track and the completed music submix are being monitored, a technique that enables the mix engineer to use the overall sound to balance the SOTs and effects submix.

Now only the announce track is left to be mixed onto track #17, while the previous music and SOTs/effects submixes are being monitored. In this way, the engineer can ride the announce track, and change equalization between punch-ins, without having to concentrate on the other elements in the over-all balance. Once the announce submix is finished, the complete mix can be retrieved from a combination of these three tracks, by setting to zero their respective input channel faders. During layback, a group-assign fader is used to balance all three tracks for overall level to one-inch master VTR.

On-Air Audio Considerations

The finished promos for the day are simultaneously transferred to two composite reels, a procedure that frees up the individual video reels for future promo production, and also enables the videotape operators to satellite feed a continuous set of priority promos to NBC's New York headquarters. In addition, one of the composite reels is shipped immediately to New York, where it becomes the primary air copy. The satellite transmission is mainly used to deliver promos airing in the next 24 hours.

In both network centers, the composite reel is used to make a pair of duplicate two-inch video cartridge copies of each promotional spot. Audio on all cart dubs is recorded at a -3 dB level, referenced to the source video's line-up tone; this reduced level compensates for the difference in loudness between commercials and network programs. The promo carts are loaded, along with commercials and public service announcements, into two RCA TCR-100 videocart machines as they are schedules throughout the day. (Note: The system format for this machine is the two-inch, reel-to-reel quadraplex video standard, with one audio track, running at 15 ips.) Both copies of each promo are placed into the two cart machines, one serving as the on-air version, and the other as backup protection. The cart machines are then remote controlled for program air by a Control Data 636 computer located in Switching Central.

* * *

Audio production of NBC Entertainment Promos is as challenging as it is fast-paced. The process has to be efficient and streamlined, with little time for experimentation, in order for both sound crews to produce over 100 promotional spots a week. Yet, at the same time, there is a never-ending search for new techniques, new equipment, and new ideas to make future promos attract the attention of television viewers. With the advent of Stereo Television, it will be interesting to see and hear how each of the networks make use of this technology in their respective promotion departments. ■■■

In addition to those individuals already mentioned in the text and photographs, I would like to thank the following people for their assistance in the preparation of this article: Craig Curtis, Jim Keller, Al Kennedy, Peggy Keppel, Dave Klandrud, Rick Rettig, and Stowell Werden. I would also like to acknowledge the efforts of audio mixer Bud Stalker and editor Dave Dunlap, who make up the other half of the NBC PPS-2 production team — *RO*.

VIDEO MUSIC EDITING . . . continued —

rhythm is established between the music and the video, certain visuals can play very well in that length of time. This also means a *lot* of edits — we probably average between 25 to 35 edits in a 20-second spot."

"What's interesting is that a lot of times we put these short images in and they don't work until we see everything else around it," concedes producer Lew Goldstein. "For example, in the promo we cut for the *Miami Vice* episode *Hit List*, we had a guy with a sniper rifle looking through a scope for eight frames, then the next shot was a tight shot of the trigger. The video only works with both shots, because the action continued into the next shot and, logically, your mind processes the visual information. If you only saw the video of the guy with the rifle by itself, you're going to wonder, 'What was that?'

"*Miami Vice*" is a unique, stylized show. Not only does the cinematography set the mood, the musical score by Jan Hammer is also outstanding. Anthony Yerkovich and Michael Mann Productions had set the tone for this show. We capsulized the same style and pacing through our promotional spots, and portrayed the elements of *Miami Vice* by utilizing a kind of Music Video approach to tell a small story in 30 seconds or less, with quick video cuts to music, and only minimal announce copy; i.e., 'Miami Vice . . . Friday!'

"For the series premiere promos we decided, as Duke has mentioned, to use as our theme Phil Collins' recording of 'In the Air Tonight.' We emphasized the theme with slick video images: two vice cops driving their black Daytona Ferrari convertible through the dark shadowed streets of Miami, with the wind blowing in their hair. When you think of darkness and people lurking in the shadows, evil and death immediately comes to mind. In this particular show, evil and death are prevalent throughout.

"Early research sampling of the Phil Collins' promo indicated that younger people in the 25- to 35-year-old age group were able to understand what the show was about from the short brief cuts contained in the promo. The research group knew the actors were cops; they knew there was lots of danger, intrigue, some sex, and lots of action. When combining these terrific visuals and great music, you end up with a very exciting promotional piece!" □□□

FACILITY SPOTLIGHT: CENTURY III TELEPRODUCTIONS
Designing and Equipping a New Audio-for-Video Post-Production and Sweetening Studio

by Vin Gizzi

Century III Teleproductions, currently New England's largest video production and post-production facility, contacted this writer in Fall of 1983 to discuss the design of a new audio-for-video suite. At that time, Century III was already one of the major East Coast video operations, and growing rapidly. It had become aware, as have many video companies, that audio mixing services beyond the capabilities of a typical video edit room increasingly are being requested by its clientele. While video editing has become more sophisticated, and user friendly, the lack of equally sophisticated sound completion capabilities has come to stand out more clearly as a distinct problem for many video post-production companies.

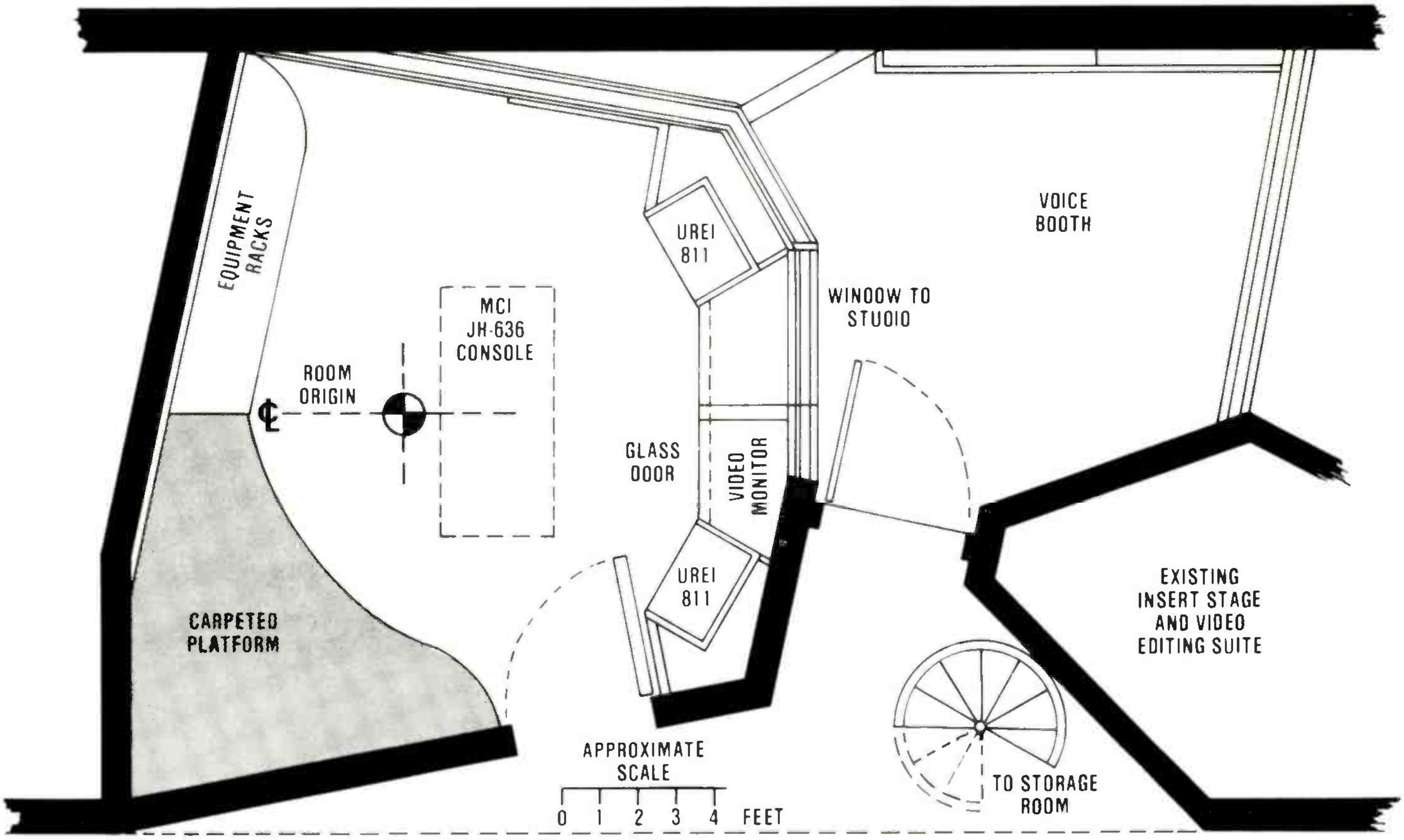
This article describes the design and equipping of Century III's recently completed Audio-Sweetening/Mix to Picture studio.

According to Century III president Ross Cibella, "We had been pioneers in many areas of the production and post-production business, and wanted to lead the way in audio as well. We were as much concerned with design as with equipment in this project, because we had to have a room quantitatively capable of *very* high-level work."

Defining that "high level of work" and deciding how it would be accomplished was the first order of business in the design process.

The type of sound work required in a facility is determined very much by the character of the company itself and, in that regard Century III has an interesting history. In 1977 a young, ambitious entrepreneur named Ross Cibella opened an audio studio in Boston that was geared toward the advertising market. Above and beyond running a successful company, he hoped to accomplish two things: to develop strong relationships with the people producing commercials; and to establish himself in the eyes of the local financial community as an adept businessman. He reasoned that if clients, investors, and bankers were impressed by the performance of his audio company, they would follow, and support him in a larger and more comprehensive video facility.

His instincts proved to be right, and in 1979 Cibella opened Century III



LAYOUT OF CENTURY III TELEPRODUCTIONS' NEW "AUDIO SWEETENING/MIX TO PICTURE" ROOM

Teleproductions. Starting with one video editing room and a staff of six, the facility grew very rapidly. Today Century III includes four video edit suites, two sound-mixing rooms, a film-to-tape transfer suite utilizing Bosch and Dubner equipment, and such special effects devices as the Quantel Paintbox and Ampex ADO. In addition, Century's facilities also are made available to corporate as well as commercial clients, through a new division whose orientation is more towards the sales and marketing presentation needs of larger organizations.

The company is active in shooting and completing television comercials, both on film and videotape. A popular and cost-effective hybrid is to shoot on 16mm and 35mm film, transfer the camera negative to tape, and then edit and release the commercial on video. Century III also turns out television entertainment programming, a recent example being *Healthbeat* for Metromedia.

Such a wide variety of work, and the lure of that new field called Music Video, indicated the necessity for a sound-mixing installation of considerable sophistication and flexibility.

Design Concepts

An audio-for-video mixing facility must satisfy several requirements that are not usually met in a single audio control room. The audio monitoring system must be equal to that of a state-of-the-art recording studio. The reason for this is not only because any particular project may involve mixing or remixing a program destined for a "Hi-Fi" video release format (videodisc, Beta Hi-Fi, FM simulcast); virtually every project edited in the facility passes through the mix room for some form of correction or processing. As a result, such a room function as the final quality-control check for audio tracks. For this reason, it was considered important that the new room should comprise an extremely accurate monitoring environment, much like the situation required for disk mastering.

Mixing capabilities must also match those of a recording studio, since audio post sessions may involve the remixing of music multitrack tapes to picture. Consequently, the console must be equipped with comprehensive signal-routing capabilities to handle the elaborate "ping-ponging" and track layering used while building up elements for a video mix. Audio processing gear also must run the gamut — from digital reverb and effects devices, to notch filters and noise gates needed to clean up noisy dialog tracks.

Audio/video synchronization capabilities should also equal those provided in the video edit rooms since clients often move directly from the edit room to a mix session, and expect equivalent performance. The mix session may also require the replacement and re-editing of a great deal of audio material to match video edits. It is not uncommon during a complicated video editing session to pass certain scenes through many recorded generations to accomplish a variety of optical effects. And while modern VTRs can maintain excellent quality and signal-to-noise, the re-recorded video signal, degradation of audio material during the transfer is a different story altogether.

Multiple audio generations become quite obvious even on Dolby-encoded one-inch videotape. It is common practice, therefore, to replace all or most of a program's audio track in a separate audio sweetening session, scheduled after the edit is complete. This procedure implies the use of an audio editing system that offers most of the features and flexibility found in a good video editing system.

Of course, the audio tape machines used in an audio-for-video mix room must be state-of-the-art to preserve quality during transfers, and allow noiseless and gapless punch-ins. The ability to punch-in or "pickup" on an audio track is an important feature to have available during a mix because usually it is impossible to mix an entire program in just one pass. It is

often impractical to mix more than a minute or so of program at a time, because of the number of adjustments and changes needed in an active or heavily edited scene.

Lastly, an audio-for-video mix room must be capable of accommodating fairy large groups of people (there are a lot more cooks involved in a television show or commercial than in a record project); all of them need to see and hear what's going on during the session.

Recording Equipment Selection

Century III's VP of audio, Rob Hill, had strong feelings about the production equipment needed for the new Mix to Picture room: "The equipment selected was based on projected trends in audio-for-video, and the advent of Stereo TV. We chose a [Sony] MCI JH-636 board with 32 inputs, and we're retrofitting every input with custom parametric equalizers from Troisi Engineering. These have a high-pass filter that is continuously variable from 31 Hz to 16 kHz — that's a big help in cleaning up location tracks."

Hill had been an engineer at Century III Audio, and later at Intermedia Studios, a Boston recording facility that Cibella purchased in 1978. Intermedia was also a success but, knowing that his future lay in the television business, Cibella sold it in 1983, bring in Hill over to Century III Teleproductions to run the new audio-for-video department.

The final equipment package includes the JH-636 console with automation, Otari MTR-90 24-track, Studer A810 and Otari MX-5050 MkIII ¼-inch stereo machines, the former with center-track timecode. A BTX Softouch SMPTE synchronizer runs all the audio transports, in addition to RCA 800 one-inch and Sony BVU-800 ¾-inch video machines, and remote Magna-Tech Model 636 35mm mag transport. Outboard equipment is extensive, featuring the now-common array of digital processors, including a Lexicon 200 reverb, 1200 Time Compressor, and Prime Time II, as well as a bank of compressors, limiters, and noise gates. Monitoring is via UREI 811 Time-Align loudspeakers and Auratone Sound Cubes, powered by Bryston 4B and Crown D75 amplification. A motorized Kloss Novabeam projection screen is available for lowering between the UREI monitor speakers.

"The Mix to Pix suite had to interface with five video edit suites, two stages, radio production and our duplicating center," Hill points out. "When you consider that interfacing includes audio, video, and control lines you get some idea of the complexity and scope of this project."

Other rooms at Century III include SuperSuite Edit C (*above*), which features a Soundcraft console, and (*right*) Film-to-Tape Transfer Suite.

Acoustic Design Approach

A space had already been designated for the audio suite, and Century III management decided to sacrifice some noise isolation in favor of retaining maximum floor space for the rooms. Very little music recording at high levels was planned for the studio, and the function and location of adjacent offices and facilities presented no leakage problems.

After examining the site, I decided that I could meet most of my design criteria for a control room by adding one new side wall that would produce left-right symmetry without eating up too much floor space. The existing walls were solidly constructed of two layers of gypsum on each side of two-by-six wood studs, and would be adequate for noise isolation and would be stiff enough to provide a foundation for the subsequent acoustic treatment. A new front wall was needed in the control room, in addition to extensive work in the studio area. Although the isolation requirements for neighboring rooms were relaxed somewhat, it was important that the mix room be isolated from mechanical noise generated by equipment elsewhere in the building.

The new audio suite, which is located on the second floor of the building, would lie directly beneath the 30-ton air conditioning units — all of Century III's rooftoop system. The units produce a great deal of noise, most of it low-frequency, and much of it was finding its way into the proposed building area. I specificed new spring mounts for all air-conditioning units which, combined with resilient couplings installed on each section of duct leaving or entering a unit, eliminated structure-borne noise problems. Acoustic noise was another matter, however, and an additional ceiling was needed in the mix suite to control it. In order to retain maximum headroom, new ceiling joists were resiliently hung from the existing structure, and faced with two layers of ⅝-inch gypsum board.

Another troublesome source of noise proved to be the main electrical service panel for the building, which was located in the area intended for the voice studio. Since the relocation of hundreds of electrical cables was considered completely impractical, provisions had to be made to deal with the abundant 60 Hz hum radiating from the service boxes, while still allowing regular access to them. The solution was to close off as much of the service area as possible with a heavy wall. To effect the required sound isolation, doors made of a sandwich of chipboard and sheetrock

— the Author —
Vin Gizzi is a principal of Benchmark Associates, a New York City firm specializing in the design of technical facilities.

were then fitted to the openings. The doors are hung from piano hinges along one side, and close against wood stops faced with neoprene seals.

Because of the limited depth available in the room, it was decided to lay out the control room with a client area located to the left of the mixing engineer, rather than in back of him. This layout resulted in a room similar to many recording studio control rooms, with tape machines being situated to the engineer's left, and outboard equipment behind him.

I knew immediately that, in order to fit all necessary elements into such a small space, it would be necessary to combine architectural, acoustic and mechanical functions in a complex way. Overhead polycyndrical diffusers in the mix room double as membrane absorbers, controlling the reverberation time of the room in the 100 to 200 Hz range. The diffusers also house low-voltage spotlights, mounted in small sealed cavities, that provide illumination at the console. The geometry and finish of the diffusers — random-width oak boards with a high-gloss varnish — also provide a visual focus for the room's interior design.

A suspended ceiling panel over the room's central area conceal air-conditioning ducts, and holds recessed lighting for the client area. The panel is faced top and bottom with two inches of Owens Corning 703 rigid fiberglass, covered with stretched fabric.

The overall acoustic design was aimed at keeping the amount of absorptive material to a minimum by placing it carefully. This was done for two reasons: the need to conserve every inch of space; and to keep the room as diffuse and live as possible. Most of the sound absorption material is located in the ceiling to eliminate disturbing overhead reflections originating from the monitor speaker positions. Ray diagrams were used to locate areas of the side walls that could be sources of early reflections at the mixing position; these are also covered with absorbent material. A small section of the rear wall is also treated to prevent flutter echoes between it and the control-room window.

The front wall features a large arched window divided functionally, but not visibly, in two; the left side looks conventionally into the studio, while the right side forms the face of a blacked-out cavity that holds the video monitor. The actual division of space happens behind the glass, so that no seam or mullion is visible from the front. If the recording studio is not in use and lights are off there, the observer sees only a large tinted panel with the video monitor "floating" behind it.

The Acid Test

Thanks to the high level of craftsmanship practiced by Century III's own building crew, the operating staff considers that the project turned out beautifully. Sessions in the room have been described by clients as smooth and remarkably free from the usual start-up bugs. Synchronizing audio tape machines to remote VTRs and even remote film dubbers is now a routine procedure.

The facility has been very heavily booked since it opened, and it looks as if Century III's assessment of its market was correct. Client reaction has been very good, says Cibella. "We're especially pleased with the acoustic performance of the mix room. Mixes relate very well to the real world, which our clients consider very important. Clients also appreciate the quality of the product they're getting, and the speed with which it's produced.

"Of course the real measure of a room's success is in its popularity," he continues. "Mix to Picture is in use about 16 hours a day, which is terrific. But it's very gratifying to use — having tried to be relatively visionary in planning this room — to have our instincts proved out, not only about our marketplace, but also the design we thought would work for it." ■■■

The Directory

R-e/p's Product Listing of TAPE MACHINES AND SYNCHRONIZERS

Even though the company's products are no longer being manufactured, listings are provided for Ampex Corporation and Soundstream, to provide suitable information for purchasers of used equipment.

A.C.E.S.
U.S. Distributor: Mammoth Marketing
P.O. Box 6493
Thousand Oaks, CA 91359
Phone: (805) 496-2969

TR-45

Tracks/Speeds: Four-track; 15 and 30 ips.
Formats: Half-inch.
Frequency Response: 20 Hz to 20 kHz, ±3 dB.
Distortion: 0.8% (tape).
Signal-to-Noise Ratio: -62 dB A weighted.
Wow and Flutter: 0.04% typical at 30 ips.
Selected Standard Features: Switchable EQ; remote return to zero.
Price Range: $6,450.

TR-24, TR-16

Tracks/Speeds: 16- and 24-track; 15 and 30 ips
Formats: Two-inch.
Frequency Response: 44 Hz to 18 kHz, +1.5 dB/-1 dB (record/play).
Distortion: 0.8% (tape) at 320 nWb/m.
Signal-to-Noise Ratio: -64 dB average; 60 dB A weighted.
Wow and Flutter: 15 ips = 0.06% maximum; 30 ips = 0.04% maximum DIN weighted.
Selected Standard Features: Remote standard, nine-memory auto locator optional; EQ switching follows tape speeds.
Price Range: TR-16: $14,950; TR-24: $18,950; (Remote Auto Locator: $1,000).

TR-25, TR-225

Tracks/Speeds: Two-track; 15 and 30 ips.
Formats: Quarter-inch and half-inch.
Frequency Response: 20 Hz to 20 kHz, ±1.5 dB (record/play).
Distortion: 0.8% (tape).
Signal-to-Noise Ratio: O.O3% typical at 30 ips.
Wow and Flutter: 15 ips = 0.06% maximum; 30 ips = 0.04% maximum DIN weighted.
Selected Standard Features: Auto-switching EQ; remote zero return.
Price Range: $4,900.

1610, 1610/8

Tracks/Speeds: Eight- and 16-track; 15 and 30 ips.
Formats: One-inch.
Frequency Response: 45 Hz to 21 kHz, +2 dB.
Distortion: 0.8% (Tape).
Signal-to-Noise Ratio: -64 dB at 15 ips; -69 dB at 30 ips.
Wow and Flutter: 0.05% at 30 ips.
Selected Standard Features: Nine-memory Autolocator; auto switching EQ.
Price Range: 1610: $11,950 including locator remote; 1610/8: $9,950 including locator remote.

TR-532

Tracks/Speeds: Eight-, 16-, 24-track
Formats: One- and two-inch.
Frequency Response: 30 Hz to 20 kHz, +1 dB/-3 dB at 15 ips (record/play); 50 Hz to 20 kHz, +1/-3 dB at 30 ips.
Distortion: 0.8% (Tape).
Signal-to-Noise Ratio: -63 dB at 30 ips A weighted (24-track).
Wow and Flutter: 0.04% weighted.
Selected Standard Features: ATC 35 memory locator.
Price Range: TR-24: $39,550; TR-16: $29,200; TR-8: $25,950.

AKAI
U.S. Distributor: IMC
1316 E Lancaster
Fort Worth TX 76102
Phone: (817) 336-5114

MG1212

Tracks/Speeds: 12 audio plus two data-control
Formats: Custom half-inch cassette.
Frequency Response: 50 Hz to 20 kHz, ±3 dB.
Distortion: 0.5% 0 dB, 1 kHz; -3% +12 dB, 1 kHz.
Signal-to-Noise Ratio: -94 dB (NAB auto), reference 3% THD at 315 Hz.
Wow and Flutter: 0.03% WRMS.
Selected Standard Features: Built-in 12-channel mixer; digital bussing; auto punch in/out; auto mute; auto location; dbx noise reduction.
Price Range: $6,995.

AMPEX CORPORATION
401 Broadway
Redwood City, CA 94063
Phone: (415) 367-2011

ATR-116/124

Tracks/Speeds: Eight-, 16-, 24-track; 7½, 15 and 30 ips.
Formats: One- and two-inch.
Frequency Response: Operating level 370 nWb/m using Ampex 456 tape. At 30 ips: ±2 dB, 40 Hz to 30 kHz; at 15 ips: 25 Hz to 20 kHz.
Distortion: (Applies to overall record/reproduce distortion, reference 370 nWb/m.) Even-order distortion is less than 0.1% at 1 kHz; third harmonic distortion at 1 kHz less than 0.3%; SMPTE IM distortion less than 4% of 740 nWb/m for NAB equalization; less that 3% at 740 nWb/m for IEC/AES equalizations.
Signal-to-Noise Ratio: Unweighted. 16-track: at 30 ips, -72 dB; at 15 ips NAB, -69 dB. 24-track: at 30 ips, -69 dB; at 15 ips NAB, -66 dB.
Wow and Flutter: Unweighted. 30 ips NAB 0.03% RMS; at 15 ips 0.04%.
Selected Standard Features: Single point search-to-cue; variable speed shuttle, monitor memory function; four switchable EQs; 16-inch reel capacity; variable speed operation; auto-bias (optional).
Price Range: Not applicable.

ATR-100

Tracks/Speeds: Full-, two-, and four-track; 3¾ to 30 ips.
Formats: Quarter- and half-inch.
Frequency Response: Operating level 370 nWb/m using Ampex 456 tape: ± 2 dB, at 30 ips, 35 Hz to 28 kHz; at 15 ips, 20 Hz to 20 kHz.
Distortion: (Applies to overall record/reproduce distortion.) Even-order distortion 1 kHz at 370 nWb/m is less than 0.1%; third harmonic distortion at 1 kHz less than 0.3% at 370 nW/m; SMPTE IM distortion less than 1% at 370 nWb/m.
Signal-to-Noise Ratio: Unweighted. Full-track: at 30 ips -77 dB; 15 ips NAB -73 dB. Two- and four-track 30 ips: -72 dB; at 15 ips NAB -69 dB.
Wow and Flutter: NAB RMS unweighted 30 ips 0.03%; at 15 ips 0.04%.
Selected Standard Features: Optional version for disk mastering; half-inch, two-track mastering operation (option); multi- or single-point search-to-cue (option); variable speed operation; selectable four-speed operation; 14-inch reel capacity; rack mountable.
Price Range: Not applicable.

ATR-800

Tracks/Speeds: Full-, two-, and four-track; 7½, 15 and 30 ips.
Formats: Quarter-inch and half-inch.
Frequency Response: Operating level 370 nWb/m using Ampex 456 tape, ±2 dB; at 30 ips: 50 Hz to 24 kHz; at 15 ips: 30 Hz to 20 kHz.
Distortion: Record/repro: Even-order distortion 1 kHz at 1,040 nWb/m is less than 0.2%; third harmonic distortion at 1 kHz less than 0.3% at 370 nWb/m; SMPTE IM distortion less than 1.5% at 370 nWb/m.
Signal-to-Noise Ratio: Unweighted. Full-track: 30 ips: -73 dB; 15 ips NAB: -70 dB. Two- and four-track 30 ips: -68 dB; 15 ips: -66 dB.
Wow and Flutter: Unweighted: 30 ips NAB RMS 0.03%; at 15 ips 0.04%.
Selected Standard Features: Single point search-to-cue; variable speed shuttle, monitor memory function; four switchable EQs; 16-inch reel capacity; variable speed operation; auto-bias (optional).
Price Range: Not applicable.

COMPUSONICS
323 Acoma St.
Denver, CO 80223
Phone: (303) 698-0060

DSP-2002

Tracks/Speeds: Two-track; digital recording/editing system.
Formats: Hard-disk based; 48 kHz sam-

The RPG™ Acoustical Diffusor

TRC Studios

Acorn Sound Recorders

Otis Conner Productions

There has always existed a need in architectural acoustics for an attractive, modular surface treatment which would efficiently diffuse, i.e. uniformly distribute rather than absorb or attenuate, the acoustical energy in an enclosed space. RPG Diffusor Systems has recognized this need and pioneered the development of the RPG Acoustical Diffusor, which provides a highly diffusing surface that efficiently backscatters sound over a broad range of frequencies, with uniform wide-angle coverage. These desirable diffusive properties cannot be obtained to the same degree with conventional surface treatments, which employ geometrically irregular shapes, polycylindrical columns an/or alternating reflecting and absorbing panels. Many years of theoretical research and development, experimental testing and psychoacoustical evaluation have verified the effectiveness of the RPG Diffusor System. RPG Acoustical Diffusors consist of a series of wells of different depths, based on quadratic residue sequences, and are constructed from hand-rubbed lacquered hardwood and anodized aluminum for a truly handsome appearance.

The RPG Diffusor represents an acoustical design component which was formerly unavailable. Architectural acoustic designers now have at their disposal all of the required acoustical ingredients (absorption, reflection and diffusion), for exemplary and reproducible room design. Virtually any critical listening or performance environment can be enhanced by appropriate use of the RPG Acoustical Diffusor.

- **Auditoriums**
- **Churches**
- **Theaters**
- **Conference Rooms**
- **Concert Halls**
- **Rehearsal Spaces**
- **Orchestral/Choir Shells**
- **Audio/Video Recording Studios**
- **Radio/TV Announce Booths**
- **Mobile Studios**
- **Recording/Broadcast Control Rooms**
- **Disc Mastering Control Rooms**
- **Film Mix/Editing Control Rooms**
- **Exhibit Demo Rooms**
- **Acoustical Ceiling Systems**

Below we have listed a few of the growing number of users who have joined the RPG Diffusor System:

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ASTORIA MUSIC
New York, NY
CHICAGO RECORDING CO.
Chicago, IL
MASTERMIX
Nashville, TN

NEW AGE SIGHT AND SOUND
Atlanta, GA
OTIS CONNER PRODUCTIONS
Dallas, TX
RCA DIGITAL EDITING SUITE
New York, NY
SIGMA SOUND
New York, NY

JIMMY SWAGGART WORSHIP CENTER
Baton Rouge, LA
TELE-IMAGE
Las Colinas, TX
TRC STUDIOS
Indianapolis, IN
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Chicago, IL

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Largo, Maryland 20772
301-249-5647

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For additional information circle #125

pling frequency.
Frequency Response: 0 Hz to 20 kHz.
Distortion: Less than 0.01% third harmonic at 1 kHz 10 V peak to peak.
Signal-to-Noise Ratio: Better than -90 dB.
Wow and Flutter: "Ummeasurable."
Selected Standard Features: Computer audio console; recording and signal processing, and SMPTE timecode interface, editing facilities; menu-driven software environment; stereo/mono audio under software control; random access and playback in programmable sequence (no tape rewind or cueing); optional data reduction for increased storage.
Price Range: $34,000.

dbx, INCORPORATED
71 Chappel Street
Newton MA 02195
Phone: (617) 964-3210

Model 700
Tracks: Two-track, stereo.
Formats: NTSC standard video format on half-inch VHS or ¾-inch U-Matic; Companded Predictive Delta Modulation.
Frequency Response: (Sinewave or pink noise, 100 millivolts input, reference record position) 20 Hz to 20 kHz, ±0.5 dB.
Distortion: Total harmonic distortion 1 volt input, 1 kHz: less than 0.05%.
Signal-to-Noise Ratio: Dyamnic range (maximum RMS signal at 1 kHz to A-weighted noise, 20 Hz to 20 kHz: 110 dB, typical.
Wow and Flutter: Less than 0.01% unweighted; less that 0.006% weighted RMS.
Selected Standard Features: Microphone gain selectable from 20 to 60 dB; input gain control; pre- and post-clip LEDs indicate overload before or after gain adjustment on each channel; main output signal switch selectable between digital decoder and VCR analog audio output (analog scratch track synced to the digitized signal); switch-selectable meter for each channel; headphone jack with volume control; plus video lock, LED standby, LED video unlock, and error correct LED.
Price Range: $4,600.

FOSTEX CORPORATION
15431 Blackburn Avenue
Norwalk, CA 90650
Phone: (213) 921-1112

Model 250
Tracks/Speeds: Four-track; 1⅞ and 3¾ ips; built-in mixer.
Formats: Compact cassette.
Frequency Response: 20 Hz to 18 kHz (40 Hz to 14 kHz, +2 dB/-3 dB at 0VU).
Distortion: Mixer section: better than 0.05% at 1 kHz nominal level; Recorder section: 1.5% at 315 Hz, 0VU overall.
Signal-to-Noise Ratio: Mixer overall -75 dB weighted; recorder -71 dB weighted with built-in Dolby C.
Wow and Flutter: 0.07 weighted peak (IEC/ANSI).
Selected Standard Features: Four input, four-track portable studio.
Price Range: $1,300.

Model A-2
Tracks/Speeds: Two-track; 7½ and 15 ips.
Formats: Quarter-inch.
Frequency Response: 40 Hz to 22 kHz, ±3 dB at 15 ips; 40 Hz to 20 kHz, ±3 dB at 7½ ips NAB equalization.
Distortion: THD less than 1% at 1 kHz 0VU.
Signal-to-Noise Ratio: -63 dB at both speeds, referenced to 1 kHz, 3% THD level.
Wow and Flutter: ±0.06% peak (IEC/ANSI) weighted at 15 ips, 0.10% peak (IEC/ANSI) at 7½ ips.
Selected Standard Features: Port for auto switching of noise reduction; fully remotable.
Price Range: $850.

Model A-4
Tracks/Speeds: Four-track; 7½ and 15 ips.
Formats: Quarter-inch.
Frequency Response: 40 Hz to 22 kHz, ±3 dB at 15 ips; 40 Hz to 20 kHz, ±3 dB at 7½ ips NAB equalization.
Distortion: THD less than 1% at 1 kHz 0VU.

Signal-to-Noise Ratio: -63 dB at 15 ips, and -60 dB at 7½ ips referenced to 1 kHz, 3% THD level.
Wow and Flutter: ±0.06% peak (IEC/ANSI) weighted at 15 ips, 0.10%; peak (IEC/ANSI) at 7½ ips.
Selected Standard Features: Noise reduction can be auto switched using rear port; full IC logic transport.
Price Range: $1,450.

Model A-8/A-8LR
Tracks/Speeds: Eight-track; 15 ips.
Formats: Quarter-inch.

Frequency Response: 45 Hz to 18 kHz, ±3 dB.
Distortion: THD less than 1% at 1 kHz 0VU.
Signal-to-Noise Ratio: -7 dB weighted, referenced to 1 kHz, 3% THD level.
Wow and Flutter: ±0.06% peak (IEC/ANSI) weighted.
Selected Standard Features: IC logic servo-control transport fully remotable; three-motor transport with DC servo capstan; integral Dolby C noise reduction; auto sync switching.
Price Range: A-8: (records four tracks at a time) $2,000; A-8LR: (records eight tracks simultaneously) $2,500.

Model B-16
Tracks/Speeds: 16-track; 15 ips.
Formats: Half-inch.

Frequency Response: 40 Hz to 18 kHz, ±3 dB.
Distortion: THD 1% at 1 kHz.
Signal-to-Noise Ratio: -80 dB weighted, -60 dB unweighted with built-in Dolby C.
Wow and Flutter: ±0.05% peak (IEC/ANSI) weighted.
Selected Standard Features: SMPTE ready; integrated Dolby C; real-time counter; IC logic transport; available with half-inch eight-track playback and 30 ips speed.
Price Range: B-16: $5,900; B-16D: $6,800.

Model B-16DM
Tracks/Speeds: 16-track; 15 ips.
Formats: Half-inch.
Frequency Response: 40 Hz to 18 kHz, ±3 dB.
Distortion: THD 1% at 1 kHz.
Signal-to-Noise Ratio: -80 dB weighted, -60 dB unweighted with built in Dolby C.
Wow and Flutter: ±0.05% peak (IEC/ANSI) weighted.
Selected Standard Features: Direct-drive, PLL capstan motor; integrated Dolby C; synchronizer ready; real-time tape counter; three-head machine; 16 independent channels of monitoring; remote including headphone amplifier.
Price Range: $9,600

JVC COMPANY OF AMERICA
41 Slater Drive
Elmwood Park, NJ 07407
Phone: (210) 794-3900

VP-900
Tracks: Two-track; PCM digital processor
Formats: Three-quarter or half-inch videotape.
Frequency Response: 10 Hz to 20 kHz, ±0.5 dB
Distortion: Less than 0.02% at 1 kHz, 19 dBm output.
Signal-to-Noise Ratio: Better than -90 dB.
Wow and Flutter: "Below measurable limits."

Selected Standard Features: Optional pre-emphasis.
Price Range: $20,400.

LYREC
U.S. Distribution:
Mammoth Marketing
P.O. Box 6493
Thousand Oaks, CA 91359
Phone: (805) 496-2969

TR-55
Tracks/Speeds: Two- and four-track; 15 and 30 ips.
Formats: Quarter- and half-inch.
Frequency Response: 30 Hz to 19 kHz, ±2 dB at 15 ips (record/play).
Distortion: 0.8% (tape).
Signal-to-Noise Ratio: –68 cB A weighted at 15 ips.
Wow and Flutter: 0.04% maximum RMS weighted.
Price Range: TR-55 2/2 half-inch stereo two-track $11,350; TR-55 2/2.5 quarter-inch stereo two-track $7,950; TR-55 4/5 half-inch four-track TBA.

TR 532
Tapes/Speeds: Eight-, 16-, and 24-tracks; 15 and 30 ips.
Formats: One- and two-inch.
Frequency Response: 30 Hz to 20 kHz, +1 dB/–3 dB at 15 ips; 50 Hz to 20 kHz, +1 dB/–3 dB at 30 ips.
Distortion: 0.8% (tape).
Signal-to-Noise Ratio: –63 dB, A weighted at 30 ips, 24-track.
Wow and Flutter: 0.4% weighted.
Selected Standard Features: ATC 35 memory locator.
Price Range: TR-24: $39,000; TR-16: $29,200; TR-8: $25,950

MITSUBISHI ELECTRIC CORPORATION
U.S. Distributor: Digital Entertainment Corporation
87 Sand Pit Road
Danbury, CT 06810
Phone: (203) 743-0000

Mitsubishi X-800
Tracks/Speeds: 32 tracks; 30 ips.
Formats: One-inch.
Frequency Response: 20 Hz to 20 kHz, ±1 dB; 50 Hz to 20 kHz, +1 dB/–0.5 dB.

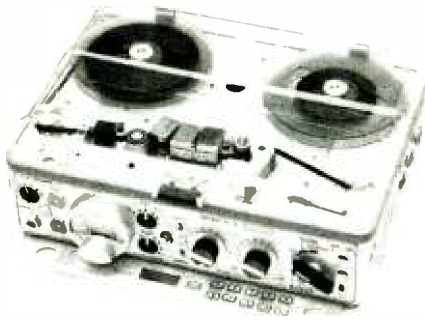
Distortion: Less than 0.05%.
Signal-to-Noise Ratio: Better than –90 dB.
Selected Stadard Features: Full autolocate unit including : operation by int/ext SMPTE code or TACH; memory for up to 99 takes times/events; memory for four-channel groups; memory for four channel setups; variable operation of ±10%; automatic punch-in and -out; 99 take time memories; search-to-zero; search stop; search play; "auto-mark" memory; 9.6 kHz SMPTE input; digital ping-pong panel; and razor editing.
Price Range: List $170,000.

Mitsubishi X-80
Tracks/Speeds: Stereo 15 ips, digital mastering machine.
Formats: Quarter-inch.
Frequency Response: 20 Hz to 20 kHz, ± 1 dB; 50 Hz to 20 kHz, +1 dB/–0.5 dB.
Distortion: Less than 0.05%.
Signal-to-Noise Ratio: Better than –90 dB.
Wow and Flutter: "Immeasurably low."
Selected Standard Features: Cut and splice editing; error-correction; VU and peak metering; synchronizable with DEC VCO (option); analog cue track; SMPTE channel; monitor outputs; XLR input/output; remote autolocater unit (option); portable console configuration (with studio-cart option); varispeed.
Price Range: List $27,000.

NAGRA MAGNETIC RECORDERS, INC.
1147 North Vine Street
Hollywood, CA 90036
Phone: (213) 469-6391

IV-STC
Tracks/Speeds: Two audio and one SMPTE

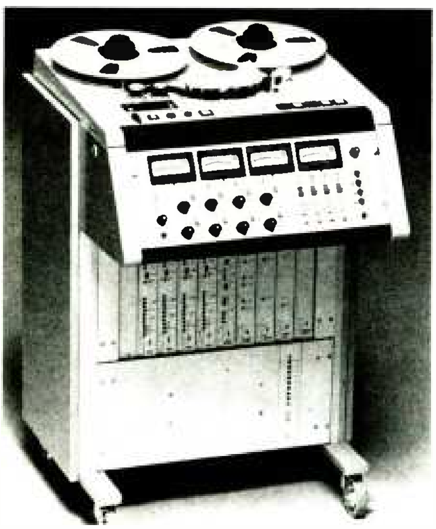
timecode.
Formats: Quarter-inch.
Frequency Response: 30 Hz to 20 kHz, ± 1 dB.
Distortion: At MPL below 1% third harmonic



Signal-to-Noise Ratio: –74 dB at 15 ips NAGRAMASTER; –71 dB at 7½ ips NAB.
Wow and Flutter: ± 0.05% at 15 ips.
Selected Standard Features: Center-track SMPTE/EBU timecode.
Price Range: $7,932.

OTARI CORPORATION
2 Davis Drive
Belmont, CA 94002
Phone: (415) 592-8311

MTR-10, -12
Tracks/Speeds: Two- and four-track; 3¾ to 30 ips.
Formats: Quarter- and half-inch.
Frequency Response: 18 Hz to 27 kHz, +5 dB/-2 dB at 15 ips.



Distortion: Less than 0.15% 1 kHz referenced at 250 nWb/m.
Signal-to-Noise Ratio: –73 dB, 3% third harmonic noise floor, 18 Hz to 30 kHz unweighted.
Wow and Flutter: Less than 0.04% DIN 45507.
Selected Standard Features: Microprocessor controlled transport; selectable calibration; full-function auto locate and remote available; optional MTR-10/12 quarter-inch, center-track head module for SMPTE/EBU tracking; MTR-12 handles 12-inch reels, and is optimized for 15/30 ips speeds.
Price Range: MTR-10-2: $6,450: MTR-10-4: $8,450; MTR-12-2: $6,800; MTR-12-4: $8,800.

MTR-20
Tracks/Speeds: Two- and four-track; 3¾ to 30 ips.
Formats: Quarter- and half-inch.
Frequency Response: 35 Hz to 28 kHz, ± 2 dB at 30 ips.
Distortion: 0.2% 1 kHz referenced at 250 nWb/m.
Signal-to-Noise Ratio: –75 dB 3% third harmonic noise floor, 18 Hz to 30 kHz unweighted.
Wow and Flutter: 0.03% 30 ips NAB unweighted.

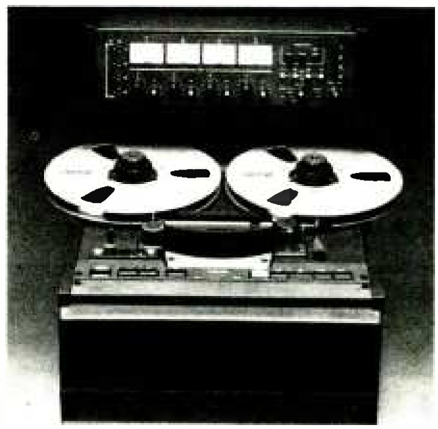
Selected Standard Features: Auto alignment; four programmable speed/EQ combinations; RS-232/422 port (option); extensive programmable functions; optional center-track SMPTE timecode.
Price Range: $11,000 to $13,000.

MTR-90
Tracks/Speeds: Eight-, 16-, and 24-track; 15 and 30 ips.
Formats: One- and two-inch.
Frequency Response: 20 Hz to 20 kHz, +1.5 dB/-2 dB at 15 ips.
Distortion: Less than 0.015% 1 kHz referenced at 250 nWb/m.

Signal-to-Noise Ratio: –74 dB at 30 ips, 3% third harmonic noise floor, 18 Hz to 30 kHz unweighted.
Wow and Flutter: Less than 0.04% DIN 45507.
Selected Standard Features: Pinch-rollerless PLL capstan/microprocessor controlled servo DC transport; RS-232 interface; transformerless input/output.
Price Range: 24-channel: $38,950; 16 channel: $27,000; Eight-channel: $21,950; 16/24 (prewired for 24): $31,650.

MX5050B-II
Tracks/Speeds: Two-track; 7½, 15 and 30 ips.
Formats: Quarter-inch.
Frequency Response: 30 Hz to 20 kHz, ± 2 dB at 15 ips.
Distortion: Less than 0.05% 1 kHz referenced at 250 nWb/m.
Signal-to-Noise Ratio: –72 dB at 30 ips, 3% third harmonic noise floor, 30 Hz to 18 kHz unweighted.
Wow and Flutter: Less than 0.04% NAB weighted.
Selected Standard Features: Transformerless balanced; +4 or –10 dBu operating level; microprocessor controlled real-time counter; three speeds with switchable pairs.
Price Range: $2,295 (also available in full-track and quarter-track formats).

MX5050BQ-II
Tracks/Speeds: Four-track; 7½ and 15 ips.
Formats: Quarter-inch.
Frequency Response: 30 Hz to 20 kHz, ± 2 dB at 15 ips.
Distortion: Less than 0.07% 1 kHz referenced at 1 kHz 250 nWb/m.
Signal-to-Noise Ratio: –66 dB 3% third harmonic noise floor, 30 Hz to 18 kHz unweighted.
Wow and Flutter: Less than 0.04% NAB weighted.



Selected Standard Features: Mike/line mixing; selectable headphone monitoring; microprocessor-controlled counter-LED display.
Price Range: $2,295.

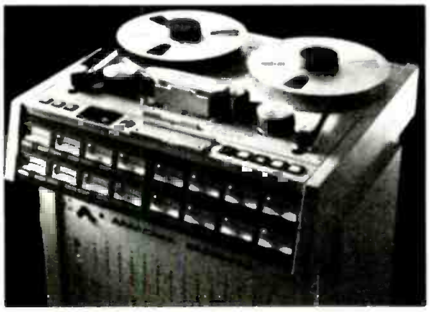
MX5050 MARKIII-8
Tracks/Speeds: Eight-track; 7½ and 15 ips.
Formats: Half-inch.

Frequency Response: 40 Hz to 25 kHz, ±2 dB at 15 ips.
Distortion: Less than 0.07% 1 kHz referenced at 250 nWb/m.
Signal-to-Noise Ratio: –68 dB at 30 ips, 3% third harmonic noise floor, 18 Hz to 30 kHz

unweighted.
Wow and Flutter: Less than 0.06% NAB weighted.
Selected Standard Features: Exterior oscillator input; dump edit; cue mode; CB-116 Autolocator and CB-110 remote controller available; also available in four-track, half-inch and two-track, quarter-inch versions.
Price Range: \$5,295; CB116: \$695; CB110: \$650 (four-track: \$3,895; two-track: \$2,795).

MX-70

Tracks/Speeds: Eight- and 16-tracks; 7½, 15, and 30 ips.
Formats: One-inch (half-inch optional.)
Frequency Response: 55 Hz to 22 kHz, ± 2 dB at 15 ips.



Distortion: Less than 0.2% referenced at 1 kHz at 250 nWb/m.
Signal-to-Noise Ratio: –68 dB one-inch 16-track; –72 dB one-inch eight-track.
Wow and Flutter: Less than 0.04% DIN 45507.
Selected Standard Features: Four models: one-inch 16-track; one-inch eight-track; one-inch eight-track prewired for 16 tracks; any of three models convertible to half-inch eight-track.
Price Range: One-inch 16-track \$14,950; eight-track (prewired for 16-tracks) \$13,500; one-inch eight-track \$12,000.

L.J. SCULLY
138 Hurd Street
Bridgeport, CT 06644
Phone: (203) 368-2332

LJ-12

Tracks/Speeds: Two- or four-track; 3¾, 7½, 15, and 30 ips.
Formats: Quarter- or half-inch.
Frequency Response: 40 Hz to 12 kHz, ±1.5 dB at 3¾ ips; 35 Hz to 18 kHz, ±1.5 dB at 7½ ips; 30 Hz to 22 kHz, ±1.5 dB at 15 ips; 40 Hz to 28 kHz, ±1.5 dB at 30 ips.
Distortion: Less than 1% THD at 250 nWb/m at 15 ips.
Signal-to-Noise Ratio: –68 dB at 15 ips, unweighted 30 Hz to 18 kHz reference 510 nWb/m.
Wow and Flutter: Less than 0.04% at 15 ips (DIN 45507 weighted).
Selected Standard Features: Four speed, microprocessor controlled.
Price Range: \$7,000+.

SONY PROFESSIONAL PRODUCTS COMPANY
Sony Drive
Park Ridge, NJ 07656
Phone: (201) 930-1000

MCI/JH-24

Tracks/Speeds: Eight-, 16-, and 24-track; 15 and 30 ips.
Formats: One- and two-inch.
Frequency Response: 30 ips AES 36 Hz to 26 kHz, +1.5 dB/–3 dB; 15 ips NAB 30 Hz to 26 kHz, +1.5/–2 dB.

Distortion: Reference fluxivity 510 nWb/m 1kHz: fundamental third harmonic distortion: 30 ips AES less than 0.35%; 15 ips NAB less than 0.5%
Signal-to-Noise Ratio: Reference 510 nWb/m unweighted 20 Hz to 20 kHz: 30 ips AES –67 dB; 15 ips NAB –63 dB.
Wow and Flutter: 15 ips less than 0.04% DIN 45507 weighted; 30 ips less than 0.03% DIN 45507 weighted.
Selected Standard Features: Full function remote control.
Price Range: Eight-track: \$17,000; 16-track: \$22,685; 24-track: \$33,729.

MCI/JH-110C-8

Tracks/Speeds: Eight-track; 7½, 15, and 30 ips.
Formats: One-inch.
Frequency Response: 30 ips AES 40 Hz to 28 kHz, +0.75 dB/–2 dB; 15 ips NAB 30 Hz to 24 kHz, +0.75/–2 dB; record/reproduce 7½ ips NAB 30 Hz to 20 kHz, +0.75 dB/–2 dB.
Distortion: Reference fluxivity 510 nWb/m 1kHz fundamental third harmonic distortion: 30 ips AES less than 0.35%; 15 ips NAB less than 0.52%; 7½ ips NAB less than 1.6%.
Signal-to-Noise Ratio: Reference 510 nWb/m unweighted 20 Hz to 20 kHz: 30 ips AES –66 dB; 15 ips NAB –64 dB; 7½ ips NAB –63.
Wow and Flutter: 7½ less than 0.03% DIN 45507 weighted; 15 ips less than 0.20% DIN 45507 weighted; 30 ips less than 0.15% DIN 45507 weighted.

Selected Standard Features: Full function remote control, 10 memory autolocator included.
Price Range: \$11,470.

The Directory

MCI/JH-110C-2

Track/Speeds: Two-track: 3¾, 7½, 15 and 30 ips.
Formats: Quarter-inch.
Frequency Response: 30 ips AES 40 Hz to 28 kHz, +0.75 dB/–2 dB; 15 ips NAB 30 Hz to 24 kHz, +0.75/–2 dB; record/reproduce 7½ ips NAB 30 Hz to 20 kHz, +0.75 dB/1.5 dB.
Distortion: Reference fluxivity 510 nWb/m 1kHz fundamental third harmonic distortion: 30 ips AES less than 0.35%; 15 ips NAB less than 0.52%; 7½ ips NAB less than 1.6%.

Signal-to-Noise Ratio: Reference 510 nWb/m unweighted 20 Hz to 20 kHz: 30 ips AES –66 dB; 15 ips NAB –64 dB; 7½ ips NAB –63.
Wow and Flutter: 7½ less than 0.03% DIN 45507 weighted; 15 ips less than 0.20% DIN 45507 weighted; 30 ips less than 0.15% DIN 45507 weighted.
Selected Standard Features: All DC transport; built-in autolocator.
Price Range: $5,400.

PCM-3324

Tracks/Speeds: 24-track, PCM DASH-format.
Formats: Half-inch; 15 ips.
Frequency Response: 20 Hz to 20 kHz, +0.5 dB/–1 dB.
Distortion: Less than 0.05%.
Signal-to-Noise Ratio: Better than –90 dB.
Wow and Flutter: "Undetectable."
Selected Standard Features: Complies with DASH format, switchable 48 kHz and 44.1 kHz sampling frequencies; autolocate at any cue point.
PRICE RANGE: $133,600.

PCM-3102

Tracks/Speeds: Two-track, PCM DASH-format.
Formats: Quarter-inch.
Frequency Response: 20 Hz to 20 kHz.
Distortion: Less than 0.05%.
Signal-to-Noise Ratio: Better than –90 dB.
Wow and Flutter: Equal to crystal accuracy of 0.001%.
Selected Standard Features: Switchable 44.1 and 48 kHz sampling frequencies; built-in timecode generator.
Price Range: "Call for price."

PCM-1610

Tracks/Speeds: Two-track, PCM digital processor.
Formats: U-Matic digital audio recording tape.
Frequency Response: 20 Hz to 20 kHz.

Distortion: Less than 0.05%.
Signal-to-Noise Ratio: Better than –90 dB.
Wow and Flutter: "Undetectable."
Selected Standard Features: Instant VTR interfacing; synchronized operation; digital editing; built-in SMPTE timecode generator.
Price Range: $25,000.

SOUNDCRAFT ELECTRONICS
1517 20th Street
Santa Monica, CA 90404
Phone: (213) 453-4591

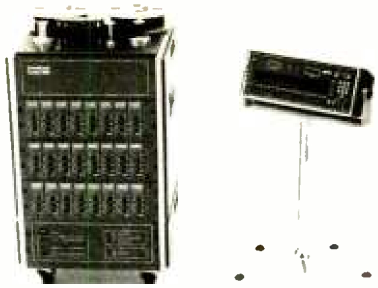
Series 20

Tracks/Speeds: Two-track; 15 and 30 ips.
Formats: Quarter- and half-inch.
Frequency Response: 50 Hz to 24 kHz, ±1 dB at 30 ips; 30 Hz to 20 kHz, ±1 dB at 15 ips.
Distortion: 0.08% at 0VU.

Signal-to-Noise Ratio: –66.5 dB at 15 ips; –69 dB at 30 ips referenced at 510 nWb/m.
Wow and Flutter: 0.04% DIN.
Selected Standard Features: Microprocessor control; programmable audio.
Price Range: $6,500 to $7,600.

SCM-760 MkIII

Tracks/Speeds: 16- and 24-track; 15 and 30 ips.
Formats: Two-inch.
Frequency Response: ±2 dB, 40 Hz to 20 kHz at 15 ips; ±2 dB, 50 Hz to 24 kHz at 30 ips.
Distortion: 0.08% at 0VU.
Signal-to-Noise Ratio: –63 dB replay; –64 dB sync with transformer.
Wow and Flutter: 0.04% DIN.
Selected Standard Features: Noise reduc-



tion switching; nine-memory autolocator; frequency or voltage external reference capstan PLL motor.
Price Range: $17,950 to $24,950.

SOUNDSTREAM
2505 Parley's Way
Salt Lake City, UT 84109
Phone: (801) 486-4701

Soundstream Digital Recorder
Tracks/Speeds: Eight-track PCM digital transport; 30 ips
Formats: One-inch
Frequency Response: DC to 21.5 kHz.
Distortion: Less than 0.25% at all frequencies less than at 20 kHz and below, and levels up to +20 dBm.
Signal-to-Noise Ratio: Better than -90 dB.
Wow and Flutter: "Unmeasurable."
Selected Standard Features: None, all machines are the same.
Price Range: "Negotiable."

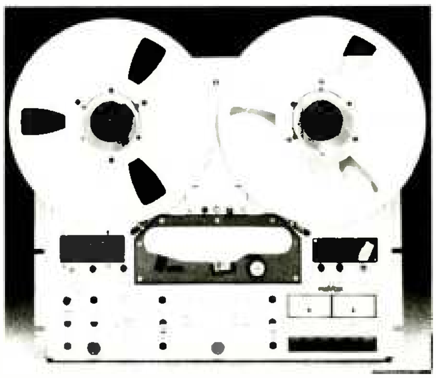
STEPHENS ELECTRONICS
2025 North Lincoln
Burbank, CA 91504
Phone: (818) 842-5116

Model 821B 104A QII
Tracks/Speeds: Four-, eight-, 16-, 24-, 32-, and 40-track; 15, 30, 60 ips.
Formats: Half-inch
Frequency Response: 15 ips: ±1 dB from 25 Hz to 18 kHz; 30 ips: ±1 dB from 30 Hz to 25 kHz.
Distortion: At 1 kHz less than 0.8% reference to 200 nWb/m.
Signal-to-Noise Ratio: 1040 nWb/m (3% THD) unweighted, wideband. Tape Stopped: -80 dB at four-, eight-, and 16-track; -77 dB at 24-track; -76 dB at 32-track: -75 dB at 40-track. 15 ips: -74 dB at four-, eight-, and 16-track; -71 dB at 24-track; -70 dB at 32-track; -69 dB at 40-track. 30 ips: -75 dB at four-, eight-, and 16-track; -72 dB at 24-track; -71 dB at 32-track; -70 dB at 40-track.
Wow and Flutter: 15 ips 0.03% WRMS filter; 30 ips 0.02% WRMS filter.
Selected Standard Features: Autolocator, interfaces to BTX Shadow, CMX 300, 340, etc. Capable of handling 14-inch reels.
Price Range: Four-track $16,000, to 40-track $64,000.

STUDER REVOX
1425 Elm Hill Pike
Nashville, TN 37210
Phone: (615) 254-5651

B67
Tracks/Speeds: One- or two-track. Type A: 7½, 15, 30 ips; Type B: 3¾, 7½, and 15 ips.
Formats: Quarter-inch.
Frequency Response: 15 ips: 30 Hz to 18 kHz, ±2 dB.
Distortion: Maximum 1%.
Signal-to-Noise Ratio: 15 ips two-track -62 dB above 185 nWb/m.
Wow and Flutter: 0.06%, weighted at 15 ips.
Selected Standard Features: Adjustable tape tension; locking tape tension arms; selectable NAB or CCIR EQ; optional Pilot-tone versions.
Price Range: Two-track portable: $4,800.

PR99 MkII
Tracks/Speeds: One- or two-track; 3¾, 7½, and 15 ips.
Formats: Quarter-inch.
Frequency Response: 30 Hz to 22 kHz at 15 ips.



Distortion: Maximum 0.3% at 255 nWb/m at 7½ ips.
Signal-to-Noise Ratio: -66 dB ASA-A weighted, 7½ ips at 510 nWb/m.
Wow and Flutter: 0.08%, at 7½ ips (DIN 45507).
Selected Standard Features: Real-time counter; autolocate; zero locate; vari-speed; balanced inputs and outputs.
Price Range: $2,250.

A800 MkIII
Tracks/Speeds: Eight-, 16-, and 24-track; 15 and 30 ips.
Formats: One- or two-inch.

Frequency Response: 30 Hz to 20 kHz, ±2 dB.
Distortion: 1% maximum.

The Directory

Signal-to-Noise Ratio: –74 dB (30 ips at +6 dB, unweighted).
Wow and Flutter: Maximum 0.04% at 30 ips.
Selected Standard Features: Microprocessor transport control; 14-inch reel capacity; timecode channel; master bias control.
Price Range: $38,590 to $69,500.

A810

Tracks/Speeds: One- or two-track; 7½ and 15 ips.
Formats: Quarter-inch.
Frequency Response: 20 Hz to 20 kHz at 15 ips.
Distortion: 0.06% maximum.

Signal-to-Noise Ratio: –71 dB ASA-A weighted at 15 ips.
Wow and Flutter: Maximum 0.05% at 15 ips.
Selected Standard Features: Microprocessor control of transport and electronics; center-track SMPTE timecode option; programmable features; four tape speeds.
Price Range: $5,990 to $9,700

A80VU MkIV

Tracks/Speeds: Two-, four-, eight-, 16-, 24-track; 3¾, 7½, 15, and 30 ips.

Formats: Quarter-, half-, one-, and two-inch.
Frequency Response: 50 Hz to 20 kHz, ±2 dB.
Distortion: 1% maximum (via tape at 1 kHz, NAB equalization, at 510 nWb/m).
Signal-to-Noise Ratio: –68 dB (24-track at 30 ips); –74 dB (half-inch two-track at 30 ips).
Wow and Flutter: Maximum 0.04% maximum at 30 ips, IEC 368/DIN 45507.
Selected Standard Features: Tranformerless in/out; Dolby HX Pro compatible; 20-address memory auto locator.
Price Range: $10,500 (two-track) to $35,000 (24-track with auto locator and remote).

TANDBERG OF AMERICA
P.O. Box 58 Labriola Court
Armonk, NY 10504
Phone: (914) 273-9150

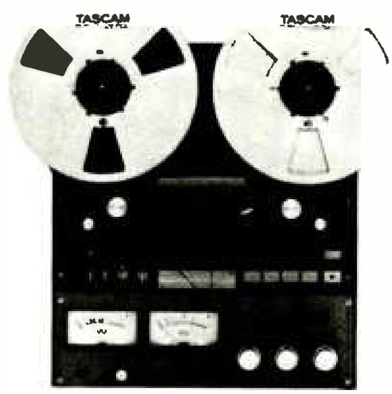
TCD 910

Tracks/Speeds: Two-track; 7½ and 15 ips.
Formats: Quarter-inch.
Frequency Response: 20 Hz to 20 kHz.
Distortion: Less than 1%.
Signal-to-Noise Ratio: –73 dB.
Wow and Flutter: 0.1%.
Selected Standard Features: 8-bit, 32 EPROM microprocessor; full auto locator system; RS-232 interface; vari-spool; real time/mode counter.
Price Range: $1,795.

TASCAM
7733 Telegraph Road
Montebello, CA 90604
Phone: (213) 726-0303

42-NB

Tracks/Speeds: Two-track; 7½ and 15 ips.
Formats: Quarter-inch.
Frequency Response: 30 Hz to 25 kHz, ±2 dB at 0VU.



Distortion: 0.08% at 0VU 1 kHz, 250 nWb/m reference.
Signal-to-Noise Ratio: –65 dB unweighted (0 to 100 kHz).
Wow and Flutter: 0.05% NAB weighted.
Selected Standard Features: Response in sync equal to repro, +4 dBm XLR in/out; single connector for SMPTE timecode edit systems.
Price Range: $2,295.

44-OB

Tracks/Speeds: Four-track; 15 ips.
Formats: Quarter-inch.
Frequency Response: 40 Hz to 25 kHz, ± 2 dB at 0VU.
Distortion: 0.08% at 0VU 1 kHz, 250 nWb/m reference.
Signal-to-Noise Ratio: –64 dB unweighted (0 to 100 kHz).
Wow and Flutter: 0.05% NAB weighted.
Selected Standard Features: +4 dBm XLR

in/out; equivalent response in sync or repro head; single connector for SMPTE timecode edit systems.
Price Range: $2,995.

48-OB

Tracks/Speeds: Eight-track, 15 ips.
Formats: Half-inch.
Frequency Response: 40 Hz to 25 kHz, ±2 dB at 0VU.
Distortion: 0.08% at 0VU 1 kHz, 250 nWb/m reference level.
Signal-to-Noise Ratio: -64 dB unweighted (0 to 100 kHz).
Wow and Flutter: 0.05% NAB weighted.
Selected Standard Features: +4 dBm XLR in/out; equivalent response in sync or repro mode; single connector for SMPTE timecode edit systems.
Price Range: $4,495.

52-NB

Tracks/Speeds: Two-track, 7½ and 15 ips.
Formats: Quarter-inch.
Frequency Response: 30 Hz to 26 kHz, ± 2 dB at 0VU.
Distortion: 0.08% at 0VU 1 kHz, 250 nWb/m reference level.

Signal-to-Noise Ratio: -70 dB unweighted (0 to 100 kHz).
Wow and Flutter: 0.04% NAB weighted.
Selected Standard Features: +4 dBm XLR in/out; equivalent response in sync or repro head; single connector for SMPTE timecode edit systems.
Price Range: $3,495.

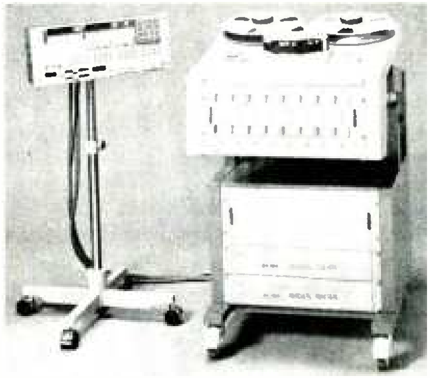
58-OB

Tracks/Speeds: Eight-track, 15 ips.
Formats: Half-inch.
Frequency Response: 30 Hz to 26 kHz, ±2 dB at 0VU.

Distortion: 0.08% at 0VU 1 kHz, 250 nWb/m reference level.
Signal-to-Noise Ratio: -70 dB unweighted (0 to 100 kHz).
Wow and Flutter: 0.04% NAB weighted.
Selected Standard Features: +4 dBm XLR in/out; equivalent response in sync or repro head; single connector for SMPTE timecode edit systems.
Price Range: $5,995.

MS-16

Tracks/Speeds: 16-track; 15 ips.
Formats: Half-inch.
Frequency Response: 30 Hz to 25 kHz, ±2 dB at 0VU.



Signal-to-Noise Ratio: -70 dB unweighted (0 to 100 kHz).
Wow and Flutter: 0.04% NAB weighted.
Selected Standard Features: +4 dBm XLR in/out; equivalent response in sync or repro head; single connector for SMPTE timecode edit systems.
Price Range: $8,995.

Portastudio 244

Tracks/Speeds: Four-track; 3¾ ips.
Formats: Standard compact cassette.
Frequency Response: 20 Hz to 20 kHz, ±1 dB
Distortion: THD 0.05% at 1 kHz nominal level.
Signal-to-Noise Ratio: One mike-in to line-out: -68 dB (IHF, A weighted); -76 dB (unweighted).
Selected Standard Features: Four-input, four-track portable studio.
Price Range: N/A

AEG TELEFUNKEN
Bucklestrasse A-5
D-7750 Konstanz, West Germany
Phone: (011) 4975312370

M15A

Tracks/Speeds: Two-, four-, eight-, 16-, 24-, 32-tracks; 7½, 15, and 30 ips.
Formats: Quarter-, half-, one-, and

two-inch.
Frequency Response: 30 Hz to 18 kHz, ±1.5 dB at 15 ips.
Distortion: Maximum 0.4%, referenced to 400 nWb/m.
Signal-to-Noise Ratio: -66 dB RMS, A weighted, referenced to 400 nWb/m.
Wow and Flutter: Maximum 0.04% at 15 ips.
Selected Standard Features: N/A
Price Range: N/A

M21
Tracks/Speeds: Two-track; 7½, 15, and 30 ips.
Formats: Quarter-inch.
Frequency Response: 20 Hz to 20 kHz, ±1.5 dB at 15 ips.
Distortion: Maximum 0.4%, referenced to 400 nWb/m.
Signal-to-Noise Ratio: -73 dB (stereo 0.75mm) RMS, A-weighted, referenced to 1,020 nWb/m.
Wow and Flutter: Maximum ±0.04% at 15 ips.
Selected Standard Features: Ultra long-life heads.
Price Range: $6,500 to $8,000.

UHER OF AMERICA
7076 Vineland Avenue
North Hollywood, CA 91605
Phone: (818) 764-1120

4000 Report Monitor AV
Tracks/Speeds: 7½, 3¾, 1⅞, 15/16 ips.
Formats: Mono (half-track) quarter-inch.
Frequency Response: 20 Hz to 25 kHz at 7½ ips; 20 Hz to 16 kHz at 3¾ ips; 25 Hz to 13 kHz at 1⅞ ips; 25 Hz to 6 kHz at 15/16 ips.
Distortion: Unspecified.
Signal-to-Noise Ratio: Better than -66 dB at 7½ ips; better than -64 dB at 3¾ ips; better than -57 dB at 1⅞ ips.
Wow and Flutter: (DIN 45507) less than 0.15% at 7½ ips; less than 0.20% at 3¾ ips; less than 0.25% at 1⅞ ips
Selected Standard Features: Portable and mains operation; certain models with built-in monitoring capability; built-in phantom powering; five-inch reel capacity.
Price Range: $1,620.

4200 Report Monitor
Tracks/Speeds: Same as above.
Formats: Stereo (half-track) quarter-inch.
Frequency Response: Same as above.
Distortion: Unspecified.
Signal-to-Noise Ratio: Same as above.
Wow and Flutter: Same as above.
Selected Standard Features: Same as above.
Price Range: $1,772.

4400 Report Monitor
Tracks/Speeds: Same as above.
Formats: Stereo (quarter-track) quarter-inch.
Frequency Response: Same as above.
Distortion: Unspecified.
Signal-to-Noise Ratio: Better than -64 dB at 7½ ips; better than -62 dB at 3¾ ips; better than -56 dB at 1⅞ ips.
Wow and Flutter: Same as above.
Selected Standard Features: Same as above.
Price Range: $1,772.

ADAMS-SMITH
34 Tower Street
Hudson, MA 01749
Phone: (617) 562-3801

System 2600
Timecode Types: LTC/VITC/SMPTE/EBU standards.
Lock Accuracy: 1/1000 TV frame.
Memories: Up to 120 timecode addresses.
Transport/Control Commands: FF; RW; play; stop; pause; record-in; record-out; for one master and up to eight slave transports.
Other Primary Features: Modular-based and may be configured to exactly fit user requirements.
Price Range: $3,000 to $25,000 depending on modules selected.

ALPHA AUTOMATION
2049 West Broad Street
Richmond, VA 23220
Phone: (804) 358-3852

The Boss 8400
Timecode Types: SMPTE/EBU/LTC/VITC; 24, 25, 30, Drop frame
Lock Accuracy: 1/300 of a second, ±50 microseconds
Memories: See below.

Transport Control Commands: Full remote transport commands
Other Primary Features: Auto editor based on BTX Shadow synchronizer; up to four Shadows can be connected to the system, to provide control of four audio transports; 54 scratch pad registers; 81 location registers; sychronizer expandable to 16 devices.
Price Range: $16,900.

APPLIED MICROSYSTEMS LTD.
Town Mill, Bagshot Road
Chobham, Woking,
Surrey, England
Phone: (09905) 6267

I-CON
Timecode Types: SMPTE and proprietary

code.
Lock Accuracy: ±2 milliseconds.
Memories: 9 cue storage points.
Transport/Control Commands: Controls record selects (on Tascam 85-16, 58 and 48; Fostex B16), and normal tranport controls
Other Primary Features: RS-232 interface.
Price Range: Approximately $2,800.

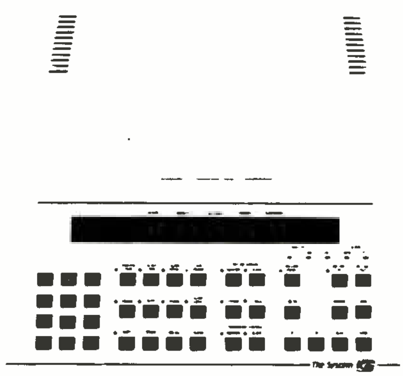
AUDIO KINETICS
4721 Laurel Canyon Blvd. #209
North Hollywood, CA 91607
Phone: (818) 980-5717

Q.Lock 310 C
Timecode Types: SMPTE/EBU 24, 25, 30 Drop frame.
Lock Accuracy: 50 microseconds.
Memories: 10.
Transport/Control Commands: Full remote transport commands.
Other Primary Features: Auto Record, dedicated application S/W, option 64.
Price Range: 310.2: $11,950; 310.3: $14,450.

THE BTX CORPORATION
75 Wiggins Avenue
Bedford, MA 01730
Phone: (617) 275-1420

Shadow II Plus Synchronizer
Timecode Types: SMPTE/EBU/LTC/VITC; 24, 25, 30 Drop frame.
Lock Accuracy: 1/3000 of a second, ±50 microseconds.
Memories: 16 general purpose memories ("Scratchpad").
Transport/Control Commands: All controls available via RS-232/422 interfaces.
Other Primary Features: Performance improvements over Shadow II; variable park tolerance.
Price Range: $5,595.

The System Synchronizer/Controller
Timecode Types: SMPTE/EBU/LTC/VITC 24, 25, 30 Drop frame.
Lock Accuracy: 1/3000 of a second, ±50 microseconds.
Memories: 16 "Scratchpad" registers.



Transport/Control Commands: Play; fast forward; rewind; loop; record in; record out; goto; chase.
Other Primary Features: All functions of Shadow and Shadowpad in one compact unit.
Price Range: $5,595.

Softouch Controller
Timecode Types: SMPTE/EBU/LTC/VITC 24, 25, 30 Drop frame.
Lock Accuracy: 1/3000 of a second, ±50 microseconds.
Memories: 16 general purpose; also 99 loops.
Transport/Control Commands: Play; fast forward; rewind; loop; record in; record out; goto; chase.
Other Primary Features: 16 programmable softkeys; pre-programmed loops and records.
Price Range: $6,500 to $20,000.

Shadowpad Controller
Timecode Types: SMPTE/EBU/LTC/VITC 24, 25, 30 Drop Frame.
Lock Accuracy: 1/3000 of a second, ±50 microseconds.
Memories: 16 "Scratchpad" memories.
Transport/Control Commands: Play, Fast Forward, Rewind, Loop, Record In, Record Out, Goto, Chase.
Other Primary Features: Pre-programmed, subframe-accurate loops and records.
Price Range: $2,500

OTARI CORPORATION
2 Davis Drive
Belmont, CA 94002
Phone: (408) 438-0598

EC-101
Timecode Types: SMPTE/EBU/LTC/VTC.
Lock Accuracy: 50 microseconds over ±50% play speed.
Memories: N/A (see CB-121).
Transport/Control Commands: See reference CB-121.
Other Primary Features: Chase synchronizer plug-in module for MTR-90 transport offset storage in 1/80th frame increments; RS-232/422 port (option).
Price Range: $3,295 (retrofit kit necessary for existing MTR-90s).

CB-121
Timecode Types: SMPTE/EBU and LTC/VTC.
Lock Accuracy: 50 microseconds over ±50% play speed.
Memories: N/A.
Transport/Control Commands: Enable-disable synchronizer; select display mode; hold or run display; enter offsets; capture offsets; and trim offsets up or down.
Other Primary Features: Remote controller for EC-101.
Price Range: $495.

SONY PROFESSIONAL AUDIO PRODUCTS CO.
Sony Drive
Park Ridge, NJ 07656
Phone: (201) 930-1000

AVSP-500 SYNCMASTER
Timecode Types: SMPTE/SMPTE Drop frame/EBU.
Lock Accuracy: ±50 microseconds.
Memories: 200-event EDL.
Transport/Control Commands: Depends on interface chosen, and machines in network.
Other Primary Features: Multi-format code generator, complete display facilities.
Price Range: $12,000 to $15,000.

STUDER REVOX
1425 Elm Hill Pike
Nashville, TN 37210
Phone: (615) 254-5651

TLS 4000
Timecode Types: SMPTE/EBU, video composite, pilot frequencies 20 Hz to 20 kHz.
Lock Accuracy: 40 microseconds.
Memories: 10 address and offsets.
Transport/Control Commands: Lock; wait lock; instant lock; lock plus pilot; slew mode; resolver pilot mode; loop; cue and goto (autolocate).
Other Primary Features: Modular system for easy expansion; RS-232/422 interface for external control; two local control unit options.
Price Range: $3,800 to $5,390.

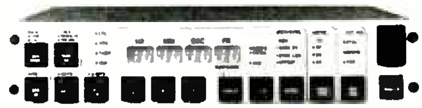
SYNCHRONOUS TECHNOLOGIES
1020 West Wilshire Boulevard
Oklahoma City, OK 73116
Phone: (405) 842-0680

SMPL Lock
Timecode Types: Drop- and non-drop frame: 24 and 25.
Lock Accuracy: 1/80th frame.
Memories: 12.
Transport/Control Commands: Locate, full remote.
Other Primary Features: Autolocates and controls MIDI-equipped units; automatic-punch, eight-point event sequencer; timecode generator; timecode reader;

metronome.
Price Range: $2,000

TIMELINE, INC.
458 Minneford Avenue
City Island, NY 10464
Phone: (212) 431-0330

LYNX Timecode Module
Timecode Types: 30 Drop Frame, 25, 24 (accepts mixed codes).
Lock Accuracy: 50 microseconds
Memories: Contained in external controller.



Transport/Control Commands: Self-contained "chase" mode. Other modes from external controller.
Other Primary Features: Contains timecode generator and wideband reader.
Price Range: $2,990, pro net.

EQUIPMENT ASSESSMENT

YAMAHA RM2408 RECORDING AND PRODUCTION CONSOLE

Reviewed by Peter Butt

The Model RM2408 is the latest recording mixer to be introduced to the American market by Yamaha, and features 24 input/output channels that drive a direct out, eight assignable mix busses with odd/even panning, two echo sends, and stereo monitor panning. Each of the eight mix busses and echo sends features a bus master control, plus ports for insertion of outboard signal processors by way of normalled phono-type jacks. Echo may be returned to the stereo mix and/or to any of the eight mix busses by either assignment switching or by panpot. Each major output bus is metered by a buffered analog VU. The eight Program bus outputs may be directed to either "A" or "B" output ports, allowing a 16-track tape machine to be unambiguously assigned from the RM2408 mix assigns. A 6-by-24 tip-sleeve ¼-inch patch bay provides access to the pre-fade, post EQ insert/return points for each I/O channel.

As might be assumed, the smaller Model RM1608 embodies the same features as the RM2408 reviewed here, with the exception that it has only 16 I/O channels compared with the 24 of the larger model, and a smaller patch bay.

The RM2408 functions in any of three basic modes, each of which may be selected by activating one of three corresponding illuminated switches located on the Talkback module. In *MULTI* mode, the mike or line input of all I/O modules is assignable to the eight Program Mix busses, as for session tracking and overdubs. In *2TRK* mode, the mike or line inputs of all I/Os are pannable to the Stereo program busses. In addition, assignments to the eight Program mix busses are available via the bypassed stereo level control in either *MXD* or *2TRK* modes of operation; the resultant ability to use the Mix assignments as additional effects or subgroup sends should prove valuable to an inventive mixer. In *MXD* mode, the inputs of all I/O channels are connected to the tape-machine outputs, and the I/O outputs pannable to the Stereo program busses.

The Control Room monitor module permits monitoring of the Stereo program bus outputs; a two-track tape playback input to the console; Echo Return 1 or 2; or an adjustable Solo. The Studio Monitor output may access either the Stereo program busses; the two-track tape return; or be muted. (The Studio Monitor muting does not override talkback to the studio.) The Solo pick-off point is the same signal as appears at I/O Direct Out: that is, post-fade, post-EQ, post-insert.

The I/O Monitor mix insertion is pannable to the Stereo Program busses connector line table and used as stereo monitor mix in *MULTI* mode for Control Monitor, and may be selected as either pre- or post-fade origin. Only Echo Send 1 may be selected as pre- or post-fade, while Echo Send 2 is selectable as a post-fader or I/O center Stereo Monitor send.

The console is powered by a separate power-supply assembly that may be rack mounted in a 5.5-inch panel space. Since the nearest EIA rack panel sizes are 5.25 and 7.00 inches, some filling of a panel gap will be necessary for the fastidious. Power connection to the consoles is made via a multiconductor cable having multipin, metal-shell connectors with threaded retaining bushings for mechanical security. The supply and system run cool to the touch, rising only a few degrees above ambient temperature.

Microphone inputs are via the conventonal three-pin XL-type connectors, with pin #2 to be taken as the HI signal terminal. [Pin #2 is live or "hot" in accordance with IEC 268-12, the standard for XLR-3-style connectors — *Editor.*] Internal +48 VDC phantom power may be enabled to individual microphone ports, via the PH switches on each input channel; a single switch located on the rear connector panel removes phantom power from all inputs. Polarity inversion of the microphone signal only is accomplished by the traditional PH switch activation.

The I/O modules includes a three-band peak/notch equalizer allowing a maximum of about ± 15 dB of boost or cut in a fairly symmetrical manner. I/O input may be selected as either mike- or line-level input. While the microphone input features a switchable 20 dB pad variable pre-amp gain that is adjusover a nominal 60 to 20 dB range, the input port is passed through a phono and is not gain trimmable. I/O module patch insertions are at a nominal -10 dBv level, and are accessed through the patch bay. A channel mute switch located near the I/O fader serves to enable or disable the channel signal feed, and causes the illumination of a tally LED when the I/O channel signal is enabled. The Post-Fade, Pre-Pan output of each module is available at another rear-panel phono connector designated as Dir Out.

The PGM Mix outputs 1A through

. . . continued overleaf —

A BIG PART OF OUR MICROPHONE TEAM COMES IN EACH BOX

Most companies concentrate their resources on innovative hardware and elaborate processes in an effort to achieve the "ultimate" product.

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8A are available to those respective I/O channel monitor feeds, while PGM Mix 1B through 8B are available to I/Os 9 through 16 monitor feeds, respectively. PGM monitor feeds for I/Os 17 through 24 are said to be inactive in the User Manual, where in fact they connect to the respective line input. Another error in the documentation appears in the single-line block diagram shown on Page 13 of the manual. Access to the Mix bus summation output is indicated on the drawing, labeled PGM SUB IN 1-8. It is just as well that these do not exist, since the signal polarity at these points relative to PGM OUT A and B would necessarily be inverted. There is nothing illegal or immoral about accessible inverting signal ports, but they can cause confusion and frustration far beyond what their conceptual simplicity might imply.

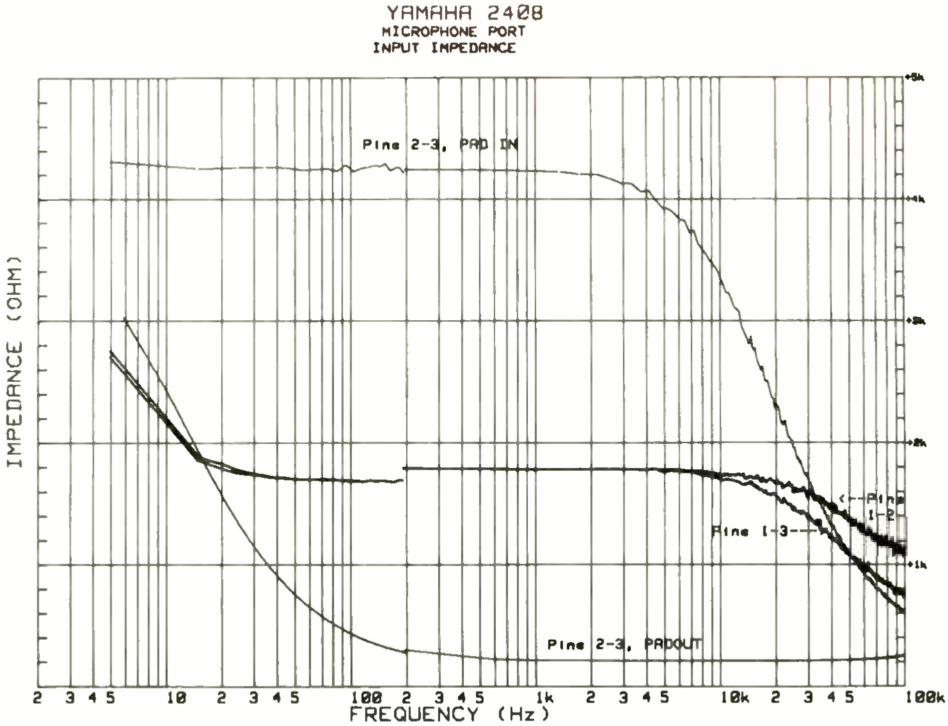


Figure 1: Microphone input port impedances. Driving generator source resistance is 150.0 ohms.

System Assessment

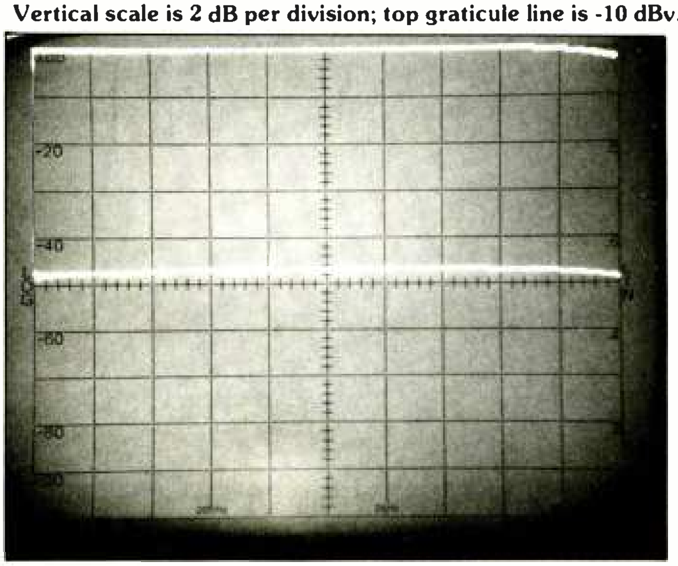
The RM2408 microphone input port is an active differential transformerless pre-amplifier, the port input impedance magnitudes of which are shown in Figure 1. The signal source impedance for all data shown for this test series is 150.0 ohms, a commonly encountered source resistance for most microphones and many other signal sources. A microphone connected to the RM2408 input would be "looking" into a load impedance magnitude of only about 205 ohms at 1 kHz; this could be a higher value to the advantage of microphone transient response, frequency response, distortion, and noise performance. I feel that this unusually low mike input port impedance is a major contributing factor to the marginal equivalent input noise performance observed (see Table 1). A microphone load impedance of about 1,200 ohms through most of the audio spectrum below 20 kHz should be achievable without extreme design or cost measures being taken.

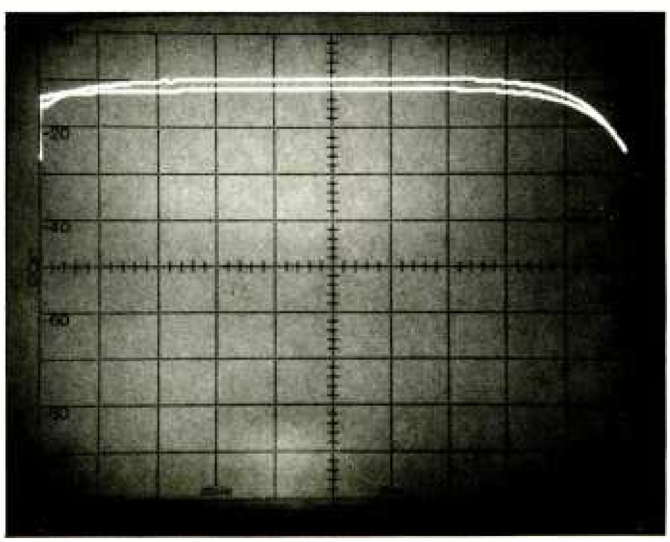
Insertion of the 20 dB input pad raises the microphone load impedance to about 4,120 ohms at 1 kHz. The wide difference in microphone loading between a padded and unpadded condition would result in very perceptible differences in microphone signal coloration; this is especially the case of a dynamic mike. The rise in unpadded terminating impedance magnitude in the region below about 200 Hz implies that there would be a roll-off in low-frequency response in the unpadded case, yet no such radical change in impedance for the padded case.

The microphone input circuit is very well balanced below about 10 kHz. The impedance curves for pins 1-2 and 1-3 match very closely in magnitude, and this balance is confirmed by the common mode rejection ratio data shown in the sweep photograph of Figure 2: The microphone input circuit CMRR holds at about 47

Figure 2: (*left*) Microphone input port common mode rejection. The top trace is the normal mode response as observed at the I/O insert send. The bottom trace is the response at the same output port under common mode input drive conditions; vertical scale is 10 dB per division, and horizontal scale log 20 Hz to 43 kHz. Top graticule line reference is -10 dBv.

Figure 3: (*right*) Logarithmic 20 Hz to 43 kHz frequency sweep of microphone input at extremes of pre-amp gain settings. Vertical scale is 2 dB per division; top graticule line is -10 dBv.



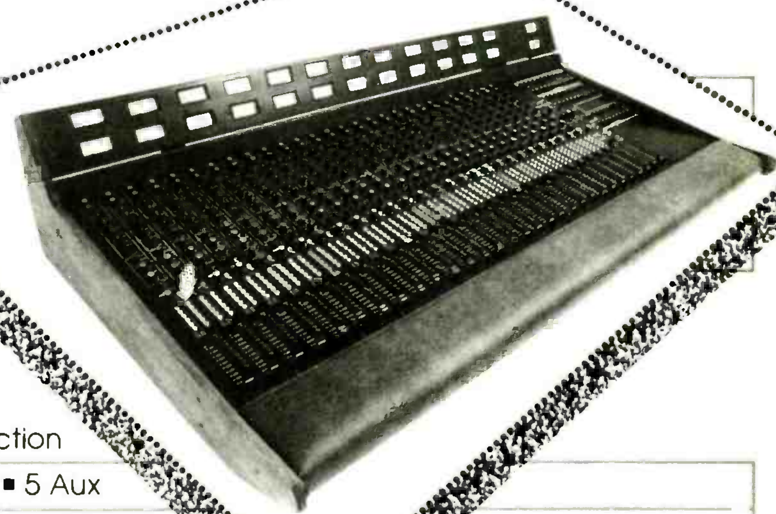


That British Sound

▶ **ITAM #1610**: 1" 16 track
Recorder/Reproducer **$11,950.00**
■ 1" 8trk., pre-wired **$9,950.00**

BOTH TAPE MACHINES FEATURE:
+4dBm IN/OUT ■ 15/30ips
■ Full-function 9 cue position remote-autolocator ■ stand
■ 50% range vari-speed

◀ **ACES TR-24**: 2" 24 track
Recorder/Reproducer
$19,950.00
■ 2" 16trk., pre-wired 24 trk
$14,950.00



▶ **ACES ML24:**
I/O console, 32 in x 24 buss,
Integrated wired patch bay
$17,025.00

■ **ACES SM16**: Split console,
32 in x 16 buss **$15,665.00**

BOTH CONSOLES FEATURE: Fully modular construction ■ Audiofad long-throw conductive plastic faders ■ 5 Aux sends ■ 5 band switchable EQ ■ Input LED PPM's ■ Stand ■ +48v phantom power ■ LED display (optional) ■ Two year parts warranty ■ Other frame sizes available ■ Many options available

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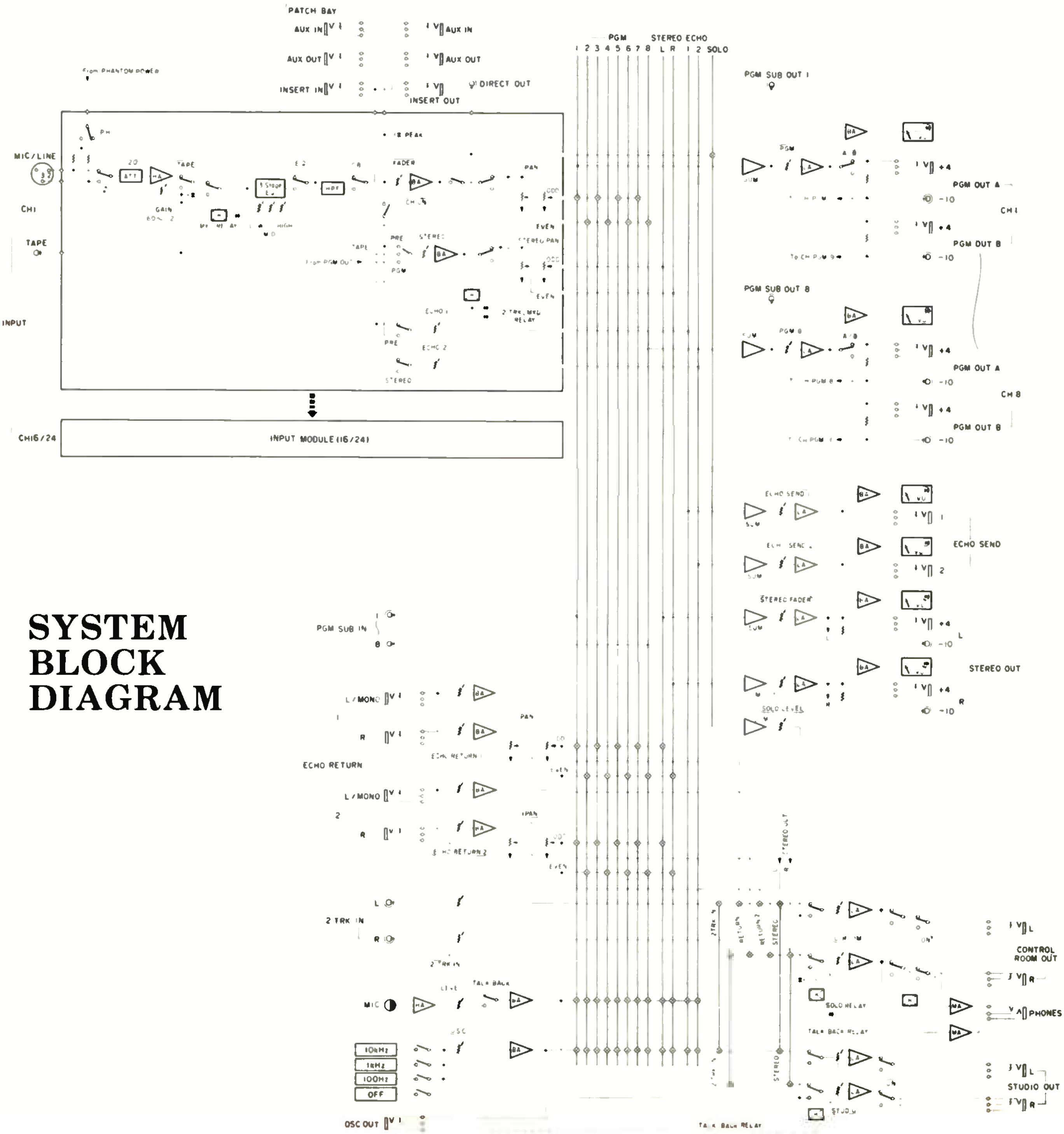
YAHAMA RM2408 CONSOLE REVIEW

dB or better over the 20 Hz to 43 Hz range of the logarithmic sweep. Stability of microphone circuit frequency response over a similar frequency range is shown in Figure 3; the upper trace is the I/O Insert Send response for an indicated unpadded pre-amplifier gain of 60 dB, while the lower trace is the result for a pre-amplifier gain of 20 dB. The frequency sweep is logarithmic, 20 Hz to 43 kHz, and the vertical scale 2 dB per division. The top graticule line represents -10 dBv. It can be seen that the differences between the two extremes of gain setting are fairly minor.

The complex transfer function of the microphone input circuit for several cases of input channel configuration are shown in Figure 4. One will first notice the difference in low-frequency response between the condition of a padded and unpadded microphone pre-amplifier input port. We inferred that this might be the case from the impedance magnitude traces of Figure 1. There is about 7 dB greater relative response at 2 Hz for the padded case compare with the straight input. The severe cusp of the unpadded mike input absolute phase curve implies that a leading phase (not polarity) shift occurs somewhere slightly below 2 Hz.

In the padded case, this amount of phase shift is not approached until a considerably lower frequency. The magnitude response curve lying between the two mike circuit responses is the I/O module response for a padded Mic Input signal as observed at the I/O Insert Send port with the Equalizer circuit IN, and the EQ setting at their null positions. Insertion of the Equalizer results in a slightly



SYSTEM BLOCK DIAGRAM

SUMMARY OF YAMAHA RM2408 SPECIFICATIONS

Description	Quoted	Observed
Frequency Response:	0, ±3 dB, 20 Hz to 20 kHz; 0, ±0.05 dB, 30 Hz to 15 kHz.	Mike In/Direct Out; 0, -3 dB, 6 Hz to 90 kHz; 0, -0.5 dB, 20 Hz to 30 kHz.
Total Harmonic Distortion:	Less than 0.1% at +4 dBv; 20 Hz to 20 kHz.	0.020% at 20 Hz; 0.023% at 50 Hz; 0.021% at 1 kHz; 0.018% at 10 kHz; 0.020% at 15 kHz; (all for +4 dBv IHF load, Mix Bus output.)
Microphone Equivalent Input Noise:	-128 dBv EIN 150 ohm Rs, Max; Input Grain "-60"; 20 Hz to 20 kHz.	-126.3 dBv to -126.9 dBv, range of four mike inputs, taken at channel insert send; 20 Hz to 20 kHz bandwidth.
Residual Output Noise:	-95 dB; All Faders Down (Parameters as above.)	-78.5 dBv Stereo Bus High-level output; pan center; all 24 channels assigned; 20 Hz to 20 kHz; 150 ohm Rs.
Hum and Noise:		
PGM Master at Maximum; one channel fader at Nominal	-64 dB (Parameters as above)	-72.7 dBv; Bus Out.
Stereo Master at Maximum; Channel Stereo Sends at Minimum:	-70 dB (Parameters as above)	-72.4 dBv; Stereo Bus Out.
Stereo Master at Maximum; one channel Stereo Send at Nominal:	-64 dB (Parameters as above)	-72.5 dBv; Stereo Bus Out.
Echo Send Master at Maximum; Channel Sends at Minimum:	-76 dB (Parameters as above)	-80.8 dBv; Low-level Echo Send Out
Echo Send Master at Maximum; one channel Stereo Send at Nominal:	-75 dB (Parameters as above)	-75.1 dBv; Low-level Echo Send Out.
Crosstalk:	-70 dB at 1 kHz; adjacent input. -70 dB at 1 kHz; input to output.	-78 dB at 1 kHz adjacent input; Maximum gain, mike-in to direct-out. -80 dB at 1 kHz into direct-out; fader minimum (see text).
Channel Equalization:	±15 dB maximum; Peak/Notch High: 2 kHz to 20 kHz Mid: 0.35 kHz to 5 kHz Low: 50 to 700 Hz.	±14 to ±16 dB maximum Peak/Notch Center-Frequency 35 Hz to 22 kHz (See figures 11A and 11B).
High Pass Filter:	12 dB per octave; 80 Hz HP	-3 dB at 74 Hz; 10 dB per octave.
Talkback:	Mike, pre-amp, level control; PTT to PGM, STEREO and ECHO busses.	Yes (No remote).
Oscillator:	Switchable sine; 100 Hz, 1 kHz, 10 kHz.	*f (Hz) Level (dBv)*: 107 -9.0; 1042 -9.9; 9692 -9.6
VU Meters:	12 Illuminated meters; PGM Bus 1 thru 8, Stereo L and R (0 VU = +4 dB), Echo Send 1 and 2 (0 VU = -10 dB).	Confirmed
Peak Indicators:	LED on channel strip lights when Pre-fader level reaches 3 dB below clipping; LED on PGM meter lights when Post-Master Fader level reaches 6 dB below clipping.	Confirmed.
Phantom Power:	+48 VDC to XL pins 2 and 3; pin 1 returns.	Confirmed. Master switch selects all channels.
Dimensions: (W × H × D)	1301 × 279.6 × 769mm (51.25 × 11 × 30.25 inches).	
Weight:	55kg 121.3 pounds.	
Power Requirements:	120 V, 60 Hz; 150 W.	Confirmed. Fixed U-type grounding AC power cord.
Power Supply Dimensions: (W × H × D)	480×140×300mm (18.875 × 11 × 11.75 inches)	AC power cord.
PSU Weight:	8kg/17.6 pounds	
ADDITIONAL		
Polarity:		XL pin #2 High; All other input ports and all output ports are single-end, ungrounded condition "HI." A postive-going transition at any input port HI yields a positive-going transition at all accessible output ports, monitors.

Price: Suggested retail price, including PSU: RM2408 $9,900; RM1608 $6,600.

Manufacturer: Yamaha International Corporation. P.O. Box 6600 Buena Park, CA 90622, (714) 822-9011.

higher magnitude drop at the high- and low-frequency extremes of the microphone input function, relative to the padded microphone function with the Equalizer OUT. Its absolute phase companion is also shown. The magnitude and absolute phase response functions for the case of unpadded, minimum-gain microphone input with the High Pass filter IN should be obvious by casual inspection. The nominal 80 Hz break occurs at about 75 Hz in this case, and the roll-off slope approaches about 10 dB per octave in the 10 to 20 Hz range.

High-level input port impedance magnitudes are shown in Figure 5; the I/O module Tape input impedance plateau resides at about 7.2 kohms in the mid-band region, rising below 10 Hz and shelving to about 1.5 kohms above 100 kHz. The typical I/O insert return impedance magnitude is about 9.7 kohms, showing a slight rise to about 14 kohms below 50 kHz. The typical Mix insert return impedance measures about 22.4 kohms, declining to about 12.5 to 13 kohms at the high end of the band.

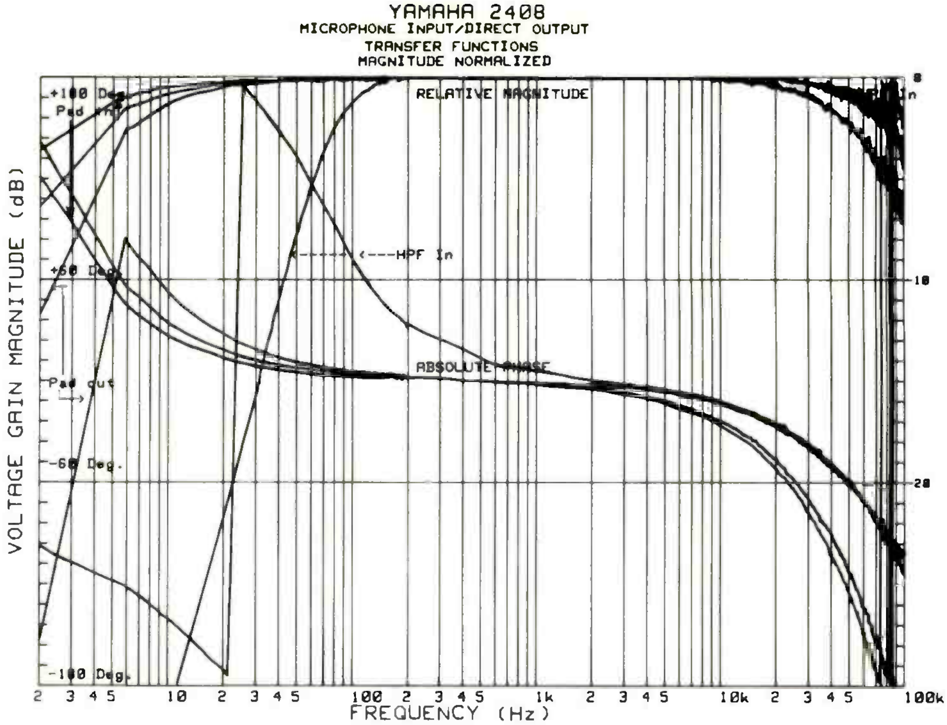


Figure 4: Transfer functions, microphone input driven by a 150.0 ohm generator. Output port is Direct Out with indicated features activated as indicated.

The major output port source impedance magnitudes are shown in Figure 6; the highest source impedance measured was about 800 ohms for the I/O Direct output port, while Insert sends and Mix outputs measured 400 to 500 ohms. The rise in source impedance magnitude at the low end of the band shown indicates that the driver output DC blocking capacitor reactance becomes significant below 20 Hz in all cases. The interface of devices having fairly high input impedances is definitely indicated, whether the low-level phono outputs or the ¼-inch phone jacks are used as sources. Any load less than about 5 khoms would tend to unacceptably degrade the RM2408's performance. Obviously, the higher the load impedance, the better will be frequency response and low-end distortion performance of the console in actual application.

TABLE 2: VU METER CALIBRATION VERSUS FREQUENCY

Frequency (Hz)	Meter Reading (dB)
5	-0.75
10	0.0
20	-0.1
40	0.0
50	0.0
100	0.0
200	0.0
500	0.0
1k	0.0
2k	0.0
4k	0.0
8k	+0.2
10k	+0.2
12.5k	+0.6
16k	+0.9
20k	+0.8
25k	+0.6
30k	+0.6

Note: The metered channel output was adjusted to constant level versus frequency as indicated by external standard meter.

□□□

YAHAMA RM2408 CONSOLE REVIEW

The response of the non-program ports is an important, although infrequently emphasized, aspect of mixing console performance. Upon reflection, one can easily understand that if the monitor, effect, cue and foldback sends alter the nature of the primary signal characteristics, the perception of the program performance will be altered accordingly. Even if a recording console's main program channel is impeccably transparent, this virtue will be of little practical importance if the mixer perceives it to be distorted, or of non-ideal magnitude or phase response. Attempts to correct the perceived defects will result in the degradation of the erstwhile impeccable signal in a manner that approximates reciprocation of the imperfections judged to exist by the engineer, producer or artist.

Figure 7 shows the unpadded, minimum-gain microphone signal transfer function as observed at the Echo Send, Bus #1 and Control Monitor ports. The Bus #1 trace is the reference against which we shall compare the Monitor and Echo send functions in judging the acceptability of the non-program signal ports. Bus #1 passband (+0, -3 dB) extends from slightly below 5 Hz to about 60 kHz. This passband is restricted further from about 8 Hz to about 30 kHz, as observed at the Control Monitor and Echo send ports. By the previous argument, a broader Monitor and Echo response would enhance the quality of the RM2408's performance, both as perceived and realized. We would also prefer to have as little phase shift as could be managed through the system in all respects.

A multiple overlay of the console's transfer function absolute group delay characteristics is shown in Figure 8; group-delay plateaus for each of the paths are substantially flat and have values less than 10 microseconds for cases except for the Stereo Mix bus. The program channels do not show the drop in signal delay time commonly observed in many audio signal devices at high frequencies.

Declining group delay with rising frequency is a result of rates of phase-shift lag versus frequency that deviates from a linear relationship within the band of interest. Such a situation is called a "non-linear system phase response," and explains why systems having linear phase response behavior display the constant group delay over large portion of their passband. The audible consequences of this kind of phase response is conducive to more accurate reproduction of tran-

Figure 5: Magnitude impedance of three high-level inputs. Generator source impedance is 5,047 ohms.

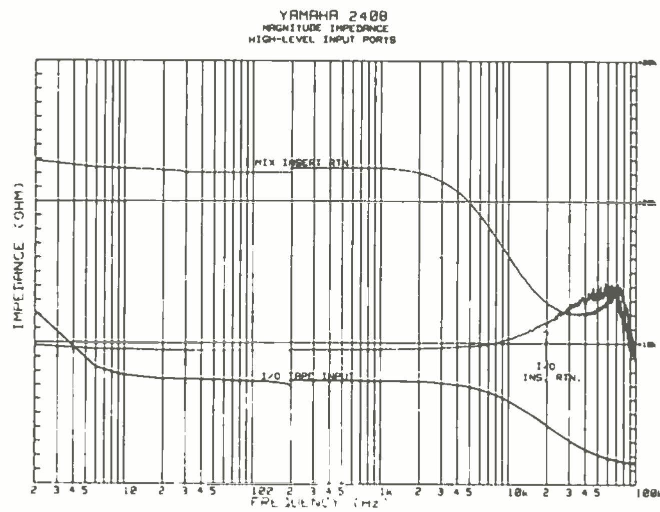


Figure 6: Magnitude of impedance of high-level (+4 dBv) output ports. Driving generator source resistance is 150.0 ohms.

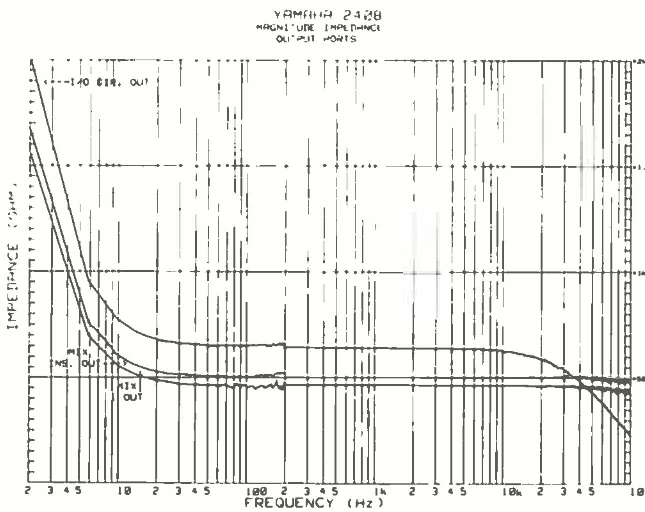


Figure 7: Microphone input transfer functions as observed at various non-program output ports. Driving source resistance is 150.0 ohms; all loads are 10 kohms.

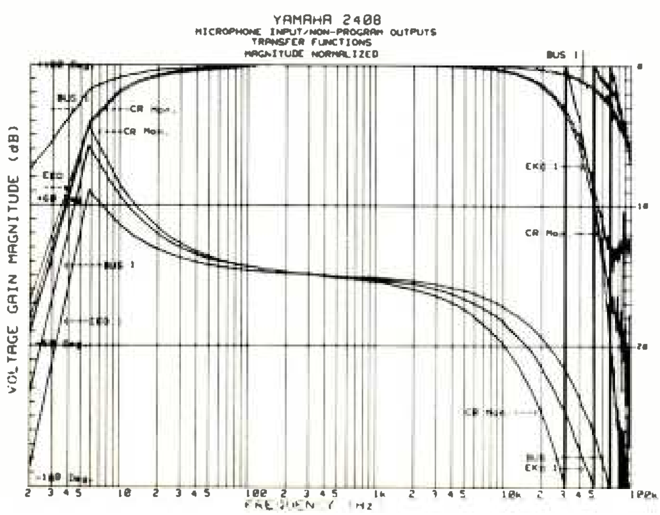
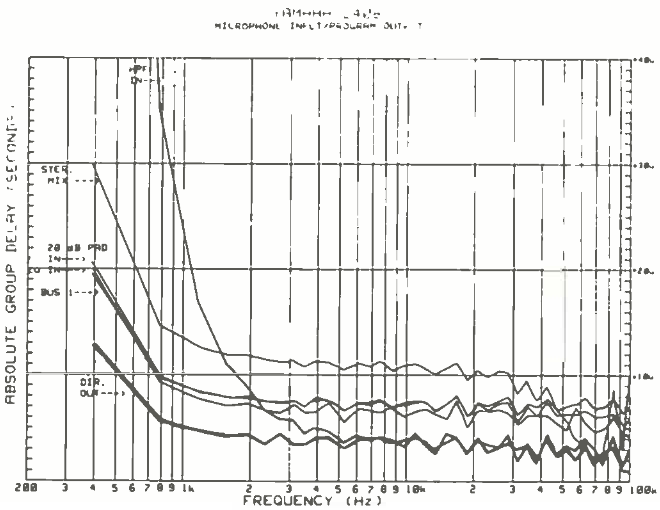
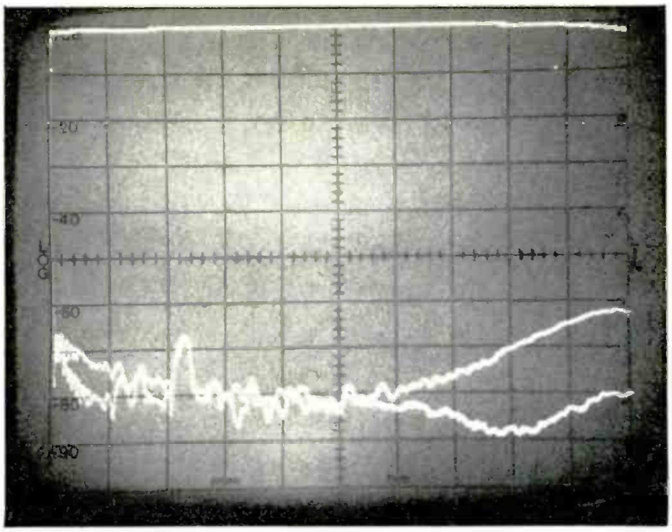


Figure 8: Group delay performance of various ouptut channels as driven from a microphone input.





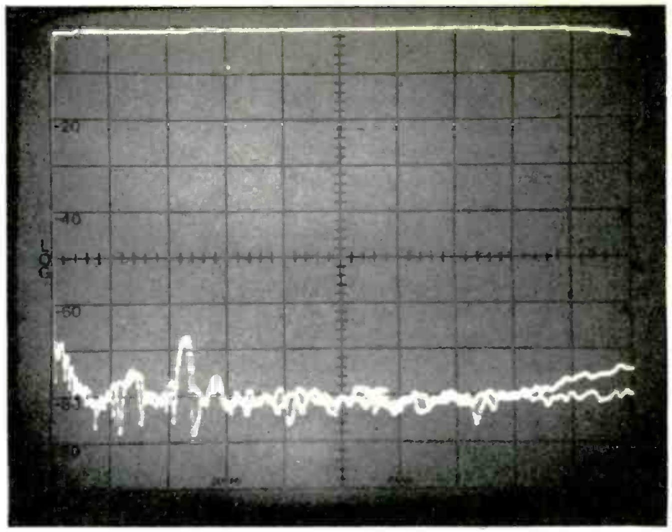


Figure 9A: (*left*) **Crosstalk between adjacent I/O channels, microphone pre-amps set to their maximum gain. Top trace is the Direct Output of the center channel. The lower traces are the Direct Outputs of the flanking I/O channels, their microphone inputs loaded by shielded 150 ohm loads. The sweep is the log 20 Hz to 43 kHz range; vertical scale is 10 dB per division. Top graticule reference is 0 dBv.**
Figure 9B: (*right*) **Identical data as in Figure 9A, for a different set of I/O modules.**

sient signal components, and to the more precise imaging of sound sources within the stereo panorama. The data shows that the RM2408 displays linear phase characteristics in many of its signal paths, a highly desirable feature that is not often observed in lower-priced mixing systems. Indeed, I have not observed it in many top-line designs.

One means of achieving this desirable linear phase property in a multi-stage signal channel is to design each succeeding stage of the system so that the upper and lower band edges are determined by single-pole networks that coincide in frequency. The single-pole lowpass characteristic is a degenerate case of all linear-phase transfer functions. If this is done for a number of cascaded stages, without regard for the gain of each stage, the resulting system phase response will approach that of a Bessel-Thomson (constant delay, maximally flat) filter. Such a design adds substantially nothing to the cost of a unit, since the objective can be achieved merely by determining and specifying the proper values of resistive and capacitive components for each cascaded stage; it seems as though someone at Yamaha may be aware of this design technique.

The departure of the stereo mix bus group delay response from its fairly flat 10 to 12 microsecond plateau below 20 kHz may be an oversight on the part of some member of the RM2408 design team, or it may be that the limit of that design philosophy had been reached and could not be carried further than the primary program signal paths within the physical and budget restraints of the system. Whatever the reason, the effort is worthwhile as far as it goes.

The ability of a console to route signals to their proper destination is only one measure of its effectiveness as a signal manipulation tool; another is its ability to prevent the appearance of signals where they are not wanted. Figure 9A shows crosstalk data for I/O 23 flanked by I/Os 22 and 24, each with their microphone pre-amps set for maximum gain. The sweep is logarithmic, 20 Hz to 43 kHz, 10 dB per division, vertical. The top line is the driven channel, I/O 23, direct output, 0 dBv reference level; the bottom trace I/O 22's direct output; and the center trace I/O 24's direct output. Crosstalk rejection at about 1 kHz runs about 80 dB for both adjacent channels, rising to a worst case of 64 dB for I/O 24.

Figure 9B presents the same sort of data as that shown in 9A, but for the case of I/O 22 driven, I/Os 21 and 23 idle at maximum pre-amp gain.

Panpot leakage for extreme pan to odd PGM Mix bus for a single I/O channel feed is detailed in Figure 10, again showing the log frequency sweep and 10 dB per division vertical scale. The top trace is the panned PGM Mix bus #1 output, +4 dBv reference level; the center trace shows bleed through the bus #1 assign switch with the bus panpot set to full clockwise, odd bus pan; and the two lower traces that appear to be overlaid as one are the signal bleed to bus #2 with the bus assigned and not assigned for the full clockwise panpot position as before. The rated 1 kHz crosstalk attentuation for each of the three conditions is in the range of 84 to 86 dB.

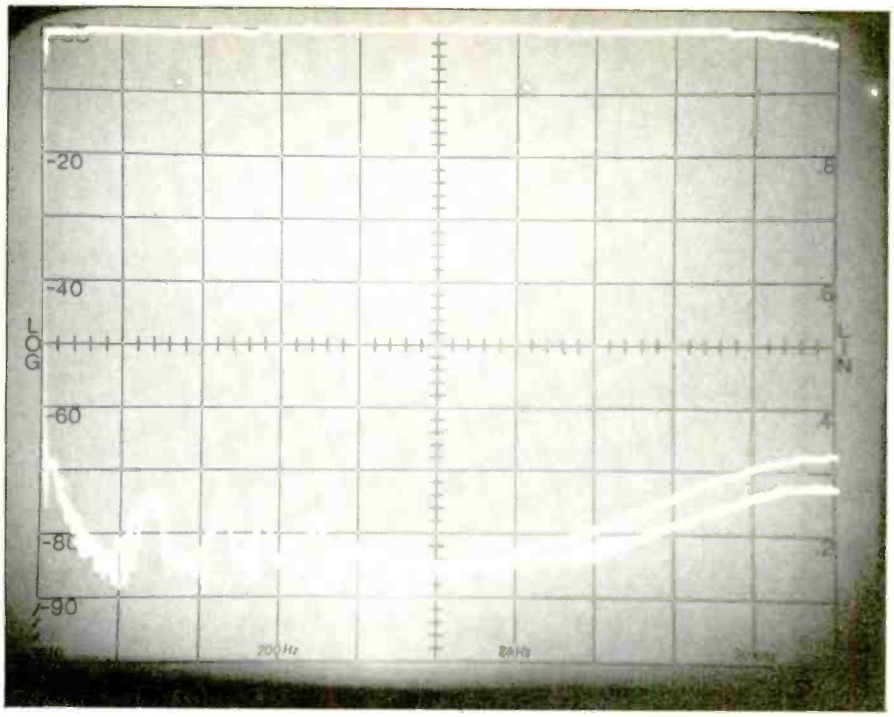
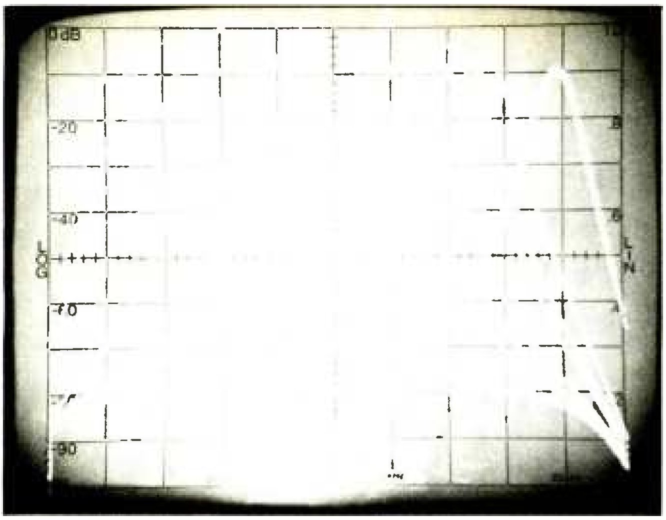


Figure 10: Signal leakage through panpot and Program Mix bus assigns. Top graticule line is 0 dBv. Vertical scale is 10 dBv per division; sweep log 20 Hz to 43 Hz.

The I/O equalizer peak/notch curves are represented in Figures 11A and 11B; the sweep is again log frequency and the vertical scale 2 dB per division. The maximum peaking settings for various frequency settings are given in Figure 11A, while the maximum notching settings are shown in Figure 11B. The relatively flat trace shows the I/O channel response with the equalizer set for what was taken to be a null position. The curves appear to be fairly symmetrical and well-behaved through-



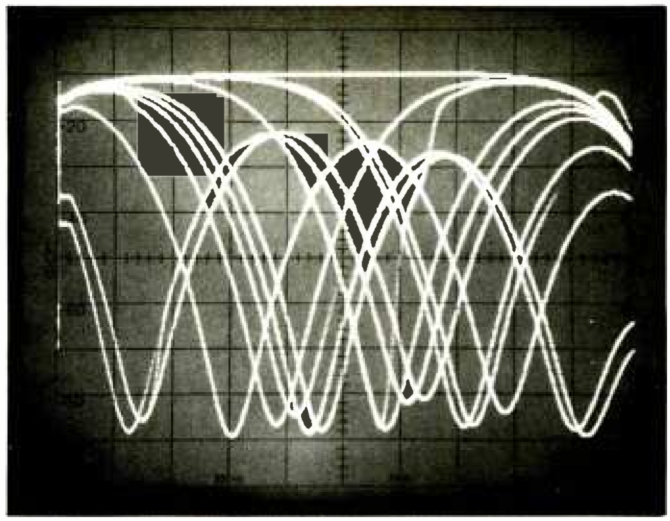


Figure 11A: *(left)* **I/O channel equalizer peaking at maximum boost for various frequency settings. Vertical scale is 2 dB per division; sweep logarithmic 20 Hz to 43 kHz. Bottom trace is the equalizer response with controls set to a nominal "null" position.**

Figure 11B: *(right)* **I/O channel equalizer notching at maximum cut for various frequency settings. Vertical scale is 2 dB per division; sweep logarithmic 20 Hz to 43 kHz. Top trace is the equalizer response with controls set to a nominal "null" position.**

out their ranges, although there does seem to be a slight variation in the maximum peak/notch height/depth with changes in center frequency.

Table 2 gives the response of a typical PGM bus VU meter response for a constant signal level as observed at the output of the relevant program feed, for a constant reading as indicated by a standard meter. The absolute calibration of all of the meters was about 0.5 dB below a true (±1%) +4 dBv (average) output level into a 10 kohm load. The slight rise in meter indication at higher frequencies should be corrected, since the +0.9 dB error around 16 kHz at times could be misleading.

Summary

The Yamaha RM2408 is clearly a significant advance in the field of inexpensive, lightweight audio mixers. It is ruggedly constructed of what appear to be quality components. The change to an in-line monitoring system enhances the flexibility of the mixer over what was possible with the previous PM-1000 and PM-2000 units. Buffering of the VU meters is a step that has helped yield the excelled THD performance quoted in the Specification tabulation of Table 1. Microphone pre-amp equivalent input noise ratings could not quite be observed, perhaps because of the relatively low pre-amp input impedeance that results in what is substantially a terminating load for the microphone. I do object to the radical difference in microphone loading with insertion of the 20 dB pad. This is another matter where simply designing the pad for the proper input/output impedances would have resulted in the desired attenuation without so significant a change in either microphone loading or apparent pre-amp source impedance.

The choice of single-ended high-level input ports is one that I would only grudgingly accept were I to choose components for a prospective system installation. Obviously, cost is an important factor in the design and marketing of a console in this price range; I fully recognize that frills cannot be suffered without consequence of the market place. After all, how many musicians and small-studio operators will pay out any hard-earned cash for something so exotic as a "differential input"? I will candidly admit to prejudice in this matter, but only due to very bad past experiences.

It is possible to configure a very quiet, buzz- and RFI-free system of large proportions centered around a console having differential high-level input ports, even though the high-level output ports may be single-ended. Such is not the case for even a medium-scale, totally single-ended console that is the center of even a fairly small installation. Often, liberties must be taken with the grounding system to obtain initial operation of such a system. Reliability of the system, when new components are added, becomes problematical as the number of current-carrying shields and returns grows with the number of AC power cords and, presumably, ground lifts. The existence of even one device having high primary-to-secondary power transformer capacitance can upset the delicate balance that may have been previously achieved by painful trial and error. The potential for unpleasant electrical shocks to unwary personnel is always a risk in that kind of situation.

The earlier Yamaha PM-1000 consoles featured balanced, floating input and output ports through use of transformers, and it was this detail that made them so widely accepted by the sound-reinforcement fraternity. Single-ended signal ports are deadly in a hostile environment where power grounding merely conforms to local safety ordinances.

The use of phono-type connectors is another matter I have been conditioned to regard with distaste. I have had many problems with the reliability of phonos, especially where they are often connected and disconnected — gold plated or not. This is true of my home sound system, and is doubtless true of the reader's experience as well. Convenience is a marketing factor, so I will rant no more on this subject.

The clipping levels at the +4 dBv output ports was observed ranging from +20.3 to +20.6 dBv into a unit IHF load of 10 kohms: this is inadequate headroom (16 dB) for live acoustical instrument recording that is at all demanding. Because the low-level phono output ports are derived from resistive dividers, the margin before clipping is not enhanced by the dropping of the phono port output level to -10 dBv; the same 16 dB of headroom applies.

The relatively high source impedances of 430 to 600 ohms make the driving of any but high input impedance ports risky, as mentioned above. The still higher phono port source impedance, listed by Yamaha as 2,000 ohms calculated at about 4,700 ohms, bodes ill for lengths of commonly used cable that are longer than about 30 feet. Assuming the use of conservative 30 pF per foot (100 pF per meter) cable, a 30-foot (10m) run would roll-off at about 37 kHz when driven from a 4,700 ohm source.

The Yamaha RM2408 represents a degree of external beauty and a measure of fairly reasonable performance within the limits of its price. This writer's paranoia and insecurity aside, it will likely serve the moderate scale needs of a home recordist or small commercial recording facility in a cost-effective manner. ■■■

MANUFACTURER'S REPLY:

Yamaha International Corporation, Combo Products Division, replies as follows:

As pointed out in Peter's review, the RM2408 compares very favorably with the much more expensive consoles that he is accustomed to in his daily work. At just $9,900 suggested retail, the RM2408 has a wide range of uses, including recording and many broadcast production applications. The review does not contain Peter's specific test procedures, so it is difficult to comment on his observed specs. The RM2408's hum and noise are measured with a -6dB per octave filter at 12.47 kHz: equivalent to a 20 kHz filter with infinite dB per octave attenuation.

LINNDRUM 9000 DIGITAL DRUM MACHINE AND SEQUENCER

Reviewed by Bobby Nathan

To many studio engineers and producers, one of the most exciting new MIDI-equipped products to be launched recently is an improvement on a drum machine that was not only considered to be the first, but is still regarded by many to be the best. The original Linn Drum LM-1 was feared by all who could not master it. Drummers found it easy to have mixed emotions about its place in the recording studio. Engineers, who had built reputations on the strength of their trademark "drum sound," also were not too thrilled about the drum machine's existence. While users felt drum machines could never replace the live drummer, nevertheless the drum machine did find a home in the studio. It has spawned many imitations — some with less features and a lower price, and others with new features. The existence of drum machines has created a staggering amount of interface problems and, of course, with these problems came an even more staggering array of devices to solve them.

Although the LM-1 was first on the market, it was not until Roger Linn introduced the LinnDrum — for almost half the price of the original LM-1 — that the drum machine era had been truly established. And Linn Electronics' newest entry, the LinnDrum 9000, is now twice the price of the "industry-standard" LinnDrum.

At a time when the quality and advanced features of today's drum machines seem to be costing less, one would ask why Roger Linn has taken the gamble to market the LinnDrum 9000 *expressly* for the recording studio and financially secure producer/artist/session players? At first glance the 9000 is just another drum machine but, after a closer look, it is clear that it is no ordinary drum machine.

Front-panel Features

The LinnDrum 9000 has 18 velocity sensing pads to enable drum patterns to be programmed. Each pad (or instrument) has completely independent tunable pitch, velocity, volume and panning controls. Each sequence/pattern will remember all pitch tunings, velocity, volume and panning settings. As a result, if you append sequences, the pitch, velocity, volume and panning will switch automatically. The 9000 is the first velocity-recording drum machine equipped with only one pad for high-hat, but with a most clever slider (or optional foot pedal) to enable every high hat nuance between open and closed to be duplicated. To me, a high-hat sound is an immediate giveaway that you are listening to a drum machine on a particular record. After hearing the LinnDrum 9000's high hat playing a pattern, with not only velocity and dynamics but all the open and closed timbres, it's pretty hard to tell you're listening to a machine — it sounds like a drummer with great time.

The same also holds true for the digitally recorded sounds of kick, snare, claps, two congas, sidestick, cabasa, cowbell, tambourine and, especially, the four tom-toms. Unlike the LinnDrum, on the LinnDrum 9000 it is possible to play all four toms and the two congas simultaneously. There is also a "Tap" button which will allow you to listen to, for example, a cassette in the control room and, by tapping in two ¼-note taps (in time with the cassette), to read the bpm (beats per minute) of the cassette's tempo and simultaneously set the LinnDrum 9000's tempo.

Another interesting little button is labelled "Repeat." When activated, the repeat button will allow you to press any of the 18 pads and have a repeating trigger for the instrument chosen. The rate is set by the error correction amount (¼ note thru 1/32-note triplet). An example would be programming in a press roll on the snare pad for as long as you hold the snare pad, with the loudness of the press roll being determined by how hard you press the pad.

A back-lit Liquid Crystal Display (LCD), which can be read easily in low light environments, shows pattern and measure numbers. The 9000 also features a "Help" button that causes the LCD to display the appropriate help message for whatever function you're currently working with.

Erasing an instrument, measure or pattern/sequence is easy: you simply press "Erase," and the LCD prompts you. For example, the display reads "Erase measure," "1 or all" and you enter the currect info. Future software updates will allow the LCD to display various screen editor functions.

There is no song mode as we have come to know it, with part numbers and corresponding pattern numbers; instead, there are only sequences. To write a song, you would append various sequences onto the end of an existing one. Instead of the display showing part numbers, it reads measure numbers. This latter feature I find to be most suited to the needs of a composer. You can also insert or delete different sequences via "Measure Number In-point" to "Measure Number Out-point" style editing. There is also a scratch-pad built into the software to aid in programming a sequence into a song.

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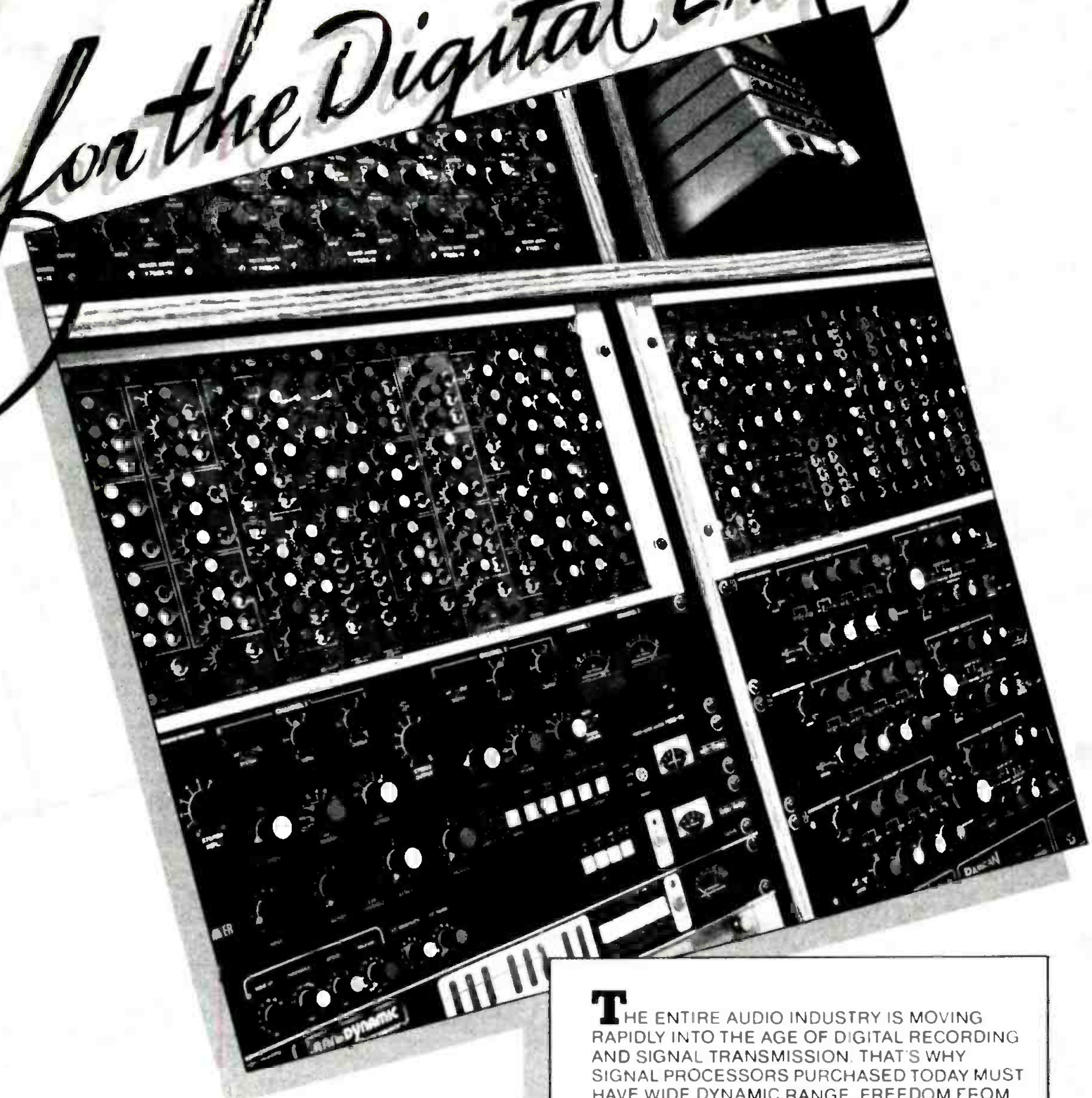

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The new LinnDrum 9000 did not forget the live drummer either, since patterns played by a live drummer on either Simmons or compatible pads can be recorded, as well as real drums with sensors attached. The unit comes with six trigger ins and outs; additional trigger inputs can be purchased as an option. Unlike the LinnDrum, the 9000's trigger inputs respond virtually instantaneously; they can also read analog triggers, allowing audio from tape with dynamics to also trigger the unit.

Data storage is by means of a new design in cassette-dump technology. In contrast to many other units, the cassette dump accepts a very wide range of level from tape. A 3½-inch Sony disk drive is available as an option for memory storage. Unlike the LinnDrum, the 9000 enables new, alternate sounds to be loaded from cassette or floppy disk. Sound good so far? The LinnDrum 9000 will also include, as an optional extra, a sampling card that enables the user to sample up to four drum sounds at one time, and to store the samples on cassette or disk. Obviously, you could program a song and choose between hundreds of stored samples at any time during the project.

The LinnDrum 9000 now incorporates a programmable metronome that is variable from ¼-note to 1/32 triplet; the tempo of each pattern is also programmable. When merging two patterns together of different tempos — for example 124 to 117 bpm —the number of bars it will take to change tempo can be programmed, making severe changes a smooth affair. The 9000's sync tone is still a standard FSK 48 beat per ¼-note tone, but the unit will be able to read its own FSK and output all industry standard clocks: ¼ note thru 1/64-note triplets.

Future options include a sort of MIDI Clock/SMPTE-type of sync tone. Once the master multitrack tape has been striped with the sync tone, you can fast-foward, rewind, and drop

SUMMARY OF LINN 9000 DIGITAL DRUM MACHINE SPECIFICATIONS

Number of sounds: 18, including bass, snare, hi-hat, four toms, two congas, two ride cymbals (regular and bell), two crash cymbals (standard and splash), cowbell, claps, cabasa, sidestick, and tambourine.

Sequences: 100 drum and 100 synth.

Note capacity (keyboard recorder): Over 7,000 (expandable).

Note capacity (drums): Over 24,000 (expandable).

Internal memory: 64K (expandable).

Storage media: cassette or optional installed 3.5-inch disk drive.

Cassette interface: Mike-in, line-in, line-out.

Sync-pulse rate: preset at 48 cycles per quarter note.

Tempo range: 48 to 250 bpm.

Audio outputs: Mix outputs line level, +4 dBm VU, +14 dBm peak with all drums playing at maximum volume level into 20 kohm line input. Direct outputs — line level, typical 0 dBm VU.

External audio inputs: Line level, 2K ohm impedance. 20 Hz to 28 kHz.

Power requirements: 110V or 220V AC; 50 or 60 Hz.

Size (H × W × D): 6 by 24 by 14 inches.

Weight: 29 pounds.

MIDI-In, MIDI-Out, and **MIDI-Thru** jacks.

Trigger In: Six dynamic-sensitive for internal drum pads or virtually any audio source. (Expandable to 12 inputs.)

Trigger Out: Two independently programmable jacks provide an electrical trigger pulse (+5V) suitable for sychronization with external gear.

Footswitches: Two inputs independently programmable for hi-hat open/close, start/stop, record punch-in and -out, erase, repeat, error correct defeat, tap tempo, or sequence increment/ decrement.

Cassette: Intended for loading and controlling data from a cassette recorder, the MIC OUT is specifically designed to accommodate cassette players with microphone input only.

Sync: During playback, the 9000 will sync itself to tape by the OUT sending a tone that can be recorded onto one track of the tape machine.

High-Hat Pedal: This feature allows the voltage control pedal to remotely control the hi-hat decay.

Click: Separate output for recording click.

Individual Outputs: Direct outs for all drum and percussion voices, thereby allowing individual signal processing of each ouput desired.

Audio Out: Left and right outputs for stereo mix.

Audio In: Two general purpose inputs to the mixer may be used for outboard gear.

OPTIONS:

Audio input card for sampling (recording) your own sounds.

3.5-inch on-board **disk drive** to store/reload sequences and/or sounds.

64K or **128K static RAM** memory expansion, up to 256K.

SMPTE reader and generator card.

Additional six trigger input card (total: 12).

Price: $4,990 retail for basic machine without options listed above.

Linn Electronics, Inc., 18720 Oxnard St., Tarzana, CA 91356. (818) 708-8136.

the multitrack into play anywhere on the tape, and in song mode the 9000 will act like a SMPTE transport by follow-chasing the multitrack tape and be perfectly sync with the tape every time. This is a real track saver for an eight-track studio, since the drums do not have to be printed on tape, and the latter does not have to be started from the top of the tune every time for the drums to synchronize.

MIDI Sequencer

As if all these features weren't enough, the LinnDrum 9000 is not *just* a drum machine. Instead, it should be considered as a multitrack tape machine for MIDI-equipped synthesizers. The sequencer section includes 99 sequences, each with 32 tracks, and features Copy a Track, Append a Track, Copy a Sequence, and Append a Sequence functions. Error correction must be determined before recording, but with the punch in/over modes, you can change the error correction for different riffs. A method you might like to try to error correct after the fact would be by patching a MIDI cable out of the MIDI-Out jack and into the MIDI-In jack, and bounce your first track to another track with error correction selected for the new track. In this way you can record in real time and choose to quantize a track later. Or you can save various performances on different tracks and then choose which track to use; you can mute playback of any track or solo it; you can either overdub on an already existing track or erase it by recording over it; you can change the MIDI channel status of any given track after it has been recorded; and so on.

The way the built-in sequencer records the data is more like a drum machine, than other sequencers. You start off by pre-determinating how many bars your sequence will be. Then, when recording on a track in that particular sequence, that track will stay in record and loop. For example, if you are trying to record a four-bar pattern that has a pickup on the *3-and* of bar #4, you can start playing there, and keep recording until the *3-and* of bar #4 comes around again. If you stop playing there you will hear what you just played in. If you missed a note you can add it the next time it comes around. You can even overdub pitch bend, or modulate the next time it comes around. Or you can punch-in on any track anytime without erasing what was there, unless you choose to do so. Oh, I forgot to tell you, the sequencer also records velocity. And, depending on how you compose, you can record a sequence or sequences assembled into a song first, and then program the drums to your sequences, or vice-versa.

The controls for the sequencer are more like those of a tape recorder — you have Fast Forward, Rewind, Play, Record, and Stop. There is even a programmable autolocate button to search out a particular measure number from which to start. While listening to any sequence, you can punch-in, record, punch-out, or fast foward/rewind to a particular place (measure number) in the sequence.

The 9000's MIDI capability is totally together, since it works extremely well with the Yamaha DX7 (possibly the true test of a unit's versatility). A MIDI Echo function is provided — not to be confused with digital delay or tape slap — an example of Echo use would be the setting up of a DX7 as a master MIDI keyboard to write a bass line onto track #2A the 9000's sequencer. Track #2 is set to MIDI channel #3. A Roland Super Jupiter synthesizer controller is then plugged into the sequencer's MIDI-Out jack, and set to receive only data on MIDI channel #3. The MIDI Echo feature allows the user to monitor the sound of the Super Jupiter, while playing the DX7 keyboard during record and playback. (In fact, this topic deserves a separate article of its own, which I will try to cover in an upcoming issue of *R-e/p*.)

Because the 9000 is a drum machine and a sequencer built into one unit, the need for an interface box is nonexistant. The inside of the 9000's mainframe is laid out in a similar way to the IBM PC, with extra slots for memory expansion, printer interface, monitor interface, and a modem option that would allow for a musician in L.A. to compose songs with another composer in New York. (In essence, a studio musician at home with his 9000 could download a song from the studio, add his part, and then upload it back to the studio via modem and then to tape.)

The biggest frustration with all drum machines/sequencers has been getting them to interface with other devices. The next biggest problem that has plagued us is that, up to now, sequencers have not been user friendly. The set-up time required to establish a sequencer/drum machine interface has been most time consuming. Now, with the first stand-alone drum machine/sequencer available for what I consider to be a reasonable price, it remains to be seen what affect it will have on the ever-changing recording studio industry. ■■■

NEW SERIES 600 RECORDING CONSOLES FROM SOUNDCRAFT

Responding to the upsurge in 8- and 16-track recording facilities, the Soundcraft Series 600 is available in 16-, 24-, and 32-channel input mainframes, and features eight group outputs paralleled for 16-track operation. The standard console is fitted with a full 16-track monitor section. In addition, added flexibility is offered by means of direct channel outputs.

LED bar graph metering is standard, and the console is switchable for -10 dBv or +4 dBm operation to interface with the majority of tape machines. Six aux sends are provided, with the capability of each pair to be used either pre- or post-fader, or pre- or post-EQ on each input module. Stereo input modules are also available for audio/video post production.

The Series 600 is priced as follows: 16-channel $6,950; 24-channel $8,750; and 32-channel $10,950.

SOUNDRCRAFT ELECTRONICS, INC.

For additional information circle #143

MODEL 1772 CONDENSER CARDIOD MIKE FROM ELECTRO-VOICE

Designed for use in problem sound reinforcement systems where reflection and speaker placement demand high gain-before-feedback, according to Jim Edwards, E-V's market development

manager, the new Model 1772 "maximizes gain-before-feedback through a smooth, peak-free frequency response, a fine-tuned cardiod pick-up pattern which reduces room reverberation and rejects unwanted background noise, and innovative transducer positioning for exceptional sensitivity."

With wide dynamic range and 137 dB headroom, the new mike is said to reproduce the highest sound pressure levels without overload or distortion. Condenser design and internal LF filters minimize handling noise, and an intergral pop filter protects against wind, breath and pop.

Weighing less than 11 ounces, the 1772 features a durable Memraflex grille and a zinc diecast case with non-reflecting epoxy finish. Battery or phantom power (24 to 48V) options are available.

ELECTRO-VOICE, INC.

For additional information circle #144

AXE ANNOUNCES DI-400 FOUR-WAY DIRECT BOX

The DI-400 Quad Direct Box is an AC-powered, rack-mounted version of the DI-100 direct box, and contains four separate line-level DIs with variable gain. Because the unit is capable of sending a +4 dBm signal, with plenty of headroom, direct to a multitrack, it can eliminate the need for the mixing console altogether. The rack-mounting capability eliminates cables and boxes on the floor when recording synthesizers in the control booth.

The unit can also be wall mounted in the studio's main room, or located on the floor for easy access whenever multiple direct boxes are needed in a central area — such as rhythm section or synthesizer dates, as well as live concerts.

Suggested retail price of the DI-400 is $900.

ARTISTS X-PONENT ENGINEERING

For additional information circle #145

SCV MODEL PC-80 PHASE MEASUREMENT SYSTEM

The Model PC-80 is a portable, absolute phase measuring system allowing the checking of any audio system, including microphones, loudspeakers and loudspeaker systems, console patch points, plus studio wiring.

The system, which consists of a generator and a detector, allows measurement to be made eithe acoustically, by generating and sending a 1 kHz special wide-band pulse to a self-contained speaker and reading it with the detector's built-in microphone, or electrically through standard XLR-type connectors. In the latter mode, the generator features an output level control of 0 to 1 VRMS at 1 kohm load impedance for measurement of wired systems. Modes are switchable from internal mike, line or external mike. LEDs are provided to indicate in-phase (green), reverse phase (red) as well as power on (yellow).

The frequency spectrum is as follows: electrical mode — 1 Hz to 20 kHz (generator and detector); acoustical mode — 200 Hz to 5 kHz (generator) and 100 Hz to 15 kHz (detector). The two separate units each accept standard 9-volt batteries allowing operation of up to 50 hours each.

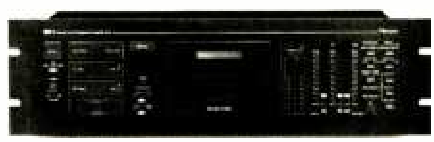
Suggested list price of the PC-80 is $299.

SCV INC.

For additional information circle #146

NAKAMICHI INTRODUCES THREE-HEAD PROFESSIONAL CASSETTE DECK

The MR-1 features a unique Asymmetrical dual-capstan diffused-resonance transport, which is said to be so accurate that no pressure pad is required to maintain tape-to-head contact. A discrete three-head configuration provides perfectly accurate azimuth alignment, extended bandwidth and exceptional

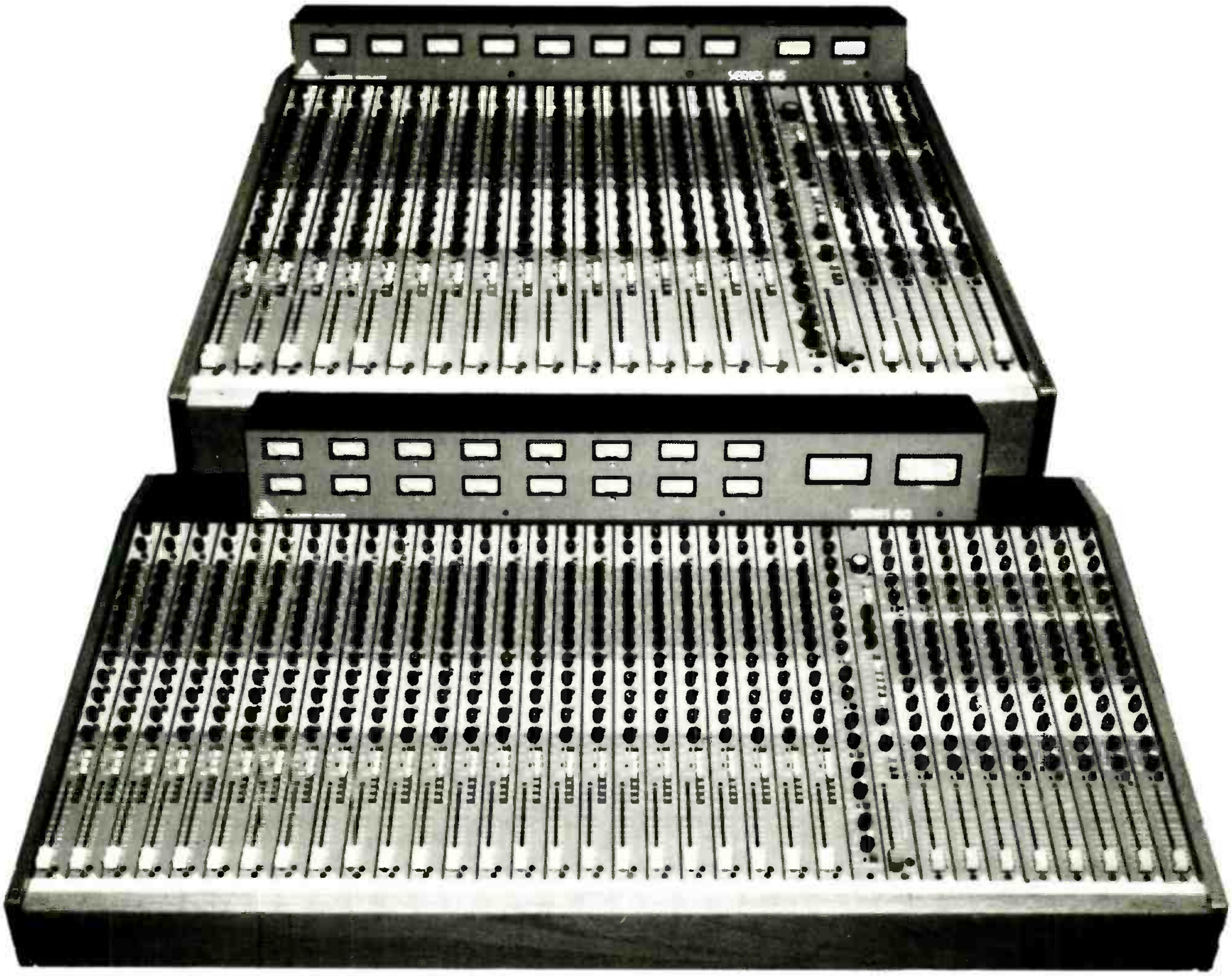


dynamic range, according to Nakamichi.

Front-panel ¼-inch, balanced line-input jacks simplify temporary connection, while rear-panel balanced XLR-type input and outputs provide permanent connection. The balanced input are 600-ohm impedance and rated input level is +4 dBm with 16 dB of headroom; minimum input level is rated at -6 dBm. Balanced line ouputs are designed for termination in a standard 600-ohm load, and have a source impedance of 100 ohms (balanced). Nominal output level is identical with nominal input level (+4 dBm), with 16 dB headroom to saturation. Also, a front-panel headphone jack drives up to 100 mW into a four-ohm load, and is provided

SERIES 65

THE MIXER THAT DOESN'T LIVE UP TO EXPECTATIONS



If you're looking for a compact, economically priced mixer that will grow with your needs for 4, 8 or 16 track operation, you've probably been disappointed with the lack of 'Big Studio' facilities they have to offer. **Series 65** is a 'Big Studio' mixer in a compact frame at a very economical price. Fully modular, these are just some of the stunning features: Four band E.Q. (incorporating 2 swept mid ranges), eight auxiliary sends (balanced), separate mic and line inputs (balanced), stereo solo, auto-mute, monitor equalisation (three band incorporating a swept mid range and routable to group or monitor), four echo returns and unique 'Group assignment' which allows 8 or 16 track recording without the need for cross patching or paralleling outputs. Technically the console offers the same sonic quality that has made **TRIDENT** a legend among world class recording studios around the world. Check out the **SERIES 65** today. In a world where you'd expect alternatives, there is only one : **TRIDENT.**

Everything Audio
Southern California
(818) 995-4175

Leo's Professional Audio
Northern California
(415) 652-1553

RMS Sound
Northwest
(503) 239-0352

Comcast
Indiana
(317) 849-7050

Southwest Pro Audio
Texas
(512) 443-4567

Trackside Engineering
Southeast
(404) 436-3024

Ear Pro Audio
Arizona
(602) 267-0600

Audio Techniques
New York City
(212) 586-5989

SEE US AT BOOTH #633 NAB

TRIDENT U.S.A. INC.
280 Mill Street Ext, Lancaster, Mass 01523 U.S.A.
Tel: (617) 365-2130/(617) 368-0508 Telex: 951546

TRIDENT

New Products

with a headphone volume control independent of the line-output level control.

The unit features Dolby B and C noise reduction and provisions for external NR processors via phono jacks. Dual 16-segment, peak-reading meters with an attack time of 100 milliseconds and a decay time of 2 seconds indicate recording/playback levels in 2-dB steps from -20 dB to +10 dB, re: nominal level.

Frequency response is a quoted 20 Hz to 20 kHz, ±3 dB, at the -20 recording level. The geometry of the Crystalloy playback-head core is said to virtually eliminate contour effect ("head bumps") for smooth bass response; the head's sub-micron gap resolves wavelengths that correspond to recordings at 20 kHz and above. Wow and flutter is a quoted 0.027 WRMS, ±0.048%, weighted, peak.

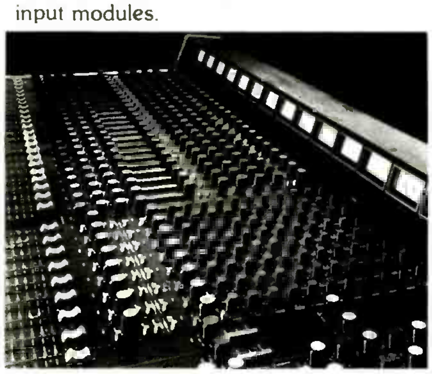
The MR-1 carries a suggested retail price of $895.

NAKAMICHI U.S.A. CORP.

For additional information circle #148

WHEATSTONE MTX-80 LIVE/RECORDING CONSOLE

The latest addition to the Audioarts Engineering line of recording and sound-reinforcement consoles, the MTX-80 features continuously variable gan, phantom power, mike phase reverse, three-band semi-parametric EQ, channel on/off, direct outs, and assignable sends on all input modules.



The eight sub-groups include group on/off switches and 11-by-8 matrix outputs to provide a multitude of send for video, multitrack recording, and remote sound-reinforcement feeds. The group and master left and right modules include auxiliary inputs; the master left and right also include sweepable highpass filter and talkback.

All input and output modules utilize the company's long-throw M-104 fader and include highpass filters, patch in/out, a comprehensive solo system, and peak indicators. The console comes standard with master effects send module, effects return module, and control room module.

WHEATSTONE CORPORATION

For additional information circle #149

SHURE SM98 INSTRUMENT MICROPHONE

Described as the first professional-quality miniature unidirectional condenser microphone designed specifically for instrument and amplifier miking, the new SM98 combines the convenience and adaptability of small size with professional performance capabilities. Reporting on the early professional acceptance of the SM98, Sandy Schroeder, Shure marketing manager/professional entertainment and general audio products, remarked: "Soundmen found placement of the SM98 quick and uncomplicated, especially around a drum set. Musicians commented on the SM98's 'natural' clean sound, and the freedom afforded by its undistracting small size. Compared with widely used, high-performance conventional instrument microphones, the SM98 offers the positioning and performance advantages of miniaturized components, and advanced condenser technology which has generally higher performance than dynamic microphones."

The mike utilizes advanced technology in its low-noise, low-distortion pre-amp, allowing it to withstand close miking of drums, brass instruments, amplifiers, and other high SPL sources without distortion. It features a flat frequency response with switchable low-end rolloff. Due to its small size — half-inch diameter, 1¼ length — the SM98 is said to offer a near-perfect cardiod polar pattern at all frequencies.

Included is a unique swivel adapter that allows the microphone to be used with all standard stands, booms, and goosenecks. Specialized mounting kits for Shure's SM83 are easily adapted to the new model, as is Shure's Z-bracket for amplifier miking.

The pre-amp is powered by two standard 9-volt transistor batteries or a Simplex (phantom) power source, and features on/off and a low-end rolloff switch.

User net price on the SM98 is $250.

SHURE BROTHERS, INC.

For additional information circle #151

AUDIO+DESIGN/CALREC ANNOUNCES COMPEX 2 COMBINED LIMITER, COMPRESSOR, EXPANDER/GATE

Like the original Compex, the new model offers separate compression and expansion/gating, as well as overall peak limiting. In contrast, the dynamics control is effected with an extension of softer ratios and thresholds to -60 dB below normal operating levels — thus wide dynamic programs can be processed in a subtle manner throughout the program with ratios as low as 1.25:1, A+D/C claims.

Other features include a choice of Log/Lin compressor release times, plus a unique AGC auto release characteristic. In the expander section, the ratio may be continuously varied from 1:1.2 plus auto mode for use with wide-band material. A further function of the expander/gate section is the Gate-Hold circuit, which can be triggered to delay the selected release time by anything up to two seconds, to allow a fast release to be used and a selected beat sequence to be repeated through the rating window.

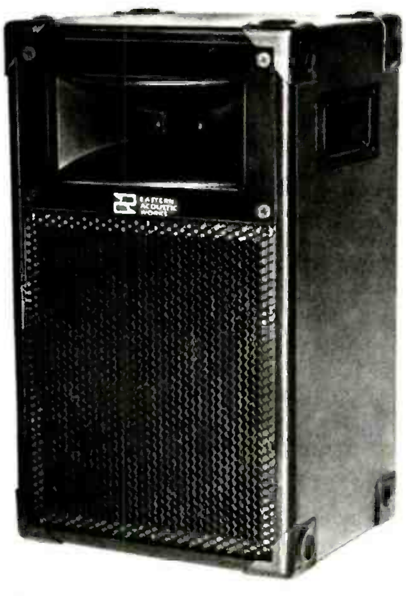
AUDIO+DESIGN/CALREC, INC.

For additional information circle #152

EAW ANNOUNCES FR-102 FULL-RANGE LOUDSPEAKER SYSTEM

Offering "high efficiency and wide coverage in a ultra-compact package," the FR-102's wide horizontal coverage and extended HF bandwidth are attributed to its new RCF N-252F compression tweeter. The N-252F utilizes magnetic dampening fluid, polymer film diaphragm, and constant horizontal geometry horn, to provide "exceptionally smooth response" up to 18 kHz with coverage greater than 120 degrees. The advanced high-temperature magnetic dampening fluid is said to significantly lower distortion at high sound pressure levels, while increasing long-term power handling.

While the small size of the FR-102 limits LF response to 70 Hz, active equalization is said to enable the useable response to be extended to 50 Hz. Additionally, the unit can be used with an optional subwoofer for applications where low bass is required.



Unusually smooth transition response — ±1.5 dB, 150 Hz to 12 khz — is claimed to be achieved through the use of a third-order equalized crossover network with large air-core inductors and 5% toleranced resistors and custom capacitors that provide 18 dB per octave slope.

The FR-102 is described as being ideal for close-field playback applications where space is limited; highly portable sound reinforcement systems, under-balcony fill systems; and, with a subwoofer, small high-level dance systems.

EASTERN ACOUSTIC WORKS

For additional information circle #153

NEW OPTIONS FOR THE KURZWEIL 250 DIGITAL SYNTHESIZER

The options include a user sampling program, an expanded 8,000-note sequencer memory, and a new block of preset sounds that triples the number of instrument voices pre-programmed into the keyboard. Also announced is a new personal computer interface for the Model 250 that enables users to store musical creations onto floppy disk.

The new Sound Modeling Program, designed for use with the Apple Macintosh personal computer, allows the sound of any acoustic instrument to be recorded, and played back on the 250 keyboard. The 250's resident programming features then can be used to modify the sampled sounds in a variety of ways. The Sound Modeling program allows from 20 to 100 seconds of sound to be sampled, at a rate ranging from 5 to 25 kHz.

The 250's 12-track sequencer is now expandable to a total of 8,000 notes — more than twice the memory available in the standard unit. The expanded sequencer accesses board editing capabilities that allow a single track, a single note, or an entire recorded sequence to be modified, and can also be used to automatically transpose the stored sequence to a different music key, or to speed it up or slow it down.

The 250's initial 30 instrument voices are now supplemented by a new Sound Block of 15 voices, provided in the form of a ROM board that is retrofitted to the unit. Sounds now available through the Sound Block include a choir of human voices, woodwinds, harp, and electric bass.

The new computer interface enables the memory storage capacity of the instrument to be expanded using an Apple Macintosh. Known as the MacAttach Communications Package, the interface consists of a cable and disk file management program, and allows sampled sounds, keyboard setups, sequences, or lists of setups to be stored and called from the Mac's memory.

KURZWEIL MUSIC SYSTEMS, INC.

For additional information circle #174

IMS INTRODUCES SERIES 400 AUDIO ROUTING SWITCHERS

Built-in GPIB, RS-232, and RS-422 interfaces allow an audio synchronizer, event controller, or video computer editor to control the Series 400 Smart Switcher and its functions. The unit provides up to 1,024 crosspoints in only 10½ inches of rack space, and is field-expandable to virtually any matrix size.

The Series 400 memory capability is said to speed up the process of patching and mixing audio effects, by allowing a microcomputer to access any number of tape machines and special effects sources. The operator can check on routing status from a read-out of the source and destination of all connections.

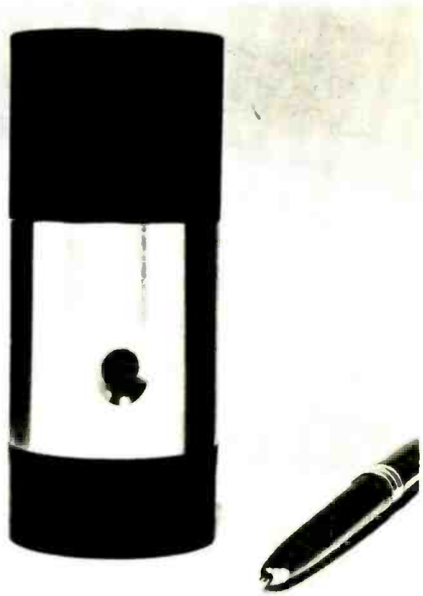
"More and more, the market is demanding increasingly sophisticated audio effects, especially in audio-for-video applications," says Jerry Kearby, IMS president. "Recording studios and video post-production houses are looking for routing switchers for their audio chains that are as flexible as their video switchers."

INTEGRATED MEDIA SYSTEMS

For additional information circle #156

SPANTA MODEL SLC-1 SOUND LEVEL CALIBRATOR

The Model SLC-1 is a pocket-size, battery operated unit that provides direct calibration of SPL level meters and other sound measuring systems, and is designed to fit one-inch and half-inch microphones. Calibration frequency is 1 kHz, providing the same value for A, B, C, D and linear weighting networks. The excitation SPL is 94 dB.



Calibration is effect by fitting the unit snugly over the microphone, pressing the pushbutton and adjusting the sensitivity of the sound measuring equipment until the meter indicates the correct SPL. Calibration accuracy is a quoted ±0.3 dB.

SPANTA, INC.

For additional information circle #157

MODEL EQ213 STEREO EQUALIZER FROM SCV

The Model EQ213 2/3-octave, 13-band, stereo equalizer is intended for disk mastering, broadcast, TV and motion picture post-production, control room and PA applications. The unit features minimum phase active networks that are interanlly frequency adjustable, and ultra-low-noise integrated circuitry resulting in a quoted THD and IMD of 0.01%, and a SNR of 94 dB/

Each band provides a cut and boost of 12 dB (24 dB total) via long-stroke, center-detented controls. In addition, each channel features a level control that allows adjustable gain from infinite attenuation to an overall increase in 6 dB, with unity gain at the center-detent position.

Inputs are 100 kohm electronically balanced, and outputs 600-ohm balanced.

Other features includes a ground lift switch, and a security cover to protect all controls against inadvertent disturbance. The unit's frequency response is 20 Hz to 20 kHz, 0, -1 dB with 3 dB down points at 10 kHz and 28 kHz. Center frequencies range from 63 Hz to 16 kHz with better than 2% accuracy. Maximum output level is +20 dBv at 1 kHz, 600 ohms. The unit's suggested list price is $843.

SCV INC.

For additional information circle #158

SHURE RE-INTRODUCES MODEL 520D "GREEN BULLET"

First introduced over 40 years ago, the "Green Bullett" has become the preferred microphone choice of contemporary harmonica players. In addition to fitting comfortably into the harmonica player's palms without obstructing his playing technique, the 520D's specially controlled frequency response — 100 Hz to 5 kHz — is said to be ideally suited to the harmonica; it delivers the optimum amount of distortion necessary to produce the "dirty" sound that blues, rock, and country players desire.

The 520D is a controlled magnetic, omnidirectional dual-impedance microphone. In addition to hand holding, it can be mounted on a microphone stand, boom, or gooseneck. The microphone has been completely retooled and revitalized to guarantee its original performance and dependability.

User net price of the Model 520D "Green Bullett" is $91.75.

SHURE BROTHERS INC.

For additional information circle #159

STANDARD TAPE INTRODUCES NEW SERIES OF TEST TAPES

The first new series is manufactured to international (IEC) frequency characteristics, and the second to domestic (NAB and AES) frequency characteristics. STL will continue to manufacture its existing series of NAB characteristic test tapes utilizing STL original programs.

STANDARD TAPE LABORATORY, INC.

For additional information circle #160

MODEL 525 DUAL-GATED COMPRESSOR/LIMITER FROM SYMMETRIX

Applications for the Model 525 are said to include any situation where dynamic-range controllers are presently found — that is, anywhere there is a need for automatic gain control. The manufacturer claims that the 525 will excell in sound reinforcement applications where the expander/gate circuitry will aid in feedback suppression, and in studio vocal tracking where the units' freedom from pumping and distortion will be greatly appreciated.

Two channels are featured in a 1¾-inch rack package; the unit is capable of simultaneous compress/limit and expand/gate functions. Symetrix has incorporated its proprietary fast RMS level detection circuitry, which is said to offer the best of both RMS and peak level sensing. Also featured are newly developed program-dependent attack and release circuits in both the comp/ limit and expand/gate sidechains, which provide instantaneous automatic adjustment of attack and release parameters.

By arrangement with Valley People, the unit incorporates that company's TA-104 VCA.

SYMETRIX, INC

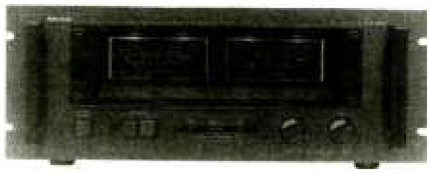
For additional information circle #161

For additional information circle #162

New Products

NIKKO AUDIO ALPHA 650 POWER AMPLIFIER

The Alpha 650, which features "Exclusive Terada Circuitry," is rated at 300 watts per channel (minimum RMS, driven into eight ohms, 20 Hz to 20 kHz, with less than 0.08% total harmonic distortion). In a strapped mono mode, using a built-in bridging line transformer (BLT) circuit, the amp is rated at 500 watts RMS at eight ohms.



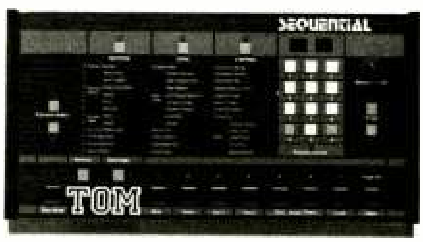
Designed for commercial applications, the 650 offers XLR inputs, RCA outputs, banana plugs, and ¼-inch mono phone jacks. The amp is said to be stable down to two ohms, and featuers a two-speed thermal controlled fan, responding to microprocessor commands.

NIKKO AUDIO

For additional information circle #166

SEQUENTIAL UNVEILS TOM DIGITAL DRUM MACHINE

A fully programmable drum machine featuring eight digitally recorded instrument sounds, the new unit provides programmable volume, tuning and stereo pan individually for each drum and cymbal sound.



Up to 99 rhythm patterns can be programmed, and each one can be from one to 99 measures long. Total memory capacity is more than 3,000 notes. Once a number of rhythm patterns have been recorded, the user can link them together (up to 99 different patterns) to create a whole song. Songs can be edited, copied, and appended in the same manner as patterns. Memory contents is retained even when power is off thanks to a backup battery.

TOM can be interfaced with almost any basic sequencer system through its multimode clock input and output. The unit also features MIDI, which enables volume and tuning to be programmed in real-time from any velocity-sensitive keyboard instrument.

SEQUENTIAL

For additional information circle #167

COMPLEMENTARY PHASE PARAMETRIC EQUALIZER FROM MEYER SOUND

The CP-10 is a 10-band stereo parametric equalizer featuring five bands of equalization per channel, with an additional high and low shelving cut filter for each channel. Front panel controls include boost or cut (±15 dB); center frequency selection (10:1 range), bandwidth control (0.1 to 1.1 octave); and individual in/out switches for each frequency band.

The Complementary Phase design is said to ensure that phase shift due to equalization is predictable and minimal. In fact, one side of the equalizer can be used to correct a curve set up in the other side, the manufacturer claims, and the net result is not only a flat response, but also a time delay of less than 1 millisecond.

Any frequency between 60Hz and 6 kHz can be controlled by two equalization circuits per channel (or four EQ circuits in mono operation); 20 Hz to 60 Hz and 6 kHz to 20 kHz are controlled by two EQ circuits and two shelving cut filters (or four and four in mono operation).

The cut and shelving high- and low-pass filters are so described because as each is turned from flat to maximum cut the turnover frequency shifts and the slope steepens, providing the user with a flexible tool for house-curve tailoring or bandwidth limiting. (Using the shelving cut filters at maximum attentuation reduces the bandwidth of the equalizer to approximately three octaves between 5 kHz and 500 Hz, with a filter slope of 6 dB per octave above and below those frequencies.)

MEYER SOUND LABORATORIES, INC.

For additional information circle #168

DELTALAB MODEL CE-1700 COMPUEFFECTRON® FROM ANALOG & DIGITAL SYSTEMS

An improved version of the CompuEFFECTRON microprocessor-controlled, real-time processor, the new Model CE-1700 adds a non-volatile memory and the capability of more than 100 user—programmable effects to the 128 pre-set programs currently available.

According to Dick DeFrietas, who will become ADS VP for professional products following the establishment of DeltaLab as a division of Analog & Digital Systems, the improved CompuEFFECTRON was developed in response to requests in the field for more programmable flexibility in digital delay effects. "With the CE-1700 NVM, the operator can create his own effecs, call up the presets when desired and sequences more than 100 programs in multiple sequences, recalling them via the front-panel touch controls or remotely with an external foot switch. For studio use, the user can override the program mix for an 'effects-only' output."

A user can program new effects through a keypad, which are retained in the non-volatile memory until erased or re-programmed. The 128 pre-set programs include "patches" for flanging, doubling, chorusing and echoing, as well as custom patches for more unusual effects.

The unit features a quoted 20 kHz bandwidth at all delays up to 1.7 seconds, and a dynamic range of up to 90 dB with less than 0.2% distortion. Delay range is from 1 to 1,723 milliseconds, and modulation is greater than 100-1 with a speed ranging from 0.05 to 10 Hz. The sampling feature will record and store up to 1,700 milliseconds, and either play it back or repeat it on command.

ANALOG & DIGITAL SYSTEMS, INC.

For additional information circle #169

DX-1 SOUND SAMPLING SYSTEM FROM DECILLIONIX

Designed as a digital sound sampling system for Apple II series computers, the DX-1 allows any sound to be entered and played back ina variety of ways, including musically over at least a five-octave range. Software is provided for sequencing sounds under program control, or the DX-1 can be played "live" on the computer keyboard in several modes.

Menu formatted for ease of use, the software includes features for triggering and external synchronization to other electronic musical instruments and devices. User-recorded sounds can be saved to diskette.

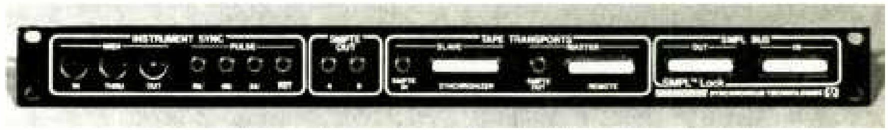
The DX-1 system, which includes a PCB, software manual, connnecting cable, and other accessories, sells for $349.

DECILLIONIX

For additional information circle #170

SYNCHRONOUS TECHNOLOGY UNVEILS VIDEO/AUDIO/MIDI CHASE-LOCK SYSTEM

The new computer-based system integrates several Audio/Video/MI functions which, in the past, have been performed by separate components. Functions include chase-lock and 10-point autolocator for audio/audio or audio/video transports; digital drum set/sequencer synchronizer; SMPTE timecode readers; timecode generator with house sync; and timecode derived metronome.



The looping mode and Automatic Punch In/Out are said to simplify overdubbing and sound editing and assembly for film or video frame rates, and drop frame or non-drop formats.

The device also allows MIDI control of digital drum sets, sequencers and digital keyboard recorders. When SMPL-Lock sends autolocate information to the tape transports, similar data is sent along the MIDI bus so that these instruments are "parked," waiting for the system to start them and keep them in sync with the tape transports at their new location.

Owners of the SMPL System can upgrade with the addition of the multi-machine interfacing panel. Total system cost is under $2,000.

SYNCHRONOUS TECHNOLOGIES

For additional information circle #171

New Products

MARSHALL ELECTRONIC ANNOUNCES AMBIENCE EFFECTS SYSTEM

The new AES-357 is said to utilize new technology to make possible true room simulation, ambience generation, stereo field generation, and related effects. The unit has a large display which shows the parameters for the 1,000 rooms and ambient environments that it is capable of generating, combined with two multi-function, high-resolution bar displays. The unit is claimed to exhibit a true 20 kHz frequency response, with resolution exceeding 16 bits.

The unit's processing technology was developed to overcome an important limitation of conventional digital systems, the company says. Other systems must time share a D-to-A converter for all output taps, severely limiting any specialized processing on any one tap. In the AES-357, each tap is converted to audio with its own circuitry; allowing specialized processing to be dedicated to each one. This facility is said to allow unprecedented control over frequency response, frequency curves, panning, amplitude and transient power responses. Further, relative and absolute phase relationships of each of these outputs is controlled to a resolution exceeding anything previously possible.

Still another new applied concept is the selection of a fully processed, amplitude, phase and frequency tailored output matrix being re-injected into another matrix not at one point, but at several chosen points simultaneously. This capability is said to allow generation of the smoother, softer reflections experienced in actual rooms.

The device also can be used alone for ambience generation, room simulation, stereo imaging manipulation and stereo synthesis; or in a special mode that will add room simulation to a current digital reverb unit.

MARSHALL ELECTRONIC

For additional information circle #175

ART UNVEILS DR2 DIGITAL REVERB

The new unit features three user presets along with front-panel KILL that mutes the reverb output; three room choices, a plate, a medium-size room and a large hall; four pre-delay settings of 0, 25, 50 and 75 milliseconds; three high-

frequency damping settings; three settings of room position (this acts like the blend on the Model 01a, controlling the mix between early reflections and later reverberation); and six settings of decay time per room. (Decay time may vary from 0.1 8 seconds, depending on the room and RT60 setting.)

In addition to the balanced mono input and stereo output facilities (as in the 01a), the DR2 also has a mono mixed output with a front-panel reverb level control, which allows it to be used in-line where normal reverb send/return adjustments are not available. The DR2 also has a level switch to allow signal level matching at lower levels. A dry kill switch is provided for the mono output, to allow external mixing to be used with the mono output.

Suggested retail price of the DR2 Digital Reverberation System is $1,095.

APPLIED RESEARCH & TECHNOLOGY INC.

For additional information circle #176

OBERHEIM ANNOUNCES "STRETCH" FOR DX DRUM MACHINE

The new add-on provides space for four rows of three drum voices each, which can be selected from the existing catalog of DX voices. The Stretch also features a MIDI Out port, to allow increased interface possibilities. Other features includes Punch In/Punch Out, Cue Tempo, Record Countdown, Drum Output Enable/Disable, Programmed Click, and Selective Loading of Songs and Sequences via an improved cassette interface.

OBERHEIM ELECTRONICS

For additional information circle #177

TANNOY NFM8 CLOSE-FIELD MONITOR

The NFM8 is a ducted-port system employing an all-new eight-inch, dual-concentric driver. The unit consists of a polyolefin copolymer LF unit and a nitrogen-blown one-inch HF dome coupled to a compression horn lens. The HF unit is further cooled by ferro fluid for high power reliability. The concentrically mounted diaphragms are precisely phase-aligned by Sync Source, which ensures total phase coherency at all angles.

The cabinet is constructed from MDF, the same high-impact material used in the company's live-performance systems. MDF was chosen to ensure a solid enclosure to prevent any interference with the low-frequency response. The enclosures are finished in a suede black epoxy; in addition, two sides of each cabinet employ acoustic isolation pads that provide a non-slip finish to all console surfaces while in the horizontal or vertical position. The pads also isolate the monitor from the console in order to eliminate all buzzing or rattling.

TANNOY NORTH AMERICA, INC.

For additional information circle #178

AUDIO PRECISION UNVEILS SYSTEM ONE AUDIO TEST EQUIPMENT

The new computerized System One is said to perform tests three to 10 times faster than the fastest equipment previously available. Nearly all common audio tests are run automatically by simply selecting the test from a menu, results being graphed while the test is made.

Comprising a computer interface card and software designed to run on an IBM PC or Compaq, the unit provides automated or semi-automated frequency response, wow and flutter, distortion and SNR measurements. Residual noise is quoted at less than 1.5 microvolts, distortion less than 0.001%, level measurement accuracy 0.1 dB, generator output level 26.6V open-circuit, (+30 dBm into 600 ohms), and 0.05 dB flatness.

The modular system can be ordered in a large number of different configurations, and modules and options can be added later by the customer. Price of a typical configuration, including high-level balanced generator and measurement modules for level, frequency, noise, and harmonic distortion is $5,225.

AUDIO PRECISION, INC.

For additional information circle #179

NEW SERIES OF SONY CASSETTE COPIERS FROM EDUCATIONAL ELECTRONICS CORPORATION

The Models CCP-110 and -112 are one-to-one and two-slave add-on versions, respectively, and duplicate audio cassettes at 16 times normal speed; they record both sides of a C-60 monaural cassette in approximately two minutes.

Features include the Sony-developed brushless and slotless (BSL) motors for capstan drive, channel-select, audio end and short tape indicator, auto rewind, and built-in erase head. As with all other Sony copiers, the units carry an industry-exclusive two-year head wear warranty.

EDUCATIONAL ELECTRONICS CORP.

For additional information circle #180

URSA MAJOR MSP-126 DIGITAL STEREO PROCESSOR

The MSP-126 perfoms a multitude of stereo processing functions, including stereo synthesis from monaural sources, precise image manipulation, ambience simulation, plus individual and cluster repeats. Applications include music recording, film and video soundtrack production, electronic music synthesis, and broadcasters that need to produce high-quality stereo source material. By utilizing PCM technology, the MSP-126 is said to deliver uncolored response over a 20 kHz bandwidth.

The unit contains eight pre-programmed modes, each of which can be adjusted with two 16-position controls. The eight modes are as follows:

- Multi-tap stereo processing, which creates a streo image from a mono source with flat response and complete mono (left, right, and left-plus-right) compatibility, adjustable from mono to full-width.
- Comb filter stereo processing, which creates stereo using comb filters, with left-plus-right compatibility, adjustable from mono to full width.
- Pan pot, which places the apparent location of a source anywhere in the stereo field using time delay; position and overall image width are adjustable.
- Binaural manipulation, which is similar to pan pot, but in the binaural mode for headphone applications, and adds front-to-back depth to the image via digital "reflections."
- Room generates the early reflections of a room or concert hall, with adjustable delay and dry/wet mix.
- Delay cluster generates a cluster of signal repeats, with adjustable pre-delay and mix.
- Repeats generates from two to 10 equally spaced repeats, alternating between channels, with adjustable overall length, and rising or falling gain.
- Scale provides a stereo comb filter whose "teeth" are at precise musical intervals, adjustable up a chromatic scale from unison to an octave plus a minor third.

Because the MSP-126 is totally software driven, new programs can be developed for it at any time.

Pro-user price for the MSP-126 Multi-Tap Stereo Processor is $2,500.

URSA MAJOR, INC.

For additional information circle #181

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1 Sony BVU 200A $4K
2 White EQ 1/3-oct. #4001 $1K pr.
2 White EQ 1/6-oct. #4303 $1.5K pr.
CONTACT: Marty Newman at (212) 687-4180

EQUIPMENT FOR SALE
10 brand-new JBL 2365 60-degree by 40-degree Constant Directivity Horns. Never used. 1/3 off list price. For information call Dana Yuricich or John Ruck at Showscan Film Corp., Marina Del Rey, Ca.
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AMPEX ATR 102 w/VSO $6.000; UREI 811's $1,400/pair; 30 channels Roland VCAs: make offer; Altec 729 Dual 1/3-octave EQ; Ampex/GRT ½-inch bin loop cassette duplicator, four slaves, all new heads: $15,000; 4 RE-20s: $175/each. Call David or Roger: (415) 328-8338

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Gotham Audio Corp. is disposing of used & demonstrator equipment. Neumann tube mike systems, fully reconditioned and guaranteed; Telefunken M15A 24/32-track and ¼-inch master recorders; VMS-80 disk mastering lathe; Telcom c4D cards; and Danner faders. Call for equipment list today: Contact Russ Hamm (212) 741-7411, 10-5 EST.

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NEOTEK SERIES III: 28×24 with 16 busses, six echo returns, custom "pre-return" submaster fader, extra patch bay, producer's desk, leg set, patch cords, spare parts, exc. cond. Also: White 32-band EQs, $550 ea. (312) 864-4460.

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Soundhound, Inc.
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(212) 575- 8664.

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It's possible now with the SCV absolute phase checker! Available for only $299, with case. Exclusively from:
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For additional information circle #184

The Department of Music Production and Engineering is now accepting applications for two positions in its department.

MUSIC PRODUCTION: Applicants must have professional experience in music production (film, TV, records, jingles, etc.) and the music business. Appropriate degrees or equivalent professional training required. Previous teaching experience is desirable but not required. Teaching responsibilities will include production courses in records, TV, film, jingles, etc. Salary and rank commensurate with qualifications.

RECORDING ENGINEER: Applicants must have extensive professional experience in the recording industry (film, records, TV, etc.), must be active recording engineers having recorded various musical styles and combinations (vocal, solo, orchestra, etc.) Appropriate degrees, theoretical background or equivalent professional training required. Previous teaching experience is highly desirable. Teaching responsibilities will include courses in mix, multitrack recording, etc. Salary and rank commensurate with qualifications.

Berklee College of Music is a private four-year institution with an educational mission of practical career preparation in the various contemporary styles of today's professional music world. The 200 or more internationally respected faculty work with 2,500 students from over 75 countries and the U.S.

Please send resume, letters of reference, and supportive background materials to:
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For additional information circle #183

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News

- *continued from page 36* . . .

multitrack because of the "unique" sound texture offered by analog.

Gatica estimates that a total of approximately 115 tracks were recorded during the project, and that prior to the final mix he had 62 simultaneous digital tracks on the pair of X-800s, leaving two open tracks for the final mix.

A video shot at the A&M vocal session is scheduled to be premiered at the forthcoming Grammy Awards. The single, "We Are The World," is scheduled for imminent release; a benefit album containing various songs donated by the artists involved is also currently under consideration.

FIRST COMPUSONICS DIGITAL SYSTEM INSTALLED AT VITELLO AND ASSOCIATES

Utilizing computer technology to digitize and manipulate audio information, the new recording and playback system enables up to 30 minutes of stereo material (or 60 minutes of mono) to be sampled at a 50 kHz sampling frequency, and stored on a 140 Mbyte hard disk. According to company president Paul Vitello, the DSP-2000 currently is being used at his North Hollywood facility to produce stereo sound effects for 125 episodes of *Voltron*, a new animated television series that will feature a stereo soundtrack.

Digitized and stored sound effects can be edited by the system, and replayed against a SMPTE timecode tracks to an accuracy of 33 milliseconds (1/30 frame); future enhancements will provide synchronization accuracy to the data-block level of 1/13th of a frame, a capability considered essential for dialog editing. Also under development is a complete digital editing and mixing system for video and film soundtrack post production.

Further details from: Vitello and Associates, (818) 505-0061.

"PRO-AUDIO UPDATE" WEEKLY ELECTRONIC NEWSLETTER NOW AVAILABLE ON IMC

R-e/p is now providing a weekly electronic newsletter on the IMC EMail Service, titled "Pro-Audio Update." Compiled by editor Mel Lambert, the new service can be accessed from any IMC command-level prompt by typing "REP."

Currently, the newsletter comprises five regular categories: Latest News Items; Equipment Deliveries; People on the Move; Convention/Exhibition Schedule; and Coming in the Next Issue. Additional pages are added as necessary; recently the service was augmented with exclusive details of the Quad-Eight/Westrex business disruption, full details of which can be found elsewhere in these News pages.

For further information about the IMC EMAIL Service, contact International Management Communications at (212) 757-0347, or (213) 937-0347. To be included in future editions of "Pro-Audio Update" message IMC822/REP-US.

OTARI ORGANIZES MTR TECHNICAL PROGRAM

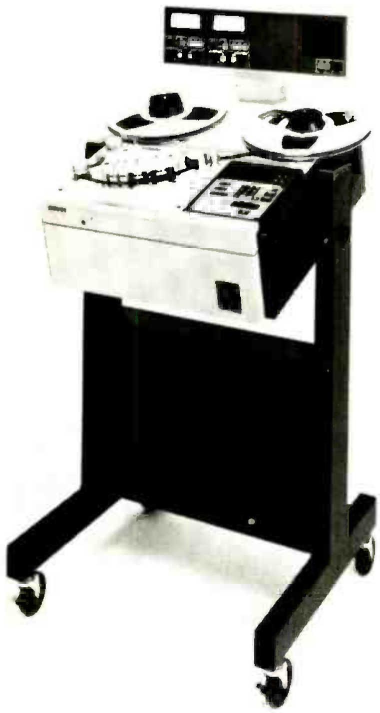
The week-long MTR User Training Technical Program will be held at the Howard Johnson Resort Lodge, North Hollywood, CA, March 3 thru 9. The seminar program comprises three days devoted to the MTR-90 Series I and II multitracks; and two days covering the MTR-10/12 two- and four-tracks.

More details from Barry Ross at Otari Corporation: (415) 592-8311.

The new Compusonics DSP-2000 digital system at Vitello and Assoc. (*L to R*) chief engineer Gary Simpson, Compusonics' John Stoutner, and Paul Vitello.

NEW ANALOG TAPE MACHINE SERIES FROM SONY

The company will unveil a new generation of MCI analog tape machines at the forthcoming AES Convention, Hamburg. The APR-5000 Series is based on the same transport featured in the PCM-3102 DASH-format digital two-track, and is available in mono/full-track formats (APR-5001), plus stereo two-track with and without center-channel timecode (APR-5002 and -5003, respectively); half-inch two-track headblocks and guides also will be available.



Separate 16-bit and 8-bit microprocessors handle tape transport logic and control, plus communications with external synchronizers and editing systems. Built-in, non-volatile memory can be used to store a variety of tape-type and EQ settings, and allows various function controls to be re-assigned under softkey designations. The transport features a ceramic capstan, constant tension, a built-in SMPTE timecode reader, and will accommodate 12-inch reels.

NEW AUDIO TEST EQUIPMENT COMPANY FORMED

Audio Precision, Inc., was formed recently in Beaverton, Oregon, by four former Tektronix engineers and managers, Bob Metzler, Dr. Richard Cabot, Bruce Hofer, and Bob Wright. While not yet ready to disclose full details of its first product, the company says that their first offering will provide "significant advancements in ease of use, measurement speed, performance levels, and cost-effectiveness over anything presently available."

By focusing purely on pro-audio markets, and using the latest high technology tools and concepts, the company expects to offer instruments of direct practical relevance to the recording and production industries.

AUDIOFORCE NOW RENTING MITSUBISHI X-800 DIGITAL MULTITRACK

The recent acquisition by Audioforce, Inc., New York, brings to three the number of rental agencies offering Mitsubishi X-800 recording systems; also included are Audio Affects, Los Angeles, and Nashville's Digital Associates.

"We always buy the equipment that people ask for, and they ask for digital all the time," says Sid Zimet, Audioforce president. When not booked on sessions, the X-800 will be housed at New York's Clinton Studios, which also owns a Mitsubishi multitrack.

The new X-800 was first installed at Chicago's Audio Media Studios in late December, with Zimet supervising the delivery and instructing the client in the machine's operation. "And that took less than an hour, which is a remarkable thing about digital — it's easy to use. To say it's reliable is frankly an understatement," he adds. "We've rented out the Mitsubishi X-80 two-track for over four years, and now have three of them in service. My experience is that with a few minutes of simple instruction, they're all set — I never get a call until the project's finished. In the equipment rental business I couldn't ask for more."

With an introductory rate that

includes the free use of an X-80 two-track, Zimet expects to keep the system busy on extended projects much of the time. Housing the deck at Clinton Studios should help, he offers.

"We couldn't be more pleased with the arrangement," explains Clinton president, Bruce Merley. "We've had a great deal of interest in 64-track ability from our commercial clients. With two Mitsubishi X-800s here we can offer greatly enhanced digital service."

UNIVERSITY OF LOWELL ADDS RECORDING COURSE

The university's College of Music has introduced a Minor in Sound Recording Technology for Electrical Engineering Majors, to complement the complete revision and upgrading of its recording degree, the Bachelor of Music: Emphasis in Sound Recording Technology. The revised curriculum for primary study in SRT includes intensive work in musicianship; sound recording technology with hands-on production experience; video production; electrical engineering; acoustics; physics; mathematics; and directed general education. The BM: SRT degree is said to prepare students for entry-level positions in all disciplines of the recording industry requiring musical abilities and critical listening.

The new Minor in SRT emphasizes repair and maintenance techniques; equipment functions and operation; aural perception of equipment functions; and research and development.

Full details from: College of Music, University of Lowell, Lowell, MA 01854. (617) 452-5000.

SMPTE OFFERING STEREO TELEVISION SEMINAR

"Stereo for Television — A Whole Different Ball Game," a one-day seminar focusing on stereo parameters for production in film and tape, jointly sponsored under the auspices of the School of Cinema-Television, University of Southern California, and the Society of Motion Picture and Television Engineers is scheduled to take place at the USC campus on Saturday, May 11.

The one-day seminar will include production and post-production considerations for film and tape, and a technical overview of system requirements. Seminar sessions, conducted by experts in their field, will also explore aspects of stereo perception, dynamic range, presentation environment, etc., as well as how to convert existing Music Videos and theatrical programs for Stereo Television.

Enrollment fee for the seminar is $45 (including luncheon); to register call (213) 743-7469.

STOP PRESS: QUAD EIGHT/ WESTREX BUSINESS DISRUPTION; BTX CORPORATION TAKEOVER BY CYPHER DIGITAL

• At press time, the following statement was received from Cam Davis, president of Quad Eight/Westrex, regarding the current status of the company's business disruption:

"Quad Eight/Westrex has temporarily shut down its United States manufacturing operations while discussions regarding its possible acquisition are pending. It is anticipated that U.S. manufacturing operations will resume next week. Operations in the UK continue as normal."

The company would neither confirm nor deny rumors that current overtures are being made by Mitsubishi for the purchase of Quad Eight/Westrex. As a result of the failure of another recent acquisition attempt, Davis says, the company is "experiencing difficulties with its primary lending institution."

• On Monday, February 11, the BTX Corporation was acquired by Cypher Digital, manufacturer of a wide range of timecode products. The BTX product line, including timecode synchronizers and controllers, will be absorbed into the Cypher manufacturing capability, according to a company spokesman.

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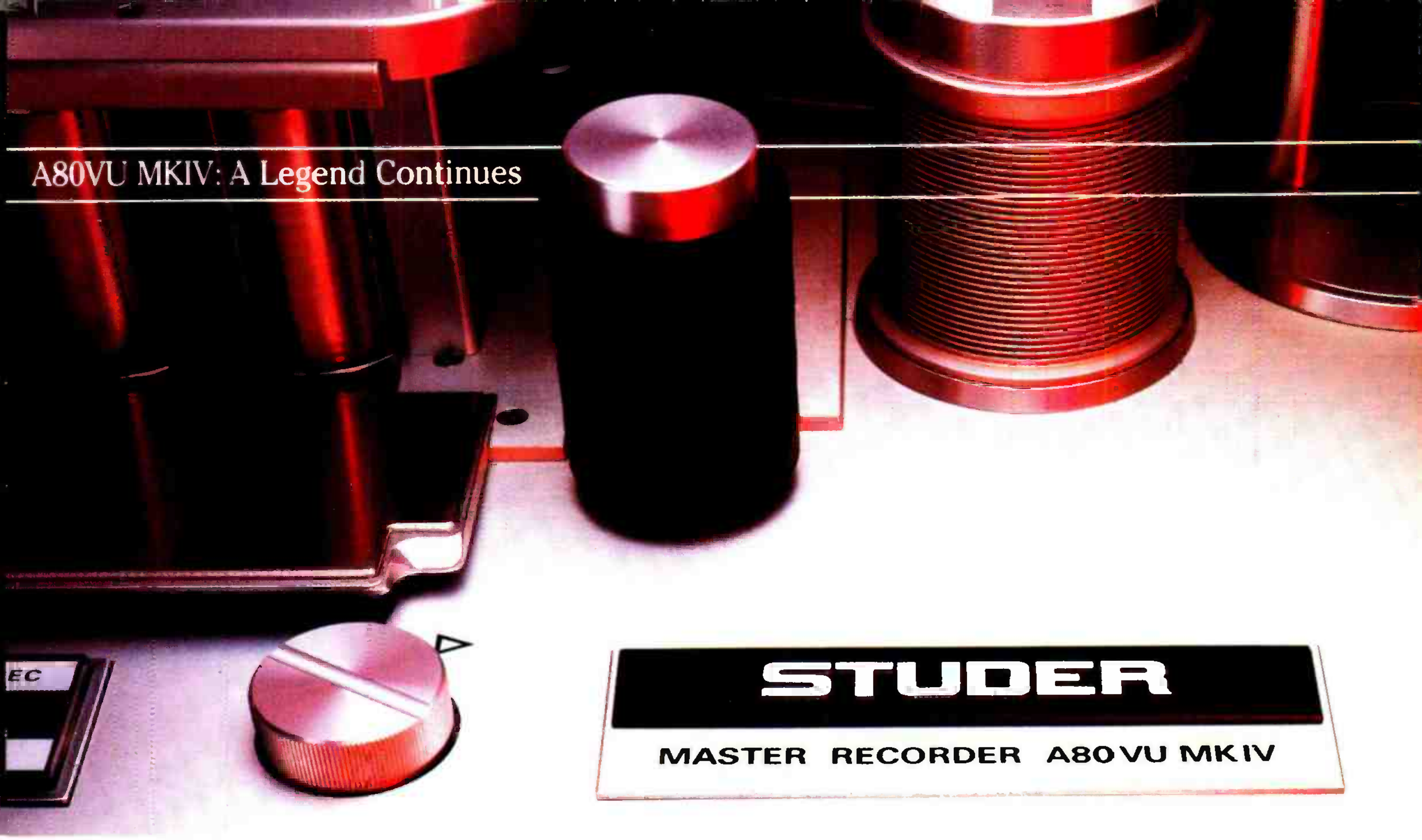

This is our newest multitrack. It is also the most affordable multitrack in Studer history.

For the fourth time since its inception, we've changed the A80VU. We've improved the sonic performance, tape handling, and durability. And we've substantially lowered the price.

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Uh-oh. Something Is Missing. Yes. The transformers are gone. They've been replaced in the input and output stages with new high performance active balancing circuitry. Other MKIV improvements include a new master bias oscillator, extended record headroom, and a new record and bias driver compatible with all present and future high-bias requirements. Record electronics are now fully compatible with Dolby HX Pro* requirements.

Smoother Shuttling. Hardier Heads. The MKIV's new tape tension control system provides smoother tape handling, while a new extended wear alloy for record and play heads greatly increases head life.

Never Lower. The list price of the A80VU MKIV 24-track is lower than any of its predecessors. And that's in straight dollar figures, without adjusting for inflation. What's more, the A80VU MKIV now has a list price lower than most of its competition.

No Hocus-Pocus. How could we make the A80VU MKIV better and lower the price at the same time? Simple. We make it in Switzerland, and you pay for it in dollars. The favorable exchange rate does the trick. That means you get advanced electronics, Swiss precision, and low price. If you act now. This can't go on forever.

Your Time Has Come. If you've always wanted a new Studer multitrack but thought you couldn't afford one, your time has finally come. Call today and find out why the A80VU MKIV is one of the most advanced recorders available at any price. And then ask about our new lower prices. Be prepared for a pleasant surprise.

For more information, call or write: Studer Revox America, 1425 Elm Hill Pike, Nashville, TN 37210; (615) 254-5651.

*Dolby HX Pro is a trademark of Dolby Laboratories.

STUDER REVOX

For additional information circle #193