JANUARY/FEBRUARY 1972

VOLUME 3 — NUMBER 1

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RECORDING engineer/producer

relating recording science • to recording art • to recording equipment

IN THIS ISSUE:
CLASSICAL RECORDING

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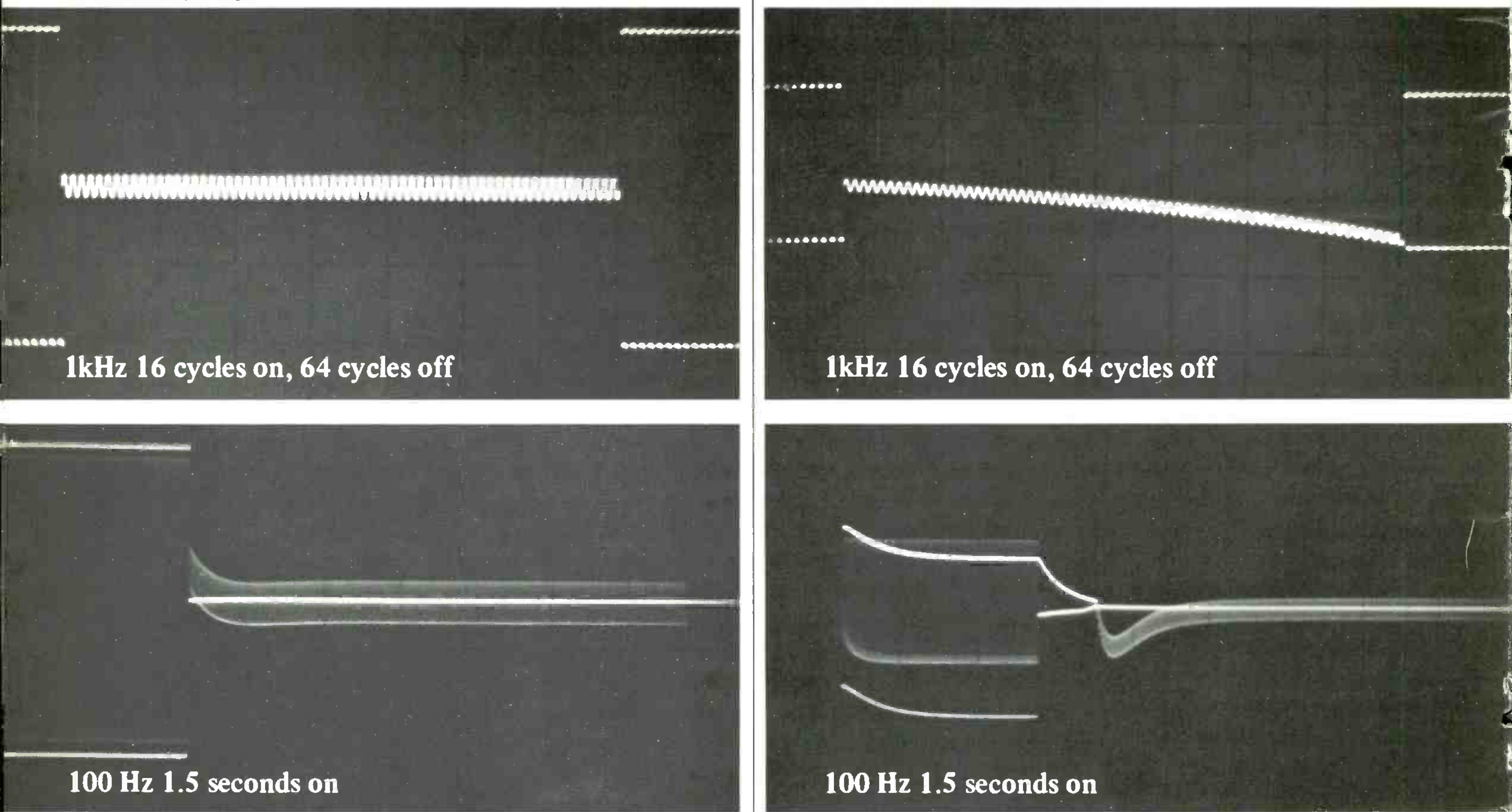

1kHz 16 cycles on, 64 cycles off (FLICKINGER 535) | 1kHz 16 cycles on, 64 cycles off (NEXT BEST COMPETITOR)

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CREDITS – cover: TRICI VENOLA
Page 24: WAYNE YENTIS

RECORDING
engineer/producer

–the magazine to exclusively serve the recording studio market . . . all those whose work involves the recording of commercially marketable sound.

–the magazine produced to relate . . . RECORDING ART to RECORDING SCIENCE . . . to RECORDING EQUIPMENT.

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LETTERS and LATE NEWS

From: Woody Langley
KTTV
Hollywood, Calif.

Although perhaps not suited for your "Fighting Murphy's Law" column, I'd like to extend a description of a weird effect which could be called preverberation — a reverb effect which occurs before the fact. Recorded material, particularly voice only, is threaded backwards around the capstan and puck (as described in the current R e/p) to run in reverse and this material is transferred to another machine. While the transfer is being made, the recording machine is placed in a playback mode and the pot is opened to give the desired amount of reverberation. When this copy is then played backwards, voila!, reverb occurs first. Application is limited only by experimentation for such things as single-word preverb. Normal tape reverb or echo can also be added to this doctored master for a truely psychedelic effect.

Being in television, I'd particularly like to see more articles on TV audio. How about an interview with Bud Lindquist of CBS who turns out beautiful sounds for the woefully inadequate home set. Also something on multiple track TV recording for remixing and relaying in sync onto the mono videotape track (how it's done, practices established by the various networks, etc.) along with conjunctive information and reviews on such systems as Ampex's Auditec. And material on miking and baffling techniques for music shows, particularly where bands are required on stage and the horrendously compounded problems of open booms and talk mics in such situations.

Again, many thanks for filling a gaping void and I'll be looking forward to each new R e/p with "unlimited" anticipation.

From the READERS

An editorial material rating of the most useful feature article, as gathered from the Reader Service Cards received prior to press time.

NOVEMBER/DECEMBER ISSUE:

ROY HALEE	29.10%
PHASE & THE SINGLE MIKE	48.21%
24 TRACK COMES TO L.A.	3.54%
SOLID STATE SWITCHING	17.73%

From: R. B. Adems
Symphonic, Inc.
Reading, Pa. 19601

A simple method of carrying cords which will become tangled and would do damage to the mike and speaker cables is to wind them on the wooden wash line boards available for less than $1.00 at most hardware stores. They have the handle built in and do not twist the cord and always leave the cords ready for use. The handle is useful for hanging and storing and you can label the boards as to cable length, type, use if for condenser, dynamic, moniter etc.

From: Daniel Kuryliak
Eastern Sound Co., Ltd.
Toronto, Canada

Keep on chugging! After reading 'Murphy's Law' in the December issue, I dug up some interesting ideas that were conceived around our studio. These could be good protection against "the slings and arrows of outrageous fortune" – Murphy's Law!

For those of you who wear out a lot of expensive test tapes and a lot of time checking playback & eq. before each session, here's a little tip – take off that knob that's constantly changing position and get the same value in a locking type from your local supplier, Ohmite make these, and it doesn't take much time to install them either. After the initial line up all you have to do is a weekly routine check.

AUDIO ENGINEERING SOCIETY TO HOLD 42ND CONVENTION AT LOS ANGELES HILTON.

The 42nd Convention of the Audio Engineering Society will be held at the Los Angeles Hilton Hotel, Los Angeles, California, May 2–5, 1972. Technical sessions on new developments in the field of audio engineering will be offered, concurrently with exhibits of professional audio equipment. Send titles and abstracts to: Leon A. Wortman, Ampex Corporation, 401 Broadway, Redwood City, California 94063. For exhibit information, contact: Jacqueline Harvey, Exhibit Manager, Audio Engineering Society, 124 East 40th Street, New York, New York 10016.

BOOK REVIEW

Acoustics of Studios and Auditoria, by V.S. Mankovsky, 395 pages, $15.00.

Dr. Mankovsky has succeeded in presenting the whole of acoustics, and sound insulation, and noise control as a neat and logical sequence, with sufficient basic theory to give the student a thorough understanding of the subject and enable him to apply his knowledge to new situations. At the same time there is a mass of design information which makes it a useful reference work.

A very attractive feature is the fact that all the essential theory is dealt with by the use of College level mathematical methods. Equally useful is the way in which the statistical, geometrical and wave-mechanical aspects of room-acoustics are treated in parallel, clearly showing the most suitable areas of application for each of these viewpoints.

Dr. Mankovsky is a lecturer in the Leningrad Institute of Cinema Technitions. The work has been edited and introduced by Dr. C.L.S. Gilford, Professor at the University of Aston, Birmingham, who was formerly adviser on acoustics to the B.B.C.

ACOUSTICS OF STUDIOS AND AUDITORIA is available from **RECORDING engineer/producer.** Send check or money order for $15.00 plus 35¢ (postage and handling) to R-e/p, Dept M, Post Office Box 2287, Hollywood, Ca. 90028.

SCHEIBER AND ELECTRO-VOICE ANNOUNCE ISSUANCE OF FOUR-CHANNEL PATENT. In a joint statement made today by Peter Scheiber of Audiodata Company and Electro-Voice, Inc., a Gulton subsidiary, it was announced that U.S. Patent Number 3,632,886 has been issued to Scheiber covering encoding and decoding matrix techniques for four-channel recording and broadcasting. As previously announced, Scheiber and Electro-Voice have agreed to pool their efforts in the protection of patents, licensing, and manufacture of equipment using developments from both firms.

Howard Durbin, E-V's senior vice president and technical director, states, "It is our belief that this patent is basic and will cover all current or announced matrixing systems. This is a tribute to the pioneering work done in this field by Peter Scheiber and our subsequent contributions. The Electro-Voice Stereo-4™ system, developed in conjunction with Leonard Feldman and Jon Fixler's Industrial Patent Development Corp., the first production matrix technique on the market, has received wide acceptance. Our association with Feldman and Fixler, of course, continues. This with our latest development: equipment capable of completely compatible decoding using any type encoded material, coupled with the patent issuance, means that the four-channel industry is on a solid basis to enjoy immediate growth and consumer confidence.

"We intend to issue licenses to assure the broadest possible base of manufacture for four-channel encoding and decoding hardware as separate components or in-house circuitry. It will continue to be our policy to include any license fees as a part of the selling price of integrated circuitry. Our basic interest is in establishing the four-channel concept as an industry with playback equipment in all price classes. The E-V Stereo-4 system has the flexibility to provide that, and to present preferred results to the recordist, broadcaster, and equipment manufacturer."

SROKA NEW AMPEX MARKETING MANAGER. A.A. Sroka has been named marketing manager of the recently formed audio/video systems division of Ampex Corporation, it was announced by Charles A. Steinberg, vice president — general manager.

Sroka, formerly general manager, professional audio products division, is responsible for domestic sales of Ampex audio and video broadcast products and closed circuit television products. These include video recording systems, television cameras, professional audio recorders and related accessories.

Ampex recently consolidated four divisions — video products, professional audio products, educational and industrial products, and Videofile information systems — into the audio/video systems division.

Sroka has named Richard Sirinsky, national sales manager; Donald V. Kleffman, manager of product marketing; and Thomas E. Scholten, manager of distribution policies.

Regional managers are Kenneth Herring, western region, located in Glendale, California; Frank Nault, south central region, located in Dallas, Texas; Al Slater, northeast region, located in Hackensack, New Jersey; Donald Smith, midwest region, located in Elk Grove Village, Illinois; and Frank Benson, southwest region, located in Atlanta, Georgia.

SHURE ANNOUNCES PUBLICATION OF NEW PROFESSIONAL PRODUCTS CATALOG. Shure Brothers Inc., Evanston, Illinois, has published a new comprehensive catalog describing the company's full line of microphone and circuitry products for broadcasting, recording, motion pictures, and professional sound reinforcement.

Included in the catalog are illustrations and technical specifications, extensive discussions of microphone types and microphone selection, and a variety of applications in which Shure Professional Products have been used.

For a free copy of the Shure Professional Products Catalog, Number AL312B, write: Shure Brothers Inc., 222 Hartrey Avenue, Evanston, Illinois 60204.

RUFKAHR, ALTEC SALES MANAGER. Robert A. "Bob" Rufkahr has joined Altec in the position of sales manager, according to an announcement by Don Davis, the company's vice president for marketing. Rufkahr comes to Altec from the Ampeg Company, Inc., manufacturers of portable sound systems and related equipment, where he served for four years as vice-president-marketing.

Rufkahr will have sales responsibility for Altec professional sound equipment and University sound products; musical sound products, telecommunications and

inter-communications equipment manufactured by the Altec Division of LTV Ling Altec, Inc.

NEVE APPOINTS TITCOMB. David Neve, general manager of Rupert Neve, Incorporated, announces the appointment of Rodney D. Titcomb as marketing manager. Mr. Titcomb, formerly with Westinghouse Electric Corporation, will be responsible for the marketing activities of both Rupert Neve, Incorporated, and Rupert Neve of Canada, Ltd. An electrical engineering graduate, he brings 13 years of marketing experience to his new assignment.

Circle No. 105

Bill Porter and friend.

"We are using nothing but Shure SM53 microphones—at my request." That's Bill Porter, master sound engineer for his *United Recording Corp. of Nevada* out in Vegas, telling us about a recent series of live concerts at which he officiated as the man in charge of sound. "I've been using Shure microphones in recording sessions for quite a few years. We used SM60's for Buddy Rich's 'Mercy, Mercy' live recording at Caesar's Palace . . . and have had numerous comments on the drum sound. We use the 545 and 546 on all live and recording sessions for guitar, drums, and piano." We're proud of the fact that with all the equipment from which he can choose, Bill calls on his "tried and true" friends from Shure.

Shure Brothers Inc.,
222 Hartrey Ave., Evanston, Illinois 60204.

SHURE

Circle No. 106

"Recording classical music requires great understanding of the composition and the conductor, and this must be understanding without interference." Carson Taylor.

"Our duty is to be able to understand to the greatest ability, objectively, what it is they (the conductor and orchestra) are doing . . . and get that technically onto tape so that it doesn't spoil it . . . doesn't detract from it . . . doesn't change it." Max Wilcox.

"The classical engineer is saddled with what is there, and out of this he must make a record." John McClure.

RECORDING CLASSICAL MUSIC

as written from tape recorded interviews
by ROBERT BUSHNELL, President
Bushnell Electronics Corp.

Here are three viewpoints. The essence of each being virtually the same in describing the externality of the classical recorder to the actual musical performance. These are the views of: Columbia Masterworks' JOHN McCLURE, an engineer turned producer; R.C.A.'s MAX WILCOX, a classical musician and conductor turned producer; and, Capitol/Angel's CARSON TAYLOR, an engineer with a classical music education.

Each man continues to underscore the importance of complete rapport with the music and the musical organization. *Taylor,* "I must understand the music to the best of my ability, the problems inherent in the score, therefore I must read the score. I must become extremely familiar with what the conductor wants, his interpretation, what it is he expects to have as the sound coming from the orchestra. Not just the emotional variations, but mainly the sound balance from the sections. Every conductor balances the orchestra differently.

McClure: "You are part of the creative team . . . not the musicianship as you may be on a pop date . . . you can't say 'hey man, let's change this, let's erase that track and add something else' . . . so the relationship to the conductor is a very important thing."

Wilcox: "You have to be very sensitive to the demands of the players who function as a unit . . . because the great orchestras are a homogeneous mass of personalities. When you have to suggest a change, something for the sake of the recording, you have to understand the affect of the change on the orchestra."

As each man continued from these first statements of professed strict allegiance to the fidelity of the performance to a discussion of their own philosophies and perogatives in recording a performance, it becomes apparent very quickly, that several key decisions must be made early in the recording process that will be disruptive of the normal concert, and even the concert rehearsal process. These changes have to be *sold,* more or less, to the conductor and to the orchestra.

The point is made by each in his own way, but as Max Wilcox put it very succinctly, "even though the program is played exactly the same, a recording and concert hall experience are two different listening experiences. At a concert the listening experience is psycho-acoustic. Each of the senses is participating to create a complete performance. The listener can experience, the microphone cannot. For example, you

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Circle No. 107

can experience, the microphone cannot. For example, you can, often, in the concert hall see parts of the orchestra playing feverishly, and you don't really hear them. Say, during a grand climax, you see the strings playing away, you know they are there, but the brass and percussion are overwhelming them. At the performance you have an optical advantage. In the recording if you don't hear them, you just don't hear them. Your mind is not filling in what's being lost. We have to strive for clarity. A lot of this depends on the hall acoustics."

So, one of the earliest decisions to be made concerns the recording environment. John McClure puts it neatly, "the room makes an enormous amount of difference. You can take an indifferent or small orchestra and put it in a really good room and it will sound first class and big. You can put a great orchestra in a bad hall and it will sound bad. For instance, Carnegie Hall is very nice for listening to concerts, but not good at all for recording. When you get up on the stage and clap, and you hear a distinct clap coming back to you, not as decay, but as slap, then you can guess you are in trouble. A change in hall may be the answer."

Carson Taylor gives us two pretty good case histories of this. "In Cleveland's Severance Hall I had the advantage of listening to the orchestra in rehearsal. I couldn't believe my ears. The woodwinds were very silvery and shimmering, they had a beautiful sound. But, the violins sounded terrible.

"Why? Because they had the stage extension up, and the orchestra was as far out on the stage into the hall as it was possible to get. I walked around the hall and I determined, at least in my own opinion, that the hall had a very bad sound, within the hall. But the stage, when I walked around it, had a perfectly magnificent sound.

"The woodwinds were being projected from within the stage, while the violins were not being projected at all from where they were being played out in the hall.

"I said, 'Maestro (Dr. George Szell), we have to move the entire orchestra as far back in the stage as we can go.' 'From there we will build out.' I explained that this would bring the whole front row of the violins inside of the stage area. Also, I asked to move the viole to the outside and the celli to the inside."

"Dr. Szell wanted to hear the difference before he would relocate the orchestra. We made a short take. He listened to the tape for about a minute and okayed the change, by saying, 'we are in business'."

"Then, in recording the Chicago Orchestra, we went looking for an alternative for Orchestra Hall, which is totally impossible (for recording). After scouting around we decided on Medina Temple. This is an old, very large, very barny, Shriners hall, but it does have a good acoustic ambient. Unlike Severance Hall, Medina has a 'thrust stage', that is it has no stage as such. So there in Medina, your sound isn't from the stage but from the auditorium. As soon as I walked into the hall I could tell there was one serious problem in the bottom end. The low frequencies needed to be built up.

"In this case we had to move the orchestra forward out into the hall, ahead of the curtain line, and ahead of the flies up above, where the various curtains are kept. The difficulty I heard in Medina was that most of the sound at the back of what would have been the normal stage area was going up and right into the flies above the rear part of the orchestra, eating up the low frequencies.

"After moving the orchestra forward we built a wall of baffles about two feet behind the horns. The baffles had to be high enough to stop a major portion of the sounds from the horns and woodwinds and the rest of the instruments back there from going back to the rear wall and returning.

"Flies and orchestra pits in multi purpose auditoriums are always a problem. We would have been able to use Chicago's Auditorium Theatre, the most acoustically gorgeous building I have ever heard, if it hadn't been for the huge concrete orchestra pit which produces a terrible slap, and the fact that it was impossible to find a suitable place to build a control room."

"As John McClure describes flies, "They act as an excellent anechoic chamber. They are deadly."

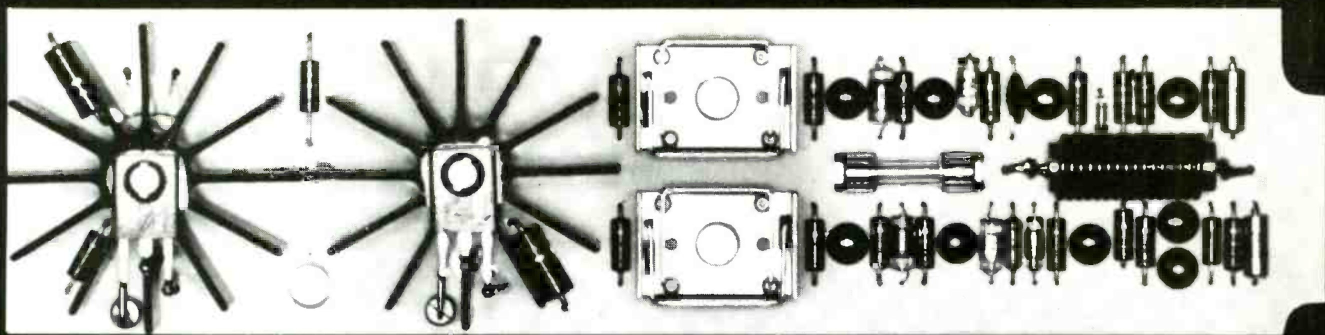
In his discussion of the difference between a good recording hall and a good listening hall, Max Wilcox describes the Academy of Music, where the Philadelphia Orchestra plays its concerts, as "a beautiful place, gives a beautiful but quite small sound with almost no reflections. It is not good for recording. We record the Philadelphia in Town Hall (Philadelphia) which has one rather peculiar acoustic property. During the early fall and the late spring it is magnificent. However, in the middle of the winter, after the heat has been on in the building for a few months, once the plaster walls are thoroughly dried out, the walls tend to pass the high frequencies, and the hall loses some of its reverberation. The low frequencies transmit better than the high frequencies do in dry air. In the early fall and late spring, you know, this area is very humid. Moisture builds up in the building, especially over the summer, and you come into the hall and you have reflections all over the place. You can mike fairly low, fairly close to the orchestra. Say, nine to ten feet up. You have a gorgeous sound, nice proximity, it's beautiful.

"During the winter we have installed commercial humidifiers and have noticed that the reverberation time is extended, but the high frequency transmission in the hall has stayed."

The selection of the hall, and to some degree, the relation of the orchestra and the music itself to the recording environment is very much the responsibility of the classical recorders . . . only a small part of their perogative, as it turns out.

There are no firm do's and don'ts as to what of modern recording technology may or may not be used in a classical recording. As John McClure puts it, "I think it is the policy of the individual producer to use equalization, limiting or any processing methods. I think that it also varies with the conductor and the kind of music.

"A conductor who is very much into the technical end of recording, there are very few, I think, that concern themselves about it. Someone like Stokowsky who is very interested in and concerned with the techniques of recording. He would definitely specify; 'I would like this reverbed,' or 'leave it like it is.' Someone like Leonard Bernstein is not technically oriented. The intermediate would not be of concern to him, as long as the record we put out sounded great.



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"That there is such a thing as pure recording that shouldn't be lumped in with the use of limiters and equalizers, which are somehow suspect as adding to or subtracting in some way from the pure effort of the orchestra, I don't agree with that.

"If you have a problem and need a limiter then use a limiter. If you need an echo chamber use it. We have used echo on some recordings made in Philharmonic Hall (New York), which is not as live as we would like it to be. The problem is that if you echo everything that is on each channel, as a unit, certain instruments project different ways from the stage and normally excite more reverberation in the hall then the instrument requires that is sitting right next to them, or in front of them, or in back of them. Once they are all locked together on a track it gets difficult, and this is where you can run into trouble . . ."

"And, yes, we use Dolby on all of our classical recordings, and a surprising percentage of our 'pop' too."

Wilcox, and Paul Goodman, his engineer on the Philadelphia Orchestra recordings agree that processing devices and techniques are important tools. Says Goodman, "Dolby is a tool, not an answer. I know Max feels as I do, it's one of the handiest tools we have. Dolby is very sensitive to level, you have to be exact. With it you don't get the masking; the tape hiss is not masking out the very small figures that are in the score and being played, and are heard in the concert hall, but are not getting on the tape, over the tape hiss. Since we started using Dolbies our settings for the lower string instruments are as much as 2 to 3 dB lower as far as the pick-up is concerned, with the same balance. The end result without the low frequency energy is that I have a louder record by about 2 or 3 dB. Low frequency energy is basically what will limit the level on a record." Wilcox: "The Dolby doesn't color sound any more than any amplifier in the board or any other piece of electrical equipment you add to the circuit. The change of sound, if there is any, is well within accepted limits. What we are really talking about is coloration. Individual loudspeakers have far more coloration than anything like the Dolby.

"I'd say, though," continues Wilcox, "that 95% of the sound on the released record is the sound just as we recorded it by setting-up right and establishing the orchestral balances according to the score and the conductor, at the sessions.

"Occasionally, we may have to reach for a particular thing, if we should have missed, should we say, a horn passage. We can then take a good selective equalizer on a good clean tape and reach for that basic frequency area, and that should pull it right out.

"We also use a little subtle equalization to return the low end to where it should be . . . a natural level. With more microphones being used all the time you have a lot of off-axis pick-up, and you get a low end build-up beyond what the orchestra itself is doing. With a little equalization we are able to return the bottom end to a normal level without disturbing the balance of the orchestra.

"At R.C.A. we are using 8 track recording, at the moment, for our classical product. Four tracks for the basic orchestra, two tracks for the helper mikes; which includes the brass on one track and percussion on another . . . strictly as helpers. Or, if it's a concerto, or a solo type of work those two tracks would be used for the artist. Then we use two tracks for ambience, useful for quad."

Carson Taylor's views on audio processing are consistent with his more purist philosophy of recording classical music. "I must say that with the exception of very minor EQ on a couple of microphones, in the case of the recordings done at Medina, because we needed to filter out low frequency rumble that is inherent in the building, we don't use any artificial coloring. I have to say that none of those records were ever equalized in the remix, they were never equalized in the cutting channels. We have never used . . . it is my firm desire to not add any artificial reverberation of any kind. It is my desire to record the orchestra as it sounded in the hall."

Regarding noise reduction, Taylor has, since the system was introduced, been using the 3M Dynatrack system. He does not use Dolby. As Taylor puts it, "the Dynatrack system produces at least the same amount of noise reduction as a Dolby, and there is no dependence on compression and expansion, which I can hear in the Dolby. In the Dynatrack system the switching between the high level track and the low level track is so instantaneous that it will follow the click of a Spanish dancer's heels. You cannot detect it."

MIX VS. RE-MIX

If the emphasis in 'pop' and 'rock' use of multi-track recording techniques is on the re-mix or mix-down; with commonly anywhere from 5 to 7 hours of re-mix for every hour of recording, the emphasis of the classical recorder, even using multi-track, is still on the *mix* in the hall.

John McClure: "The *mix* actually starts, in a way, when you read the score and decide how you are going to seat the orchestra. This isn't so easy with a new piece of music like Bernstein's MASS, with all the different things going on. I knew that down at the Kennedy Center if we tried to get a good mix in the hall with close to 200 performers, we would be able to equal the national debt in no time. So, this one we did 16 track, using the Location Recorders van. We recorded everything dry, in roughly three days. That meant that virtually all of the choices, in localization of movement, of spectrum, of balance were left to the mix-down. The MASS involved making, probably 200 times as many choices in the mix-down as you would probably make in doing an established symphonic recording. I can mix-down a 4 or 8 track, an average symphony, in a day, tops two days. The MASS took almost 5 weeks of mixing, 10 hours a day. But remember this was a new piece of music, the dimensions of which were unclear. So this was an unusual case for classical work.

"Bernstein, of course, did come in and listen to the results, and made some suggestions, things that he wanted to hear. Which we were able to do.

"Ordinarily we do our orchestra recording on 8 track so that we have more individual control. What we have is usually 6 channels of orchestra. We try to isolate the various choirs, (sections) and then have 2 or possibly 3 channels of ambience.

"I would like to do an experimental 16 track session with the Philharmonic using some established music. I know we will run into at least one very serious problem; splicing 2 inch tape. In the classical game we are used to being able to splice almost any musical attack. We won't be able to do that with 2 inch tape. For example if we have a beautiful take or movement, with just a couple of *clams* which we just can't leave in, we tend to make cover takes of these passages and then just find the note and splice that in. You just can't be sure you can do that with 2 inch tape. So there we have to be a little cautious in using 16 track. In classical we aren't able to build 16 tracks, layer after layer, with *sel-sync* and *punching-in* and all that. We have to do everything right at once or rely heavily on splicing."

The Wilcox/Goodman team emphasizes another factor of importance in establishing, as they say, "95% of the mix at the recording session. When you have worked with the same orchestra, in the same hall you get to a point where if you have run your board settings and marked them down to get a reference point for different kinds of music, you can almost get a *take* first shot, if it's okay orchestrally. That's certainly the way Mr. Ormandy likes to work.

"But generally our mixdowns are very simple. We just do a little touching up. If we want to change dynamic range we do it in the hall. We try to do most of the things in the hall. It all depends on what you are recording. You adapt your techniques to the kind of music. In a Mozart symphony you don't want it to be terribly echoey, boomy kind of thing. You will lose the violin details of the score. However in a big Strauss thing, some of his instrumental line is not meant to be heard, *tat-tat-tat,* like that. Strauss was a very elaborate orchestrator, but he certainly didn't mean that every little sixteenth note should be heard. What you are looking for in Strauss is the whole thing, an orchestral blend.

"What we try to do is to generally spread out the orchestra a little more, but still to have the orchestra feel as normal as possible within the demands of the recording. What we do is to move the brass section back a little bit. The woodwind section a little further away from the strings. About 6 feet further away. This way we can cut down on leakage, and maintain better control. For the sake of this control you can't move the orchestra too far apart. They have a need to hear themselves. Move them too far and you will have one section playing half a bar behind another.

"We don't hesitate to ask for changes if the reasons are good, and they will definitely help the recording. However you have to be sensitive to the demands of the players."

On the subject of the mix, according to Carson Taylor, "We have done many recordings going direct to two track. If you have prepared well it is no problem. We now use four tracks (from the eight track Dynatrack), which is a matter of convenience. I still basically record the orchestra on two tracks, but we are faced with solo artists on various occasions. For instance, on the Brahms Double Concerto with the soloists David Oistrakh on violin; Rostropovich on cello. George Szell and I had producers from here and England to satisfy. The 2 solo tracks gave us a chance to change the balance between the soloists themselves, and between the soloists and the orchestra. But, we recorded the orchestra only on two tracks.

"Multi-track is one convenient back-door to get out of, when you need it.

"As far as I am concerned the 'mix' that we want on a record starts when we think of the orchestra and the score in relation to the hall. The trademark of George Szell was that the first violins must be above everything and they must be driving and strong, all the time. This had to be the basis of the mix we gave Dr. Szell.

"With Guilini, he doesn't want this, he wants a very evenly balanced orchestra with accents put only in the places where he considered the composer judged they should be. I have to know these things to get the mix . . . if he is doing Brahms the fact that the second violins are just as important as the first violins, because you have an interplay between the two . . . therefore I would use a special seating arrangement which puts part of the second violins in the first section, and part of the first violins in the second section.

"Knowing the music, the mixer should not hesitate to make these suggestions. It would probably not occur to the conductor on behalf of the recording. This is part of the classical mixer's creativity."

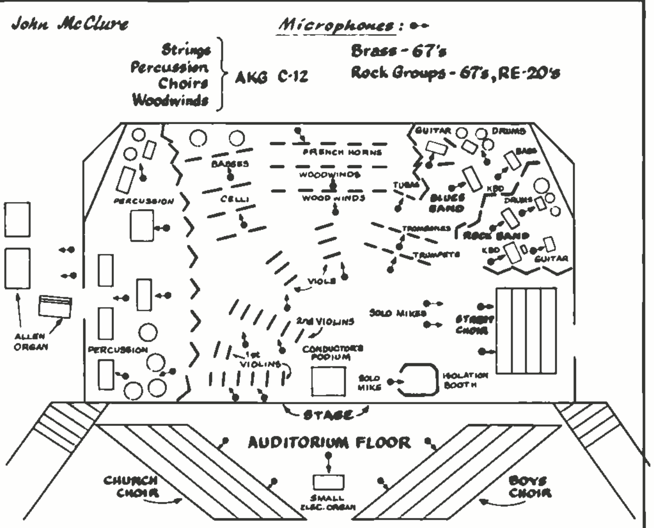
In his approach to re-seating as an aid to the mix John McClure has certain starting points. "For one thing, as many orchestras do, they have the basses strung out in a line across the back of the whole string section, along the wall. I prefer to hold the basses to a kind of section. Generally I like to keep the low strings on the right and the timpani on the left, so that you have bass weight on both sides of the orchestra. I also like to put the brass section on the right side opposite the violins. They are not playing in the same audio spectrum. This way it gives you a little more control for equalization, if you need it. They don't get in each others way. For example, if the timpani is overwhelming the violins on the left, I can get rid of some of it with low frequency attenuation.

"If we did a piece that would lend itself to Quad, like a symphonia for double orchestra or something like that, I would make sure that they were reseated so that there would be an antipodal effect. In the few pieces that are written for separate choir or off stage brass, or brass in the back of the hall, I would make sure that those voices were out somewhere away from the orchestra. Things like Mahler's Third, and the Berloiz Requiem that are scored for separate brass . . . it is a startling and delightful effect, one I wished the composers would use more, now that we have quad.

"As you can see, we do use baffles where we need them, (the interview was being done during a break in the recording of the Columbia Brass Choir and Organ at St. Georges Church, 16 Street and Irving Place in New York). I have baffles around the french horns just to deaden them a little bit. French horns are a bit of a problem. They go right through everything gobbling up the power, but they aren't being heard that well. I think it's because of the pure sine wave quality of the sound, which gets to bouncing, adding and subtracting. Not at all like a trumpet with its full harmonic structure which is heard very well. The trumpet reads very neatly on the meters, not so the french horns. Generally things that have more second and third harmonics tend to be heard more easily at lower power."

On the subject of miking, each man generously supplied a microphone set-up diagram which each felt demonstrated his basic philosophy.

McClure, "We are miking sections. From a 'pop' point of view, it is distant miking, but compared to the old mono days, and the early stereo days it's probably close miking. On the violin sections we are 8 to 10 feet above them with 2 or 3 microphones. On the woodwind section somewhat the same with 2 microphones. The only instrument we mike individually is the solo instrument, piano or solo violin in a concerto. The celli and basses we do with 2 mikes, each. The brass and percussion sections with one each.



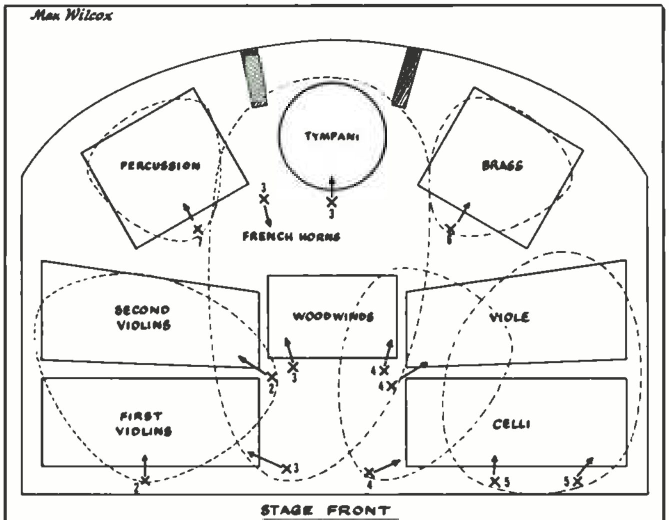
BERNSTEIN MASS - STAGE LAYOUT
Kennedy Center Concert Hall-Washington, D.C.

Track No.	Track No.	Track No.
1, 2 PERCUSSION: Timpani, Xylophone	6 BRASS: Trombones, Tuba	11 VOCAL SOLO
3 WOODWINDS: Flutes, Clarinets	7, 8 STREET CHOIR and Street Solos	12, 13 CHOIRS (Church or boys choir)
4 WOODWINDS: Oboes, Bassoons	9 STRINGS: Violins, Viole	14 ROCK BAND
5 BRASS: Trumpets, Fr. Horns	10 STRINGS: Celli, Bass	15 BLUES BAND
		16 SPARE: Organ, Etc.

"We use mostly AKG C-12's, with U-67's where we need a lot of isolation, like on the brass. I used to do quite a bit of recording in the late 50's and early 60's with a little Sony, the C-37, really great!

"There is an enormous difference in philosophy among various companies on the sound of the timpani. Some prefer a very distant, echoey, hally sound with no particular localization. I like the excitement of the noise of the skin. Bernstein is pretty much in that direction too. He loves presence in everything. The problem with timpani is the fact that it produces so much volume that it leaks. Since most halls are so much more reverberant on the low end, the timps tend to fill the hall and choke up the other channels, so, by placing an accent mike to pick-up the skin sound of the timpani, close, you can definitely locate it in the mix . . . as to its source in the mix. So we mike the timpani pretty closely."

Max Wilcox's basic miking philosophy, as he puts it, "Is to be sure that our stereo recording really is stereo, and not just two tracks of mono. Even if it's a soloist, it's two mikes, regardless of phasing problems. They are not put both in the center on pan pots in the re-mix, they are completely left and right. I get my centering at the session, because I listen that way. I position the individual, the instrument, and my microphones in such a manner that I will have a center, not a pinpointed center, but a general center.



Legend

× *Microphone* ⟶ *Direction of Emphasis*

--------- *Approximate coverage of microphone family*

Track Assignments For Microphones Numbered:	Orchestra Section Coverage	Right/Center/Left Sound Source Emphasis
1 (2 mikes) (Not Shown)	Ambient, Hall	Left/Center/Right
2 (2 mikes)	First Violins Second Violins	Left/Left Center Left/Left Center
3 (4 mikes)	First Violins Second Violins French Horns Woodwinds Timpani*	Left Center Left Center Left Center Left Center/Center Center/Left Center
4 (3 mikes)	Woodwinds Viole Celli (front)	Center/Right Center Right Center/Right Right Center
5 (2 mikes)	Celli Basses	Right Center/Right Right
6 (1 mike)	Brass	Right Center
7 (1 mike)	Percussion	Left Center
8 (2 mikes) (Not Shown)	Solo	Where Applicable

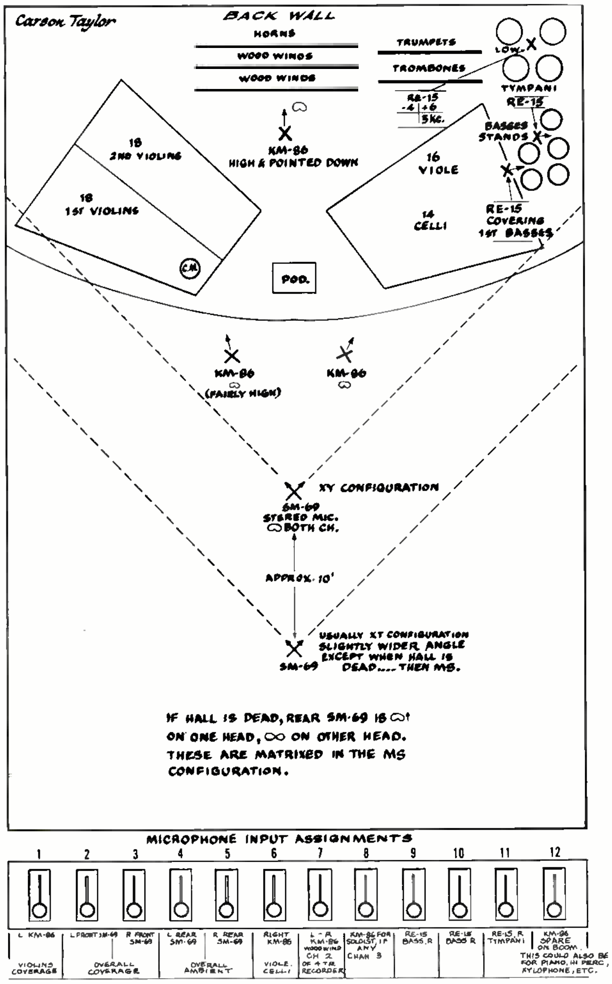
*May be split between tracks 3 and 4.

"The size of the room you are recording in affects how high or how low you are going to work microphones. In a smaller room where the sounds are coming back to you pretty quickly you work your mikes lower and, maybe, a little closer. A small room won't reinforce your orchestra in the way a large concert hall does where you can work the mikes in a high position, and work the back mikes in omni to get that big concert hall sound."

The overall effect of Wilcox's stereo miking technique, as he puts it, "is to get a *seamless* kind of pick-up pattern."

Carson Taylor begins describing his miking philosophy as, "in reality a single microphone technique. The main microphone is a stereo M/S microphone, in the diagram, the closer to the orchestra of the two SM-69's.

"This is the reference point for everything else. Now, that mike has to be positioned so that basically if you dropped off every other mike . . . and that's the first thing I do, I open only that one mike, and I have to get a satisfactory sound from that one mike, disregarding the fact that, say, I don't hear enough of the contrabassoon, or something like that. I must hear basically the over all sound of the orchestra through that one mike. From there the others are added to pick up the details, to fill in the holes until I finally get an overall balance.



"So you can see I do not mike closely. I do not mike sectionally in the orchestra. Close miking produces very objectionable mechanical sounds . . . key clicks, etc.

"With the SM-69 you always have an in-phase signal; as far as your basic signal goes it's in-phase. The mono of an S/M system is as close to the stereo as any system you can get.

"To listen to a symphony orchestra in the best possible spot you would have to have wings to fly out into the building and hang there. The spot which differs in each building is somewhere out and above the main floor audience. This is where we suspend the main SM-69 mike. In Severance Hall this will be in cardioid, it's actually working 'x-y' due to the fact that the building itself has poor acoustics; the good acoustics remember are in the stage. The rear SM-69 is in exactly the same line 15 feet behind, running 'm-s.' So, in this case we had a cardioid going straight on, and a figure eight behind it.

"Regarding sweetening microphones, I have one over the timpani, always a dynamic, something like an RE-15. Then we have the KM-86's set so that they can strengthen the weak parts, but always keeping in mind that *perspective line.* If you have an oboe solo, for example, he cannot sound larger than the orchestra. If the mike makes him appear to move forward out of his position in the section you have already used too much of the sweetener. So I only set those mikes there, at the points where they will strengthen the weak parts, but never move the instrument out of its position (perspective line). Sometimes I will use, it depends on the auditorium, a couple of bass mikes to strengthen the bottom end. Just to slightly enhance the definition, that's all.

"Just one other thing about miking, we always keep one extra mike mounted on a boom on a long cable. This is another back door you have left for yourself to get out of. Everything you plan may not work out just exactly the way you figured. You have to be ready with alternate plans. I always like to have two plans ready. You have to leave yourself a back door.

"You have to know what you are hearing," Taylor continues: "The control room (monitoring environment) is the biggest headache when you are moving from hall to hall. I monitor on JBL S-7 monitor speakers. When we set up our control room I spend quite a bit of time listening to my own personal reference tape, a compilation of several excerpts of things that I have done, that I am used to. Things that I know very well. So that when I play them back in the new control room environment I have a pretty good idea of its efficiencies and deficiencies. That's strictly for getting the idea of what the monitor itself in that location is going to sound like. I don't know how you could do it otherwise."

Interestingly, Wilcox uses acoustically suspended monitor speakers, KLH Model 5's. "I am basically a musician, so when I hear a piano that I know sounds like a piano, or a violin that sounds like a violin on these speakers that's what I am looking for. I chose these speakers as musical instruments. They are about as flat as they can be, for their size. If the strings don't sound bright on these speakers, it's cause they aren't bright. It's not because the loudspeaker is pre-equalized, as those are that some of the 'pop' people use.

"The balances between an acoustically suspended speaker and a free cone speaker are vastly different. There isn't as much presence pushing out in the mid-range in an acoustically suspended speaker, as there is in a free cone speaker. On these we aren't kidding ourselves at all."

These are three professionals, men among perhaps only another handful in the world, whose recording work is not geared to a relatively short period on the 'charts.' These are men who stamp their professional beings on 12 inch vinyl, to endure, to be acknowledged, to be criticized, and to be amortized over the decades that define anything classical.

How fitting to end with the quote: "We aren't kidding ourselves at all."

. . . END

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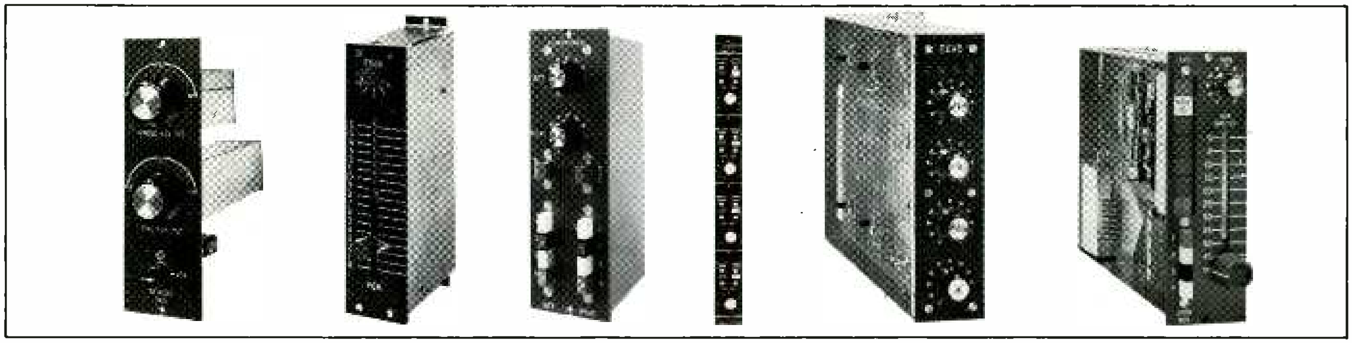

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- **Echo Facilities permit selection of internal or external reverb devices and a bright or soft timbre.**
- **Color-coded knobs enable fast and easy matching of input channels with correct output selector modules.**
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"We're fortunate in having our relationship with our clients. They hire us because they want us to be innovative. They give us time to experiment. The only clients we work with now use Moog. We don't like to take on a client who isn't going to be involved in electronic music."

The man speaking is Robert Margouleff, who with his partner Malcolm Cecil, is precipitating the transformation of modern music, as the synthesizer comes of age.

Stevie Wonder, a long time resident of upper crust of popular "soul" singers, is the focal point for this transformation.

Formerly accompanyied by the traditional Motown complement of horns, strings, and drums, Wonder has involved himself completely in synthesized accompaniment, returning to traditional instruments only for drum and occasional full grand piano parts.

Margouleff, Cecil and Wonder are unabashedly attempting the creation of a mutant musical form:

Bob Margouleff began his entrance into the professional music world as a singer, working with Leonard Bernstein, Aaron Copeland, Eugene Ormandy, and the Boston Symphony Orchestra.

Malcolm Cecil began as Principal Bassist with the BBC orchestra, and has played with Stan Getz, Roland Kirk, Ginger Baker, and Johnny McLoughlin.

They met two years ago, and found an immediate rapport. Their partnership led quickly to an album, "Zero Time," a critically acclaimed assault on the limits of contemporary electronic music, including within its scope a synthesized vocal intoning distinguishable words and phrases.

Between them they own one of the world's largest Moog systems, an expanded Series III synthesizer.

They are both competent engineers, using their electronics knowledge to expand and aid their employment of the Moog.

When they met Stevie Wonder, their own album had already been released. Their orientation, though unquestionably melodic, had been away from the imitation of other instruments. As Margouleff put it:

"Most people who are into purely electronic music seem to be doing imitative playing. There's nothing wrong with it, if you want to go that way. "Switched On Bach," for example, is a brilliant record, but it's very much imitative playing . . . . what I call a "past oriented" trip. I think the advantage of a synthesizer in creating electronic music is in discovering the sounds which the machine is capable of doing on its own. The idea is not to be imitative but original."

However, with the entrance of Stevie Wonder, things changed a bit. The synthesizer was now to serve a dual purpose: the creation of its own sounds, and the imitation of the horn and string sections so familiar to the "soul" audience.

But this slight shift in approach was not at all a compromise. After all, this was no longer pure electronic music, justified in its independence of other instruments. Rather, this was a mutation that was taking place. The horn sounds were still there, but they had an unfamiliar brilliance. Other instruments too, were mimicked, but all with a slight edge, an indefinable difference that only a synthesizer could have created.

There were other sounds too, sounds that had never been heard before, sounds that only a synthesizer could produce.

Wonder wrote the songs and the arrangements. He played everything himself, everything being the synthesizer in a hundred different formats, with a hundred different voices. He overdubbed drums and grand piano.

"Music Of My Mind" is the end result, on TAMLA/MOTOWN.

The album is an exceptional engineering acheivement not only for its synthesizer, but for many other reasons as well:

In the course of the album's production a highly effective echo noise reduction system was developed, which in addition

Cecil — Wonder — Margouleff

ELECTRONIC SOUL BY GEORGE KOCH

Circle No. 110

to its noise supression capabilities, was found to produce a variable delay in the live chamber employed. As this is of more than passing interest to any engineer or producer who has battled the incessant hiss of a chamber, it will be described in some detail:

1) Echo send buss "OUT" (see illustration) is interrupted, and a D.B.X. in RECORD mode is inserted.

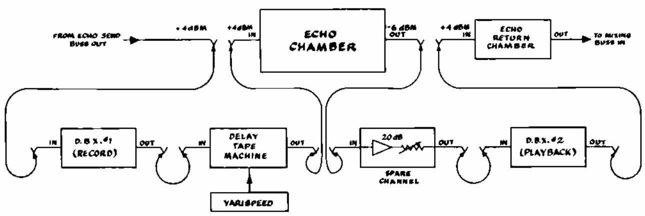
2) The output of this D.B.X. enters the input of a tape machine and v.s.o. configured for tape delay.

3) The output of this delay unit enters the input of the live chamber.

4) The chamber output enters either a spare board input, or any suitable amplifier with 10-20dB of gain. The gain of this amplifier controls the effective reverberation time of the chamber, since it is inside of the D.B.X. loop, and the decoding of the D.B.X. signal is gain-sensitive.

5) The output of this amplifier enters the input of a D.B.X. in PLAYBACK.

6) The output of the D.B.X. is brought to a chamber return with normal facilities for E.Q., buss selection, and the like.



ECHO NOISE REDUCTION SYSTEM PATCH

NOTE: BY VARYING THE GAIN ON THE SPARE CHANNEL IN THE RETURN LINE, EFFECTIVE CONTROL OF THE CHAMBER REVERBERATION TIME CAN BE OBTAINED - LOW GAIN → SHORT REVERBERATION, HIGH GAIN → LONG REVERBERATION. THUS LIVE CHAMBER CHARACTERISTICS CAN BE ALTERED CONSIDERABLY, MUCH AS THE REVERBERATION TIME OF AN E.M.T. PLATE CAN BE ALTERED.

The D.B.X. utilized in this system is the noise reduction system. Due to a peculiarity in its decoding mode, it produces an effective variable delay on the chamber employed. The higher the gain of the amplifier following the chamber, the lower is the level that will operate the D.B.X. decode section, thus changing the reverberation time by apparently making it longer. Natural chamber time will be retained when the level at the output of the D.B.X. in RECORD mode equals the input of the D.B.X. in PLAYBACK mode.

The function of the tape delay prior to the chamber is to permit proper rhythm to exist in the echo process, by varying the delay of the signal to be processed until it is in time with the music. The total effect is a quiet, variable delay, in-tempo echo.

Margouleff and Cecil's engineering approach is comprehensive. As Bob explains, "When we record we supervize the process from its inception right through to the mastering. Very few engineers seem to do that. When an album is finished in a studio, it is mastered somewhere else by someone else. The recording engineer doesn't know anything about it, and he goes on to his next project. In the end, the product suffers. Our responsibility ends with the completion of the master, not before then."

And the process of recording, what does it involve? Cecil explains:

"The way we record, and we joke about this, we call *8 track stereo.* We record everything in stereo, so when we mix, there are not several narrow sounds. We have spread. We can close the spread up, or broaden it, and position it anywhere we want."

Margouleff continues the thought, "The stereo matrix is something many seem to take for granted. They put one thing on the left, one on the right, and so on."

"A board should be played like an instrument. The array where you place the instruments is very important, not only in terms of balance, but also in terms of getting the different sonorities separate. If you intend to use a track, it should have meaningful information on it. If it has meaningful information on it, it should be where you can hear it. It's part of the musical context, and therefore has to be heard. During the mix I don't want to have to pull it down to where it just barely becomes audible. If it's just barely audible, it's noise. This is where a careful separation of sonorities becomes critical."

"Further, how things work together in the final stereo matte is dependent on how the instruments are played and miked. The Moog comes directly into the board at 600 ohms, through a one-to-one transformer. The different tracks created by the Moog are highly controllable in terms of their sonority, since each sound is created completely. But when you deal with standard instruments, that control is lost, and must be accounted for by careful mike choice and/or equalization. For instance, for recording a piano on a soft gentle piece, you might use a U-87 and a U-47, giving you a large dynamic range. But if the pianist is going to be playing hard, and there are a lot of other tracks going into the cut, it isn't going to come through. Instead, you might use an RE-20 or an RE-15, for a brighter sound."

Bob Margouleff pauses and looks to Malcolm Cecil for his nodded agreement. It is an unnecessary courtesy. At the board they function like one body and mind with four hands. Away from the board the eerie unity continues, the only disparity being Cecil's english accent. Cecil continues:

"There are many factors involved in getting a clean separation of tracks in the final mix. Stevie's voice, for instance, presented some difficulties. It is highly trained, but for the stage, not the studio. He has a tremendous dynamic range. The microphone we've found that works best, that stands up to his dynamic range and yet gives a clear, listenable sound, is a KM86i, with almost no equalization. On other occasions however, we have used an RE-20 and an RE-15. It all depends of course, on the song and the particular sound we're looking for."

"Incidentally, we avoid pads on microphones. We've found that they unquestionably change the sound. They change the damping factor of the microphone. We'd rather use variable gain on the amplifiers."

These are some of the details involved in the process of creating ELECTRONIC SOUL. A new music is in the making. The synthesizer is coming of age, and it is not where it was expected to be, in the rarefied air of avant-garde and high brow music forms. It is down deep in the nerve endings of man, in the music that makes him dance.

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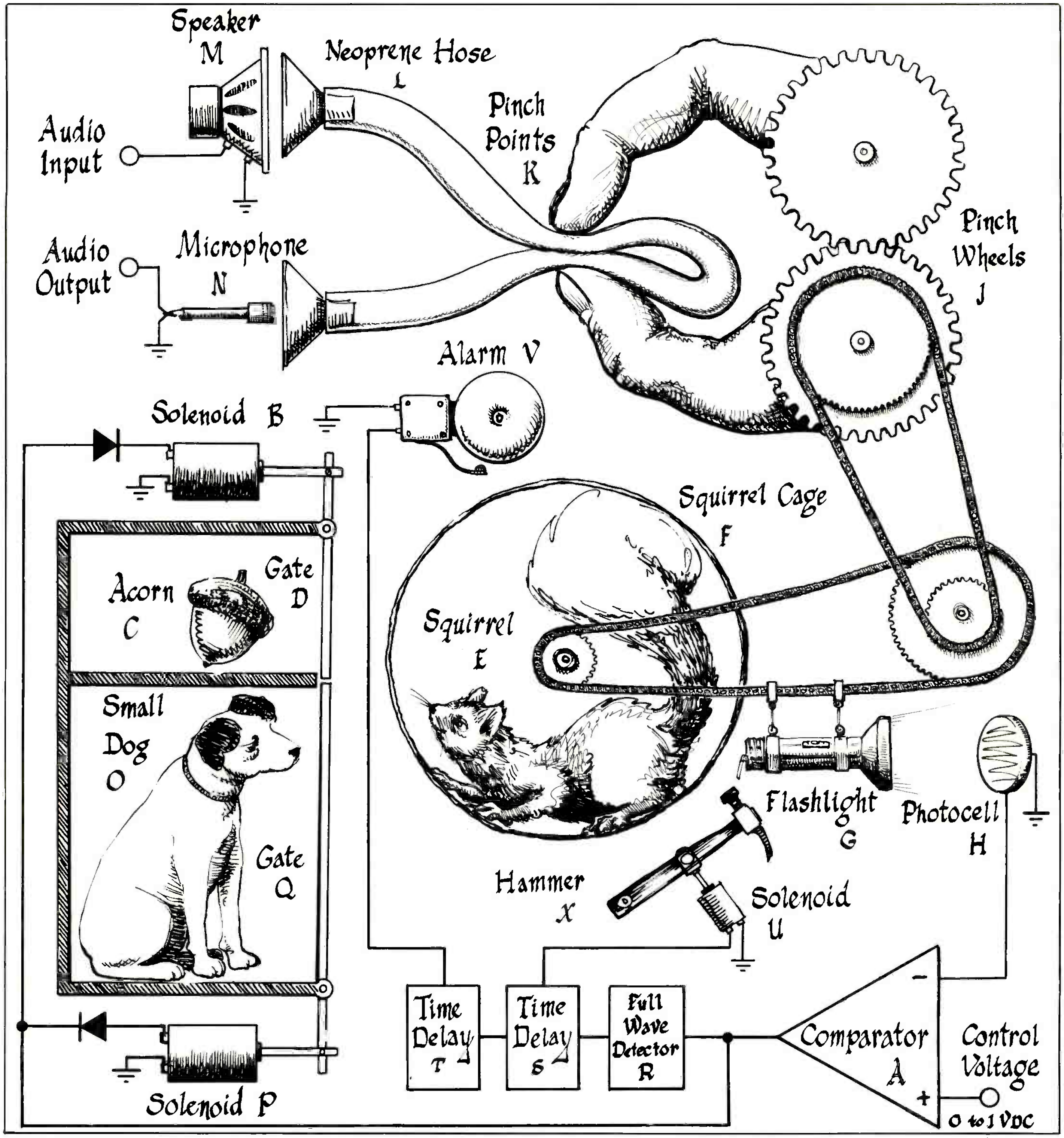


THE DESIGN OF AN "RP" TYPE VOLTAGE CONTROLLED AMPLIFIER

The above illustration depicts a typical "RP" (Rodent Powered) precision voltage controlled amplifier (VCA). The design of the device is such that a DC control voltage may be used to accurately control the gain (or loss) of the audio portion of the circuitry. In this design, the desired transfer function (volts vs gain) is accomplished with a high degree of isolation between the control and audio circuits.

If quality ball bearings are employed on the rotating mechanisms, years of reliable usage may be expected, with only routine feeding and cleaning of droppings. It should be noted that feeding should be done **only** when use of the device is not anticipated for several hours. Otherwise, sluggish performance may result.

THEORY OF OPERATION

Assume that a control voltage of +.5VDC is applied to the positive input of comparator (A). Further assume that photocell (H) is generating a DC voltage of .2VDC which is applied to the negative input of comparator (A). The output of comparator (A) will now go positive, closing solenoid (B), opening gate (D) and exposing acorn (C).

Squirrel (E) will attempt to reach acorn (C), thereby rotating squirrel cage (F) in a counterclockwise direction, bringing flashlight (G) toward photocell (H). This "forward squirrel motion" will also cause pinch wheels (J) to be rotated in the direction which widens pinch points (K) and allows more sound to pass from speaker (M) through neoprene hose (L) and into microphone (N).

As flashlight (G) moves toward photocell (H) the voltage produced by photocell (H) and applied to the negative input of comparator (A) will, of course, rise! When this voltage equals the .5VDC present at the control input, the output of comparator (A) will drop to zero, thereby opening solenoid (B) closing gate (D) hiding acorn (C) and stopping the forward motion of squirrel (E).

Now, assume that the applied control voltage is reduced to .1VDC (in an attempt to reduce the audio gain of the device) the output of comparator (A) will go negative, closing solenoid (P) opening gate (Q) and exposing small dog (O). You guessed it! Squirrel (E) will turn and run the other way, rotating squirrel cage (F) in a clockwise direction. This "reverse squirrel motion" will cause flashlight (G) to move away from photocell (H) and will cause pinch wheels (J) to rotate, thereby closing pinch points (K) and causing an increased resistance to the flow of sound from speaker (M) to microphone (N) through neoprene hose (L).

Again, when the squirrel motion causes the voltage produced by photocell (H) to equal the applied control voltage, the output of comparator (A) will go to zero, causing gates (D) and (Q) to be closed. If squirrel (E) should fail to stop at this time, comparator (A) will cause solenoid (B) or (P) to activate either gate (D) or gate (Q) (whichever one opposes the squirrels unauthorized acitivty), thereby correcting the error.

A certain amount of ambient squirrel activity or "hunting" may be expected, but is easily reduced to livable proportions by proper selection of squirrel (E).

FAIL SAFE PRECAUTIONS

It is possible, particularly when the device is first used in the morning hours, that squirrel (E) may be asleep and may not respond to acorn (C) or small dog (O). Precautions have been taken to avert a malfunction should this condition occur.

The output of comparator (A) is applied to full wave detector (R) and then to time delays (S) and (T). Should the output of comparator (A) remain either positive or negative for over five seconds (indicating lack of squirrel activity), time delay (S) will cause solenoid (U) to close thereby causing hammer (X) to sharply strike squirrel cage (F). Should this fail to restore squirrel activity within an additional 5 seconds, time delay (T) will operate alarm (V) which will sound loudly until squirrel (E) awakens.

END RESULT

As can readily be seen, by the above discussion, we have evolved a system by which the gain (or loss) of the audio portion of the circuitry may be made to vary in direct proportion to the applied DC control voltage. By proper adjustment of the mechanical mechanisms, a highly accurate, linear relationship of control volts vs audio gain may be realized.

VCA=VOLTAGE CONTROLLED AMPLIFIERS BY PAUL BUFF

President, Allison Research

WHAT ARE VCA'S AND WHAT DO I NEED THEM FOR?

If you read my preceding article "The Design of an R.P. Type VCA," you know that a VCA is an amplifier whose gain (or loss) is controlled by an externally applied DC control voltage. Such a device, while simple in configuration, can be an extremely valuable tool for extending the usefulness and adaptability of audio installations. The Allison Research VCA-1PC will be used as a basis for the discussions to follow.

VCA'S IN USE

First, let's take a look at what a recording console really is. Primarily it is a very large mound of amplifiers, switches, wires and gain controls. Gain controls for controlling echo, gain controls for controlling gain, gain controls for controlling equalization, panning, amount of compression, etc., etc., etc. In short, essentially every control board operation performed by the engineer/producer consists of switching the electrical signal, through the proper wires to the proper gain controls.

Okay, in the first place, why do we use an electrical version of the sound we are processing? How about hydraulic microphones feeding little pipes with valves instead of volume controls? (Say, that gives me an idea for my next article!)

Now seriously, we use electricity because of its extreme speed, accuracy, dependability and adaptability. Wires are used because they are what carry electrons around best. We use amplifiers to produce the needed electrical signal levels, and switches and gain controls to interface man with machine.

Let's look at the three basic catagories of switches.

1. Human operated, mechanical switches (the kind with buttons and levers and things)
2. Electrically operated, mechanical switches (relays and other things that go click all by themselves)
3. Voltage controlled switches (the kind that perform millions of operations per second and make digital computers possible)

The voltage controlled amplifier is essentially the counterpart of the voltage controlled switch. The mechanical gain control is necessary (as is the mechanical switch) in order to interface man to machine, but many many gain control operations are much better performed electronically than mechanically.

ELECTRONICALLY GANGED FADERS

Take a real life example: A friend of mine had occasion to need to control the gain of eight isolated audio channels simultaneously. His solution was simple. He fitted sprockets to eight potentiometers, mounted them all in a row and placed a chain around them to link them together. It worked of course, but a three inch knob was required to control the monster. Naturally, all the expected difficulties, such as poor tracking, large size and total lack of adaptability were present. (For instance, how do you add a channel or two?)

Take a look at Fig. 1 for the voltage controlled version. Here are its advantages:

1. No mechanical linkages to wear out
2. $\pm$ 1dB tracking over a 60dB range
3. Small size, ease of mounting, low operating torque etc.
4. Extreme adaptability because:
   (a) Unlimited additional channels may be added at will
   (b) VCA's may be mounted at a distance from gain control without long audio lines.
5. Noise free operation (potentiometer noise can be filtered from control signal with a simple RC network)
6. Drastically reduced mechanical failure possibility

THE SUB-MASTER MATRIX

Now, let's examine a slightly more complex situation. Say you are given sixteen isolated audio channels and eight "sliders." Further assume that it is desirable to be able to control any combination of isolated audio channels by any combination of sliders. Using conventional mechanical gain controls, a total of 128 potentiometers will be needed (Eight sliders with sixteen gangs each.) Also, you will need 256 shielded audio cables and a rather complex switching matrix. Since the potentiometers carry audio signal, each one will have to be a high quality, low noise element. Total cost? Around $7,500.00. Size (assuming you can buy 16 gang sliders)? About the size of eight shoe boxes. Adaptability? Forget it,

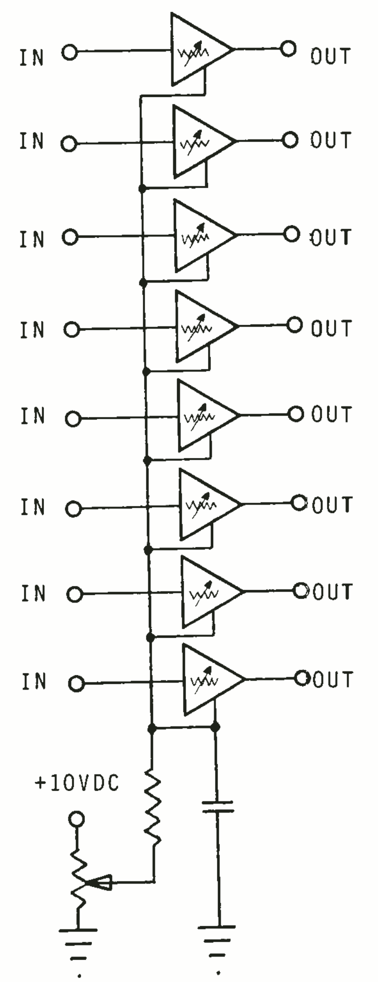


FIGURE 1
AN EIGHT CHANNEL VOLTAGE CONTROLLED ATTENUATOR

you bought it, now it's yours! Ever wonder why elaborate "grouping" sub-master systems are not available on today's control consoles?

Take a look at Fig. 2 for a 16 by 8 "grouping matrix" using VCA's. Here, only 8 potentiometers (of lesser quality) are needed, in addition to 16 VCA's. The gain controlling elements may be located at each audio source. (Scratch 256 shielded cables!) All interconnections may be unshielded bussing wires. Total Cost? Less than $800.00. Size? Total size equals about one half a shoe box. Adaptability? Sure, what do you want? Just add a VCA for each new audio channel, or a slider for each new control channel. You say you want an overall master for the whole mess? Just vary the 10 volt source and you've got it. Tracking? That's automatic, since each VCA-1 tracks any other VCA-1 within ±1dB. Reliability? Well, let's see. Eight potentiometers and no mechanical linkages, as opposed to 128 potentiometers and a plumber's nightmare. Suddenly, it becomes quite possible to consider the use of highly flexible "grouping" sub-master systems.

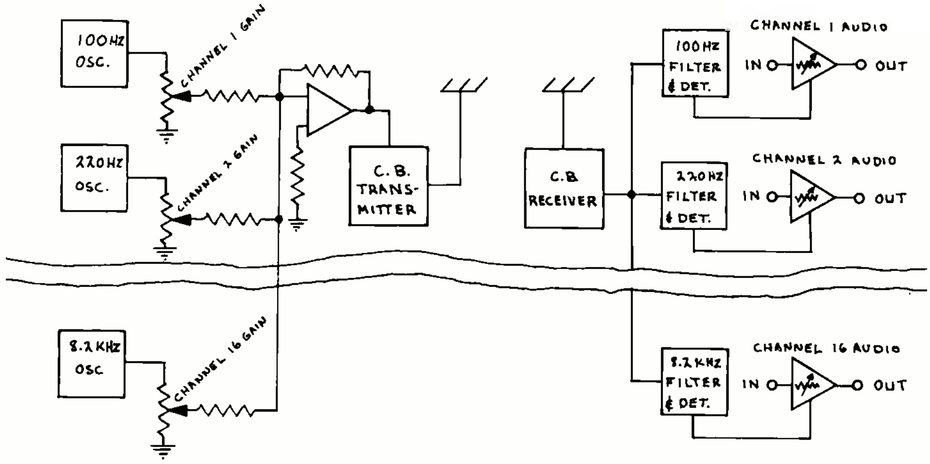


FIGURE 3
A REMOTE RADIO CONTROLLED MIXING CONSOLE

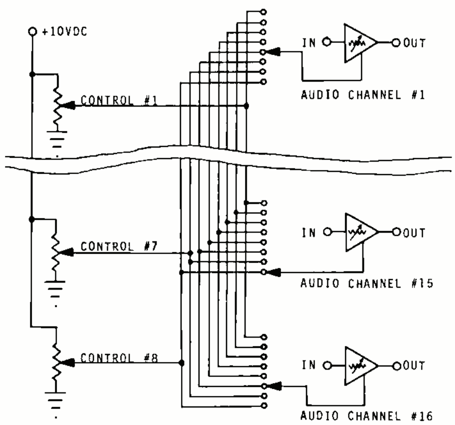


FIGURE 2
A 16 CHANNEL BY 8 SUB-MASTER "GROUPING MATRIX"

REMOTE CONTROL CAPABILITIES

One of the simplest uses a VCA might perform would be in remote control applications. Since the control circuitry carries only a DC signal, long connecting lines may be used without any degradation of the audio signal. Certain applications may suggest the use of remote control systems which do not require connecting lines at all. This could be done quite simply with a radio control system such as diagrammed in Fig. 3. The implications of such a system should be of acute interest to those engaged in the sound reinforcement field. The mixing engineer could simply take up residence in the audience, with his ultra miniature mixing console, and go about his duties without cumbersome cables or complicated set up. On stage the microphones can go direct to the remote controlled receiver and then on to the speakers. The same scheme might be used, with equally dramatic results, for remote control of stage lighting, etc.

COMPUTER MEMORY MIXDOWN SYSTEMS

Now, here is where it is really at. In my book, the interface of mixing engineer to modern mixdown console has just about reached its limit. Consider a fairly standard 16 track to quad mixdown. You're given 16 input modules, each with the following variable controls.

1. Gain. 2. Echo send. 3. Left/right panning. 4. Front/rear panning. You are further given four output modules with gain and echo return controls. Additionally, you have master gain, master echo sends and returns, a myriad of switching and a full compliment of equalization controls. In front of this conglomeration, you seat one homosapien with two appendages of suitable configuration for operating the afore mentioned controls. You ask him to make a mental judgment of which adjustments to perform and then to perform them accurately. Then you ask him to remember and repeat certain adjustments, while changing others. You speak to him in vague ways and expect him to interpret and act with precision. My question is this. How much accuracy can you expect of this person? 100%? Absolutely not! What happens when several music components require the use of his mind and his hands at the same instant? I think we all know what happens. Compromise. We settle for the best we can do.

That's where computer memory mixdown comes in. It does not make any decisions, but simply gives the engineer an infallible memory and as many additional "hands" as he needs. Fig. 4 shows a block diagram of one typical channel in such a system. It should be noted that one VCA channel will be used for each mixing function which is to be automated, be it gain, echo or panning. (This article will not concern itself with automated E.Q. or switching.)

A typical automated mixdown might go something like this. First, all "manual/automatic" switches would be set in manual position. The engineer/producer would then go about the initial "rough mix" in routine real time fashion. While this mix is being performed, the subsequent control voltage, which is applied to each VCA, is converted to digital form and is recorded on a spare track of the master tape as a dynamic record of mixing activity. Now, if the switches are placed in "automatic" position and the tape replayed, all mixing activity will be redone, automatically, as directed by the previously recorded information track. Should the engineer/producer now decide that some of the mixing functions were good and some needed to be redone, they would proceed as follows: on the acceptable functions, the switches would be left on "automatic," and on the unacceptable functions, they would be placed on "manual." The engineer would now perform new mixing activity on the manual channels, while leaving what he had previously done on the "automatic" functions.

The combination of old and new mixing information would now be assigned to a second spare track of the tape. Again, the engineer/producer can place all switches on "automatic" and make an assessment of the results. Should more changes be necessary, they may be accomplished in similar fashion, this time moving the mixing information back to the first spare track. The mixing information can be "ping-ponged" back and forth between two spare tracks, as many times as is necessary to complete the mix to everyone's satisfaction. Once the mix is complete, the mixing information becomes a permanent part of the master tape. Obviously, this sort of automated mixing cannot be performed without suitable voltage controlled amplifiers.

SUMMARY

I have attempted to show some of the uses which voltage controlled amplifiers might perform in professional audio applications. There are others, too numerous to go into at this time, but one thing is for certain, VCA's will play an important part in the design of tomorrow's audio equipment.

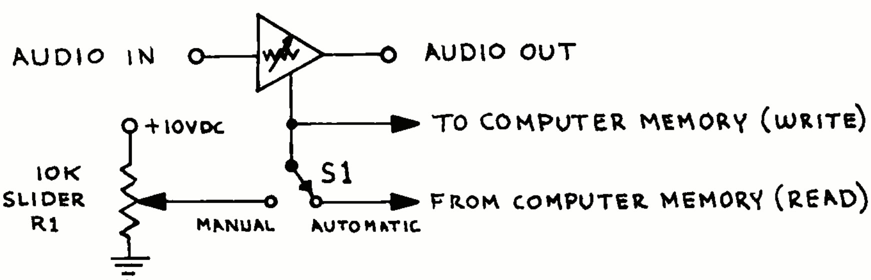


FIGURE 4
TYPICAL COMPUTER MEMORY MIXDOWN CHANNEL

16-track means Dolby System

There isn't any other way.
If you want to produce good, low-noise stereo releases from 16-track master tapes, the Dolby System is the only way to get them without changing the sound. Even if you start with the best tape. That's it.

Figure it out: every time two tracks are mixed together, directly or indirectly, their noise adds – cutting 3 dB from the signal-to-noise ratio of the original tracks.

Mixing sixteen tracks down to two will lose at least 9 dB this way – much more than can be recovered by the use of any special tape or speed. Since the signal-to-noise ratio depends upon the area of tape used per unit time, using the Dolby System is like running your tape at 12 feet per second! Or, if you prefer wide tapes instead of high speeds, you can use tape 20 inches wide! Either way, the Dolby System is by far the better deal.

In fact, the Dolby System will give you 10 dB of noise reduction, plus the same reduction of hum, crosstalk or any other sound that wasn't on line-in.

Putting it all together.
Sixteen Dolby units stack in just 28 inches of rack space. In a pinch, they can be stacked anywhere that a pair of bookshelf speakers will fit, because total volume is just over three cubic feet. The regular lines to and from the recorder will go to the Dolby units instead; XLR connectors for the cables are supplied with the noise reduction units.

Lining up the system is just like checking line level in and out of your recorder. Put a standard NAB or DIN test tape on the machine and set studio line level; then put on a reel of blank tape and set record level. You adjust everything from the front, using meters built into the Dolby units. You can still use any tape you want, record at any level you want, except that from then on you get a 10 dB better signal-to-noise ratio.

Using the system.
It is good practice to start every reel of tape by recording a section of Dolby Tone, which is generated by a distinctively modulated oscillator built into each noise reduction unit. There will never be any question of correct level with that reel again, at your own studio or any other one to which you send the tape. Decoding will therefore be perfect – for 100% maintenance of signal integrity.

Working up from the noise level, you can take some of the 10 dB as head room – for more relaxed, easier recording. You will be surprised how quickly you become addicted to the luxury of nearly dead silence *and* sparkling clean peak levels free of distortion.

With the Model 361 units, there is no extra operating bother, since the operating mode is automatically switched by your recorder. The Dolby Tone can be actuated from your mixing console.

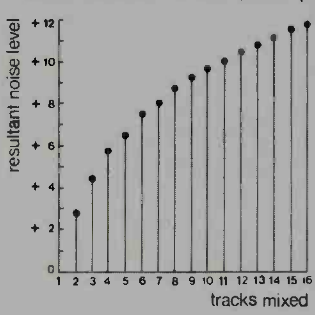
Can you count on it?
The characteristics of your recorder and recording tape will probably vary more than the Dolby System. If you ever want to reassure yourself that the system is working perfectly – good engineering practice – we make an inexpensive device that tests a noise reduction module and the chassis in about one minute (NRM Test Set, Cat. No. 35). The test set is an easy to use plug-in unit; you can do the testing whenever you wish *without* external test equipment. If the meter reads "Test" in all 24 switch positions; the module is ok. If any test result is out of spec, pop in a spare module and let us worry about the other one – you'll get a quick replacement.

Special tape – that was 10 dB ago.
The purpose of the Dolby System is to let good engineers come closer to perfect recordings. Used with today's best tapes, it gives skilled engineers a chance to show what tape can really do.

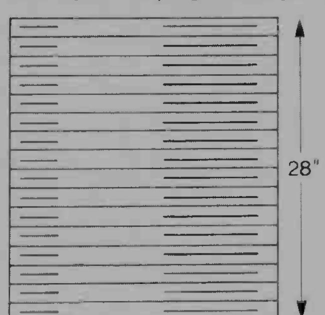
Take the distortion treatment that ordinary recordings give to drums. Good producers – those who ask to compare recorder line-in and line-out – learned a long time ago that non-Dolby tapes actually change the sound of drums because of the high peak levels needed to stay above the noise, even with the best tape you can buy. With the Dolby System, you can record at more conservative levels and get a perfect line-in, line-out match. If your business is recording sound (instead of re-arranging it), Dolby is the only way to record drums – or any other sound.

The bad news.
Sixteen of the Model 361 Noise Reduction Units sell for $11,840; a spare noise reduction module is another $425; the NRM Test Set $140.

What do you do next?
First, whether you think you'll buy now or later (or maybe not at all), get someone from Dolby over for a demonstration. Satisfy yourself that what is printed here is correct. Call (212) 243-2525 and we'll arrange to set the system up in your studio.



Any way you look at it,
more tracks mean more noise.



The entire sixteen units will take less space than a pair of bookshelf speakers.

Dolby Laboratories Inc
333 Avenue of the Americas
New York NY10014

Circle No. 112

How to get something for nothing . . .
"Loudness enhancement" or "the impression of enhanced intensity" is the phenomena regarding the fact that the ballistic characteristics of the human ear cause us to sense a loudness increase, in excess of the sum of the energy supplied to the two speakers, when delay is inserted in one channel. This impression of more "liveness," or that the sound has more "body," and that the sound source seems to gain volume has been studied by many with varying results.

THERE WILL BE A SLIGHT DELAY

by M.T. (Bill) Putnam, President
the URC Companies:
United Recording Corp.
Western Recorders
Coast Recorders
UREI

AUDIO DELAY LINES . . . the latest entry into the field of audio program processing accessory equipment introduces the possibility of an important new dimension of program enhancement. The new audio delay systems forecast their uses to be as convenient to mixers and producers as have been equalizers, filters, and dynamic controllers (limiters, compressors, gates, etc.) which are well known and are in general use in today's studio operations.

The basic techniques relating to the application of audio delay devices are not necessarily new. But, because of the new developments which have evolved audio delay devices in new forms; Wide Band Acoustical devices, Digital devices, the "Bucket Brigade" devices, all of which have now become, or are about to become, practical in both size and cost, a fresh look at the practical applications of audio delay offers the promise of being extremely rewarding . . . especially in the search for that ever elusive "new sound."

A bit of background. Many years ago when fair quality wire recorders first became available and were then quickly followed by magnetic tape recorders in the early 1950's, it was discovered that the enhancement of reverberation which came about by delaying the send feed to the echo chamber came about via the fact that we were simulating an approximation of the "early sound" first reflected before the onset of reverberation. The time delay limitations related to the record to play-back head spacing vs. tape speed.

The "early sound" is the term commonly used to identify the first few reflections arriving at the listeners position (or microphone) before the onset of reverberation. This "early sound" more readily defines the size of the room than does the reverberation. The length of persistence of the multiplicity of hundreds of random reflections that follow this early sound defines the liveness of the room, and constitutes the decay envelope . . . or the reverberation time of the room.

The length of time required for this sound to decrease 60dB in level is called T_{60}, or reverberation time.

Tape feedback recirculation was also quite common, particularly before the availability of so many electro-mechanical reverberation devices, and when acoustic chambers were still a luxury. Using tape feedback we didn't really simulate long colorless reverberation, but rather used it simply to gain a liveness effect, since to recirculate enough signal at 15ips to achieve even a small decay time was to inevitably end up with too much recirculation gain and, then, suffer the periodicity and coloration (comb filter effect) which comes with any single dimension delay system.

With reference to some of the delay phenomena we will discuss which require very short delays of a few miliseconds (M.S.), the tape speeds required coupled with the inconvenience place practical limits on the regular use of this method. In short, tape recorders are a rather cumbersome and sometimes noisy, as well as an expensive way to create short delays.

Another early approach (25 years ago) was the classic analog method involving a large number of low-pass filters and loss make-up amplifiers and equalizers. One such system provided 10 M.S. of delay requiring approximately two full 6 foot racks of filters and amplifiers. The cost for a single channel figures out to be a little over $1500 per milisecond of delay. (Never really made it, somehow.)

DELAY METHODS TODAY

WIDE BAND ACOUSTICAL DELAY LINES: Acoustical delay lines have been with us (or without us) for more than five decades, but until Dr. Duane Cooper of the University of Illinois evolved a practical system which provides a very useful amount of delay, 16 M.S. or more, in a very flat system (±1-1/2dB, 40Hz to 10kHz, at a very good signal to noise ratio), little favor was gained by this family of devices. (This two channel unit is now available, called the UREI "COOPER TIME CUBE," at a cost of approximately $425 per channel.)

DIGITAL DELAY SYSTEMS: The use of digital processing systems were experimentally used by Dr. Schroeder of Bell Labs, and others, using half million dollar computors in early analog to digital, digital to analog methods for providing experimental wide band audio delays. Equipment of this type is now manifest in $3000 to $5000 single channel processors with time base multiple outputs, as represented by the Gotham Audio "Delta Tee," and the Melcor digital time delay line, among others.

ANALOG "BUCKET BRIGADE": State of the art methods also indicate a high order of feasibility and practicality for what is known as the "bucket brigade" technique. The term "bucket brigade" describes the way in which the device, an analog shift register, transfers information from stage to stage in response to timing signals. Analog information is stored in a series of capacitors as charge deficit. The transfer circuits are MOS transistors which allow the charge deficit to move from one capacitor to the next. There are IC chips in experimental production which emphasize the interest and the practicability for less expensive ways of achieving increments of wide band audio delay. This group of devices will probably end up in a middle cost bracket, somewhat lower than the digital lines, but more expensive than the acoustical version.

Having discussed the history of time delay methodology, and then the equipment now available (or on the horizon) it is all rather meaningless unless we put time delay to the practical use which will enable us to take another step toward capturing that ever elusive "new sound." Since our purpose is to discuss fundamentals relevant to sound recording and reproduction, the application of time delay to public address system needs, i.e., to provide shorter differences of arrival time, (live direct sound from the stage vs. amplified sound), will be omitted from this article, other then to say that larger delays, 100 M.S. and more, are required in large auditoriums to reduce the interference. Digital delay lines are particularly suitable for these P.A. applications.

It is important to review some of the psycho-acoustic phenomena and the basic engineering physics relating to human hearing and sound perception which creates the practical need, and prescribes the practical uses for time delay devices.

The basis for understanding some phenomena which most prevanently relates to the application of time delay devices was encompassed in the early research work of such scientists as Dr. Helmut Haas, E. Meyer, G. R. Schodder, BeKesy, Barkhausen, Madsen, Damaske and others.

The acoustic characteristics which produce certain aural effects, basic to all stereo and quad mixing, were well documented by Haas and these others in studies which outline: 1) "the precedence effect" or the "Law of the First Wavefront," 2) the subjective integration of sound by the ear which produces "loudness enhancement," 3) notation of intensity increase required by delayed sound to relocalize the sound to its original position, 4) that the localization of the source then becomes vague and unstable, 5) that the limit of "sound fusion," or the maximum of

FIGURE 1

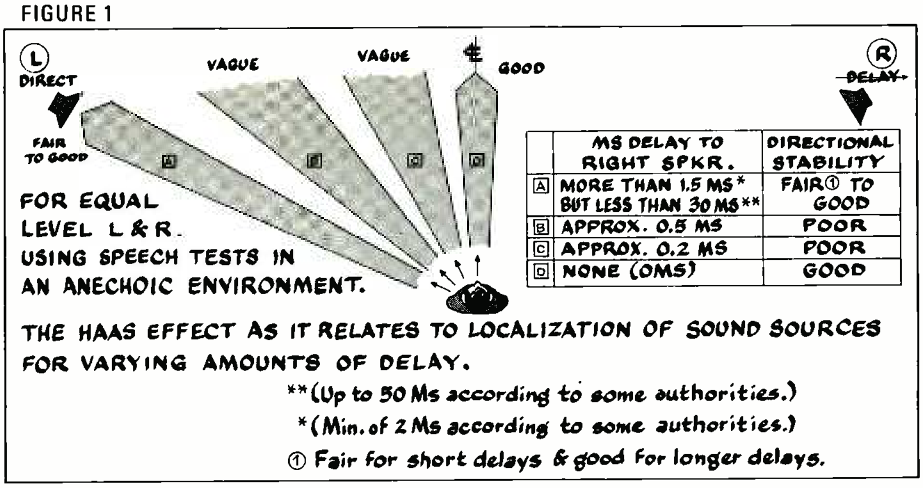


FIGURE 1, shows graphically the effect of introducing varying amounts of time delay to the right delay allowable before a second distinct echo was detectable, was approximately 30-35 M.S. (Other authorities felt that perhaps 50 M.S. was a more realistic limit.)

Dr. Haas' experiments were conducted using speech material and noise as program sources. Too, Haas' data was established in an anechoic environment (atop a building), and in our mixing practice these effects become diluted, to some degree, by the reverberation time of the listening area. Therefore, the recognition of these effects may be somewhat less distinguishable when listening to various types of program material in the typical mixing environment. However, overall, the data accumulated provides the scientific basis for understanding the delay phenomena which creates the "spatial illusion," or the illusion of "instrument doubling," as well as other aural effects relating to time and space, in both stereo and quadraphonic recording.

channel (R). The level to the right and left channels is equal. The change in sound source localization, shown by the shaded areas A - B - C - D, as heard by the listener at the center position, is the result of introducing delay only to the right channel. Again, to emphasize, the level to both left and right channels remains equal.

The directional stability or point source localization is good and positive with no delay to the right channel, as seen by the shaded area defined 'D' in the diagram.

With 0.5 M.S. delay, and 0.2 M.S. delay introduced, respectively, to the right channel the directional stability or sound point source location (shaded areas 'C' and 'B') becomes poor or vague.

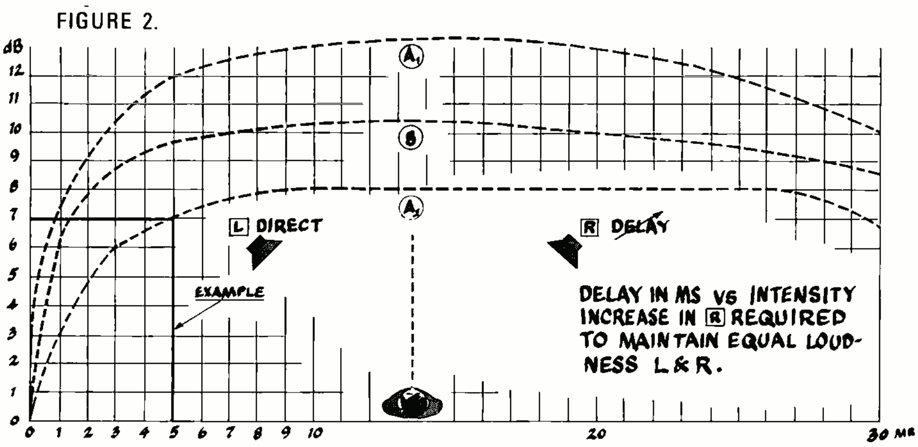
When the delay to the right channel is still relatively short, in the order of 1 or 2 M.S., the localization is still not well defined, but the sound seems to come mostly from the left speaker, as shown by the shaded area 'A' in figure 1. Continuing to refer to shaded area 'A', as the delay is increased up to 30 M.S. this localization is substantially improved. Beyond 30 M.S. delay a separate and definable echo can be distinguished. The ability to detect this distinct repeat, of course, depends on the program material. For example, in the case of slow legato musical phrases a longer delay can be tolerated before detection of the echo.

It is important to point out again, that these results tend to become diluted as the reverberation time of the control room or mix-down room becomes a factor; but the basic principle is valid, as we can apply it in a meaningful way . . . as time delay devices become a more common tool in recording.

How then, if a signal is fed in phase, equally to two speakers, separated a few feet, and the increase in sound intensity is the classic 3dB, how then can we get an apparent increase in loudness beyond this classic 3dB by the simple expedient of adding only a small delay to one speaker?

Many sophisticated explanations have been presented over the years in answer, and in essence they

FIGURE 2.



A_1, A_2 - UPPER/LOWER LIMITS OF HAAS DATA (1949)
B - MEYER/SCHODDER DATA (1952)

FIGURE 2, shows the results of separate data from Dr. Haas and collectively by E. Meyer and G.R. Schodder relative to the increase in intensity required at the right channel as a function of varying amounts of delay to produce apparent equal loudness from both left and right channels (center localization).

For example, according to Dr. Haas' lower limits, with 5 M.S. delay to the right channel, the right channel must be raised 7dB in level to relocate the apparent sound source to vague center localization.

suggest that our human hearing mechanism intergrates the sound intensities over short time intervals similar to ballistic measuring instruments, and that because of the delay to one channel the subjective loudness increase is in excess of the sum of the electrical energy supplied to both speakers.

. . . a psycho-acoustic something for nothing.

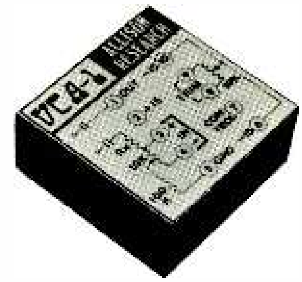
In recapitulation of the results of studies concerning this impression of "enhanced intesntiy," the following shows the variation in some data collected over the years:

	SUBJECTIVE INCREASE IN LOUDNESS	NOTE
*Haas 1949	3 DB	No enhancement except enlargement of sound source.
Aigner-Strutt 1934	5–6 DB	For equal band width direct and delay.
Aigner-Strutt 1934	6–9 DB	With delay band width restricted.
Meyer/Schodder 1952	4–5.6 DB	Enhancement starts at approx. 15 M.S. and greater.
Cross section of data from various authorities 1950–1970	4–6 DB	Some data based on mult. reflections shows increases to 10 DB.

*Later researchers felt that the SPL levels in Haas' experiments were too low.

VCA-1 VCA-1PC

VOLTAGE CONTROLLED AMPLIFIER MODULE & P.C. BOARD ASSEMBLY



Here are the specs...

SPECIFICATION	MIN	TYP	MAX	UNIT OF MEASUREMENT
MAXIMUM OUTPUT LEVEL (CLIPPING)	+18			DBM INTO 30 KOHMS (MAY BE MODIFIED TO FEED 600 OHMS BY ADDING ONE EXTERNAL RESISTOR - R5)
MAXIMUM INPUT LEVEL		+18		DBM (WHEN STRAPPED FOR UNITY GAIN)
TOTAL HARMONIC DISTORTION (THD)		.10	.18	% @ +10DBM INPUT (R1 = 100K)
DISTORTION IS A FUNCTION OF INPUT LEVEL AND IS INDEPENDENT OF CONTROL VOLTAGE		.18	.32	% @ +14DBM INPUT (R1 = 100K)
		.33	.50	% @ +17DBM INPUT (R1 = 100K)
OUTPUT NOISE @ CONTROL VOLTS = 0 VDC (OFF)		-94	-91	DBM (20HZ TO 20KHZ)
OUTPUT NOISE @ CONTROL VOLTS = 1.00 VDC (ON)		-67	-64	DBM (20HZ TO 20KHZ)
LINEARITY (TRACKING) (A LINEAR TRANSFER FUNCTION OF CONTROL VOLTS VS. GAIN)				GAIN TRACKS THE CONTROL VOLTAGE WITHIN ±1DB OVER THE CONTROL VOLTAGE RANGE OF .001VDC TO 1.00VDC (60DB) WHEN ADJUSTED WITH EXTERNAL TRIM POT (R4)
CONTROL SIGNAL FEEDTHROUGH (WHEN NULLED WITH EXTERNAL TRIM POT (R2)		80	60	DB BELOW RATED OUTPUT @ 1KHZ
MAXIMUM ATTENUATION (CONTROL VOLTS = 0VDC)	100	120		DB (20HZ TO 100KHZ)
FREQUENCY RESPONSE (AUDIO SECTION)				±1DB 10HZ TO 400KHZ
FREQUENCY RESPONSE (CONTROL SECTION)				±1DB DC TO 400KHZ
ACCURACY OF REFERENCE GAIN (ON GAIN) @ R1 = 100K, CONTROL VOLTS = 1.00VDC	-1	0	+1	DB
INPUT IMPEDANCE (AUDIO SECTION)				APPROXIMATELY EQUAL TO THE VALUE OF R1
INPUT IMPEDANCE (CONTROL SECTION)	100	400		MEGOHMS
POWERING REQUIREMENTS				±15VDC @ 15MA
SIZE				1.06" BY 1.06" BY .56" EPOXY MODULE
CONNECTORS				10 .032"-DIA GOLD PLATED PINS

Hey, you bona-fide audio equipment manufacturers who missed our $20 introductory price, the offer is still good; one to a customer, please.

ALLISON RESEARCH

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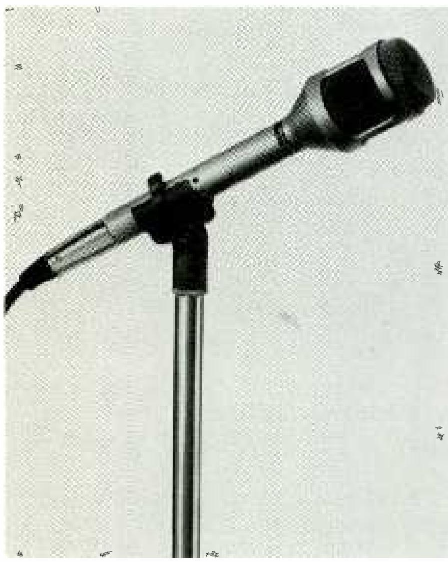
GOTHAM AUDIO DEVELOPMENT CORP., NEW YORK, N.Y.

Circle No. 115

NEW PRODUCT NEWS

SHURE BROTHERS INTRODUCES PROFESSIONAL QUALITY MICROPHONE WITH NEW "POP" PROTECTION. The new microphone, called the Model SM54, incorporating all of the performance characteristics of the SM53, features a series of special acoustic filters that virtually eliminate problems of explosive speech and breath sounds. This new feature is particularly important since the SM54 has minimal proximity effect, allowing extremely close voice pickup without excessive low frequency response.



Other features include a broad, true cardioid pickup pattern with a wide front working angle at all frequencies that effectively eliminates tonal coloration. The unit also sharply reduces stand, cable and handling noise through a built-in shock mount. The SM54 has the same rugged design as the SM53 and can even be dropped on its nose without damage to the element or alteration of the sound quality. The case is of aluminum with stainless steel mesh grille and is finished in a neutral, "no-glare" finish for on-camera use.

Professional net price of the Model SM54 is $162.00.

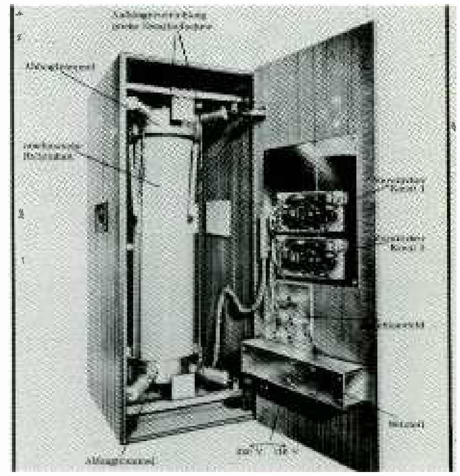
SHURE BROTHERS INC., 222 HARTREY AVENUE, EVANSTON, ILLINOIS 60204.

Circle No. 116

AKG HAS INTRODUCED ITS NEW REVERBERATION UNIT MODEL BX-20E. Reverberation is produced by the torsional vibration of a specially treated coil spring. The torsion elements consist of several units with microscopic and macroscopic variations of the line parameter interspersed with controlled damping and supporting elements. Electronic damping, utilizing motional feedback, permits variation of the decay time from 2 to 4.5 sec.

The BX-20 is a two channel unit with independent control of decay time of each channel allowing maximum flexibility in stereophonic or monophonic use. The two inputs can be paralleled, and either channel can be used separately; channel separation is 60 dB.

In view of its dimensions (40" h x 17" w x 19" d) and its weight (105 lbs.), the unit is suitable for on-location and remote use. It may be operated either from a 110 v a.c. or 24 v d.c. source. AKG, 100 EAST 42 STREET, NEW YORK, N.Y. 10017.



Circle No. 117

Circle No. 118

to all recording and broadcast studios

THE SANSUI QSE-1 IS *ALL* YOU NEED TO ENCODE 4 FULL-FIDELITY CHANNELS —AND NOTHING ELSE.

Just add it to your existing equipment for instant conversion and here's what you have going for you:

(1) It yields **accurate** sound-source location in every direction for startling live-sound ambience.

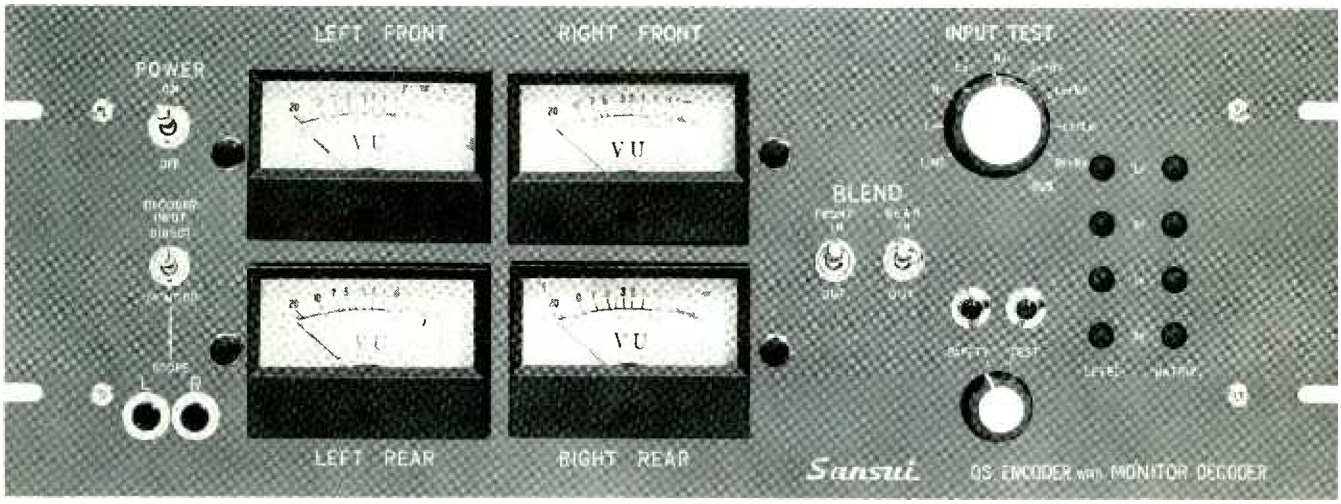
(2) It's in broadcast and recording use **today** with outstanding results.

(3) A **complete** line of complementary Sansui home hardware is available now. In fact, thousands of Sansui decoders are in users' homes already.

(4) It's compatible with 2-channel stereo and other four-channel matrix systems.

To be more specific:

Its ingenious ±"J" phase shifters completely eliminate the signal dropouts and shifts in sound-source location that plague other matrix systems. Its symmetrical treatment of all four channels can **accurately** pick up **and relocate in reproduction** any sound source over a full range of 360°—so there are no limits to total freedom and flexibility in using creative studio and psychoacoustic techniques. And present standards of frequency response, signal/noise ratio and dynamic range are maintained.



The Symbol of Sansui 4-Channel Sound.

It reproduces flawlessly on present two-channel stereo and monophonic equipment. And it will produce four-channel output not only through matching Sansui hardware, but through all other available decoders—and there are 600,000 of them world-wide today.

Thousands of them are Sansui QS-1 Synthesizer/Decoders that will decode it flawlessly. So will any of the full line of matching Sansui 4-channel receivers and converters for existing two-channel systems—made by the most respected name in stereo today throughout the world, and a recognized pioneer in four-channel sound.

Can you afford **not** to make this simple addition? Experiment with one right now. Learn what other recording and broadcast studios everywhere, now working with the QSE-1 Encoder, are finding out for themselves. Confirm their astonished conclusions.

For full details, contact your nearest Sansui office now.

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Sansui Audio Europe S.A.	**Belgium**	Diacem Building Vestingstraat 53-55. 2000 Antwerp. Tel.: 315663-5. Cable: SANSUIEURO ANTWERP. Telex: ANTWERP 33538.
	Germany, W.	6 Frankfurt am Main, Reuterweg 93. Tel.: 33538.
Vernitron Ltd.	**U.K.**	Thornhill Southampton S09 5QF. Southampton 44811. Cable: VERNITRON SOTON. Telex: 47138.

Circle No. 119

TEAC INTRODUCES MODEL TCA-43 MULTI-TRACK MASTERING TAPE DECK. Incorporating the new "SIMUL-SYNC" feature of the TCA 43, recordings may be made independently and synchronized to produce discreet program material on all four tracks. By using the head function selector switches, any of the four tracks may be synchronized to any or all of the remaining three tracks. This is possible by utilizing the recording head for play-back monitoring thereby eliminating the time delay usually encountered by using the play-back head for monitoring. The result is complete flexibility in producing special effects recordings. Multiple tracks can be synchronized and the resultant four channel sound blended during play-back to produce any desired sound mix.

While the professional OVERDUB feature of this "SIMUL-SYNC" deck is one of the features of primary importance, this model also has the capability of discreet four channel record and play-back as well as two channel 1/4 track stereo record and play-back with automatic reverse. It can also accommodate 1/2 track play-back.

The use of the optional accessory known as the AX 20 mix-down panel can provide left, center or right distribution of each of the master tracks. The AX 20 allows the final "SIMUL-SYNC" recording master to be mixed down to stereo or mono and copied onto a tape with a second recording deck. TEAC CORP. OF AMERICA, 7733 TELEGRAPH ROAD, MONTEBELLO, CA. 90640.

Circle No. 120

ARP ANNOUNCES THE SOLOIST. The Soloist is a preset keyboard-operated synthesizer that offers 18 different, realistic instrument voices, from tuba to piccolo, bass guitar to violin, and more. In addition to these traditional instruments, the ARP SOLOIST is capable of dozens of very un-traditional sound effects.

The new model is available as a result of the growing demand for a small, moderately-priced synthesizer. The Soloist weighs 25 pounds and measures only 30" x 10" x 4".

ARP INSTRUMENTS, A DIVISION OF TONUS, INC., 45 KENNETH STREET, NEWTON HIGHLANDS, MASS. 02161.

Circle No. 121

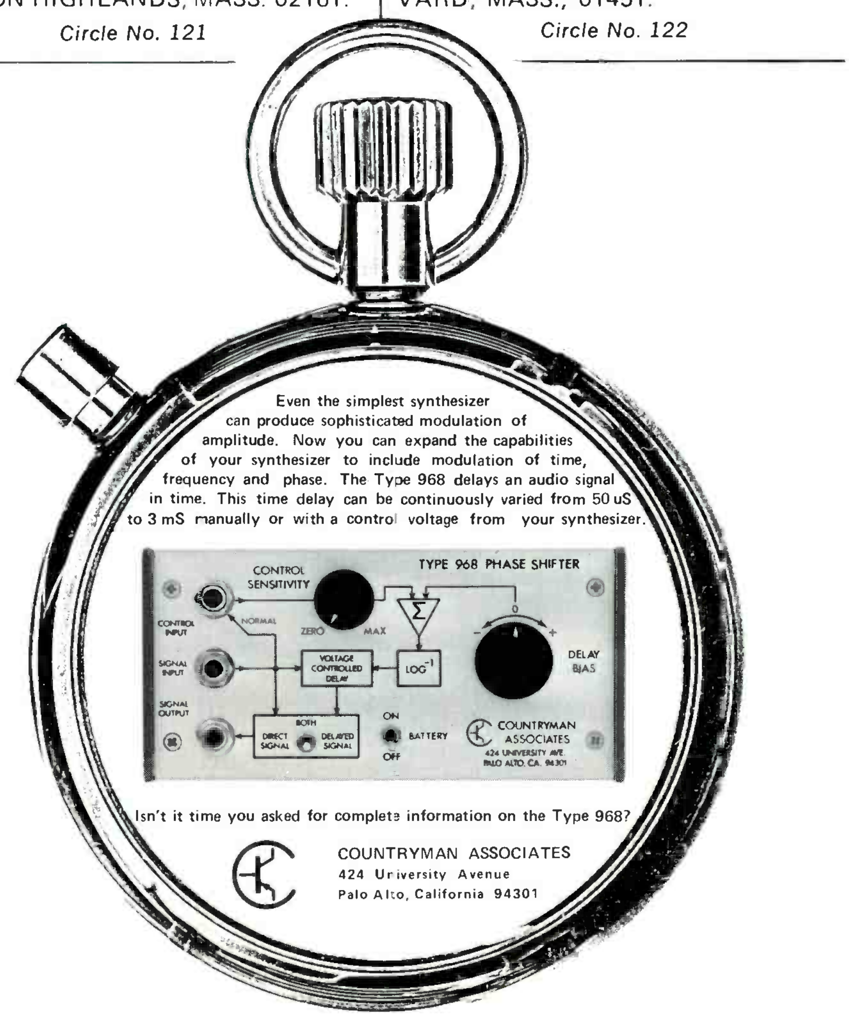
DBX 187 TAPE NOISE REDUCTION SYSTEM.

The DBX 187 four channel noise reduction system, said to decrease tape noise by 30dB, decreases hum, hiss and crosstalk due to the recording process. Straight line decibel compression before recording and complimentary expansion on playback gives perfect transient tracking with no pilot tone or critical level matching. Hiss and asperity noise at normal levels are reduced by 10 db giving an extremely clean, clear sound. This system is useful for multi-track recording, stereo mixdown, telephone line transmission and echo systems.

$1950 for four channels with remote control capability.

DBX INC., BOLTON ROAD, HARVARD, MASS., 01451.

Circle No. 122



Circle No. 123

FAIRCHILD ANNOUNCES 3rd GENERATION 'REVERBERTRON.' The new "Reverbertron," Model 659A, features instant selection of three decay times. It uses six electro-mechanical delay lines, each isolated against building rumble and environmental noise, and each tuned differently to produce a natural reverberant effect.

Smaller PC boards and a generation unit now only 3¼" high — it was formerly 19 inches — permit stacking several Model 659A's in the space previously required by a single system. Because it measures only 7" overall and is low in weight, the new model can also be carried conveniently into the field. An exclusive knob-controlled system-lock feature prevents vibration and damage in transit.

Among other of 11 improvements in electronics of the Model 659A are a lower input level requirement (down to – 30 dbm); an improved S/N (65db); full range equalization with bass, midrange and control presence peak selection, high and rolloff; metering of all signals; local and remote selection of three degrees of reverberation (dry, premix 1 and premix 2), and continuous mix control. FAIRCHILD SOUND EQUIPMENT CORPORATION, 15-58 127 STREET, COLLEGE POINT, N.Y. 11356.

Circle No. 124

BURWEN WIDEBAND GAIN CONTROLLER. The GC101 Wideband Gain Controller is a two quadrant transconductance multiplier, divider, squarer, square rooter, and gain controller. It has the unique property of being able to control gain over a 60 dB range with 1% or .1dB accuracy, extremely low distortion, low noise, and wide frequency response. The GC101 delivers ±10 V at 10 mA, provides response from dc to 12 Mc at unity gain, and from dc to 70 kc at 60 dB gain. Noise is 130 mV rms in a 20 kc bandwidth at unity gain

and only 6 u V rms at –60 dB gain. Because its harmonic distortion typically .04% at X = 2 V rms does not increase at low gain, the device opens up new applications in automatic gain control, sweep generation, modulation, and filtering. BURWEN LABORATORIES, 12 HOLMES RD. LEXINGTON, MA. 02173.

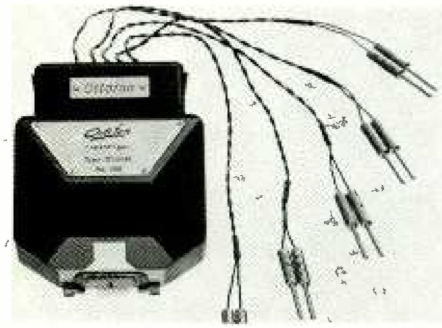
Circle No. 125

GATELY ELECTRONICS ANNOUNCES THE AVAILABILITY IN THE UNITED STATES OF THE NEW ORTOFON 701 DISK CUTTING SYSTEM.

This new disk cutting system is presently being used by several European re-

Circle No. 126

cording companies and features a 500 watt rms amplifier to assure full level cutting of the very highest frequencies. In disk cutting, the power requirements vary as the cube of the frequency. That is for equal level 8 times the power is required at 20 kHz as at 10 kHz. When used with the Ortofon DSS-661 cutting head the response is flat way beyond 20 kHz being only 3dB down at 36 kHz.



The 701 cutting amplifier incorporates protection against cutter head burn out. All adjustments are available on the front panel. To show the advantages of this new system a demo record and tape are available to existing cutting studios so they may use their present equipment to cut a test record from the demo tape and compare the results obtained with their present equipment to the results obtained on the demo record cut using the new Ortofon 701 system.

GATELY ELECTRONICS, 57 WEST HILLCREST AVENUE, HAVERTOWN, PA., 19083.

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WANTED: Young engineer with strong track record to Cheif at one of the largest independent recording complexes in the South. Good pay and outstanding opportunities. Contact: 205–871–4221.

NOTICE

INDIVIDUALS seeking employment in the recording industry may submit their qualifications for FREE publication in *RECORDING engineer/producer.*

Listings will be limited to 30 words, and will be limited by available space. Listings will be selected for publication on the basis of earliest postmark. Listings will not be automatically repeated or carried over to the succeeding issue.

Young Engineer seeks employment with 8 or 16 trk. studio. Background including musical, 4-trk., electronics, etc. I shoot for new sounds, diplomacy, efficiency. Eric Enfield, Laddie's Recording, 1105 E. St. Charles Rd., Lombard, Ill. 60148 (312-629-1140)

Engineer (25) finishing Army tour desires position in mixing or installation/maintenance. Electronics degree. Exp. in location recording and high power PA. Resume on request. John Probst, 609 S. Walter Reed Dr. Arl. Va. 22204. Phone: 703-920-0568 aft. 4 pm.

Engineer/Mixer, four years experience, familiar with 16 track. Johnathan Solak, 6 Van Ness Road, Binghamton, New York 13905. (607) 797-3909.

Young — audio production mixer — tech. 8 yrs. pro experience heavy audio production/tape background & studio techniques in general. B.A. degree in music. Contact Pat Coghlan Jr., 18200 Valley Vista Blvd., Tarzana. Tel. 343-5169 — 780-4787 message.

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Wanted to sell the GARNER Model 202 high-speed professional audio tape copier. Makes five 1200 ft. copies in four minutes. Single capstan drives all tapes. Solid State electronics.

Write Philip Mullin
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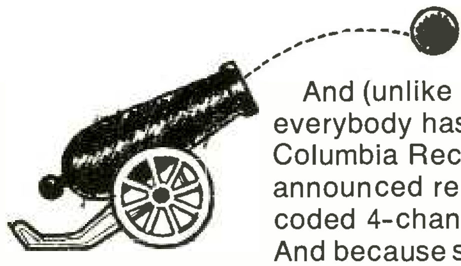
Circle No. 131

A report in the 4-channel war of the matrixes:

The war is over!



And (unlike real wars) everybody has won. Columbia Records has announced release of encoded 4-channel records. And because support from major record companies is essential to 4-channel, we welcome them. Columbia now joins the many pioneering record manufacturers who've already produced thousands of 4-channel discs.

We must admit that at first we were concerned. Because while most of the original matrixes were basically compatible, these new SQ discs were different. Which could have led to a battle of the matrixes and even more confusion in the marketplace.

But we knew our matrixing system was best, so what to do about this promised flood of seemingly incompatible discs? The answer: a new "universal" E-V decoder now in production. Not only does this improved decoder handle our STEREO-4[T.M.] (and all similarly-encoded material) but we've added sophisticated circuitry to decode SQ records accurately. It even does some things decoders built solely for the SQ format don't, like more correctly controlling the position of a front-center soloist.

So, now the E-V Decoder is the only one for *all* matrix 4-channel programs. And now — more than ever — matrixing (encoding four channels of sound into two) continues to grow as the method to get 4-channel sound on records, FM, and tape to the listener . . . now and in the foreseeable future.

What about our "old" EVX-4 Decoder? Well, despite the algebra, it actually decodes SQ records remarkably well. It just doesn't offer complete rear directionality from these different discs. But unless it is directly compared with our improved decoding this has proved a minor issue for many listeners.

And what should the independent recording studio do now? Switch to the SQ system . . . or stay with STEREO-4? Well, with our universal decoder in the wings, we can be dispassionate (or nearly so). And our new encoder is equally universal. With a choice of STEREO-4, SQ, or a compromise encoding. You choose the encoding that best suits the recording at hand.

Just call or write and we'll be happy to talk to you (in English or algebra) about the relative merits of all of the current or proposed encode/decode combinations. Including what to listen for, and problems to avoid. And we'll tell you in detail how the STEREO-4 encoder fits so well into the present picture.

So, hopefully, order is restored. Record companies can now get on with production of software in increasing quantities . . . while we provide fully compatible hardware (plus some new models coming that will assure a permanently bright future for STEREO-4 matrixed sound). And consumers can begin to fully enjoy the fruits of all our labors.

Peace.

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Circle No. 132

When you buy MCI's new JH-416 mixing console—priced at a phenomenal $18,500 for the 16-track model—you'll have enough money left over to buy our JH-16 recorder ($16,500) . . . and still be paying less than what you'd expect for a comparable mixing console alone. Expandable to 24 tracks (total: $24,100), the JH-416 makes possible a complete studio package of heavy hardware at unheard-of savings. And to save even *more*, consider starting out with an 8-track version of the JH-416 ($12,900), which you can build on later.

We'll match our mixer—its specifications and functions—with any competitive model, even the $40,000-and-up jobs. For example, each input module of the JH-416 features: Illuminated straight-line fader • four-knob equalizer • individual track/quad monitor assign with pan and level control • two cue feeds • two echo feeds • solo/preview • 16-track, plus direct bus assignment • overdub switching • sub master on each bus • plus much, much more for much, much less.

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