ISSN0011-7145

THE SOUND ENGINEERING MAGAZINE

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DECEMBER 1979
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Coming Next Month

• Next month, we finish the old year by concluding our series on microphones, with features on the Pressure Recording Process, and the use of instrumentation microphones in recording sessions. Next, we take a look at some of the latest developments in analog audio. At least one manufacturer is bucking the digital tape recorder trend, and we'll find out a little more about this in the January issue of **db—The Sound Engineering Magazine.**

About The Cover

• Microphones on the cover: (Top row; left to right) Neumann U-87, Shure SM-58, Electro-Voice RE-20, Sennheiser 441; (Bottom row; left to right) Audio-Technica AT-813, Sennheiser 421, Shure SM-81, Neumann KM-84.

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THE SOUND ENGINEERING MAGAZINE

DECEMBER 1979 VOLUME 13, NUMBER 12

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

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db, the Sound Engineering Magazine is published monthly by Sagamore Publishing Company, Inc. Entire contents copyright © 1979 by Sagamore Publishing Co., 1120 Old Country Road, Plainview, L.I., N.Y. 11803. Telephone (516) 433 6530. db is published for those individuals and firms in professional audio-recording, broadcast, audio-visual, sound reinforcement, consultants, video recording, film sound, etc. Application should be made on the subscription form in the rear of each issue. Subscriptions are $9.00 per year ($18.00 per year outside U.S. Possessions and Mexico; $10.00 per year Canada) in U.S. funds. Single copies are $1.95 each. Controlled circulation paid at Brattleboro, VT 05301. Editorial, Publishing, and Sales Offices: 1120 Old Country Road, Plainview, New York 11803. Postmaster: Form 3579 should be sent to above address.

db Letters

To The Editor:

A few comments on Irving Joel's article, "An Automatic Broadcast Console":

The console Joel describes is fine in the automatic mode. It is a substantial improvement over conventional program automation systems, since it places the operator in the same room as the system, with the controller in front of him. It also eliminates the need for a separate piece of gear for live inserts.

However, I feel that the console is not human-engineered well enough for operation in the *semi*-automatic mode:

First, Joel states that "low profile" console design is unnecessary, due to the proliferation of "combo" operation. There are objections to his reasoning: In many stations, the news originates in another studio. In that situation, good visibility is of prime importance, not only for cueing, but to help the newscaster and operator see each other in any on-air dialogue. Additionally, having a high, essentially flat surface in front of the operator tends to make the operator feel closed in, and such a surface close behind the microphone may have undesireable effects on its directional characteristics and/or tonal quality.

Second, while Joel's system of alert leds and other indicators is good, it could be improved, along with the location of the "play" and microphone "on-off" switches. The relative positions of the input modules with their associated push-buttons and the VU meters can cause problems; when the operator starts equipment, the meters can be hidden by his arm. Experience (and discussions with other engineers) leads me to believe that it is not efficient to place frequently-used push-buttons at a height requiring the operator to raise his arm(s) off the table top.

The clock/timer is something no control room should lack. But the location on Joel's console could be improved. Ideally, the clock/timer should be directly over the VU meters. The second choice would be on the *right* side of the front panel. Right-handed people have a tendency to look to their right, and are more comfortable doing so. It is important that combo operators (who may be non-technical) be as comfortable as possible. (Of course, if there are a lot of left-handed operators at the facility where this console is to be used, the timer and clock are correctly positioned.) The headphone jack shows a similar dislocation. Many types of stereo headphones *must* be worn with the cord coming out on the wearer's left side, or they don't fit properly—which

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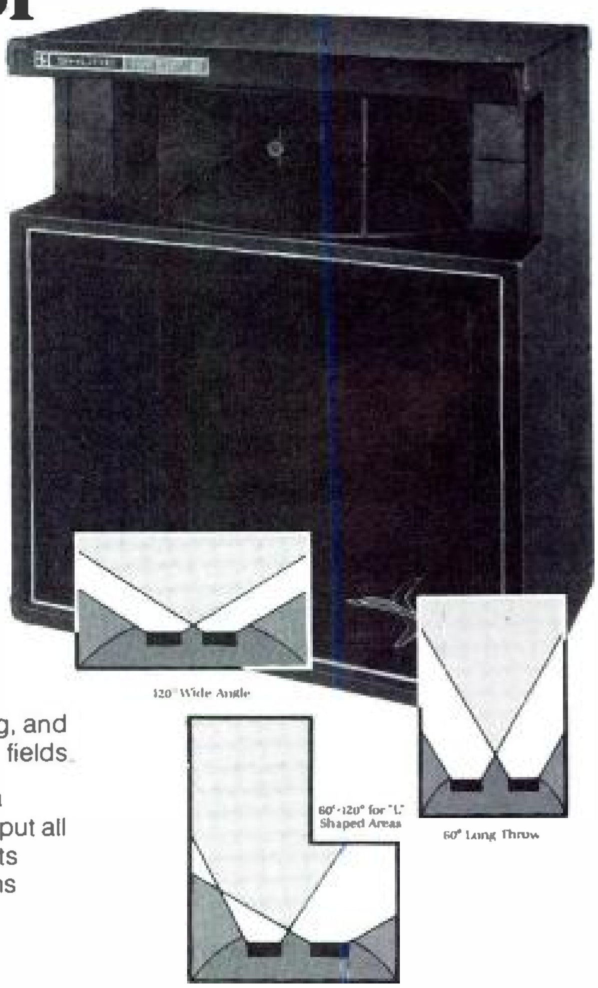

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may adversely affect their frequency response. With the headphone jack on the right side, the headphone cord has to run across the desk, the operator's lap, or both.

Some provision should be made for a copy stand at eye level in front of the operator. It is difficult to speak clearly while looking down, let alone see anything on the front of the console. A built-in copy stand would fulfill Joel's goal of keeping the weather forecast from in front of the VU meters.

While I agree with most of Joel's reasoning, I think the execution of his thought could be better. One final question remains: With all of the information this console provides for the operator, shouldn't some sort of outside thermometer and indicator lights (leds?) for the telephone be provided? I have built a few control rooms, and worked in many, and the above items rank right behind the time, and the remaining length of what's on the air in importance.

I would enjoy any feedback from Irving Joel, or anyone else interested in the human-engineering of consoles and control rooms.

Robert E. Barmore, Jr.
Staff Technician, WLBC AM/FM
Muncie, IN

Mr. Joel Replies:

I'd like to thank Mr. Barmore for his comments on my article.

A great deal of attention during design was given to human design engineering and function accessibility. The console is supported by two formica covered wooden end bells and the height can be reduced to suit individual installations.

Normal operation of the console in the semi-automatic mode is performed with one button marked START/NEXT EVENT, which can be easily reached without lifting one's elbows off the table. If for any reason this position is not suitable, a remote START/NEXT EVENT button connection is provided for and the button could be located at any remote location desirable. This is also true of the earphone jack which can be remotely located and placed well under the desk to completely get the cord out of the way. It so happens our mechanical design engineer is left handed but I don't feel this entered into the decision of the clock placement. Rather, consideration for the operator has placed them close to the microphone controls.

We felt that the microphone ON/OFF buttons should be located as close to the fader as possible, hence the position—and it might be wise to note here that all ON AIR functions are completed only after the switch has been released, which allows the operator to put his finger on a button anticipating a cue and being as-

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sured that it will not switch until it is released.

The copy stand is placed under the console for easy access and when reading copy, the VU meters are in full view. The angle of the stand is adjustable. In actual operation, we found this position to be quite workable.

As far as the telephone is concerned, we provide muting while the microphone is on the air and feel that the minimum wattage for the phone light should be 150 watts, not easily available in the console.

"S & H" Crowhurst

TO THE EDITOR:

I have read and enjoyed **db** for a number of years now, but each and every month my attention has been drawn more to the articles by Norman Crowhurst.

The content of his column is invariably interesting, but of increasing importance to me is the discrepancy between the middle initial assigned to Mr. Crowhurst on the contents page and on the heading of his column.

I have been following this for probably two years now, and I am amazed each and every month that the initials don't agree.

Please...let us all in on the secret! Is he an "H" or an "S"???

Possibly your magazine will become even more enjoyable with this mystery solved.

ROBERT W. CAMBRELENG
Engineering Department
WOR-TV, New York

db replies:

Finally!

In July, 1977, the style of our column logos was updated. And somehow, Norman H. became Norman S. (Our typesetter is a not-so-distant relative of Edsel Murphy.)

Of course everyone here at **db** *caught the error immediately. (Then why didn't you guys fix it?—typesetter.) However, we decided to wait until the letters started pouring in. In fact, we even had a little contest arranged—$20,000 a week for life, to the first person who contacted us to point out the goof. Unfortunately, your letter arrived just minutes after the contest deadline. (The deadline was fixed at five minutes before the first letter arrives.)*

In any case, Norman's "H" has been restored to its rightful place, and Robert W. Cambreleng gets a free one-year subscription to **db** *for being so attentive. After all, now that the mystery is solved,* **db** *should be even more enjoyable—so, it's the least we can do.*

TO THE EDITOR:

I was quite impressed with your September article on the ITC series "99" tape cartridge system and cartridge evaluation. We are currently involved in a project of deciding upon a tape cartridge to use when we start carting up our a.m. music in stereo. I was pleased to see the results of Mr. Nikanorov's tests as they reflected what we had discovered out here in "radio land" using just a "primitive" ITC RP series stereo cart recorder. We had purchased over 1,000 of a newly introduced cart that was billed as being the ultimate cart for stereo usage and phase stability. We found after testing for phase response that up to 25 per cent of each case of carts were not meeting specifications of the manufacturers. Approximately a dozen carts that I rejected were sent to this company's lab and were verified as defective due to various reasons.

1) Case Warpage.

2) Incorrectly manufactured plastic tape guides.

We decided to give this company one more shot at producing us one good case of carts with no rejects, as they have been very concerned and cooperative. The result was again 25 per cent failure out of a case of 24 carts. To check these carts against another brand I ordered a random case of 24 carts from the "top" rated cart, per Mr. Nikanorov's list. The result was 100 per cent of these carts producing excellent phase response. Both companies' carts were of the same length and checked in a manner in which our original supplier approved and recommended.

We are currently considering returning all 1,000 plus carts to the first manufacturer and going with the second because of what appears to be the increased superior phase stability of the later.

One additional point I would like to make in regard to the series "99" ITC machine. This unit will "mask" or cover up for these manufacturing inconsistancies of any cart. If you have a cart that is recorded correctly out of the series "99" machine how does one know that these "mechanical" problems that result in phase discrepancies between one brand might cause drift and playback problems down the line?

I believe in the series "99" and we have one on order with "ELSA," however, I wonder if certain manufacturers are hoping it will cover up their manufacturing or quality control problems? I want to be sure that I start out with a good cart and I plan to check every one out that we use with the RP series machine. After all these stereo carts are demanding a premium price, shouldn't we expect the claimed performance from the manufacturer?

Chris J. Cain
Engineering Director
Midwest Family Stations
Madison, Wisconsin

NORMAN H. CROWHURST

db Theory & Practice

More Systems Design

• As last month's column showed, any time you want to design a new system, you must consider how it correlates with systems already accepted and in existence. The advent of the 45 rpm EP and the 33 rpm LP are an example of different systems in this regard. The 45 rpm disc, with its large center hole, was an attempt to design a completely new, independent system, not related to prior systems at all. The 33 rpm disc took an alternate approach, to optimize the existing form.

But does that mean such systems will stay separate? Now we find systems—record players—with a variety of ways in which they can accommodate both. One way is to put those little spiders in the center of 45 rpm discs, so they will fit on the standard phonograph spindle. Another way uses adaptors on the phonograph spindle, to enable it to accommodate either type of record in its changing.

That relates to the mechanical aspects. A similar thing happens in the electronic part. Then we have the part that bridges both—the transducers. Disc is an electromechanical recording medium, in which a stylus mechanically follows a groove, and produces electrical signals, or vice versa, at recording. Tape uses electromagnetic properties of the tape to transcribe.

Disc has the advantage over tape, in that the groove is 3-dimensional, so it is possible to devise the 45/45 stereo system, in which two channels of "information" can be conveyed independently, using directions of movement mutually at right angles, to one another, and to the direction in which there is relative movement along the groove by the stylus. The fact that the groove moves, while the cartridge stays put does not alter the relative movement involved.

Tape, on the other hand, does not have the depth capability or, if it does, nobody has yet found how to use it. You have, essentially a surface of tape, which moves relative to a scanning magnetic gap. But you can stack up these heads, or gaps, to get a lot of them onto a relatively narrow piece of tape. And, to enable a lot of program to be stored in a small space, the tape itself can be made very thin.

THE SPACE PROBLEM

This space problem is somewhat aggravated, every time we decide we want to record, or convey, more channels of information: when we move from mono to stereo, and when we move from stereo to quadraphonic. And even more, of course, when we move to multi-track facilities, or however many tracks we decide to use.

But as usual, there is an offsetting factor. The modulation, power requirement, or whatever corresponds to it, needed to convey a given "loudness" impression in mono, has been shown to be more than that needed in two channels to convey an equivalent impression in stereo, which means that stereo doesn't necessarily need twice as much "space" as mono.

Now we come to the question of how we extend the already established stereo, in its various forms of 2-channel implementation, to quadraphonic. It is generally conceded that the loudspeaker position is in the form of a quad: 2 in front and 2 behind. The question is, how do we convey the 4 signals that are to be distributed to these 4 speakers, so the reproduction does what we want it to?

HOW TO FIT 4 CHANNELS

For the conversion from mono to stereo, both disc and radio came up with a way to put 2-channel information over a single channel: disc by using mechanical movements mutually at 90 degrees to each other; radio by using an ultrasonic switching frequency, 38 kHz. How can we extend that to squeeze in 4 channels?

Some systems designers thought of other ideas that had been tried for stereo. For disc, for example, one alternative offered had been to use a higher frequency subcarrier to do substantially what stereo f.m. did. Now that we have 45/45 for stereo, couldn't we get the other 2 channels in by using a subcarrier on each? What came to the rescue, once again, is a closer consideration of the natural relationship of signals.

Why are the back channels needed? Ideally, of course, they should be completely independent, so it does not matter how their content inter-relates. But how does it normally tend to inter-relate? In most situations, the front channels convey the complex source of the original sound, to present, for example the impression of various instruments in an orchestra, located left, right and various points in between.

And what comes from the back speakers? Reverberation of the same sounds presented from the front speakers, to convey an illusion of the auditorium in which you are listening to that orchestra. And what is characteristic of reverberation? Some proponents of pseudo stereo had already exploited this, by taking mono, splitting it between two loudspeakers to represent "direct" sound, then delaying it, and splitting it between the same two loudspeakers, but this time out of phase, to represent the "reverberated" sound.

And that form of pseudo stereo certainly "had something." It could not convey impressions of left and right, with points in between, of course. But it did convey the impression of reverberant sound, as a separate entity from the direct sound, as well as the real thing could. What does this mean?

If you've conducted phasing tests, you'll know. Putting the same program into two loudspeakers, in phase, makes the apparent source of the sound a position in between the two units. But putting it out of phase produces what has been called the dissociation effect: the sound seems to come from nowhere in particular: all around you, but not from the loudspeakers.

PSEUDO QUADRAPHONIC

This is done by "fooling" our hearing faculty. And, regardless of whether it's nice to fool mother nature, it works. Since the advocates of pseudo stereo demonstrated that bit of trickery, a similar group of people, who wanted to make "super stereo" have produced similar effects, using stereo program as the source, to generate pseudo quadraphonic.

Do you begin to see a pattern that we could use? Using the stereodisc, 45 45 as reference, lateral movement of the stylus represents center front as the source of sound, equal level in left and right, in phase. From the effects already observed, equal level in left and right, but out of phase, could well represent center back. That would be vertical movement of the stylus. Extending that combination, circular movement one way would be back right and circular movement in the opposite rotation would be back left. That represents left and right at 90 degrees.

This is, in fact, the basis for the quadraphonic system adopted. It uses logic circuitry to determine the phase relationship of various frequency components of the left and right signal, as well as their amplitude components, and to route them accordingly. Amplitude differences shift left or right, phase differences shift front or back, essentially.

Now, having deduced that logic system as applicable to produce a consistent means of coding and decoding 4 channels into 2, it doesn't matter whether they are being transcribed on disc or not, because it is not the manner of movement that "does it," but the amplitude and phase relationships associated with that manner of movement. So we have a basis for converting stereo multiplex into quadraphonic multiplex. In fact, if the radio f.m. is transmitting a stereodisc that was quadraphonically recorded, the radio won't know the difference.

The decoding system will reconstruct the quadraphonic, regardless of whether there is a stereo f.m. link in the transmission, or not. While man-made systems cannot actually think, a useful device in thinking through ideas for new systems is, "How can it know the difference?" Coded "instructions" built into the system must be unambiguous, that is the point. If they are not, then the system has no means of "knowing the difference."

Now, while all this has been going on, in systems that are essentially analog (using computer technology terms for the moment), meaning that we are dealing with the qualitative shape of waveforms, and their analysis in frequency and phase, tremendous advances have been made in digital technology, which deals with coded position and time.

EXAMINING THE HUMAN HEARING FACULTY

How can this be applied to audio? Many old time audio engineers probably reacted that it was inappropriate, if not impossible. But if they had studied how our hearing faculty functions, maybe they would not be so quick to make that judgment. For the human nervous system transmits impulses with timing differentiations that cannot resolve anything less than a few milliseconds. So how is it we can hear sounds representative of frequencies up into the kilohertz, whose period goes down to less than a tenth of a millisecond?

Human hearing must somehow analyze the content in these frequencies, at the ear, and then code that information in nerve impulses that the human system can handle, for transmission to the brain, where further analysis in the brain's "computer" puts it together, so we can identify the sound that caused it, and probably its direction from our head at the time. If the human hearing faculty can do that, why could not an electronic system do something similar.?

For years, the reason was a matter of complexity: it would take millions of vacuum tubes, and some extremely sophisticated circuitry. Even its logistics would require the Empire State building to house it. But a lot has happened since that was true: solid state, integrated circuits, microcircuitry, large scale integrated circuits (in which "large" does not refer to physical size, but to the amount of circuitry squeezed into a chip that can rest easily on the tip of your little finger). And the precision achieved by such devices is unbelievable. That is the attractive thing.

Here, analog audio designers have been trying to get distortion down to tiny figures, like fractions of a percent that represent much better than 60 dB, and digital devices can calculate mathematical functions, instantly, out to many more decimal places—many times more accurate than the best sliderule, used by a person with fantastic eyesight.

Obviously, the possibility is there, and we have a natural antecedent for it, in our own hearing faculty. So what were we waiting for? Some imagination, to see how to use it. When somebody else has done it, it always seems obvious. But for the first person who does it, it requires imagination, creativity. It does not just happen, because it's obvious. That is perhaps the single most cogent flaw in the evolution theory: it does not answer the question, "Where do ideas come from?"

But even imagination applies principles. We need to follow a problem-to-solution approach. Where does audio most need precision help? Organ makers had one place: tuning the instrument. Electronic dividers could solve one part of that, by deriving lower octaves from the top octave of master oscillators. But tuning the top octave was still a skilled, and time consuming process. And of course, it could "go off." How nice it would be to have an organ that "tuned itself" automatically.

Well, most of today's electronic organs do just that. And the process is digital. Starting with a single master oscillator, operating up in the megahertz region, precision counters derive the whole set of 12 master oscillator frequencies in absolute lockstep with that single master. Musicologists define pitch, which electronics people call frequency, in terms of semitones, each of which is the twelfth part of an octave, and "cents" each of which is one hundredth of a semitone. A musical instrument tuned within a few cents is well tuned.

The preprogrammed counter, that produces a set of 12 master tones from that single master frequency in kilohertz, tunes the whole octave within a few cents—far closer than most tuners could do it, whatever they might like to think! And it does it instantly—it is, in fact, built in. And the tuning of the whole organ can be changed, still in tune, merely by changing the frequency of that one master oscillator. How's that for a hard job made easy? And there's more. That's just one application. We'll pursue others another time. ■

db Calendar

DECEMBER

11-14 **International Entertainment Exposition.** Las Vegas Convention Center, Las Vegas, Nevada. For more information contact: American Expositions, Inc., One Lincoln Plaza, New York, NY 10023. (212) 691-5454.

JANUARY

5-8 **1980 International Winter Consumer Electronics Show.** Las Vegas, Nevada. Las Vegas Convention Center, Jockey Club Hotel, and the Grand Ballroom of the Las Vegas Hilton. For Information contact: William Glasgow, Show Manager, Consumer Electronics Shows, Two Illinois Center Suite 1607, 233 N. Michigan, Chicago, IL 60601. (312) 861-1040

FEBRUARY

1-2 **The 14th Annual Television Conference of the Society of Motion Picture and Television Engineers (SMPTE).** Toronto, Canada. Sheraton Centre Hotel. For more information contact: SMPTE TV Conference, 862 Scarsdale Avenue, Scarsdale, NY 10583

25-28 **AES 65th Convention** (London). London Hilton and Park Lane Hotels. For more information contact: Audio Engineering Society, Inc., 60 East 42nd St., New York, NY 10017.

26-28 **"Sound 80"** Cunard Hotel, Hammersmith, London.

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MARTIN DICKSTEIN

db Sound With Images

Experience Is The Best Teacher

• Although I had planned to do a completely different type of column this month, I can still find situations from my travel experiences that could be helpful to **db** readers. Sometimes, nice things happen and I try to relate them, so they might be of some help. Sometimes, just the opposite is the case, and I tell of these incidents so that they can be avoided. This time there are two trips to discuss.

The first was to a hotel in the Southwest. The client was a large manufacturer, and the occasion was a yearly sales meeting for almost 200 regional managers. Arrangements for the show were started several months before the meeting date. The show was to consist of a live portion, on stage, between speeches, on each of the three days of the conference, with a multi-image slide/film show on a rear screen, which was a single stretch material 22′ wide. The slide presentation was to be programmed and projected on 3 screens, that is, 3 side-by-side images.

When the client asked me to assist them on this project, they told me that they would design the stage-set themselves and have it put up by a local contractor with whom they had already been in contact. The set-up was to consist of the rear screen in the center of the stage (with space for projection equipment behind) and enough room in front for the live acts. The local contractor was to put up drapes on both sides of the screen, extending out beyond the ends of the stage to carry out the motif of the program. Symbols of the motif were to be hung on the drapes with the client's art department designing and executing all the art work. The contractor was also hired to set up the stands needed to raise the projectors to the level of the center of the screen. I submitted a drawing to help the contractor in this work. (At first, I suggested to the client that it could probably be done more easily by the people at the hotel, but they insisted that the contractor would construct something special according to the client's design in keeping with my requirements.)

PRIOR ARRANGEMENTS

During my discussions with the hotel on the show's audio-visual requirements, several months prior to the date of the meetings, I was told that they only had a limited amount of equipment, that the ceiling-speaker sound system was poor and they were in the process of installing a better system, and that my best bet would be to work with an outside audio-visual contractor for the necessary equipment. I knew one of the AV rental companies in the area from a previous successful presentation and arranged with them to have the necessary equipment delivered and set up, and explained the situation with the drape contractor.

The client's art director was to arrive a day before anyone else to be sure the stage was set up according to his design. I was to arrive the following day, and the arrangement was that the screen, drapes, and equipment would be in place ready for me to set up the soft-ware, and ready for me to test all systems.

When I arrived, the AV contractor had set up the screen according to my specifications, but the drapes and risers and the local contractor responsible for these were nowhere in sight. The AV contractor had waited for my arrival, but since there was nowhere to set up the equipment, this could not be done. The client insisted that his contractor would be there shortly, but this didn't happen. Finally, after having waited for several hours in vain, the AV contractor asked to be relieved of that part of his contract and left, leaving all the necessary hardware. It wasn't until several hours later that the draping contractor finally arrived and started to cut material for the drapes, with no thought of setting up the risers. With the help of the client, we got the contractor to start setting up a place for the equipment, leaving the drapes to be done later.

COMPLICATIONS SET IN

The contractor started by setting up a massive rigging of heavy steel pipes, and putting rough boards on top for the equipment. The level of the boards turned out to be almost 9" too high (considering that the projectors would have to go on top with slide projectors stacked 3 high, and the film projector having a lens height approximately 9" above its base). When told of this, the contractor said his steel piping would only allow a lowering of about 6", and that the screen would have to be raised. They refused to consider any other solution, as they had a contract and they were going to finish the job. Fortunately, we could raise the screen, and this seemed the easy way out not to waste any more time. We were already about 9 hours behind schedule.

TIME & COST

When I told the client, whose art director stayed with us all the way through the ordeal, that it could have been done in a small fraction of the time with the AV contractor and the help of hotel personnel; he admitted his error, and he realized, even more, how wrong the whole idea was when he saw the "special" set-up that was rigged. The drapes were not put up completely until several hours after the rigging, incidentally, so the total cost for the complete job, if paid for by the hour, must have been enormous.

LESSONS TO BE LEARNED

The lesson here is that it could save an awful lot of time and money to just have a simple set-up of platforms and stacked tables, usually available in-house at most hotels.

The second trip was to a large city on the east coast. The meeting was to take place in a large conference room with rear screen projection. The technician at the site told me on the phone, in a conversation well in advance of the presentation date, that all the equipment I needed would be available, including a remote slide control for the speakers at the podium. It was a simple type of presentation with a single slide projector and a 16mm film projector, as well as a ¼-inch tape. When I arrived, I found that he was right. The equipment was there. However, he had never had a presentation before in which film and slides were mixed. It was necessary to turn a large front-surface mirror to show slides, then film. Both projectors were permanently set and it was the mirror setting that determined which projector would hit the screen. This meant that there was a pause of almost five seconds or more, depending on how smoothly he could turn the mirror from one position to the other and get it locked-in with as little wobble as possible.

Another annoying condition was that the sound system, with the tape recorder, was in another room. This meant that he would have to go from the projection booth to a control room to start the tape, and then come back when it was necessary to start a film (after turning the mirror).

And to top it all off, the projector was a random-access type with two drums using their special slide mounts, and the technician did not know how to load the slides. It also turned out there were other presenters whose presentations also included slides, and they did not know how to load the slides either. Since the projector was the type which used an internal mirror (in addition to the large one), the slides in one drum had to be loaded normally (as for front projection, because of the double reflection), and the other had to be mounted for rear projection (using only the single large mirror to reverse the image). I finally had to load all the slides, using my own blanks to separate the speeches as no one had their own blanks—including the technician.

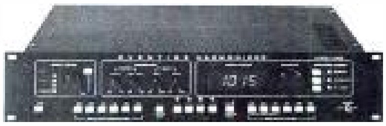
An interesting lesson. Know as much as you can about the different types of equipment, get as much information beforehand including types and model numbers so you can be fully prepared for anything, and keep your cool in spite of all the nervousness around you. ■

db New Products & Services

HARMONIZER

• Capable of changing the pitch of an input signal by three octaves (one up, two down), the model H949 Harmonizer has two outputs, each with 400 ms. of delay. In addition to pitch change and delay, other effects include flanging, repeat, random delay (for automatic double-tracking), and a new effect called "reverse." Incorporating two different algorithms to handle the pitch change 'glitches', the user can select whichever is optimum for the program material. Delay is selected via incremental push-buttons; while pitch change is controlled either by a knob (manual mode) or by the HK940 keyboard, which varies the pitch in discrete musical steps. Switchable for 115 or 230 volts, the H949 offers a frequency response of 20 Hz to 15 kHz, ±1 dB, and a dynamic range greater than 96 dB.

Mfr: Eventide Clockworks, Inc.
Price: $2,400.00
Circle 60 on Reader Service Card



VOCODER

• A sixteen channel vocoder, the model 7702 features 16 analyzer and 16 synthesizer channels in the vocoding range, extending from 50 Hz to 5080 Hz. In addition, the performance of the model 7702 is enhanced by a high frequency channel which extends from 5080 Hz to 15 kHz, and covers the region of "s" sounds and explosive consonants. The analyzer outputs and synthesizer inputs are accessible through phone jacks. Cross patches for sound scrambling can be made and activated, or de-activated, by manual or foot control. The unit employs a built-in voiced/unvoiced sound selector, a noise generator for the "s" sounds, a hiss/buzz balance control, various mode switches, overload indicators for voice and carrier inputs, and a Mic-Line switch for the voice input. A sample/hold control holds entered vowels when desired. Maximum output level is +15 dBm and the signal-to-noise ratio is better than 70 dB. Input impedances are 20k ohms minimum, while output impedance is 600 ohms.

Mfr: Bode Sound Co.
Price: $5,600.00
Circle 61 on Reader Service Card

CASSETTE DUPLICATOR

• Employing vacuum tape chambers for safe, high speed tape handling, the Automated Cassette Duplicator (ACD) system uses a reel-to-reel bi-directional master (available in ¼-, ½-, and one-inch formats) and a line of Cassette Slave Loaders. The system is capable of reproduction rates of 32 to 64 times normal speed with frequency response to 15,000 kHz. An attachment for the Cassette Slave Loader, the Automatic Cassette Feeder (ACF) automatically performs the tasks of cassette insertion, leader extraction and threading of the leader into the splicing platform. Then, the automatic operation of the Cassette Slave Loader takes over to cut the leader, splice the tape to the leader and load the cassette with tape while recording the signals from the master. When duplication is finished, it cuts the tape and splices the leader at the end of the program material, and winds the remaining leader into the cassette, ejecting the finished product from the loader.

Mfr: Recortec
Circle 62 on Reader Service Card

PASSIVE EQUALIZER

• Boasting no insertion loss, the model 4320 passive equalizer features 27 one-third octave filters on I.S.O. centers from 40 Hz to 16 kHz. The unit provides filter attenuation up to 10 dB, an EQ in out switch, and two accessory octal sockets on the rear panel—one for a bi-amp or tri-amp crossover network; the other for the insertion of a response shaping filter.

Mfr: White Instruments, Inc.
Price: $550.00
Circle 63 on Reader Service Card

The logic behind the Revox B77.

The logic is the logic which is built-in.

It's an ingenious and highly sophisticated system—much like the human nervous system—which controls the deck's functions.

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The logic also permits full-function remote control, and an editing mode that keeps the playback circuitry live, even when the motors are stopped. You can make your splices right on-the-beat, and our built-in splicing block makes it easy.

The design and construction of the Revox B77 further guarantee smooth and accurate operation. To get the long-life advantage of ferrite without static build-up or heat degradation, we use Revox's exclusive Revodur heads, made of metal to dispel heat and static, and vacuum-coated with permalloy for durability.

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Revox offers many options with the B77 including a full range of speed configurations from 15/16 IPS to 15 IPS, variable speed control, ¼ track record/playback and more.

All this professional quality is neatly engineered to fit in a deck you can carry. After all, if you own a machine this good, it's logical to take it with you.

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Circle 33 on Reader Service Card

The Last Word on Microphones

IF YOU'RE LOOKING for the last word on microphones, we'd like to pass on the following suggestion: Keep on looking, for you won't find it in this issue of db. Or in any other issue, for that matter.

No we are not awaiting some major breakthrough in microphone technology. As far as we know, there are no immediate plans for digital microphones or automated polar responses. Of course, microphones are getting more sophisticated, but the basic designs don't seem to be changing at the rate one finds in consoles and signal processing devices.

While yesterday's console may be getting "old-fashioned," chances are yesterday's microphone is still as good as new, and you wouldn't dream of trading it in. But, we're not yet ready to write the final chapter on the art of the microphone. And probably, never will be.

There's still no "right" way to mike the drums, no "correct" equalization for the rhythm guitar pickup, and no answer to "What do I use on the (you-name-it)". However, there is one immutable rule that we can pass on to you. It's Edsel Murphy's closed-loop theorem of microphone movement. (Remember, you read it here first.)

1. Place any microphone at any distance from an instrument.
2. Describe your set-up to ten engineers.
3. Receive ten explanations of what's wrong.
4. Follow the recommendations of any one engineer.
5. Go to line 2, and repeat, until you get the message.

Now that we've all gotten the message, let's move on to some words that may help you to better cope with the Murphy's-loop syndrome.

We start off with some applications-oriented features. For example, what about a nice little heater for your favorite old condenser? Nope, this is not meant to keep it cozy on those long winter nights. It's to keep it from making impolite little noises at the wrong time. (Or, as Murphy tells us, "Microphone noise level is inversely, and perversely, proportional to signal strength.") For the latest generation of condenser microphones, there's probably no need to bother, but everyone has some older favorites, and sometimes these get a little finicky whenever the THI changes too quickly. Bob Katz' heater project may be just the thing to keep these mics behaving themselves.

Of course, you'll need good microphone cables, and there's no shortage of cable testers around these days. However, the tinkerer may want a little something extra, and Kirk Elliott's CMOS cable tester is surely just that. And, be sure to check out our directory of microphone manufacturers immediately following Kirk Elliott's article.

In his era of the twenty-four track recording studio, it's important to remember that there is a world of difference between stereophonic sound and multi-track mono. If your last extravaganza lacked a little dimension, perhaps you're overlooking some of the advantages of stereo miking. Obviously, it's not right for every session, but how about trying it out on the next string overdub? Bruce Bartlett describes how to begin.

Of course, you can't stereo-mike the strings on an ELP live gig. In fact, you just might not be able to hear them at all if it were not for Arnie Lazarus and the FRAP. When it's necessary to "go direct," and the instrument is acoustic, the FRAP Point-source-microphone may be worth its weight in gold—and you know how much gold costs these days.

Well, it seems we're just getting started on the subject of microphones, and here we've run out of room this month. So, we'll continue this next month, with a few more features, but still, no last words on the subject. ■

Eight good reasons to be a Beyer Buyer.

one The first reason is Beyer. We have fifty years experience making the world's finest microphones and headphones. And an unmatched reputation for quality, reliability and innovation. The choice of professionals everywhere.

two M 160. One of the world's best-loved and most versatile microphones. Warm, soft sound favored by vocalists and musicians alike. Dual ribbon design for high strength and fast transient response.

three Beyer headphones. A full range of high quality professional models for critical monitoring and reliable communication. DT 109 combines stereo headphones and boom-mounted microphone, ideal for on-air use and disco deejays. DT 444S wireless headphone receives sound from an infra-red LED transmitter up to 300 feet away. Full 20-20,000Hz frequency response. Six hour stereo operation on rechargeable NiCad batteries.

four The new M 400. A great performer's mic. Supercardioid pick-up pattern to minimize feedback. Rugged design for long life. Tapered frequency response with rising high end and rolled off lows, plus midrange presence boost. Built-in humbucking coil and pop filter. Dynamic design is unaffected by heat and humidity.

five Beyer microphone stands and booms. A full range of mic mounts for floor and desk use, with fixed and folding bases. Available with collapsible tubes for easy packing. Also heavy-duty stands for speaker cabinets.

six Beyer microphone accessories. Wind screens, impedance matching transformers, in-line switches, power supplies, wireless transmitters, stereo arms, goosenecks, clamps, thread adapters, anti-shock suspensions, and even a mic stand ashtray! The whole works. If you can use it with a mic, we make it.

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

eight See your dealer or write for information on our product line. You'll have many more reasons to be a Beyer buyer.

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BOB KATZ

Build a Heater for a Condenser Microphone

High humidity situations can often cause havoc with some of the older condenser microphone models; here-in are plans for a built-in heater for your condenser microphones.

IN MANY CASES, high humidity is no problem for condenser microphones. In fact, some manufacturers claim their microphones will perform in up to 99 per cent humidity. However, a high-humidity environment, coupled with an ambient temperature rise, can create precipitation problems within condenser microphones, especially older models, with stainless-steel diaphragms. Television recording engineers often encounter this problem in large, air-conditioned, heavily-lit studios. During breaks between rehearsals and takes, the lighting is often shut off, to allow the studio to cool. Before shooting, the studio may be at 60 degrees Fahrenheit, but by the time of the first break, the temperature may have risen to 75 or even 80 degrees. In conditions such as these, the microphone diaphragm can be compared to an ice-filled lemonade glass on a summer's day.

EARLY SOLUTIONS

The tv recording engineer soon learns that when his condenser diaphragm begins to sweat, he can expect to hear some bad sound. The first symptom is often an increase in the low frequency noise level of the microphone, with "shotgun bursts" of noise, resembling the sound of a constant thunderstorm. The output signal-level begins to drop and the microphone soon becomes unusable. Older, high-voltage tube microphones may even arc in high humidity.

Bob Katz is an instructor and lab supervisor at the Institute of Audio Research in New York City and a free-lance recording engineer. Initial technical help and materials were supplied by Steve Washburn, an audio and computer design engineer with Mantra Sound, Glastonbury, CT. The Nichrome wire modification described in this article has been successfully built and used by Ray Rayburn of Rayburn Electronics, New York City.

Of course, the engineer can run out to the studio (hopefully between takes) and shake the microphone like a fever thermometer. This gets the musicians' attention, and sometimes even cures the problem. A somewhat-more elegant solution is to borrow a 500-watt light, and heat the microphone over it for about ten minutes. As long as the microphone does not become too hot to touch, it is probably safe to dry it over a lamp. (One manufacturer specifies the integrity of its capsules and preamps to 60 degrees C., which is 140 degrees F.)

I used to dry my "waterlogged" microphones overnight in a dessicator until I found I could dry them with heat. I also practice two additional preventive measures:

1. Warm a microphone with body heat before bringing it from a cool control room to a hot studio. (Producers may look aghast at this practice, fearing that the engineer is becoming too attached to his equipment.)

2. Very carefully clean the diaphragm, since dust and other pollutants can cause centers where moisture will stick and take longer to evaporate. The cleaning process is not difficult; however, do *not* attempt to clean the diaphragm unless you feel confident to take the capsule apart. Even if the microphone is not under warranty, it might be preferable to send it to the manufacturer for cleaning and calibration at regular intervals.

If you wish to clean the microphone yourself, and can gain access to the diaphragm with confidence, open the capsule in a clean, dust-free environment. Obtain absolutely pure-grain alcohol. Do *not* use freon, because its low boiling point may cool the diaphragm, causing moisture to re-condense. Do *not* use any brand of rubbing alcohol, because these contain glycerine or lanolin.

Take a clean, soft camel's hair brush. Dip just the tip of the brush in the alcohol and apply it to one edge of the circular (usually) diaphragm. Gently "paint" from one side to the other. In the process you will sweep dust to the other side and off the diaphragm, as well as dissolve minor deposits that may also be there.

To keep dirt and deposits from building up on microphone

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DC-73 CARDIOID CONDENSER MICROPHONE

Cardioid condenser microphone for 48 volt Symsi powering. Features an integral electrical "pop" filter which has no effect on high frequency response, and a built-in shock resistant elastic suspension to reduce hand noise to a minimum. A large diameter circular condenser element provides full natural sound in highly rugged package with a steel mesh protective grille. A two position slide switch on the case permits selection of either flat response or 100 Hz high-pass for vocal work.

Excellent hand-held vocal microphone. Also for brass or percussion where cardioid pattern is demanded. Studio, broadcast and especially recommended for live performance. $295

DC-96 CARDIOID CONDENSER MICROPHONE

The DC-96 is a cardioid condenser microphone for 48 volt Symsi powering. The DC-96 employs the unique rectangular dual membrane capsule similar to the DC-63 and ST-8 Stereo microphones with FET pre-amplifier. Features: very low noise, high output level and extremely smooth frequency response both on and off-axis. The small size of the DC-96 further enhances its use where larger mikes become obtrusive.

For all live performances and studio work where a high quality cardioid microphone pickup is required; for both close and distant pickup. $465

VM-40 (Omni) and VM-41 (Cardioid) CONDENSER MICROPHONES

The VM-40 (Omni-directional) and VM-41 (Cardioid) condenser microphones both feature a 1.5 cm diameter circular condenser element. Small in physical size and very rugged, they operate on 48 volts Symsi power. A four position ring switch is incorporated to permit selection of either full range frequency response or 100 Hz high-pass and for each position there is a 10 dB pad.

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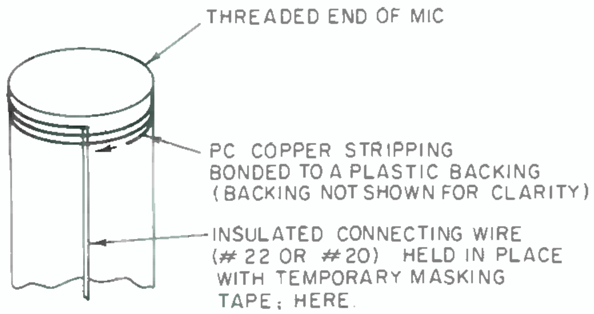

Figure 1. PC copper stripping applied to the microphone shell.

diaphragms, be sure to follow Lou Burroughs' recommendation: put windscreens on *all* vocal microphones.

A NEW SOLUTION

A humidity problem during an August recording session prompted me to consider a more-permanent solution for my microphones. This solution should work with almost any make of half- to one-inch cylindrical microphones. The session was the summer "Music Mountain" concert series in Falls Village, Connecticut. The non-airconditioned concert hall is a wooden building with many screen doors, open to the summer heat. The inside stays cool all morning, and into the early afternoon, but within minutes after the audience files in on a hot and muggy day, the hall temperature and humidity rise tremendously. That's when cool musicians, cool engineers, and cool microphones begin to sweat simultaneously.

One weekend, a microphone failed due to the temperature change. As I segued to my backup microphones, I began to think about a built-in heating element. This would function like the heater incorporated into certain calibration microphones. After a little experimentation, the following "field modification" was devised, and seems to work quite well.

RECIPE FOR A HEATING ELEMENT

There are two methods which have been proven successful. Material availability will probably determine which you use.

One method uses very-thin nichrome wire, encased in very-thin shrink tubing. The tubing electrically insulates one turn of nichrome from the next, and from the microphone body. The other method uses flat 1/16-inch wide PC copper stripping bonded to a thin, transparent backing. The plastic backing provides insulation as above, and (when the paper backing is removed) also has an adhesive which sticks to the microphone body. Described below is the PC stripping method. The nichrome wire method is easily derived from this.

Ingredients: Copper PC stripping, 1/16-inch wide and 12 inches long, with adhesive plastic backing; four feet of small-gauge insulated wire; one-inch-diameter heat shrink tubing.

1. Remove the capsule from the microphone case and place it in a safe location.

2. Remove the preamplifier from inside the microphone shell, and place it in a safe location. (This guarantees neither capsule nor electronics will be damaged by the heat of heat-shrinking.)

3. Cut two 2-foot long pieces of insulated stranded wire; strip back one end of each wire by 1/16th of an inch, and tin it.

4. Trim the plastic on both sides of the copper stripping to approximately 1/16th of an inch beyond the stripping.

5. Remove only an inch of backing from the plastic, exposing some of the adhesive, then stick it to the end of the microphone shell. (See FIGURE 1) You are beginning what is to become a fine-pitched spiral of copper ribbon that will go around the preamp cylinder, from the capsule end to less than an inch below this.

6. Using a razor blade, slice under the end of the copper ribbon and lift about a ¼-inch of it from the plastic. Tin this copper piece. Lay the previously-tinned wire along the length of the cylinder so that its tinned end lines up with the end of the copper. A temporary piece of masking tape further down the shell will hold the wire in place. Fold the tinned copper back over the wire and solder, *without melting the plastic underneath.*

7. Continue the spiral neatly around the microphone. The lines of copper should be as close to one another as possible, so as to concentrate the heating effect in the area nearest the capsule. With each revolution, the stripping will go over the connecting wire and hold it in place. When you reach the end of the strip, cut the copper on an even revolution so that the second connecting wire will line up with the first. Masking-tape the second two-foot stripped and tinned wire to the microphone, locating it to line up at the bottom of the spiral. Solder it to the copper stripping as above. (If using Nichrome wire, silver solder and a torch must be used.)

8. Check for a complete circuit with an ohmmeter. A DVM will show almost 0 ohms (0.1 to 0.2 ohm). Also, at this time check that there are no shorts from the copper to the metal shell of the microphone.

9. Cut a piece of one-inch diameter heat shrink tubing, (transparent is nice because you can see the fruits of your work) a quarter-inch or so longer than the spiral on the microphone. When heated, the tubing will shrink or curl a little at the edges, so account for that as you shrink it down. (a gas flame stove will do the trick.)

10. Remove the tape holding the connecting wires in place and tie a knot just where they come out from the tubing. This will add mechanical strength to the second connecting wire, which is not covered by the full width of the tubing.

The SP-15 has two things our best turntable doesn't have: Quartz-locked pitch control and a lower price.

You know what made the SP-10 MK2 our best turntable, and why so many radio stations use it: Wow and Flutter of 0.025% WRMS. Rumble of −78dB (DIN B). Speed accuracy within an astonishing 0.002%. And amazingly high torque for a start-up time of 0.25 second.

Yet for $300 less,* the SP-15 has exactly the same high degree of speed accuracy, the same wow and flutter and the same rumble as the SP-10 MK2 while delivering an incredible start-up time of 0.4 second.

Technics quartz-locked pitch control is pretty incredible, too. Unlike the pitch control in many other turntables, it lets you vary the speed with the unvarying accuracy of quartz. In precise 0.1% steps above or below any of the three standard speeds up to a maximum of ±9.9%. What's more, the exact speed variation you choose is shown right up front in bright digital display. And with Technics you can lock the pitch at the pitch you choose.

Another reason you'll choose the Technics SP-15 is durability. It has an electronic brake that can stop the platter in 0.4 second, even though a tracking force of 2.2 lbs. (or the weight of 250 tonearms tracking at 2 grams) can't begin to slow the platter down. And to help minimize acoustic feedback, it has a heavy-duty aluminum diecast chassis plus a double-damped platter. And when you add the optional SH-15B2 base (shown with SP-15) you'll get the extra protection needed to cope with high volume levels.

There's also Technics SP-25, a two-speed version. With the same accuracy, quartz-locked pitch control (±6%) and many of the great features of the SP-15.

The SP-15 with quartz-locked pitch control. It has the same phenomenal performance as the Technics turntables many FM stations use and discos abuse: MOTOR: Quartz-locked DC direct drive. SPEED: 33⅓, 45 and 78 RPM. STARTING TORQUE: 3.0 kg·cm. START-UP TIME: 0.4 sec. (90° rotation at 33⅓ RPM). WOW AND FLUTTER: 0.025% WRMS. RUMBLE: −78dB (DIN B). PITCH ADJUSTMENT RANGE: ± 9.9%.

The SP-15. We added quartz-locked pitch control, we subtracted from the price.

*Based on Technics recommended price for SP-10 MK2 and SP-15 (excluding bases).

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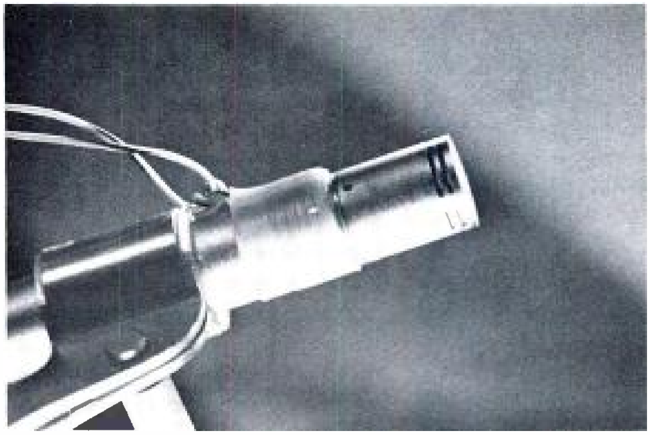


Figure 2. A condenser microphone outfitted with heater.

11. Small-diameter heat shrink tubing around the two connecting wires will keep them together. I cut them to one-foot length and put banana plugs on the ends. An XLR-to-banana adapter allows using standard microphone cable for the d.c. feed to the heating element. (The sex of the adapter is reversed to avoid "deadly error.") Return the preamp to the shell and screw on the capsule. Your microphone with heating element is now complete. Steps 1-11 take about 20 minutes.

12. Next step is to build a well-filtered, variable d.c. power supply. The simplest supply uses a step-down transformer to 12 volts or 24 volts maximum. This feeds a variable autoformer with a similar maximum voltage rating. The autoformer will enable you to "dial-up" the voltage necessary to produce the current to heat the coil. Ultimately, the voltage across each coil may end up being as little as 2 to 5 volts, but you must be able to deliver more into the feed cable(s) to overcome losses. Full wave rectification, substantial filtering (40,000 to 48,000 *uf* of filter capacitance minimizes hum induction at high currents), and an output ampere meter complete the supply.

USING THE HEATER AND POWER SUPPLY

The heating effect is reasonably proportional to the square of the current. Therefore the ampere meter on the power supply provides a monitor on the microphone temperature without a fancy thermostatic control. If you use nichrome wire, about 1 to 1½ amps of current will make the microphone feel warm to the touch (between 90 and 105 degrees F.) in a normal temperature room. As much as 4 to 5 amps may be needed if you use the PC stripping.

Start with 1 ampere and gradually increase to find the proper current temperature. Experiment to find the correlation between the ampere meter and the microphone temperature. After each adjustment, wait five minutes before increasing the current again, as the microphone takes time to come up to temperature. Once you have found the current-level for your microphones, make a note of it, and simply set the meter when you turn the supply on. Varying cable lengths will contribute to voltage losses but the required current will be approximately the same.

If you wish, four microphones can be put in series parallel. This will double the current requirement at the supply. If you do not use a thermostat, you should use equal lengths and equal type cable to each parallel leg, to guarantee equal heating.

The heated-microphones are used as before, except an extra cable goes up to each. Of course, you can no longer slide-on a microphone holder from the front. If your microphone does not permit sliding a holder on from the back, or snapping it onto the body, then you will have to use a clothes-pin type holder instead of the original slip-on type.

Using microphones with heaters will not guarantee that "warm" sound you are looking for, but will certainly help worried engineers and producers to keep cool. ■

Technics RS-M85 MK2 with metal tape. We pushed performance to a new high. But kept the old price.*

Last year you could get the precision of direct drive and the unparalleled accuracy of quartz with Technics RS-M85. The cassette deck *Audio* magazine (June '79) said "had the best tape speed characteristics ever measured in a cassette deck." This year you can get that same accuracy with the RS-M85 MK2. Along with the additional benefits of metal tape. Yet we didn't add a cent to the price.

What we did add is more dynamic range, a wider frequency response and sendust-formulation heads that easily handle the difficult jobs of recording and erasing metal tape.

One more difficult job the RS-M85 MK2 easily handles is keeping wow and flutter down to a microscopic 0.035% while maintaining excellent speed accuracy. But that's not surprising. At least not with Technics quartz-locked direct drive. This servo system compares the rotation of our direct-drive motor with the unwavering frequency of a quartz oscillator, and instantly applies corrective torque whenever the slightest speed deviation is detected.

Another one of the RS-M85 MK2's bright spots is its two-colored fluorescent (FL) bar-graph meters. A device attack time of just 5 millionths of a second proves they're fast. While no more than 0.1dB deviation from the 0 VU level proves they're accurate. And that's proof enough.

Still, the RS-M85 MK2 has even more: Like a separate, coreless DC motor for reel drive. Dolby† NR. A low-noise, highly linear amplifier section. Full IC logic controls. A 3-position bias/EQ selector with bias fine adjustment. And an optional full-function infrared wireless remote control (RP-070).

Technics RS-M85 MK2. We pushed the performance up. Not the price.

FREQ. RESP. (Metal): 20-20,000 Hz. WOW AND FLUTTER: 0.035% WRMS. S/N RATIO (Dolby in): 69 dB. SPEED DEVIATION: No more than 0.3%.

*Based on Technics recommended price for RS-M85 and RS-M85 MK2.
†Dolby is a trademark of Dolby Laboratories.

Technics
Professional Series

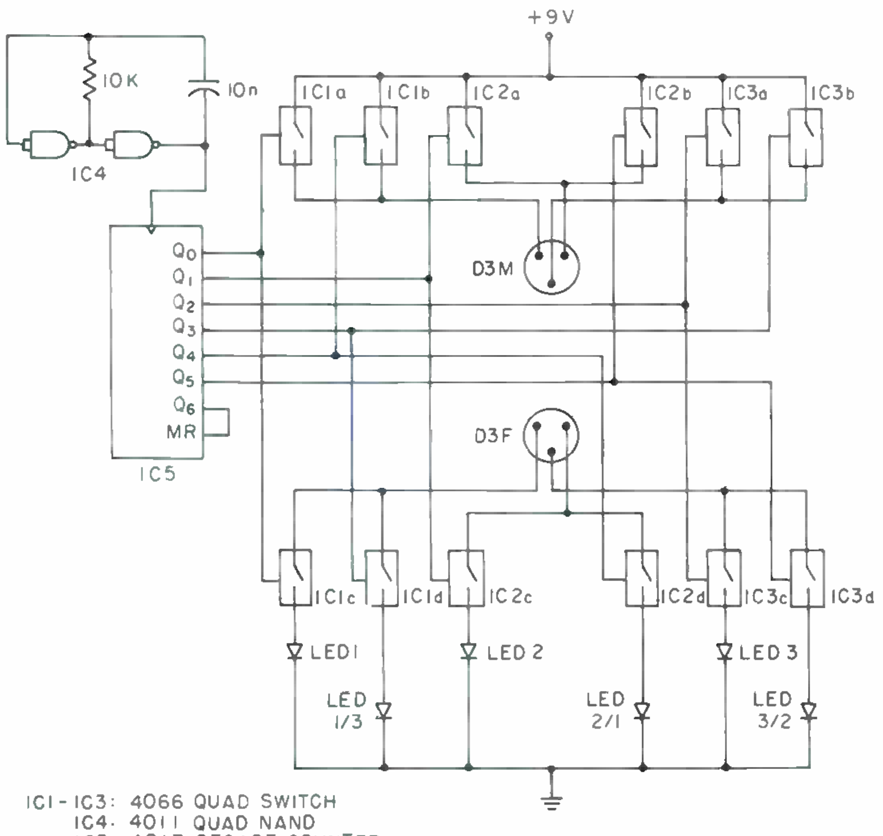
KIRK ELLIOTT

CMOS Microphone Cable Tester

Leds indicate microphone cable's status.

AUDIO CONNECTORS and cables are among the elements most likely to fail in a professional sound system. The constant use and abuse to which they are subjected, both in the recording studio and on the road, make them especially susceptible to damage. Moreover, the high pressure circumstances of recording sessions and concerts necessitate the speedy identification of any technical problems that may arise.

(Continued on page 30)



IC1 - IC3: 4066 QUAD SWITCH
IC4: 4011 QUAD NAND
IC5: 4017 DECADE COUNTER

Figure 1. Schematic diagram for the microphone cable tester.

Kirk Elliott is employed as a technician at Octopus Audio, Toronto, Canada.

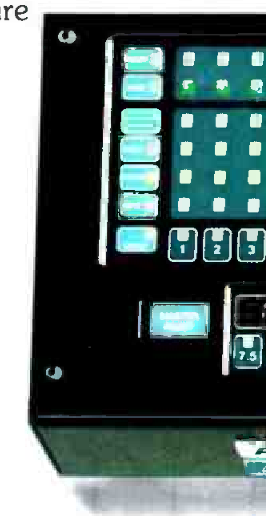
Circle 31 on Reader Service Card

Introducing a present

Once you go through a recording session with the new ATR-124 24-channel recorder by Ampex, you'll want to go through another. Because with each new session you'll discover something new you can do. Things that you can only do with a recorder that's full of features of the future.

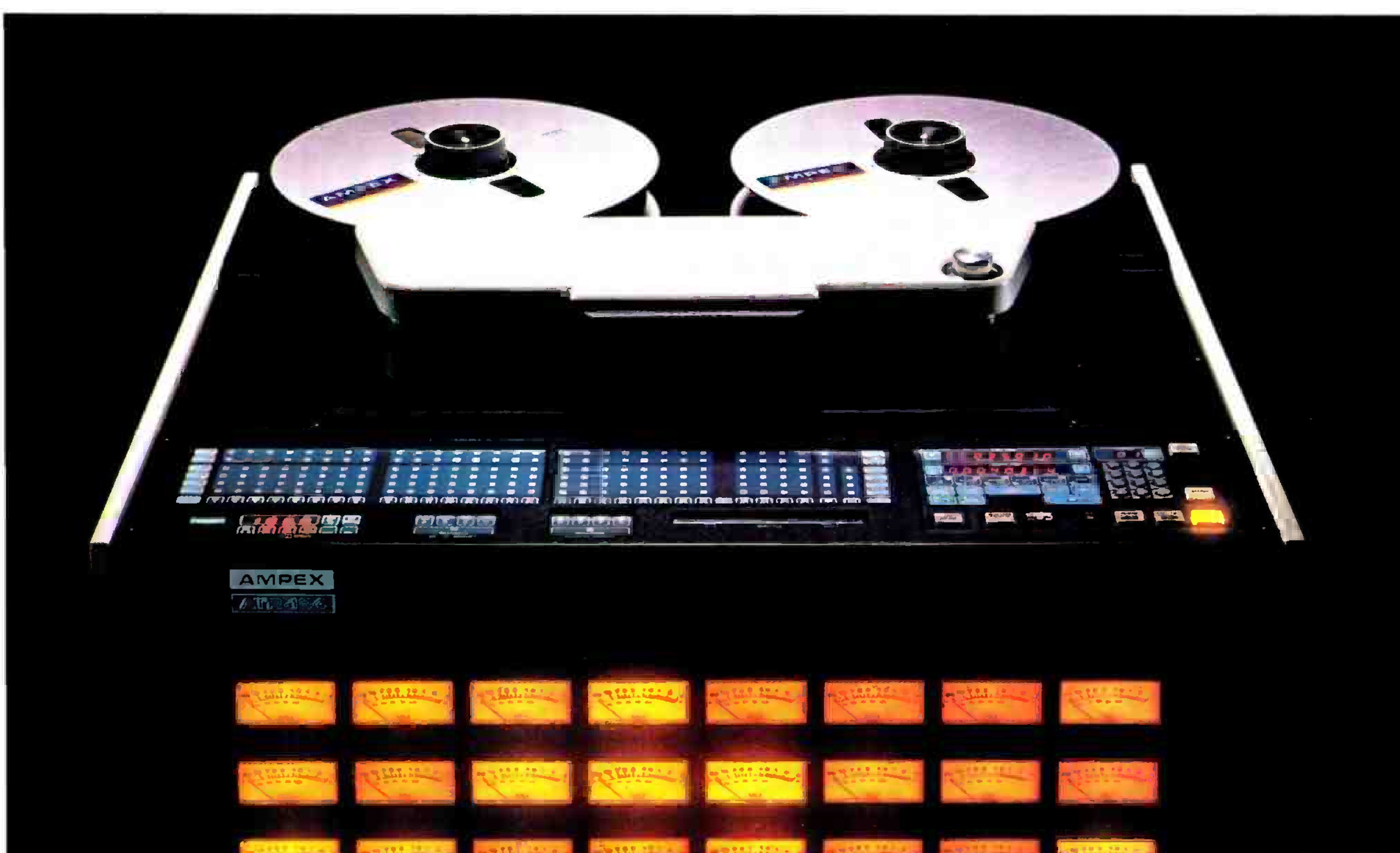
ATR-124 gives you the unheard of: Time on your hands.
Which means you can use that time to give clients more of what they're paying for—your creative skills. With the ATR-124 microprocessor-based control system, you can pre-program what you want to do ahead of time so you won't waste studio time setting things up. When their time starts, you're ready to record by touching a single recall button.

ATR-124 also lets you duplicate a technique you may have used earlier in the session without having to rethink what you did. Just touch the memory button and it'll all come back to you. ATR-124 lets you rehearse what you've got in mind, without recording it, to make sure what you've got in mind is right. Tape can be manipulated faster which means you'll get the sound you want sooner. And the chance to try something "a little different." All because of the speed and accuracy that ATR-124 puts at your fingertips.



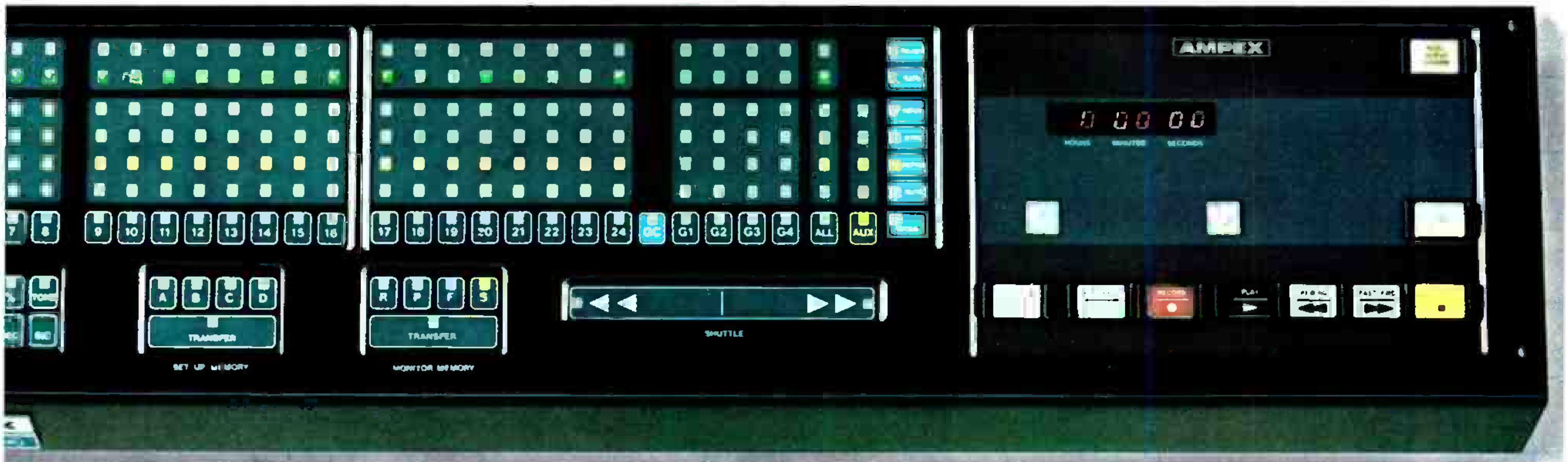
ATR-124 doesn't take away your creativity, it adds to it.
The less time spent setting up, correcting, and redoing, the more time spent creating. And when you add features that help you create to the ones that help you save time, you've got one very potent piece of audio machinery. Take the control panel for instance. It's like nothing you've ever seen. Pushpads linked to a microprocessor give you a new level of creative flexibility. Program a setup, then change it. Then change it back, all with a single fingertip.

A repeatable, variable speed oscillator for pitch correction and special effects is built in. In addition



from the future: ATR-124.

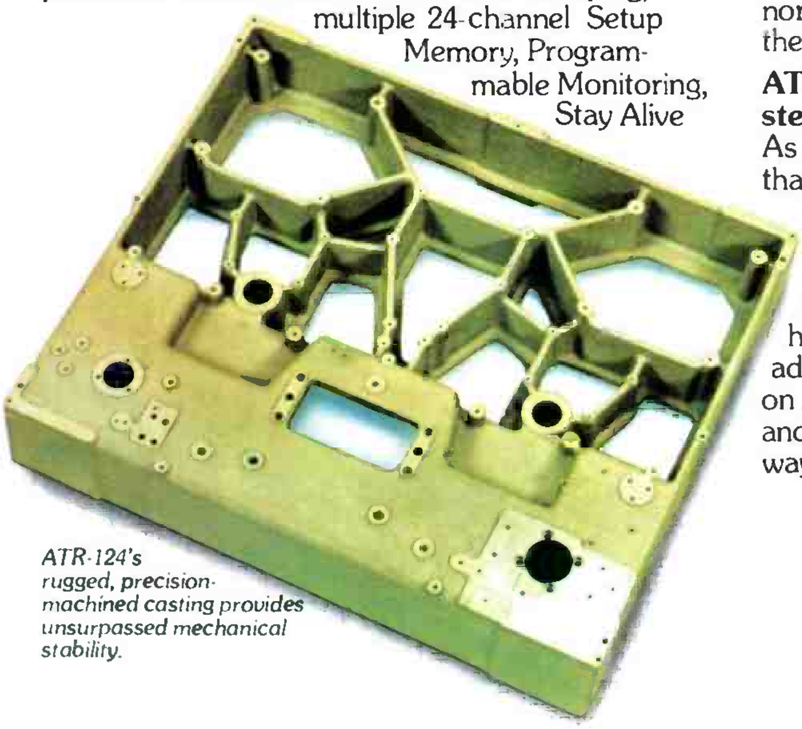
to the standard output, there is an optional auxiliary output with each channel that enhances flexibility. So don't think that ATR-124 is going to



ATR-124's Control Panel. Speed and accuracy at your fingertips.

replace anything that you do. On the contrary, it's going to improve the skills you have, if not help you develop some new ones.

ATR-124 picks up where ATR-100 leaves off.
It's only natural that the people who brought you the ATR-100 should be the ones to bring you something better. ATR-124 offers you 24 channels instead of 4. You also get many new and exclusive features. The kind that have set Ampex apart from the crowd for the last 30 years. Features like balanced, transformerless inputs and outputs; a patented flux gate record head; 16" reel capability; input and output signal bus for setup alignment; membrane switch setup panel; fingertip-operated shuttle speed control; and microprocessor-based synthesized Varispeed -50% to +200% in .1% steps or in ¼ tone steps. ATR-124 also features microprocessor-based control of Channel Grouping, multiple 24-channel Setup Memory, Programmable Monitoring, Stay Alive Memory, and Record Mode diagnostics. The point is this: If you like the ATR-100, you're going to love working with the ATR-124.



ATR-124's rugged, precision-machined casting provides unsurpassed mechanical stability.

ATR-124 options.
As impressive as the ATR-124 itself.
With the addition of a built-in Multi-Point Search-To-Cue (MPSTC), you can rehearse edits and control five tape-time actuated events and be compatible with SMPTE time code. Separately controlled auxiliary output amplifiers with each channel provide simultaneous monitoring of normal and sync playback as well as all other monitoring modes. A roll-around remote control unit can also be added to the ATR-124 which contains all control features normally found on the main unit.

ATR-124's Multi-Point Search-To-Cue (MPSTC). Provides 100 cue locations.

ATR-124. Your next step is to experience it firsthand.
As you scan the points we've covered, remember that you're scanning just a small portion of ATR-124's story. We haven't even begun to discuss the accessibility of key components for easy servicing and minimal downtime, or the features we've built in to give you greatly improved tape handling. To find out more, write to us at the address shown below. We'll send you a brochure on ATR-124, our latest audio effort. Better yet, call us and we'll set up a demonstration. It's really the only way to listen to the future.

ATR-124. Pure 24-Channel Gold From Ampex.

AMPEX

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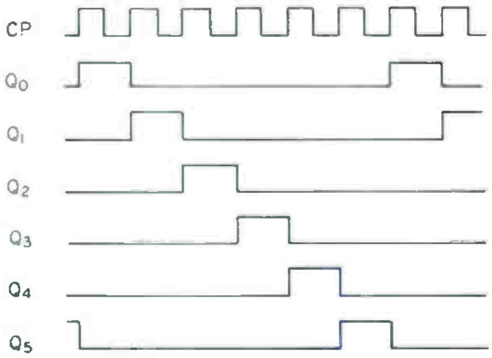

Figure 2. IC5 waveforms.

FAULTY CABLES AT A GLANCE

This cable tester allows the technician to identify faulty microphone cables at a glance. Both ends of the cable are plugged into the device, and six leds indicate the cable's status. The procedure is fast and neat, avoiding the awkwardness of interfacing ohmmeter test probes with cable plugs. Furthermore, intermittent failure is easily detected by plugging in the cable and tugging or shaking it.

Leds 1, 2 and 3 (FIGURE 1) indicate continuity, while leds 1/3, 2/1, and 3/2 indicate shorts. Any break in the connection between pins 1, 2, or 3 of the male end, and the matching pin on the female end, causes the corresponding continuity led to go out. If a short exists between any pair of conductors, the appropriate led lights up. Thus, a good cable will light up leds 1, 2, and 3, while blanking the others. Any other combination of leds indicates not only a faulty cable, but also the nature of the fault. Phase reversal (wires for pins 2 and 3 reversed at one of the connectors) is indicated by the illumination of leds 1 and 2/3, and is thus easily identified.

WHY CMOS LOGIC?

CMOS logic was chosen to minimize power consumption and allow battery operation. IC4 operates as a single square wave generator and clocks IC5, which is connected as a scale-of-six ring counter. Pins Q_0—Q_5 of IC5 produce the waveforms shown in FIGURE 2.

Each of pins Q_0—Q_5 is connected so as to sequentially enable a pair of analog switches. For example, when Q_0 goes high, it enables switches IC1a and IC1c. If a good connection exists between pin 1 of the male end of the cable and pin 1 of the female end, led 1 flashes on. When Q_0 goes low and Q_1 goes high, switches IC1a and IC1c are disabled, and led 1 goes off. At the same time, switches IC2a and IC2c are enabled, testing for continuity between pin 2 of the male end and pin 2 of the female end. Q_2 enables IC3a and IC3c, testing for continuity in the number three conductor.

Similarly, when Q_3 goes high, switches IC1d and IC3b are enabled. If conductors 1 and 3 are shorted together, then led 1/3 flashes on. Q_4 and Q_5 enable switches to test the other two possible short-circuit combinations.

IC4 oscillates fast enough that the flashing leds appear to be on continuously, even though they draw current only once every six clock pulses. Each led draws an average of about 5mA.

Although Switchcraft sockets were installed in the unit to facilitate the testing of standard microphone cables, there are other possibilities. Appropriate jacks would allow the verification of virtually any audio cable with up to three conductors, such as standard ¼-inch phone plug cables or professional RTS patchbay cords. The device has proved useful in the shop for identifying bad cables, and for giving a final check once the cables have been repaired. ■

Circle 18 on Reader Service Card

How serious are you about a power amp?

We build our Professional Series power amplifiers as if our reputation were at stake. Because it is. And so is yours, when you select an amplifier. That's why you should consider Yamaha power amps. They come through for both of us. Because we both designed them. Comments and suggestions from professionals like yourself were incorporated into the final design. As a result, Yamaha power amps excel in the areas that can make or break a power amp—performance, reliability, and flexibility. Take the P-2200 for instance.

Performance. The very conservatively rated specs tell the story. The P-2200 produces 200 watts continuous power per channel, from 20Hz to 20kHz, with less than 0.05% THD, both channels driven into 8 ohms. I.M. and THD are typically less than 0.01% @ 150W for powerfully clean sound.

Peak-reading meters accurately display a full five decades (50dB) of output level for accurate monitoring of program dynamics, transient power demands, and headroom. Frequency response is 20Hz to 20kHz, +0dB/−0.5dB, ensuring transparent highs. The high damping factor of over 300 (8 ohms, 20Hz to 1kHz) provides tighter low-frequency driver excursion and efficient power transfer.

Reliability. Large toroidal power transformers, multiple protection circuits, heavy front panels, serviceable printed circuits, massive heat sinks, and fully vented chassis are some of the reasons Yamaha power amps have a proven reputation for reliability.

Flexibility. Detented, log-linear input attenuators, marked in 22 calibrated dB steps, allow you precise, repeatable setups, accurate input sensitivity adjustments, and simultaneous adjustment of the level of two channels or programs on separate amplifiers. The P-2200 has one male and one female XLR connector plus two parallel phone jacks for each channel for convenient chaining to another amp and

adaptor-free connection to any mixer. A polarity switch satisfies DIN/JIS or USA wiring practice. The P-2200 is readily suited for monaural operation as well as 70-volt commercial applications.

The P-2201 is identical to the P-2200 except it does not have the peak-reading meters. The P-2100 and the P-2050 differ primarily in rated power output and size. Each model offers the maximum in performance, flexibility, reliability and value for the dollar in its category.

We have a technical brochure covering all four models. Write Yamaha, P.O. Box 6600, Buena Park, CA 90622. Or better yet, visit your dealer for a demonstration of the Yamaha power amps that take their job as seriously as you take yours.

Because you're serious.

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BRUCE BARTLETT

Stereo Microphone Technique

Listening tests and evaluations of several microphone-pair arrangements, for recording in stereo, provide a variety of interesting localization effects.

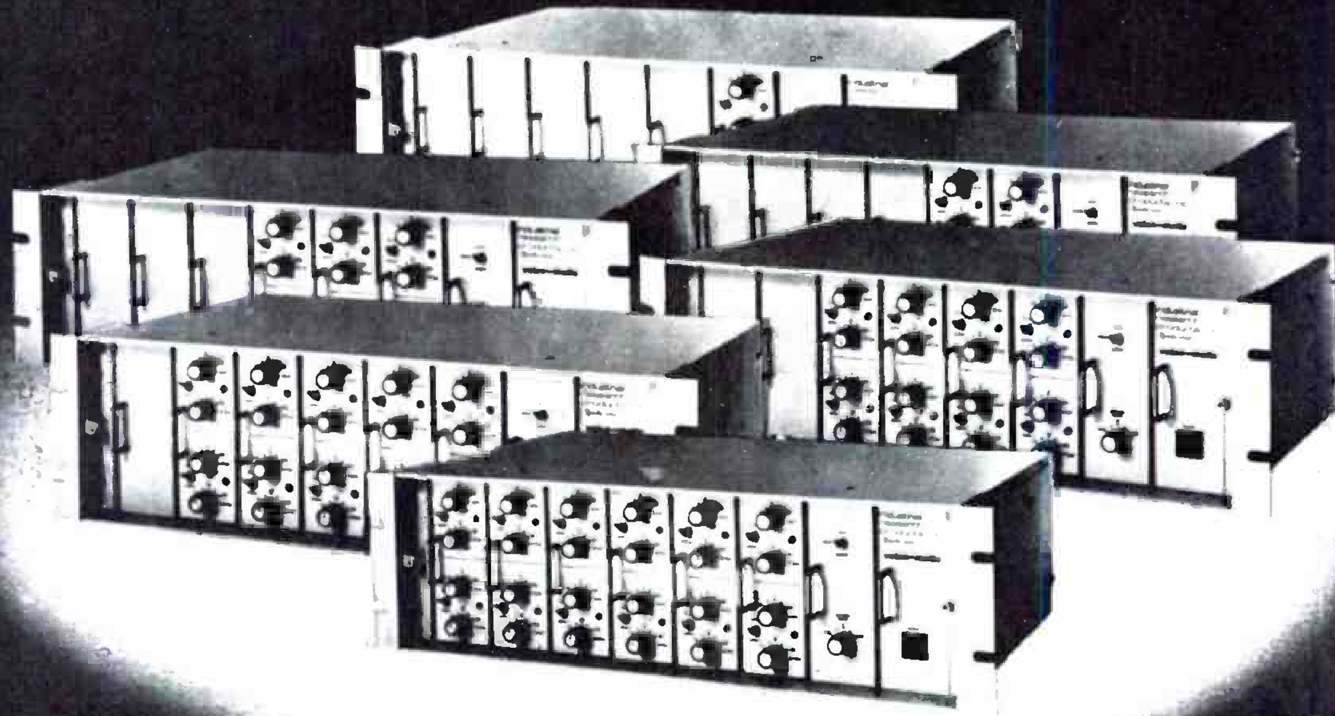
THE ADVENT OF DIGITAL RECORDING and improvements in the sound reproduction chain have drawn attention to a critical link in the recording chain: the microphone technique used during the recording session. For, despite all manner of control-room advances, the accuracy of a recording is still largely determined by the selection and placement of the microphones.

This article will explore several two-microphone stereo techniques, which capture the sound of a musical event as a whole, letting the conductor, composer, or musicians determine the musical balance. Methods which require multiple microphones or matrix devices will not be covered.

INITIAL PLACEMENT

Imagine that we are setting up two microphones to record an orchestra. We place the microphone stand(s) about 6 to 20 feet in front of the front-row musicians, and finally settle on a spot where we monitor a tasteful balance between direct sound and hall reverberation. The microphones are raised high on the stand (10 to 20 feet) to keep the front row of the orchestra from over-balancing the back row, and are tilted down to aim at the orchestra. Due to their close, high placement, the microphones

Bruce Bartlett is a development engineer, Electro-Acoustical Development at Shure Brothers, Inc., Evanston, IL.



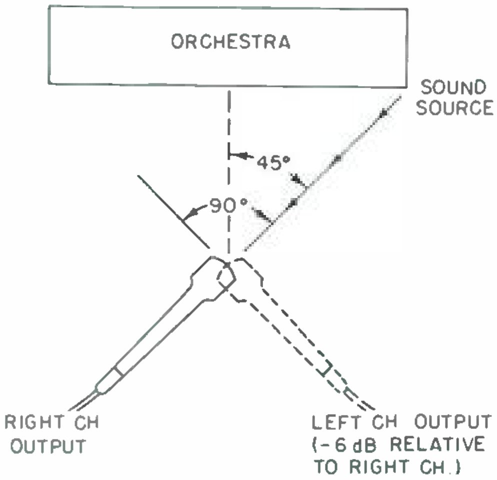


Figure 1. Coincident-pair technique. Cardioid microphones crossed at 90 degrees.

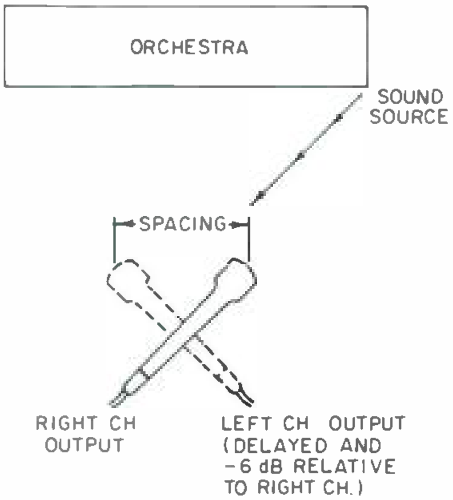
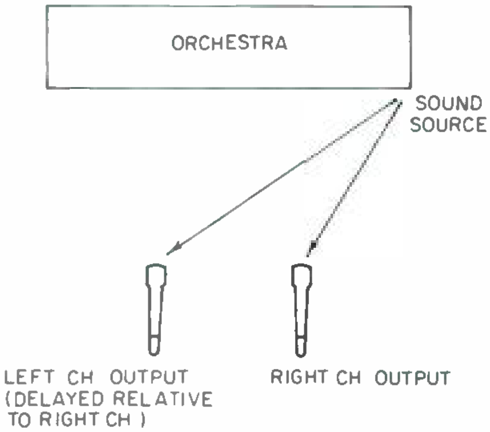


Figure 2. Near-coincident technique. Cardioid microphones crossed and spaced.

Figure 3. Spaced-pair technique.



pick up a brighter sound than an audience would hear, so some high-frequency roll-off in the playback system may be necessary to restore a natural spectral balance. (See Roy F. Allison's article in the Recommended Reading Section.)

STEREO MICROPHONE PLACEMENT: GOALS AND TECHNIQUES

One goal of stereo recording is to place the microphones so that each instrument will be reproduced in the same relative location as it was during the recording.

Suppose that we are listening to the recording over a pair of loudspeakers. We want instruments in the center of the orchestra to be reproduced exactly between the two speakers. We want instruments at the sides of the orchestra to be reproduced from the left or right speaker. Instruments located slightly off-center should be reproduced off-center, and so on. I would like to call this objective "localization accuracy" or "directional fidelity."

Also, we want the reproduced reverberation to simulate live reverberation. The reproduced reverberation should either fill the listening room (apparently coming from every direction), or it should at least spread evenly between the speakers.

Three simple methods used for stereo recording are the "coincident-pair," the "near-coincident," and the "spaced-pair" technique. The coincident-pair arrangement mounts two directional microphones with their diaphragms one above the other, angled apart to aim at approximately the left and right sides of the orchestra (FIGURE 1). Near-coincident placement also angles the microphones apart, but with the microphone grilles spaced a few inches apart horizontally (FIGURE 2). In spaced-pair recording, two matched cardioid or omnidirectional microphones are placed several feet apart, aiming straight ahead toward the group (FIGURE 3).

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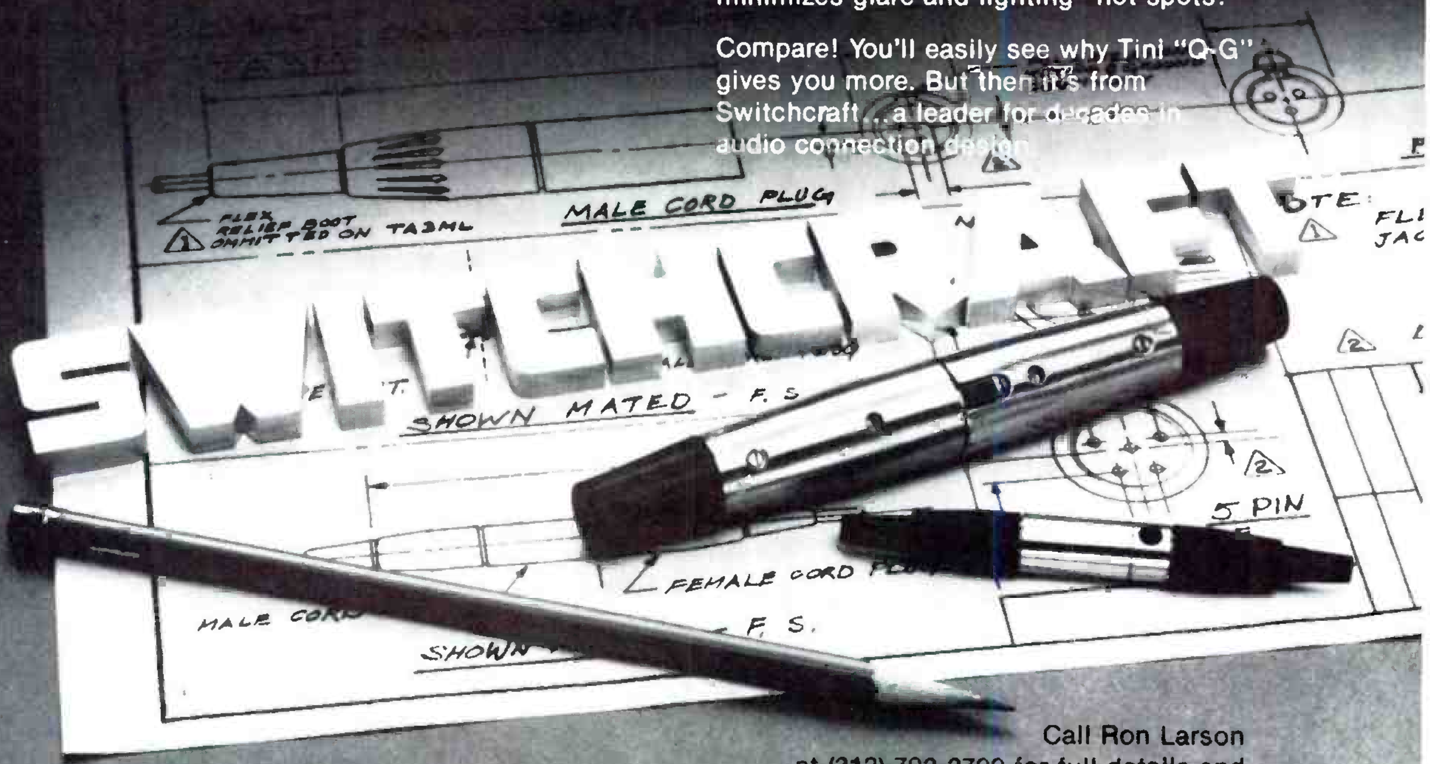

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HOW THESE SYSTEMS WORK: LOCALIZATION BY INTENSITY DIFFERENCES

The location of "phantom images" between two loudspeakers depends in part on the signal intensity differences between the loudspeakers. Suppose a musical signal is sent to two stereo loudspeakers, with the signal to each speaker identical except for an intensity (level) difference. FIGURE 4 shows the approximate sound image location between speakers vs. the intensity difference between channels. (The information in this article is based on carefully-controlled listening tests. The data is the average of the responses of ten trained listeners. They auditioned a pair of high-quality loudspeakers in a "typical" listening room, while sitting centered between the speakers at a 60-degree listening angle.) A 0 dB difference (equal level from each speaker) makes the phantom image of the sound source appear in the center. Increasing the difference places the image farther away from the center. About a 20 dB difference makes the image appear at only one speaker.

Suppose two cardioid microphones are crossed at 90 degrees to each other, with the grille of one microphone directly above the other (FIGURE 1). The microphones are angled 45 degrees to the left and right of the center of the orchestra. Sounds arriving from the center of the orchestra will be picked up equally by both microphones. During playback, there will be equal levels from both speakers and, consequently, a center image is produced.

Let's say the extreme right side of the orchestra is 45 degrees off-center, from the viewpoint of the microphone pair. Sounds arriving from the extreme right side of the orchestra will approach the right-aiming microphone on-axis, but they will approach the left-aiming microphone at 90 degrees off-axis. A cardioid polar pattern has a 6 dB lower level at 90 degrees off-axis than it has on-axis. So, the extreme-right sound source will produce a 6 dB lower output from the left microphone than

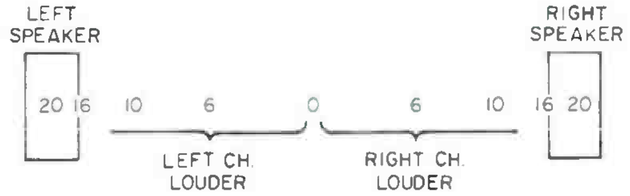

Figure 4. Approximate image location between loudspeakers vs. intensity difference between channels in dB. (Listener's perception; listener sitting centered between speakers at a 60-degree listening angle)

from the right microphone. Thus, we have a 6 dB intensity difference between channels. According to FIGURE 4, the image of the extreme-right side of the orchestra will now be reproduced right-of-center. Instruments in between the center and the right side of the orchestra will be reproduced somewhere between the 0 dB point and the 6 dB point.

If we angle the microphones farther apart, say 135 degrees, the difference produced between channels for the same source is around 10 dB. As a result, the right-side stereo image will appear farther to the right than it did with 90-degree angling. Note that it is therefore not necessary to aim the microphones exactly at the left and right sides of the ensemble.

The farther to one side a sound source is, the greater the intensity difference between channels it produces, and so, the farther from center is its reproduced sound image.

LOCALIZATION BY TIME DIFFERENCES

Phantom image location also depends on the signal time differences between loudspeakers. Suppose we send the same musical signal to two speakers at equal levels, but with one

Your seat may be three decks up in the end zone. But, acoustically, you're on the 50-yard line. A Delta-T audio delay system is what does it. By synchronizing sound from multiple speaker locations in massive Giants Stadium in East Rutherford, N.J., this advanced system provides complete acoustical intelligibility and naturalness from the opening notes of the Star Spangled Banner to the final public announcement — regardless of where your seat may be located.

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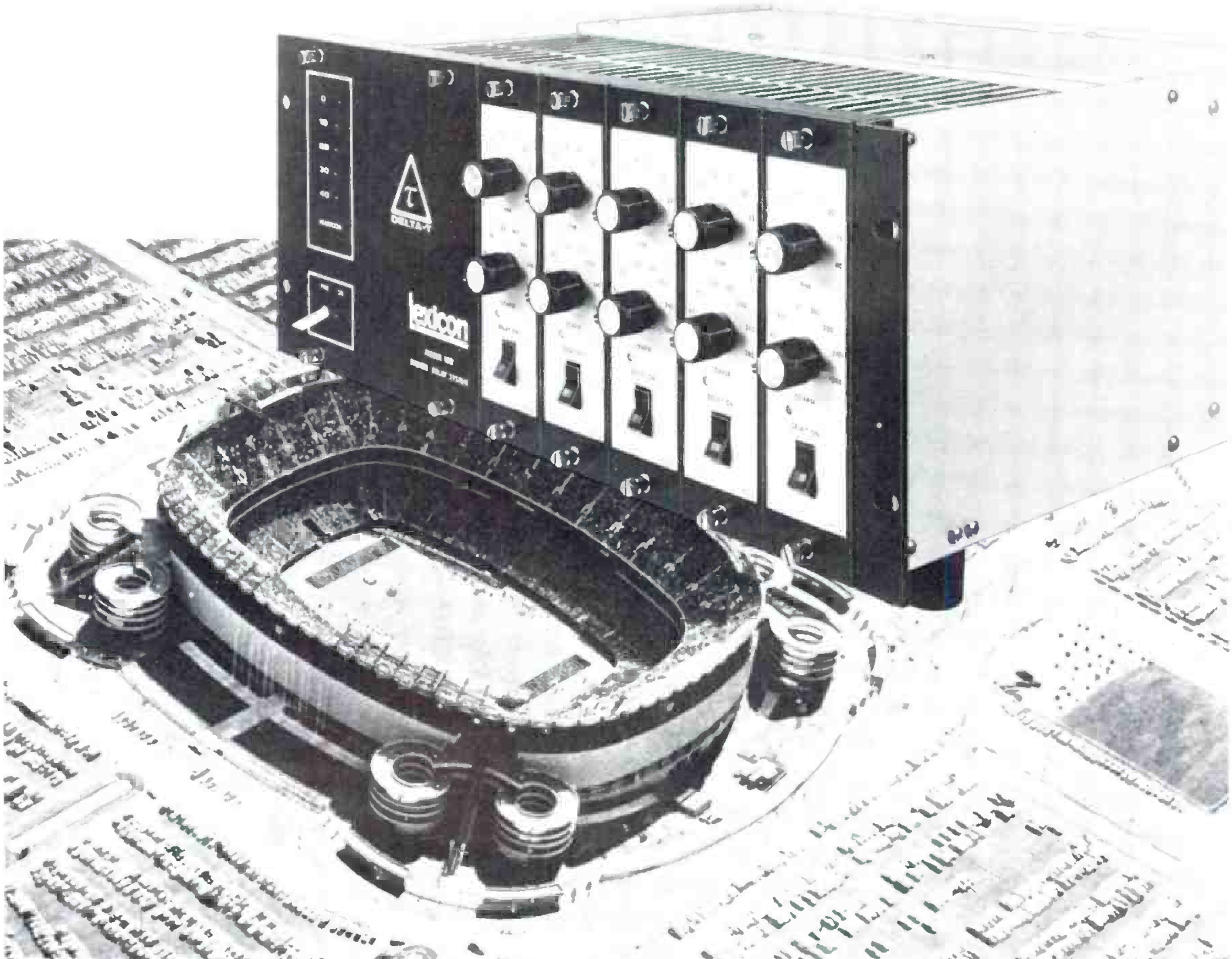

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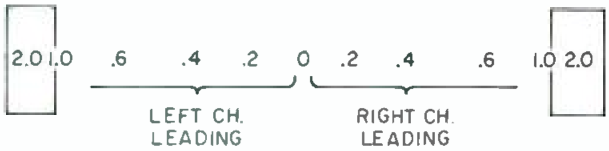


Figure 5. Approximate image location between loudspeakers vs. time difference between channels in milliseconds. Identical signal sent to both channels. (Listener's perception; listener sitting centered between the speakers at a 60-degree listening angle)

channel delayed. FIGURE 5 snows the approximate sound image location between speakers, with various time differences between channels, in milliseconds. A 0 msec difference (no time difference between speaker channels) makes the phantom image appear in the center. As the time difference increases, the phantom image appears farther off-center. A 2 msec time difference is sufficient to place the image at only one speaker.

Spacing two omnidirectional or cardioid microphones apart horizontally—even by a few inches—produces a time difference between channels for off-center sources. A sound arriving from the right side of the orchestra will reach the right microphone first, simply because it is closer to the sound source (FIGURE 3). For example, if the sound source is 45 degrees to the right, and the microphones are eight inches apart, the time difference produced between channels for this source is about 0.4 msec. For the same source, a three-foot spacing between microphones produces a 2 msec time difference between channels, placing the reproduced sound image at one speaker.

With spaced-pair microphones, the farther a sound source is from the center of the orchestra, the greater the time difference between channels and so, the farther from center is its reproduced sound image.

LOCALIZATION BY INTENSITY AND TIME DIFFERENCES

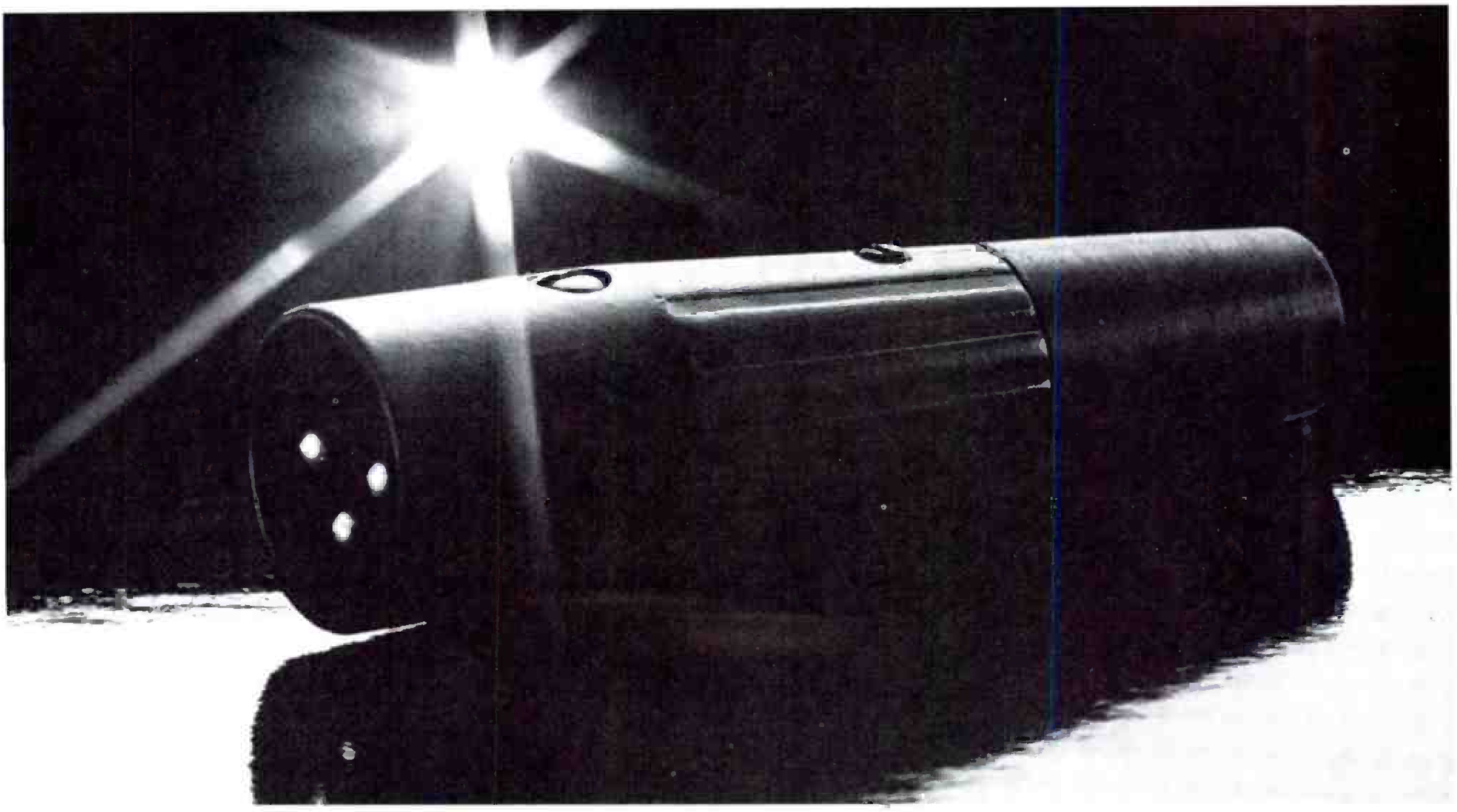
Phantom images can also be localized by a combination of intensity and time differences. Suppose 90-degree angled cardioid microphones are spaced eight inches apart (FIGURE 2). A sound source 45 degrees to the right will produce a 6 dB level difference between channels, and a 0.4 msec difference between channels. The image shift of the 6 dB level difference adds to the image shift of the 0.4 msec difference, to place the sound image at the right speaker. Certain other combinations of angling and spacing will accomplish the same thing.

In review, if a musical signal is recorded on two channels, its reproduced sound image will appear at only one speaker when (1) the signal is 20 dB lower in one channel, or (2) the signal is delayed 2 msec in one channel, or (3) the signal in one channel is lower in level and delayed by a certain amount.

We have seen that angling cardioid microphones (coincident placement) produces intensity differences between channels. Spacing cardioid or omnidirectional microphones (spaced-pair placement) produces time differences between channels. Angling *and* spacing cardioid microphones (near-coincident placement) produces both intensity and time differences between channels. These differences localize the reproduced sound image between a pair of loudspeakers.

LOCALIZATION ACCURACY

If the orchestral width as "seen" by the microphone pair is 90 degrees, then we want sources 45 degrees to one side of center to be reproduced out of only one speaker. Sources 22.5 degrees off-center should be reproduced half-way between the center of the speaker pair and one speaker.



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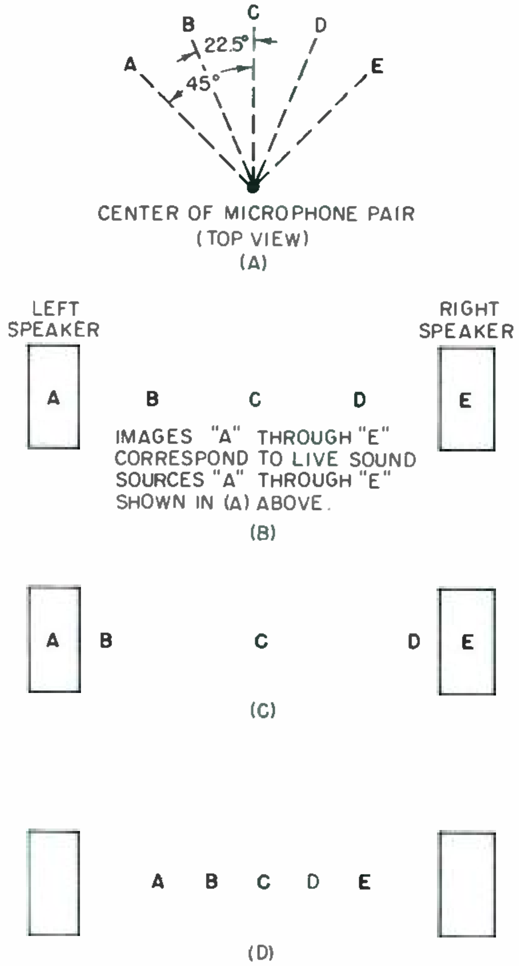

Figure 6. Stereo localization effects. (A) letters "A" through "E" are live sound source positions; (B) accurately-localized sound images between speakers (Listener's Perception); (C) "exaggerated separation" effect; (D) "narrow stage width" effect.

FIGURE 6 illustrates what we are discussing. In FIGURE 6A, the letters A through E represent live sound source positions, relative to the microphone pair. In FIGURE 6B, the corresponding images of these sources are accurately localized between the speaker pair.

Spacing, or angling, the microphones more than is necessary to achieve a full stereo spread produces an "exaggerated separation" effect; instruments near the center are reproduced to the extreme left or right, rather than slightly off-center. Instruments exactly in the center are still reproduced exactly between the speakers (see FIGURE 6C).

Conversely, too-little angling or spacing gives a poor stereo spread or a "narrow stage width" effect (see FIGURE 6D).

A listening test was performed to determine the localization accuracy of various stereo microphone techniques, for a 90-degree orchestral width. Recordings were made of a speech source at 0, 22.5, and 45 degrees relative to the microphone pair (as in FIGURE 6A). Tests were made in an anechoic chamber, and in a reverberant gymnasium. Listeners were asked to note the reproduced sound image locations for several techniques. The image locations of the anechoic and reverberant recording rooms were averaged, with results shown in FIGURE 7. Since results may vary under different listening conditions, this information is meant to be indicative, rather than definitive.

If the orchestral width is more than 90 degrees, the stereo spread of all these techniques is wider than shown in FIGURE 7. The closer to the ensemble a microphone array is placed, the greater is the orchestral width as "seen" by the microphone pair, and so, the wider is the stereo spread (up to the limit of the speaker spacing).

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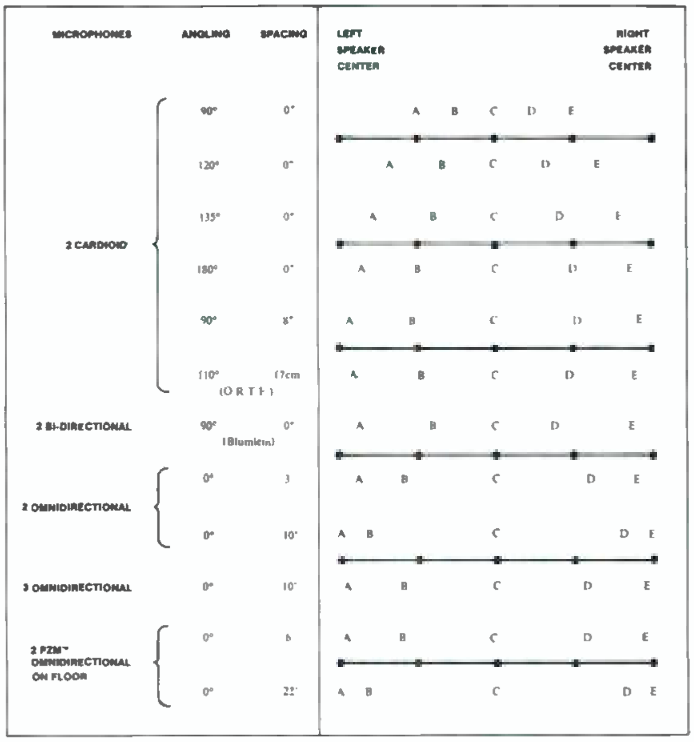


Figure 7. Stereo image localization of various stereo microphone arrays. Images "A" through "E" correspond to live sound sources "A" through "E" in figure 6A.

EXAMPLES OF COINCIDENT-PAIR TECHNIQUES

According to FIGURE 7, it may seem reasonable to angle two coincident cardioid microphones 180 degrees apart to achieve maximum stereo spread. However, sounds arriving from straight ahead will now approach each microphone 90 degrees off-axis. The 90-degree off-axis frequency response of some microphones is weak in high frequencies, giving a dull sound to instruments in the center of the orchestra. In addition, it has been the experience of another experimenter, Michael Gerzon, that 180-degree angling places the reproduced reverberation to the extreme left and right.

A 120 to 135-degree angle between microphones might be a better compromise. Gerzon has reported that the 120-degree angle gives a uniform spread of reverberation between speakers, while the 135-degree angle provides a slightly-wider stereo spread. These angles are useful where maximum stereo spread of the source is not desired. For still-wider stereo images, a near-coincident or spaced-pair technique may be required. However, the 135-degree angle just described can provide a full stereo spread if the orchestral width, or "source angle," is 150 degrees.

Angling at 90 degrees tends to reproduce most of the reverberation from the center. It gives a narrow stage width, unless the ensemble surrounds the microphone pair (180-degree source angle).

The "Blumlein" or "Stereosonic" technique crosses two bi-directional (figure-eight) microphones at 90 degrees, with no horizontal spacing between microphones (see FIGURE 8). Again, according to Gerzon, this particular set-up provides accurate localization, and gives the most-uniform-possible spread of reverberation across the reproduced stereo stage.

EXAMPLES OF NEAR-COINCIDENT TECHNIQUES

The listening tests summarized in FIGURE 7 reveal that the 110-degree angled, 17 cm spaced array (the ORTF, French Broadcasting Organization, system) and the 90-degree angled, eight-inch spaced array provide the most accurate localization. In general, near-coincident techniques give wider stereo spread than coincident techniques.

Another method (not included in the listening tests) is the Stereo 180 System developed by Lynn T. Olson. It employs two hypercardioid pattern microphones, angled

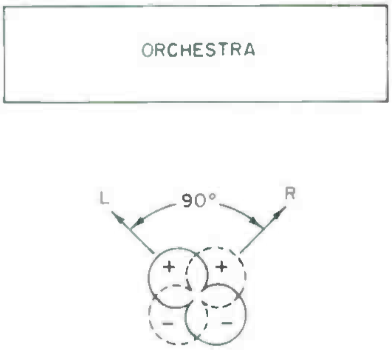


Figure 8. Blumlein technique.

135 degrees apart, and spaced 4.6 cm horizontally. The hypercardioid patterns have out-of-phase rear lobes which create the illusion that the reproduced reverberation is coming from the sides of the listening room as well as between the speakers. The localization accuracy of the array is reported to be very good.

EXAMPLES OF SPACED-PAIR TECHNIQUES

Listeners commented that the spaced-pair methods give vague, hard-to-localize images for off-center sources These methods would be useful when clearly-defined sound images are not desired (for special effect). Spacings greater than three feet give an "exaggerated separation" effect, in which instruments slightly off-center are reproduced full-left or full-right. Instruments directly in the center of the ensemble are still reproduced exactly between the speakers.

A disadvantage of the three-foot spaced arrangement is that the microphone pair is most sensitive to instruments in the center of the orchestra, with reduced pickup of the sides. Spacing the microphones farther apart, say ten feet, may be necessary to cover the whole orchestra adequately, but this would produce exaggerated separation. (An exception to our limitation of two microphones would use a third microphone between the other two, mixed in at an approximate equal level, split to both channels, to reduce stereo separation while maintaining full orchestral coverage.)

When two spaced microphones are placed on the floor, the sound source is above the plane (the floor) of the microphones. As a result, for full stereo spread, floor-placed microphones require a greater spacing than stand-mounted microphones. Listening tests showed that a six-foot spacing between microphones is sufficient for a full stereo spread, when the sides of the musical ensemble are 45 degrees away from the center of the orchestra, from the viewpoint of the center of the microphone array (see FIGURE 6A).

When floor space is available, floor placement has several benefits. No microphone stands are needed, simplifying set up and giving a nonobtrusive appearance. Sound waves reflected from the floor combine in phase with the direct sound waves from the orchestra, giving approximately a 6 dB level boost, and preventing phase cancellations.

When a floor-placed pair is used to record an orchestra, the front-row musicians are usually reproduced too loudly, due to their relative proximity to the microphones. Musical groups with little front-to-back depth, such as small chamber groups, jazz groups, or soloists, may be the best application for this system.

A special variation of the spaced-pair, floor-placed method is the "PRP" system (Pressure Response Pickup) developed by Ron Wickersham and Ed Long. Two omnidirectional pressure-response microphones are placed next to the floor, with a recommended spacing of 22 feet between microphones (although listeners reported an "exaggerated separation" effect with this spacing). Each aims downward toward a metal plate

Figure 9. A Shure A27M microphone adapter which will accommodate a variety of coincident and near-coincident stereo microphone techniques.

placed on the floor. The microphone-to-plate spacing is 0.015 inches. This arrangement gives each microphone equal frequency response at all angles of incidence. A microphone constructed in this manner is referred to as a Pressure Zone Microphone (PZM™).

MONOPHONIC COMPATIBILITY

If two recorded channels are mixed to mono, any time differences existing between channels will result in dips in the frequency response of the mono signal because of phase cancellations. Coincident techniques, in which the microphone grilles are aligned vertically, have no time difference between channels and so are mono-compatible. Other techniques will produce dips in mono frequency response which may or may not be audible. If monophonic reproduction is anticipated, it is wise to monitor the recording alternately in stereo and mono to listen for differences in tone quality.

PHONOGRAPH COMPATIBILITY

If records are made from the original tape recording, the spaced-pair technique can cause problems. The sound-path length from source to microphone is different for each microphone. The difference in path length creates low-frequency phase differences between the two channels. Strong low-frequency out-of-phase signals cause excessive vertical modulation of the record groove, resulting in serious cutting problems unless the recorded level or stereo separation is reduced. The coincident and near-coincident methods allow records to be cut from tapes with less difficulty.

HEADPHONE COMPATIBILITY

Full stereo spread on headphones can be defined as stereo spread from ear to ear. Coincident cardioid techniques produce less stereo spread over headphones than over loudspeakers. If you are monitoring your recording over headphones or anticipate headphone listening to the playback, you may want to use the near-coincident or spaced-pair techniques.

HARDWARE

With coincident and near-coincident techniques, the microphones should be rigidly mounted with respect to each other so that they can be moved as a unit without disturbing their arrangement. Angling and spacing should be user-adjustable to permit stereo spread control.

A device for this purpose is shown in FIGURE 9. It consists of two vertically-aligned rotating sleeves with studs to accept each microphone's swivel adapter. The sleeves can be rotated to vary the angle between microphones and the microphones can be adjusted axially in their swivel adapters to vary the spacing. Vertical clearance between microphones is adjustable.

MICROPHONE REQUIREMENTS

The sound source dictates the requirements of the recording microphones. Live acoustic sources produce frequencies from about 40 Hz (bass viol and bass drum) to about 20,000 Hz (cymbals, castanets, triangles). A microphone with uniform response between these frequency limits will do full justice to the music. The highest octave, from 10 kHz to 20 kHz, gives the added touches of "transparency," "air," "realism," and transient clarity to the recording. Frequencies below 80 Hz may need to be rolled-off to eliminate rumble from trucks and air conditioning, unless it is desirable to record very deep organ or bass drum sounds.

An orchestra or band is a wide-angle sound source. Consequently, sound approaches the microphones from a broad range of angles. If the tonal qualities of all instruments are to be reproduced equally well, the microphone ideally should have a broad, flat response at all angles of incidence within at least ±90 degrees. Stated another way, the polar pattern should be uniform with frequency. Microphones with small-diameter diaphragms usually meet this requirement best. Note that some microphones have small diaphragms inside large housings.

If microphones used in a coincident-pair array are noticeably more directional at high frequencies than at middle-to-low frequencies, instruments which boost the high frequencies of their spectra when played loudly will appear to shift away from the center when played loudly. Again, the polar pattern of the microphone ideally should be the same at all frequencies.

Since classical music can encompass a wide dynamic range (as high as 80 dB), the recording microphones should have very low noise and distortion.

To optimize sound image focusing, the microphone pair should be well-matched in frequency response, phase response, and polar response.

CONCLUSION

You can demonstrate to yourself the degree of localization accuracy provided by your chosen stereo microphone arrangement, for your own particular recording and listening conditions. Record yourself speaking from various positions in front of the microphone pair while announcing your position (e.g., "left side," "mid-left," "center"). If the reproduced stereo stage is too narrow, increase the angle or spacing between microphones. Do the opposite to reduce excessive separation.

We have investigated several microphone pair arrangements for recording in stereo. Each has its own advantages and disadvantages. Which method you choose depends on what sonic compromises you are willing to make. With a little practice, and with some understanding of these techniques, you can achieve quality stereo recordings. ■

Recommended Reading:

Allison, R. and Berkovitz, R., "The Sound Field in Home Listening Rooms," *Journal of the Audio Engineering Society*, Vol. 20, No. 6, pp. 459-469, July/August 1972.

Bevan, W., Schulein, R., and Seeler, C. E., "Design of a Studio-Quality Condenser Microphone Using Electret Technology," *Journal of the Audio Engineering Society*, Vol. 26, No. 12, pp. 947-957, December 1978.

Ceoen, C., "Comparative Stereophonic Listening Tests," *Journal of the Audio Engineering Society*, Vol. 20, No. 1, pp. 19-27, January/February 1972.

Eargle, J., *Sound Recording*, Van Nostrand Reinhold Co., New York, 1976.

Gerzon, M., "Why Coincident Microphones?" *Studio Sound*, March 1977, pp. 117, 119, 140.

Wickersham, R. and Long, E., "How to Make Convincing Stereo Recordings," *High Fidelity Magazine*, August 1978.

ARNIE LAZARUS

The FRAP Point-Source-Microphone

Commonly known as a "pickup," the Point-Source-Microphone is a professional-quality alternative to microphones, for use in live gigs, state-of-the-art studios, and live recordings.

THE "PICKUP" is a general term which describes a transducer that is directly attached to a musical instrument, in order to respond to the instrument's vibrational energy or pressure variation. In the last ten years, it has evolved from a thirty-nine dollar music store accessory to a state-of-the-art device, used side-by-side with two thousand dollar microphones in direct-to-disc recording. Because the term pickup is indiscriminately used to describe all these systems and also connotes the cheaper music store accessory, I would like to propose an alternate term, "Point-Source-Microphone" for transducer systems like the FRAP (Flat Response Audio Pickup).

The term "Point-Source-Microphone" accurately describes a transducer that integrates an instrument's energy in such a way as to correlate to the phenomenon we know as musical sound. Ideally, "Point-Source-Microphones" also eliminate many sounds that should *not* reach the listener, such as fingers sliding on a guitar, bow hair setting violin strings in motion, key noise of a flute, sax or clarinet, or the valve thump of a trumpet.

In the last ten years, FRAP has developed professional transducer systems based on two types of transduction: (1) vibrational and (2) pressure.

VIBRATIONAL TRANSDUCERS

Instruments such as the violin, guitar, banjo, piano and kalimba utilize a resonant body to amplify their basic sound-producing machanism. The "amplifier" is also a tone modifier, adding coloration and distortions to create the tonal qualities of the musical instrument. Since the acoustic "amplifier" operates by vibrating, we may assume the instrument's characteristics may be detected via vibration detection. Since musical instruments vibrate in three dimensions, a transducer capable of detecting all three dimensions would be in order.

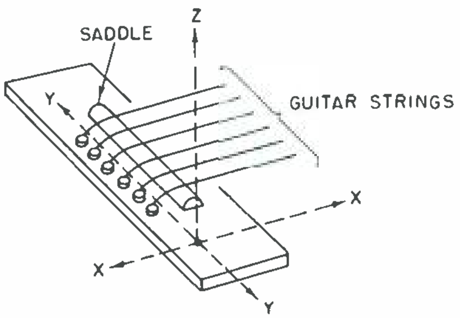
Three-dimensional transduction also lends itself to other areas of detection. In fact, FRAP triaxial transducers have been used to develop a state-of-the-art hydrophone, to record the cables of San Francisco's Golden Gate Bridge, and to reproduce and amplify the sounds of such instruments as hammer dulcimer and salt shaker percussion.

FRAP's triaxial transducers consist of three wide-bandwidth accelerometers, oriented orthogonally and vector-summed to give an instantaneous single-vector resultant output. The three vectors may be represented as:

X-axis—a line parallel to the strings
Y-axis—a line at a right angle to the strings
Z-axis—a line normal to (i.e. passing through the surface of) the guitar.

The X, Y and Z axes are illustrated in FIGURE 1. The accelerometers are piezo-electric, and display extremely-high impedance. (Piezo-electric defines any material which, upon deformation, produces a voltage. Deformation occurs by compressing, bending, or twisting, or by any combination of the above. When an instrument vibrates, the acceleration forces the piezo-electric material to deform microscopically.)

Figure 1. The three vibrational axes on a guitar.



Arnie Lazarus is the inventor of FRAP (Flat Response Audio Pickup). Working on several unusual projects involving FRAP, Mr. Lazarus has recorded the Golden Gate Bridge, designed stethoscopes for NASA, and frapped the entire symphony orchestra which toured with Emerson, Lake and Palmer in 1977.

Figure 2. The FRAP triaxial transducer.

To convert the extremely-high impedance of the transducer (seen in FIGURE 2), to generally-usable impedances, a preamp based around a discrete low noise FET input operational amplifier is required. The preamp allows users to couple the FRAP to such devices as tape recorders or console inputs without degradation or distortion, and further adds tone control and bass roll-off capability.

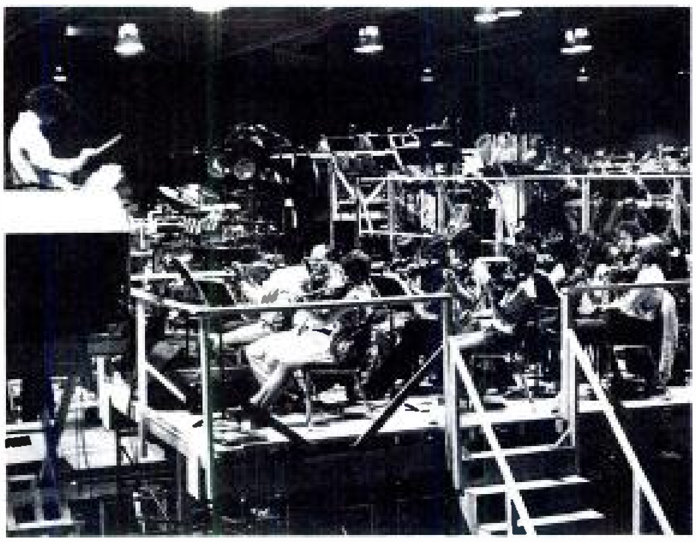
The problem of mounting the FRAP on fine instruments was solved by using an inert wax that has excellent vibrational transfer characteristics. According to *Mechanical Vibration and Shock Measurements*, published by Bruel and Kjaer in 1972, a thin layer of wax gives excellent frequency response—in fact, almost identical to mounting with a steel stud (Pages 121 and 122).

A thin layer of silicon rubber, such as Dow Corning 3145 or 738, can also yield similar results. When using wax, air voids below the 'z' element may produce a "thin" sound. This may be avoided by sliding, rather than pressing, the transducer in place.

Placement of the transducer system is as important as the method of transduction, and a great deal of research has gone into accurate prediction of placement. Today, we can specify with a high degree of accuracy the placement of FRAP triaxial transducers for entire families of musical instruments.

If the overall tonal characteristics of a musical instrument, or the transient or overtone structures do not sound correct, the FRAP is either incorrectly placed or the monitor system is deficient. (Because FRAP's transient response and dynamic range are at least as great as the instrument's, usually fine speaker and amplifier systems are required for accurate determinations to be made.)

Figure 3. The ELP Orchestra in rehearsal.



In 1977, Emerson, Lake and Palmer were scheduled to tour with a 60-piece symphony orchestra. (FIGURE 3) Their goal was to duplicate the music on their album "Works Volume I," which featured full orchestra and choir. To do this, Audio Analysts, a Montreal-based sound company, was instructed to integrate the sound of the band with that of the orchestra, so that the band would not overpower the orchestra, nor be forced to play with reduced intensity in order to hit a sonic balance. Additionally, Audio Analysts was asked to meet these specifications: (1) a natural, not electronic sound; (2) undistorted, clean sound with no feedback; (3) a living room atmosphere, even in the largest arenas; (4) stereo sound; and (5) fast sound check.

FRAP was called in to help solve these problems.

The symphony orchestra had the following complement:

Wood-Winds	Strings	Brasses
2 clarinets 4 C flutes 1 alto flute 2 bassoons 1 contra-bassoon 2 oboes 1 oboe d'amour 1 alto sax 1 soprano sax 1 piccolo 1 bass clarinet	16 violins 6 violas 5 celli 4 bass viols 1 guitar 1 piano Plus 3 percussion instruments	4 trumpets 2 trombones 1 bass trombone 4 French horns 1 tuba

All of the instruments were fitted with FRAP triaxial or Type W pressure transducers, except the brass and percussion instruments, which used microphones. Also fitted with FRAPs were Greg Lake's rare guitars—a Zemaitas 12 string, a Martin 00-45 circa 1910, and another Martin of similar vintage.

Having this "laboratory" to work with, in addition to a critical listening committee consisting of ELP, Godfrey Salmon (conductor) and Bruce Dukoff (concert master and virtuoso violinist), the excellent sound system (costing almost a megabuck) gave us an ideal opportunity to pin down some accurate placement positions for the FRAPs.

We started with the violin family. The thirty-one instruments spanned about thirty makers and three centuries. After an afternoon of experimentation and careful listening, we discovered the spot was *the same* for every one of the violin family instruments! That position was located on the back, with the 'z' axis element directly over the sound post, as seen in FIGURE 4. (The sound post of a violin is less than a quarter-of-an-inch in diameter.) If placement was off by even a fraction, the instrument would not sound right.

The string players did not use wax to attach their transducers, because the wax attachment took too long to adjust. Moreover, in the case of instruments finished with thin brittle layers of spirit varnish, some spots of varnish would cling to the transducer during removal. This almost caused a mutiny among the players and embarrassment of a spectacular magnitude to this author. (This does not occur with instruments finished with oil varnish, as is generally the case with Italian instruments).

We solved the problem by taking the wax off the FRAPs and replacing it with a gob of silicon rubber adhesive, covered with Saran Wrap. The FRAP-with-adhesive was placed directly over the sound post on each instrument, to custom-fit the contour of the back, then lifted off and left to cure overnight. The transducers were ultimately held on the instruments with rubber bands stretched across the corners—or, in the case of the bass, held in place with an elastic strap.

FRAP placement on the piano proved to be more difficult. The piano is a large sound-collecting mechanism that uses 'stiff

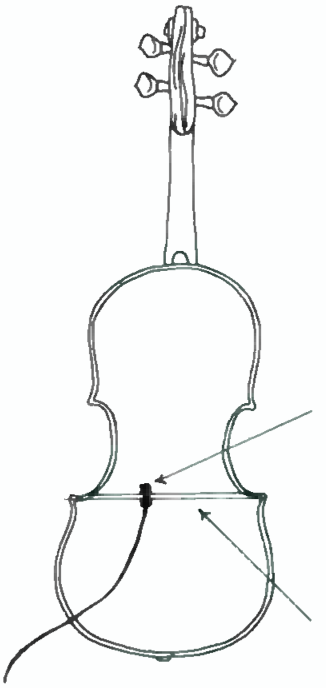


Figure 4. The transducer is placed directly over the sound post.

RECORDING LAURINDO ALMEIDA WITH FRAP AND MICROPHONE

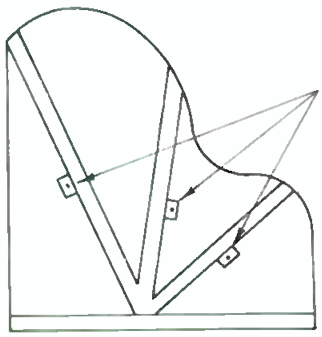
In June 1979, Laurindo Almeida was recorded direct-to-disc by Crystal Clear Records, a San Francisco-based company. Their goal was to capture all the intricacies of Almeida's instrument. On the right track, engineers used a special $2,000 condenser microphone with ultra fast, low noise, low distortion capabilities. On the left track, engineers used a FRAP triaxial transducer. As Chief Engineer Patrick Maloney wrote: "With the FRAP I got the crispness and attack so necessary for the natural projection of the guitar, while capturing the warmth and richness of the entire instrument. With the microphone, I got a slightly-different perspective, along with some room sound which I wanted in order to avoid an isolated, over-dubbed, sterile-type sound. My intention was to capture as natural a guitar sound as I could, and as I wasn't using any equalization, limiting, compression or special effects at all, it was important that the transducer be as transparent and undistorted as possible. The FRAP ... was a joy to use. And speaking of joy, you should have seen the look on Laurindo Almeida's face when he stepped into the control room and heard the sound of his guitar..."

strings' (See "Physics of the Piano," *Scientific American*, December 1965.) Moreover, many piano strings lose their fundamental almost immediately, and the overtone or 'partial' predominates. This effect, especially when amplified, can be confusing or annoying to both the listener and the player. The piano also has three basic resonance modes which have to be dealt with when used either with a P.A. system, monitors, or both. These are (1) the stage, (2) the piano, and (3) the combination of 1 and 2. These often-conflicting resonances can be reduced or eliminated by graphic or parametric equalization.

Most important, the piano sounds very different to the player than to a listener. At the player's ears, the piano sounds more percussive, and the transients display a bite or edge that often disappears with distance.

The best placement of the three-dimensional transducer is on the beams underneath the piano, in the area where the beams tie to the harp (FIGURE 5). Generally, two FRAPs are used, as well as a microphone. However, there are other possibilities, such as on the skirt of the piano or, more-or-less centered on the body. FRAP production manager Walter Rapaport—who devised these placement techniques—adds, "The main thing to keep in mind when placing the FRAP on the piano is to listen to each of these placements without preconception, letting your ears be your guide."

Figure 5. Transducers placed on the beams of a piano.



Finding the correct placement on guitars usually takes a matter of minutes for those familiar with FRAPs and guitars. However, many store salespeople reported problems, and since placement is critical to the sound, we needed to find a method which made placement fast and predictable.

After working with literally thousands of guitars over a ten-year period, including building FRAP systems for the C. F. Martin Company and the Gurian Guitar Company, we developed some 'constants' about placement. Best response is: (1) on the treble side of the bridge; (2) closer to the end of the bridge than near the strings and; (3) with the 'Z' axis element behind the saddle—*not* in line with it. Our limited experience in working with classical guitarists, due to their sense of 'acoustic purity' resulted in placing FRAPs on the crest of the bridge. However, while working with guitarist Laurindo Almeida during direct-to-disc sessions, we found the best placement of the FRAP was just behind the crest—in line with where the strings *tied to the body*.

This astounding placement led us to our breakthrough—the finding of placement constants that would work for virtually any guitar! As this is the first time this placement is being offered, I would welcome any comments or explanations.

The transducer's 'Z' axis element must lie on a line formed by the strings entering the body or, in the case of a twelve-string guitar, the line somewhere between the two lines. This coordinate will work for any type of guitar: steel string, classical, flamenco, or archtop. (The archtop generally has a trapeze tailpiece, and due to the variance of construction in relation to arching and tailpiece, these designs will probably contain some anomalies relating to placement.)

A partial explanation for this placement of the "Z"-axis element along the "Y" axis, is as follows: If one works out the static force equations of a string on a guitar, one will easily observe that the vertical force of the saddle holding up the strings is equal and opposite to that of the body of the guitar holding the strings in place. As the top of the saddle is a node, very little vibration occurs here. Therefore most of the energy is transferred to the guitar top at the point where the strings are anchored to the body.

The "X" coordinate is determined this way: If we take the string length from nut to saddle and divide by seven, we get the

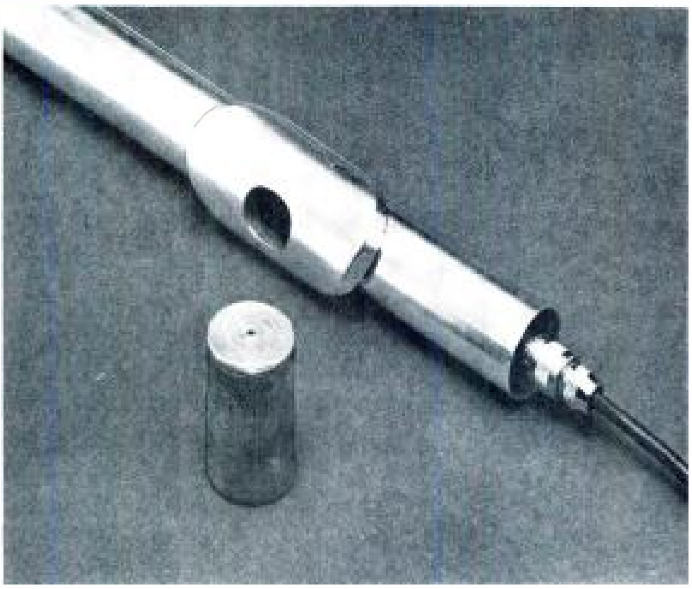


Figure 6. The Type W flute transducer.

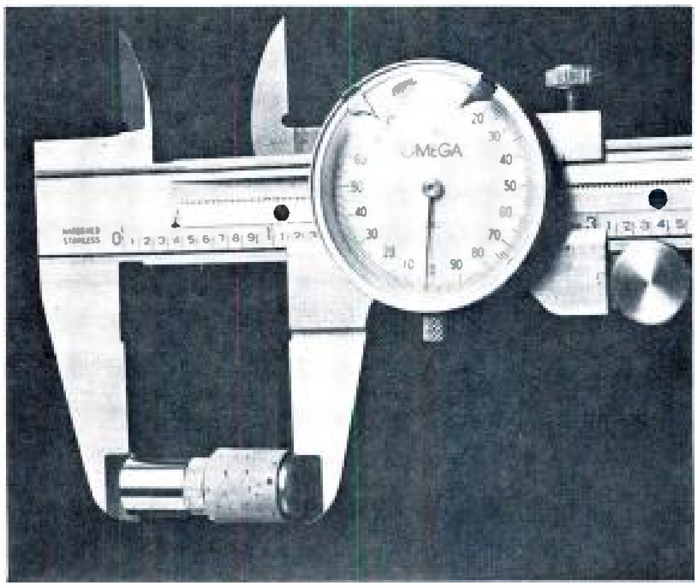


Figure 8. The Type W piccolo transducer.

distance that the vertical element would be from the low E (6th) string. The distance will be taken along the line created as the strings enter the body. My estimation is that this is good for plus or minus an eighth of an inch.

Empirically, these coordinates work, although the above explanation is by no means complete. While I'm confident about stating this as a "formula," the supportive data is still being analyzed by the author at the time of this writing. As soon as I have more-complete information, I will submit it in a letter to **db**.

PRESSURE TRANSDUCERS

Instruments such as the flute, saxaphone or clarinet depend upon a resonating air column to produce sound. The instrument derives its characteristic tones from instantaneous pressure deviations in the basic air column.

The FRAP Type W transducer detects the sound of a wind instrument via the vibrating air column. If just pressure changes are monitored, and if the transducer does not really move or flex, except at the molecular level, the structure of the instrument will not change. The transducer will respond only to sounds in the air column and not mechanical noises such as key clicks or breath intake. (Instrument structure is mentioned because as soon as the transducer is attached to the instrument and a nipple introduced to the air column, it becomes part of the structure.) The Type W transducer satisfies the above premise and the evolution of its development is worthy of note.

Type W transducers come in three versions: Flute, Screwmount and Piccolo. (FIGURES 6, 7 and 8) The flute and piccolo pose no major problems as both are primarily straight-bore instruments. The sound inside is very similar to the sound outside.

In the flute, the main problem is to create a situation where the sound is not changed acoustically, and the transducer's presence remains undetected within the flute. This is done by replacing the cork and faceplate assembly with the transducer. In order to reflect the standing wave in the air column efficiently, the faceplate must be a metal such as silver, stainless steel, etc. So as not to disturb any of the flute's subtleties or tuning, the hole in the plate that introduces the pressure to the transducer must be sufficiently small. Moreover, there must be virtually no displacement of air in the detector. The tube leading to the diaphragm must be very short or high-frequency response will suffer.

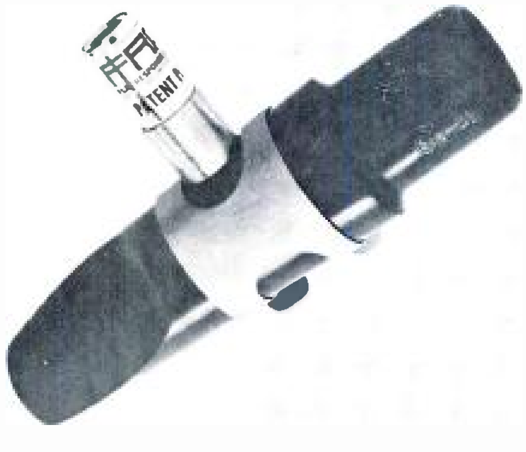
As a result of meticulously working out these details, the Type W flute transducer will give very accurate reproduction and will not feed back, even in extremely high level sound fields. As it is a non-accelerometric device, it also does not detect key noise nor does it 'hear' the air blast and 'edge' tone one gets with close miking.

A past criticism in Type W transducer's use in recording has been the lack of ambient sound or reverb. This can be remedied with artificial ambience or by using a combination of mike and transducer. The benefits of using the Type W transducer for recording are (1) overdubbing without headphones, (2) using effects that require a very smooth signal to trigger them, such as a pitch to voltage follower; and (3) performing in extremely loud environments with a minimum of problems.

The Type W piccolo transducer is exactly like the flute design but is much smaller in diameter.

A very interesting set of problems confronted us for the Type W screwmount transducer for wind instruments. In order to sample the pressure variations, the air had to enter the transducer via a nipple. If we made the tube straight, we had an acoustical low-pass filter, due to the tube resonance. The tube had to be as short as possible. If these requirements were not satisfied, the transducer would have the tubby, bassy sound associated with microphone use. The tube was kept to a minimum length and the resonance removed by turning the nipple into a horn. (Thanks here to Cliff Hendricksen, horn designer at Altec-Lansing, and Dr. James McGill, president of Digital Recording Corp., for their help and analysis.)

Figure 7. A Type W screw-mount transducer on a saxaphone mouthpiece.



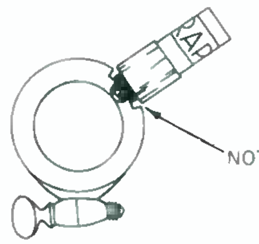


Figure 9. Installation of a screw-mount transducer.

A further problem was caused by the difference in sound between the inside and the outside of the instrument which occurs in conical-bore reeds. When we first began this project, we noticed the sound was very tubby for saxaphones and clarinets. I contacted Professor Arthur Benade of Case-Western Reserve University, one of the most knowledgeable people about the physics of musical instruments, and the author of several excellent books on the subject. Professor Benade explained that the latticing of open holes in the instrument creates a 6 dB-per-octave treble boost which flattens out at a certain frequency for each type of instrument in the family. We then devised an equalization network which is adjustable for each type of wind instrument. Thus, both aforementioned compensations worked and removed a characteristic tubbyness which masked the sound of the instrument. Ironically, to trigger a synthesizer, the opposite must occur, so that the pitch-to-voltage converter is not confused by complex upper-order harmonics.

The mounting techniques for the screwmount transducers are illustrated in FIGURES 9 and 10. Note that a double reed is included. The double reeds are different from the single reeds insofar as the toneholes are small compared to the diameter of the bore. This is different from the clarinet type of instruments which are built around the Boehm system, where the tone holes are of a size comparable to the bore. It is largely this geometry that gives the oboe and the English horn their high pressure and characteristic tone. Indeed, when an oboe was once built on a Boehm design, it sounded like a double-reeded clarinet.

Being the smallest and highest-pitched instrument of the double reeds, the oboe has the most difficult sound to reproduce with the transducer. Its constricted sound results in considerable phase shifting. In fact, Paul McCandless, reed player for Oregon, claims the only way the amplified oboe begins to sound right is when it is run through a phaser! And, according to David Friend of ARP, the oboe has a predominant tone within it which interferes with other tones at all times. Moreover, the addition of the transducer to the staple (where the reed is wound) adds enough geometric volume to put it out of tune, and an equal volume to that added by the transducer nipple must be removed from the staple by filing.

The trumpet displays similar problems as the saxaphone but instead of having tone holes to push up the treble, it has an impedance-matching system called a bell. The equalization method is the same as for the reeds—roll off the lows. The transducer is mounted with a screw into the stem of the mouthpiece, "as in the conical bore reeds." (The mouthpiece is drilled and tapped for a 10-32 thread.)

Figure 10. Double-reed mounting technique. Epoxy holds the 10-32 nut in place, and a 1/16-inch hole is drilled into the bocal.

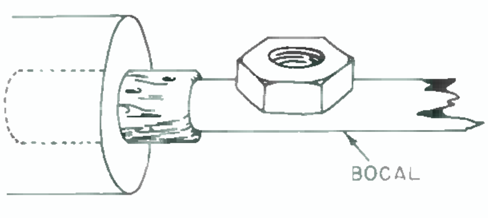


Figure 11. Carlos Montoya and FRAP, on a direct-to-disc recording.

Only one other serious problem confronted us with wind instruments—they are wet. A bubble can form across the nipple. We solved this problem by slotting the top with a jeweler's saw to break surface tension.

CONCLUSION

The ability to accurately specify placement for the FRAP Triaxial Transducer and the FRAP Type W Pressure Transducer provides a professional-quality alternative to microphones for many situations—including live gigs, state-of-the-art studio, and live, recording. The major difference between studio use and P.A. (or live recording) is that the room ambience will affect the tonal qualities of the instrument and result in some acoustic distortion. This is true because the instrument is a receiver as well as a transmitter. In a studio, the ambience is controlled. The sound vibrations impinging on an instrument such as a guitar or piano or even a horn will effect and distort the outgoing vibrations. However, it should be noted that these problems also occur when a microphone is used.

The recording studio also gives users the advantage of lower acoustical distortion. Moreover, with direct transduction, set-up time is reduced. The FRAP's built-in isolation helps keep the semblence of a group together if one does not want to lay down tracks. The FRAP's distinct advantage to microphones is that it can feed effects and synthesizers.

The mixture of a professional grade FRAP transducer and a microphone, as was done recording both Laurindo Almeida and Carlos Montoya (FIGURE 11) by Crystal Clear Records, will give the performer the best of both worlds.

To hear how FRAPs sound when used in recording, I recommend that the reader listen to the Crystal Clear Recordings of Montoya and Almeida, John McLaughlin's Shakti album on Columbia and Herbie Mann's Sunbelt album on Atlantic. ■

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Index 1979

INDEX BY TITLE

INDEX BY AUTHOR

Ron started as a singer in Philadelphia. He worked the board at several major festivals during the late '60s before entering the studio in England during the early '70s. Along the way, he began producing. As a producer and/or engineer, Ron has worked with The Who, Led Zeppelin, Bad Company, Dave Mason, The Babys, UFO and many others. His most recent project was with The Jefferson Starship.

ON MULTI-TRACKING

"I go for the whole thing. I would rather not do anything for two days than have to take the band down to three pieces and have to build it back up again. I'd rather piece the tracks together than piece the band together. I mean, there'll still be overdubs and things like that, but rock'n roll is so much a feel situation, you know?"

ON DIPLOMACY

"A lot of times, people will stand around and everybody will think the other guy likes it. Nobody will say 'Well, I don't like it.' It won't be till after a while that they find out that nobody ever liked it. They just never wanted to say anything. Now, I'm the guy who goes in there and gets it all out of them—what they like and what they don't like—so there's none of that. I can be the bad guy, sometimes. I'm just real frank and rough. If somebody's not doing something, I like to say it right then and there, so one of the band members doesn't have to say it. It might be a shock, but none of it is taken out of the studio."

ON MUSICAL STYLES

"You know, hard rock stuff is the hardest thing to record. People whacking the hell out of the drums. Guitars turned up to ten. Everything is distortion. People screaming down microphones. The harder the rock, the harder it is to record."

ON TAPE

"Consistency. That's the most important thing. You know, you can work all day for that one thing and you put that tape on and it drops out or it does something. You stay with it until it cracks up. Then you use somebody else's. And I did that a lot. I've used everybody's tape. I've been using 3M tape for five or six years, exclusively. They happen to use the same tape I do, here at The Record Plant. But if they didn't, I would have my own tape in in a second."

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