ISSN0011-7145

THE SOUND ENGINEERING MAGAZINE

db

JULY 1978
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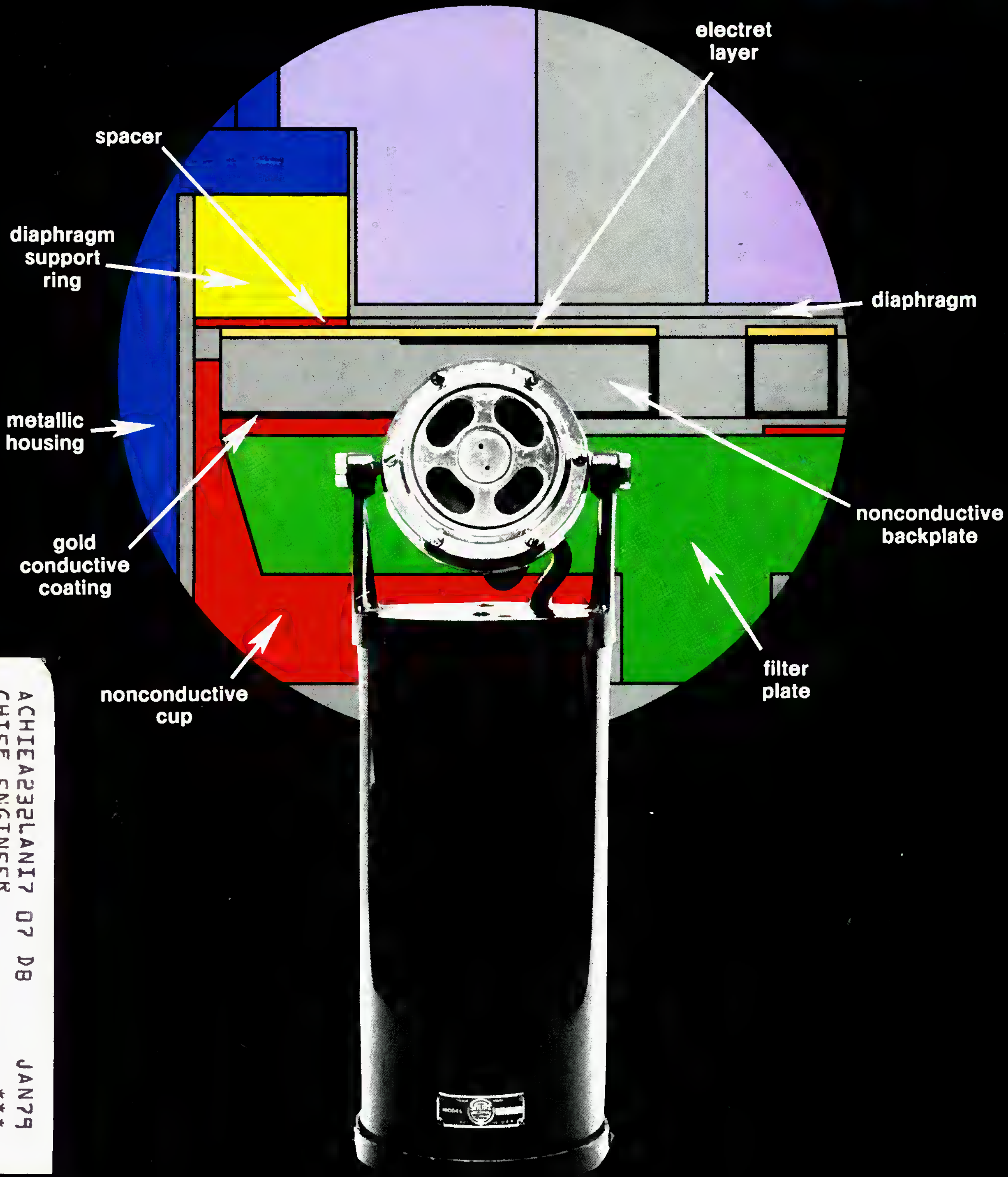

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Coming Next Month

• August is Microphones, Part 2. Among the articles planned, one by John Borwick of Great Britain will detail the Ambisonics Microphone System and what it means in the recording studio. This will complement the article by J. Howard Smith of Calrec in this issue.

The Ghent Microphone System for stereo and quadraphonic recording is explained by Benjamin B. Bauer of the C.B.S. Technology Center.

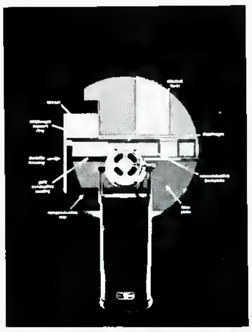
The issue will also contain a directory of electret microphones—serving as a catalog of this important product line.

A Backward Glance at Cardioid Microphones by John M. Woram will do just that. ". . . to obtain an instrument with unidirectional sensitivity pattern . . . using one transducer element only." . . . is from a 1938 patent application by a then young (and now famous) engineer/scientist.

What John could not fit into this issue's coverage of the Los Angeles AES Convention gets into August.

And there will be the usual columns and features. All this coming in August in **db, The Sound Engineering Magazine.**

About The Cover



• Old and new combine this month. The color background is a drawing that details the working parts of a modern electret condenser microphone. And just in case you think the condenser mic is something new, we have superimposed a much older mic. Can you identify it? Hint: it was made by Shure Brothers who also supplied the electret drawing. See this space next month for the answer.

db

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db Letters

THE EDITOR:

When I read Dan Keen's *Understanding Maximum Power Transfer,* I suffered alternate spasms of dismay, disbelief, rage, and severe puzzlement. What sort of drivel is this? In one swell foop (!), Mr. Keen has set audio science back into the nineteenth century. Is he a nephew of the publisher? Is there no one at **db** who passes any judgment on articles published? My primary field is not audio, but it is quite obvious that I know a hell of a lot more about it than Dan Keen does. Therefore, I am also outraged that something like that thing he wrote ever makes it into print.

I am sure that Norman Crowhurst (who does know what he is talking about) could set the matter straight in a much better manner than I can (especially considering my present mood), but in case no one else wants to tackle it. . . .

Mr. Keen is correct insofar as his statement that maximum power transfer requires matching of source and load impedance, but SO WHAT?

Thomas Edison knew the same thing, and it kept him from designing a practical power network. If the alternators at power stations were suddenly matched to their load, they would be destroyed almost instantly, because:

1. While matching source and load impedance does allow maximum power transfer, HALF the total power is dissipated in the source. A source of any significant power generally won't stand that for very long.

2. I have been working in or with audio for only 32 years, so I haven't seen everything. But I have never seen an audio amplifier with an 8-ohm source impedance. I have seen many, many, many designed to be connected to an 8-ohm load, but they certainly didn't have any such source impedance. An amplifier marked "8 ohms" is designed, hopefully, to give best results with an 8-ohm load, but it is most unlikely that its source impedance is much over 1 ohm these days—frequently it is much less.

I hope that by the July issue, someone will have straightened this mess out.

WAYNE W. DUNNING
Wichita, Kansas

Our face was red!

Mr. Dunning's letter is one of many that we received. For what may seem a final word, see this issue's **Sync Track.**

(continued)

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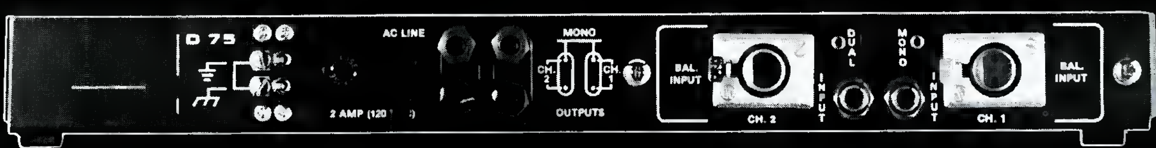

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letters (cont.)

The Editor:

I wish to direct your attention to a typographical error which occurred in my article, *Noise Level Limits in Recording Studios,* which appeared in the April, 1978 issue of your publication. Please eliminate "for the hi-fi enthusiast" at the end of the next to the last paragraph.

I'll thank you to publish this note because the sentence the way it stands sounds ridiculous.

Michael Rettinger

Ed. Note: We agree. The weird juxtaposition of words occurred during a rough day at the print shop and we were not canny enough to discern it. Mr. Rettinger is definitely not responsible for it.

db Calendar

AUGUST

Management Seminars. Contact: Heidi E. Kaplan, Dept. 20 NR, N.Y. Management Center, 360 Lexington Ave., New York, N.Y. 10017. (212) 953-7262.

3-4 Foreign Market Entry Strategies. Wharton School, U. of Pennsylvania. Washington, D.C.

7-8 New Products: A Systematic Approach. New York University. Los Angeles.

14-15 The Effective Engineering Manager. New York University. Houston, Texas.

17-18 Unlocking Creativity. New York University. New York City.

13-17 **British Musical Instrument Trade Fair.** Bloomsbury Centre Hotel; Hotel Russell; Connaught Rooms, London. Contact: British Information Services, 845 Third Ave., New York, N.Y. 10022. (212) 752-8400.

11 **National Radio Broadcasters Association Management Seminar,** Welsh Company, Tulsa, Oklahoma. Contact: NRBA, Suite 500, 1705 De Sales St. N.W. Washington, D.C. 20036. (202) 466-2030.

14-16 **J. B. L. Sound Reinforcement Workshop,** Kansas City. Contact: Nina Stern, James B. Lansing Sound, Inc., 8500 Balboa Blvd., Northridge, Ca. 91329. (213) 893-8411.

SEPTEMBER

11-13 **J.B.L. Sound Reinforcement Workshop.** Vancouver, Canada. Contact: Nina Stern, James B. Lansing Sound, Inc. 8500 Balboa Blvd., Northridge, Ca. 91329. (213) 893-8411.

New York Management Seminars. Contact: Heidi E. Kaplan, Dept. 14NR, New York Management Center, New York, N.Y. 10017. (212) 953-7262:

11-13 **Management of New Technology Projects.** Boston, Mass. N.Y.U.

14, 15 **New Products: A Systematic Approach.** Houston, Texas. N.Y.U.

21, 22 **Foreign Market Entry Strategies.** Chicago, Ill. Wharton School.

17-20 **National Radio Broadcasters' Association, Convention.** Hyatt Regency Embarcadero Hotel, San Francisco, Ca. Contact: N.R.B.A., Suite 500, 1705 De Sales St., N.W., Washington.

20-24 **Autumn Hi-Fi Show.** Cunard International Hotel, London, England. Contact: British Information Services, 845 Third Ave., New York, N.Y. 10022. (212) 752-8400.

25-29 **International Broadcasting Convention.** Wembley Conference Centre, London, England. Contact: British Information Services, see above.

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PATRICK S. FINNEGAN

db Broadcast Sound

Test Errors

● We test the performance of audio units, as well as the entire system from time to time. But tests may not always reflect the true performance of a unit in its normal program operational mode. The wiring arrangements, procedures, and instruments used in the test can often in themselves introduce errors which produce misleading results. Discussions this month will center on some of the factors which can affect and distort the test results.

ENVIRONMENTAL EFFECTS

Wherever a particular audio unit is physically placed in the system, the location will subject the unit to environmental conditions such as air temperatures and circulation, vibrations, and so forth, as well as to other conditions peculiar to that setting—for example, strong rf or noise-inducing fields. This is the particular setting in which that unit must operate during programming and some of these conditions may dictate the manner in which we must adjust the unit if it is to operate properly.

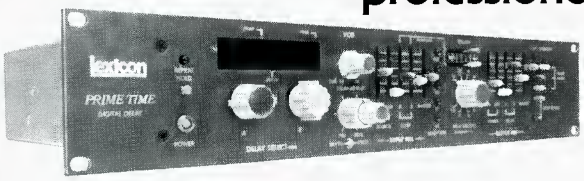
Since environmental conditions are a part of the normal operation of the unit, we must be careful of the test results if these conditions are altered while the test is being conducted. In many test setups, the environmental conditions are changed, since there is no way to avoid it. But since changing the conditions in some cases can also change the test results, it is important that we take this into consideration when we make a judgment on those test results. Consider, for example, a transistor which has become heat sensitive in an amplifier. The ambient temperature buildup in the enclosed rack or cabinet causes the stage to distort on signal peaks. Now if we are testing this unit for the distortion, but have the circuit board out in the open air on an extender card or the unit opened up on the work bench, the lower temperatures would allow the unit to check out perfectly! Changing the environmental conditions in this case have also changed the results, which can be very misleading to the troubleshooter.

When test results seem suspicious, and we have altered the environmental conditions in which the unit normally operates, we can try to simulate these conditions by the use of heat lamps, heat guns, fans, cooling sprays, and so forth. If the use of any of these devices affect the test results, then weigh those test results carefully. Try to determine if they represent the true performance of the unit with programming.

WIRING

In its normal habitat, an audio unit will be interconnected to other units through well-shielded cables, tied to a building ground system and afforded some degree of protection from rf and noise fields by cabinets, racks, and conduits. When we remove a unit to

professional quality delay plus special effects



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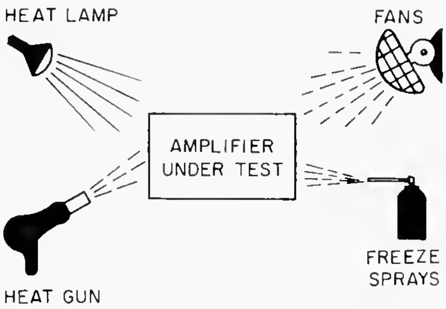


Figure 1. It is sometimes necessary to simulate the operating environment of the unit when it is in a test location.

the workbench for tests, we also remove the protection the cable system offers and substitute an altogether different arrangement. Sometimes, even though we test a unit in its normal place in the system, we substitute different input/output wiring and attach test instruments. This, too, can remove a part of the system's protection.

Perhaps the greatest offenders in producing misleading test results are the "haywire" test arrangements we set up to conduct the test. Unfortunately, there are many instances when haywire cannot be avoided for the test setup. But even when this is the case, we should try to take whatever precautions we can to reduce any ill effects such wiring may have on the test results. For example, we can use shielded cables and take care with the shield grounds, get a good ground to the unit, test instruments, and building ground. When strong rf fields are present, be especially careful with the shielding and grounding. Remember that a long test clip used for a ground is an antenna at f.m. frequencies. Any extraneous signals introduced by the wiring arrangement can produce erroneous test results that can be misleading.

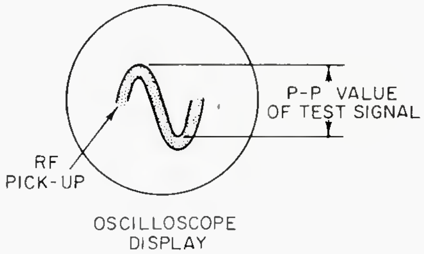
When test results seem to indicate serious performance flaws, try to determine if these are bona fide equipment faults or only distortions created by the test arrangement. An oscilloscope attached to the output of the unit or to the distortion analyzer (if used), can often be a help in identifying grounding problems, noise and rf pickup in the test arrangement. Even though we can identify the problem, we may not always be able to eliminate it completely. In that case, we must weigh the results so that true performance of the unit can be determined.

PROCEDURES

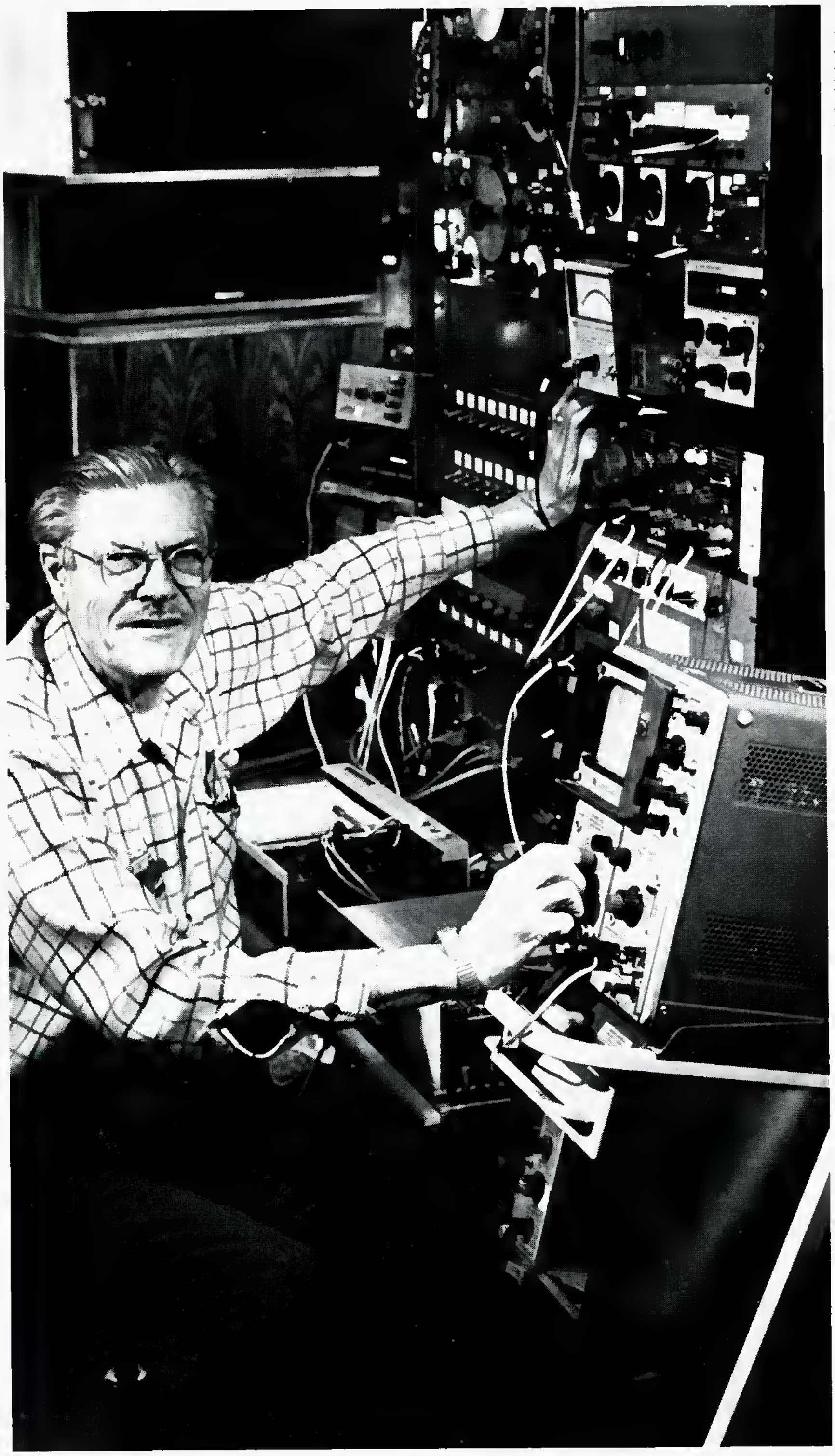
The manner in which we conduct the test, as well as the manner employed to have the unit adjusted and operated during the test, can very well produce results that don't reflect the true performance of the unit in its normal operational mode in the system. As mentioned earlier, the environmental conditions may require that a unit be operated in a specific manner. If under the test arrangement we make a number of readjustments to test the unit as it would be operated in other situations, then the test results can be far from the true performance of the unit where it must operate in the system on a day to day basis.

Consider, for example, that you have an amplifier unit in the system at a place where the input signal must be relatively high, perhaps due to a noise situation, and you operate it this way to obtain a better-signal-to-noise ratio. In its normal mode the input control is just barely open. The input control on the unit is a "T" type fader across a balanced circuit. Such an ar-

Figure 2. Rf pickup can distort the indications. Illustrated is rf pickup by the oscilloscope, which produces a "thick" trace. Use only one "envelope" of this trace for your measurement.



LISTEN.



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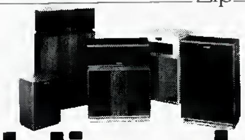

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In Canada: Superior Electronics, Inc.

Circle 19 on Reader Service Card

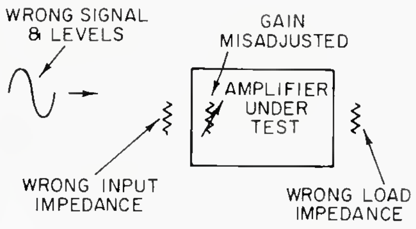


Figure 3. Some factors which can produce errors in the test results.

rangement and adjustment can roll off the high frequency response. With the unit on the bench for test, you reset the signal generator level to ordinary standard levels and readjust the control of the unit to mid-point (the way this type of control and arrangement should be operated). With the test set up in this manner, the unit checks out with a perfect response curve! But this is not a true picture of what really happens to the system frequency response during programming by that amplifier and the ordinary way it is operated on a day to day basis. A true test of the performance would require that the setup duplicate the regular way this unit is adjusted. The signal generator, however, may not be able to provide the higher level signal required. In that case, you would need to include the driving amplifier in the test and then test both units together.

OTHER CONSIDERATIONS

Remember that the unit operates with program signals at given input/output levels, and that the unit is adjusted for given input/output impedance matching and loads. If we change any of these in the test setup, we can very well influence the test results.

The test signal in many cases is a sine wave. Sine wave and complex program signals are considerably different and the unit will react differently when these signals are passed through it. The program peaks can be 8 to 12 dB higher than the same indicated level of sine wave on the test meter. These peaks might overload some stage in the unit, introducing harmonic and intermodulation distortions. Unless we are satisfied that the unit has adequate headroom, those beautiful distortion measurements with sine wave may simply not be true when program audio passes through the unit. We can easily check for adequate headroom by simply increasing the signal level from the generator another 10 dB higher than the normal input level and then checking for an increase in distortion. If there is no increase, we can be reasonably sure the original distortion measurements represent the true performance of the unit.

Aside from adjusting and supplying normal signal levels from the generator, the input/output impedances and load on the unit can affect the results in many ways, such as giving poor frequency response and poor signal-to-noise ratios. In effect, the unit is not adjusted properly at all. If the load impedance is incorrect, the signal levels through the unit will require greater adjustment than normal, and this in itself is not truly representative of the unit's performance.

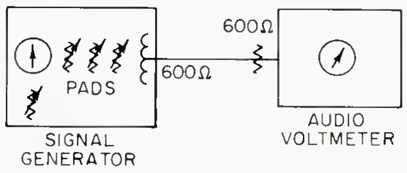
TEST INSTRUMENTS

The signal generator and the measuring devices have a considerable effect on the results of tests. The operator should have a reasonably good understanding of an instrument and its use. But even assuming that the operator does know the test equipment and how to use it, operational errors can occur that change results—for example, forgetting to change the impedance setting or the terminating resistor.

Aside from operating errors, the accuracy of the instruments enter into the accuracy of the test results. A check on the accuracy of the instruments from time to time will remove errors in this regard. Rechecking the calibration of an instrument is especially important if repairs have been made to it which could effect its precision.

Consider, for example, the audio generator with internal metering and calibrated pads. These can be checked out with the voltmeter section of the distortion analyzer or another accurate audio voltmeter. Measure the generator output directly, and use an accurate termination across the input to the voltmeter. Preset the signal level into the pads, then switch each pad in and measure the drop in signal in each case. If you want to know the actual output level the unit delivers when its output impedance is set to other than 600 ohms, switch the gen-

Figure 4. If there is any question about the signal generator output levels and pads, measure these with an audio voltmeter.



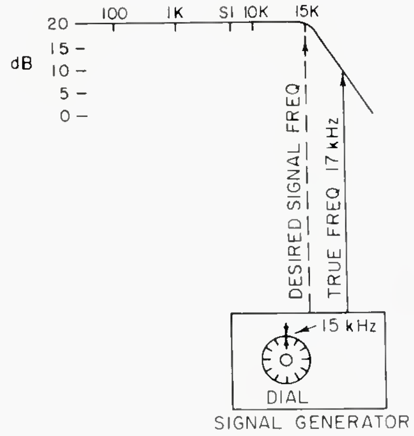


Figure 5. The output frequency of the signal generator may not be what the dial indicates. This could put the test signal on the slope of a filter curve.

erator to 150 ohms; for example, use a 150 ohm terminating resistor on the voltmeter, and again switch in the pads.

When testing an audio unit that contains filters or equalizers, the accuracy of the input signal *frequency* is important. How accurate are the dial settings on your generator? If there is any doubt, set up your frequency counter (if you have one) and actually measure the frequency of the audio signal. Check the dial reading against what the counter indicates. If there is a difference, then mark that place and check at several other places on the dial. If all are off by the same amount, it could be that the knob was not put back on the shaft in the correct place the last time it was off. But if the dial markings do not agree in some places and agree in others, that indicates a problem in the unit itself. Either mark the correct places, or better yet, send the unit back to the factory for repairs and recalibration. But even if you are reasonably sure of the calibration of the dial, if the circuit is a critical one from a frequency standpoint, you can still use the counter to set the generator frequency exactly at what it is supposed to be.

RECAP

Many factors can distort the results of tests so that a true performance measurement is not achieved. Environmental conditions, haywire hookups, procedures and test instruments can all be factors. There can always be operating errors, of course, but if we become careless, we may find that we are only making measurements—not testing the true performance of a unit. ■

JOHN M. WORAM

The Sync Track

More About Maximum Power Transfer

• A recent **db** article on *Understanding Maximum Power Transfer* (March, 1978) really convinced us that people do indeed read the magazine. While the author's math was right on, unfortunately, his choice of a practical example wasn't. As a result, we received an avalanche of mail, offering much in the way of comment on the subject of maximum power transfer.

Many of the letters were several pages long, and it's difficult to single out just one or two for publication, since practically all of them offer at least a little something of interest on the subject. So instead, I've been "volunteered" to try to sort it all out, and will do so by stealing a little from this one and that one, in an effort to straighten out the matter of getting power from here to there.

IMPEDANCE MATCHING

In today's transistorized world, many electronic devices (especially power amplifiers) have very low output impedances, often measured in fractions of an ohm. Years ago, such was not the case. There were any number of "black boxes" around with output impedances of several hundred (600 comes to mind) ohms.

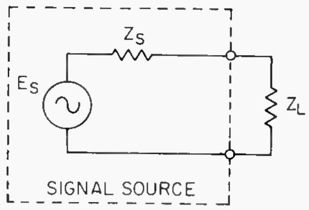
Let's look at such a black box—say, an equalizer—with an output impedance of 600 ohms. Although our equalizer is going to be fed some sort of audio input signal, for the moment we're only interested in its output. So, we can consider it as a 600 ohm "source" (of signal). In other words, looking backwards into its output terminals, we discover a "source impedance" of 600 ohms. In the diagram below, we'll identify this as Z_s. This Z_s is shown in series with the signal source itself, although there is actually no single component inside with such an identity. In other words, you can't go to the source and pick up a new Z_s if the one you have burns out.

Now, in order to achieve maximum power transfer to the next device in the signal path, this next device must have an input impedance that is equal to Z_s. "Impedance matching," it's called, and since this device becomes a "load" across the source, it's shown in our diagram as "Z_L." And that brings up a "trick" exam problem that Rouland-Borg plant manager Don Metz recalls:

Given $E_s = 10$ volts, and $Z_L = 8$ ohms, what should Z_s be, for maximum power transfer? To which the unwary student quickly responds, "Why, Z_s should also be 8 ohms, so that the impedances are matched."

Wrong! If Z_s had been given as 8 ohms, then of course Z_L should be 8 ohms for maximum power transfer. But that wasn't the problem, was it? We were given E_s and Z_L. Since $P = E^2/Z$, we'll get our maximum power delivered if the entire voltage, E_s, is dropped across Z_L. In other words, Z_s must be zero.

A load impedance (Z_L) connected across the output terminals of a signal source. The output impedance of the signal source is represented by Z_s.



Of course, as Mr. Metz points out, the problem is academic, since Z_s comes with the black box, and is not variable in usual real-life situations. In the real world, we are stuck with Z_s, and must choose Z_L accordingly. All else being equal, as Z_L gets larger, the dissipated power gets smaller. But when $Z_L = Z_s$, we do get the maximum possible power transfer, under the prevailing conditions. However, if we could now make Z_s smaller (while Z_L stays put), we'd get even more power delivered to Z_L. But that's not possible, is it?

Byron Roscoe, chief electroacoustical engineer for the David Clark Company, Inc. sums it up concisely for us: *"The maximum power theorem is only valid for the case where we are limited to a given source resistance, and the load resistance can be varied. In such a case, adjusting the load resistance to equal the source resistance will result in maximum power in the load. However, in the amplifier/loudspeaker interface, we are limited in our choice of load impedances by electroacoustical considerations in the speaker system (8 ohms being the nominal value for most stereo loudspeaker systems), and in this case the maximum power transfer occurs when the source resistance is made as small as possible, zero being the ideal value."*

He also notes that, ". . . *modern speaker systems depend on the low source impedance of the amplifier to provide electrical damping* . . ." Roscoe cites the existence of a "damping factor" rating for most amplifiers, from which the actual output (i.e., source) impedance may be determined. He defines damping factor as. ". . . *the ratio of the output voltage (loaded) to the change in output voltage when the load is removed. For example, if a certain amplifier delivers 10 volts to an 8 ohm resistive load, and the voltage rises to 10.125 volts when the load is removed, then the damping factor of the amplifier is* $\frac{10.0}{10.125\text{-}10.0} = 80$, *a typical value.*

"This corresponds to an internal impedance of 0.1 ohm, and is hardly worth taking into account, due to the small percentage of the load impedance it represents."

How did Mr. Roscoe discover the internal impedance (Z_s) was 0.1 ohm? Easy—a little Ohm's Law will do it. With the speaker in place, he measures 10 volts across 8 ohms. $I = E/Z$, so there is a current flow of 1.25 amps. Of course, this current is also flowing through Z_s, so there must be a voltage drop there also. Removing the speaker, he measures 10.125 volts across the amplifier's output terminals. So, he now knows he's losing 0.125 volts across Z_s when the speaker is reconnected. Again, applying Ohm's Law; $Z_s = E/I = 0.125/1.25 = 0.1$ ohm.

DAMPING FACTOR

A. H. Clegg, assistant general manager of Technics by Panasonic's Product Engineering Division draws our attention to the fact that the damping factor may also be measured as Z_L/Z_s. Since we know the Z_s of Mr. Roscoe's amplifier, its damping factor becomes $8/0.1 = 80$; just as it did using the earlier formula.

Professional engineer Walter Jamison mentions that the specs for Crown amplifiers list a damping factor of greater than 200 for 8 ohm loads, giving them a Z_s of $Z_L/DF = 8/200 = 0.04$ ohm.

By now, hopefully everyone is as confused as I am, but the point to come away with is that while speakers (that is, Z_L) remain at 8 ohms, amplifier source impedances (Z_s) are extremely low. Therefore, we don't have to waste a lot of power in Z_s, just to achieve that illusive maximum power transfer.

By the way, the new Institute of High Fidelity document, "Standard Methods of Measurement for Audio Amplifiers (IHF-A-202, 1978)" defines damping factor as 8 ohms/Z_s.

In conclusion, our thanks to all those readers who were kind enough to share their thoughts with us. And, to that one mid-western professor who sent us two copies of his letter, thank you, thank you. ■

MARTIN DICKSTEIN

db Sound With Images

More and More Multi-Imaging

• First, let's differentiate between multi-imaging and multi-media. The former consists of many images on the screen while the latter employs various media, possibly sound or print, or even a mixture of slides and film. Both techniques are obviously related. Multi-imaging does not exclude sound, but is a more precise term to indicate that there are more images to see at one time than a single slide projector can offer.

Although the slide projector seems to be the workhorse of the presentation business, it is limited in its operation. It projects a single image, and then takes about 1½ seconds to change to the next image. In between, there is a black screen. There are times when this type of presentation may prove to be sufficient. In fact, it might even be the best method of presentation in some instances, depending on the material, scope, and purpose of the meeting. However, ever since Expo '67 showed the way, multi-imaging (either with or without sound) has become more and more popular.

It has been said that a picture is worth a thousand words. By simple logic alone, then, does this mean that doubling the number of images doubles the value to 2,000 words? Does four images equal 4,000 words, eight images 8,000 words? What if some of the images are repeats? Does this reduce the value by one-third? Or one-half? Obviously, from this rather simple example, the actual value of multi-imaging cannot be determined precisely. How about divided attention, even when both images are different? Or when there are more images than the viewer can see at once, is attention divided evenly among them?

VARIETY OF STIMULI

It has been found that greater learning takes place with a variety of stimuli. After all, it has been estimated that each one of us gets about 100,000 stimuli each day. The Gestalt psychologists tell us that any perceptual organ responds to a complete whole in some way that cannot be analyzed rather than to the sum of all the individual parts. This gives us one good indication that multi-image is effective, but not necessarily in direct proportion to the number of images displayed. It depends on how the multiple images are used, and what they are.

It has also been shown that other elements play a great part in getting information across. For example, motion. This effect is possible because of the operation of the visual process of the eye and the brain. Movies are not really showing motion. They go by as single frames, each displayed for a finite period of time. At the standard rate of 24 frames per second, the eye seems to see motion. Actually, it retains an image for about 1/10 of a second after the image has disappeared. All we need, then, is ten frames per second for the eye to think it sees motion with no gap. This perceptual peculiarity is known as the *phi phenomenon*, or persistence of vision.

THE PHI PHENOMENON

The phi phenomenon can be simply demonstrated by flipping animation cards, usually made for children in cartoon form, at the proper speed. This was also the principle evidenced in a device used a number of years ago in the form of a rotating circular drum with a series of successive images viewed through a thin slot. You can also simulate motion with a slide projector, using dissolves on single images, but due to the slow process of changing slides, the motion is not as effective as it would be with more than two units. Actually, three or more projectors throwing onto a single screen are the usual basis for an illusion of motion.

Single images projected by a single projector in normal procession can be dull and boring if the material shown is of an uninteresting character. Using a dissolve can improve this to some extent, but if more projectors simulating action are used, the audience's attention can be riveted.

Another alternative to using three projectors on one screen, if motion is not required, is to use three screens. (It has been shown that using two screens is only slightly more effective than employing one screen, except where direct association and comparison are needed.) By utilizing three screens, it is possible to create a "ping pong" effect where the images jump from one screen to another.

This effect makes use of movement, keeping the eyes from staying in one place too long, while allowing the projectionists to change slides without an apparent black space between images. As long as the eyes are kept moving, a black screen in one position is less objectionable than a 1½ second blank space between successive images. Of course, all three screens can be used simultaneously for comparison or addition of information if desired.

TWO PROJECTORS

To make an improvement over the blank screen, the next step is using two projectors on each screen. This allows for dissolve, or fast cut, on each screen with no blank screen at all if it is not desired. The next step up, of course, is three projectors per screen. Now, motion on a single screen as well as on all screens simultaneously, or across the three screens, becomes feasible.

There are other effects (and reason for their use) possible with multi-projector systems, for example, when the visuals that have to be used (perhaps because the client insists) are not as good (either in color or sharpness or quality) as some of the ones newly shot specially for this presentation. By showing the poorer ones quickly in good balance with the better ones, the audience sees them, but does not get the opportunity to inspect them carefully. Perhaps the slides are old and show some part of the product's history or the client's factory back when he started. These can be shown as part of a flash-back sequence as prelude to the present where the slides are sharp and brilliant. Or perhaps the old slides can be reshot into a smaller format than the standard slide. This would increase their brightness somewhat but would allow less scrutiny and could be used as part of a build-up through dissolve or fast cut sequences to a present showing of the same scene. It is also possible to use the old slide in smaller format and build up to a single segmented image of many shots of the same scene. Again, less scrutiny but greater visual effect.

USES OF IRREGULARITY

The employment of the segmented slide introduces an even greater possibility for the use of the multi-projector system. It has been found that nonconformity attracts the eye of the viewer. If an image is off center, the eye is attracted to it because the image within the whole screen is not balanced. This irregularity creates a sense of unrest or stress to the eye. (It's like having a well known sequence of musical notes played up to, but not including, the last note, or like the dropping of one shoe. The ear strains for that final resolution, resulting in conformity and balance.) Paintings have been made with the center of attraction off-center on the canvas to attract the attention to where it was desired by the painter, instead of the natural inclination of the viewer, who looks to the center of the image. Or to be even more exact, the viewer's vision actually is drawn naturally, for example, to the lower left part of the picture.

The segmented slide permits a build by succession to a whole image, or to a full image consisting of individual, but separated parts. When this is done, it might be interesting to know that the human visual perception process starts at the left (for the Western world) and at the right for those whose normal reading starts at the right. Thus, for the American audience, the portion of the image that creates the highest stress is in the upper right part of the screen. An image on the screen at this position will bear more "weight" and be remembered more than one placed in the more expected part of the screen.

WHOLES MUST BE MADE OF PARTS

It is also well known that the human eye and mind combine to fill in what is left out. Wholes must be made of parts. Thus, the concept of giving image forms to the constellations was born in ancient times. The same effect takes place when images are placed next to each other. Groups are formed in the mind of the viewer. Groups are also formed from similarity; where images look alike in shape, they are automatically put together in the viewer's mind. The same thing happens when images of similar size or color are found. The eye groups them together. In like manner, the eye seems to perceive a grouping when objects are located in a line of direction. Thus, by sequencing objects (or images) in a direction, motion can be implied.

Motion can be simulated by dissolving (or cutting) from one size to another. Going from big, to small, to disappearance shows the object receding into the distance. Building of size means coming closer. Movement can also be from one side of a screen (or across three screens) to the other. A similar sequence could be effected with changes in color or brightness.

Although it has been shown that the eye can process over four million bits of information in a single second, there is a limit to the number of eye/mind boggling effects that can (or should) be shown in a single sitting. A time limit usually used as a guide is ten minutes. Beyond this, the audience becomes restless and tired; less is learned, or even visually accepted, in succeeding minutes.

There are a limitless number of effects possible, with more screens, more images, more segmentation, more variation in size and/or shape and/or color, and there are no shortcuts to learning the limitations or capabilities of the systems available or the acceptances of effects by the audience. Using balance and stress, movement, color, image brightness, shape changes and complexity (people will look at a complicated image rather than a simple one), the presentation producer is limited only by his own imagination and creativity in getting his message across. However, the most valuable of the creators' talents might well be the judgment he or she uses in selecting images and determining how they are presented on the screen or screens. ■

More on Matching

• In the February issue, this column discussed damping, including aspects of matching. Then in March, Dan Keen's article *Matching for Maximum Power Transfer*, presented a quite different aspect of the same question, perhaps a little too concisely. This is an old question that has been covered again and again, but there are always new people coming along who have not yet understood it completely.

First, we will show the conditions under which Dan Keen's formula, which is the classical one, applies. Then we will deal with its limitations. But first, we should comment on what the output rating of an amplifier means. This is where the damping factor comes in.

Suppose an amplifier has an output *rating*—it is not correct to call it an impedance—of 8 ohms. This means it is designed to feed a load of 8 ohms. Then if its damping factor is given as, say 20, this means its output *internal* impedance is 8 divided by 20, or 0.4 ohm. Now, if the rated output is, say 200 watts, then it will give 200 watts into 8 ohms output load. That would be 40 volts. (40 squared, divided by 8, is 200.)

Now, using an audio oscillator, after checking that it delivers 40 volts into 8 ohms, turn the input down 20 dB so the output voltage into 8 ohms is 4 volts instead of 40. That will be 2 watts, instead of 200. And if you remove the 8 ohm load, because the damping factor is 20, the voltage will rise to 4.2, instead of 4.

Now if, without changing the input level, you apply a matching load of

0.4 ohms, it will have 2.1 volts across it. Squaring 2.1 and dividing by 0.4 gives 11.025 watts, which is more than the 2 watts delivered into 8 ohms for that input. In fact, *for that input*, 11,025 watts is the maximum output, or power transfer. Now does that work at maximum output?

MAXIMUM OUTPUT

The amplifier is designed to deliver 40 volts into 8 ohms, which is a maximum current of 5 amps. That is how it gets its 200 watts: 40 volts at 5 amps. This usually means the amplifier will not deliver more than 40 volts *or* 5 amps without running into distortion or destroying the output transistors.

When you connect 0.4 ohms to the output, at a level of 2.1 volts, it will take 2.1 divided by 0.4, which is 5.25 amps, which may just "get by." So this means that, under the maximum power transfer condition, although it delivers 11.025 watts maximum for that input level—probably 11 watts is the real maximum—the amplifier is really capable of 200 watts output.

Now let us look at what changing the output impedance does for maximum output, as opposed to maximum power transfer. Our hypothetical amplifier has a maximum output capability of 40 volts, or 5 amps. When you use both of those maxima, you need 8 ohms, to get 200 watts. But if you use only 4 ohms, then your output will be limited to 20 volts, to give the allowed 5 amps, and that is 100 watts, instead of 200.

And if you go to 16 ohms, your maximum output is still 40 volts, but you only take 2.5 amps, again reducing your output to 100 watts, instead of 200. Dan Keen's article correctly said it is better to err on the high side, if you do not use the rated impedance. Let us see why that is. Mismatching either way, by a factor of 2 to 1, reduces the available output to half. Why would one way be better than the other?

MISMATCH UP OR DOWN?

Because of the low internal impedance, which we have assumed to be 0.4 ohm for an output rated impedance of 8 ohms, the output voltage is more directly related to input level than output current is. If you put in the input required to give 40 volts into 8 ohms, and change the impedance to 16 ohms, the voltage will rise from 40 to about 41. The current will drop from 5 amps to about 2.56 amps, and the power will be slightly more than 100 watts, or about 105 watts.

Going above 40 volts may introduce slightly more distortion than staying at 40 volts. But you will not need to turn level down much to reduce the output level from 41 volts to 40 volts.

Now, if you take off the 8 ohm load and substitute a 4 ohm load, assuming the output stage of the amplifier will still deliver it, the 40 volts will drop only to about 38 volts, which would be getting close to 400 watts. But the current would be 10 amps which, when the rated maximum output from the output stage is 5 amps, means something else has got to happen.

What happens depends on other features of the amplifier's design. It may be that the voltage will drop, and go very distorted, along with the current, which will not go much over the 5 amp maximum. It may be that the output stage will burn out. Or it may be that a protective device will take control, so that the output is disrupted in whatever way that particular protective device operates.

In any event, you can see that it is much safer to err on the high side, in which the only effect is a reduction in maximum power available. Before leaving the question of loudspeaker matching, it would be well to mention something else that I have mentioned before, that the nominal impedance of a loudspeaker, or even of many of them connected as a total load for an amplifier, is far from being a constant resistance of the value named by the rated impedance.

As the ideal arrangement is for impedance to err on the high side, it would be ideal for a loudspeaker rated at 8 ohms to have that value as a minimum. This will usually be in the range between about 600 and 1200 hertz. Below that, the impedance will rise to a peak at its main resonance. This might be somewhere between 30 and 120 hertz, depending on unit design, and how it is mounted in its enclosure. And the impedance at that frequency might rise from twice to perhaps five times or more.

So the impedance might be 8 ohms at some frequency between 600 and 1200 hertz, rising to from 24 to 40 ohms, or maybe more, at some frequency between 30 and 120 hertz. And the impedance rises again, above the low point between 600 and 1200 hertz.

Matched in that way, the amplifier would deliver its maximum power at some frequency between 600 and 1200 hertz, and something less than maximum at all other frequencies. But the overall response would be flat, which is what matters, because it would deliver substantially the same voltage to the loudspeaker at all these frequencies, although the power taken would vary.

REAL-LIFE LOUDSPEAKERS

Is that how it is? Not necessarily. Loudspeaker makers are concerned with efficiency; they want their units to be competitive with other makes. And it could be argued that the impedance rating should be an average figure, or a mean figure, rather than the minimum value. So if the voice coil is wound so the minimum impedance, somewhere between 600 and 1200 hertz, is, say, 5 ohms rising to between 15 and 25 or 30 at main resonance, and again at higher frequencies, the unit will sound louder for the same apparent level delivered from the amplifier.

It may be slightly overloading the amplifier over the range of frequencies where the impedance is lower than 8 ohms. But human hearing is not far from its maximum sensitivity over this range, so the level at these frequencies seldom hits maximum on any form of program material. Where more energy is needed at the lower frequencies, the impedance is higher, probably still well over the nominal 8 ohms, so the amplifier can thus deliver more power to the unit at frequencies where it is really needed.

As I have found so many times before, everything is a compromise in audio. You figure something one way, and there seems to be a right way to do it. But when you look a little closer, there may be justification for the way most manufacturers really do it, although at first that may have looked like cheating.

DIRECT CURRENT

Dan Keen's article pointed out that the maximum power transfer principle applies to both a.c. and d.c. Of course all audio is a.c., just the frequency changes, over the enormous range from 20 hertz to 20,000 hertz. And as we have seen, that is part of what complicates matters. Now how about d.c.?

A battery is the most common form of d.c. source. Suppose we consider a 12 volt car battery. On a cold morning, pressing the starter button may cause the voltage to dip from its open-circuit 12, or a little more, to perhaps 6 volts, as the starter drains perhaps 500 amps from the battery.

That means the starter is taking 3,000 watts to turn the engine over, which figures to about 4 horsepower. Because the voltage drops to half, you are effecting maximum power transfer. This means that 4 horsepower is the most that you could get out of that battery, on such a short-term rating as turning your motor over on a cold morning. But what will that do to the battery if you sustain it for long?

The internal voltage of the battery is still 12. The other 6 volts are lost, inside the battery. This means that, as well as the 3,000 watts delivered to the starter, another 3,000 watts is doing something inside your battery. It could be just developing heat, and 3 kilowatts is a substantial amount of heat to be generated in a box as small as your car battery. It could also be producing forces that would buckle the plates.

The combined effect of the heat and the forces of all that current will be disastrous to most batteries unless it is limited to a short duration. That relates to a d.c. source of power. How about our conventional power source, the a.c. line? The voltage at our house input terminals, is supposed to be 115. Suppose that at the substation terminals it is 125. Then to draw 115 volts at your house there must be sufficient voltage drop in the line between the substation and your house of 10 volts.

But according to the maximum power transfer theory, you should load the line down, till you only get 62.5 volts. You would not be very happy with that, would you? And the lines between the substation and your house would probably get quite warm, if you ever achieved that!

So maximum power transfer can really have a variety of meanings. What we really want, is to get the maximum power delivered where we want it. The maximum power transfer theory always dissipates half the power getting it there, one way or another, and delivers half the total available power. We would really prefer to deliver more than half the total power we have, or produce.

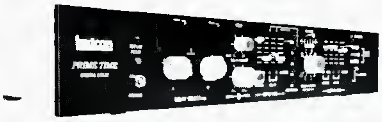
What this all points out, once again, is the importance of being quite specific about what you are saying. When we talk about maximum power transfer, we really start with some assumed conditions. We need to spell out very carefully what that condition is. ■

db New Products & Services

DIGITAL DELAY PROCESSOR

● Designed for recording studios and live entertainment, Prime Time audio time delay processor combines digital audio delays, vco time base processing, and complete mixing facilities in a single package with self-contained power supply. The mixer section allows use with an outboard reverb unit or other signal processor. An echo return mix is created by a combination of the two internal delays, input, and an auxiliary return. Delays can be recirculated by way of a second mixing bus for reverb effects. Special effects possible include a doppler pitch shift, natural double and triple tracking, flanging, and vibrato via the vco time base modulator. A delay multiplier control provides extended delays, up to two seconds; with a repeat/hold control, a rhythm track can be repeated indefinitely with no degradation of the original audio signal. The major dynamic functions are remoteable to a performer's foot pedal controls.

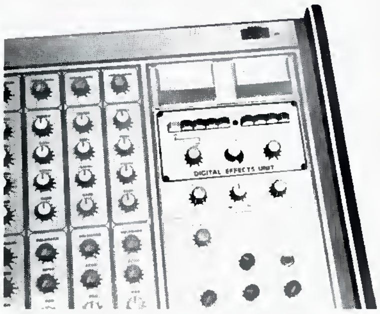
Mfr: Lexicon, Inc.
Price: $1,485. Memory Extension: $175.
Circle 50 on Reader Service Card



ECHO EFFECTS MODULE

● The feeling of power that comes from generating a wide variety of sound effects is available to the sound mixer who can create, with the push of a button, echo, reverb, flanging, vibrato, and phasing with the assistance of the CCD echo effects module. The module's "charge coupled device" is touted to eliminate the problems of poor bandwidth and high noise levels. Echo select consists of four basic delay times on pushbuttons, which may be used individually or in any one of fifteen combinations. The first button produces a very fast single repeat, ADT (Automatic Double Tracking) and other features include variab'- delay control, vibrato and vibrato speed control, as well as repeat and repeat volume. The module fits into either the HH Stereo 8 or Stereo 12 mixers.

Mfr: HH Electronics, (Audiomarketing, Ltd.)
Circle 51 on Reader Service Card



"FOR THE FIRST TIME RECORD PEOPLE AND MOVIE PEOPLE HAVE REALLY PULLED IT TOGETHER."

Quincy Jones

It's not surprising that Quincy Jones sometimes feels like he was born in a studio. He's performed on, composed for, or produced over a thousand albums. Right now he's finishing his first musical, Sidney Lumet's version of The Wizard of Oz, The Wiz, starring Diana Ross.

While Quincy is one Jones that's impossible to keep up with, we were able to catch him briefly to find out his views on the current recording scene, his latest work, and "Scotch" 250 Mastering Tape.

The only thing Dizzy Gillespie, Andy Williams, Peggy Lee, and Ringo Starr have in common is that they've all worked with you. How can you work in so many musical styles?

"I don't get hung up in any bags. When I was studying in Paris, a teacher told me once, there were only twelve notes, so you should find out what everybody's done with them, because they're the same twelve notes that Palestrina was scuffling with. So I can live with the best of all different areas. I like that, you know. The menu is broad, man—eat everything."

There are a lot of movie scores that have turned into some pretty hot albums lately, Saturday Night Fever, for example...

"You know why I think it's happening? It's just a guess... for the first time record people and film people are basically the same people and they've really pulled it together.

"Of all the films I did, the thing that bugged me the most was that we'd be in the studio and the music would boom down at you, and when you got to the theatre it was almost like a rumor, all the bottom end and the top end falls off. Then Dolby came along and they got *A Star is Born, Star Wars, Close Encounters,* and *Saturday Night Fever.*

"Those are successful record-wise because for the first time people actually hear the music in the track, really hear it. We've got a new kind of sound system now with Dolby. Emotionally it hits you from a place you're not even aware of."

Is it technically harder to achieve what you want in a musical as opposed to doing a score for a dramatic film?

"Oh yeah, in *The Wiz* we've got choral things that go up to 80 and 120 voices, so to get a good lip sync we decided to use just two voices for guide tracks, almost like a Polaroid. After their mouths are moving in the right way, then we sit down and put the sweetening on the dance and singing numbers."

So the music is composed simultaneously with the filming?

"They've been sending me out dailies on videotape from New York because the color really turns me on. You get it at 2 o'clock in the morning and look at the reel about ten times. You have to eat it. That's the best homework you can do for a film."

You're a big user of "Scotch" 250. Do you find that it has a clean sound? That's one of the things we've been selling the tape on.

"That's right.

"It's like with film stock, you know. When you've got 800 people out there on a set, I don't care what happens on that performance, if it isn't recorded on camera, it's all over. And it's the same in the recording studio; everything else is superfluous.

"No matter how great a song we get, or performance or balance or anything else, if that same thing isn't reproduced and captured on that tape, nothing we do means a thing,

"That's why we stay with 'Scotch.'"

"SCOTCH" 250 MASTERING TAPE.

The tape the masters use.

3M COMPANY

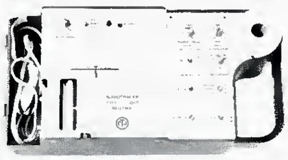
QUALITY COMPACT SPEAKER

● Sophisticated components, usually found in larger speakers, are included in the Impulse stereo system, measuring 24 x 14 x 9 inches. The speaker contains a vinyl-coated, lightweight spun aluminum cone mid-range and offers a frequency response of better than 39-21 kHz ± 3 dB, with elimination of phase errors within the frequency area where its computerized transducers are intended to be used. It provides for moderate efficiency, with a high power handling capacity of 200W rms/program material, can be driven with an amplifier using as low as 15 watts.
Mfr: Kustom Acoustics
Price: $199.00.
Circle 52 on Reader Service Card

THREE SPRING REVERB

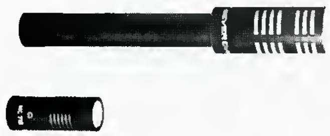
● The same tonal characteristic as created by putting two Type-4 reverbs in parallel is achieved through the three-spring design of Type-9 reverb, intended to accompany high quality musical instruments, guitar amplifiers, mixers, and studio reverb applications. The manufacturer claims that the addition of the third spring smooths out the peaks and valleys, offering a cleaner reverb. For especially effective application, it's recommended that the Type-9 be used with a BBD or CCD electronic design equipment.
Mfr: Acoutronics
Circle 53 on Reader Service Card

AUDIOTRACER



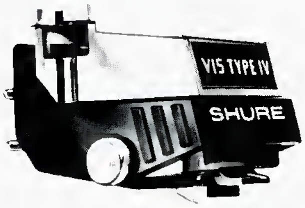
● An available permanent record of the level response of any audio device or system, electronic or electroacoustic, is possible through the use of the 3201 Audiotracer, which measures and thermographically records audio-frequency or time phenomena versus linear or dB amplitude. The unit consists of a voltage-controlled oscillator with switchable dual-range swept output, 5 Hz "warble" generator with switchable-width f.m. of vco; switchable 1 kHz reference oscillator; output amplifier with rms drive capability of 3 watts into 4.5 ohms and continuously adjustable output level; input amplifier; motional-feedback pen-drive amplifier and galvanometer movement with switchable amplitude range and switchable writing speed; electronically controlled paper-drive mechanism with adjustable paper speed; ceramic-tipped d.c. heated writing pen. The unit weighs 3.75 lbs. and comes with a carrying case and battery pack for field use.
Mfr: Philips Audio Video Systems Corp.
Circle 54 on Reader Service Card

DYNAMIC CONDENSER SERIES

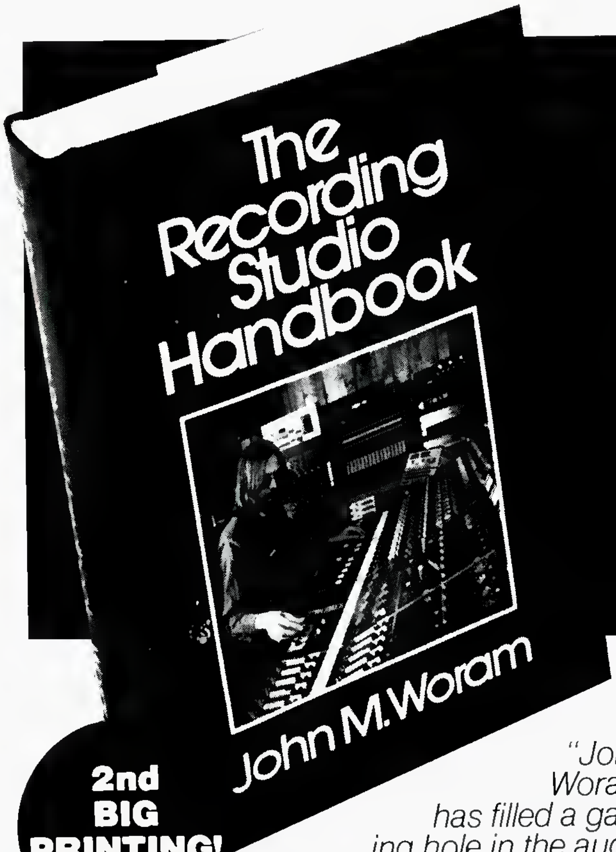


● This versatile 48V phantom-powered condensor series consists of one preamp/shaft (HV710) and four interchangeable head capsules (CK711-714). The capsules include two omnidirectional and two cardioid patterns, with one of each pattern containing a windscreen. External power supplies for both balanced and unbalanced operation are provided. A lapel clip-on condensor has its own 18V power supply.
Mfr: Beyer Dynamics
Circle 55 on Reader Service Card

STYLUS ASSEMBLY



● A revamped design aimed at correcting stylus problems is the basis for the V15 Type IV pickup system. The total mass has been reduced, utilizing a telescoped shank structure and a lightweight, high-energy magnet, which improve trackability in the mid and high frequencies. A two-function bearing system is independently optimized for low frequencies and high frequencies. A hyper-elliptical nude diamond tip creates an elongated tip-groove contact area, with a claimed 25 per cent reduction in distortion over a conventional stylus. Other important features are a built-in viscous-damped dynamic stabilizer which combines with the stylus assembly to raise the arm-cartridge resonance frequency and attenuate the arm-cartridge system resonance effect, resisting sudden warp-caused changes in motion; a brush array of electrically conductive fibers to sweep away static electricity from record grooves; and a dynamic stabilizer which acts as a shock absorber.
Mfr: Shure Bros., Inc.
Price: $150.
Circle 56 on Reader Service Card



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And the Journal of the Audio Enginering Society said: ". . . a very useful guide for anyone seriously concerned with the magnetic recording of sound."

The Recording Studio Handbook
by John Woram

The technique of creative sound recording has never been more complex than it is today. The proliferation of new devices and techniques require the recording engineer to operate on a level of creativity somewhere between that of a technical superman and a virtuoso knob-twirler. This is a difficult and challenging road. But John Woram's book charts the way.

The Recording Studio Handbook is an indispensable guide. It is the audio industry's first complete handbook that deals with every important aspect of recording technology.

Here are the eighteen chapters:

- **The Decibel**
- **Sound**
- **Microphone Design**
- **Microphone Technique**
- **Loudspeakers**
- **Echo and Reverberation**
- **Equalizers**
- **Compressors, Limiters and Expanders**
- **Flanging and Phasing**
- **Tape and Tape Recorder Fundamentals**
- **Magnetic Recording Tape**
- **The Tape Recorder**
- **Tape Recorder Alignment**
- **Noise and Noise Reduction Principles**
- **Studio Noise Reduction Systems**
- **The Modern Recording Studio Console**
- **The Recording Session**
- **The Mixdown Session**

This hard cover text has been selected by several universities for their audio training programs. With 496 pages and hundreds of illustrations, photographs and drawings, it is an absolutely indispensable tool for anyone interested in the current state of the recording art.

Use the coupon at the right to order your copy of *The Recording Studio Handbook.* The price is only $35.00, and there's a 15-day money-back guarantee.

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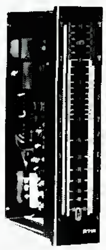
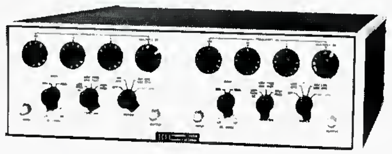
PEAK PROGRAM METER



• This version of the peak program meter utilizes a 2-channel bar-graph display, available with 100-200 and 300 element spreads. Some versions emit an orange display with this changing to red above "0" level. Designated RTW, the meter has been designed with a minimum of mechanical movement, thereby decreasing the possibility of distortion created by the unit itself. The device offers a total of eight configurations, four with additional dB gain. Measuring range is —50 dB to + 5 dB.
Mfr: Track Audio, Inc.
Circle 57 on Reader Service Card

LOW FREQUENCY FILTER



• Ultra low frequencies are within the province of Model 3343 dual channel filter. The device is digitally tuned over the range from 0.01 Hz to 99.9 kHz and has 48 dB slope per octave. 96 dB per octave can be achieved when both channels are operated in the same mode, at the same cutoff frequency setting and cascaded. Channels can be operated in either high-pass or low-pass mode, giving a frequency response of an eight order Butterworth. A front panel switch changes the frequency response to r.c. optimum for transient filtering. Switch tuning permits cutoff frequency calibration accuracy of ± 2 per cent and 3-digit resolution. The unit operates with batteries or from an a.c. line.
Mfr: Krohn-Hite
Price: $2,000.
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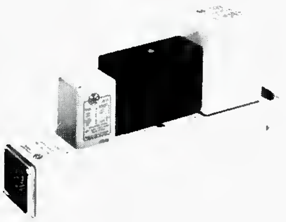
SOLID STATE RETROFIT



• Langevin 116 retrofit—30 dBm mic line or 40 dBm power amplifier, has a frequency response of 30 Hz to 15 kHz. Claimed thd is 0.25 per cent and input noise is —127 dBm. Gain is +45 dB (adjustable). Model 117, solid state 39 dBm line amplifier has a frequency response of 50 Hz to 10 kHz (± 1 dB).
Mfr: Opamp Labs Inc.
Price: $150.00.
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MINI LIMITER

• Compact Mini-Limiter measures 12¼ x 1½ x 4½ in. and weighs four pounds, is designed for p.a. or live recording. Inputs on the single-channel limiter are provided for line and low-Z balanced microphones. The front panel controls include an input level slide and five pushbuttons, controlling line or mic selection, limiting, slow or fast attack and two for release. An l.e.d. indicator is activated when 3 dB or more of limiting occurs. The mic input is an XLR connector; other single jacks are standard phone jacks leading to line input, stereo link connection, output for zero dBm and an output for —30 dBm.
Mfr: Audiomarketing Ltd.
Price: $250.00.
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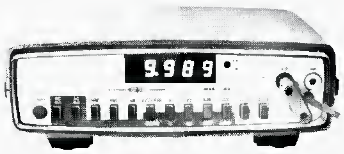
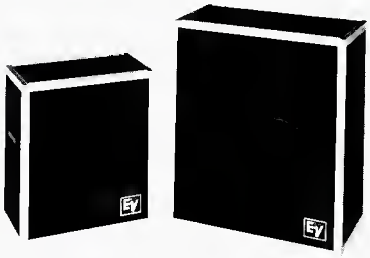
PORTABLE MULTIMETERS



• Applicable for both bench and portable field use, Models 7141A/B 4½ digit multimeters are available with a choice of current or dBm measuring modes. All models feature auto and manual range selection; d.c. voltage from ± 10 microvolts to ± 1000V is measured in five ranges, a.c. voltage measurements from 10 microvolts to 750V in five ranges. A true rms a.c. converter permits accurate measurements of triangles, pulses, square waves, or distorted sine waves up to 20 kHz. The resistance mode offers six ranges allowing measurements from 0.01 ohms to 20 megohms. The units will also measure dBm from —60 to +60 dBm in five manual ranges. Model 7141A has a d.c. accuracy of ± 0.02 per cent of reading ± 0.05 per cent of full scale for six months. Model 7141B offers a d.c. accuracy of ± 0.02 per cent of reading ± 0.01 per cent of full scale. Options include an analog meter and a battery pack.
Mfr: Systron-Donner Corp.
Price: $395-$500.
Circle 61 on Reader Service Card

STAGE SPEAKER SYSTEMS

● Theories previously restricted to low frequency reproduction have been applied to a vented midrange cone speaker on the S15-3 three-way loudspeaker. High sound pressure levels up to 116 dB can be achieved without the use of a horn midrange driver. This permits the unit to be more compact than would be necessary with the usual large horn. It is claimed that the units are virtually flat down to 50 Hz. The upper limit of S15-3 and its companion two-way speaker, S12-2, is 16,000. The units are designed for touring, including the extra touch of a metal grille cloth.
Mfr: Electro-Voice
Price: S-12-2: $350. S15-3: $550.
Circle 62 on Reader Service Card



A.M. PROCESSING

● Maximum improvement of a.m. broadcast signal is claimed for the OPTIMOD-AM, model 9000 audio processing system. Placed between the console and transmitter audio input, the system processes the signal completely through a series of six basic blocks, consisting of a six-band limiter, a polarity follower, an input conditioning filter, a broadband compressor, a program equalizer, and a peak limiting circuit. When required, these functions can be bypassed for proof of performance. The unit comes with a rear-panel jack which will accept an adapter device for a.m. stereo.
Mfr: Orban Associates, Inc.
Price: $3,995.
Circle 63 on Reader Service Card

MANUAL MULTIMETER

● An option of automatic or manual range selection of a.c. and d.c. voltage ranges between 2.0V and 1000V, as well as resistance ranges between 2kΩ and 20MΩ are possible with Model 462 3½-digit multimeter. Manual selection adds a.c. and d.c. voltage ranges of 200 mV and four current measurement levels, both a.c. and d.c., between 2 mA and 2000 mA, totaling 23 ranges. The compact unit, designed for either bench or portable use, is powered by batteries. Total weight is 1 lb. 8 oz.
Mfr: Simpson Electric Co.
Price: $185.00.
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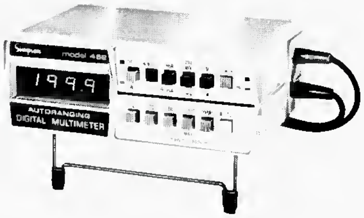

db New Literature

CIRCUIT ANALYZER

A 6-page brochure, describing the Model 123 Circuit/Analyzer covers the major components of the system, suggested applications, and specifications. Mfr: Accutest, 25 Industrial Ave., Chelmsford, Mass. 01824.

ELECTRONICS BOOKS

Slanted toward technicians, engineers, and hobbyists, this publisher's catalog includes titles encompassing reference and instructional material. Mfr: Parker Publishing Company, Inc., W. Nyack, N.Y. 10994.

A/V PRODUCTS

More than 290 accessories, such as storage, retrieval, security, and mobility equipment for a/v use are listed in this catalog. Mfr. H. Wilson Corp., 555 W. Taft Dr., South Holland, Ill. 60473.

PHOTOPHONE SYSTEMS

The linkage of sound and pictures in the post-production operations of television and motion picture creation is facilitated by servo systems described in this brochure, offering product information and a system planning guide. Mfr: RCA Photophone Systems, 2700 W. Olive Ave., Burbank, Ca. 91505.

SOUND LEVEL METERS

Equipment to measure noise and vibration is described in a 19-page booklet, including color photos and a handy chart for matching model numbers with specific applications. Mfr: B & K Instruments, Inc., 5111 W. 164th St., Cleveland, Ohio 44142.

A.V./LIBRARY FURNITURE

A full-color catalog lists a collection of a.v., library, and television adjunct furniture. Mfr: Bretford Mfg. Co., Inc., 9715 Soreng Ave., Schiller Park, Ill. 60176.

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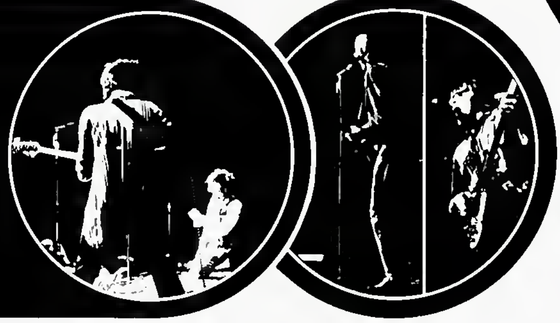

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Editorial

On Microphones and Paint Brushes

"The audio engineer and his microphone may be compared to the artist and his paint brush."
Lou Burroughs
MICROPHONES:
DESIGN AND APPLICATION

That being so, is there anything to be said here that will guide the engineer in his quest for the ultimate microphone? Perhaps not. In the real world, there are neither all-perfect microphones nor paint brushes. To get through the job at hand, you'll need some selection (of brushes or mics, depending on what you're doing). And the selection must be left up to the artist (you), not to us.

But to make that selection intelligently, a little more knowledge may come in handy. And so, we've asked some of the experts on the subject to share their thoughts with us, and with you.

For instance, what about the electret condenser microphone? Not so many years ago, these were little more than laboratory curiosities. But now, some authorities expect the elecret to eventually replace other condenser designs entirely.

Robert B. Schulein—an experienced electret-watcher—describes some of the basics of electret technology in **Electrets and Condenser Microphones.** This, our "cover story," is based on his participation in the development of the SM-81, Shure Brothers' entry into the manufacture of electrets.

Next, J. Howard Smith describes the unique **Sound Field Microphone,** which gives the user as many as four separate outputs. Lately, there has been renewed interest in simpler miking techniques, as some of us re-discover that, when it comes to microphones, "more" is not *always* synonymous with "better." For example, most sound sources occupy space as well as direction, a point (sorry about that!) which the Sound Field technique takes into consideration, and which—very often—multi-miking does not. Of course, if you *must* "fix it in the mix," that's possible too. But that's Smith's story, and you'll have to turn to his article for more information.

The Sound Field Microphone achieves its versatility by manipulating "first-order directivity characteristics"—a little phrase that perhaps is not on the tip of everyone's tongue. Our application note on **Plotting Polar Patterns** may be of some help here. For those readers with more math than microphones, there should be no trouble combining polar patterns to come up with still other polar patterns—on paper, anyway.

And, once you've discovered your own miracle microphone, don't forget that you'll need a good (quiet!) preamplifier to do it justice. J. W. Dorner reminds us that resistors are not totally passive devices. It turns out the little devils are active noise makers, especially when things get hot. So, unless you do all your recording at absolute zero, don't overlook **Input Noise in Microphone Preamplifiers.**

We'll have a bit more to say about microphones next month, but now we move on to Los Angeles for a look at **The 60th Audio Engineering Society Convention,** held there in May of this year.

And, to bring the issue to a close, Marshall King brings us Part II of his **The Technician and His Union,** a look at the trials and tribulations of "getting organized." Basement studios take note—it could happen to you too! J.M.W.

A NOTE FROM THE PUBLISHER

Effective with this issue, John Woram becomes editor of **db** Magazine. He's actually been at the job for a while now, and this issue represents the full effort that makes this recognition due. Look for even bigger and better **db**'s in the future. L.Z.

ROBERT B. SCHULEIN

Electrets and Condenser Microphones

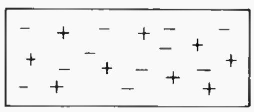
Improved materials, resistant to environmental decay, have brought the electret microphone into practical use

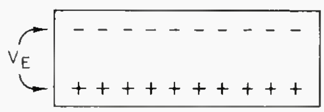
In 1935, when Shure Brothers introduced its Model 40 series of condenser microphones, the term "electret" was already about 50 years old. In 1885, Oliver Heaviside coined the term to describe an electrified insulator with its two ends oppositely charged. Electret (ELECTRicity + magnET) was chosen simply as the electrostatic analog to the word magnet. It wasn't until the 1920's, however, that the concept and its applications attracted serious consideration, as investigators became more interested in producing electrets with the ability to hold a charge for long periods of time.

In Japan, Professor Motoaro Eguchi fabricated electrets, using a combination of carnauba wax and rosin. During the Second World War, the Japanese used these in some of their military communications microphones. Samples recovered in the Pacific puzzled many of the Allied technical personnel who, unfamiliar with elecret theory, tried to determine their principle of operation.

Yet, as early as 1938, wax electret condenser microphones had been sold in the United States, under the name "No-voltage Velotron." However, as the Japanese discovered during the war, all of these microphones had the serious problem of poor environmental stability. For at this stage in the development of the electret, the microphone lost its charge when exposed to unfavorable—

Figure 1. The principle of an electret. At (A), uncharged electret material; at (B) charged electret material.


(A)



(B)

Robert B. Schulein is Chief Development Engineer at Shure Brothers, Inc., of Evanston, Ill.

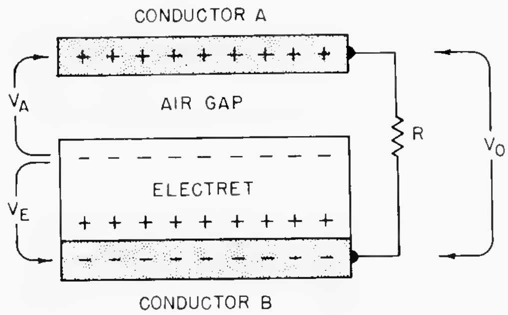


Figure 2. A condenser microphone with electret bias.

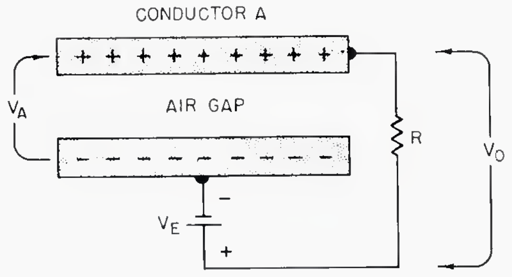


Figure 3. A condenser microphone with external bias.

though realistic—conditions of temperature and humidity. This was in contrast to the excellent charge stability demonstrated under controlled laboratory conditions. And so, for the time being at least, the electret was put aside. In fact, it was not until the early 1960's that the electret materials in use today were finally identified and evaluated.

At Bell Telephone Laboratories, G. M. Sessler and J. E. West suggested the use of Mylar* (in 1962) and Teflon* (in 1965) as electret materials. At the time, it was discovered that Mylar has good mechanical properties, making it suitable as a microphone diaphragm. However, in humid environments, its performance as an electret turned out to be poor. Teflon, on the other hand, has excellent charge-storage properties, though it lacks the mechanical properties of Mylar.

During this period, Shure Brothers began its own comprehensive electret research program, investigating the means of providing a stable bias voltage for condenser microphones. The following discussion reviews the most significant aspects of this work, as it applies to the development of high-quality condenser microphones, suitable for use in demanding professional applications.

ELECTRET PRINCIPLES

In describing condenser microphones, there has been much confusion of terminology. Over the years, terms like condenser, true condenser, air condenser, electret, and electret condenser have been used, and it is unfortunate that so many have been applied, just to describe two particular designs that are actually quite similar.

In both, the transducer is a variable capacitor (instead of a moving coil or ribbon), and they differ only in the method by which that capacitor is biased. Either the bias voltage is built-in (supplied by the electret material) or, it is furnished by some external d.c. source. In either case, the theoretical performance of the resulting condenser microphone should not be influenced by the method of bias.

Before discussing the electret concept in more detail, a few words may be in order regarding why any condenser microphone requires a charging source, or bias, in the first place.

CAPACITANCE

The concept of capacitance (C) is based upon the relationship between Q and V. Q is the charge stored upon, and V is the voltage between, two conductive elements. The formula for capacitance may be written: $C = Q/V$.

For a parallel-plate capacitor, as in the case of a condenser microphone, capacitance may be expressed in terms of the dimensions and material of the capacitor. Specifically; $C = EA/d$. Here, E is related to the dielectric between the capacitor plates, while A is the plate area, and d is the spacing between the plates. If these two expressions are combined, the following formula may be written for the voltage across the capacitor; $V = Qd/EA$.

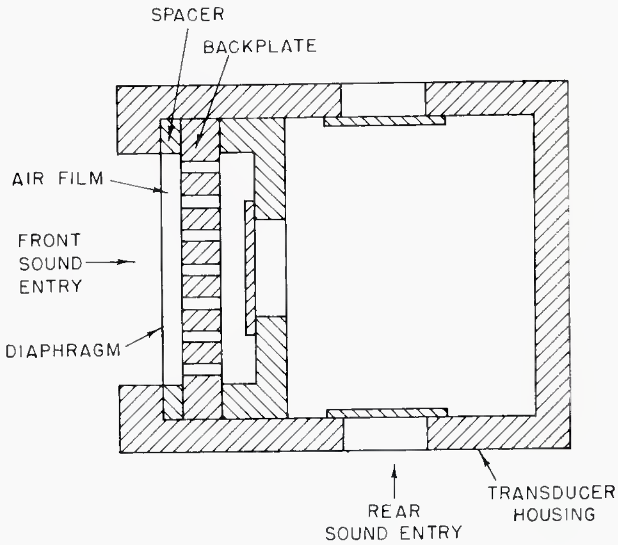
It is this relationship that suggests how a condenser microphone can be made. For if the charge on the capacitor (Q) is held constant, the voltage will vary directly with the spacing of the elements (d). Therefore, the purpose of the bias supply is to maintain this constant charge on the capacitor plates.

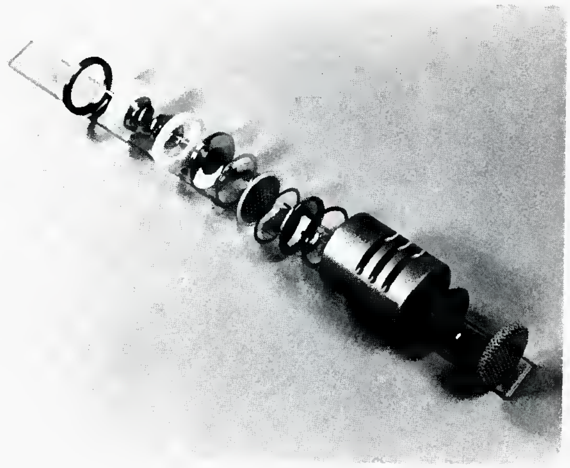
UNDERSTANDING THE ELECTRET

To understand just what an electret is and how it can be used to bias a condenser microphone, consider the electret materials shown in FIGURE 1, where two samples of an insulating material with electret properties are shown. One is in its normal uncharged state and the other is charged. In the charged state, the material has been altered so that an excess of positive and negative charges exists on opposite surfaces. As a result of this charge redistribution—and dependent upon its magnitude—an effective bias voltage (V_E) can be considered to exist across the electret.

Now, if the material is sufficiently thick and a very good insulator, it should be difficult for the charges to drift back toward each other, which would return the material to its initial uncharged state. It is this particular

Figure 4. A simplified cross-sectional view of a uni-directional condenser transducer.





Exploded view of a condenser transducer using an electret charge layer.

decay process—strongly influenced by temperature, humidity, and various forms of contamination—that has received so much attention since the early electret experiments of the 1920's. The best electret materials are, of course, those in which the charge migration process occurs very slowly, and those which are not seriously influenced by high temperature or humid environments.

In FIGURE 2, we have taken a charged electret layer and made electrical contact with one surface (conductor B). The opposite surface is placed in close proximity to conductor A. A capacitor is thus formed. Now, if a resistor (R) is connected between the two conductors, a condenser-transducer, with output voltage V_O is defined. Just as Kirchhoff's Voltage Law can be used to determine unknown voltages in electronic circuits, it can be shown that a potential difference of V_A (the bias voltage) exists across the air gap in the condenser-transducer. In the steady-state condition, no current will flow through resistor R, and therefore the potential difference between conductors A and B will be zero. Consequently, the potential difference across the air gap (V_A) must be equal, but opposite, to that of the effective voltage across the electret material (V_E).

If a relative motion is now imparted to the capacitor plates (by moving conductor A, for example) there will be a tendency for the steady-state charge condition to change. On the other hand, if resistor R is very large (say 2,000 Megohms), the charge on the capacitor will not have sufficient time to change value. Therefore, our arrangement of two conductors and a charged electret material satisfies the necessary conditions for a condenser microphone. In addition, the bias voltage (V_A) is directly determined by the magnitude of the charge stored in the electret.

At this point, a parallel can be drawn to an externally-biased condenser microphone, as shown in FIGURE 3. Here, the bias potential is supplied by the battery V_E, and consequently the charge situation is similar to the example of the electret charge layer.

Based upon this analysis, it is interesting to note that regardless of how the bias voltage is supplied, an output voltage will be generated whenever there is relative motion between the two transducer sections. Therefore, either electrode may remain fixed. This is of particular significance in the case of the electret transducer, for it indicates that the electret material could just as well be placed against the stationery electrode, or backplate, as on the diaphragm itself.

This design option was recognized by Eguchi, who had little choice but to place his wax electret layer on the backplate, due to the difficulty of fabricating a wax diaphragm. The same option still makes sense today, based upon the electrical and physical properties of the best diaphragm and electret materials available. For example, Teflon and Aclar make excellent electrets, but are not good as diaphragms.. In fact, our tests indicate that a Teflon diaphragm with an electret charge layer would have to be at least 1 mil thick to provide a stable charge. But a diaphragm of this thickness would result in a definite performance compromise. Its excessive mass would create an unacceptable roll-off of high frequency response.

Other materials, such as polypropylene and polyester terephthalate (Mylar) make much better diaphragms, and consequently permit a more nearly optimum design. However, despite the design advantages of placing the electret material on the backplate, most of today's electret microphones are not manufactured this way. The task of attaching the electret material to the stationary backplate is complicated by the fact that this element is perforated with many holes, as part of the microphone's acoustical design. Not only do the holes in the backplate serve to control frequency response, but in addition they give the microphone its directional properties. Therefore, a properly-applied electret layer must precisely conform to this hole pattern, requiring a sophisticated and well-controlled manufacturing process.

ELECTRET ADVANTAGES

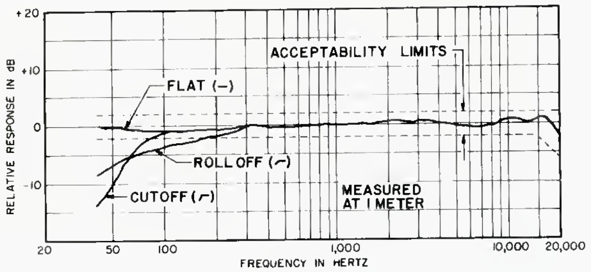
So far, we have seen that the electret form of biasing produces the same results as an external bias supply. To many users, this may be of minimal interest, for the advantages of the electret system may not yet be apparent. There are however, several important performance advantages associated with the electret approach.

The sensitivity of a condenser microphone is directly related to its bias voltage which, with today's electret technology, is inherently stable. Designs that do not incorporate an electret require some other very stable source of external bias or the added complication of a voltage regulated d.c.-to-d.c. converter to maintain constant sensitivity. Also, the electret condenser microphone can be more easily designed to work over a wide range of operating voltages and impedances, and at the same time maintain consistent performance. This is particularly important in field applications where standard simplex (phantom) powering voltages may not be available. In addition, present UL (Underwriters' Laboratory) standards specify that voltages greater than 42.4V peak or d.c. are not permitted at the terminals of connectors, such as those used with professional microphones. Consequently, those condenser microphones requiring a 48V d.c. simplex supply may not be allowed in some applications where UL approval is needed.

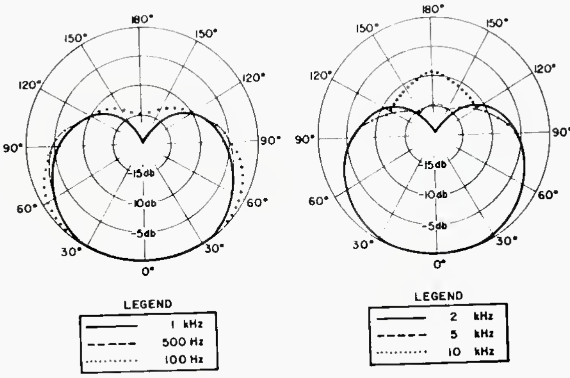
Beside powering considerations, the electret offers the advantage of design simplicity, consequently increasing the reliability of the microphone's impedance-conversion electronics. Because of the high-impedance requirements of such circuits, it is important to minimize the number of components required, since each one represents a potential noise source, which may be aggravated by adverse conditions of temperature and humidity. (For more on this subject,see *Input Noise in Microphone Preamplifiers* in this issue.—Ed.)

ENVIRONMENTAL STABILITY

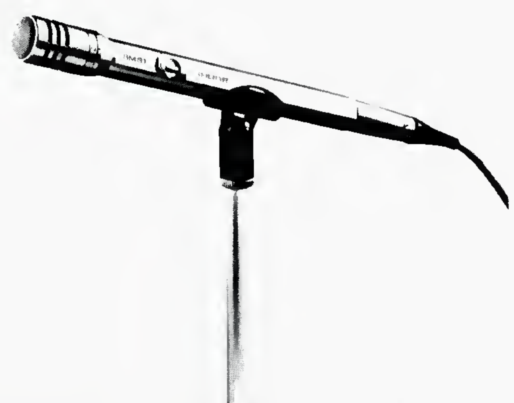
As previously indicated, the primary hesitation among designers in using electrets in professional condenser



On-axis Frequency Response of Condenser Microphone



Polar Response Characteristics of Condenser Microphone



Example of a modern condenser microphone using electret technology.

microphones has been a concern for stability in the presence of extremes in temperature or humidity. With a solution to this problem in mind, an electret biasing technique was formulated in our laboratories about seven years ago. This involved materials and construction methods designed to produce the required ultra-stable charge layer. To evaluate the design performance, a large quantity of microphones was constructed, and a variety of environmental tests were carried out. As a control, one group of microphones was stored in an environment approximating normal studio conditions, and these were evaluated periodically. During the seven year test cycle, the average decrease in sensitivity due to charge decay was 0.15 dB. If this is extrapolated over a ten year period, the average decay will be 0.23 dB.

Other microphones were given accelerated life tests, in which they were subjected to temperatures as high as 165 degrees Fahrenheit, and humidity levels of nearly 100 per cent. Under these conditions, charge decays of about 1 dB per year were observed. Of course, these conditions represent an extreme, never (hopefully!) reached in actual practice.

These data should be compared with that obtained from some of the first electret microphones, manufactured and sold in the early 1970's. Most of these are no longer commercially available. Under the same accelerated life test conditions, these microphones lost all of their charge in less than one week!

OTHER CONSIDERATIONS

Thus far, our discussion has been limited to the electret charge layer with emphasis on principles of operation and stability. Only the basic aspects of the electret charge layer have been discussed. However, there are many other factors that must be dealt with in manufacturing a stable electret layer. For example, it is necessary to insure a uniform charge distribution on the surface of the electret layer. This problem is unique to electrets, where improper charging techniques can result in a non-uniform charge distribution. If, for example, the charge density is too high in the center of the backplate, a condition of instability may result in the collapse of the diaphragm. Fortunately, this is an unlikely occurence if a uniform charge distribution exists. Consequently, it has been found necessary to develop special techniques of charging and charge stabilization.

The diaphragm, or moving electrode, also plays a significant role in the stability of a condenser microphone, regardless of the technique employed to establish a bias voltage. Stable diaphragm performance must be achieved, and constant diaphragm tension must be maintained, at temperatures as high as 165 degrees Fahrenheit (74° C) in order to guarantee reliable performance under the most demanding professional conditions. To meet these requirements, the diaphragm structure is coated with gold on both sides, and stabilized prior to assembly by a controlled sequence of exposures to high-temperature wet and dry atmospheres. FIGURE 5 is a detailed close-up view of such an assembly.

A professional condenser microphone is, of course, more than just a transducer. In order to adapt the high-impedance output of such a device to the user's microphone input preamplifier, a carefully-designed electronics section is also required. Here, the requirements are the same as those of a conventional condenser design. The microphone must not only operate with minimal noise and distortion, but must also be unaffected by radio frequency interference, stray electrostatic and electromagnetic fields, as well as the unavoidable physical abuse often suffered in the field.

IN SUMMARY

Studio condenser microphones have undergone continuous refinements over the past sixty years in nearly all aspects of their design. In each case, improvements have been directed not only toward better performance, but toward reduced complexity and increased flexibility. Today, the technology exists to eliminate the need for an externally supplied bias voltage. Modern electret technology can be applied to studio-quality condenser microphones in a no-compromise fashion, not only to reduce complexity, but to provide the user with a more versatile product. ■

*Trademarks of E. I. du Pont de Nemours & Co., Inc.

J. HOWARD SMITH

The Sound Field Microphone

Four cardioid condenser capsules arranged on the faces of a tetrahedon, through in-phase and out-of-phase matching, reflect true ambience.

THE CALREC TYPE CM 4050 Sound Field Microphone is based on an application of the mathematical theory of *Sampling,* in which a closely-spaced array of capacitor capsules and associated matching electronic circuitry completely characterize the first-order directivity of the sound reaching the microphone. (The term, "first-order" refers to the mathematical derivation of various polar pattern formulae. Most practical microphones exhibit first-order polar patterns. Ed.)

The Sound Field Microphone's output may be mono, stereo, quadriphonic or Ambisonic, and can be rotated continuously through 360 degrees horizontally, and tilted ± 45 degrees vertically.

CARDIOID OPERATING PRINCIPLES

Before explaining the operating principles of the Sound Field Microphone, let us examine the principle factors governing the operation of a standard cardioid microphone. It may be worth noting that the earliest cardioid microphones (c. 1933) contained two transducers whose outputs were combined within the microphone housing; one was omni-directional (usually a moving coil)—the other, bi-directional (typically, a ribbon element).

In fact, today a "theoretically perfect" cardioid microphone may be thought of as operating in these two modes simultaneously: first, as a pure pressure-sensitive device with an omni-directional polar response, producing a positive-going signal for an increase in pressure level, irrespective of direction (FIGURE 1). At the same time, it functions as a pressure-gradient (i.e., sensitive to the direction of pressure flow) device, with a figure-8 polar response producing a positive-going signal for pressure flow from 0 degrees (front), no signal at all for pressure flows from 90 degrees or 270 degrees (sides), and a negative-going signal for pressure flow from 180 degrees (back). (FIGURE 2)

As these signals are produced around a single diaphragm, the resultant output of the microphone is a 50-50 mix of the two which, with addition of the in-phase information (from the front) and cancellation of out-of-phase information (from the rear), gives the familiar first-order cardioid polar pattern. (FIGURE 3).

Unfortunately, in real life the two apparent systems vary both from each other and/or from theoretical norms of frequency response, sensitivity and polar pattern. Consequently, their summation does not produce the perfect cardioid pattern, or a level frequency response over the entire audio spectrum.

By skillful design and scrupulous manufacture, it is possible—particularly in capacitor capsules—to keep these variations to a minimum, and so produce the excellent performance specifications of the modern condenser microphone. But even these are some way below theoretical perfection, and must always be so, because corrections that improve the performance of one system will adversely affect the other.

THE SOUND FIELD PRINCIPLE

The Sound Field microphone monitors the three-dimensional sound field around itself, using four very high quality, well-matched cardioid condenser capsules on the faces of a regular tetrahedron (FIGURE 4). The technique allows the component parts to be isolated, individually corrected, and re-combined to produce an output (or outputs) that its designers consider to be as close to the theoretical ideal as it is likely to achieve, for some time to come.

FIGURE 5 is a graphic representation of the four-capsule array, with the nominal front looking straight out from the paper. FIGURE 6 shows the Sound Field Microphone suspended in the middle of a room. The faces of the tetrahedron have been "exploded" to clarify their relative orientations, which are:

Capsule 1—Left-front, up (L_f)
Capsule 2—Right-back, up (R_b)
Capsule 3—Right-front, down (R_f)
Capsule 4—Left-back, down (L_b)

If the cardioid outputs of capsules 1 and 2 are combined out-of-phase, the result is a figure-8 which is free of any distortion caused by the addition of the original omni-directional and figure-8 components within the capsules.

J. Howard Smith is the Managing Director of Calrac Audio, Ltd., of Hebden Bridge, England.

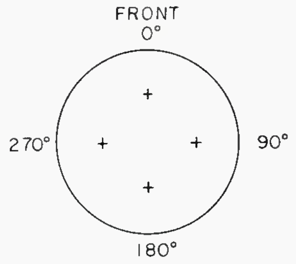


Figure 1. An ideal omni-directional polar pattern.

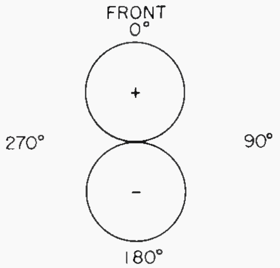


Figure 2. An ideal bi-directional polar pattern.

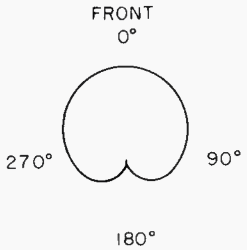


Figure 3. Cardioid polar pattern, produced by combining the polar responses seen in Figures 1 and 2.

as these are equal and, combined out-of-phase, will cancel out. The axis of the resultant pattern is 45 degrees to 225 degrees along the horizontal. The same procedure with capsules 3 and 4 produces 315 degrees to 135 degrees horizontal. Both combinations are shown in FIGURE 7.

Repeating the process with capsules 1 and 3, and 2 and 4, gives patterns at 315 degrees to 135 degrees (1 and 3) and, 45 degrees to 225 degrees (2 and 4) vertical, as shown in FIGURE 8. Note that, at this stage, the positive lobes of each of these patterns is angled either forward or upward.

FURTHER COMBINATIONS

The next stage of the process is to add the forward-oriented horizontal patterns in-phase, to produce one figure-8 pattern on a horizontal (front-to-back) axis of 0 degrees to 180 degrees. The same patterns are combined out-of-phase, to produce another figure-8 at 90 degrees to 270 degrees (i.e., left-to-right). A similar single in-phase treatment of the two vertical components produces one vertical figure-8 pattern.

Throughout all combinational processing stages, great care must be taken to maintain the relative gains accurately, in order to guarantee optimum pattern shapes.

There now exists three pressure-gradient (bi-directional) signals, representing the three basic components of direction; front-back, left-right, and up-down. Finally, it is only necessary to add the outputs of all four capsules in-phase, producing a single omni-directional pressure signal.

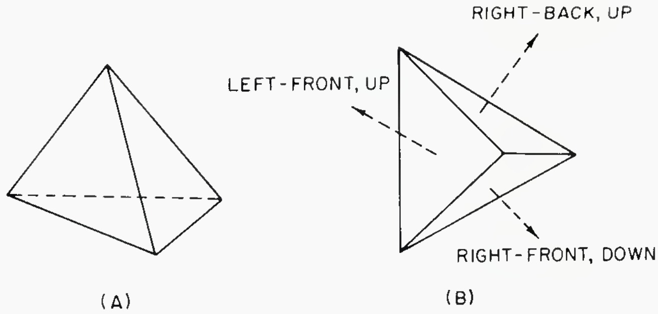


Figure 4 (A) A regular tetrahedron (in effect, a three-sided pyramid.) (B) The same tetrahedron, oriented so that its sides face in the directions shown by the dashed lines. (The fourth side faces right-back, down).

We now have the four signals required to construct any first-order characteristic—from figure-8 through cardioid, to omni-directional—looking in any direction through 360 degrees horizontally and vertically. At this stage, phase correction is applied in order that any two or more microphone patterns so constructed are truly coincident. That is, they apparently occupy the same point in space.

THE "B" FORMAT

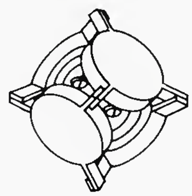
These four signals, known as "B" Format, and designated W (omni-directional), X (front-back), Y (left-right), and Z (up-down), may be directly recorded on tape for processing later on. This allows experimentation with capsule angle, pattern, pan, and tilt, without in any way affecting the actual recorded "B" Format signals. Naturally, in a live broadcast situation this facility would not be practicable, but would create unprecedented freedom for experimentation during pre-broadcast rehearsals.

THE SOUND FIELD CONTROL SYSTEM

Since the European debut of the Sound Field Microphone at the 59th convention of the Audio Engineering Society (Hamburg, Germany, February 28 to March 3, 1978), additional development work has taken place, and the complete system now comprises two units—the actual microphone, and the Sound Field Control Unit. The Control Unit provides the following facilities:

1. Input gain switching, over a 50 dB range.
2. Capsule mute switching (test purposes only).
3. Microphone normal/inverted switching.
4. Master fader (rotary).
5. Metering (PPM or VU) switchable to W, X, Y or Z.
6. Rotate—360 degrees of continuous horizontal rotation.
7. Tilt—± 45 degrees of continuous vertical rotation.
8. Vertical Dominance. If the sound field is imagined as a sphere, the Vertical Dominance function will

Figure 5. The four capsules of the Sound Field Microphone.



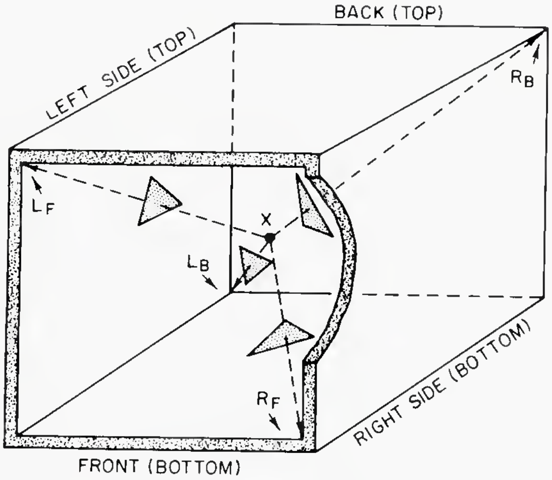


Figure 6. "X" marks the spot where the Sound Field Microphone is suspended in the middle of a room. The faces of its tetrahedronal shape have been "exploded" to clarify the relative orientation of the four capsules on the single tetrahedron.

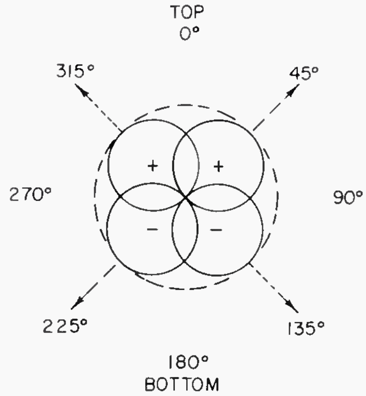
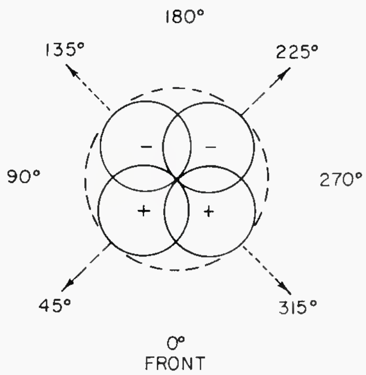


Figure 8. Dashed line: combination of capsules 1 & 3. Dotted line: combination of capsules 2 & 4.

extract a cone of information, the apex of which is at the center of the sphere, and whose angle is determined by the rotary potentiometer (45 degrees maximum, up or down). The feature is useful for getting rid of the unwanted sounds of ventilating systems or audience reactions, while maintaining an otherwise-spherical sound field.

9. Output Mode Selector—mono, stereo, quadriphonic or Ambisonic.
10. Angle. In the stereo mode, this function controls the capsule angle of a virtual pair, ranging from 0 degrees (mono) to 180 degrees. Similarly, in the quadriphonic mode, the virtual four capsules may be scissored between 0 degrees (both pairs on a 0 degree to 180 degree axis), and 180 degrees (both pairs on a 90 degree to 270 degree axis). In the Ambisonic mode, the control relates to loudspeaker position; it should be adjusted to the angle subtended by the left-front and right-front speakers at the central listening position.
11. Polar Patterns—Variable through all first-order characteristics, from figure-8 through cardioid, to omni-directional.
12. Gain—Acts as master monitor gain during "B" Format recording.

Figure 7. Dashed line: combination of capsules 1 & 2. Dotted line: combination of capsules 3 & 4.



The Calrec Sound Microphone, and the NRDC (National Research and Development Corporation) Ambisonic System are the subject of United Kingdom Patent No. 1494751 and United States of America Patent Nos. 3997725 and 4042779, together with all corresponding patents in other countries, and all other patents pending. ■

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This bibliograph makes no attempt to be complete and the author apologizes to those kindred spirits not mentioned.

Plotting Polar Patterns

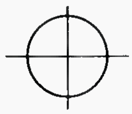
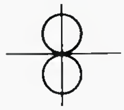
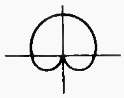
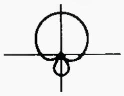
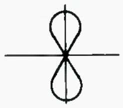
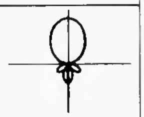
● In this issue of **db**, author J. Howard Smith refers to "first-order" directivity characteristics. For our mathematically-inclined readers, we asked Shure Brothers' R. Schulein to supply the formulae for an assortment of theoretical polar patterns, from first- to sixth-order. These are given below, along with our quick sketches of the patterns.

Also given are Distance Factors—an indication of the permissable increase in working distance of various microphones, as compared to an omni-directional microphone with the same zero-degree sensitivity. At these relative distances, the signal-to-noise ratios for the two microphones are the same.

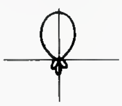
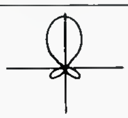
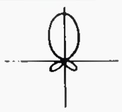
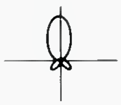
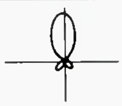
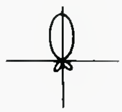
Remember, this is all theory—in practice, you may come up with somewhat different results. And, to get more separation on the acoustic guitar, don't rush off in search of a sixth-order cardioid microphone. In the real world, most—if not all—microphones are first-order. Sorry about that!

THE EDITOR(S)

MICROPHONE POLAR CHARACTERISTICS

CHARACTERISTIC	BI-DIRECTIONAL	OMNI-DIRECTIONAL	CARDIOID	HYPER CARDIOID	SUPER CARDIOID	2nd ORDER BI-DIRECTIONAL	2nd ORDER CARDIOID
POLAR RESPONSE PATTERN							
POLAR EQUATION	1	$\cos\theta$	$\frac{1}{2}(1+\cos\theta)$	$\frac{1}{4}(1+3\cos\theta)$	$.37+.63\cos\theta$	$\cos^2\theta$	$(1+\cos\theta)\cos\theta$
RELATIVE OUTPUT AT 90° (dB)	0	$-\infty$	-6	-12	-8.7	$-\infty$	$-\infty$
RELATIVE OUTPUT AT 180° (dB)	0	0	$-\infty$	-6	-11.8	0	$-\infty$
ANGLE AT WHICH OUTPUT IS ZERO	-	90°	180°	110°	125°	90°	90° 180°
DISTANCE FACTOR (DSF)	1	1.73	1.73	2	1.93	2.24	2.74

MICROPHONE POLAR CHARACTERISTICS

CHARACTERISTIC	2nd ORDER HYPER-CARDIOID	2nd ORDER SUPER CARDIOID	3rd ORDER CARDIOID	4th ORDER CARDIOID	5th ORDER CARDIOID	6th ORDER CARDIOID
POLAR RESPONSE PATTERN						
POLAR EQUATION	$(.6+\cos\theta)\cos\theta$	$(.775+\cos\theta)\cos\theta$	$(1+\cos\theta)\cos^2\theta$	$(1+\cos\theta)\cos^3\theta$	$(1+\cos\theta)\cos^4\theta$	$(1+\cos\theta)\cos^5\theta$
RELATIVE OUTPUT AT 90° (dB)	$-\infty$	$-\infty$	$-\infty$	$-\infty$	$-\infty$	$-\infty$
RELATIVE OUTPUT AT 180° (dB)	-12	-18	$-\infty$	$-\infty$	$-\infty$	$-\infty$
ANGLE AT WHICH OUTPUT IS ZERO	90° 127°	90° 141°	90° 180°	90° 180°	90° 180°	90° 180°
DISTANCE FACTOR (DSF)	2.83	2.81	3.4	4	4.3	4.9

JOSEF W. DORNER

Input Noise in Microphone Preamplifiers

Thermal noise is a factor which must be accounted for in determining the efficiency of preamplifiers.

ANYONE FAMILIAR with audio electronics knows that a microphone preamplifier is required in order to amplify the low level signals which arrive from the microphone. Of course, this must be done with a minimum of distortion, and a maximum signal-to-noise ratio.

What else needs to be said? Well, perhaps a bit more, since now and then even technical reviewers have been seen to come to conclusions which depart considerably from the actual facts.

THERMAL NOISE

As every technician knows, the laws of physics set limits in all areas of science, and it is impossible to go beyond these limits, one of which is thermal noise. At temperatures above absolute zero (−273° C), thermal noise is generated by the continuous random motion of electrons within an electrical conductor. The voltage so generated has a uniform spectrum level—meaning that it contains all frequencies at equal levels over a specified bandwidth (white noise). It increases in direct proportion to the temperature of the conductor and to its electrical resistance.

Josef W. Dorner is an audio engineer in Zurich, Switzerland.

THE NYQUIST EQUATION

Some fifty years ago, H. Nyquist developed a formula for calculating thermal noise. The following simplified version of the equation lets us determine the thermal noise voltage in a resistor at room temperature (20° C = 68°F).

$$E_n = 1.27 \bullet 10^{-10} \sqrt{RB}\ *$$

E_n = Noise voltage, in volts, rms
R = Resistance, in ohms
B = Bandwidth, in hertz

If we calculate the noise voltage for a 600 ohm resistor and a bandwidth of 20 kHz, we obtain:

$$E_n = 1.27 \bullet 10^{-10} \sqrt{600 \bullet 2 \bullet 10^4} = 0.44 \text{ microvolts}$$

This represents the noise voltage presented to the input of an amplifier when it works from a 600 ohm source, since any 600 ohm transducer will produce this noise voltage due to thermal agitation of the electrons in its signal-generating conductor. Relative to the 0 dB reference of 0.775 volts, the absolute open circuit noise voltage of a 600 ohm source can therefore be expressed as having a level of

$$-20 \log \frac{0.775}{0.44 \bullet 10^{-6}} = -124.92, \text{ or } -125 \text{ dB}$$

VOLTAGE AMPLIFICATION

In order to evaluate the quality of a preamplifier with regard to its noise performance, one measures its noise output after having established its voltage amplification. Voltage amplification (as described in ASA C 16.29 1957/ 1.2.3) is the ratio of the voltage across a specified load connected across the transducer's output, to the voltage across the input to that transducer. In order to obtain the amplification factor in decibels, we apply the following equation:

$$20 \log \frac{E_{OUT}}{E_{IN}}$$

For the purpose of demonstrating this measurement on a piece of well-known recording equipment—designed to work with microphones of the rather unorthodox impedance of 600 ohms, an audio millivoltmeter was connected to the output of the microphone preamplifier, and with its gain control fully open, a low-level signal was fed into its input, from an audio signal generator. By adjusting the input level, the output was set to read +6 dB, with reference to 0.775 volts, or, expressed in other terms—1.55 volts.

The input voltage for this condition was found to be 0.16 millivolts. By entering these figures into the above-mentioned equation, a ratio of almost 10,000 (actually, 9687.5:1) is obtained between the two signals, and this stands for an amplification factor of about 80 dB (79.72 dB). Of course, one might set the amplification to some other easily-measured value by adjusting the amplifier's gain.

With the voltage amplification thus established, the signal source was disconnected, and the input terminated with a 600 ohm resistor, so as to simulate a 600 source. The noise voltage now appearing at the amplifier's output is a combination of the resistor's thermal noise, plus the noise which is generated in the amplifier's first stage.

In our test case, a noise voltage of 5.46 millivolts, or −43.04 dB was measured. In order to be certain that a true rms reading is obtained, one must know that for the widely used average-responding rms calibrated meter, a correction factor of 1.127 (+1.04 dB) must be applied to the meter reading. By applying this rms correction, the level rounds off to −42 dB. Now, by subtracting the 80 dB amplification factor (−42 — 80), the figure of −122 dB, or 0.615 microvolts is obtained—a value which stands for the amplifier's equivalent input noise (EIN). It differs by only 3 dB from the theoretical minimum value—truly an excellent figure, which expresses at the same time the Noise Figure of that amplifier.

It goes without saying that the above-developed calculations apply equally well to any other combination of amplifier gain and source impedance. ■

*ADDENDA

The complete Nyquist equation is:

$E_n = \sqrt{4kTRB}$
k = Boltzmann's constant ($1.38\ 10^{-23}$ joules per °C)
T = Absolute temperature, in °K (= °C + 273)
R = Resistance, in ohms
B = Bandwidth, in hertz

JOHN M. WORAM

The 60th Audio Engineering Society Convention

The word's out—digital!

FROM ANALOG TO DIGITAL

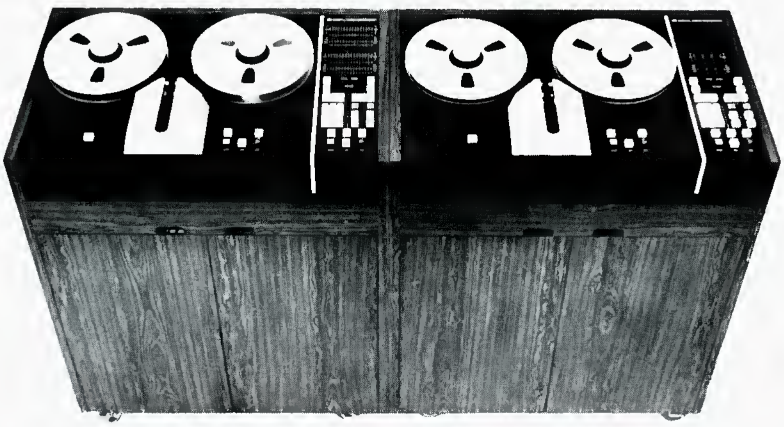
Mitsubishi, 3M, Technics, Soundstream, JVC. What do these five companies have in common? Why, digital tape recorders, of course! And all were at the recent 60th convention of the Audio Engineering Society (2-5 May), demonstrating their various versions of the tape recorder of the future.

How far off is that future? Well, that depends. Digital recordings are already being made in this country, with Soundstream probably the first to have its machines used for regular commercial releases.

With Soundstream first, 3M certainly qualifies as biggest. Their Digital Audio Mastering System records 32 audio channels on Scotch 265 one-inch digital tape. The system includes an auxiliary 2-or 4-channel digital tape recorder for mixdown.

How much will all this cost? Probably "under $150,000" ($149,999.95?) according to 3M. But for now, the system

The 3M Digital Audio Mastering System.



is not for sale. Instead, 3M promises to have three systems available later this year on a lease-rental basis only. Much to their credit, the company acknowledges the still-evolving nature of digital technology. With standards still in a state of flux (or whatever the digital equivalent of flux is), it is a certainty that there will be significant updates required on any digital recorders delivered within the next few years. Therefore, the lease-rental route seems to make the most sense, at least for the present.

Soundstream may be thinking along the same lines, since it is now offering its four-channel digital tape recorder on a contractual basis. Soundstream supplies equipment and personnel for on-location recording, plus editing services at its Salt Lake City headquarters.

The company has recently perfected its half-speed playback capability, and has taken advantage of the JVC Cutting Center's ultra-modern half-speed disc mastering facility to produce some "super discs" which may become the ultimate high-fi demo. At the recent convention, at least one such system was partially demolished when it couldn't handle the Soundstream/JVC test pressing. And that conjures up a mind-boggling prospect: After all these years of complaints about lousy pressings, wouldn't it be a laugh if (in the distant-but-rosy future) the lp record finally became one of the *strongest* links in the audio chain?

An interesting feature of Mitsubishi Audio Systems' two-channel PCM recorder is that two analog tracks are also recorded on the same piece of quarter-inch tape, along with the digital program. Why? Well, at this stage in the emerging state-of-the-art, some features of conventional analog tape recorders are not yet possible on digital machines. For example, one cannot "rock" the tape back and forth slowly, to find a particular spot. Even if this were possible, digital tapes usually require electronic editing anyway, so there would be little point to doing so. However, it would be nice to high-speed fast-forward (or rewind) to find (by ear) the middle of take 3, or whatever. Not possible with digital (yet). So, the analog tracks are used for these purposes..

And, it happens that the Mitsubishi system does indeed permit traditional "cut-and-splice" editing. That means the analog tracks can be used to find convenient edit spots, as just noted. It also means you can do the editing on any convenient analog tape recorder, keeping the digital machine free for recording purposes. But just remember to make those edits 90 degree butt joints. The PCM system doesn't like the traditional 45 degree cuts.

Perhaps it would be proper to view the analog + digital convenience of the Mitsubishi system as an "interim" system, very handy until the entire audio world is digital (when and if). By then, electronic editing (and other digital goodies) will be commonplace, and there will be no need for analog assistance. But in the meantime, those two analog tracks can certainly be helpful.

With minimal fanfare, Technics quietly demonstrated their version of a PCM two-channel tape recorder. At first glance, the machine appears to be merely another variation of their RS-1500 isolated loop analog tape recorder. In fact, it uses the same transport system as the RS-1500, but there the resemblance ceases. As of this writing, Technics has not announced its formal entry into the digital arena, and this machine is probably best viewed as an advanced laboratory prototype.

Likewise, JVC presented a prototype of its digital system—in this case, an add-on unit for use in conjunction with the company's VHS format videocassette recorder.

What about Ampex? So far, they haven't shown anything digital. But it's a pretty safe bet they've got something up their sleeve. When their sensational ATR-100 was introduced a few years ago, it took the audio press, and others, by complete surprise; previously, there hadn't been so much as a whisper. Now, when you query them about things digital, you get a broad smile and something non-committal like, "Hmm," or "We'll have to look into that." So, all you digital watchers, be prepared.

Mitsubishi Audio Systems' PCM Recorder.

Cybersonics' Disc Master 2002.

OTHER DIGITAL HARDWARE

By now, digital delay lines have been around for ages, and are well on the way to being taken for granted. When they were first introduced, they were one input/two (or more) output devices, with delay times of up to 200 milliseconds as about standard. They were immediately put to use to supply "tape delay" and for vocal doubling.

At the L.A. show, Lexicon—a pioneer in digital delay—demonstrated a prototype of the latest generation in digital delay technology. Their Model 224 Reverberation Synthesizer does just what the name suggests. Since real reverberation is, after all, nothing more than a series (an incredibly complex series of course) of delays, it follows that reverberation might be digitally synthesized.

As usual, 'tis easier said than done, since each room has its own distinctive pattern of seemingly-random echoes, decay times, and what-have-you. A little feedback around the delay line just doesn't fool anyone. In fact, you might need a computer program to develop a convincing reverberation characteristic.

And that's where the 224 comes in. It's set up to accept seven programs, three of which simulate concert halls of various sizes and shapes. Apparently, additional programs will become available, tailored to various applications. There is also a depth control which ". . . is similar to adding a time delay before the reverberation, but is much more complex and natural in its operation." Still other controls allow separate adjustment of hi-, mid- and low-frequency reverberation time. Lexicon expects to make the first production deliveries of the Model 224 during December of this year. The bottom line is $5,000.

DeltaLabs' DL-1 Digital Delay Module is a one input/two output device with delays variable from 5 to 160 μsec. Front panel controls are used to vary two of the outputs, or these may be disabled (for tamper-proof operation) by duplicate internal controls. Presumably, the third output is only adjustable from inside.

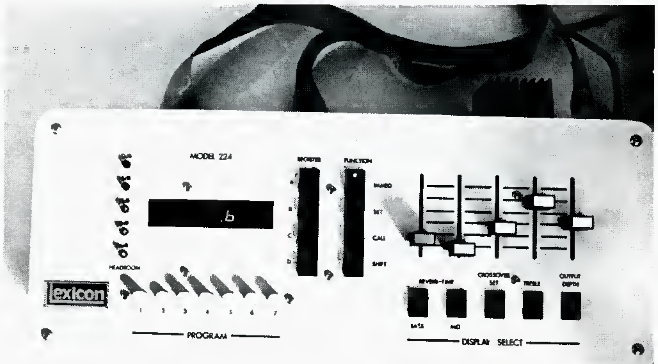
Along with the usual applications of digital delay, the DL-1 spec. sheet suggests "Haas Effect Stereo Panning," for better localization. For example, a signal to originate at, say, the extreme left, is also fed to the right, via a delay of about 20 μsec. Since the signal arrives at the left speaker first (no delay), the listener localizes the sound as extreme left. The delayed right channel "helps" reinforce this impression. Try it yourself, and experiment with different delay settings.

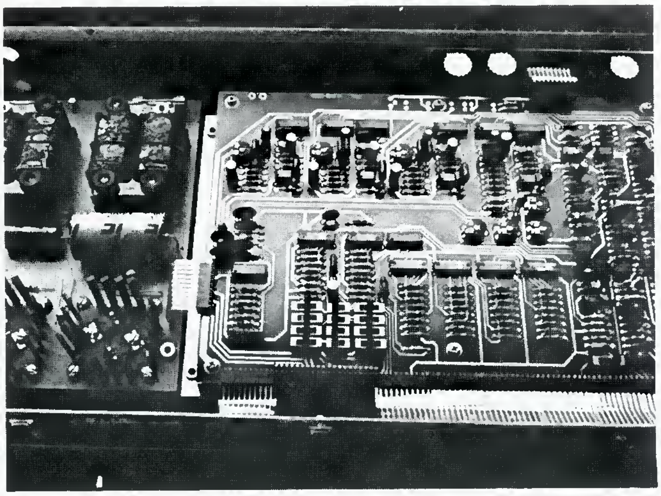
UREI's Model 927 Digital Delay Line has four outputs, each adjustable in 1 μsec. increments up to a maximum of 127 μsec. Each delay time is set by thumb-wheel switches, with a separate toggle switch for delay in/out. In addition, there are separate level controls for each of the four outputs. To keep busy fingers away, there's a screw-on see-through face plate security cover.

A SPACE STATION AT URSA MAJOR

No, we haven't been done-in by the typesetter again. Ursa Major is a new outfit, and the Space Station is their entry into digital audio. According to the catalog sheet, it was designed by an engineer with a broad knowledge of analog and digital technologies but ". . . obviously, not a hell of a lot of modesty." Another one of his shortcomings seems to be a penchant for telling the truth.

The control panel on Lexicon's Model 224 Reverberation Synthesizer.





The innards of DeltaLabs' DL-1 Digital Delay.

For along with a thorough description of all the strange and not-so-strange effects that are possible, is the warning that Ursa Major has ". . . not produced a perfect product. With some sources, especially solos of flute, voice or notes approaching pure tone, its limitations can be heard in a form of noise that varies with signal intensity, pitch and the degree of feedback."

However, from a brief listen at the convention, it seems as though Ursa Major really has its act together. Now, if only they could knock off the honesty.

ANALOG EFFECTS

Although anything with the word "digital" in it draws attention these days, analog is certainly not dead by any means.

Neutrik Products from Philips Audio Video Systems announced its Model AD4 analog Delay Unit with four discrete outputs. A single potentiometer controls all four delays, over a 4:1 range. Thus, at a minimum setting, the delays are: 12.5, 25, 37.5 and 50 μsec. At maximum, these lengthen to 50, 100, 150 and 200 μsec. The AD4 will sell for $795.

If you've ever felt the need for Limited Spin, Hall Effect, Hollowing, Cardboard Tube, Motorbiking or Dalek Voicing, you'll probably want to check out Audio and Design Ltd.'s new S24 Time Shape Module. This is another in the company's "Scamp" series, which includes an assortment of equalizers, compressors and such.

On the off-chance that some of our readers can't tell a cardboard tube from a Dalek voice, we'll just say here that the S24 is a Special Effects Generator gone biserk. (Dalek Voice?) With separate controls for flanging, delay, envelope follower, spin, etc., plus a built-in limiter, you can probably create any effect that occurs to you, and a lot of others that don't.

A NEW LATHE

It looks as though our convention coverage will spill over into the next month's issue, since there's a lot more to talk about, but not too much space left. However, we can't leave you without mentioning the debut of—of all things—a new cutting lathe!

It seems as though every convention brings with it a rash of new consoles, but when was the last time you saw a new lathe? Cybersonics, Inc. drew a lot of attention with its Disc Master 2002, especially among direct-to-disc enthusiasts, who were attracted by its compact size.

One of the logistical problems of direct-to-disc work is the transportation of a fairly massive lathe (or lathes) to the recording site. The DiscMaster 2002 is only 35½ in. wide, 27½ in. deep and 16 in. high, and weighs about as much as a well-fed recording engineer (some 250 lbs.).

According to the Cybersonics brochure, the lathe features a "compact utilitarian design" in which "simplicity is the key." A quick look at the front panel confirms this: there are very few knobs and switches visible. And this invites the question: is there enough control here to meet the sophisticated demands of today's tape-to-disc technology? Obviously, the answer will come in time. But while we're waiting, we wish Cybersonics good luck with their enterprising new Disc Master, which should sell for about $46,000. ■

MARSHALL KING

The Technician and His Union—Part II

Avoiding the "one armed paperhanger" syndrome.

Of the three main unions serving our industry—IATSE (International Alliance of Theatrical Stage Employees), NABET, and IBEW—the first of these is undoubtedly the largest, having established an early foothold when commercial film-making was in its puberty. In addition to sound, it covered most of the allied arts, such as camera, lighting, makeup, wardrobe, editing, projection, and stagecraft.

The IBEW, on the other hand, which is just as old (or older), concerned itself at the outset with electrical wiring for industry, particularly those dealing with railroads. Thus its name: the International Brotherhood of Electrical Workers. It was only later that it began to encompass work in radio, motion pictures, television, and the recording fields. Today it is notable in our industry for being the bargaining agent for the technicians of CBS Broadcasting.

The baby of the trio is NABET, a bargaining unit organized specifically for the work implied in its title: the National Association of Broadcast Employees and Technicians. A large part of NABET's work today is representing the workers of the other two major broadcast networks, NBC and ABC. However, in the past decade, NABET has made several successful forays into the making of independent motion pictures, one of which was *Easy Rider*. In addition to this, NABET has recently been able to take advantage of internal strife in other unions to get a foothold in several motion picture studios.

WHO REPRESENTS US?

Who represents us? has been a primary concern of workers since the very beginning. The record shows that sound technicians over the years have been approached for organization by the most diverse labor groups, from the Teamsters to the ACEW and nearly everything in between. How we choose to go with one group rather than another is a riddle often based on gut feeling more than reason. Meanwhile, the alert business agent of any local union is forever on the lookout for ways to enlarge his ranks.

Thus it was that Jack Coffey, of IATSE's Sound Local 695 in Hollywood, which has traditionally represented *film* technicians, began a one-man effort to organize *videotape* technicians whenever and wherever he could.

Outspoken, controversial, determined, Coffey's convictions have won him both friends and enemies, neither of whom does he consider more important than the ideas for which he fights. Such fighting has led to his dismissal or suspension as business agent on more than one occasion, with the invariable result that his tenacity thus far has boomeranged him back into reinstatement.

Whether Coffey is "in" or "out" at the time this goes to press, the fact remains that the career of every union business agent is forever plagued by a precarious position. On

Marshall King is a sound mixer at CBS Television.

Freelance work often entails being forever on the go and a willingness to work under fast moving and difficult conditions.

one hand he is faced with the possibility of a disgruntled membership, while on the other he has the parent body, the International Office, constantly reminding him not to move too fast.

To the technician in smaller operations where it is expected that he do everything—yesterday camera, today audio, tomorrow editing—some of the foregoing issues may seem irrelevant or amusing. Yet, one of the foremost goals of the union is the protection of craftsmanship and the improvement of working conditions. Once gained, benefits can be easily lost through lack of vigilance when employers force expediency in honor of the budget. Keeping an eye on this matter is often a full-time job, and every business agent in the country looks to his membership for assistance.

MINIMUM CREWS

Pressure from producers' guilds is very strong indeed, and outside of membership apathy, it's a prime cause for union gains being subsequently lost. For example, until recently the minimum sound crew in film consisted of four men: the mixer, the recordist, the boom operator, and the cableman. A half dozen years ago the recordist was dropped, his job going to the mixer. But this was not the end of it. After being reinstated into office for the second time, Jack Coffey returned to find that the membership of his Local, under the guidance of a trustee chosen by the International Office, had ratified a contract with the producers that made provisions for a minimum *two-man* sound crew.

Is the union's efforts to enforce a guaranteed number of technicians on a given production just a form of featherbedding? Or does it have something to do with craftsmanship and better working conditions? In all fairness to the producer it must be admitted that he would be derelict if he didn't watch his budget and hire only the number of people necessary to do the job. Therein lies the dispute: how many people *are* needed to do the job?

Labor's ears are still ringing from a blast appearing in the *Los Angeles Times* by a well-known producer whose outrage was based on his being told, upon arriving at the outskirts of his desert shooting location, that he could not drive his own car to the set, but would have to be taken there in a vehicle operated by a union member. Is this comparable to the film sound man striving for a minimum four-man crew?

Technicians who have been there insist that it is not the same. Most are ready to show that, far from being featherbedding, a well-rounded crew not only reduces the pressure on each member but affords some hope for the best professional product. "Oscars and Emmies may not be the uppermost goal of a technician," one mixer has said. "but his good reputation *is*, even without the awards. And that kind of craftsmanship is always tied to good working conditions."

"It's true," he continued, "that we have all jumped in and performed *extra work* when the production would otherwise have been held up at great expense. But sometimes that leads to dire results, for somone on the production staff will later remember that Joe Blow did two jobs at once during the emergency, so why not schedule it that way all the time? Bit by bit, it leads to the one-armed paperhanger syndrome."

A distinction should be made between the technician's problems which are a matter for the union and those which are purely the earmarks of his trade. The union can ensure that meal periods are properly given and that out-of-town expenses are paid for, but it cannot be called upon to relieve the technician of the daily dilemmas that result from a battle of wills, from personality conflicts, or from differences in creative authority.

THEY CRIED A LOT

An example that comes to mind are certain problems that existed on a well-known soap opera, a program where the leading actress and the audio mixer cried all the time. The actress did it because she was paid to, and the mixer did it because the leading man thought it was sexy to whisper his lines as the boom mic was trapped twenty three feet above the floor in order to stay clear of the chandelier which the art director spent the weekend designing in his elfin workshop.

One of the show's early directors became hysterical whenever the boom operator deliberately ran his boom arm through the key lights during rehearsal in order to find where the shadows were so he could avoid them during taping. The director was affectionately referred to as "Tinkerbell," for he would fly out of the control booth in a rage without once touching the floor, which surprised no one at all, for that was his normal way of getting around—1,000 miles to the gallon on unleaded fairy dust. He was so excitable that he had to take uppers to calm down. As stated, these are not matters of concern to the union; they are incidents that must be dealt with by the personalities involved.

CHANGING TRENDS

Trends are forever changing in our industry, just as in others, and a look at certain recent practices in film sound can show us how conditions can take an unexpected change, in any branch of our craft, with regard to working conditions and product quality.

One noticeable trend, now that the major studios and their huge centralized home lots have largely given way to smaller, independent companies which form and unform faster than cumulus clouds on a spring day, is for the production sound mixer to supply his own equipment. True, he may pass the expense of this equipment (rented or owned) on to the producer who will gladly pay it, for it relieves him of a major headache. But how does this affect the quality of the sound?

It *can* mean that the mixer is obliged to put together an array of rented equipment with which he is not totally familiar. It *can* mean that since he is now his own maintenance man, he might be caught with failing equipment and no time to fix it. It *can* mean that, since he cannot own or rent every possible kind of audio device, he may find himself using whatever he has at hand, when the situation calls for gear he doesn't have or couldn't find available. It *can* mean that he is obliged to own a van and a dolly, and that he must now deal more with Teamsters and other Locals in order to do his work. It *can* mean that he is living under a new kind of pressure, far removed from the day when he merely deposited his mind, body and experience on the location site—a new situation whereby working conditions, wages, jurisdiction and craftsmanship all take on a new meaning. And it is almost certain to mean that the business agent of his Local has new problems to deal with.

Jack Coffey mentioned yet another trend in film audio: the increasing use of rf mics, due to pressure from the cammermen. "The way it works is this," he said. "Cameramen have gotten wise to the rf mics without knowing a hell of a lot about it. For them it's a new-found freedom. They don't have to give way to the boom mic and its shadows, so they press for its use. In older days the soundman would tell the cameraman, and everyone else, to stay the hell out of his department. But now if the cameraman can convince the director that production will go faster if the rf mic is used, then that's it. You can see what effect this can have on the quality of sound."

STAFF VS. FREELANCE

What are some of the distinctions between a staff job and the freelance technician? While there are many similarities, there are many more differences, particularly in the area of *hustle*. It seems to be a fact that a full-time, year-round staff job imparts a feeling of security and long-term thinking which does not always go with freelance work.

A good freelance mixer, or camerman, or boom operator seldom works for union scale. He can command more, sometimes twice the going rate, by virtue of his being able to travel on a moment's notice, by his knowing all of the equipment of most manufacturers, by his readiness to take full responsibility for his part of the final product, by his ability to think in terms of how the producer can get the job done better or more quickly with no slowdowns or breakdowns.

A staff technician, on the other hand, often finds himself in the role of middleman, whereby many decisions (and motivations) are taken away from him by his employer who takes on the foregoing responsibilities with the producer. The result can lead to a tendency where the staff technician, weary of getting his hands slapped for going out on a limb, backs off from his own ideas and turns the proper knob at the proper time.

If there is a creative freedom and a better wage scale in freelance work, what are the shortcomings? One is the matter of reciprocity. Is the New York mixer free to work in the jurisdiction of Las Vegas or Chicago? Or is he told that he can't touch any of the equipment, that he must be only an "advisor" to a local mixer of their own choosing? It's a hassle that's been raked over more than once. Often it is decided that the freelance technician will be allowed to work in these "visited" areas provided that a technician from the local union is paid to stand by. These are possibilities that should be discussed before the show ever hits the road.

Another requirement for the freelancer, which is usually not faced by the staff technician, is that he must keep himself in demand and always on the scene. Whether or not this leads him into a political posture is hard to say, but if it doesn't keep him alert and affable it does mean

A Hollywood sound crew from IATSE Local 695 sets up equipment for televising a luau *at the opening of Disneyworld in Florida.*

that he has some kind of genius that producers will pay anything to get. Many freelancers, lacking true genius, have found it better to be alert and affable, and to have an efficient phone answering service.

Being forever on the road, living out of a suitcase, is also a way of life the freelance technician has to accept. For many, such traveling is the very essence of life, but for others it's a home-wrecker. Most of all, the technician, being away from the protective eye of his local union, shouldn't be obliged to do two jobs when he has contracted for one. Sometimes he is coerced into doing so, particularly where the true scope of his work wasn't clearly spelled out during the production meeting back home. And even when it is, it's sometimes harder to convince a producer that the crew will be short-handed than to convince an Indian that Columbus discovered America.

Doug Adam, assistant business agent of IATSE Local 695, cites one recent occasion where a crew was sent to videotape a show in a New Orleans nightclub. "When they got there and found they were short-handed," Adam said, "the mixer informed the producer that he needed a third man in the audio crew. No cableman or audiotape man had been provided, and the boom operator was obliged to walk backward carrying a fishpole during a 50-foot camera move while the mixer held him by the belt to guide him backward over the cables and around the furniture, while at the same time operating a Nagra recorder slung over his shoulder.

"On top of this," Adam continued, "the producer required the playback of an audio tape and suggested that perhaps one of the stagehands could do it. With that, all work came to a stop and the business agent was flown out from Hollywood to set the matter straight. Here's another case where the guys had to be especially vigilant when working away from home."

So it may be that the freelance technician works under a kind of pressure that is subtle in its implications. He is, by the very nature of his being freelance, in a position where getting repeated calls for work from producers is his life's blood. Yet, unlike the relatively secure staff technician working in a station where the producer does not pay his salary, he is obliged to be his own watchdog against careless acts by a producer who will, hopefully, hire him again and again. It's a paradox to be sure, but if he fails to demand strict adherence to fair working conditions he is reducing his professional well-being at a logarithmic rate.

JOB CLASSIFICATION

A final aspect pertaining to wages, working conditions and craftsmanship is the matter of *job classification.* In smaller local companies it is common for the technician, especially the beginner, to wear many hats. Today he is working camera, tomorrow audio, yesterday video. Per-

A wireless microphone being prepared to be used in a field remote where the performer will move across open ground.

haps this is the way it should be when he has yet to find out where his best inclinations lie. Certainly from the standpoint of management this is most desirable, for when anyone can do any job, flexibility in the employee pool is very high.

Yet, labels do form as a result of our performances and we migrate into fields of prime activity. Beyond this, even excellence may prevail in some special area as a result of talent or extended effort, or both. Now we are classified. Should we be paid accordingly?

On the surface the answer would appear to be an obvious "Yes!," coming especially from the achievers who work alongside those content to just get by. But how does this work out in actual practice; how is it regarded by our unions and employers?

Generally speaking, there is a high degree of job classification (and the varying pay scales that go with it) in film work governed by the IATSE, but in network television done on videotape there is a very small amount of classification in NABET contracts and virtually none in the IBEW. While the latter condition has not been a cause for overt internal strife, there have long been differing opinions regarding its propriety. To oversimplify, there is no monetary reward for those who extend themselves and no particular penalty for those who don't care to; there is one pay scale for all. Advocates of job classification point out that in the absence of monetary rewards the motivation factor can be of low magnitude.

On the other hand, IBEW's Andy Draghi has serious reservations regarding the idea of classification and varying pay scales.

"Look at it this way," he says. "Supposing after many years of pressure in the control booth, which every mixer has known, there comes a time when you either *must* or *want to* slow down. So you get out of the booth and get yourself assigned to a less demanding job. Now if you have classification, this means you're going to take a cut in pay. I don't think that's fair. It seems to me that you should continue to be rewarded for all your years of work by getting the same pay when you have to pull back and slow down a bit."

But advocates of classification see this as an unsatisfactory reward, for, after finally withdrawing to a less strenuous position, he would find himself alongside technicians who for the same twenty years have been getting the same pay without once venturing onto the firing line.

"It's very simple," one cameraman told me. "Let me have the heavy pay while I'm doing the heavy work. If I can't cut it after all these years then I *should* get less money."

From another angle, Draghi sees classification as a possible cause of divisiveness among technicians. But those who argue in favor of it point out that since management has created its own star system by "selling" certain diligent technicians to certain irascible producers, those technicians should be paid above the norm.

Draghi's answer to this is that the existing parity in the network technicians' pay can be blamed on the company rather than the union. "We negotiate minimum salaries, not maximums," he said. "The company can and should pay above scale for the star system they support while decrying it out of the other side of their mouths."

Perhaps it is enough to say that classification or the lack of it brings its own rewards: a chance for higher income in the former and the likelihood of greater security in the latter. Which *modus operandi* pertains to us depends on where we work and on the structure of our union contracts.

THE UNION SCORECARD

Is the union doing its job? The answer depends on who gives it. Every union leader I have spoken to has the same lament: it's not the members' differences of opinion which wear them down so much as it is their having *no* opinion, which is what happens when technicians don't attend meetings and participate. IBEW's Draghi is relentless in his belief that no Local is better than the members who comprise it. Prominently displayed on the wall of his office is a list of "Ten Ways To Kill A Union." Some of them are:

1. Don't come to meetings.
2. When you attend, arrive late.
3. Never accept an office; it's easier to criticize than perform.

The face of organized labor has always been changing, but more so perhaps in the technical electronic field than any other. This is not strange, considering the constantly shifting state-of-the-art. Yesterday's breakthrough is today's cliché. And if some new device replaces it all tomorrow, who will have jurisdiction over its use?

Fortunately, efforts are still being made between the various unions in our industry to join forces and share in the benefits of cooperation. ■

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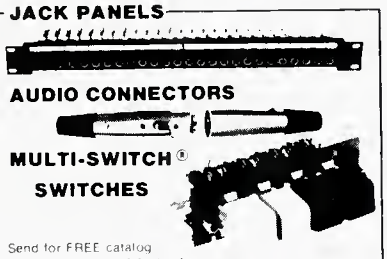
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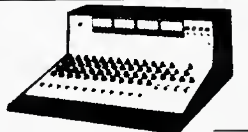
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db People/Places/Happenings

• Assuming responsibility for domestic sales of professional products, **Emory E. Straus** has been appointed marketing manager at **White Instruments Inc.** of Austin, Texas. Mr. Straus had previously been a manufacturers' representative in Chicago.

• Succeeding recently retired **Robert L. Werner, Eugene E. Beyer Jr.** has been appointed Senior Vice President and general counsel of **RCA.** Mr. Werner will divide his time between continuing to serve on RCA's Board of Directors and practicing law in partnership with **Shea, Gould, Climenko & Casey,** of New York City.

• **Marc Sorenson** of **Cal-West Marketing** has been elected 1978 chairman of the **Electronic Representatives Association,** Consumer Products/Distributor Division, Southern California Chapter. Other division officers include **Scott Bassett,** vice chairman and program director, and **Larry Fiege,** treasurer.

• **William L. Fowler** has been elected president and chief executive officer of the **Altec Corporation,** Anaheim, Ca. Mr. Fowler, who has been with the firm for two years, had been serving as vice president and general manager of the company's Altec Lansing International Division. Another appointment at Altec is that of **Peter K. More** to the position of Far Eastern regional manager. At Altec Lansing Sound Products Division, **Charles B. Black** has been named chief electronics engineer.

Moving up from the post of sales promotion manager, **J. Michael Wood** has been appointed retail advertising/sales promotion director at **Radio Shack,** of Fort Worth, Texas. Mr. Wood is a former English and Journalism teacher.

• The position of general sales manager at **Electro-Voice,** Buchanan, Michigan, has been assumed by **David Rothfeld.** Mr. Rothfeld came to Electro-Voice from **Unicord,** a Gulf & Western company. Two other additions to the marketing staff are **Bill Smith** and **Greg Silsby.** Mr. Smith will serve as national consumer products sales manager and Mr. Silsby will coordinate sales of professional microphones and monitors.

• Serving as chief liaison person with retail outlets, **Paul A. McGuire** has assumed the post of sales manager for national accounts at **Audio-Technica U.S. Inc.** of Fairlawn, Ohio. Mr. McGuire came to A-T from **Electro-Voice.**

• **Donald E. Prewett** has been elected president and chief executive officer of the **Phase Linear Corporation,** of Lynnwood, Washington. Mr. Prewett has been with the firm for six years, his most recent assignment as executive vice president.

• **William Gautreau** has been appointed director of materials control, responsible for purchasing parts and materials, at **Switchcraft, Inc.** of Chicago. Mr. Gautreau joined Switchcraft last December after long-term service with **Raytheon.**

• Exhibit space for the fall Technical Conference Equipment Exhibit of **S.M.P.T.E.** is now available. The Conference is scheduled for October 29-November 3 at the Americana Hotel in New York City. Contact S.M.P.T.E. Exhibit Dept., 862 Scarsdale Ave., Scarsdale, N.Y. 10583. (914) 472-6606.

• The entire cutterhead division of **Holzer Audio Engineering Corporation** has been acquired by **International Cutterhead Repair** of Teaneck, N.J., owned by **Sharon Rand Burch.** Mrs. Burch was at one time employed by HAECO, where she received her training as a cutterhead repair technician.

• **Robert Dreisbach,** formerly director of engineering at **Magnavox Government and Industrial Electronics Company,** died on May 21 in Fort Wayne, Indiana at the age of 71. Mr. Dreisbach was the owner of twenty patents in audio engineering, encompassing developments in loudspeakers, record changer, phonograph pickups, microphones, radios, acoustics, sound reproduction, and electronics. In his honor the Magnavox Company has established the Robert E. Dreisbach Acoustic Laboratory.

• The **Audio Engineering Society** has awarded its gold medal in recognition of outstanding achievements to **Daniel R. von Recklinghausen,** vice president of research and development for **KLH/Burwen Research.** Cited were Mr. von Recklinghausen's achievements in the field of f.m. receiver technology. He had served as chairman of the A.E.S.'s IEEE subcommittee on frequency modulation receivers.

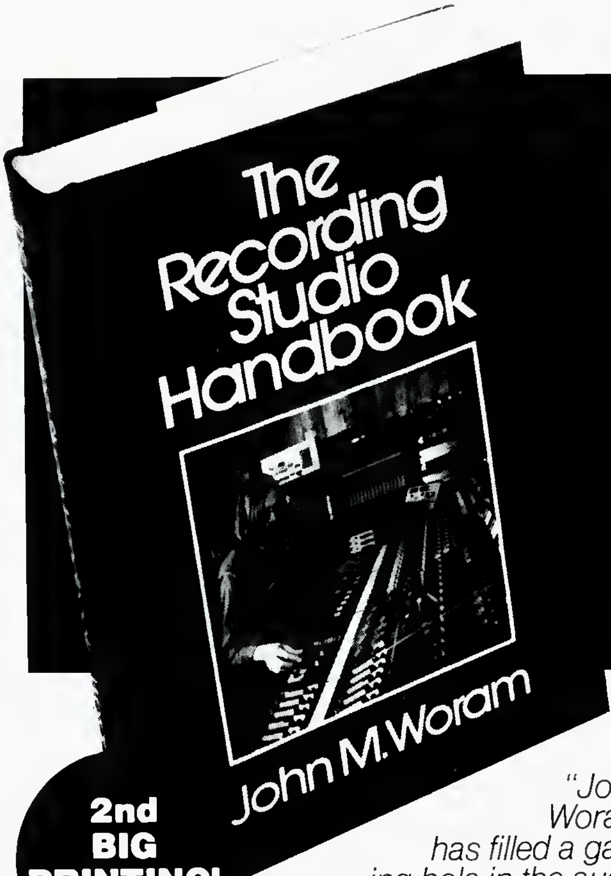
• A new studio complex, **Bee Jay Recording Studios,** has been completed in Orlando, Fla. The studio offers not only a 32-track capacity, but sleeping accommodations for frostbitten northern visitors. The address is 2500 Silver Star Rd., in Orlando.

• The official name of **IEE/Schadow, Inc.** has been changed to ITT Schadow, Inc., according to vice president **Robert G. Inglis.** The plant is located at 8081 Wallace Rd., Eden Prairie, Mn.

• Long-time musician-engineer **Bill Lazarus** has been named manager of disc recording at the **Burbank Studios** in Burbank, Ca. Mr. Lazarus' career includes stints as senior engineer at **Paramount Recording Studios,** general manager at **Angel City,** and special projects engineer for **Motown Record Corporation.**

• Increased production of consumer electronic products has led the **Markham Company** of Van Nuys, Ca. to retain **William Mayhew** as sales manager for key accounts and distributors. Mr. Mayhew comes from **Fairchild Consumer Products.**

• Public school involvement with audio education is underway in Denver, where the **Denver Public Schools** sponsor a two-level intensive course in audio engineering in their Career Education Center. The center is a new vocationally-oriented school which aims at providing high school students with job entry-level skills. The audio department boasts a multi-track recording studio with mixdown facilities and an electronics lab.



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