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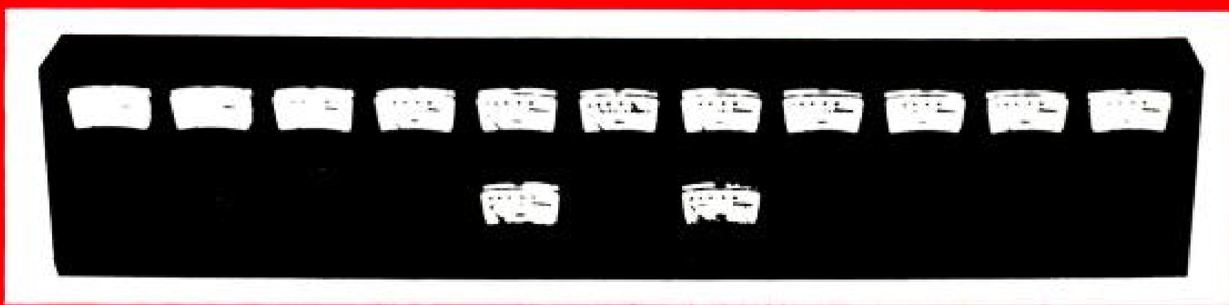
THE SOUND ENGINEERING MAGAZINE

JUNE/JULY 1968 75c

The Gain of Audio Amplifiers

A Multi-Channel Studio Console

Picture Galleries—NAB and AES



Coming Next Month

● Next month—a combined July/August issue devoted to the special problems of disc playback. The burning question of compatible stereo discs as they affect the mono broadcaster will be explored by John Bubbers. David Greene will look at these discs from the viewpoint of what improvements can be made by the master recordist. Arnold Schwartz has prepared a definitive article on disc playback tracing distortion.

In addition, David Hancock will discuss a new ultra-high-quality ribbon microphone.

And there will be our regular monthly columnists: George Alexandrovich, John A. McCulloch (freshly returned from his honeymoon), Norman H. Crowhurst, and Martin Dickstein.

Next month in **db, the Sound Engineering Magazine.**

About the Cover

● This posterization of an audio console may be seen in more normal view in Clair Krepps' article on his notable studio installation. It begins on page 22.

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THE SOUND ENGINEERING MAGAZINE

June 1968 • Volume 2, Number 6

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The Audio Engineer's Handbook

GEORGE ALEXANDROVICH

● Last month's installment discussed some basic functions of the tape recorder. It described alignment procedures for the reproduce and record functions. Now we will turn to an equally important function—the erase function.

It is, of course, possible to make perfect recordings without using the tape machine's erase. All that is necessary is to use virgin tape and use it only once. But present technology often requires some tracks to be re-recorded or remixed with others. So the erase function becomes as vital as record and reproduce.

To record any track of a modern multi-track machine in perfect synchronism with previously recorded tracks requires that the monitoring of the earlier tracks be done with the record head. The track that is being recorded must have its erase working and the record head recording. All other channels should have their erase disconnected with their record heads connected to the playback amplifier for monitoring.

Erase must be specifically designed and working so that a sufficient erasure of the previously recorded information (perhaps an unsuccessful take) takes place without affecting the adjacent tracks. Erase must also be designed so that no audible clicks are recorded on any tracks when the transport is started or stopped. The final master tape must be free of unwanted signs that it was recorded in parts at different times.

In a professional machine, the erase head is either fed from a separate oscillator, from the bias oscillator as part of an erase oscillator tank circuit, or directly fed by the bias oscillator. In this last instance the inductance of the erase head is put to work by resonating the head with the oscillator frequency.

In machines with the bias oscillator doing double duty it is vitally important that the oscillator have the lowest possible distortion, since it can become a source of noise in recording. Bias current produces more noise than the erase function if the distortion is high. This is because during the erase process the tape is saturated, while recording bias only *liquefies* the magnetic domain, thus allowing it to be easily magnetized by the audio frequencies. The oscillator that is common to both functions is more prone to higher distortion because it is loaded more heavily. (There is a higher possibility of the erase head loading the oscillator output—producing higher distortion.)

The first check of the erase function is for proper erasure of recorded material. If erasure is incomplete check the erase head for cleanliness. Then check to see that the tape-to-head contact is good. If incomplete erasure persists, check tape alignment and finally the oscillator itself.

If the hiss level is high it is necessary to first determine what has caused it. It could be record-head bias. Place the machine in the record mode and slip a sliver of paper between the tape and the face of the record head. If the noise subsides, it is the bias oscillator that is at fault.

If the noise persists, however, do the same thing with the erase head. (When it come to checking the erase and bias waveforms with a 'scope, be sure to use a low-capacity, high-impedance probe in order to avoid loading down the oscillator.)

PARASITIC ERASURE

While the erase function is being checked, the selsync function and the ability of the machine to record tracks selectively can also be tested. In any multi-track machine you should test the effects of the erase function on the adjacent channels (parasitic erasure).

During the recording session, when new tracks are being dubbed on to the previously recorded tracks, it is recognized that not all tracks will be perfect the first time. The recording engineer (and talent) must make up his mind on the spot as to the usefullness of a particular take—and live with this decision to the end.

This means that once a track has been recorded and kept, it must not be affected in any way by parasitic erasure from another track. There must be no clicks when the machine is started or stopped or the entire tape will be ruined.

The frequencies most affected by a weak current from an adjacent track are the high ones. So conduct your tests with a recorded tape carrying about a 10 kHz signal on its tracks. Now measure the output, note it, and compare it with the same output with an adjacent channel recording in selsync. Even 0.5 dB reduction is excessive when you consider that in six passes the level may drop 3 dB. Tape misalignment or excessive erase current may be the fault if loss is present. A more complete selsync performance test consists of determining if all switching functions are accomplished without any side effects.

SUMMING UP

The tape machine can be cared for by doing relatively little. But even a small effort on the part of the maintenance man will pay off nicely. Periodic cleaning, since a tape machine easily becomes fouled, will result in lowest flutter and good head contact. If it is appropriate to your machine, check the motor pulley contact to the flywheel rim as well as the ability of the motor to decouple from the flywheel rim when power is turned off. Unless this is properly accomplished, semi-permanent deformation in the compliant surface of the flywheel rim will result in audible flutter.

Basically, therefore, maintenance centers around the concept of *preventive* maintenance. Once damage occurs to a part there is little to do but replace it, either in the studio or by return to the factory. It is thus strongly advisable to comply with the manufacturer's instructions as closely as possible.

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Letters

The Editor:

Without trying to detract from Leon Wortman's article Motion-Picture Sound Systems in the April issue, I would like to clarify a few points that he brought up. Mention was made of *The Jazz Singer* being "the first motion picture employing sound." This is one of those popular legends such as Robert Fulton inventing the steam boat.

As far as I know, the first motion picture with sound was shown to Thomas A. Edison at his laboratory on October 6, 1889. It featured one of the men who had finished the development of the motion-picture sound system, Mr. W. K. Laurie Dickson. It was lip-synchronized, and its subject was welcoming Mr. Edison back from his European trip. Like the Vitaphone process of later years it was a system that synchronized a film to a phonograph disc—only it was a phonograph *cylinder* in Edison's time.

In 1913, Mr. Edison released a talking picture show that employed phonograph synchronization (and utilizing a remarkable mechanical amplification device for the sound). It was shown for several months at Kieth's Colonial Theater in New York, and later elsewhere; however, it did not catch on for some reason.

The first public showing of the Warner Brothers' Vitaphone process took place in New York, at the Manhattan Opera House on August 6, 1926. The first words, introducing the process, were spoken by Will Hayes. Several short subjects besides the Hayes film were shown: these were films featuring celebrities such as Mischa Elman. The feature film was *Don Juan*, which had accompanying music and sound effects, but no dialogue. It was not a phenomenal success, but good enough to allow the Warners to make *The Jazz Singer*—released about a year later.

Mr. Wortman erred slightly when he said: "With magnetic tape came the opportunity for stereophonic sound in theaters." The most famous example of pre-magnetic-sound-track stereophonic sound is found in the Walt Disney feature, *Fantasia*, whose original stereophonic process was called "Fantasound." Although audiences liked what they saw, *Fantasia* was not a financial success because (besides heavy production costs) when it went on road shows, all the necessary reproducing equipment had to go along with it. Since the equipment weighed nearly 1500 pounds and took up enough space to fill half a freight car, I am amazed that they even tried it at all.

World War II put an end to *Fantasia's* runs, though it has been shown in isolated theaters since then.

In addition to *Fantasia*, there were other directional sound systems tried before magnetic sound tracks were released on theatrical films. In 1937 a demonstration of stereophonic sound with motion pictures was given by Bell Telephone Laboratories. It was a public exhibition that featured a two-channel, true stereophonic system, and it utilized twin variable-area sound tracks.

Warner Brothers released a film in 1937 that had multiple sound tracks. It was called *The Eternal Road*. In it, separate tracks were used for music and for dialogue. There was, however, no localization; rather, it enabled the orchestra to appear to be spread out behind the screen through the use of multiple speakers.

I could cite other examples, but I believe I have made my point. Stereophonic sound did not *have* to wait for magnetic-sound tracks, and indeed, in many cases, it did not.

Just as optical sound has offered improvement over original phonographic methods of sound reproduction, so too has magnetic sound offered advantages over optical. So, I would be the last one to throw brickbats at Mr. Wortman. His article is informative and useful—I am merely writing this to throw a little light on the earlier aspects of motion-picture sound than were covered in his article.

Stephen A. Kallis, Jr.
Acton, Mass.

Mr. Wortman responds: If my assumption is correct that Mr. Kallis' knowledge and information of the history of "talking pictures" is accurate, then I am most grateful for the knowledge he has given to me. With respect to his polite disagreement with my statement: "With magnetic tape came the opportunity for stereophonic sound in theaters." I believe Mr. Kallis' statements reinforce my own. Historic films such as Fantasia *and* The Eternal Road *did much to prime and stimulate audiences. However, they were neither financially nor physically feasible. They did provide the initial excitement but not the real opportunity.*

My thanks to Mr. Kallis for sharing his knowledge with us.

Leon A. Wortman
Marketing Manager
Professional Audio Products
Ampex Corporation
Redwood City, Calif.

ABOUT db OR dB

In the April issue, our Editorial discussed the question of preference among our readers for the symbols of our trade. Early returns so far are much in favor of retention of earlier standards. But there are exceptions; from the already quoted Mr. Kallis (above):

I have no preference between the (actually correct) dB and (the commonly used) db; I most emphatically endorse the use of Hz, however. Its utility over c/s, cps, cs/sec, or cy/sec is small, but much too often I have seen that parameter referred to only as "cycles". Hertz is much more clearly defined than is cycles. (How many radio stations sign on giving their broadcasting frequency in kilocycles per second ?) Though at that, some do sign on giving their frequency in kiloHertz, which is correct. I'd vote for Hz, mV, and μA.

Stephen A. Kallis, Jr.

The Editor:
It is my feeling that the symbol cps or cs/sec. should be retained in technical writing as it is more meaningful to the reader. This is why I and my publishers have retained the symbol cps in the forthcoming Second Edition of the Audio Cyclopedia to be released in September 1968.

H. M. Tremaine D.Sc.

Is the term cycles less clear in its meaning than it should be? Our Merriam-Webster defines a cycle as "An interval or space of time in which is completed one round of events or phenomena that recur regularly and in the same sequence; . . ." There may be valid argument for the adoption of hertz as the standard nomenclature—but we feel clarity is not an overriding factor. More next month. *Ed.*

The Editor:
I have just read with interest your editorial in the April, 1968 issue. Your reference to the March IEEE Convention held in New York and the corresponding lack of interest and equipment devoted to the audio field was of great interest to me.

As you probably know, the National Association of Broadcasters (NAB) holds its annual convention during March or April of each year. The 1968 Convention was held at the Conrad Hilton Hotel, Chicago, Illinois, April 1-3 with a combined engineering/management attendance of approximately 5000. The 1969 Convention will be held in Washington, D.C., March 23-26, at the Shoreham and Sheraton-Park Hotels.

An important part of this over-all effort is devoted to approximately 70,000 square feet of exhibit floor space. We are extremely pleased that a good portion of the equipment on display is directed to the latest techniques and developments in the field of audio and its application to broadcasting. Although other organizations may be de-emphasizing audio, I would like to take this opportunity to assure you that the display of audio equipment is an important part of NAB annual conventions.

George W. Bartlett
v.p. for engineering
National Assoc. of Broadcasters

Last month's coverage of the NAB, continues in this issue. The exhibition of equipment at the NAB Conventions is the single largest devoted to professional audio (even though audio per se is only a part of the over-all exhibition). Of course, NAB is devoted to those aspects of audio involved in broadcasting. The Audio Engineering Society is the only other professional audio national organization that has equipment exhibitions; two per year are held—one in New York and the other in Los Angeles although the AES is involved with all aspects of professional audio, the emphasis seems to be more on areas outside broadcasting. The result is that both groups succeed in complementing each other beautifully. *Ed.*

The Feedback Loop

JOHN A. McCULLOCH

The Feedback Loop invites your questions on any subject pertinent to professional audio. Address your queries to The Feedback Loop, **db Magazine,** *980 Old Country Road, Plainview, N. Y. 11803. Please enclose a stamped, self-addressed envelope. Mr. McCulloch will answer all letters in this column or by mail.*

● When a microphone or other component is out-of-phase, the first reaction is to correct the condition. For normal operation it is certainly desirable to maintain all components in a correct phase condition. However, there are some operating situations in which the use of out-of-phase components is of specific assistance to the user.

Theoretically an out-of-phase condition will completely cancel identical waveforms, but because of small differences in mechanical or practical electrical systems we can not expect *complete* cancellation. Nevertheless, the degree of cancellation that is realizable is sufficient to produce many different and useful controls for the user.

Two microphones, with responses nearly identical, will produce an interference curve somewhat similar to Figure 1. In this case the microphones were connected in parallel, and out of phase to each other. They were positioned side-by-side. Generally they would not be suitable for use in this position. But—do you have several of the same microphone? This particular configuration may be used to audibly check similar microphones. If the response and level drops noticeably, and only reduced extreme high frequencies are heard, the microphones may be considered as closely matched.

Using an oscillator it is also possible to check amplifiers and other components that are supposed to be identical. While it may not be an exact method of calibrating components, it is valuable in a quick checkout of facilities.

Separating the microphones causes the various responses in Figure 2. By using differently separated distances, it is possible to control the lower frequencies in a room, and thus improve the recording or broadcast. Note that these curves were made with a single sound source. Several users have reported that this method is able to satisfactorily control crowds and other broad-source sounds. One concrete example is the use of two omnidirectional microphones spaced a few feet apart, each being used at six to eight inches by an announcer. Crowd noise at a fight was so well reduced that an additional microphone had to be employed to produce crowd background.

Another use of the out-of-phase condition is for a stage play. In one installation, seven microphones were placed in the footlights across the front of the stage. The intent was to open only the microphone directly in front of the performer or performers. As additional gain was required in the room, the operator chose to place the *center* microphone on a control, then microphone *one* in parallel with microphone *five*; mics *two* and *six*, and *three* and *seven* formed the other combinations. Each pair of microphones was connected out of phase with respect of one to the other. Thus when mic *one* was opened mic *five* also came up, and the room was partially canceled. It is an admittedly awkward method to mix, for the operator must remember the duality of the microphones. However, the result of this arrangement did give an additional 6 dB more gain to the system (FIGURE 3).

Special effects might be another re-

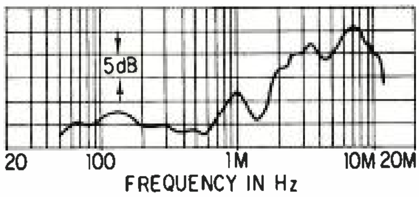

Figure 1. An interference curve caused by nearly identical microphones placed side-by-side and out of phase with each other. The single sound source is two feet away.

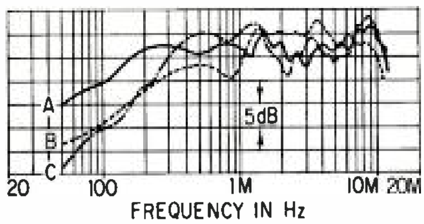

Figure 2. Conditions are identical to those in Figure 1 except that at (A) the mics are four feet apart; at (B) they are one foot apart; at (C) the distance is two feet. In all cases the single sound source was two feet in front of the left mic (and thus further from the right mic).

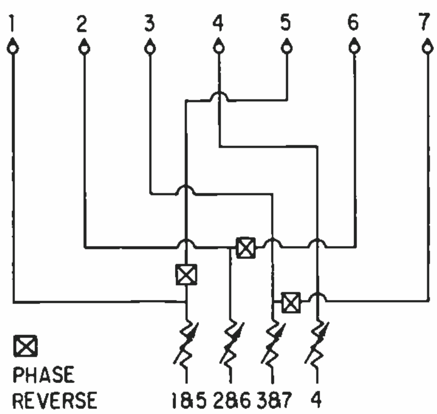

Figure 3. A system that achieves greater gain from individual mic placements in a stage-footlight position.

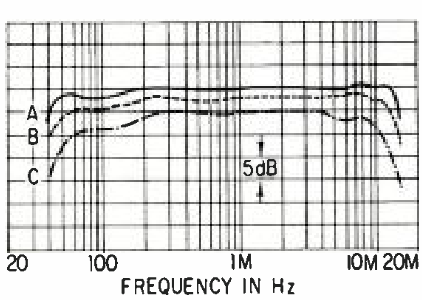

Figure 4. The normal open-circuit response of a microphone is shown at curve (A). This is properly matched at 250 ohms and is the same for open circuit. At (B) loading effects of 150 ohms are shown while at (C) heavy oad ng with 50 ohms is shown.

sult of the combination; for example two dissimilar microphones can be used to create a new voice, or to make an instrument into a sound effect. The high overtones of an instrument might be selected, and just that portion of the performance kept. Gadgetry, to be sure, but if used creatively, certainly a valuable tool.

LOADING

Any *harmful* condition which would normally be avoided, or corrected when found to exist, should also be explored for possible advantages. Such a case is the effect of loading a microphone. Perhaps the same result is true for all components, but I have only calibrated the effects on dynamic microphones. For example: Figure 4 shows the normal open-circuit response of a microphone, in curve (A). Curves (B), and (C), are the result of loading, and heavily loading the same microphone. Not only response is lost, but the level of the microphone is reduced. The loading applied here was purely resistive, but coils also affect the microphone. Do you need both high and low roll-off, and a pad? Here it is! If you are recording voice, and must use a microphone with too wide a response (and no filters available), some control may be achieved by this method.

ASSISTANCE IS NEEDED

At the present time the only connector having a written EIA wiring specification is the UA. Because of recent requirements in the manufacture of equipment, and the trend to potted construction, I have become involved in researching the present wiring practice in the XLR type of connector. Seemingly standard is the use of pin 1 for the ground, or shield. Pins 2 and 3 are signal, but without a designation of phase for balanced-line construction. I am asking the readers of **db** to assist in determining the preferred wiring of this connector. To enable a quick tally, with a promise of a report on the result and the probable issuance of a standard, please return the information requested below (a post-card will be good enough):

1. Type of Installation (t.v., f.m., a.m., recording, film, p.a., etc.)
2. Number of microphones in use.
3. Number with XLR-type connectors
4. Present wiring practice—pin 1____, pin 2____, pin 3____ (use GND, HOT, LOW designations.
5. Name, firm, and address.

Theory and Practice

NORMAN H. CROWHURST

● Quality monitoring has always been a problem, I suppose. One reader's approach, at least, is to use electrostatic speakers. But even with these, he's not quite satisfied. He thinks they're the cleanest thing there is, but still leave something to be desired.

Now I don't want to get into any argument about which is the best monitoring system. I'll satisfy everyone (I hope) by saying there are several ways to go about it, any of which can be as good as the person who puts it together wants to make it. And then in the final comparison, since none of them is perfect, the choice is a matter of taste.

But this particular reader raises some questions in wanting to improve the electrostatic approach that bring in matters that belong in this column. So we'll deal with them. First he says the electrostatics (he's using the British-made **Quads**) sound perfectly good at low level but could be cleaner when he pushes them a little harder.

Here we must discriminate between different ways that speakers can sound *unclean*. This may be more obvious in poor quality speakers, but the same distinction still persists when you're trying to get out that last little bit of audible distortion.

In the early days the more obvious distortion was over-emphasis of certain frequencies—resonances. The flatter a speaker's frequency response, the cleaner it sounds, for this reason. It doesn't emphasize any frequencies more than others. For this kind of cleanness, a high-quality electrostatic is undoubtedly unsurpassed. It's relatively simple to make one that's free from coloration.

Oddly enough, we can learn to live with coloration better than we can with the other kind of distortion. It's like wearing rose-colored spectacles. After you become accustomed to seeing everything pink, you can distinguish most of the differences between colors that you could without the spectacle coloration. Likewise, you get to ignore the note that always gets hit a little harder than all the others.

But distortion that changes the shape of the waveform, introducing harmonics and other spurious frequencies that weren't in the original at all, takes a little more effort to ignore, especially as your critical faculties develop.

If you don't know what makes an automobile motor run, you probably would never notice a noisy tappet. But as you learn about the noises the motor should make, that little tappet noise, every twice around of the mainshaft, starts to bother you.

So, when you push an electrostatic too hard, the dielectric which the movement of the diaphragm has to compress, exerts a non-linear restoring force, and distortion becomes noticeable, if still small. This was what is bothering the reader who wrote to me. And he had some suggestions he wanted me to help him implement, to overcome the effect.

His first suggestion was to use a.c. bias instead of, or as well as, d.c. bias, as in tape recording. If a.c. bias can linearize the magnetization of tape, which has a non-linear magnetizing characteristic, why couldn't it linearize the non-linear properties of an electrostatic loudspeaker?

Possibly it could, if the non-linearity in the speaker dielectric were to have the same, or similar, mechanical characteristics to the magnetic characteristics of the tape. But it doesn't. The magnetic characteristics have non-linearity combined with hysteresis—a delay of the magnetization in following the magnetizing current. And the distortion is associated with this hysteresis, essentially.

Look at a transfer curve, for magnetization against magnetizing current, taken slowly, so you can see what happens (Figure 1). If you start from a demagnetized condition, the first response is slow, (A). Then it gets faster, (B), and finally it slows down again, (C), as you run into saturation.

Now, when you start to remove the magnetizing current, the magnetization doesn't immediately respond. When the magnetizing current is completely removed, considerable magnetization remains, (D). Reversing magnetizing current speeds up the change, but finally slows down again, (F), when saturation is reached in the opposite direction.

Let's separate the hysteresis from the distortion effect (Figure 2). If we had only non-linearity, there would be no loop. The up and back trace would cover the same track. If we had no non-linearity, the hysteresis would make what would otherwise be a straight line open out into a perfect ellipse.

The magnetization characteristic combines both, but the non-linearity is essentially coupled with the hysteresis. Smaller loops take a different shape

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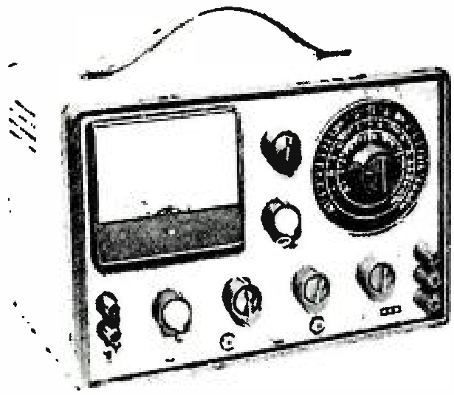

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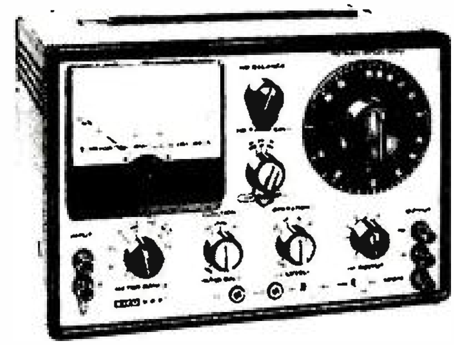

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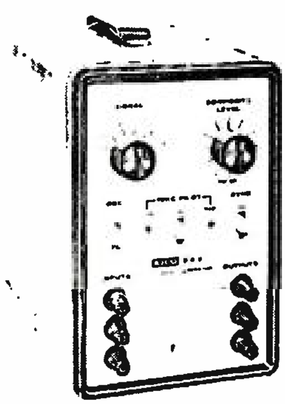

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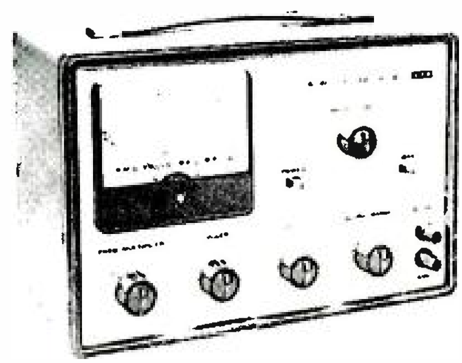

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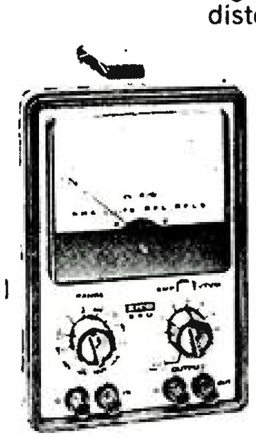

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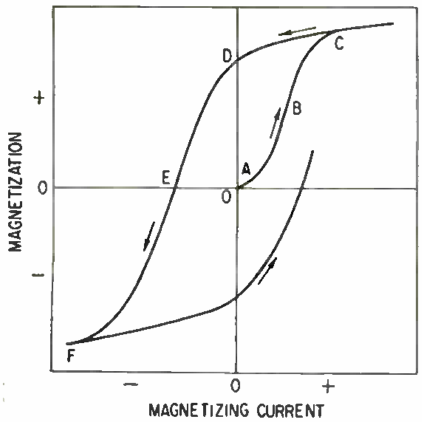

Figure 1. The transfer curve of magnetization against magnetizing currents.

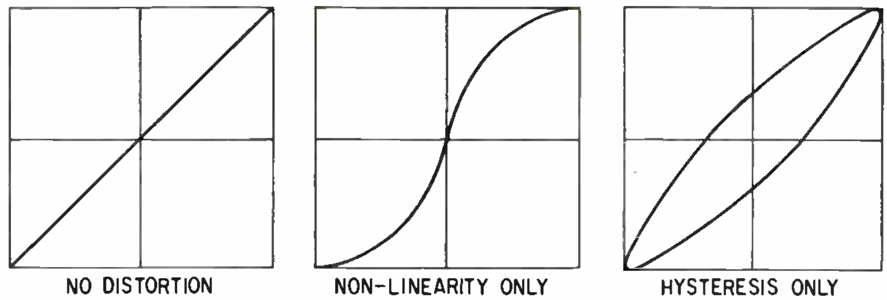

Figure 2. The separation of hysteresis from the distortion effect.

(Figure 3), from the bigger ones. This fact enables ultrasonic bias, using a head gap whose width is such that several cycles of bias occur while a spot on the tape is passing the width of the gap, to virtually *demagnetize* the tape to a polarized value that represents the instantaneous value of the audio.

This procedure has no possible counterpart in the electrostatic-speaker dielectric. If a bias is applied, it will be stressing the dielectric uniformly all the time, instead of dying away, as it does when the tape moves away from the gap. Presumably the ultrasonic bias will also produce an ultrasonic output from the speaker, which could worry the neighborhood bats. But it cannot help work out the distortion that occurs to all waveforms uniformly.

If the electrostatic speaker dielectric does have any hysteresis, which would be due to mechanical viscosity of the material, combined with the elasticity which is its desirable property, this will produce a time delay in all movement, related to the viscosity, and not to the distortion properties caused by compression not being directly proportional to applied force.

So, we see, high-frequency bias cannot help an electrostatic, as it does the tape recorder.

Our reader had really been searching for an answer. His second idea was to use motional feedback. Let's say that this idea could help, if we could find a way of applying it. His idea was to utilize the fringe part of the diaphragm, isolated from the main part, as a capacitor microphone. He didn't draw me a picture, but I suppose Figure 4 would illustrate what he had in mind.

Would this work? One of the arguments he suggested was that over-all feedback in an amplifier is better if it includes the output transformer. This is true, but an output transformer is a little different kind of device from the electrostatic speaker. There are several difficulties.

For the idea to work, the whole diaphragm would have to be rigid enough to be certain that, at all frequencies, the whole of it moved together, with a uniform flexing motion. The only way this could be guaranteed would be by making the movable part of the diaphragm far more rigid than it is practical to make so large a surface needing to move at audio frequencies; and then provide a flexible part as suspension, which couldn't even be part of the feedback pickup.

This is a pretty impossible task, from the practical viewpoint. The good electrostatics we have would not do with any easily-made modification. The unit that could use this approach would have to be a completely new design, using new theory from that employed in existing units.

Then there's the problem of isolating the motional feedback electrically. Both have to be very high impedance, and the transfer efficiency is extremely low. It would be virtually impossible to pick up an electrical output that was usable, without more amplification than a feedback loop can be engineered to work with, and without having some stray breakthrough from the output drive, to the feedback pickup, electrically.

We think of using shielding to stop this. But no shielding is perfect. It only attenuates the effect. And for this application, the attenuation would have to be something like 200 dB, which is a physical impossibility, possibly more so than making that much gain stable with feedback.

Now I've said my piece about that, perhaps I've given designers an idea. Modern technology always seems to be finding a way of doing something that yesterday we said was impossible. Anyway, I'll say that some highly sophisticated technological development is needed, before it is possible.

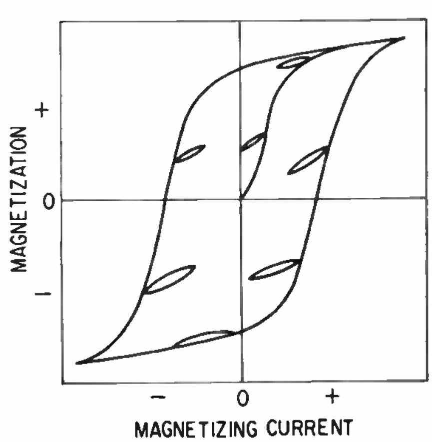

Figure 3. When linear distortion and hysteresis are combined, the non-linearity couples with the hysteresis.

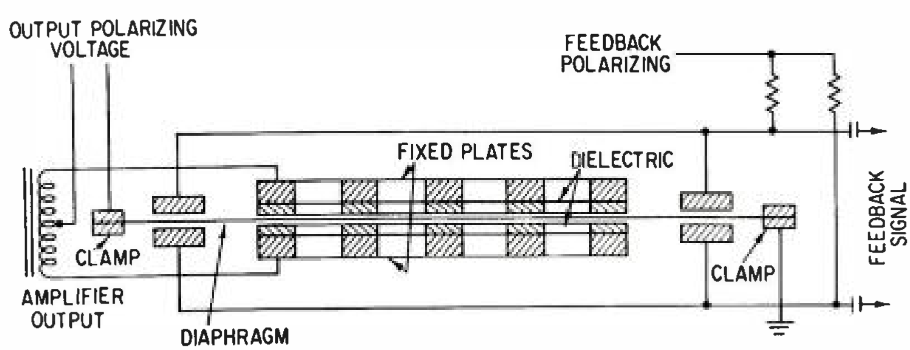

Figure 4. Would this method of using motional feedback in an electrostatic speaker reduce distortion?

Editorial

THE FUTURE OF PROFESSIONAL AUDIO is extremely bright. The demand for skills is on an upward spiral and is reflected in rising wages. As leisure becomes an ever larger part of the average American's life, the call goes out to the electronics field to fulfill expanding home entertainment needs.

Television, radio, film, and recording (both visual and sonic) have open-ended futures, growth possibilities limited only by the human imagination. . . and the availability of skilled personnel.

Where will tomorrow's engineers come from?

As automation of audio production continues, two distinct classes are developing: the technician—a semi-skilled operator, and the engineer—the man who can create, innovate, and service the tools of tomorrow. Salaries will divide sharply along these lines.

Education is the key. On-the-job training is invaluable, but it simply does not substitute for the fundamentals learned in the classroom. Fortunately, as professional audio has grown, it has begun to command the attention of degree-level courses that turn out qualified people.

John McCulloch, in THE FEEDBACK LOOP, recently asked for the names of institutions offering this kind of education. **db** is building a list for publication. We want it to be as complete as possible. If you know of courses of value in our fields, write John McCulloch. He will spread the word.

•

WHY is June **db** so late?

This issue is our first with a new printer; sharp-eyed readers will note some necessary changes in style as a result.

From here on, we do intend to get **db** to you in the month of its issue. You will receive a combined July/August issue in the early part of August. The next issue, September, will arrive at the beginning of that month, and you can expect all subsequent issues on time. —*L.Z.*


SIMPLE
Scully

The Gain of Audio Amplifiers

MELVIN C. SPRINKLE

Almost anyone who designs, uses, or buys audio amplifiers, occasionally misuses terms that describe the gain of amplifiers. The author describes the basic concepts of amplifier gain, how it is defined, and how measured.

THE BASIC PURPOSE of an amplifier is to increase the available power. Amplifiers are used to drive loudspeakers or other amplifiers, or to perform some other useful function, because in most cases the energy that is available from a source such as an f.m. tuner, phonograph pick-up, or even an audio oscillator is too small to do the job. Thus, in the usual case, amplifiers are associated with a *power* increase, although in some applications amplifiers are used for isolation or for *voltage* increase. In such cases the power increase is incidental to the intended function.

The gain of amplifiers is usually expressed in decibels as a matter of convenience. The gain, in decibels, is defined by the classic relation:

$$dB = 10 \log_{10} P_2/P_1 \qquad (1)$$

where: P_1 = reference power
P_2 = some other power

As a matter of convenience in amplifier work, P_2 is usually taken as the larger power so that the power ratio is a number larger than one. As can be seen from the above equation, the decibel is a logarithmic unit which provides several important advantages:

1. Quite often the gain of amplifiers runs into inconveniently large numbers which are tedious to handle and are conducive to mistakes. The logarithmic nature of the decibel considerably reduces the size of numbers required to express gain.
2. When amplifiers are connected together in cascade (one after the other), the total gain (when expressed in other than decibels) must be obtained by multiplying the individual gains. By using the logarithmic decibel system the gains are added, and the numbers used are smaller as mentioned above.
3. If devices which produce or cause loss are used, the losses, expressed in decibels, can be combined in an addition or subtraction process to evaluate the system performance.

Whenever one hears the term *gain* (or one of the other commonly used terms mentioned in the opening paragraph) used in connection with an amplifier, the property that is meant is almost always the *transducer gain* which is commonly called *insertion gain.*

These terms have been defined[1] as follows:

Insertion Gain: Resulting from the insertion of a transducer in a transmission system, the ratio of the power delivered to that part of the system following the transducer to the power delivered to that same part before the insertion of the transducer.

Transducer Gain: The ratio of the power that the

Mr. Sprinkle is a project engineer with Page Communications Engineers, Inc. of Washington, D.C.

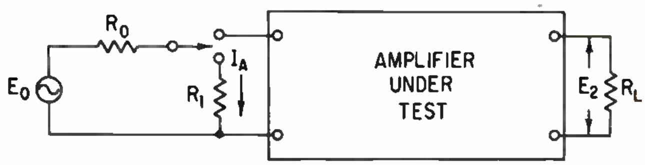

Figure 1. The measurement of insertion gain.

transducer delivers to a specified load under specified oper. ating conditions to the available power of a specified source-

Transducer: A device capable of being actuated by waves from one or more transmission systems or media and of supplying related waves to one or more other transmission systems or media.

Transducer, Active: A transducer whose output waves are dependent upon sources of power, apart from that supplied by any of the actuating waves, which power is controlled by one or more of these waves.

Thus an amplifier is an active transducer, and the property under consideration (transducer gain) will be called hereafter by the more popular precise term, *insertion gain.*

It will be seen that the insertion gain is directly the opposite of insertion loss.

As an example of insertion gain, consider an audio oscillator of 600-ohms internal source impedance which, when connected to a 600-ohm load resistor, delivers one milliwatt of power to the resistor. This is the reference power, P_1. Then, the load *is removed* from the oscillator and an amplifier is connected in its place. The 600-ohm resistor is then connected to the amplifier's 600-ohm output so as to become the amplifier's load. The power developed in the resistor is now measured and we find it to be five watts. Then the insertion gain is:

$$\text{Insertion gain in dB} = 10 \log_{10} P_2/P_1 = 10 \log_{10} 5/0.001 = 10 \log_{10} 5000 = 37 \text{ dB} \qquad (2)$$

Now suppose that the amplifier is designed for a rated load impedance of 16, instead of 600 ohms (like a loudspeaker). Note that the definitions of insertion gain and transducer gain are based upon *power.* Therefore, with the proper load for the amplifier's output, the power is measured; the insertion gain of an amplifier is *not* affected by its rated load impedance.

Seeley[2] has pointed out several important points with regard to the insertion gain of amplifiers:

1. The amplifier whose gain is of interest *replaces* a load on a generator or source. Gain is the ratio of the power obtained by using the amplifier to the power obtained without the aid of the amplifier.

2. The power absorbed by the input circuitry of an amplifier has *nothing to do* with the measurement, and here is where some of us are misled. The input circuit of one amplifier with 30 dB gain might draw one amount of power from the source, while another amplifier with the same gain might draw 20 dB less (power from the source). The power absorbed by the amplifier's input circuit plays *no part* in the definition of gain and its (the gain's) measurement.

3. The gains of two amplifiers are additive when expressed in decibels. This is not only due to the logarithmic method of expression, but also to the particular definition of gain. We use gain rather than other possible expressions for amplification because gains *are* additive when appropriate precautions are observed. It is very important that when amplifier A with 30-dB gain is followed by amplifier B with 10-dB gain, the gain of the combination is 40 dB.

4. Gain *always deals with power.* Measurements of gain may involve voltage measurement or voltmeters but impedance is always taken into account so that power is the essential quantity.

A term which is frequently heard is voltage amplification, often used synonymously with the term *voltage gain* (which should be avoided).

Voltage amplification is defined as the ratio of the magnitude of the voltage across a specified load impedance connected to a transducer to the magnitude of the voltage across the input to the transducer[1]. IEEE does not define *voltage gain.* Voltage amplification may be expressed numerically as 100 or 100x (100 times) or, more commonly, in decibels (20 $\log_{10}$ of voltage ratio) which would be 40 dB for a voltage ratio of 100. Many times, when someone uses the term *gain,* the property that is meant is voltage amplification. *It is emphasized* that voltage amplification is *not* the same as insertion or transducer gain, although, as will be shown later, in some special cases they may be numerically the same.

FIGURE 1 shows the basic system for insertion gain measurements. E_o is a zero-impedance generator (which can be an audio oscillator). If the output voltage of an oscillator is maintained constant regardless of the amount of power drawn from it (or the impedance of the load which may be connected to it, the source is said to have a zero effective internal impedance.

R_o is the impedance of the source from which the amplifier will work. The value is established by the designer of the system. In testing commercial amplifiers, the value of R_o may be deduced from representative applications of the amplifier, or values may be assigned for typical applications. The value of R_o is very important as the insertion gain depends greatly on its value.

R_1 is the initial value of load resistance in which the reference or initial power is developed. Since the definition for transducer gain refers to the *available power* from a specified source, the case for maximum power applies and R_1 will thus be equal to R_o.

R_L is the amplifier's load resistance across which the amplifier develops a voltage E_2. It is presumed that R_L is the proper value for the tap to which it is connected on the output of the amplifier.

For initial conditions, the switch is thrown so that E_o is connected across R_o and R_1 in series. Let P_1 be the power dissipated in R_1 and let I_1 be the current through R_1.

Then, $$P_1 = I_1^2 R_1 = \frac{E_o^2 R_1}{(R_o + R_1)^2} \qquad (3)$$

and $$P_2 = \frac{E_2^2}{R_L}; \qquad (4)$$

but P_1 was defined as the available (maximum) power and since under these conditions

$$R_1 = R_o \qquad (5)$$

$$P_{1m} = \frac{E_o^2 R_o}{4 R_o^2} = \frac{E_o^2}{4 R_o} \qquad (6)$$

The insertion gain then is:

$$\frac{P_2}{P_{1m}} = \frac{\dfrac{E_2^2}{R_L}}{\dfrac{E_o^2}{4R_o}} = \frac{E_2^2 4 R_o}{E_o^2 R_L} \qquad (7)$$

or, expressed in decibels:

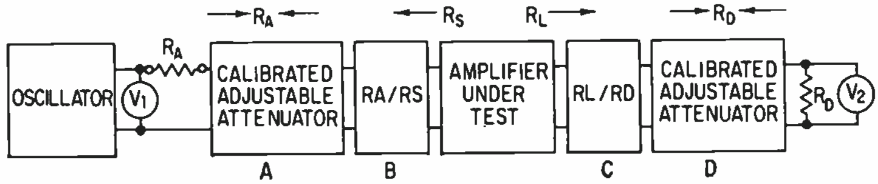

Figure 2.
The EIA measuring system for insertion gain.

Insertion Gain =

$$10 \log_{10} \frac{P_2}{P_{1m}} = 20 \log_{10} \frac{E_2}{E_o} + 10 \log_{10} \frac{R_o}{R_L} + 10 \log 4 \quad (8)$$

or

$$\text{Insertion Gain} = 20 \log_{10} \frac{E_2}{E_o} + 10 \log_{10} \frac{R_o}{R_L} + 6 \text{ (dB)}. \quad (9)$$

On the basis of Equation (9), a procedure may be established for the measurement of insertion gain:

1. Set up the amplifier as shown in Figure 1, with the proper values of R_o and R_L, and an oscillator. R_1 is not used. The oscillator frequency should be around 1000 Hz.
2. With the amplifier's gain control at maximum, adjust the output of the oscillator to give a convenient voltage at E_2. The value of the voltage may be such that either the amplifier is delivering rated power in the mid frequency region, or as much as 10 dB below rated power.
3. Read and record E_2, measuring with a voltmeter whose impedance is high with respect to R_L.
4. Without touching the oscillator, read and record E_o (the oscillator output voltage), again with a voltmeter with high impedance.
5. Insert the values read into Equation (9) and calculate the amplifier's insertion gain.

Note that the first term in the formula is a *voltage ratio* which may be measured with an a.c. vacuum-tube voltmeter. If the instrument used has decibel scales, then, in this case, the formula's first term is the *difference* in decibels between the decibel values read from the instrument for each of the two voltage measurements.

The circuit as shown in Figure 1 is for the unbalanced case. For balanced input or output amplifiers, appropriate measures, such as the use of isolation transformers must be employed to preserve the balanced condition.

While the test setup of Figure 1 will enable insertion-gain measurements to be made, in general it will be found inconvenient because the small signals required for high-gain amplifiers necessitate the use of very sensitive voltmeters, plus elaborate precautions to prevent contamination of the measurement signal with hum and noise which would invalidate the measurement.

Thus it is the usual practice to use calibrated attenuators in conjunction with a more rugged and less sensitive meter (or a higher range on an a.c. vacuum-tube voltmeter), sometimes a v.u. meter. Such an arrangement of attenuators, meter(s) and terminations is called a gain set; these are standard equipment in radio stations and audio laboratories. The gain set permits accurate and repeatable measurements on audio equipment, including the insertion gain of amplifiers.

Reduced to its simplest form, the gain set is a loss box containing attenuators which introduce loss in 10-dB, 1-dB, and, often, 0.1-dB increments. It also contains source resistors and terminations for the attenuator, impedance-changing networks, meter(s), amplifier termination resistors, and. sometimes, post amplifier attenuators. The amount of source loss is usually 110 dB, and those gain sets with 0.1-dB loss-change will usually have 111-dB loss.

Figure 2 is a block schematic of the elements of a complete gain set. This is taken from EIA Standard RS-219, **Audio Facilities for Radio Broadcasting Stations.** It consists of an audio oscillator (which is usually separate from the gain set) across whose output is connected an a.c. voltmeter. R_A is a source resistor which provides the necessary source impedance for the calibrated attenuators (since the output of an oscillator maintained constant is an effective zero-impedance source). The impedance of the calibrated attenuators is also R_A. Next is a network which has two functions: to provide a termination impedance for the variable attenuators and to provide a source impedance (Rs) from which the amplifier is to be fed. For the case of the unbalanced input amplifier, the network will usually be a simple resistive L pad. The design and construction of such a pad has been covered in the referenced literature.[4,3] For amplifiers requiring a balanced source, a repeat coil (isolation transformer), unterminated, is used either ahead of, or following the impedance matching network. It should be noted that a terminated transformer would provide in the matched impedance case, a source impedance of half the circuit impedance. If the transformer is ahead of the matching pad, the pad should be of balanced configuration. Following the amplifier is another impedance matching network or pad, the purpose of which is to provide the proper load for the amplifier and also to match the output circuit attenuator if such is used. In the case of large high-powered amplifiers it may be desirable to use a calibrated attenuator on the output circuit, and such is shown on the diagram. If the amplifier under test has a balanced output, a repeat coil, unterminated, should be inserted in the output circuit between balanced and unbalanced sections. Following the output attenuator is the load resistor which terminates the output attenuator and across which is a high-impedance voltmeter.

Using the test setup of Figure 2, the insertion gain is given by the formula:[3]

Insertion Gain =

$$A + B + C + D + 10 \log_{10} \frac{4(E^2)^2 R_A}{(E_1)^2 R_D} \quad (10)$$

where:

A is the loss, in decibels, of the calibrated input attenuator;
B is the loss, in decibels, of the input impedance-matching network, including repeat-coil losses if a coil is used;
C is the loss, in decibels, of the output impedance-matching network, including repeat coil losses if a coil is used;
D is the loss, in decibels, of the calibrated output attenuator;
E_1 is the voltage read at V_1;
E_2 is the voltage read at V_2;
R_A is a resistance equal to the iterative impedance of the calibrated input attenuator;
R_D is a resistor equal to the iterative impedance of the calibrated output attenuator; it is also the final termination of the system.

Equation (10) may be rearranged:

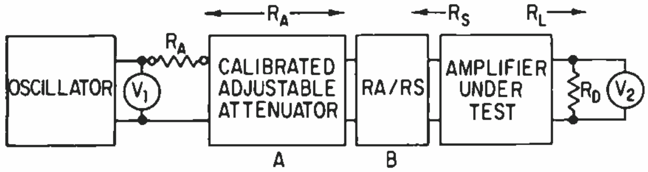

Figure 3. This simplified EIA measuring system can also be used to determine insertion gain.

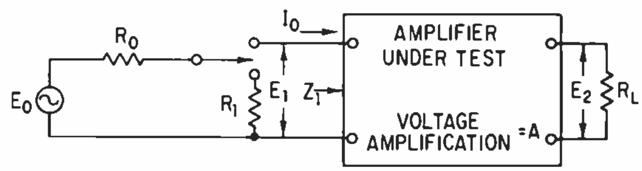

Figure 4. This is the relationship that exists between insertion gain and voltage amplification.

Insertion Gain =

$$A + B + C + D + 20\log_{10} E_2/E_1 + 10\log_{10} R_A/R_D + 6\ (\mathrm{dB}) \qquad (11)$$

It will be observed that this equation is the same as (9), with the exception that the losses in A, B, C, and D have been added.

A simplified version of the same circuit is shown in FIGURE 3, with symbols as before. In this version the output matching network and calibrated output attenuator have been eliminated. Instead, the load resistance for the amplifier, R_D is connected directly to the amplifier's output. Using this arrangement, the insertion gain is the same as that of Equation (11) with the exception that losses C and D are not used.

Examination of Equation (11) shows that, if the oscillator signal and attenuator losses are adjusted so that the source and load voltages are the same, and, if the source and load impedances are the same, the insertion gain becomes equal to the losses in the attenuators and matching networks plus six decibels. Under these conditions the insertion gain is six decibels greater than that expected by considering the commonly heard expression *gain equals the loss.*

Commercially made gain sets may not have the voltage measuring points at the same circuit locations as those indicated in FIGURES 3 and 4, and therefore the gain figures obtained may not agree with those made by the methods presented herein.

RELATION OF INSERTION GAIN TO VOLTAGE AMPLIFICATION

As previously mentioned, and further emphasized here, voltage amplification is *not* the same as insertion gain. It is also emphasized that for this reason the term *voltage amplification* is better than the more commonly heard *voltage gain,* since the latter term tends to continue the confusion.

Refer to FIGURE 4 which is the same as FIGURE 1 with a few more parameters considered.* E_1 is the voltage which is impressed on the input of an amplifier, Z_1 is the input impedance of the amplifier and I_1 is the current which flows through Z_1 when E_1 is applied.

The voltage amplification of the amplifier is A, defined as follows:

$$A = E_2/E_1 \qquad (12)$$

The current drawn by the input circuitry of the amplifier when connected to E_o through R_o is:

$$I_o = \frac{E_o}{R_o + Z_1} \qquad (13)$$

then

$$E_1 = I_o Z_1 \qquad (14)$$

and

$$E_1 = \frac{E_o Z_1}{R_o + Z_1}. \qquad (15)$$

Now, from (12)

$$E_2 = AE_1 \qquad (16)$$

*This material is after Seeley, op. cit.

and

$$P_2 = \frac{E_2^2}{R_L} = \frac{A_2 E_1^2}{R_L} = \frac{A^2 E_o^2 Z_1^2}{R_L(R_o + Z_1)^2}. \qquad (17)$$

As shown earlier in Equation (6) the reference power P_{1m} is:

$$P_{1m} = \frac{E_o^2}{4R_o}$$

Insertion Gain is:

$$\frac{P_2}{P_1} = \frac{\dfrac{A^2E_o^2Z_1^2}{R_L(R_o + Z_1)^2}}{\dfrac{E_o^2}{4R_o}} = \frac{A^2E_o^2Z_1^2}{R_L(R_o + Z_1)^2} \times \frac{4R_o}{E_o^2} = \frac{A^2Z_1^2 4R_o}{R_L(R_o + Z_1)^2} \qquad (18)$$

Insertion Gain in decibels =

$$20\log_{10}A + 20\log\frac{Z_1}{R_o + Z_1} + 10\log\frac{R_o}{R_L} + 6\ \mathrm{dB} \qquad (19)$$

Equation (19) shows that the insertion gain differs considerably from the voltage amplification. Also, to determine the insertion gain, not only must the source impedance R_o be known, but also the impedance looking into the amplifier's input terminals. Z_1 is the *magnitude* of the input impedance in the event that reactive elements are present.

Seeley[2] has pointed out several interesting cases which occur if the ratio of R_o to Z_1 is varied.

Case I. Assume that R_o equals Z_1. This case occurs when an amplifier is fed from a source impedance equal to its input impedance.

then equation (18) becomes:

$$\text{Gain} = \frac{A^2R_o^2 4R_o}{R_L(2R_o)^2} = \frac{A^2R_o}{R_L} \qquad (20)$$

$$\text{Gain in decibels} = 20\log A + 10\log\frac{R_o}{R_L} \qquad (21)$$

Case II. Assume that R_o not only equals Z_1 but that R_o also equals R_L. This case occurs when an amplifier has the same input and load impedances and is fed from a matched source,

then equation (21) becomes:

$$\text{Gain in decibels} = 20\log A \qquad (22)$$

This is the *only* case in which the insertion gain and the voltage amplification are numerically the same.

Case III. Assume that Z_1 is very much greater than R_o. This occurs when a high-input-impedance (say 500,000 ohms) amplifier is fed from a low-impedance (cathode-follower) source. In Equation (18), the term $(R_o + Z_1)$ can then be considered to be Z_1 with only slight error under these conditions.

then equation (18) becomes:

$$\text{Gain} = \frac{A^2Z_1^2 4R_o}{R_L(R_o + Z_1)^2} = \frac{A^2Z_1^2 4R_o}{R_L Z_1^2} = \frac{A^2 4R_o}{R_L} \qquad (23)$$

$$\text{Gain in decibels} = 20\log A + 10\log\frac{R_o}{R_L} + 6\ \mathrm{dB} \qquad (24)$$

Note that equations (21) for the equal-impedance case and (24) for the low source-mpedance case differ by six decibels. This effect occurs in amplifiers equipped with input or microphone transformers. If the microphone transformer is terminated so that impedances are matched, the amplifier falls into **Case I.** If the termination is removed, the input impedance is increased and the amplifier falls into **Case III** with a six-decibel increase in gain. It is for this reason that amplifiers with microphone transformers are operated with unterminated secondaries; such conditions give a 6-dB signal-to-noise-ratio improvement, since the gain is increased by 6 dB but the amplifier noise is not.

It can be shown that changing the primary strapping on amplifiers equipped with microphone transformers so as to change the input impedance produces no change in insertion gain, provided that at the same time the sending impedance is also changed to the proper value. For example, suppose we have an amplifier which has a voltage amplification of A and is equipped with a microphone-to-grid input transformer having primary impedances of 30, 150, and 500 ohms. Consider first the 30-ohm strapping for a 30-ohm microphone:

Voltage amplification of amplifier	A
Impedance ratio of transformer	30 to 70,000 ohms or 1:2333
Voltage ratio of transformer	1:48.3
Input impedance (secondary terminated)	30 ohms
Source or sending impedance	30 ohms

then using equation (22),

$$\text{Gain} = \frac{A^2 Z_1{}^2 4R_o}{R_L(R_o + Z_1)^2} = \frac{A^2 \cdot (30)^2 \cdot 4 \cdot 30}{R_L(30+30)^2} = \frac{30A^2}{R_L} \qquad (25)$$

Now, suppose we restrap the primary for a 500-ohm input and change R_o to 500 ohms. Then:

Input impedance (secondary terminated)	500 ohms
Sending impedance	500 ohms
Impedance ratio of transformer	500:70,000 or 1:140
Voltage ratio of transformer	1:11.83
Voltage amplification	0.245A

(this occurs because the voltage stepup in the input transformer is reduced by the factor 11.8/48.3 or 0.245)
then the gain is:

$$\text{Gain} = \frac{A^2 Z_1{}^2 4R_o}{R_L(R_o + Z_1)^2} = \frac{0.060A^2 \cdot 500^2 \cdot 4 \cdot 500}{R_L(500+500)^2} = \frac{0.060A^2 \cdot 500}{R_L} = \frac{30A_2}{R_L} \qquad (26)$$

It will be observed that equation (26) gives the same result as (25); thus, the insertion gain is the same.

The same considerations apply to output transformer strapping, so that, as has been mentioned, insertion gain does not change with a change of tap provided that the proper load is used simultaneously.

THE BRIDGING AMPLIFIER

In certain applications, amplifiers are used to bridge across an audio line for the purpose of picking a small amount of signal from the line and amplifying it to perform some other function, such as driving a monitoring loud speaker. A bridging amplifier does *not* terminate a line; it merely samples the signal and usually has an input much higher than the nominal impedance of the line which it bridges. It is essential that the connection or removal of the bridging amplifier produce a minimum of effect on the circuit being bridged. Accordingly, a bridging amplifier has a high input impedance which is usually provided by a high-impedance input transformer, often equipped with build-out resistors. The resistors are used to increase the impedance of the source seen by the transformer to a value representative of its primary impedance, thus preserving its frequency response.

Like all amplifiers, the bridging amplifier has gain and we should be able to define and measure the bridging gain in a precise manner. By its nature, the bridging amplifier does not perform the same function as an amplifier used to provide increased power from a source such as a microphone, so that a slightly different approach to gain is used.

The *gain of a bridging amplifier* is defined[1] as the ratio of the power a transducer delivers to a specified load impedance under specified operating conditions to the power dissipated in the reference impedance across which the input of the transducer is bridged.

In measuring the bridging gain of an amplifier, a known power representative of operating practice is set up in a resistor having the same resistance as the nominal impedance of the circuit that the amplifier is to bridge (generally 150 or 600 ohms); the amplifier is then connected and the amplifier's power output is measured.

The bridging gain is then (in decibels):

$$\text{bridging gain} = 10 \log_{10} P_a/P_z$$

where:

P_a is the power output of the amplifier

P_z is the power in the resistance across which the amplifier is bridged.

It should be noted that, since the input impedance of a bridging amplifier is high, the bridging amplifier is essentially a voltage-operated device. Since, for a given power level in an audio line or bus, the voltage will depend upon the nominal impedance of the line, the line impedance must be specified when stating the bridging gain.

In some cases a high-input-impedance amplifier may be used either for bridging a line or as a general-purpose amplifier. In such cases, two different values of gain should be quoted in the amplifier's specification: (1) the insertion gain and (2) the bridging gain.

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Ampex AG-440. For the recording engineer who wants to fly.

If you're moving out, taking off with the hot young groups who ride the charts, you deserve the multichannel recorder that will let you be your most creative. The one with the greatest versatility. The one with the fewest technical hangups. The one that will pay for itself fastest.

There's only one like that: the incomparable Ampex AG-440. Read why recording engineers who are helping to create today's big sound prefer to work on an Ampex. Then return to this thought: An AG-440 can be working for you in a matter of weeks; you can use our deferred payment or lease plans, and pay for it out of current earnings.

Now. Ready to fly? Call your local Ampex distributor. If his number isn't on your chart, call our world headquarters, (415) 367-4400 (collect).

The MOJO Courtesy Reprise Records Div. Warner Bros./7 Arts

ADD TRACKS ANYTIME, EVEN SPLIT YOUR SESSIONS

The Ampex-pioneered Sel-Sync® System made multi-channel mastering possible. It gives you the freedom to develop the right sound even after the talent has gone home, to add new dimensions with layers of experimental sound. And if only half the group can make today's session, let the other half do their thing later. It's easy. AND REMOTE SEL-SYNC SYSTEM makes your AG-440 even more versatile. It was previewed at the Spring AES Show. Plan for it on your new board.

LESS DOWN TIME, MORE ASSIGNMENTS, BETTER INCOME

The AG-440 series is the result of ten years of multi-track experience. We know that easy maintenance is a must . . . that's why it has up-front plug-in electronics, tiltable top plate, and the simple straightforward transport. And because you can switch easily from ¼" to ½" tape, you can assign a variety of jobs to this machine, make more money for the studio.

PERMANENT ALIGNMENT SAVES TIME, SAVES MASTERS

If you've ever wrestled with fuzzy sound caused by misalignment, you know how important our heavy die-cast frame is to the AG-440. It provides an inflexible anchorage for tape guides and heads so that alignment and tracking will not present problems to hamper your creativity. And the stable, drift-free electronics keep your signal clear and true.

ADVANCED HEAD DESIGN DELIVERS CRISP SOUND

High crosstalk rejection means your tracks stay separated — until you're ready to mix. Sounds stay bright and brilliant. New "deep gap" head design increases head life dramatically. This exclusive Ampex feature allows your AG-440 heads to maintain full frequency response many times *longer* than conventional heads.

ANNOYING SCRAPE FLUTTER ELIMINATED

The AG-440's extremely low inherent wow and flutter give you full freedom to create the biggest or most delicate sound. Now, in addition, you can get rid of annoying midrange scrape flutter, which has plagued recording engineers in the past. One built-in jewel bearing idler cuts scrape flutter by as much as 75%. A second (optional) virtually eliminates it.

MAXIMUM DYNAMIC RANGE MEANS MINIMUM RESTRICTIONS

Much of the excitement of today's young sound comes from driving the instruments off the top of the scale. To capture these wild experimental sounds, you need the widest possible dynamic range . . . and the AG-440's distortion free swing gets the most from the tape. Coupled with maximum S/N. This means you can mix down and dub through many generations without excessive degradation. And, your multi-channel AG-440 records and plays back an honest 30 to 18,000 Hz (±2dB @ 15 ips).

LET YOUR AG-440 PAY FOR ITSELF WHILE YOU USE IT

Ampex offers a unique choice of lease or extended pay plans. You could, for instance, get a one-channel AG-440 for as little as \$50 a month. Then, because of its modular design you could build it to 2, 3, or 4 channels. Or you can get a complete AG-440-4 or AG-440-8 *now* and pay for it out of current earnings. It's the only way to fly!

Write for your copy of the AG-440 "Flight Plan." It contains full description and specifications.
Professional Audio Products Division, 401 Broadway
Redwood City, Calif. 94063

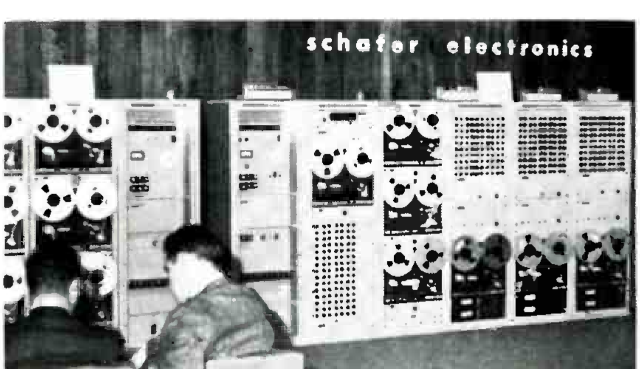
AMPEX

The NAB Picture Gallery

LAST month, space limitations prevented us from printing all the photographs we intended. Here is the balance.

Nortronics Co., Inc. At the right is Joseph Dundovic, marketing engineering manager of the tape-head manufacturer. A complete line of professional head replacement systems for a variety of machines is offered.

Richmond Hill Div., Riker Video Industries. Manufacturer of a wide range of fully transistorized equipment for television broadcasting industry. They also showed an **Electrodyne** audio console.



Schafer Electronics. Complete broadcast auotmation systems. Their computer technology and operation can completely automate the broadcast day.

Seeburg Music Library, Inc. Automated background music systems for offices, hotels, restaurants—as well as telephone line and multiplexing—are available.

Shure Bros., Inc. Microphones, disc reproducing equipment, accessories, and a compact new mixer were shown. Robert W. Carr, manager of professional products is at center.

Sony Corp. of America. Sony offers vtrs, cameras, monitors, viewfinders, and raw tape.

Sparta Electronic Corp. Theirs' is a broad line of audio broadcast equipment. Consoles, tape systems, record playing systems, and (through their **Vega Electronics Co.**) wireless and other microphones.

Tektronix. Oscilloscopes and waveform monitors specifically designed for broadcast application, particularly in color video, are offered.

Telex Corp. Viking and **Magnecord** tape recorders and duplicators, and Telex headsets are displayed. Just left of center is Paul Bunker demonstrating a Magnecord 1020.

Ward Electronic Industries. Audio consoles and intercoms, as well as video switching systems, station-break programmers, phase equalizers, etc. are included in their product line.

An Engineer-Producer Multi-Channel Console

CLAIR KREPPS

The author operates one of the first studios using multi-track recording techniques. Here is his solution to the problems found on the path from microphone to tape.

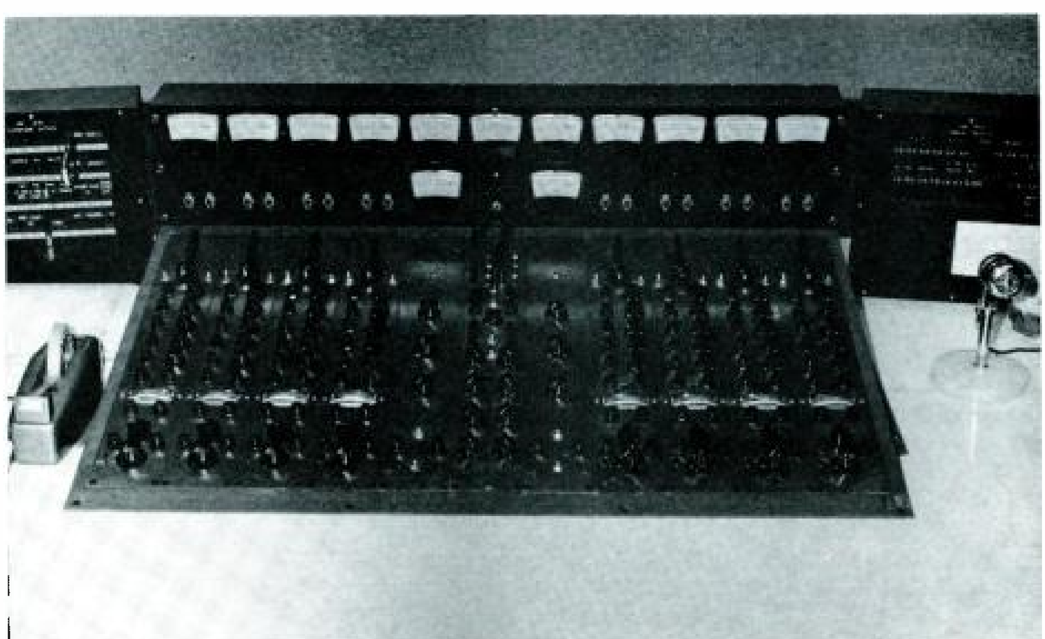


Figure 1. The complete console. In this forty-inch space (plus the four vertical pedestals) facility exists for the use of up to twenty-eight mics capable of recording 12, 8, 4 track and mono, each with independent control. In addition to full console function, this unit is the equivalent of fifteen tape record/reproduce amplifiers.

Clair Krepps is president of Mayfair Recording Studios, Inc., in New York City.

BEFORE ACTUALLY describing the instruments, let us briefly examine the planning that preceded their design and construction. I must begin by sharing any credit with my brother Edgar. Ed has had years of experience designing electronic instruments to be mass produced for the market place. My years have been spent in practical sound recording, studios, disc mastering, film scoring, and working with musicians and producers. This led to many discussions and decisions as our ideas evolved into a practical reality.

This list is a distillation of these discussions in the order of their importance.

1. The system must be completely functional to properly service the present-day popular music production techniques.
2. Clean audio power must be over abundant and easily transferred if the operator wishes to use it for special effects.
3. The basic design must be arranged for easy modifications if it is ever necessary to update it because of yet unforseen changes in the recording business.
4. The cost should be as low as possible without compromising quality or function. The sound recording business these days moves ahead so fast that the day you open your facility you are on your way to obsolescense—not that your equipment isn't doing the job, but the recording business is ever-changing.

To describe the console, the body consists of eight audio control units—*each* with the following functions. (FIGURE 1.)

1. Three microphone inputs and their level controls.
2. Selector switch; mic, tape head, phono and three aux inputs.
3. Compressor with its controls.
4. Echo feed (10 watts), and complete echo mic pre-amp with *off-on* switch, each separately controlled.
5. Its own mechanical echo chamber mounted to the underside.
6. Complete shelf-type equalizer
   high boost—16, 8, 4, 2 kHz
   low boost—500, 200, 100, 50, 25 Hz
   high atten—16, 8, 4, 2 kHz
   low atten—200, 100, 50, 25 Hz
7. Dip filter at 4 and 3 kHz.
8. Peaking equalizer, switchable to 3 or 1.5 kHz bandwidth, at 10, 5, or 3 kHz.
9. The two equalizers, functional in any combination you might want simultaneously.
10. Pan pot with related four-channel selector switches, can be used in any possible combination for eight position stereo discs.
11. Its own vu meter and power supply.
12. Its own speaker and 35-watt amplifier.
13. The output (600/ohms, transformerless) is then carried by wire to a box consisting of a relay and a few resistors and condensers and one coil (trap) and recorded on tape *without* any further amplification.

MIXING OR COMBINING AMPLIFIERS

In the center of the console are five of these, each with its own level controls, vu meter and related switching. The output of each is identical to the audio control units and they may be patched in to the 8,2,4, or single-track record heads. By inserting the speaker patch cord (center of console FIGURE 1) they may be heard on any of the eight speakers.

All photographs by Edward J. Smith

Figure 2. The tape-deck console. Each of the three pedestals has a bias-erase generator amplifier, eight f:e t. reproduce preamps, and the components of the passive selsync record section. The author is seated at the right.

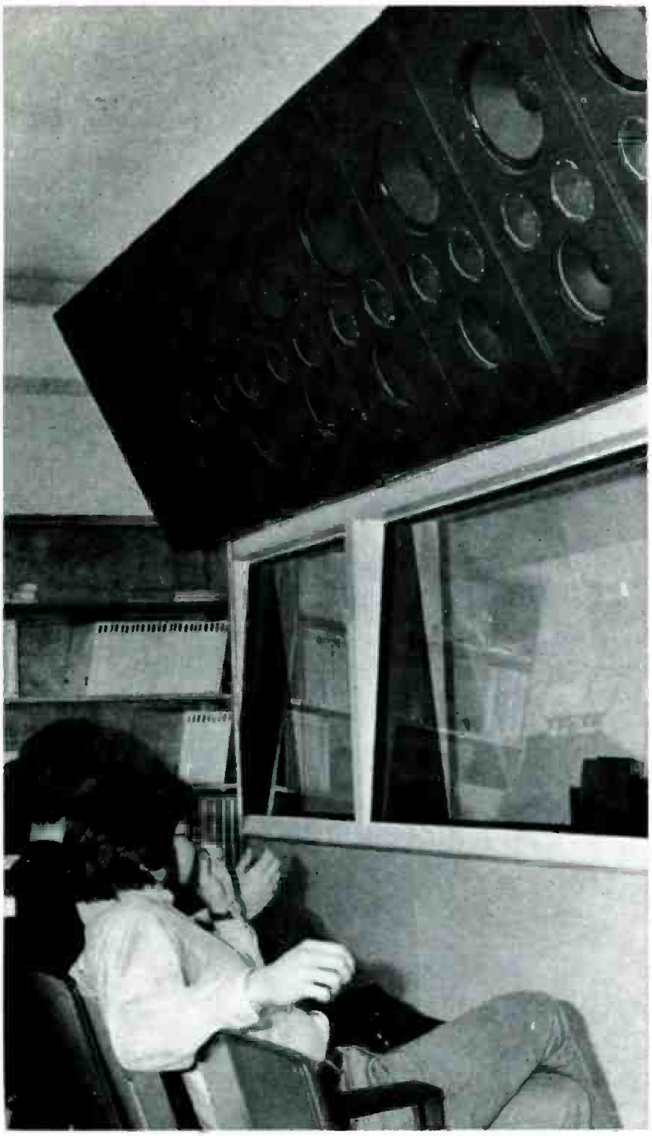


Figure 3. The speaker mounting is directly in front of the console. The photograph was taken as the popular **Orpheus** group (MGM Records) was listening to a playback.

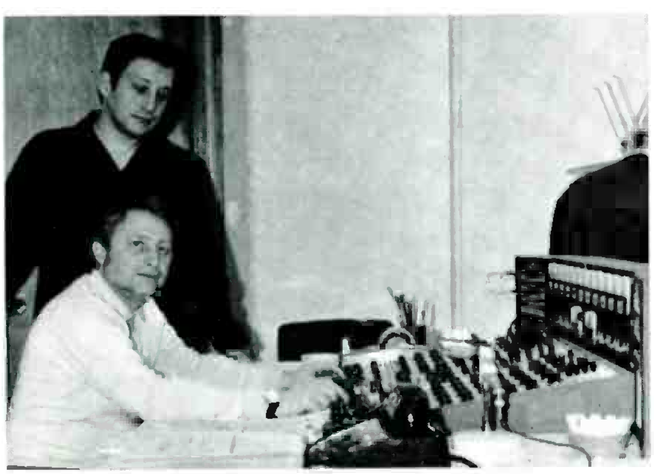


Figure 4. Standing, Alan Lorbor, musical arranger-producer of **Orpheus** and **Ultimate Spinach.** Seated at the console is Eddie Smith, Mayfair Recording's audio engineering producer. Both are listening to an eight-track playback.

The main uses of these five units are:

1. Use them for simultaneously recording and monitoring 8 track, 4 track, 2 track and mono; maintaining independent levels for each.
2. Any combination of channel combining for the proper operation of the 8-track selsync recorder.
3. Return of the live echo chambers (each has its own mic preamp).
4. Five channels of studio or headset feed in any combination for overdubbing or adding to the 8-track tape. No other amplification is necessary.

TAPE DECK CONSOLE

At the present four tape decks are used in each studio; 8 track selsync, 4 track selsync, 2 track, and mono. The four-track head assembly also has a three-track play head installed; the two-track deck also has a mono play head; the mono deck also has a two-track play head. This is a great convenience for the operator and producer (no need for extra tape machines) and saves management the price of three decks doing same job. Each deck is the same and can easily be modified for 8-track or 12-track recording for just the price of the head assemblies.

CONSTRUCTION OF TAPE-DECK CONSOLE

The front supports consist of three Bud pedestals (Figure 2.) In *each* are mounted:

1. Eight-channel fet preamps on a 19-in. by 3½-in. chassis.
2. Bias and erase generator and amplifier with its own power supply.
3. Passive record panel, consisting of relays, resistors, condensers, traps, switches, etc.

Each of the four decks gets its bias and erase from the one generator. This keeps the distortion and hiss at a minimum when it becomes necessary to go from tape generation to generation. If one generator fails there are always two standbys. In addition if the 8-track preamp fails, two identical ones are always available in the four-track and two-track pedestal.

SPEAKER SYSTEMS

The speakers (Figure 3) are cones in a sealed box with a 500-ohm crossover network (our own design). No monitor amplifiers are necessary with this console if you are satisfied to monitor at a comfortable living room level. Unfortunately, the younger performers want to hear it at the threshold of pain, so eight 35-watt transformerless amplifiers are available in the pedestals of the console. Having a speaker as well as a vu meter in each of the eight channels is a convenience greatly praised by both engineers and producers. Any number of speakers with its amplifier can be used in any channel. (One should hear 270 watts of audio power driving the 32 cones in a control room!) As the young performers bring in larger guitar amps (auditorium sound systems) we find it commonplace to parallel two or three speaker-lines to equal their studio volume.

SUMMARY

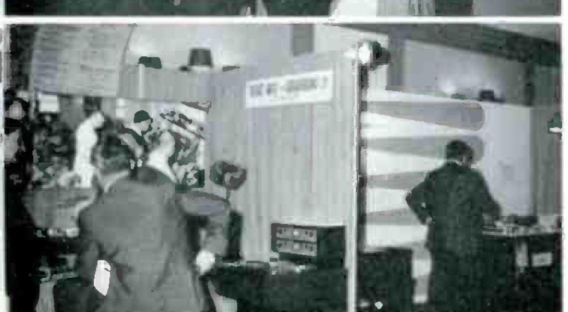
These consoles are actually eight complete mono consoles that will amplify a microphone, tape play head, or phono pickup, providing two separate outputs of ten watts each across 600 ohms—more than enough to record on a tape head, after you have shaped the sound to your taste. The individual audio control units are designed for studio use, mastering channels, tape duplication, film recording or any sound use where maximum gain is required in a small package.

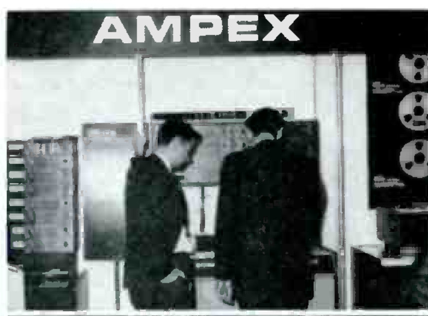
Picture Gallery: West Coast AES Convention

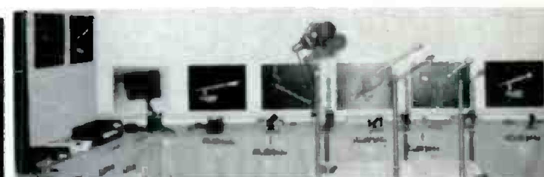
ON this and the following pages are the results of our roving camera's visit to the Thirty-fourth Audio Engineering Society's Convention held in Hollywood, California on April 29th through May 2nd.

Each Convention seems to become better and bigger than the one before. This was the best yet. We must congratulate the west coast group, particularly its tireless president, Donald B. Davis for the fine organization and control that was given to this show.

On this page—a montage of scenes at the exhibition. Following, is a display of the *products* that were shown. Many, of course, premiered at this show.

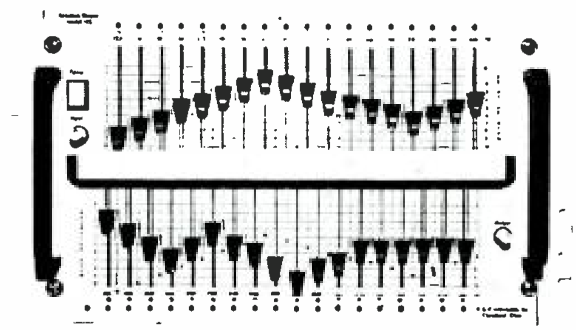






THE following illustrations highlight much of the new material shown at the AES Convention. Each product photo is keyed to the Reader Service Card at the rear of this issue. Circle the appropriate number for further information.

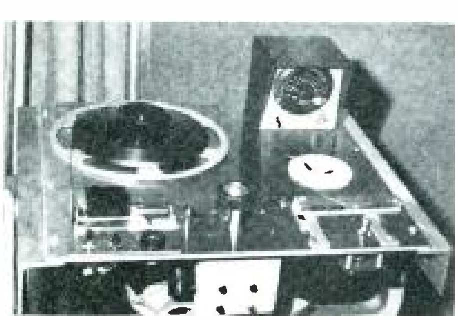
Fairchild Recording Equip. Corp. Console system. Unit is built around individual modules with a +18 dBm output. Full versatility is provided in each module. Price: $525 for a complete module—including built-in compression and equalization. Price goes down as function is removed. Circle 98 on Reader Service Card.



B & K Instruments. Graphic Frequency Response Shaper. Divides the spectrum into 36 one-third octave bands. The vertical attenuators then form a visual configuration of the set-in shape. A detachable memory bar allows a previously established shape to be re-created. Price: $3000. Circle 81 on Reader Service Card.

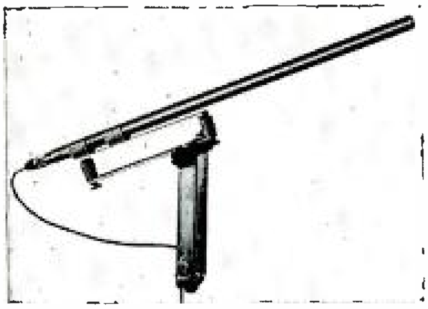
Hewlett-Packard, Inc. 8062A Sound level meter and 8055A octave band filter. Precision sound level meter with condenser mic, has A, B, and C weighting curves. Two versions available, one battery and line, the other a.c. line only operation. Price: 8055A-$520; 8062A-$720 8052A-$670 (a.c. only). Circle 94 on Reader Service Card.

Universal Audio. Response Plotter. Draws frequency response automatically. Generator and receive set can be placed apart by tranmission lines. The units are all-electronic, compact, and relatively low cost. In the illustration—left, sweep generator; right, frequency response meter; below, unit under test. The remote 'scope plots response. Price: $600. Circle 89 on Reader Service Card.



Harvey Radio Company, Inc. Tape Packaging Machine. Takes bulk tape and winds it onto individual hubs or cassettes. An efficient operator can turn out 500 units or more per day. Tape cut is automatic from a cue signal on tape or elsewhere. Price: $1700. Circle 92 on Reader Service Card.

R. T. Bozak Company. Column Speaker System. Very high power levels with broad-band capability and the dispersion characteristics of a column. A 100-watt input will result in 125 dB of spl at four feet. The column will handle up to 200 watts. Price: $695. Circle 87 on Reader Service Card.

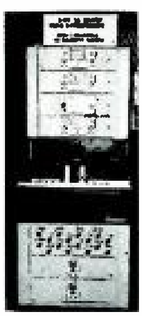


Sennheisser. MTH-804 Ultra directional microphone. Has extremely narrow angle of acceptance. Price: $377. Circle 99 on Reader Service Card.

Ampex Corp. Model 1000 recorder. Eight channels on two-inch tape. Designed as a master unit for duplicating systems. Price: $17,000. Circle 97 on Reader Service Card.

Orban Associates (R.A. Moog). Stereo Synthesizer. Creates a distortion-free stereo synthesis offering maximum fidelity to the mono source. The unit shown is a prototype. Production units will look slightly different. Price: $895. Circle 93 on Reader Service Card.

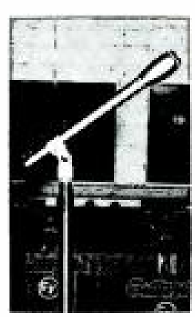
Scully Recording Instruments Co. Model 82 Recorder. An eight-channel reproducer that can be used for mix down of tapes. This is also a two-channel selectable record unit. Any two of the eight may be used. Takes one inch tape. Price: $8850. Circle 84 on Reader Service Card.



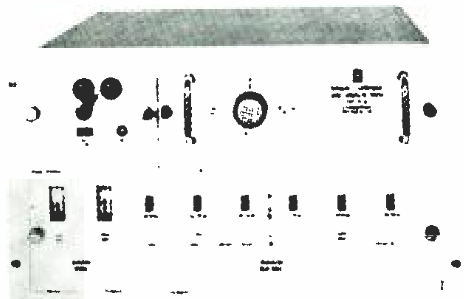
3M Company. 16 tracks on two-inch tape on an NAB reproducer. There are four selective channels of record. Availabile in the fall with the price to be announced at that time. Circle 91 on Reader Service Card.

Harman-Kardon Inc. ProcasT 5600 mixer/preamplifier. Output +18 dBm, less than 0.25 per cent distortion, −125 dBm noise. Price: $310.00. Circle 95 on Reader Service Card.

McMartin Industries. School Sound Console. Modern styling in a complete control console. Price to be announced. Circle 90 on Reader Service Card.

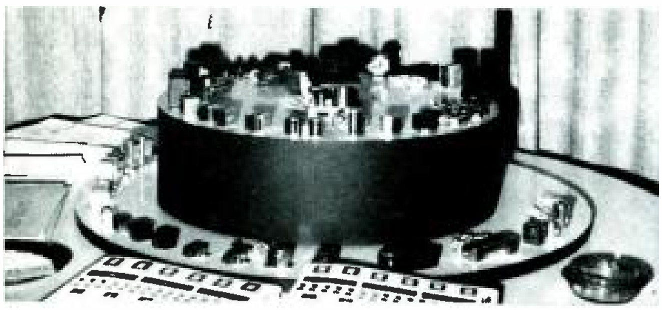


Electro-Voice, Inc. RE-55 microphone. Steel cased, with a 150Ω voice coil. Tranformerless. Extremely smooth response over the range of 40-20,000 Hz. Price: $120. Circle 96 on Reader Service Card.

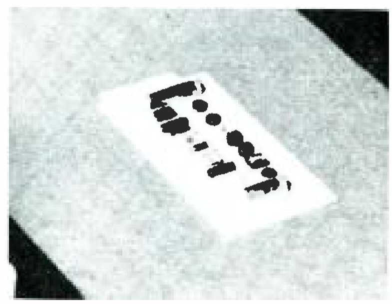


Gotham Audio Corp. EMT 970—A delay device employing a two-speed magnetic disc. Price: $2190 mono and $2590 stereo. Circle 100 on Reader Service Card.

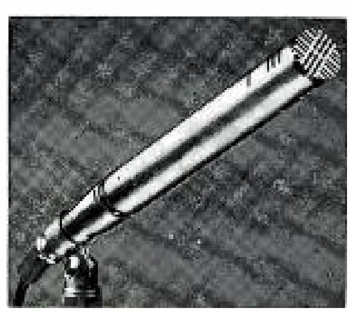
Shure Brothers, Inc. M67 Mixer. This is a four-channel mic mixer and remote amp. May be used for remote broadcasting or as an add-on mixer. Line output is at +18 dBm, mic imputs are low impedance, and the unit is fully transistorized. Distortion is less than 0.5 per cent harmonic at +20 dBm 20-20,000 Hz. Noise is at least 60 dB below a +6 dBm output. Price: $147. Circle 85 on Reader Service Card.



Lipps. A complete line of professional-grade heads for both replacement and OEM use. Price: dependent on the head. Circle 88 on Reader Service Card.

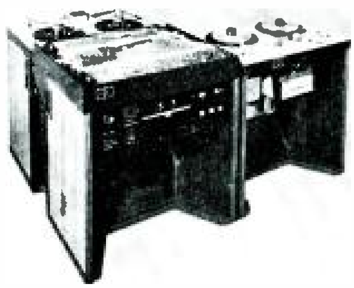


Spectra-Sonics. Isolation module/amplifier. A tranformerless isolation system that prevents ground loops. Operates, in effect, as an isolation tranformer with gain. Eliminates the problems of conventional transformers. The unit is mounted on a plug-in card. Price: $69.00. Circle 86 on Reader Service Card.



Norelco. D224 Microphone. A two-way cardioid dynamic mic. Two elements are combined with an electrical crossover to provide a 20-20,000 Hz response. At 90° off axis, response is still flat. Price: $185. Circle 80 on Reader Service Card.

Dolby Laboratories. Remote Changeover System. Enables one A301 unit to be used on recording and playback with two channels. Previously, convenience dictated two A301s, one in stereo record, the other in stereo play. The recorder can have remote control of all switching, so that the unit will be in record or reproduce in conformance with the recorder. Price: $445. Circle 82 on Reader Service Card.



Gauss Electrophysics. High Speed Duplication System. Three pieces are involved in a minimum system: the master recorder, slave unit, and the tape loop bin. With the bin, an endless loop is used on the master at 32 times speed, reproducing a 1200 half-hour program in less than one minute. The bin itself can be used with Gauss and other master units. Price: for complete basic system under $30,000. Circle 83 on Reader Service Card.

Sound with Images

MARTIN DICKSTEIN

CCTV LENSES

● In closed-circuit t.v., as in photography and in projection systems, one of the vital elements of the system is the lens. Although many people are aware of the functions of a lens and its various characteristics, we might do well to review some important facts for those who are not familiar or who may have become rusty.

FOCAL LENGTH

The lens is used, of course, to focus a scene on the surface of the camera pickup tube just as it does in a photo camera with film. The distance between the optical center of the lens and the point behind the lens at which the image is focused is called the focal length of the lens. Thus, if a scene at infinity is in focus at a point 1 inch behind the lens, the focal length is 1 inch. This is considered to be the *standard* lens and is sometimes furnished with the cctv camera.

Knowledge of the relationship of the focal length to the width of coverage and the relative magnification is important in determining the lens to use for a scene at any given distance from the camera. A 1-inch (25mm) lens will cover a scene half as wide as the scene is distant from the camera. This would mean that at 10 feet from the camera, a scene 5 feet wide would be covered by a 1-inch lens. At 20 feet, the scene coverage, in width, would be 10 feet.

A ½-inch lens (wide-angle lens) would then cover a scene twice as wide as a 1-inch lens at the same distance from the camera. Conversely, a 2-inch lens (50mm) would pick up a scene 2.5-feet wide at a 10-foot distance. A simple relationship to help remember this would be to consider that a ½-inch lens picks up a scene as wide as it is far; work all other focal lengths from that. Thus, a longer focal length lens (telephoto) will have a smaller angle of coverage but the image will appear correspondingly larger—the image of a 50mm lens will be twice as large as that of a 25mm lens.

The actual formula for the required computations is: focal length (in inches) divided by scanned-area width (on the vidicon) is equal to distance of subject (in feet) divided by width of field (feet). The standard cctv vidicon has a scanned area width of ½-inch. Substituting this into the formula makes it:

$$\frac{\text{F.L. (inches)}}{\text{0.5 (inches)}} = \frac{\text{Distance (feet)}}{\text{Width of field (feet)}}.$$

In order to obtain the height of the viewing area, the formula to remember is that:

$$\text{Height (feet)} = \tfrac{3}{4}\ \text{Width}.$$

The height of the vidicon scanned area is 3/8-inch to conform to this relationship.

SPEED

The second characteristic of lenses that is important for images on the monitor is *speed designation.* The normal lens has a speed designation of *f1.9.* This factor closely resembles the pickup of the human eye, so that, assuming the camera and monitor to be working perfectly, the lens will see approximately what the eye will.

If it is desired to pick up a dimly lit area, the lens used should be faster to permit more light through. This would mean using an f/1.4 or f/.95 lens. The lower the designation, the faster the lens and the more light it will pass. In order to determine the *f* number of a lens, it must be remembered that *f* is related to the focal length of the lens and to the diaphragm opening. The diaphragm of a lens is similar to the iris of the eye. When the eye sees a brighter light, the iris closes to protect the eye. In lenses, the diaphragm is either adjusted manually or operated electrically by an automatic iris control. To arrive at the *f* number of the lens, therefore, the formula is:

$$f\ \text{(lens speed)} = \frac{\text{Focal Length (inches)}}{\text{Lens Diameter (diaphragm wide open) (in inches)}}$$

A 1-inch lens, then, with a diaphragm opening of 0.526 inch would have a speed of *f/1.9*. As the diaphragm diameter is decreased, the *f* rating of the lens increases and the slower the lens.

An interesting relation exists between the *f* ratings of lenses and the amount of light they will pass through, a relationship that is well known to sound men. . . .the *inverse-square law.* For example, a lens with an *f* number of *2* will pass one fourth the amount of light as a lens with an *f* of *1*.

DEPTH-OF-FIELD

This term, depth-of-field, is a description of the distance of a scene that is in apparent focus measured from the nearest point in focus to the farthest point. The depth of field is related to the focal length of the lens (the shorter the focal length the greater the depth of field), the *f* stop of the lens (the higher the *f*, the greater the depth of field) and the distance of the subject from the lens (the greater the distance, the greater the depth of field).

ZOOM LENSES

In discussing lenses, it is normal to talk about a lens with a particular focal length. However, there are available lenses that permit changing the focal length by simply turning a ring on the lens. This is a zoom lens. The focal length can be adjusted manually with a lever or by remote control with a motor drive connected to the ring. (Focus and diaphragm are also adjustable on the zoom lens, either manually or by the remote control of a motor drive.) Zoom lenses are made in different ratios, the most common being 4 to 1 (25mm to 100mm) or 10 to 1 (15mm to 150mm).

MAGNIFICATION

One more relationship between object width and focal length can be shown by taking into account the magnification of the lens. Knowing that the width of the vidicon scan area is ½ inch and assuming that the object width is 5 feet (60 inches), the magnification would be

$$\frac{\tfrac{1}{2}\ \text{in.}}{60\ \text{in.}} = 0.008.$$

Using the formula

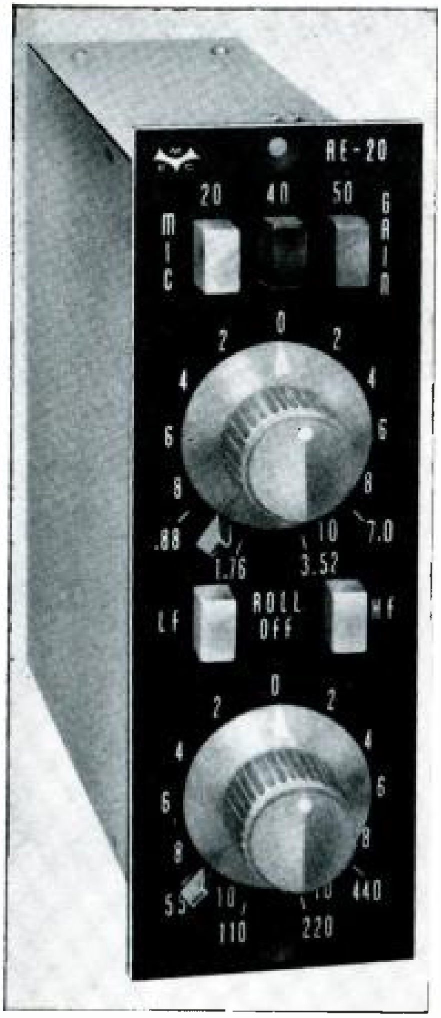
$$\text{F.L.} = \frac{\text{Object Distance (Inches)} \times \text{Magn.}}{(\text{Magn.} + 1)^2}$$

we find that for an object at 30 feet from the camera with a width of 5 feet, the focal length of the lens to be used is 2.88 inches. The actual lens to be used then would be a 3-inch lens.

Although the terminology and values of lenses used in cctv might be the same as that used in photo making and even though the mounting threads on a standard cctv camera are the same as that in a 16mm movie camera, the two lenses are not interchangeable. The photo lens should not be used on a cctv camera unless the interchange is tried first.

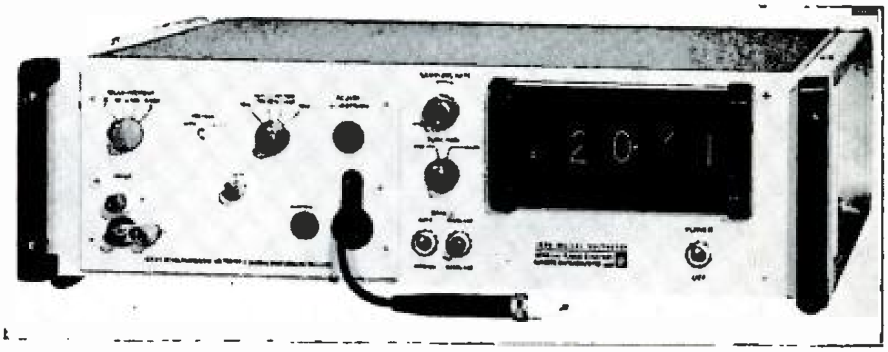
New Products and Services

AMPLIFIER EQUALIZER



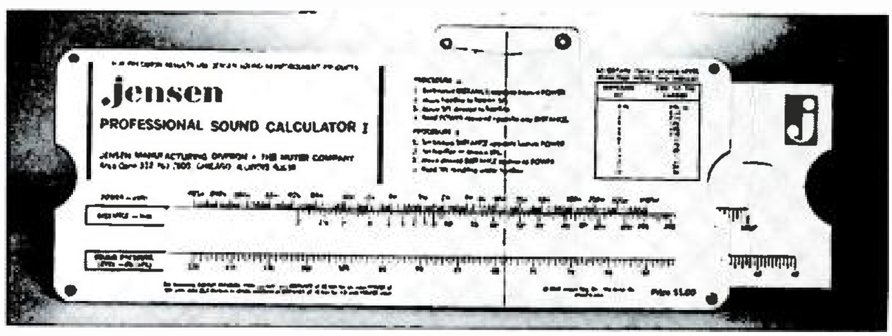
● Microphone preamplifier and amplifier equalizer are combined in this compact unit. Known as the AE-20, the unit is designed for individual mic channel use. The preamp portion is an extremely low-noise transformer input stage, with variable gain switching of 20, 40, and 50 dB. The equalizer portion is an active, lossless unit which uses a bridged T-notch filter switched from feedback loop to input for boosting or dipping at the selected frequency. A total of four high- and four low-frequency selectable equalization points, with up to 10 dB boost or attenuation in 2 dB steps, are available.
Mfgr: Melcor Electronics Corp.
Price: $346.00
Circle 52 on Reader Service Card

DIGITAL VOLTMETER

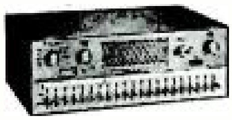


● The use of two plug-ins and one adapter make this new model 1820 dvm a highly versatile general-purpose lab voltmeter. Voltage can be presented either in dB (re 100μV) or volts; range is selected automatically. The decimal point is positioned and the proper measurement units are selected and displayed automatically as well. The unit may be used as a d.c. multimeter/uhf voltmeter as it reads a.c. (peak reading, calibrated in rms) and d.c. voltage and current, and resistance to $\pm$0.1 per cent accuracy in most cases. Useful high-frequency measurements may be made to 1.5 gigahertz; high input impedance reduces the effects of loading. As an a.c./d.c. millivoltmeter, the 1820 measures both a.c. (average) and d.c. with microvolt sensitivity. Direct current can be measured with picoampere resolution and, with a standard 'scope probe, a.c. can be measured with nanoampere resolution. With an adaptor added to either plug-in, the 1820 converts to a fully-balanced differential voltmeter with better than 100-dB common-mode rejection.
Mfgr: General Radio Co.
Price: $1985.00. The appropriate plug-in is an additional *$525.00 or $550.00.*
Circle 57 on Reader Service Card

SOUND CALCULATOR

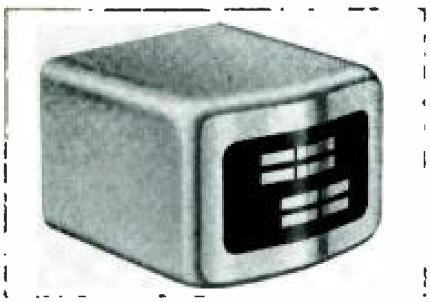


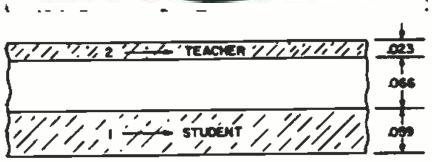
● A truly useful double-sided 7½-inch slide rule has been designed for quick and easy computation of all problems normally encountered in designing loudspeaker system installations. Fixed and movable slider scales, along with a transparent sliding hairline indicator, provide the necessary means for converting sound pressure in dynes/sq. cm. to sound pressure level (spl in dB) and for computing spl or distance or power input to the loudspeaker, for any known or specified condition such as loudspeaker sensitivity, required spl, etc. A "Range Extender" table shows spl change for large ratios of distance and power. An spl summation table shows total spl for two known signal levels. Detailed procedures on front and back of the slide rule describe typical computations. Precision is more than adequate for the practical problems encountered. The rule is shipped complete with a tech sheet describing its use. Cost is one dollar, postpaid. Write to: Jensen Manufacturing Division, The Muter Company, 5655 West 73rd Street, Chicago, Illinois 60638. Ask for the Jensen Professional Sound Calculator I.



●The model SX390 offers full dual-channel operation, separate program and intercom amplifiers, all-silicon transistors and diodes, three low-impedance mic inputs, three high-impedance aux inputs. There is a total of twenty station selector keys (expandable in multiples of twenty-one—up to sixty-two) and an all-call switch. The individual talk-listen switches have been designed to provide over 1,500,000 trouble-free operations. Separate accessories that base mount under the unit include an a.m./f.m. tuner, 4-speed record changer, and several system expander units.
Mfgr: Rauland-Borg Corp.
Circle 51 on Reader Service Card

CASSETTE TAPE HEADS

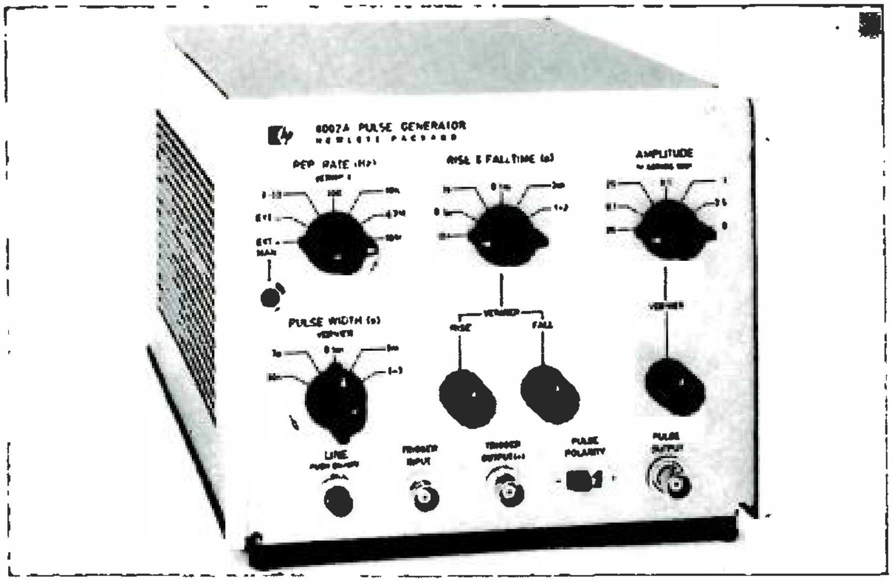




CASSETTE LANGUAGE LAB TRACK DIM. (inches)

●This new model ZW4J head is bi-directional, and performs record, play, and erase functions in both directions. Machines equipped with the new head will be able, for the first time, to play all information in a cassette without the need to flip it over. (On conventionally recorded cassettes the information on the second set is directionally reversed from the first set—requiring a reverse-play machine.) The new heads are fully compatible with the Norelco/Philips system. The heads have usable frequency response of 50-12,000 Hz at the normal cassette speed of 1⅞ inch. Inductance is up to 300 millihenry. Heads will be available for entertainment, language lab, and instrumentation applications. There is a total of sixteen different models proposed, with three in current production. The one for language lab application places one track in position for standard play on Philips-type mono machines; the second track, for the teacher, is located at the far edge of the tape (as in the line illustration).
Mfgr: Nortronics Company, Inc.
Circle 54 on Reader Service Card

PULSE GENERATOR



●This new variable rise- and fall-time pulse generator, model 8002A, is said to perform over a wider range of rates and functions than other instruments comparably priced. The unit offers repetition rates from 0 to 10 mHz, when controlled by external triggers, or from 0.3 Hz to 10 mHz, with the internal rate generator. A vernier control sets the repetition rate anywhere within a 30:1 range on any of five ranges. This pulse generator can drive both low-frequency and fast circuits. The pulse width is adjustable between 30 nanoseconds and 3 seconds in five 30:1 width ranges. Square waves as well as impulses and other functions can be generated by this instrument at any repetition rate. It offers rise and fall times variable from 10 nanoseconds to 2 seconds, a range compatible with the wide range of pulse widths and repetition rates. Pulses can be shaped appropriately to match the performance of the circuit under test. Pulse rise and fall rates are linear within 4 per cent, pulse amplitude between 0.02 and 5 volts into a 50Ω load—positive-going or negative-going is available on a front-panel switch. An internal switch removes the resistor, allowing 10 volts into 50Ω from a driving impedance of 300Ω. The output is always short-circuit protected. The source impedance of 50Ω remains constant with the resistor switched in even during rise and fall times. The generator also supplies a 2-volt trigger pulse 180 nanoseconds in advance of the main pulse. A built-in delay line can be switch by-passed to provide a 35-nanosecond advance.

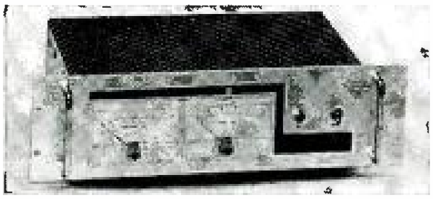
Mfgr: Hewlett-Packard
Price: $700.00
Circle 56 on Reader Service Card

MONITOR AMPLIFIER

●The model 1035 is a rack-mounted, high-quality monitor amplifier designed for input levels normally found on jack fields and telephone lines. It is completely self-contained, with its own power supply and speaker, yet only occupies 3½ inches of vertical space in a standard 19-inch rack. Important specifications include: Input impedance —150Ω, 600Ω, and 20 kΩ balanced; input level— −20 to +30 dBm; output capability—5 watts. The unit uses dual or single jack inputs, operates off standard 117V 50 or 60 Hz a.c., has on/off and gain controls, and is of all-silicon solid-state circuitry.

Mfgr: Comrex Corporation
Price: $135.00
Circle 55 on Reader Service Card

POWER SUPPLY



● The ADM PS-10 is a totally silicon-transistorized 24 volt, 10 ampere power supply. Regulation (line and load combined) is on the order of 0.0025 volts, with a noise figure of 50 μV. Circuit protection prevents damage to the unit in the event of a short circuit. Voltage and current are metered, the d.c. voltage is presented with either a positive or negative ground, and the a.c. input may be either 105-125 volts or 210-250 volts 50/60 Hz. The unit takes up 5¼ inches of rack space.
Mfgr: Audio Designs and Mfg. Inc.
Price: $395.00
Circle 53 on Reader Service Card

DRIVER SPEAKER

● This new version of the long-popular 288 driver loudspeaker carries the designation model 288D. It is suitable for use in theaters, auditoria, factories, gyms, and the like—anywhere high-level music, speech reinforcement, and paging systems are required. The driver may be used with any one of the Altec multicellular horns and will provide smooth response from 500 to 16,000 Hz. Impedance is 16Ω.
Mfgr: Altec Lansing
Circle 50 on Reader Service Card

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by Norman H. Crowhurst. 1967. This book gives all the basics needed to become a successful PA operator, in any situation where the reinforcement, relay, or distribution of sound can provide a service. It shows how to properly install, operate and service public address systems. All aspects of the subject, from survey to the selection of appropriate equipment, to installation, to routine operation and the maintenance of a finished system, are covered. Attention is given to solving problems encountered in providing successful service. The book's systematic and practical approach makes it highly useful to radio-TV servicemen, hobbyists, and PA equipment manufacturers. 136 pages; 6 x 9; illus; softbound.

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CLOSED CIRCUIT TV SYSTEM PLANNING

by M. A. Mayers and R. D Chipp. 1957 This book discusses in detail the vitally important and rapidly expanding concept of closed circuit TV systems, its utility and functioning. This book is not an engineering or a technician's text — it is written for management. It explains and illustrates the kind of systems that are available and their applications. 264 pages; 8½ x 11; illus; clothbound.

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by Alec Nisbett. This is a handbook on radio and recording techniques, but the principles described are equally applicable to film and television sound. It describes how the highest standards may be achieved not only in the elaborately equipped studio but also with simple equipment out on location. 264 pages; 60 diagrams; glossary; indexed; 5½ x 8½; clothbound.

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by A. E. Robertson. 1963. This book, primarily written as a training manual for technicians, will also prove valuable to all users of quality microphones whether for broadcasting, public address systems, or recording of all types. There is now an almost bewildering array of microphones of differing characteristics, and new designs are constantly being produced. The author makes no attempt to catalogue these but concentrates mainly on the principles of operation. He only describes actual microphones if they illustrate an important feature or have some historic significance. The book is intended for the user rather than the designer; mathematics have been omitted from the main body of the text. 359 pages; 6 x 9; illus.; clothbound.

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by *William A. Rheinfelder*. **1964.** Written for students as well as circuit design engineers interested in low-noise circuit design. Throughout, the book gives a multitude of time-saving graphs and design curves for the practical circuit designer. Simple derivations of all important formulas are also presented to help the reader obtain a deeper insight into the fundamentals of practical low-noise design. 128 pages; 6 x 9; illus.; clothbound.
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WANTED — REPRESENTATIVES. Active, qualified representatives to cover advanced audio line. Submit organization resume, territory normally covered and other lines handled. **Fairchild Recording Equipment, 10-40 45th Avenue, Long Island City 11101, New York**

Salesman to call on RECORDING STUDIOS to sell professional audio equipment in Metropolitan N.Y. area. Experience preferred. Send resume to Box E6, **db** Magazine, 980 Old Country Road, Plainview, N.Y. 11803.

REQUEST FOR PERSONNEL: Systems Engineer—aggressive young man. Systems sales oriented, construction or industrial experience helpful, but not essential. All inquiries answered. **Engineered Sound, Incorporated, 1900 North 25th Drive, Phoenix, Arizona 85006**

People, Places, Happenings

● **Brigham Young University,** in Provo, Utah will hold its second annual Recording Seminar on August 12 to 16. There will be lectures and demos by leaders in recording, music, and equipment. Last year's seminar was a huge success; this year promises to be even better. For full information write to **Mr. Kaye Jensen, Recording Seminar, Box 41, HFAC, Brigham Young University, Provo, Utah 84601.** And they suggest that it might also be a proper time for a Rocky Mountain vacation too!

● **George W. Tillett** has been appointed executive vice-president of **Audio Dynamics Corp.** according to an announcement made recently by **Peter E. Pritchard,** president. Mr. Tillett joins Audio Dynamics directly from the **Fisher Radio Corp.,** where he was director of engineering at their Pennsylvania plant. He was formerly technical director of **Wharfedale** and also chief engineer of **Heathkit's** British facility.

● **Ampex Corporation** has announced record sales for the fiscal year ending April 27th, although earnings were substantially below the year earlier. According to **William E. Roberts,** Ampex president and chief executive officer, a strike of machinists and production workers at the company's Redwood City, California plant prevented Ampex from continuing the sales and earnings growth trend of the past six years. He predicted a full recovery of the growth pattern in both sales and earnings in the current year's first quarter and fiscal year. He went on to predict a minimum compounded growth rate of 15 per cent per year for the next five years.

● Faster and better customer service is claimed for the new facility opened by **Perma-Power** in Long Beach, California. Several of their manufacturing facilities will be housed in this single building. **Ampli-Vox** cordless p.a. systems, t.v. service accessories and the company's garage door opener group will all have their products manufactured here. According to **Richard Goldstein,** the firm's president, the 6300 square foot facility began full operation on May 1 of this year.

● **Eli Passin** has been appointed national sales manager of **Gotham Audio Corp.** according to **Stephen F. Temmer,** president of the company. Mr. Passin comes to Gotham from **Harvey Radio** where he was sales manager of the professional audio/video division. He has been associated with professional audio since 1950, when he was appointed program director of radio station **WJRH** at Lafayette College. After graduate school and the U.S. Army, he joined Harvey, beginning as an assistant manager of their high-fidelity department.

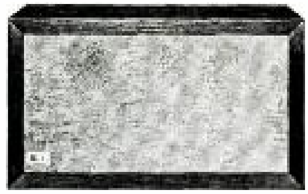
● **Lee D. Webster** has been named president and chief executive officer of **LTV Ling Altec, Inc.,** subsidiary of **Ling-Temco-Vought. Alvis A. Ward,** who has held that post, has been advanced to chairman. Mr. Webster's new assignment has been made concurrent with the acquisition of **Escon, Inc.,** of Dallas, a company which he served as president and chief executive officer. Escon is engaged in a variety of electronics and plastics production and marketing organizations.

● A management realignment at **The Muter Company** will permit more intensive planning for future growth. **Herbert T. Rowe** continues as president and chairman of the board of The Muter Company. He will devote much of this time to policy and planning activities of Muter. **Herbert F. Kuhlow** continues as executive vice-president of Muter, all divisions and subsidiaries of the company will report and be responsible to him. He also assumes responsibility for the operations of the **Jensen Manufacturing Division** in the capacity of vice president and general manager. He will make his headquarters at Muter's Chicago offices.

● **Rupert F. Goodspeed** has been appointed general sales manager, **Philips Broadcast Equipment Corp.,** Paramus, New Jersey, according to **Anthony R. Pignoni,** director of marketing. Mr. Goodspeed came to Philips Broadcast early in 1967, assuming the duties of broadcast product manager. In his new position he will head the national sales organization for the company's various product lines, including broadcast television, c.c.t.v., audio, and motion-picture projection systems. The company's marketing structure is comprised of a network of regional sales and service offices and numerous dealers and distributors throughout the U.S.

● **Superscope Inc.** exclusive distributor of **Sony** tape recorders, magnetic tape, microphones and accessories, set new first quarter sales and net income records for the three months ending March 31, 1968, according to **Joseph S. Tushinsky,** president. Net income rose to 21.5c a share as compared against 18.5c for the same period (adjusted for the same number of shares) last year. "This has been the best first quarter in the history of the company," according to Mr. Tuchinsky. He has attributed the increases to the expanding market for tape recording equipment.

● **Joseph Tritsch** has been recently appointed as vice-president and general manager of **Engineered Sound, Inc.** of Phoenix, Arizona. Engineered Sound, Inc. is an organization devoted to systems design, construction, c.c.t.v., m.a.t.v., high-frequency audio systems, and fire and security systems. In this new position, Mr. Tritsch will supervise the sales, electrical design, and installations of this newly formed group.



AR-3a

Acoustic Research announces a new speaker system.

In 1959, our first advertisement for the AR-3 stated, "it has the most musically natural sound that we were able to create in a speaker, without compromise." This judgment was supported by distinguished writers in both the musical and engineering fields. Hirsch-Houck Laboratories, for example, agreed that "the sounds produced by this speaker are probably more true to the original program than those of any other commercially manufactured speaker system we have heard." For nearly nine years the AR-3 has been the best speaker we could make.

However, technical development at Acoustic Research, as at many companies in the high fidelity industry, is a never-ending search for improvement. After much effort we have found a way to better the performance of the AR-3. The new speaker system, the AR-3a, has even less distortion, more uniform dispersion of sound and still greater power handling capability. The improvement can be heard readily by most listeners; it has been brought about by the use of newly designed mid-range and high-frequency units, and a new crossover network. Only the woofer and the cabinet of the AR-3 are retained in the new system. The AR-3a is priced from $225 to $250, depending on cabinet finish, and is covered by AR's standard five-year speaker guarantee.

Detailed information on conversion of an AR-3 to an AR-3a is available from

ACOUSTIC RESEARCH, INC., 24 Thorndike St., Cambridge, Mass. 02141