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NOVEMBER 1972 $1.00

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COMING NEXT MONTH

• David Robinson, the noted British engineer, appears in our pages for the first time with a construction article on A Stereo Phase Indicator. He describes the circuitry required to create the proper integration of signals and thus prevent random meter waving. The unit is easy to build and a very useful instrument.

Each summer, the Stratford Festival in Stratford, Ontario creates Shakespearean productions in the classical manner. An intricate sound distribution system has been created for the theater by its Technical Director, Robert Scales. In this article, he details the uses of multichannel techniques in the quest of the classical theater.

Part 2 of Marshall King's provocative essay on hearing and listening as practiced by the audio professional. A stimulating discussion of the thing, we take for granted.

db Visits—SCULLY. Our camera poked its lens into Scully's shiny, new plant in Northern California and saw their new faciilties for making multichannel tape recorders. If you knew the old plant, you will note quite a difference.

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THE SOUND ENGINEERING MAGAZINE

NOVEMBER 1972 VOLUME 6, NUMBER 10

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George Alexandrovich

THE AUDIO ENGINEER'S HANDBOOK

More on the Subject of Pads

• A couple of months ago you may have read my column about resistive pads. When you read it it was already a couple of months old because publishers need time to put an issue together and print it. Reaction from readers usually comes month or so after the issue appears. If I wish to reply or print your ideas another two months go by before you can read it. With these delays understood, I'd like to share with you the efforts of audio engineer John W. Wood from Valdosta, Georgia who sent me an interesting letter about how he figures pads. In his letter he expressed the hope that other readers may benefit from considering his method. I thought that it was a fine idea since a reason for having a magazine is to share with others your experiences and your knowledge, as well as learning from the others.

John's method consists of using a graph which he precalculated and plotted for symmetrical T or H pads. *Symmetrical* means that source and load impedances are equal. This happens most of the times in balanced lines where transformers are used. And the place where transformers are found is on the inputs and outputs of systems as well as in the patch bays. Since all inputs and outputs often appear on the jack bay we can relate this method to equalizing levels in the patch system so that all points of the system selected for easy access have the same levels.

Let's look at the method of using the graph. Look at FIGURE 1 which shows the pads (A) and (B). (A) is an unbalanced pad and (B) is balanced. Note that the difference between the two is only in splitting the R1 resistor into two legs of the balanced H pad. In calculating a balanced pad one should first find the value for R1 for an unbalanced pad and then divide it into two. One important condition for using this method is to have Rs equal RL.

Next we have a graph which was calculated for a 600-ohm circuit, but using arithmetic you can convert it to any other impedance. FIGURE 2 shows the graph. One coordinate (Y) is calibrated in decibels of loss or attenuation. The other (X) is representing the values of common-point impedance. Common-point impedance is a resistance measured in the pad shown in FIGURE 1 (A) across the resistor R2. This reading of common-point resistance combines not only the resistance of R2 but also the effects of all other impedances such as

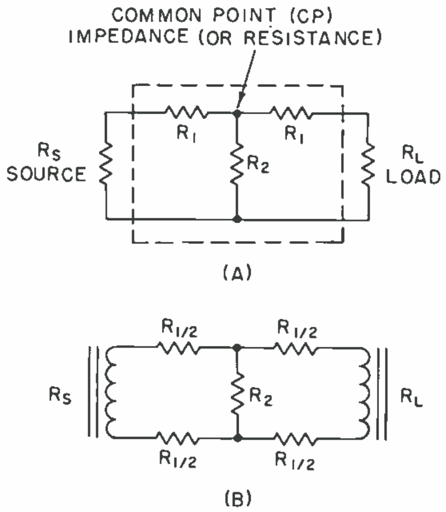

Figure 1. At (A) an unbalanced pad; at (B) balanced.

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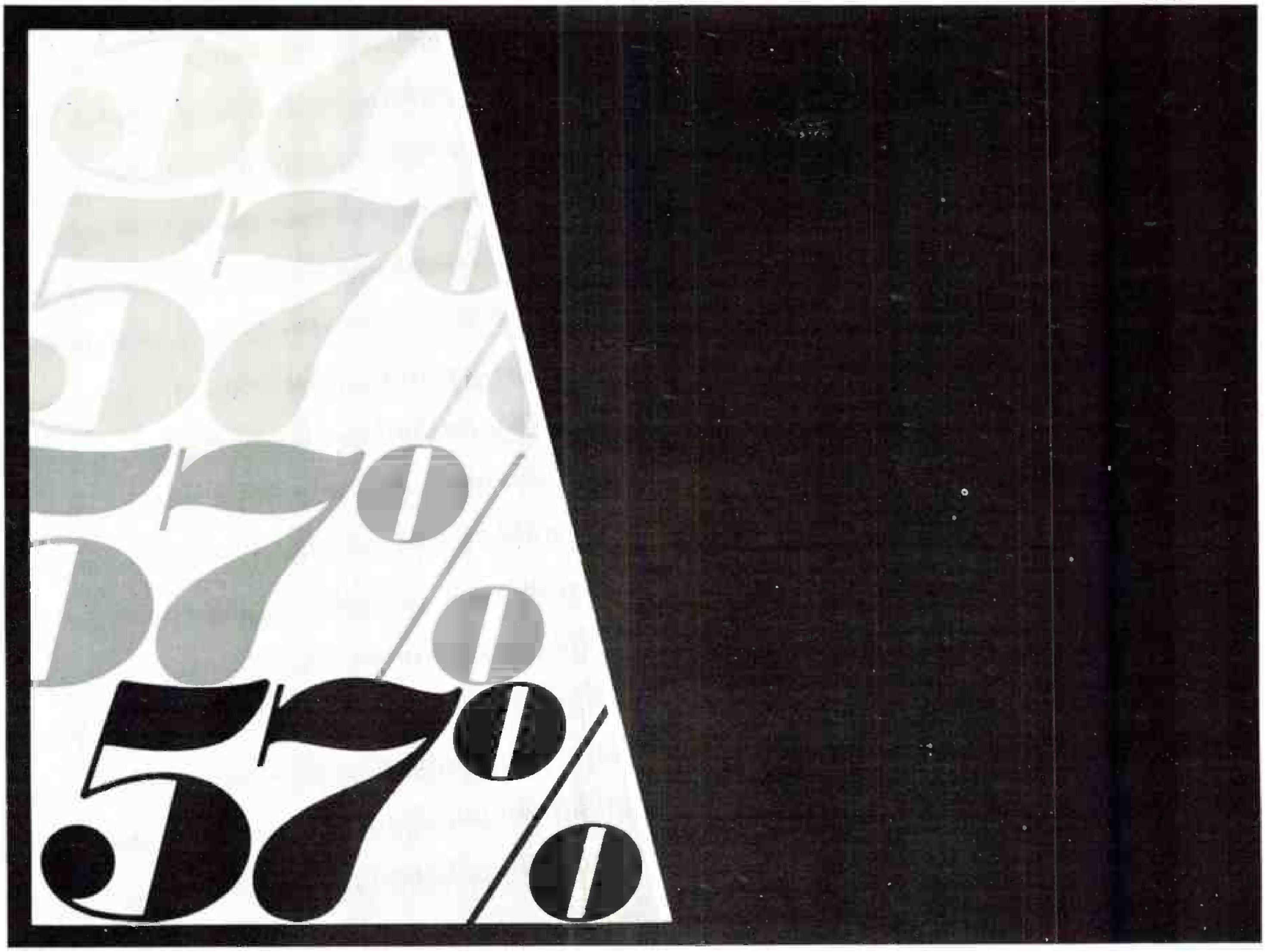
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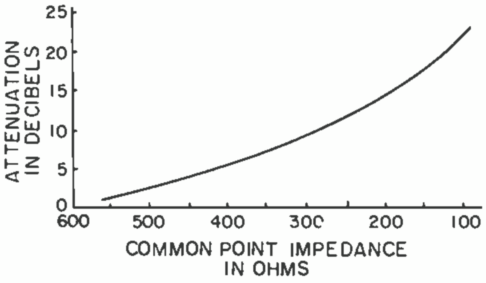

Figure 2. The graph for determining the common-point impedance.

source and load impedances and series resistors.

If we are designing a pad the first thing we want to know is how many dB of attenuation we require. The graph was calculated for a maximum of 20 dB but for all practical applications this may suffice.

From the required attenuation in dB we find on the graph the value of common-point impedance. This becomes *step 1* in finding the values of pad resistors.

Step 2 consists of finding the value of R1 which is:

$$R_1 = R_s - R_{cp} = R_L - R_{cp}$$

In *Step 3* you calculate the value of R2:

$$R_2 = \frac{R_{cp} \; (R_1 + R_L}{R_1 + R_L - R_{cp}}$$

Since to begin with you knew what the attenuation will be and you have calculated the values of R1 and R2, a pad can now be built. If the pad has to be an H pad divide the value of R1 by two and you have all the resistor values you ought to know for a balanced pad. If the pad you design has to be a 150 ohms pad then the calculation should be as follows:

After finding the value of Rcp (common-point resistance) for 600 ohms divide the Rcp by 4 (because impedance ratio between the 600 ohms circuit and 150 ohm circuit is 4:1). By the same token if design impedance is 50 ohms then Rcp value should be divided by 12 using same reasoning. All other calculations should be done in the same way using the new Rcp value. (If the design impedance is, let us say 10 k ohms, then you should multiply Rcp by 16.6—and so on.

Unfortunately this method is not so accurate if you decide to put such a pad after a low output impedance amplifier feeding a 600 ohm line. Most modern operational amplifiers are class AB with plenty of negative feedback and with the output impedance almost zero. As a matter of fact it is quite easy to make a negative output impedance amplifier so that when you connect the load, the voltage across the output terminals goes up. This is accomplished simply by the p.c. layout or otherwords how and where you connect the feedback conductor. But this is a subject for another time. In the mean time you have to use other methods to calculate the pad.

Unfortunately, the graph was not extended beyond the 20 dB mark—but nothing prevents the use of two pads connected in series. Resistors are cheap, besides quite often one may wish to change the attenuation by switching one pad in and out.

What I have described is one of the methods as it was sent to us by John Wood. Many of us use different types of handbooks and we find that methods of deriving pad information differ from book to book. Some books have short cuts—some have different sets of curves and plots or nomographs—some have special formulae—and some just have tables. Find the method you feel you are most comfortable with and use it.

I wish to thank all of you who have helped me to make my columns more interesting and meaningful. In the future anytime I receive material which could be used for this column I will gladly do so with full credit extended to the originator of the idea or material. ■

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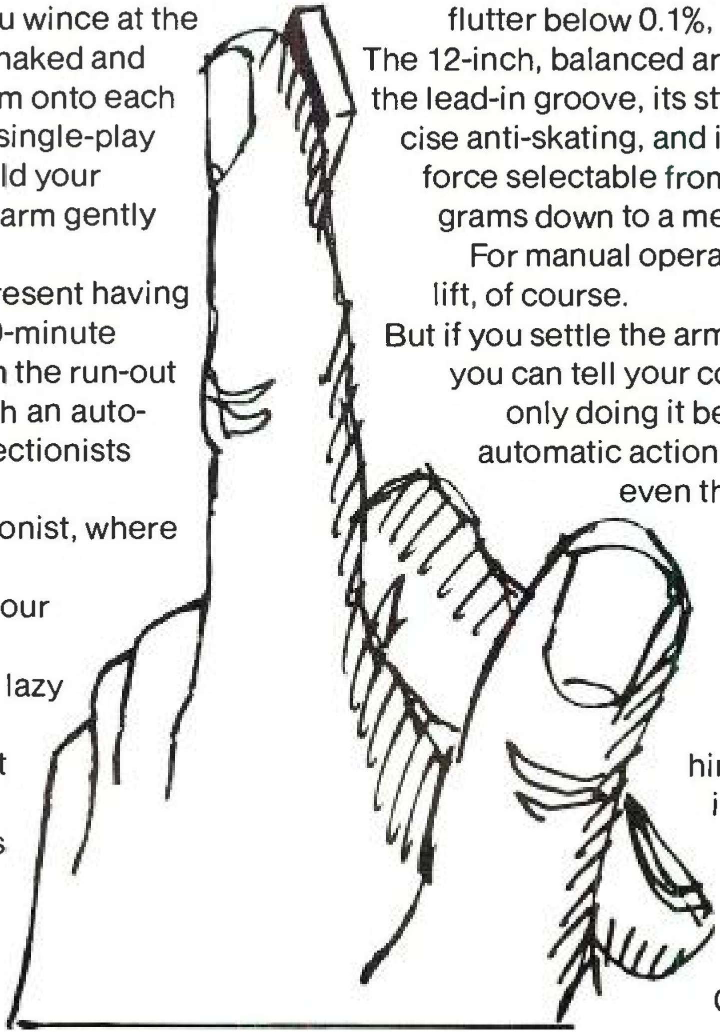

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Norman H. Crowhurst

THEORY AND PRACTICE

● The response to this column in the July issue was quite encouraging. There was also some response to my article in the same issue, about the amplifier power spiral. One of the latter responses was from a professor who has written to me before. He commended me on my treatment and then good-naturedly took me to task for a couple of "errors," that any of his sophomores would easily spot.

These errors, as you will probably guess, were in my usage of words. I referred to the curve that represents constant watts dissipation as an *exponential* curve. And I referred to the voltage at which the collector-base junction breaks down as being a *sort of zener* voltage. The good professor told me the curve is not exponential, but hyperbolic, and that the voltage is not zener, but avalanche.

Of course, using definitions the way they are applied in his school and dozen others across the country, I am sure he is correct. But it is an interesting point—or couple of points—if you give it a little thought, instead of merely reacting on the basis of preconceived definitions. And it links quite closely with one of the problems about education, that I was discussing with the state mathematics specialist only the other week.

What perturbed this specialist was the difficulty in bridging the gaps between different modern math programs, due to their usage of a different vocabulary, or applying different definitions to the same words. In the course of the conversation, he said that probably standardization is more successful, or further advanced, in the disciplines used in industry, than in education. Then, knowing my background, he suggested that I would know more about that, and asked my opinion.

My response was that, while I have worked with standards committees for an aggregate of a great many hours, spread over several decades, in an effort to standardize definitions, I do not believe it can ever happen, because the rate of advancement of the technology means that new terms are always being invented, independently in different places, so that, although older terms may get standardized, there are always newer ones that have not been.

My suggested better remedy, from the educational viewpoint, was that students should learn to correlate between treatments or texts that do use terms differently, instead of trying to put cast-iron definitions on everything. But of course, that does not excuse me for those two serious misuses in my July article!

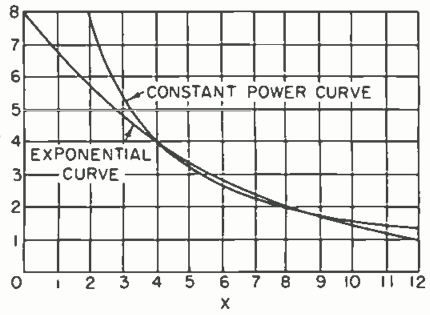
However, I would like to make a couple of observations about those usages. Taking the word *exponential*, if a horn that is defined as having that expansion law qualifies for the word, then the constant power curve does not, unless the word can be applied to more than one kind of curve. Since exponents can be used in various ways, that is possible, but evidently not accepted.

An exponential curve follows an equation of the form, $y = ke^{ax}$, while the constant power curve can be represented by the equation $y = kx^{-1}$. Curves representing such equations are shown at FIGURE 1, where the exponential form is reversed for better comparison. But is the constant power curve a hyperbola? I presume the good professor's sophomores would jump all over me for even asking the question!

The equation that I learned for a hyperbola takes the form $y^2/a^2 - x^2/b^2 = 1$. Such a curve is symmetric about both x and y axes and has asymptotes at an angle to both (FIGURE 2). Now here is where my comment to the math specialist comes in. Obviously, using that standard form for a hyperbola, the constant power curve is not one.

However, as any readers who are math buffs can fairly readily check,

Figure 1. Comparison of constant power and exponential curves.



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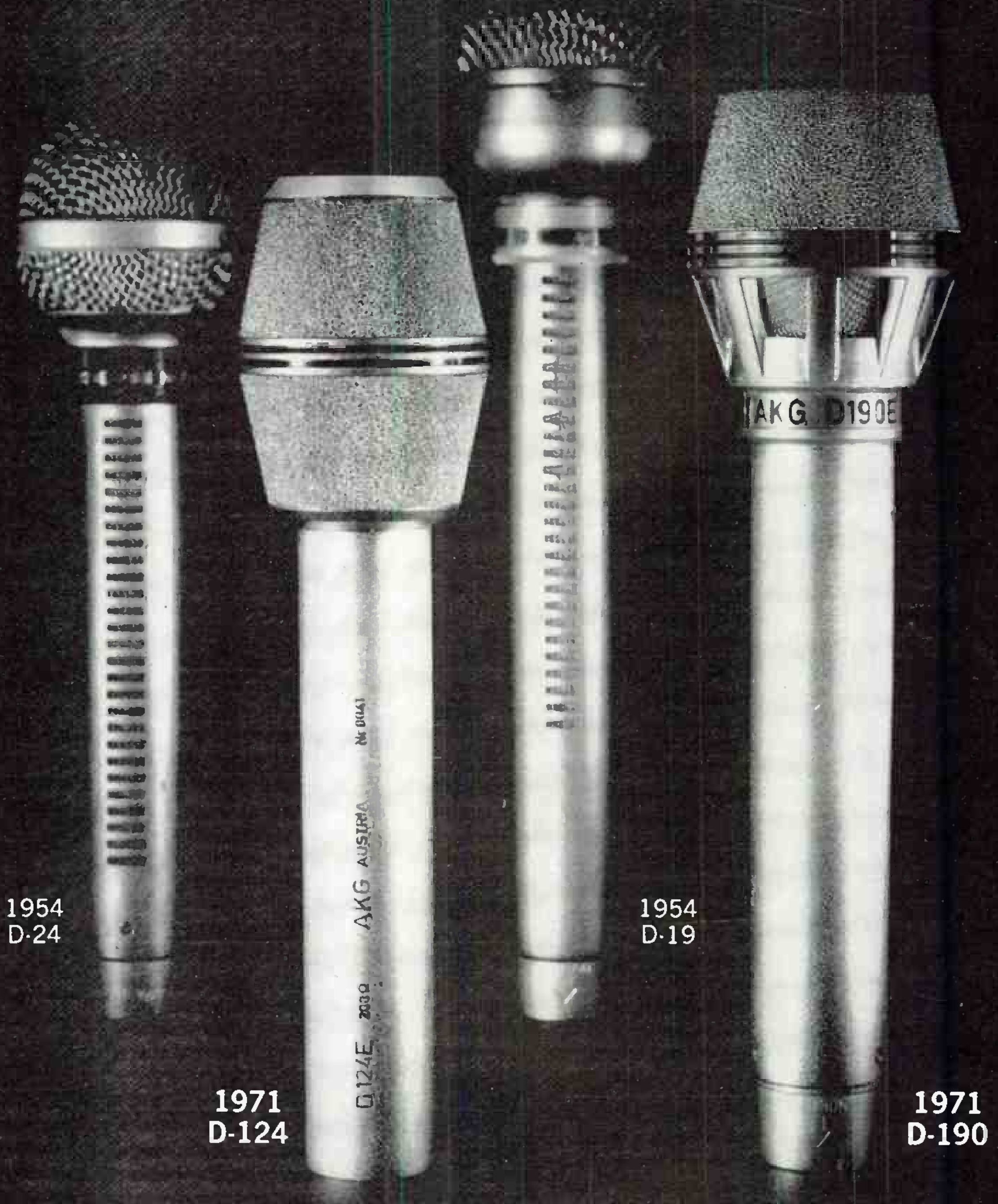

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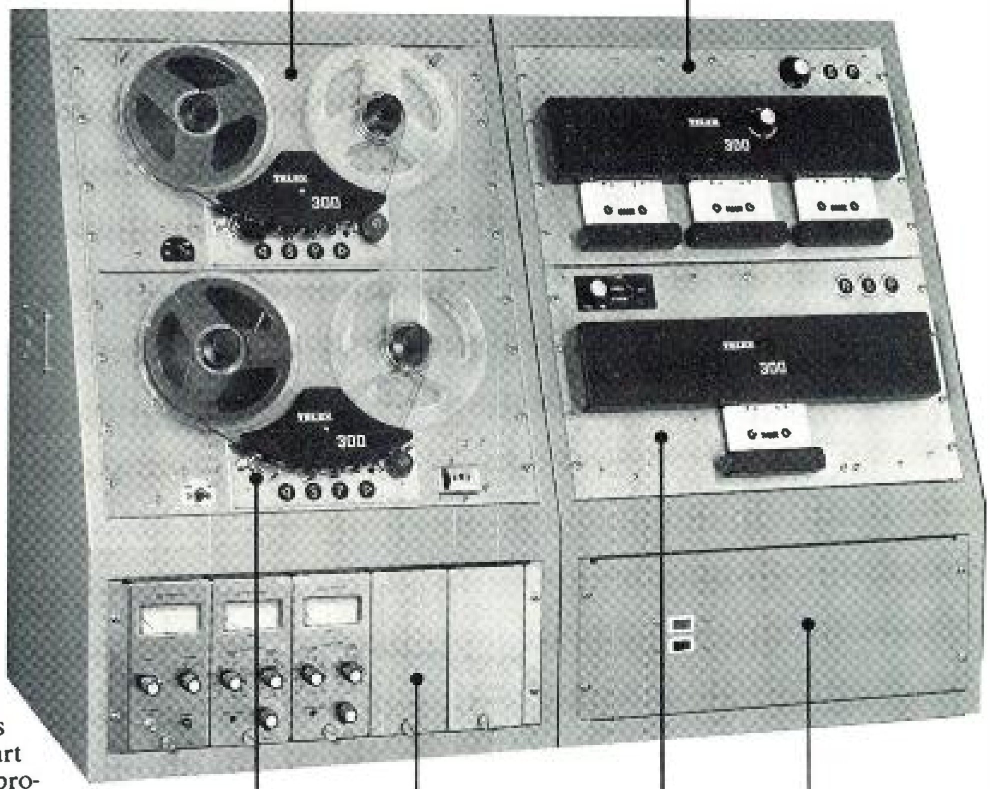

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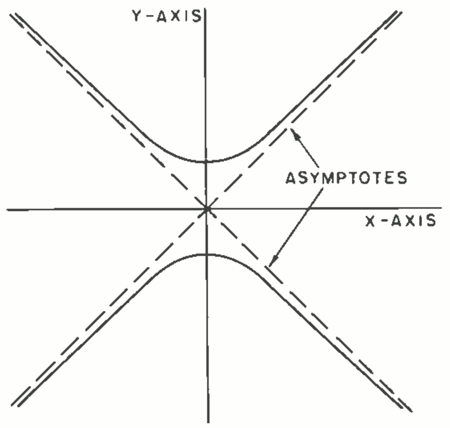


Figure 2. Hyperbolas, as conventionally drawn.

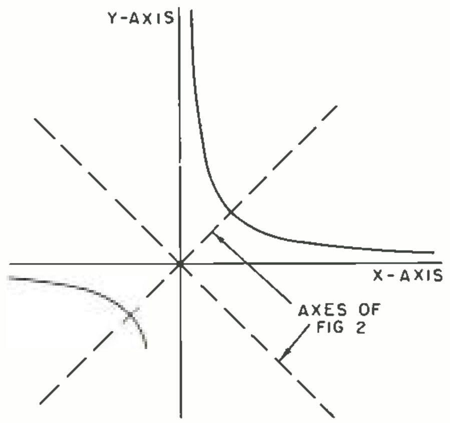


Figure 3. Rotation of axes of hyperbola makes it the same as the constant power curve.

the constant power curve is a hyperbola, if you make the x and y axes the asymptotes (FIGURE 3). This is just a matter of rotating the coordinates used for reference. Now, fairly obviously, changing the coordinates does not change the shape of the curve, *per se*, so it is a hyperbola.

But that could, in some texts, be a matter of definition. This is quite similar to the definition of *vector* in modern texts. The way I learned it, a vector was a line starting from an origin and terminated with an arrowhead, that indicates magnitude and direction of a quantity. That was generally in a two-dimensional field. If time was also a dimension, then the vector would be regarded as rotating.

We realize that vectors could be extended into more than a two-dimension-plus-time field, but they were still, in our heads, the same kind of animal. Now I look at one dictionary of technical terms and find vector defined in terms of n-dimensional matrices. After explaining this, another paragraph says, "a vector is often indicated graphically by means of an arrow . . ." and then describes the form with which I am more familiar. Here again, we have two terms of reference for the same thing. But the educational bias shows in the definition: one reference is stated as what the thing is, the other is what it may be indicated by. As I see it—and I'm open to correction—either is only a term of reference by which to define it. So why get nit-picky?

I hope the good professor's sophomores do not get too shut in on arbitrary definitions, so that they are unable to interpret material written before those definitions were standardized!

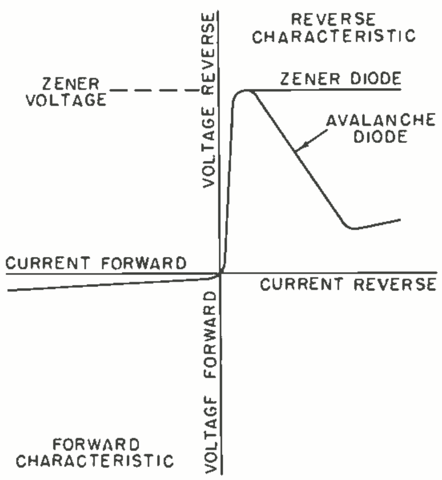
Now about that zener/avalanche bit. I had to look that up, to see what accepted usage is. One usage is the voltage associated with the reverse volt-ampere characteristic of a semiconductor, in which the voltage remains substantially constant over an appreciable range of current values, from which we get the application to *zener diodes.*

But it is also likened to the Townsend discharge (that stirred some memories of my early electrical courses) or electron avalanche, being the voltage at which such avalanche is initiated (FIGURE 4). In fact, chasing definitions back and forth, between voltage and current, zener and avalanche, there seems to be considerable interchange of usage.

From my earlier work, struggling with standards committees, it seems to me that a result of this definition-oriented educational system we seem to be saddled with is that, if one can take a wrong, or at least illogical, turn, one does. My efforts through all those struggles, which included the establishing of usable definitions of quantities like insertion gain and mismatch loss, was to achieve consistency and unambiguity.

When reverse voltage is applied to a semiconductor, a point is reached at which its characteristic abruptly changes. That point is referenced as a voltage and thus, in my head at least, is appropriately designated a zener voltage for that semiconductor element. What happens to current after that point is passed may vary. If voltage holds substantially constant over a range of current, that will also be zener voltage. But if the avalanche current pulls voltage down, so to speak, the voltage at which the avalanche is initiated is still a zener voltage.

Figure 4. Essential features of semiconductor characteristics, relevant to avalanche or zener properties.



The avalanche region, where current rises while voltage drops, represents a negative resistance condition, making an avalanche diode fundamentally different in its use than a zener diode. But in going over accepted definitions, *zener current* is, as one would expect, the range of current, over which zener voltage is substantially constant, in a particular semiconductor device. But a comment is added in one book of definitions, to the effect that this may also be called *avalanche current.*

Now, do you want to insist that zener current is what flows in a zener diode and avalanche current is what flows in an avalanche diode? If so, what flows in a transistor must be transistor current! But in the article in question, I was not talking about normal transistor usage, but the effect of exceeding the normal reverse voltage into a region that is normally dangerous.

When I started looking that up, I began to think that maybe transistors always behaved like avalanche diodes under this condition. I found no evidence for such a conclusion. The way I would expect to find it, from having worked with the little beggars, is that they have differing reverse characteristics beyond the usual range. As they are not intended for use there, they are not specified.

Diodes made as avalanche or zener diodes have those characteristics carefully controlled during production. What would happen in a rectifier diode in the corresponding region is not usually specified. But any diode has a zener voltage—or reverse breakdown voltage—whatever happens beyond that. Now, you may prefer to use the term reverse breakdown voltage, reserving zener only for those cases where constant voltage is maintained over a range of current. But in my mind, it only matters so long as we know what we are talking about.

As Charles Lutwidge Dodgson, writing as Lewis Carroll, had Humpty Dumpty say to Alice, "When I use a word, it means just what I choose it to mean—neither more nor less!" ■

Martin Dickstein

SOUND WITH IMAGES

Sound & Light—Up To Date

● What brought this subject back to mind was an experience of this past summer which left me a bit baffled (which is easy to do, I admit).

By *back* to mind I mean only that in the December, 1968, issue of **db**, this column had a general background story on the techniques involved in putting together such a complex audio/visual presentation. The equipment which had been used up to that time was outlined, and the general philosophy of the program was outlined. Since that time, many things have happened in the choice of sites for such presentations, the success and failures of these endeavors, and, of course, the greatly improved technologies involved in recording and presenting a visual and aural spectacular such as this.

Although *Son et Lumiere* (Sound and Light in French) began in France in 1952 and has since spread from castles and historical buildings in that country to ruins in Greece, sites in Egypt, Holland, and many other locations in Europe and Africa, it was not until this form of entertainment reached this country that it began to meet up with hardships, failures, and other difficulties. Perhaps there were difficulties with setting up the presentations in Europe and Africa, but evidently they are more successful than not since there are so many of them—and almost all of them still going many years after their opening.

One can only surmise some of the reasons for the failures of sound and light in this country unless information from research is available. For example, it is probable that the cost of labor and equipment in this country is quite a bit higher than it is in either Europe or Africa. This means that a higher admission price would probably result. Sophistication of the American viewer being what it is, perhaps not too many would travel out of their way to see this type of presentation. It would be necessary, therefore, to choose sites to which the public would come anyway for historical reasons or because the location is a popular tourist attraction itself. The added attraction of the evening show, when the site would normally be closed to the public would be an extra inducement to come to that attraction.

Perhaps the cost of the talent used for voices and the musical portions of the program are more expensive here than in other locations. This, too, could increase the price of admittance to the sound and light show. Add to this the cost of manufacture of equipment, installation labor, maintenance personnel, and recording facilities and the price is even higher. Then there's the problem involved with getting necessary permission for the show from the government, if it is a government-owned property, and the satisfying of the requirement that there be no visible equipment during the day when the general visiting public is going through the site.

This latter stipulation means that all lights and speakers must be mounted so that they can be hidden by natural-looking camouflages such as bushes or benches, or they must be taken down after each performance and reset before the next one. The difficulties are greater, naturally, with outdoor presentations than with those taking place indoors. The far greater number of sound and light presentations in Europe and Africa take place outdoors with the lights shining on the "star" of the show, the building or edifice, and the public sitting under the stars in the open air. This also raises problems such as weather, crowd movement in and out of the "theater", surrounding light and sound that can prove distracting during the show—and of course there's always the problem of sufficient parking space with easy-to-get-there approaches and roads whether the presentation is outside or not.

Perhaps the greatest deterrent to successful sound and light shows could be vested in the viewer. The show must be exciting and dramatic. The interest of the veiwer must be kept through the entire program, as during a good t.v. show. In the sound and light shows, the site is made the hero of the script. The background is presented by dramatic acting or narration and the story of the site must either be interesting and exciting or must be *made* that way lest the attention of the audience wilt in heat of the evening and cast eyes upward to look at the stars or to see what the weather is threatening to do next. Outdoor shows are always at night, naturally, and in most cases, the people remain stationary in a sort of on-the-grass theater setting. There are always additional problems if the program requires that the people move from one viewing position to another, although it is true that this feature can add to the interest of the audience.

The first presentation in the United States of this form of entertainment opened at Independence Hall, Philadelphia, in 1962. Within a couple of years the show closed, reopened at a later date and then closed again. Another attempt was made in Saint Augustine, Florida, at the Castillo de San Marcos around 1964. The history of the oldest fort in the oldest city in this country seems interesting and exciting, but the show did not last long.

In 1965, the presentation at the battleship North Carolina opened at its permanent berth at Wilmington, in the state for which it was named. The show begins with the building of the ship in 1937 in Brooklyn, through its trial runs and its battles in the Pacific to its serious damage suffered in the Solomon Islands in World War II. Cannon, machine guns, airplanes and torpedoes are all heard with horns, sounds of men running to battle stations and anti-aircraft shots stimulating the original real thing. The show lasts a little over an hour. After 5 years, the show is still running during the regular season, from the beginning of June to Labor Day. Since the ship is moored in the water and the public sits on shore to see the show, the lighting and sound effects take place both on and over the water and on land. Two separate power requirements were fulfilled. The equipment for both light and sound distribution is stationed both on shore and on the ship.

This presentation can be considered successful without referring to the monies involved, although it is probably true that if the program were not supporting itself somehow, the show would have closed, just as its predecessors. Perhaps the script is a good one; perhaps the public still remembers the saga of the capital ship and its adventures in the relatively recent world conflict, and comes to hear the story retold and to see tongues of flame spurt forth and hear the thunder of the gun batteries. Whatever the reasons for the attendance, the show will continue to present these reasons again next year. We were not surprised that when we called the company that helped install and design the presentation, they offered us

New Tools for Professional Monitoring

THE 8050A REAL TIME ANALYZER provides continuous audio spectrum scanning every 30 ms, 40 Hz to 16 kHz on ISO center frequencies. Has *fast* RMS response for visual program monitoring and *slow* RMS response for room equalization. Cost is about 1/3 of competitive models.

THE 9860A ACTIVE EQUALIZER has phase and amplitude transfer characteristics identical with those of the Altec passive equalizer. Permits detailed equalization from 40 Hz to 12.5 kHz. High and low-pass functions, 18 dB/octave, permit more precise finishing of frequency extremes. Cost is about 1/2 of comparable passive devices.

THE 8080A PINK NOISE GENERATOR plugs into standard Altec mixers. An ideal flat and stable noise source for room/speaker equalization with 24V power supply or battery. Cost is about 1/5 of standard noise generators.

Circle 23 on Reader Service Card

THE 771B BIAMPLIFIER has a switchable 12 dB/octave crossover at 500, 800, and 1500 Hz. Can be adapted to most studio monitoring systems and coaxial speakers. Acoustical balances can be exactly controlled by separate HF and LF gain controls. The LF section delivers 60 Watts and the HF section 30 Watts continuous sine wave power.

For more detailed data and specs write Altec, Professional Studio Products, 1515 S. Manchester Ave., Anaheim, Calif. 92803.

ALTEC

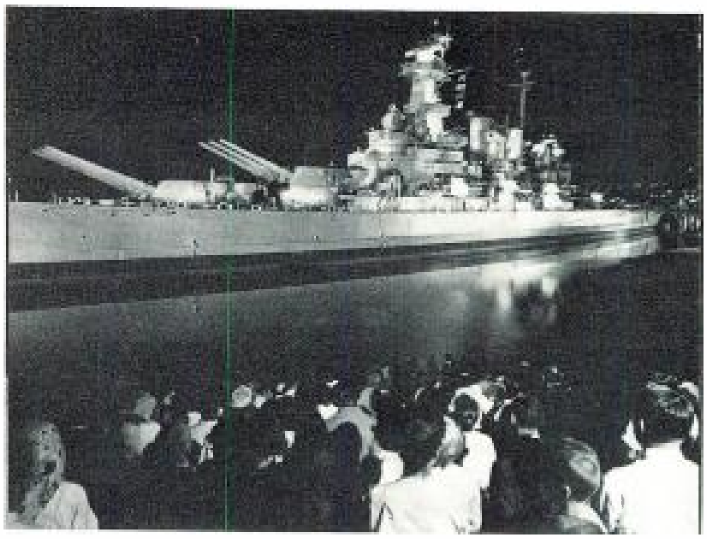


Figure 1. The famous battleship North Carolina retells the saga of its exploits during World War II in a sound and light presentation. The sound of gunfire and the zooming of enemy airplanes, the voices and sounds of the men who served aboard her, and the flames of the big guns along with programmed changing lights all add to the realism of the drama.

both written and pictorial assistance to describe their show. Perhaps we will detail this in a future story, but for the present let it be sufficient to indicate that material was provided by Philips-Norelco and the installation was discussed quite readily. (See Figure 1.)

Another successful program in sound and light is given at the exact replica of Independence Hall constructed at Knott's Berry Farm just south of Los Angeles, California. Here, the show takes place inside the building, not outside as at the original structure. This means that the show can be given in any weather under temperature-controlled conditions, and there is no need to wait till dark for a single performance. The dramatic play is presented through an 11-hour day and each show runs only 17 minutes. At this rate, the total attendance could be well over a half-million since the opening on July 4, 1966.

Now that we think about it, there are more successes with sound and light than "un-successes". Another show on which we received a good deal of research assistance from Seymour Krawitz & Co. and which is still running is at Ford's Theater in Washington, D.C. This one began in 1970, and takes place indoors, and deals in its story with the theater and the events surrounding the shooting and death of Abraham Lincoln. The actual theater is used for the show and, like the others, has no live action. The entire program is recorded on tape, acted by famous actors, and controlled by cues on the tape and a complex console. This presentation is also a near-future story in these pages.

Here's what $1,890 buys you:

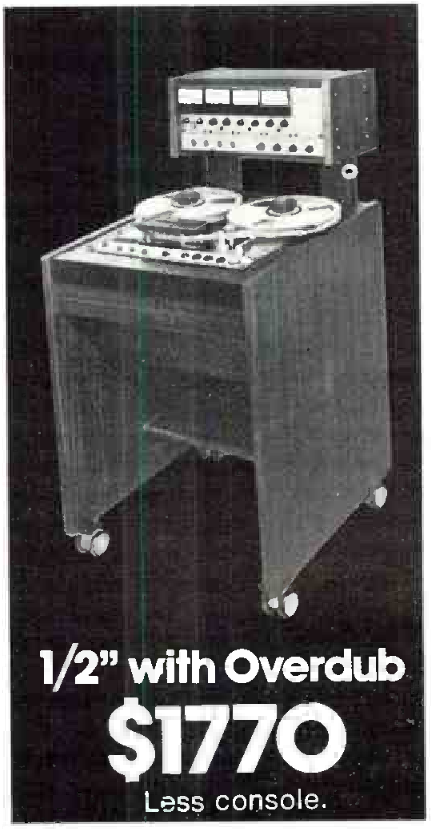
One each floor-standing TASCAM Model 10 Mixing Console. Manufactured by TEAC Audio Systems.

Eight input modules, each providing a 3-position input pad, feedback-type mic attenuator, line attenuator, 3-position input selector, hi- and lo-pass filters, hi, lo and mid-band equalizers, complete echo-send and receive circuitry including pre- and post-selection, channel and pan assignment pushbuttons, a pan pot, and a unique straight-line fader.

Four sub-master modules, each providing a straight-line fader, source/tape monitor control, and separate monitor level controls.

One master module with a straight-line fader. Four 4″ VU meters with fast-acting LED peak indicators.

Plus, pre-wired facilities for up to four additional input modules and other optional accessories including talkback, remote transport control, quad panner, and headphone monitor.



Circle 21 on Reader Service Card

And if all that seems like an awfully good buy for your money, here's what $1,770 for our Series 70 Recorder/Reproducer buys you:

½″ transport, 4-channel modular electronics, with overdub, plug-in head nest with ferrite heads and scrape filter, remote control connector, touch-button solenoid/relay controls with anti-spill circuitry, heavy-duty 6-pole torque motors, and indirect drive 2-speed hysteresis-synchronous capstan motor.

Both units available with studio line impedances optionally.

Since you're still impressed by these values, don't forget the sales tax. That's the one accessory that's not optional.

TASCAM CORPORATION
5440 McConnell Avenue
Los Angeles, California 90066
(213) 390-3566

All of which leads up to the experience of this past summer. I called Boscobel (in upstate New York, named from Boscobello, Italian for Beautiful Woods) to receive permission to write a story on the technical aspects of the sound and light show, which is presented from June to Labor Day and has been running since 1964 (another success). Permission was not granted. After a lengthy discussion it was learned that in the past stories that were written about the display somehow did something objectionable as far as the management was concerned. In one or more cases, the town in which the home of historical figure Morris Dyckman, around which the story is written, was moved from Garrison, N.Y. to several other adjacent locations. In other instances, the copy, which was promised to the management for checking prior to printing, never arrived and the story went "as is", including incorrect information or inappropriate comments.

We were told that there would not be permission granted to print any further stories. However, I was asked, at the end of the lengthy phone conversation when I would be coming to see the show. I thought that this meant that management was going to give permission for the story, but when I arrived, no such luck was forthcoming. Free admission was given as a press privilege (and when I said I would insist on paying as long as I could not do the story, this was strenuously refused), but still no story. I was given some sparse information on the background of the restoration of the building and the surrounding area, the production details of the show, but nothing tantamount to any detail at all concerning the technical aspects of the presentation.

What was learned, in essence, was that approximately one half million dollars ($500.000) was spent in setting up the sound and light show with 35 musicians, the voices of Helen Hayes and Gary Merrill doing the narration, about twelve other actors and actresses portraying the characters of the story, an extremely complex computer by RCA controlling the lighting and directional distribution of sound sources on cues from a magnetic tape, and a complex sound and lighting system to depict the play itself. All of this was a part of the restoration of the historic site which included moving the building, piece by piece from its original location some 15 miles to the South. It was also learned that it takes two men to run the show, one inside the control room in a building a short distance away from the home site itself and one outside on the grounds with the audience. By two-way radio the outside man

Figure 2. Looking back toward the parking lot, this is the walk down which the audience comes toward the house in the Boscobel Sound & Light presentation.

Figure 3. The Dyckman home. The visitors stand here for a short time during the opening narration. Lights illuminate the house in changing patterns during the show. Speakers for sound distribution are mounted in trees and at the house.

could tell the inside man if anything was not happening as it should, such as lights not turning on or off or in synch with the audio, or if the sound were not coming from where it was supposed to, or was too loud or soft. Otherwise, the program was completely automatic, requiring only the pressing of a start button for the show to begin.

It was also learned, from another source, that the tape system is four-track, three for sound and one for cues. During the show, we could tell that there were speakers in trees and on the grass areas and in the bushes, and we know that there were small speakers mounted under the benches on which the audience sat during part of the show. Controlled illumination was mounted and hidden in approximately the same locations as the speakers, some lights to illustrate action during the portrayal and some to light up the house during narrative passages. It can only be guessed that there were over 80 speakers including some sound columns at or on the house (out of sight, of course) and over 300 lights either with on/off or dimmer action. (We did count quite a few mosquitoes.)

The production lasts approximately 50 minutes and tells the story of the house, beginning the tale with Hudson discovering his river, the problems of the settlers with Indians and the events in the lives of the members of the Dyckman family through the end of the Civil War. The audience stands briefly facing the rear of the house while the narrative begins the story. Then, a short musical interlude leads the audience to the next location during which the people face a wooded area in which the next action takes place. This location permits the public to look out over the Hudson and across to West Point. (We had been informed that the musical bridge during the move of the people had originally been made too short for the time it actually took the people to move from the first position to the second, and an extensive re-recording had to be made to add to the timing of the walking music.)

At a third position, the audience is led for a brief stand facing the river and a bit of action meant to be in a wooded area along the river and on the water, and then the finale takes place with the viewers seated facing the side of the house which faces the river. To permit the audience to see the whole width of the house without obstruction, it was necessary to put the benches at quite a distance from the building. We could hear echoes as the sound of the long throw from the house came back from the wooded areas behind the benches, but the return level was only mildly distracting and not enough to disturb the intelligibility.

The synchronization seemed off in some instances, and the sound level did not always seem in proper balance between near and far sources, and the historical background of the story was rather sparse and had to be augmented with additional history not related to the tale. But in all fairness most of the audience did seem to come away with some enjoyment according to the comments they made as they left the grounds toward their cars. We had also been told that one of the reasons we were not given any technical information was to prevent others from taking advantage of the details given in the story to build their own sound and light show. We wondered as we left how many of the 200 visitors (the audience is limited to this number on purpose) would be interested in starting their own sound and light show, or how many of our readers would go out and buy an historic site with this type of presentation in mind.. True, I am being facetious. The management is entirely within its rights to refuse permission for a story or to refrain from giving out information, but I do thank them for allowing me to see the show, free. ■

NEW PRODUCTS AND SERVICES

● The AES Convention exhibits keep becoming more and more quantitative (and qualitative). We simply ran out of space last month in our attempt to list all the booths our camera had caught. So, herewith are additional ones. As with last month's listing you can receive information on the products shown, directly from the respective manufacturers, by simply circling the reader service number on the postage-paid card at the back of this issue.

• Electrodyne, part of the MCA group, showed their latest mixing consoles. *Circle 85 on Reader Service Card.*

• In addition to complete consoles, Langevin offers this assortment of components. *Circle 56 on Reader Service Card.*

• Complete control and analysis of room reverb is done with the B & K Reverb Processor. *Circle 69 on Reader Service Card.*

• RCA offers a variety of stereo and mono boards, such as is pictured for broadcasters. *Circle 73 on Reader Service Card.*

• Duncan faders can be used as attenuators for consoles and mixers. *Circle 80 on Reader Service Card.*

• EPI has created this M201A model: power and bass in studio use. *Circle 52 on Reader Service Card.*

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A ROBINS INDUSTRIES CORPORATION
75 AUSTIN BLVD., COMMACK, N.Y. 11725
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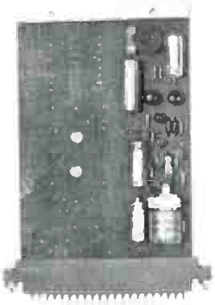

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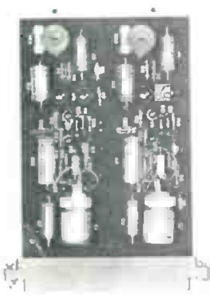

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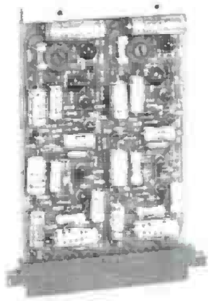

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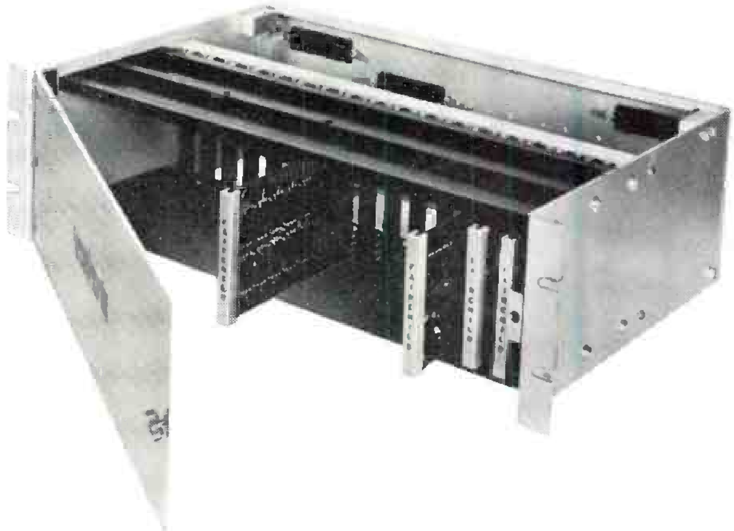
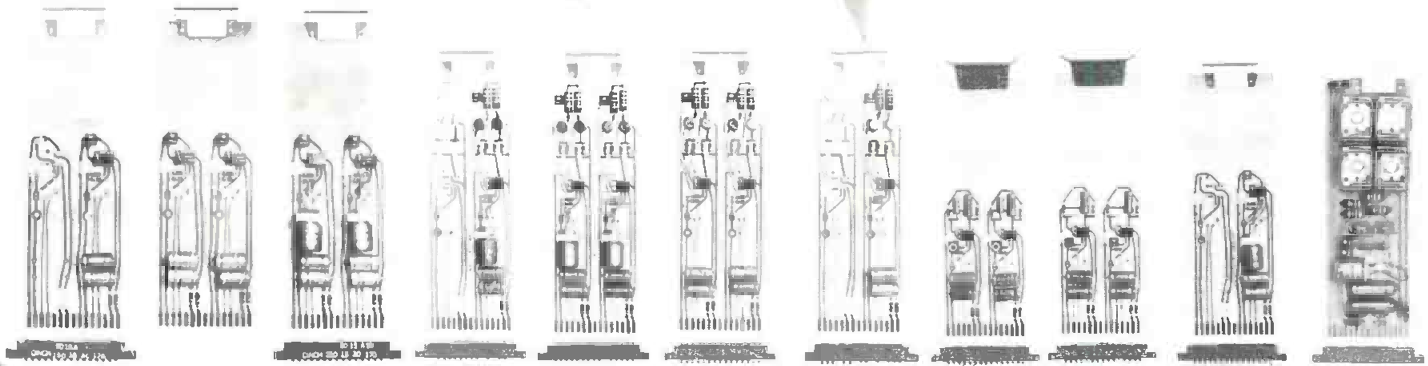




D E F G H I J K L M N

These advanced design, medium level amplifiers are versatile, compact and economical with a large variety of modular units available for any system which requires amplification. The units feature:

- State-of-the-art Op-Amp I.C. circuitry results in simplicity, reliability, high efficiency, low distortion and "short circuit proof" units.
- Very low output impedances effectively make all loads bridging loads, thus almost all units can be used as distribution amplifiers with excellent crosstalk characteristics.
- Input transformers are toroidally wound for high efficiency, magnetic coupling and minimum noise and hum pick-up.
- Modular building block design glass-epoxy p.c. boards have gold-plated contacts and are supplied with mating connector; mating card-holders, power supplies and accessories are available forming a complete design package.
- Each amplifier has power decoupling (filtering) so that interaction between circuits is minimal.
- Highest standards of workmanship, all components are mechanically fastened as well as soldered.
- All components are standard high reliability computer grade, and are open to view for ease of replacement should it ever become necessary. No encapsulated or "blind" items.

With the double amplifiers a balanced output is possible without use of transformers. Most of the units have continuously variable gain adjustable over a wide range. The 725 series, using bi-polar power supply can process frequencies from direct current to 20 kHz and allows direct coupling of stages without coupling elements.

FAIRCHILD/ROBINS ENGINEERING DATA

MEDIUM LEVEL AMPLIFIERS

MEDIUM LEVEL AMPLIFIERS

Model No.	Illus.	No. of Ampl-ifiers	Principal Use	Gain db	Input Imp. ohms	Output Imp. ohms	Max.Out-Put Level dbm	Load Imp. ohms	S/Noise db (0dbm out) Gain	S/Noise db (0dbm out) S/N	Dist. Max.%	Freq. Resp. db± (a)	Interch. Crosstalk db min.	POWER REQ. VDC	POWER REQ. @ ma	Rec. Power Supply	Dimensions H x L	Assoc. Mtg. Hardware
692AA	A	1	Line Amp	22-51	100K	3	18	150+	40	78	0.2	0.5	--	24	6	667II	3½x5(d)	692SCH or 692CF
692AA/TXI	A	1	MicPreamp	36-66	200	3	18	150+	65	60	0.2	0.5	--	24	6	667II	3½x5(d)	692SCH or 692CF
692AD	B	2	Line Amp	22-51	100K	3	18	150+	40	78	0.2	0.5	70	24	6(c)	667II	3½x5(d)	692SCH or 692CF
692AD/TXI	B	2	MicPreamp	36-66	200	3	18	150+	65	60	0.2	0.5	65	24	6(c)	667II	3½x5(d)	692SCH or 692CF
692LA30	B	2	Line Amp	0-51	10K	3	30	150+	41	60	0.2	0.5	70	48	55(c)	667T/24	3½x5(d)	692SCH or 692CF
692SPA	C	2	PhonoPreamp (b)	30-65	50K	3	18	150+	55	60	0.2	0.5	55	24	20	667II	3½x5(d)	692SCH or 692CF
725A	D	1	Line Amp	0-35	100K	0.5	18	600	35	72	0.1	0.2	--	±15	6(e)	667T/15	2½x7½(d)	725SCH or 725CF
725AD	E	2	Line Amp	0-35	100K	0.5	18	600	35	72	0.1	0.2	70	±15	6(c)	667T/15	2½x7½(d)	725SCH or 725CF
725A/T	M	1	MicPreamp	20-55	200	0.5	18	600	55	72	0.1	0.5	--	±15	6	667T/15	2½x7½(d)	725SCH or 725CF
725AD/T2	F	2	Mic Amp	20-55	200	0.5	18	600	55	72	0.2	0.5	65	±15	6(c)	667T/15	2½x7½(d)	725SCH or 725CF
725DA5/T	P	1	Dist.Amp	5-40	10K	0.5	36	8+	40	85	0.5	0.5	70	±15	120(c)	667T/15	2½x7½(d)	725SCH or 725CF
725LA	J	1	Line Amp	0-35	100K	0.1	27	8+	35	72	0.1	0.2	--	±15	120	667T/15	2½x7½(d)	725SCH or 725CF
725LA/T	G	1	MicPreamp	20-55	200-600	0.1	27	8+	55	72	0.2	0.5	--	±15	120	667T/15	2½x7½(d)	725SCH or 725CF
725LA/TB	G	1	MicPreamp	40	10K	0.1	27	8+	40	72	0.2	0.5	--	±15	120	667T/15	2½x7½(d)	725SCH or 725CF
725LAD	I	2	Line Amp	0-25	100K	0.1	27	8+	35	72	0.1	0.2	70	±15	120(c)	667T/15	2½x7½(d)	725SCH or 725CF
725LAD/T2	H	2	MicPreamp	20-45	200-600	0.1	27	8+	55	72	0.2	0.5	65	±15	120(c)	667T/15	2½x7½(d)	725SCH or 725CF
725LAD/TB2	H	2	MicPreamp	4-39	10K	0.1	27	8+	40	72	0.2	0.5	70	±15	120(c)	667T/15	2½x7½(d)	725SCH or 725CF
725SPA	K	2	PhonoPreamp (b)	45-60 (f)	47K	0.5	18	600	48	65	0.2	0.5	65	±15	6(c)	667T/15	2½x7½(d)	725SCH or 725CF
725STA	L	2	TapePreamp (g)	60 (f)	47K	0.5	18	600	48	65	0.2	0.5	65	±15	6(c)	667T/15	2½x7½(d)	725SCH or 725CF

(a) 20-20 kHz (b) Equalization curve is NAB ±½ db (c) Each section (d) Printed Circuit board with gold-plated contacts and mating connector (e) ma at full power, idling 1.5ma (f) @ 1kHz (g) Equalization conforms to standard curve for 7½"/sec ±½db

• Waters has a complete line of attenuators for every application and purpose. *Circle 78 on Reader Service Card.*

• Tapemaker showed the Livingston Sidewinder, an ingenious cartridge winder. *Circle 79 on Reader Service Card.*

• The Philips 212 turntable-arm combination offers electronic speed control. *Circle 62 on Reader Service Card.*

• The Bose 800 puts 901 components into a strong carrying case. *Circle 50 on Reader Service Card.*

JOHN WORAM

Interfacing a Dolby System

Here's what you do when you have sixteen Dolbys and more than a sixteen-track machine to use them with.

IT'S NO GREAT PROBLEM to install a Dolby noise reduction system in a modern studio. And, with the 361 units, once they're in, you don't have to give them much thought. About all you have to do is remember to turn them off if you don't want to use them.

But, what if you want to borrow a few of the units from your 16-track machine to do say, a 4-track session?—or for a 2-track playback? It's easy enough in principle. In practice, you're asking for trouble. There are 128 pairs of plugs involved in a 16-track noise reduction system. Once you begin 'borrowing' and replacing, Murphy's Law clearly states that the possibility of error increases with the square of the number of plugs touched. And, after a few moves, the tangle of wires is not to be believed, unless you've tried it yourself.

In our studio, we have the following complement of tape machines; one 16 track, one 8 track, one 4 track, and one 2 track. After a week of jockeying noise-reduction units around, we were ready to dump the whole works in the river and put up a sign saying "tape hiss is good for you." Some of our clients didn't think this would be terribly clever, so we came up with the switching system I'm going to describe.

The idea is to assign the Dolbys—as required—by simply pressing the appropriate button. Needless to say, the switching system requires a bit of logic (diodes) in order to function. In retrospect, it would have been far simpler to install a collection of switches that would route the various signals through, or around, the Dolby's as required. However, the simpler circuit would demand more on-session programming time, and we were determined to make this thing as idiot proof as possible. (The switching system to be described should not be confused with that system built into every Dolby 361 unit. The built-in system switches the unit from *record* to *playback*, and *on* or *off*. Our system merely transfers the Dolbys from one machine's playback lines to another's.)

FIGURE 1 is a simplified routing chart, showing the signal paths for any one playback track. Notice that the portion of the circuit enclosed by dotted lines is inserted into the already existing playback lines. If both the A and B relays are off, as shown, the Dolbys are inserted in the 16-track lines, and the other machines bypass the units. If all the relays are on, all the other machines pass through the various Dolbys, and the 16-track machine completely bypasses them. In our case, the other machines are an 8 track, a 4 track and a 2 track, as already noted. However, in another application, they might be any other combination of 16 or less tracks.

When the relays are on, units 2 and 4 are in the 2-track playback lines, 5 thru 8 are in the 4-track lines and 9 thru 16 in the 8-track lines. This arrangement allows us to record on any of these other machines without wiring them to program the Dolby remote control system. For example, when doing an 8-track session, units 1 thru 8 may be left permanently in the record mode, since 9 thru 16 are assigned to playback. And so on.

We have a 24-input console, 16 of which are—when in the line position—fed by the 16 track machine for mix downs. The other 8 are divided among the 4-track and 2-track machine, and a disc playback system. That means there are no inputs left over for the 8-track machine. However, if the A relays for tracks 9-16 are on, and the corresponding B relays are off, the 8-track lines will now pass through the Dolbys, and appear on inputs 9-16 in place of the equivalent tracks on the 16-track machine. It also means that the noise reduction must be turned off if it is not required. In all the other cases the noise reduction need not be turned off, since the units may be completely by-passed when not required.

John Woram is our regular monthly Sync Track columnist, our Associate Editor, and (most important) chief engineer of Vanguard Records. Next month, he'll be back in his usual columnar format.

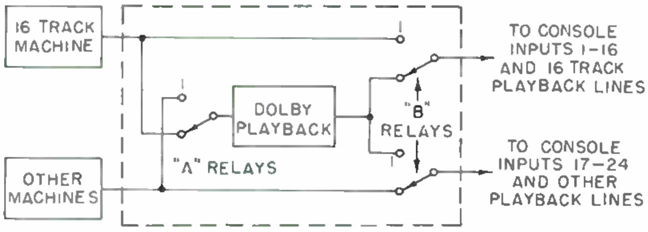


Figure 1. The simplified routing chart, showing the signal paths for any one playback track.

One other important feature—when mixing a 16-track Dolby tape down to 2 track (also Dolby) a playback unit is required, but only when listening back to the 2-track tape after it has been made. Our switching system automatically transfers two of the Dolbys to the 2-track machine's playback lines when the tape playback button on the console is depressed. When mix-down begins again, these two units are immediately returned to the 16-track lines. The same feature is also available on the 4-track machine for quad mixdowns. Of course, two or four additional record units are required for these mixdowns, but they are not permanently assigned to this control room, and so may be used elsewhere when not needed here.

It might seem more practical to switch these additional units between record and playback, rather than to borrow playback units from the 16-track machine. However, this would mean that in their absence, 2- (or 4-) track Dolby tapes could not be played back on listening or editing sessions if these additional units were in use elsewhere. Our arrangement allows us to get by with a minimum number of Dolbys on permanent duty in the control room.

By studying FIGURE 1, it's easy enough to see that the positions of relays A and B will allow any of the conditions we require. However, getting the things to do the right switching takes a bit of finagling. And things aren't helped any by insisting that the 8-track machine pass through the Dolbys for mixdowns, whether they're needed or not.

Once again, the good old truth table gets dragged out to help us over the rough spots. It's easy enough to assign four units either to the 16-track machine or the 4-track machine. But what about all the variations involved in jockeying around the 8 or the 16? The truth table seems to be the easiest way out.

Condition	A Relays	B Relays	Noise Reduction Switched Off
8 track with Dolby	1	0	0
8 track without Dolby	1	0	1
16 track with Dolby	0	0	0
16 track without Dolby	1	1	0

In the table, the first two columns refer to the condition of the A and B relays which control tracks 9-16. For 8-track mixing, the condition of Dolbys 1-8 should be independently variable, and for 16-track work, they should track the condition of relays 9-16. Note that the condition listed in column 3 (*noise reduction system switched off*) only occurs in the *8 track, without Dolby* mode. The 1 in the column indicates the relay that switches the 9-16 noise reduction off is to be energized. In the *16-track, without Dolby* mode, the action of the relays allows the 16-track machine to bypass the Dolbys, consequently the noise reduction need not be switched off.

And now, to the circuit itself, as shown in FIGURE 2. For the moment, ignore the *master noise reduction switch*, assuming it to be permanently *on*.

The basic system consists of six pushbutton switches, each with from two to four poles. The switch armatures are labeled w, x, y, z. The switches are in three groups of two labeled *1* and *2*; two each for the 2-track machine, the 4-track machine, and the multi-track (8- or 16-track) machines.

Depressing the *2-track in* button grounds the 2 track A and B relays via the *x* armature. At the same time, the *y* armature interrupts the flow of d.c. to the 16-track light. This means that, in the event that *16-track, with Dolby* had also been selected, the absence of voltage at the 16-track light would indicate that the 16-track mode is not available. Once the *2 track* button is switched *off*, continuity is restored to the 16-track light, and it will go on if *16 track* has been selected. Continuity may also be interrupted by the *z* armature of the *4 track on/off* switch, which performs the same function for the relays associated with 4-track playback.

Note the *standby switch* in both the 2- and the 4-track sections. Assuming the *in/out* switch is in the *out* position, the *standby* light may be turned on by depressing switch 2. The *y* armature illuminates the *standby* light and the *x* armature establishes continuity down to the *machine selector* switch. If this switch is in the *2-track* position and the *monitor mode* switch is on *tape*, a ground path is established back to the 2-track relays and the Dolbys are diverted to the 2-track machine for playback. When the operator returns the console to *mixer* position, the ground path is opened and the Dolbys return to the 16-track lines. Note that the *standby* light will not come on if the *in/out* switch is in the *in* position. In this condition, the Dolbys are locked into the 2- (or 4-) track lines, and *standby* becomes meaningless.

Note also that two monitor lights are shown, whereas all other functions show only one light. Actually, with the microswitch assemblies that were used, each of these lights represents two bulbs wired in parallel. The monitor lights are the exception. Here, the two bulbs are wired as shown. The left hand bulb will light up whenever the appropriate relay is actuated. The right hand bulb will go on whenever *2-* (or *4-*) *track tape* playback is selected, regardless of whether or not the Dolbys are being used.

THE MULTI/TRACK SWITCHES

The multi-track section of 2 switches fulfills the requirements outlined in the truth table. Because of the requirements of the 8-track machine, the A and B relays for Dolbys 9 through 16 are not always in the same state. The truth table shows the required state and the switches set up the appropriate condition. Note that in the mode, *8 track—Dolbys out*, the z armatures establish a path to ground, through diode *H*, for the 9-16 relays controlling the *noise reduction off* function. Diode *G* prevents the 1-8 relays from also being energized. (An explanation of all diodes follows later on.)

THE MASTER NOISE REDUCTION SWITCH

So far, we have assumed the *master noise reduction* switch to be in the *in* position. In this position, the appropriate *in* lights will come on, as required. But, assume for the moment that the 2- and 4-track switches are in the *out* position. Now, we decide to remove the Dolbys from the 16-track lines. (*16 track—Dolbys out*) For simplicity of operation, our system is strictly binary. That is, all

at the mixing desk

artist: Ralph Towner

12 noon

8 p.m.

4 a.m.

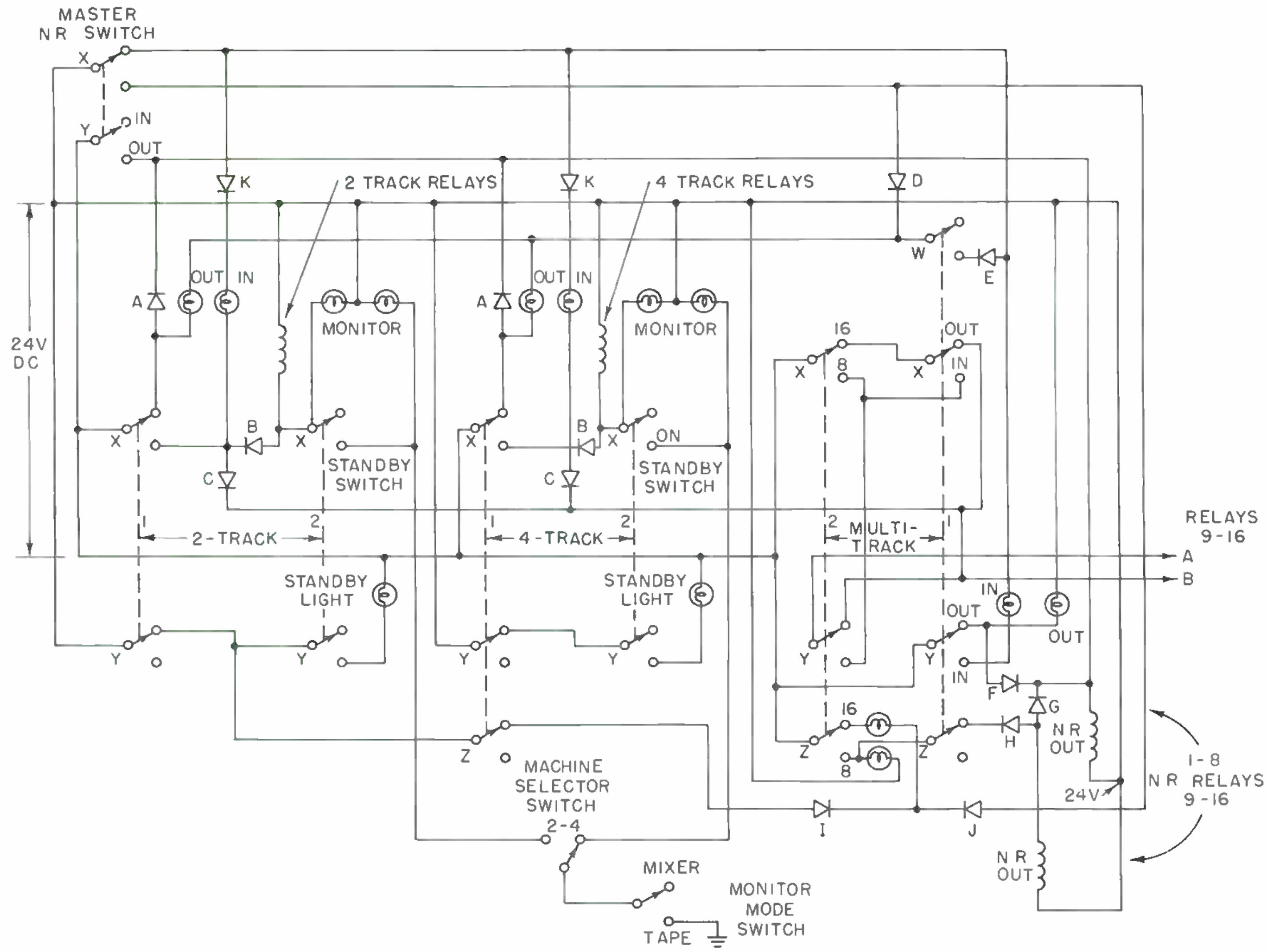


Figure 2. The complete noise reduction switching system.

conditions can be met with two position switches. The Dolbys are either in the 16-track lines or in the other lines. By definition, they cannot be in neither location. (For this to happen, there would have to be a third position on some of the switches, which would rule out simple push buttons). So, the act of selecting *16-track—Dolby out* automatically grounds the 2- and 4-track relays, via the *x* armatures and diodes c. These relays are energized, and the *in* lights come on. Voltage is removed from the *out* lights since armature *w* is opened.

In this condition, the Dolbys are locked into the 2- and 4-track lines (as well as in the 8-track lines of course). If the units are not required in these lines either, the *master noise reduction* switch must be turned *off*. In this position voltage is applied to all *out* lights, and all *in* lights are extinguished, regardless of how the various other switches may be set. Also, all the noise-reduction relays are energized, thus grounding the Dolbys and shutting the noise-reduction function off.

AN EXPLANATION OF THE DIODES

Generally, the purpose of the various diodes is to allow continuity in one direction, while preventing it in the other—hardly a new development in diode application. However, for the sake of clarity, here are some specific notes about each diode. In a few cases, there are two diodes with the same identifying letter. In these cases, both diodes perform the same function.

A allows the *master noise reduction* switch to ground both the 2- and 4-track *out* lights, while preventing the *2-track in/out* switch from grounding the *4-track out light*, and *vice versa*.

B allows the 2- (or *4-*) *track in/out* switch to ground the appropriate relays, while preventing the *in* light from coming on if the relay is grounded through the *standby* switch.

C allows *16 track—Dolbys out* to ground the 2- and 4-track relays, while preventing *2 track—Dolbys in* from grounding the 4-track relays, and *vice versa*.

D & E allows voltage to be applied to the *2 and 4 track out* lights, while preventing current flow from the line feeding *D* to the line feeding *E*, when armature *w* is closed.

F prevents the *multi-track—Dolbys out* switch from grounding the *noise reduction off* relays, while allowing the *Dolbys out* light to go on if these relays have been grounded by the *master noise reduction* switch.

G allows the 9-16 relays to track the condition of the 1-8 relays, while blocking a 9-16 ground (through *H*) from energizing the 1-8 relays.

H allows the 9-16 relays to be grounded, via armature *z*, while preventing the *8 track light* from coming on if the ground should be received through *G*.

I prevents a voltage at the *16 track light* from illuminating the *2-track standby* light in the event that there is continuity through the various switches between these two points.

J blocks current flow up towards *D* when the *16-track light* is illuminated via the lower armatures (*y* and *z*) of the *2 and 4 track in/out* switches.

K prevents partial illumination of the various *in* lights due to a current path through the de-energized 2 or 4 track relays and their lamps and thence to ground, through the *multi-track—Dolbys in* lamp. ■

MARSHALL KING

The Business of Hearing

In this, the first of two parts, the author examines hearing as it relates to the audio engineer, and particularly to him in his specialization as a t.v. audio mixer. He is quick to point out that this is one man's opinion.

One recent Wednesday morning in one of the control rooms of a prominent television station, the audio mixer on a certain comedy show appeared for work wearing a hearing aid. The reactions were as might be expected. Shock, consternation, dismay, incredulousness . . . all these and more, as both technicians and production personnel gathered in small groups and discussed *what to do*. Even the assistant director, who was older than most, had to admit that he had not seen such a professional blunder since Douglas Corrigan flew the wrong way. The general nature of the hubbub was "How does he think he can get away with it?" and "I know he's only forty-five, but if he can't *hear* why doesn't he just retire?"

It was up to the producer to tackle the problem head-on. "Get rid of the sonofabitch! I don't care if he *has* been mixing our show for six years. I'm not going to have it bandied about that a *deaf* man does our audio. How is that going to look in the trades at Emmy time? Get me someone else!"

When called upstairs, the mixer found his superiors to be very kind. They spoke softly and smiled a lot, and pretended not to notice the new hearing aid, as though he had worn it the day he was hired many years ago. "Producers are funny people," he was told with indulgence. "You know how they have their whims when it comes to personnel. But don't worry about it. This won't affect your seniority one bit, and you'll like it down in the maintenance shop. You'll still get your same salary and first pick of vacations. Sorry about your hearing."

However, this was not the end of the matter, for our hero had something to say about the situation himself. "Not so fast," he said. "It has been the company policy for years that any technician who availed himself of modern tools of the trade in order to uphold the high standards of this network would be re-imbursed for any out-of-pocket expenses incurred therein. Since the device you see in my ear definitely will allow me to deliver a better product, I hereby inform you that the re-imbursement will come to $350."

"But . . . but, you won't need your ears much down in the shop," they quickly reminded him with warmth. "Perhaps a screwdriver or two, a roll of solder . . ."

"I really don't intend to *be* in the shop," he said with respect. "I intend to continue working as a mixer at least until you remove from the *video* area all technicians who wear glasses. Correction lenses, if you please. That includes eleven video control operators, six technical directors, three tape editors and five maintenance men. Not only are they unable to *work* without their glasses, they also wouldn't have the slightest chance of finding a doorway. For your information, gentlemen, my hearing is now excellent, for I have availed myself of a modern tool of the trade."

Let me hasten to say that the above situation, to my knowledge, has not yet happened. But it may someday, somewhere. The truth is that, while the ears of the audio technician are the same type owned by other humans, they are exposed to a much more rigorous pounding from eight to eighteen hours a day for most of his working life. If we can believe that every cause has an effect, it should be no surprise that something must happen to those professional ears which may be worth looking into.

Hearing damage can be physical or psychological, can

Marshall King has appeared in our pages before. He is a television audio mixer with current credits that include the Bill Cosby Show.

be either temporary or permanent. While the ears of the whole world are being assaulted these days by loud volumes of sound from both intended as well as unintentional sources, it is the hearing faculties of those people who deal professionally with sound that is of interest to us here.

Perhaps a distinction should be made between the *noise* that has come to the public's attention as a genuine ecological enemy, and the plain *volume of sound* that has become a way of life for the audio technician. Admittedly, there may not be much difference between the two, for loudness is loudness, whether it's a power mower or an indiscriminate playback of Wilson Pickett. The *noise* that is referred to in our newspapers is that terrible cacophony which is a by-product of our civilization, such as jet planes, police sirens, tree-topping machines, and motorcycles. These sounds have earned the undying hatred of millions who consider themselves cultured, and while the control of this racket is beyond the reach of any individual, it is hoped that mass anguish rather than vigilante action will bring about governmental remedy.

But the loud volume which concerns us as audio people is that which is, to a large extent, the result of our own doing. Strictly speaking, nobody makes us run our control room speakers that loud, and there is no law that says a performer *must* peel the wallpaper with the volume from his amplifiers—or that the paying customers must sit there and listen to it. But they do, and we do, and the reasons *why* we do, rather than escape to some remote island twelve miles south of Pago Pago may be worth prodding a bit. More important, what is all this doing to our ears?

Loudness is not the only thing that can wear us down. Listening in depth can do it, too. This, combined with *continuously assaulted concentration,* the perennial bugaboo of television mixers, can have a devastating residual effect on our hearing that is too often ignored. Writing in *Playboy* magazine Max Gunther says, "Sound may also leave the brain little capacity to think." In case of the audio mixer it may be better to say that sound leaves the brain little capacity to think *about anything else.* That a violation of such deep concentration is a general assault on the nervous system and a tremendous drain on the energies can be attested to by more than one mixer who has difficulty unwinding after long sessions in the booth. Without referring specifically to audio men, ear specialist Seymore Brockman, writing in the Los Angeles *Herald-Examiner* said, "Ears seem to be wearing out faster now. Noise alone is not the culprit; stress and long working hours can contribute."

Let's look at some of the aspects of hearing that have most to do with our work: excessive loudness, ear damage, psychological fatigue, and subjective response. Since these four are irrevocably intertwined, it will do no harm to examine the last item first.

Subjective response merely means: how we act upon what we hear. If a sleeping mother hears the faint sound of her choking baby she'll spring into action at once, even though she slept through the sound of a four-car collision in front of her house. She has a narrow-pass filter set to reject all that doesn't pertain to the survival and well-being of her loved ones. It is not far-fetched to compare this to the audio mixer who is so engrossed in equalizing the sound of a Fender bass (so that it doesn't sound as someone pounding on an overstuffed pillow soaked in olive oil) that he fails to hear the buzz from a guitar amp.

Think back for a moment: it happens all the time. It's as though there is just so much you can absorb at any one instant. If you give 80 per cent of your attention to one thing there's only 20 per cent left for all the others. You may argue that you have the ability to split your hearing evenly among all elements so that nothing escapes your attention. To this I say, after counting my scars, it is

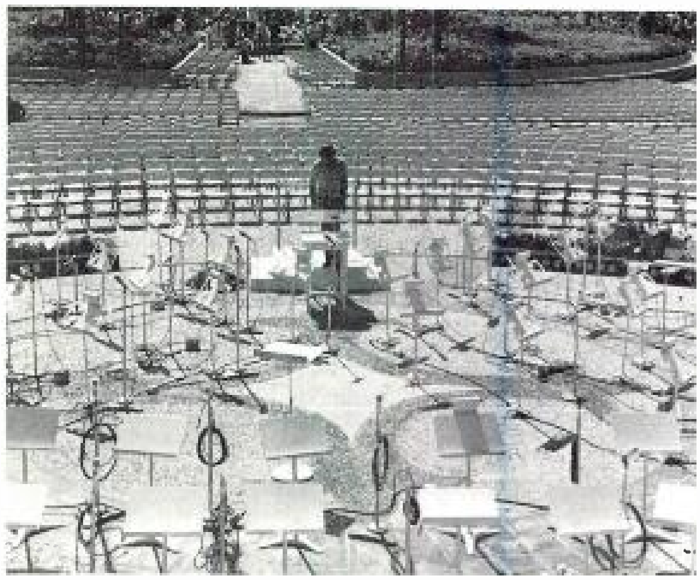


Figure 1. The calm before the storm. The tremendous volumes of sound which will flood this scene before the day is over contrasts against the morning quiet that prevails as author Marshall King checks over his mic setup for the World Symphony, conducted by Arthur Fiedler, at the televised opening of Disneyworld in Florida.

doubtful. If there are ten elements of a composite sound that you are obliged to oversee, and if you are giving equal scrutiny to each one, you are giving each one 10 per cent of your attention, which means that a fault in any of the ten has a 90 per cent chance of getting through.

While many of us may have discovered this effect independently, much work has been done in this field by experts. Writing in the *Psychological Review* (Vol. 70, 1963), J. A. Deutsch and D. Deutsch report that, "However alert and responsive we may be, there is a limit to the number of things to which we can *attend* at any one time. We cannot, for instance, listen effectively to the conversation of a friend on the telephone if someone else in the room is simultaneously giving us complex instructions as to what to say to him. And this difficulty in processing information from two different sources at the same time occurs even if no overt response is required." And if nothing else, a mixer is expected to come up with an overt response.

You may argue further, "That's not the way I work. I treat the whole thing as one sound. That way, nothin' gets by *this* cat!" The trouble with this is, if the mixer treats the audio as a single whole and doesn't know what goes into its make-up, he's in trouble. In such a case he must avoid at all costs any discussions with the a & r man, the arranger, the lead talent, the music co-ordinator, and the producer.

In all seriousness I suggest that what really happens in a mixer's mind as he works is this: After getting a rough idea as to level from each of the sound sources in the group, so they don't overload the equipment either individually or in concert with all the others, he may then turn his attention to the over-all balance. But even here I suspect he does not step back mentally during the recording and treat the thing as a single sound, as much as he'd like to. I suggest that what his mind does, in its attempt to listen to the composite effect, is to *scan* in rapid succession one element after another, much as the electron gun of a picture tube scans each element of the mosaic screen, giving each one his total attention for the short time it is visiting there.

Research made available to me by Joanne Gordon, doing graduate work in psychology at the California State University at Northridge, shows that this *selective* nature of our scrutiny has been investigated with some dramatic results. Treisman and Riley, reporting in *The Journal Of Experimental Psychology* (Vol 79, 1969), described their work in *shadowing*, a process whereby the subject is given separate messages simultaneously through phones on each ear and told to *shadow* (pay attention to and repeat back) just one of the two messages. The authors say, "they normally can report little or nothing about the verbal content of the unattended message." While the material used in this experiment were spoken words, it is probably not presumptuous to say that our ears give the same treatment of momentary priority to single elements of music.

If this is so, our attention apparently hops from the lead singer to the harmony voices to the drums to the bass to the saxes to the wah-wah guitar and back again, and again. endlessly scanning. We think we are listening to the over-all, but I feel it is impossible to do so, *in view of our position as mixers*. I'm inclined to say that the only ones capable of hearing the work objectively are those not involved in its creation: for example, the truck driver making out with the waitress in a coffee shop where our musical effort is a background from the jukebox. He hears it as a single piece, with no mosaic elements. True, his personal efforts, too, are made up of many small bits as he makes out with the waitress: whether or not she likes his rig, whether or not she is still going with that car salesman, whether or not his tattoo is impressive to her, whether or not she feels he is a good tipper, and whether or not he plays the right things on the jukebox. But the music from that jukebox is just one of the mosaic pieces which come to him as a small whole.

While it was easy for the truck driver to be objective about our music since he had no hand in its production, we, too, can put ourselves into an objective station from time to time during the recording session if we change from a *hands on* mode to a *hands off* mode. The best example of this is the difference between the actual mixing and monitoring the playback. It's true that we are still very much involved with the work even when we are monitoring a playback, but at least we are in a hands off mode, and that makes a fantastic difference in our appraisal attitude. An even greater difference in approach would be possible if we monitored our work at a later date, a thing that may be feasible in dub-down work but is seldom so in the mono mixing that most television entails.

One further example of *subjective response* will probably ring a bell with most readers who have sat at a mixing console. Not long ago when I had finished recording a certain number with Terry Gibbs and Steve Allen, my diligent cohort Andy Blyth, who sat in the booth with me and operated the tape machines, said, "What about that buzz?" I hated to admit it, but I hadn't heard any buzz. Apparently I had been so involved with the vibraphone duel between Gibbs and Allen that I tuned out all else. Not a smart thing to do. but it happens. Further proof of this *selective deafness* came a week later when the tables were reversed. Andy was learning to mix the show, for he was to be my vacation relief, and I was sitting behind as both his A-2 man and his coach. When a monologue by Rip Taylor was over and we were into the following commercial, I said, "We've got a new enemy. There's a chirp in one of the ceiling fans out there." He hadn't heard it, and proof of his extreme concentration on other matters could be seen in his palms; he was sweating so profusely through his hands that every console fader which he had used was dripping wet. I had been in a hands off mode; he definitely had not.

Perhaps some amount of pressure, and the need for instantaneous all-out judgment, is taken off the mixer in multi-track recording. for during dub-down sessions there is no need for a certain servo circuit in the mixer's head to be put into operation. This servo mechanism in the

Figure 2. Perhaps as a highlight to the loud sounds referred to in this article is this photo of Ricky Wormsbecher giving birth to one of the softest sounds known—the sound of the bass recorder. Photo by the author.

brain is the very heart of the *continuing judgment* process, a prime requirement for mixing in mono. That is, on-the-spot corrections are made as it's happening. During a television comedy show, for example, the mixer is faced with such momentary problems as: how far to dig for audience reaction, how close will the emcee hold the mic when he interviews people in the audience, will the brass be open or muted in that music cue which has been added but not rehearsed, and, will we have time to get the boom down after a wide shot or will the actor read his next line whenever he feels like it? Recording studio work, on the other hand, where multi-track audio is the rule of the day, production can be more controlled and the results a sheer delight. During dub-downs this servo in the brain is not called upon very much, if at all, for the mixer already knows what sounds are coming up, and at what levels. Only during moments of original recording is this servo in heavy operation, for the disciplined mixer at this time will mix as close to mono (or stereo) as he can, and not put all his eggs into the dub-down basket. This is not to say that multi-track mixing is like shooting fish in a barrel, but it *is* to say that mono mixing in television is much like skeet shooting with a bow and arrow—where there's no time to worry about the last missed pigeon for another is already on its way.

Admittedly, there are as many valid opinions on the matter of judgment in mixing as there are people interested in talking about it. There are a good many ideas about what is tasteful and what is vulgar (or to be more practical, what will sell and what won't), so that in the area of relativity one searches desperately for a reference point. To this extent it may be argued that, since the ears are so very subjective and otherwise hung up with preconditioning and other social gremlins, one had best lean heavily on such scientific crutches as limiters, equalizers, leveling amps, vu meters, etc. Of course this cannot be, for although a decibel is a decibel, the final criterion in the recording of a work is the way a mixer feels as he goes about his business. More bluntly, just as an umpire calls it as he sees it, the mixer must mix it as he hears it. This means that the reference point may change from session to session, which is scarcely scientific. Yet if this is not proper, is there any need for the mixer at all? Should we not turn the whole process over to the Gates Equipment Company and let the entire chain of operations, from conception to execution, become computerized or otherwise automated? If the answer is *yes*, we must go all the way and arrange for the product consumption—the ultimate listening enjoyment—to be done by computers, too, thus doing away with human intervention all the way around. If the answer is *no*, we are back at our starting point: the varying subjectivity of our judgments.

Perhaps one way to deal with unwanted variables in our work is to find that which causes peaks and troughs, and to come up with some kind of viewpoint to discourage derelict extremes while at the same time allowing for impulsive feelings of value. Certainly one way is to take frequent breaks, periods where we can get away from our work for a moment. And this doesn't mean working out a microphone problem while everyone else is taking ten. Another thing is to know when to bend and when to be firm, to fight off that kind of group thinking that too often makes the final result a committee effort. There's no denying that this may force the mixer to work as though he were independently wealthy.

Next month, Mr. King concludes with an examination of levels in the studio and their effect on the engineer. ■

WALTER JUNG

Automating the Audio Control Function, part 4

In this latest installation, the author covers a new type of programmable operational amplifier - the PRAM

IN THIS INSTALLMENT of our AUTOMATING series we look at a brand new and fascinating type of i.c., one which is much akin to the d.p.s. in that it is a very effective link between analog signal processing and the digital control of the same. Where the d.p.s. is a very effective and flexible analog switching *element*, the device we will now be speaking of is a complete four-channel digitally controlled op-amp. This device, the Harris Semiconductor PRAM (TM: Harris Semiconductor trademark for PRogrammable Analog Module), combines the functions of a four-channel analog switch and a high performance op amp in one package. With this combination in a single sixteen-pin dual inline package, the circuit architect now has a tool he might have previously hesitated to use because of the complexity and expense of the required components.

Taking a look at how the device is structured we see it's functional diagram in FIGURE 1(A). The chip consists of four op amp input stages 1, 2, 3 and 4. The outputs of these four preamps are selected by a four-input switch and fed to a unity gain output amplifier. The output of this stage is the output of the chip. You will note that each input stage is a complete differential input amplifier.

The state of the four-input switch is controlled by a decode/control section. This circuit decodes the state of the two digital input lines D0 and D1 into the four possible channel selections. A third digital line enables its use as a master control pin. The logic sense of these three control terminals is described by TABLE 1, their truth table. The

D_0	D_1	EN	Selected Channel
L	L	H	1
H	L	H	2
L	H	H	3
H	H	H	4
X	X	L	none

where L = TTL "0" ≤ +0.5V
H = TTL "1" ≥ +2.4V
X = don't care

Table 1: A PRAM truth table

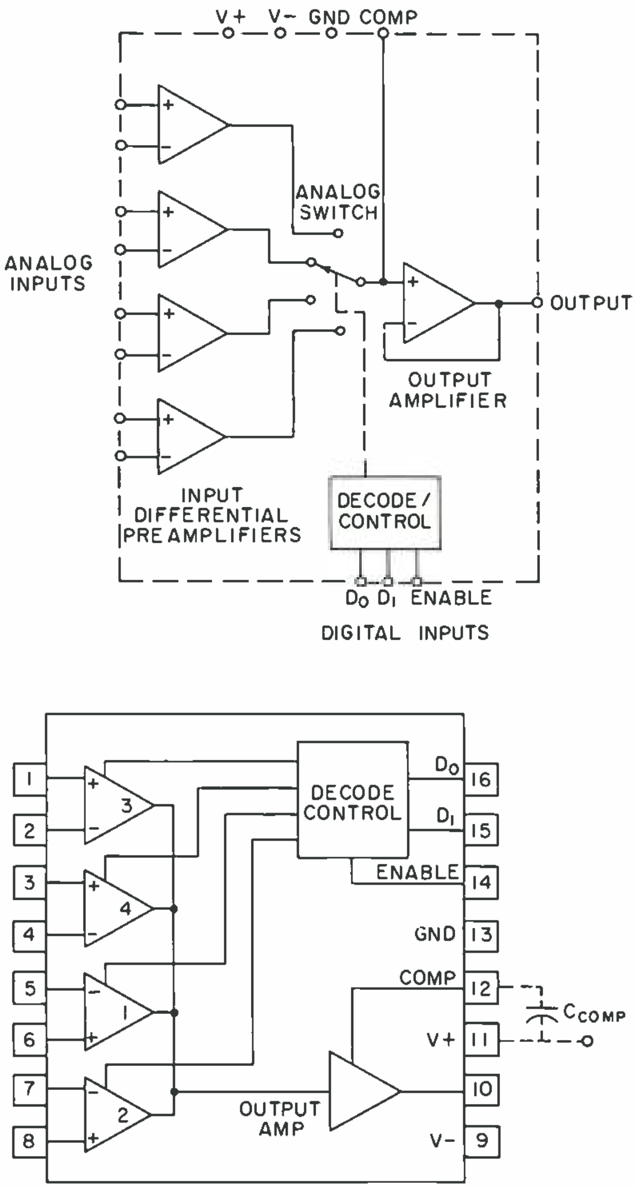


Figure 1(A). A functional diagram of a PRAM i.c. (HA-2400, HA-2404, HA-2405). At (B) pin connectors for a PRAM. Courtesy of Harris Semiconductor.

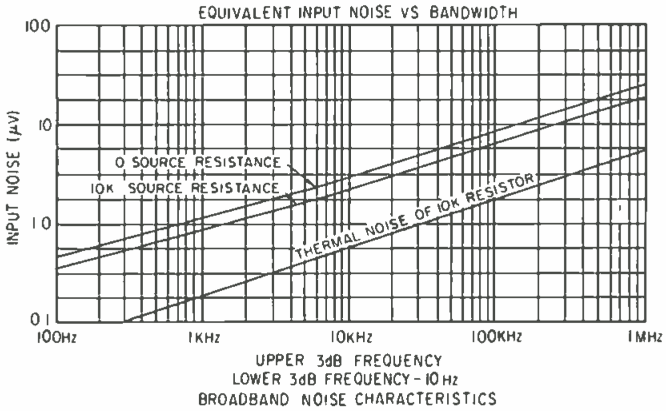


Figure 2. The equivalent noise of a PRAM amplifier. Again, courtesy of Harris Semiconductor.

digital inputs are compatible with conventional DTL/TTL logic drive. Pin connections for the PRAM are shown by FIGURE 1(B).

PERFORMANCE

Functionally speaking, a PRAM may be thought of as four independent op amps connected to a single output bus. Only the channel addressed by the digital control inputs is on, the remaining three input stages will be effectively off. The composite op-amp formed by the selected input and the output stage is quite a respectable one also. Typical specs for a HA-2405 are listed in TABLE 2. Particularly noteworthy are the excellent frequency response and slew rate specs. Gainband-width with unity gain compensation is 8 MHz and slewing rate 15 volts per μsec. At gains of 10 and above these figures may be improved to 40 MHz and 50 V/μs respectively by using reduced compensation.

A channel may be called up in a time of 100 nsec. After this short channel select interval, the output signal will slew to the new level dictated by the feedback network and at a rate determined by the compensation capacity used.

Crosstalk from an unused input channel to the *on* output is also noteworthy—it is typically −110 dB!

Noise performance of a PRAM is shown by FIGURE 2. These curves illustrate the effects of source resistance on the equivalent input noise of the amplifier, and in addition, contrast the amplifier's noise against the thermal noise of a 10 k resistor.

Parameter **Limits**

Conditions: V supply = ±15V
Ambient temperature + 25°C

(1) Offset voltage: 4mV (typical)

(2) Input Bias Current: 50mA (typical)

(3) Input Common Mode Range: ± 10V (minimum)

(4) Voltage Gain, R_L = 2K: 150,000 (typical)

(5) Gain Bandwidth (typical)
  (a) @ Av = 10, C_{comp} = 0: 40 MHz
  (b) @ Av = 1, C_{comp} = 15 pf: 8 MHz

(6) Slew Rate (typical)
  (a) Same conditions as 5a: 50V/μs
  (b) Same conditions as 5b: 15V/μs

(7) Select Delay (typical): 100ns

(8) Crosstalk (typical): −110 dB unselected input of ± 10V to output

Table 2: Capsule Specs of HA-240S PRAM.

Since the PRAM is an externally compensated amplifier, the frequency response is dependent upon the value of compensation capacitance used (Ccomp of FIGURE 1(B)). Also, since all four input amplifiers share the common capacitance used for closed loop stability, this capacitor must be selected for the lowest operating gain channel. For instance, if any one of the four channels is to be operated at non-inverting unity gain, then the maximum compensation must be used to satisfy this worst case condition. On the other hand, if the minimum gain was −1 this would halve the required compensation, since this configuration uses half as much feedback. The required compensation capacitance for inverting and non-inverting stages with gais of 1-10 are given by TABLE 3. It is easy to see the improvements in slew rate as a function of the capacitance reduction.

APPLICATIONS
MULTIPLEXER: NON-INVERTING, UNITY GAIN

The first application described using a PRAM is shown in FIGURE 3. Here four input voltages E1 to E4 are applied

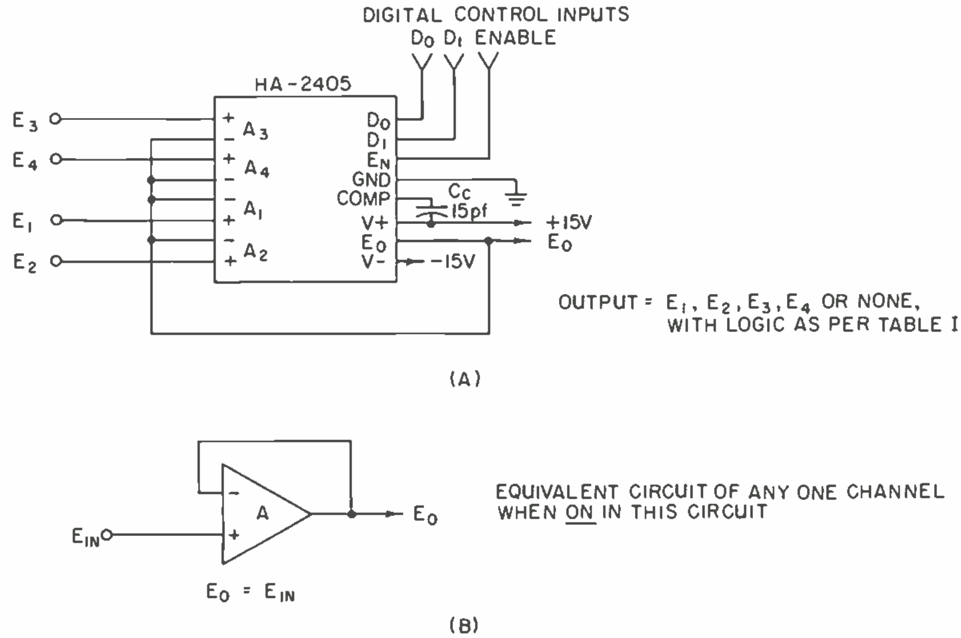


Figure 3. A four-channel multiplexer, non-inverting, unity gain scale factor.

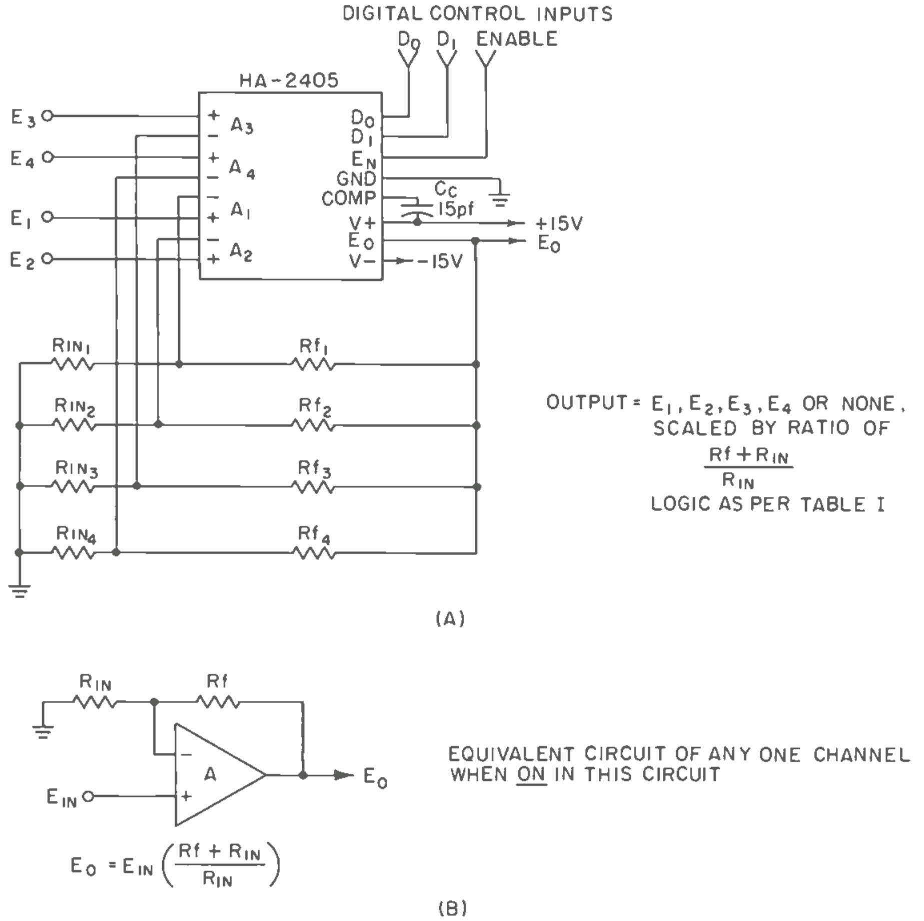


Figure 4. A four-channel multiplexer, non-inverting, with scaling option.

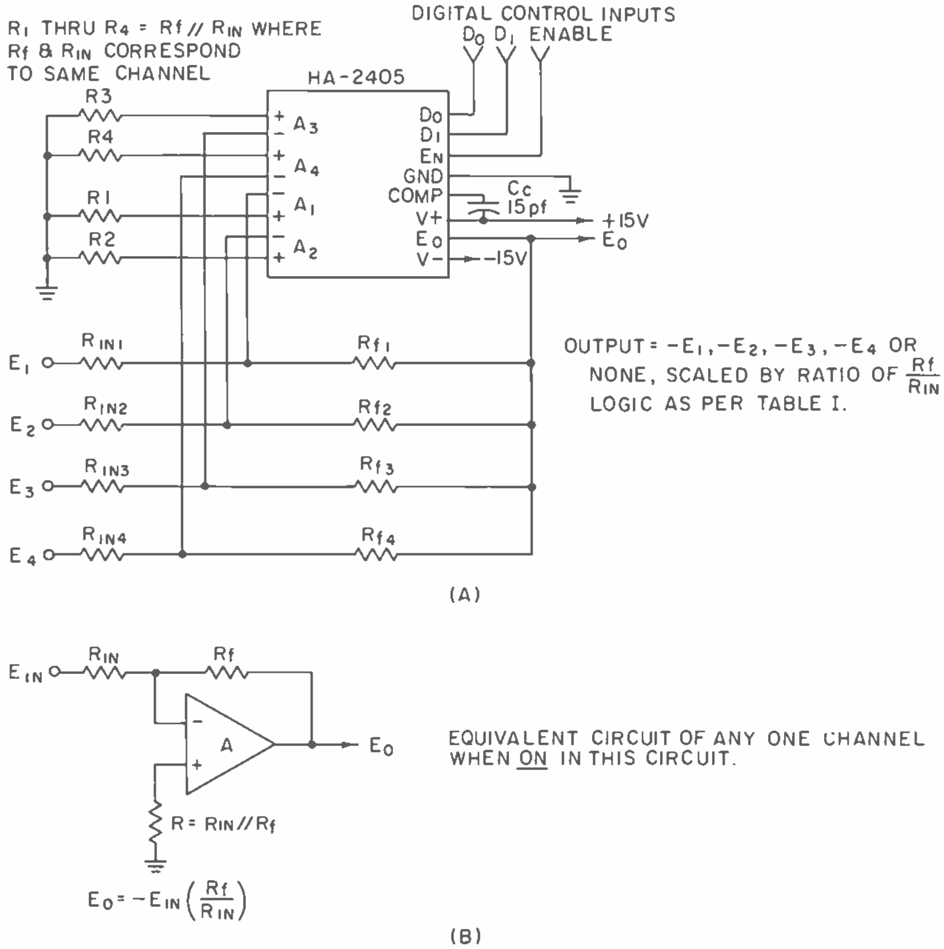


Figure 5. A four-channel multiplexer, inverting, with scaling option.

to the non-inverting inputs of the PRAM's four input amps. All inverting inputs are tied directly back to the output bus, so each stage sees 100 per cent feedback. When any stage is selected by the digital control inputs, it forms with the output stage an equivalent circuit like that shown in the insert, a unity gain, non-inverting stage (voltage follower). Thus this circuit is a four channel multiplexer, suitable for remote signal selection with the added virtues of ultra high input impedance and low output impedance.

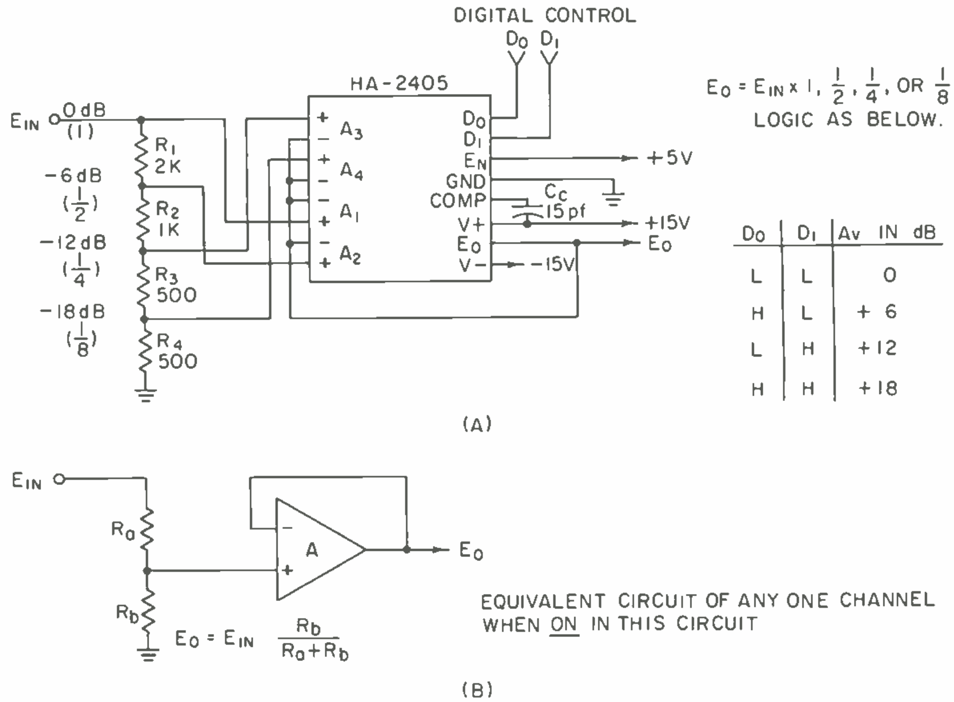


Figure 6. A programmable attenuator, non-inverting four gain states. At (B) the equivalent circuit of any one channel when on in this circuit. Ra and Rb are equal to the total series and shunt resistances the input sees.

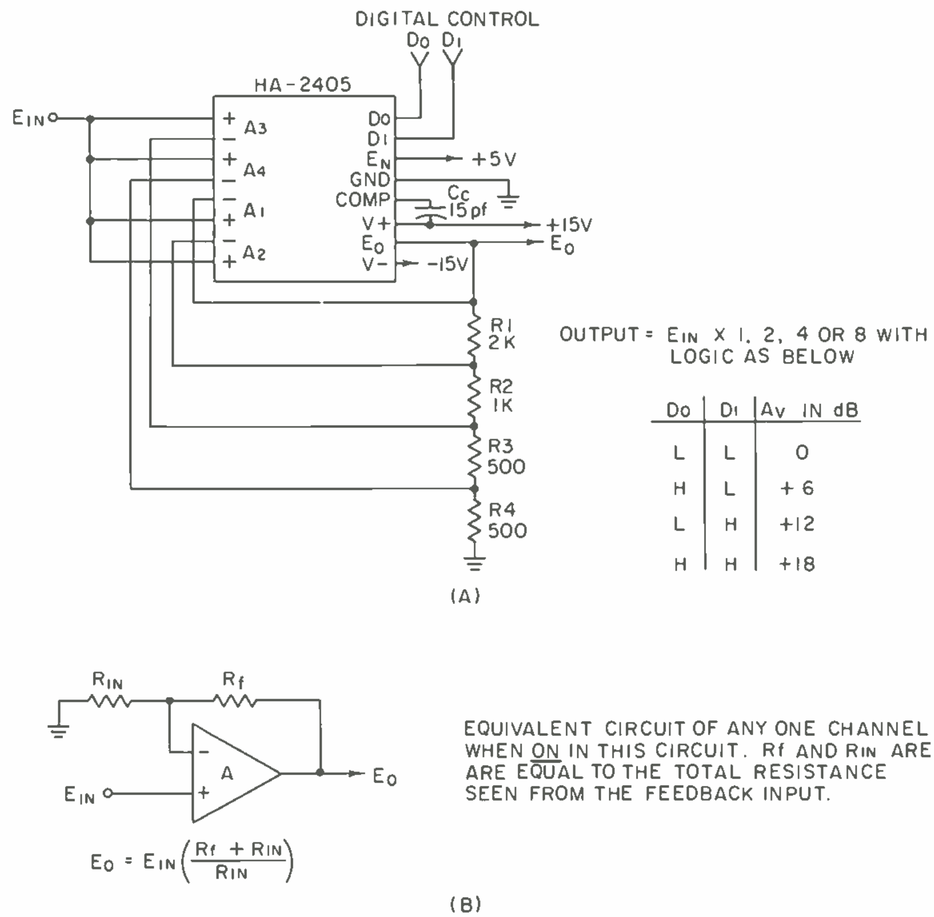


Figure 7. Almost the same as Figure 6, except for the placement of the attenuation network. At (B) the equivalent circuit of any one channel when on in this circuit. Rf and Rin are equal to the total resistances seen from the feedback input.

MULTIPLEXER: NON-INVERTING, ADJUSTABLE SCALE FACTOR

By simple modification of the basic non-inverting multiplexer of FIGURE 3 we come up with a more general form of the non-inverting multiplexer, FIGURE 4. This one has an variable scale factor by virtue of the adjustable feedback networks. These feedback networks scale up the input signal of the activated input channel according to the relation shown in the insert. Since this is still a non-inverting gain stage, input impedance remains high. Cc should be selected to satisfy the lowest closed loop gain channel in accordance with TABLE 3.

MULTIPLEXER: INVERTING, ADJUSTABLE SCALE FACTOR

Multiplexers are not limited to non-inverting configurations of course, an inverting version is shown in FIGURE 5. Here signals are applied to the inverting inputs of the four input stages through Rin, with feedback applied via Rf. Again, an individual on stage is shown in equivalent form in the insert. This configuration is also a scaling type, and individual channel gains may range from below to above unity. Again, Cc must be selected to compensate the lowest gain stage. The stage input impedance is equal to Rin.

Gain, Volts/Volt Non-Inverting	Inverting	C_{comp} pF	Bandwidth (typical) (−3dB), MHz	Slew Rate (typical) volts/μs
1	—	15	8.0	15
2	1	7	8.0	20
3	2	4	8.0	22
5	4	3	6.0	25
8	7	2	5.0	30
>10	>9	0	40÷Gain	50

Table 3: Chart of Minimum Required Compensation Capacitance for Various Gain Configurations with Attendant Bandwidth and Speed. (Courtesy Harris Semiconductor.)

Multiplexers can also be constructed by intermixing feedback configurations *within* a single PRAM stage. For instance, the largest input signal may be passed with a gain of +1, and several others scaled upward to the same level by adjusting their individual feedback networks. Should one signal be out of phase, an inverting channel may be used to bring it back to the desired phase, and also scale its gain if necessary. The key is to set up the input amplifiers to accomplish the desired forward transfer function in each input channel, then program between the channels digitally.

PROGRAMABLE ATTENUATORS AND AMPLIFIERS

Input signals to a PRAM stage may be either related or unrelated. Speaking strictly in a general sense, the multiplexers discussed above would have unrelated input signals. However, two of the most important applications of a programmable module are digitally variable attenuators and amplifiers. In a sense these are multiplexers, as they condition an input signal by various degrees of gain. So it is really a multiplexer with a related input signal—actually the same input signal at different levels. The selection of the function of attenuation or gain is governed by the placement of an attenuation network; before the amplifier for programmable attenuation, in the feedback loop for programmable gain. Once the gain ratios have been calculated (how this is done we will see shortly) a network may be used interchangeably—for gain or loss, as the requirements dictate.

PROGRAMMABLE ATTENUATION

If we turn now to FIGURE 6 we will see what is meant by the family resemblance to a multiplexer. Here the four input PRAM, wired as a selectable unity gain follower (note the wiring to the IC itself is the same as FIGURE 3) selects taps from the input divider R1-R4. This rather simple example provides a net circuit gain of 0, −6, −12 and −18 dB, digitally programmable according to the table. By now you are probably asking yourself—isn't this all obvious.? But wait—let's carry it a couple of steps further and develop a general procedure for attenuator and amplifier design.

PROGRAMMABLE AMPLIFICATION

Suppose you take the same R1-R4 network of FIGURE 6 and place it in the feedback loop—voila! A programmable times 1, 2, 4 or 8 amplifier, FIGURE 7. This circuit is virtually identical to the attenuator of FIGURE 6, the only difference being in the placement of the attenuation network. This serves to illustrate the point that the same network can serve in either case, providing identical gain differences.

This now leads us to the point where we can see the emergence of a useful tool for signal amplifier design. By designing a single network wtih taps, we can program the gain (or attenuation) of a PRAM. Given a range of dB we must be able to handle, this sets "coarseness" of the gain changes. Likewise, given a number of dB per step fixes the range. A procedure and example for a single PRAM circuit is given in the boxed example. This example, a 30 dB gain ranger, might be used as shown to program gain, or by changing to an attenuation configuration, program attenuation changes over a corresponding range.

This is certainly not the limit to gain handling range, nor is it the limit to resolution. In the next installment we'll explore a procedure for designing to a specific range of dB with a specified increment of change. All it takes is a few more parts and a simplified procedure—one step further towards automating the audio control function. ■

References

D. F. Jones *The HA-2400 PRAM Four Channel Operational Amplifier* Harris Semiconductor. Note 514, February 1972.

Programmable Gain Calculations

Let's take a programmable gain example to illustrate how the network calculations work. Suppose you wanted to remotely program a mic channel over a 30 dB range of gain from +30 to +60. You have 30 dB of fixed gain available in the preamp, so you must supply addditional stages of 0, +10, +20 and +30 dB gains. This works out rather nicely; 10 dB changes over 30 dB. So we know we must come up with a configuration to supply these requirements. In general it will look just like Figure 7, since it has four gain states—0 dB minimum with three others ranging upward, and it is non-inverting.

Selection of the gain determining resistances it facilitated by redrawing the amplifier as below into its functional equivalent. This shows the PRAM as a single amplifier with four switch selected gain taps from R1-R4. Since all resistor values are interdependent, we must solve for values which are mutually satisfactory to all gain requirements. So . . . on to step 1:

Step 1: Solve for R4 and total R1-R4 resistance to satisfy 30dB gain requirement. Solution to be in terms of "R".

Av = 30dB which is $\frac{31.6}{1}$ numerically

$\therefore \frac{Rf+Rin}{Rin} = 31.6$. In this case Rin = R4 & Rf = R1 + R2 + R3

then $\frac{Rf}{Rin} + 1 = 31.6$ or, $\frac{Rf}{Rin} = 30.6$. If Rin = R,

Rf = 30.6R

R4 = R, R3 + R2 + R1 = 30.6R

Step 2: Solve for 20dB gain ratios

Av = 20dB which is $\frac{10}{1}$ numerically

$\therefore \frac{Rf+Rin}{Rin} = 10$. In this case Rin=R3+R4, Rf = R1 + R2

restating, $\frac{Rin}{Rf+Rin} = \frac{1}{10}$ or, Rin is $\frac{1}{10}$ of total resistance

Since total divider is equal to 31.6R (from step 1),

$$Rin = \frac{31.6R}{10} \text{ or, } Rin = 3.16R$$

but Rin in this case = R3+R4 and R4 = R

$\therefore$ R3+R = 3.16R

R3 = 2.16R

Step 3: Solve for 10dB gain ratios

Av = 10dB which is $\frac{3.16}{1}$ numerically

$$\therefore \frac{Rf+Rin}{Rin} = 3.16. \text{ In this case, } Rin = R2+R3+R4$$

Rf = R1

$$\text{then } \frac{Rin}{Rf+Rin} = \frac{1}{3.16}$$

again, since total divider is equal to 31.6R,

$$Rin = \frac{31.6R}{3.16} \text{ or, } Rin = 10R$$

but Rin = R2+R3+R4

and R3+R4 = 3.16R (from step 2)

$\therefore$ R2 = 10R − 3.16R

R2 = 6.84R and R2+R3+R4 = 10R

Step 4: Solve for remaining resistor, R1

if R2+R3+R4 = 10R (from 3 above)

and total R = 31.6R

R1 = Total R − (R2+R3+R4)

R1 = 31.6R − 10R

R1 = 21.6R

Step 5: Check

A for 30dB—does $\frac{R1+R2+R3+R4}{R4} = 31.6?$

$$\frac{21.6R+6.84R+2.16R+R}{R} = \frac{31.6R}{R} = 31.6$$

B for 20dB—does $\frac{R1+R2+R3+R4}{R3+R4} = 10?$

$$\frac{31.6R}{3.16R} = 10$$

C for 10dB—does $\frac{R1+R2+R3+R4}{R2+R3+R4} = 3.16?$

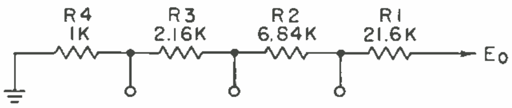
$$\frac{31.6R}{10R} = 3.16$$

D for 0dB—100% feedback, implicitly unity gain

Step 6: (finally!) substitute real values in terms of R ratios

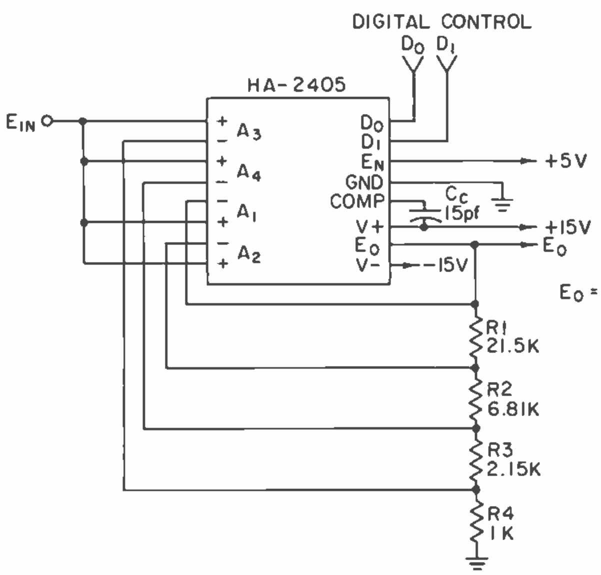
A if total divider is kept reasonably low in impedance (< 100K), R may be based on 1K. (If situation is otherwise, scale R downward, i.e. = 100Ω)

B R1+R2+R3+R4 with R = 1K yields



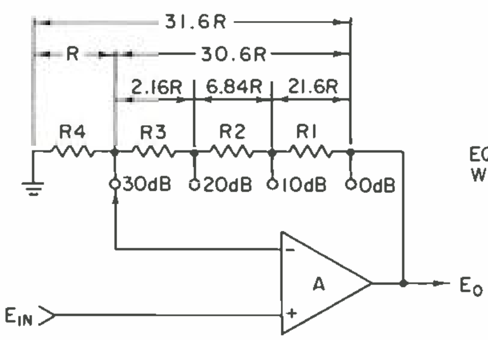
which are very close to 1% values. Since stability is usually more important than absolute accuracy, the 1% values will be satisfactory in most cases.

Step 7: The final circuit—with, would you believe only 6 parts?



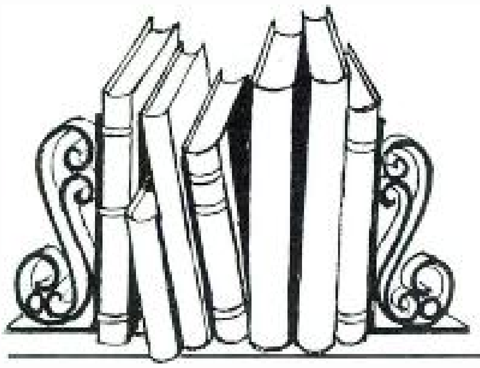
$E_O = E_{IN}$ IN SCALE WITH GAINS OF 0, +10, +20 AND + 30 dB, LOGIC AS PER TABLE

D_0	D_1	A_V IN dB
L	L	0
H	L	+10
L	H	+20
H	H	+30



EQUIVALENT CIRCUIT OF ANY ONE CHANNEL WHEN ON IN THIS CIRCUIT.

Note: Although the example shown is a programmable amplifier, the same network can be used for programmable attenuation by transposing R1 and R4 to the input (as in Figure 6).



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PEOPLE, PLACES, HAPPENINGS

LONDON BRIDGE, ARIZONA

● **London Bridge** has been reconstructed in Lake Havasu City, Arizona. But it missed something of its London environment in the desert. So each day an electronic tape system, along with discretely hidden speakers, rings out the chimes of Big Ben on the hour and half hour. Tape-Athon equipment is used. The installation was by **Hollywood Sound Systems** of Hollywood, Ca. The bridge is the actual London Bridge, moved stone by stone from London—22 million pounds of granite blocks—to the Arizona location.

● An **Electronic Music Workshop** is planned for February under the aegis of the **Boston Experimental Electronic-music Projects** of Brookline, Mass. It will be a non-mathematical approach to electronic music with an opportunity for performers, composers, teachers, technicians, film makers, and the like to learn about this art and gain practical experience in it. There is a wide variety of equipment available. The workshop includes five two-hour in-studio lab sessions (first week) and one individual studio hour arranged during the second week. The total fee (not counting possible living expenses) is $75.00.

Exact dates are February 19 to March 2, 1973. The contact is **Robert Ceely, BEEP, 33 Elm Street, Brookline, Mass. 02146.** There is a detailed prospectus.

● A most interesting demonstration was recently held at **Vanguard Records** studios. A live session was scheduled and a dual recording setup used. The group was miked with dual microphones at each location—one leading to the Vanguard control room, and the other leading to a Lamb mixer.

It was all an attempt to show that the **Revox/Lamb/Beyer mini studio**—a package consisting of a Revox recorder, Lamb mixer, and Beyer mics—is sonically competitive to a first-rate studio system. Vanguard was feeding into a **Neve** console and then to a **Scully** two-track machine. Both the Scully and Revox were run at 15 in/sec, both are two-track units. The same tape, correctly biased for each machine was used.

Of course, one mixer at the Neve and another at the Lamb produced different mixes. But both were of professional quality. The Lamb mixer was more conservative as to the level he put on the tape, so the Revox playback had somewhat more hiss on it than the Scully one. (Dolby was not used on either machine.) But other than the slight hiss difference, the audience, mostly composed of members of the **New York Audio Society,** could not clearly distinguish a sonic difference between the tapes played in syncronization (from two studio Scully machines).

● Word from **Quad-Eight** tells us that **Ron Neilson** has been appointed marketing director for the company. He comes to the company with a solid technical base of experience, tempered by capabilities in the subjective side of professional audio. Once a musician, he's been quoted thusly: "I wanted to know what really happened on the other side of that double glass window, to understand the processes which translate the artistic message."

● **E. Rorbaek Madsen,** chief engineer of **Bang & Olufsen** A/S of Denmark, is retiring at the end of this year. Widely recognized for his development work in audio, and the recipient of numerous awards, he is too spirited to long remain idle. Consequently, beginning with the new year, he will start his own international consulting firm specializing in acoustics and electro-acoustics as well as consulting for B&O. The new firm, to be known as **Akustika,** will be located at **Gimsinghoje 16, 7600 Struer, Denmark.**

● Open reel commercially-recorded tape is having a resurgence. **Ampex Stereo Tapes** has introduced a new generation of recorded open reel tapes. Selected open reel releases will be recorded using the Dolby-B noise reduction system. In addition, all open reel releases will be made on new low noise, high output stock specially formulated by the **Ampex Magnetic Tape Division.** In addition, they have installed new duplicating electronics. The combination of new tape and new electronics will create a noise improvement of duplicated tapes of 5.6 dB. Dolby-B will provide up to 10 dB of hiss reduction above that.

● An announcement from **Sony Corporation of America** tells of the appointment of **Raymond J. Steiner** as senior vice president and a member of the board of directors. He has been vice president of the consumer products division since late 1969. Prior to joining Sony, he was national sales manager of direct markets for the **Sylvania Home Entertainment Products Corporation,** where he had been for ten years.

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EVX-44 Universal Four-Channel Decoder $99.95

Now Electro-Voice offers our new universal decoder circuit built into both a stereo receiver and a 4-channel decoder.

The EVR-4X4 Receiver has every needed feature for 2- and 4-channel sound. Yet it costs no more than many 2-channel units that offer half the pleasure. And it will properly decode any matrix 4-channel input without switching. Ideal for playing SQ*, Stereo-4†, or any other matrixed records, tape or FM sources.

The EVX-44 Universal Decoder creates a 4-channel control center for existing stereo equipment. Just add a second stereo amp and two more speakers. Unique separation enhancement circuit automatically adjusts front-back separation as required by program material.

Both the receiver and the decoder are also designed to accommodate 'discrete' inputs like 8-track tape if you wish. Hear the finest in four-channel sound at your Electro-Voice showroom. Where the excitement is!

E-V 4-channel products are produced under U.S. Patent No. 3,632,886
**T.M. CBS, Inc. †T.M. Electro-Voice, Inc.*

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