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THE SOUND ENGINEERING MAGAZINE
MAY 1972 $1.00

COMING NEXT MONTH

• A special issue—devoted to four-channel sound and recording. This important topic will be discussed from a variety of viewpoints as we do a detailed wrap up of the recent Midwest Acoustics Conference. This Conference spent a complete day in explaining, detailing, and discussing four-channel recording techniques, available and coming hardware, and trends that are appearing. You will get an up-to-the minute concept of just what the four-channel scene is all about from this article.

An oscilloscope display that shows the four-channel output of a system is a real boon to the recording engineer. Donald L. Patten has developed a black box that will convert a conventional 'scope into a four-channel display 'scope. He tells you how to build it. And with it there will be no excuse for wrong phase information in any channel, unless you want it there.

Stephen H. Lampen of 3P Recording returns to our pages as he describes how he used portable professional four-channel recording equipment to record live theater.

And there will be our regular columnists: George Alexandrovich, Norman H. Crowhurst, Martin Dickstein, and John Woram. Coming in **db**, The Sound Engineering Magazine.

ABOUT THE COVER

• In the end, the tapes made by recording studios must come to the disc mastering room. This is the last audio controllable action that will be taken before the ultimate user gets his copy.

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George Alexandrovich

THE AUDIO ENGINEER'S HANDBOOK

Resistive Pads–the Practical Way

• Since most pads used today are home brewed it seems appropriate to review the most commonly used types so as to design them easily without sacrificing accuracy. Pads are used for many different purposes: isolation, matching of impedances and levels, termination, combining, and splitting.

In practice there are two types of circuits to be concerned with—balanced and unbalanced. Before we get to the design of the pads we have to define the purpose of the pad we are to construct. Most likely we are confronted with two circuits which are to be joined or interconnected. In the video field most of the circuits are of standard impedance and level and can be readily interconnected. Audio circuits are tricky most of the times. Aside from the fact that many transformer-isolated circuits require correct termination, the output from such a circuit can be permanently delegated into one line while part of the signal is diverted through switching into other circuits. Any change in loading of the source impedance will affect the level of the distributed signal. Many professional institutions prefer to permanently terminate the circuit and bridge-off with much higher impedance lines. Usually bridging means acquiring the signal by connecting the load of at least ten times larger than the source impedance. Actually, in all present-day transistorized class AB amplifiers with very low output impedances, we are bridging loads. That is why the output level of the amplifier remains constant.

Let's get back to the question of the pad. What do we want the pad to do? For one thing, should it or shouldn't it terminate the source? Secondly should it change the signal level? Third, should it present the load with the proper terminating impedance?

Let us talk first about unbalanced circuits. Several types of pads can be used. The simplest is the bridging resistor. It is a simplified form of voltage divider pad where the shunt element

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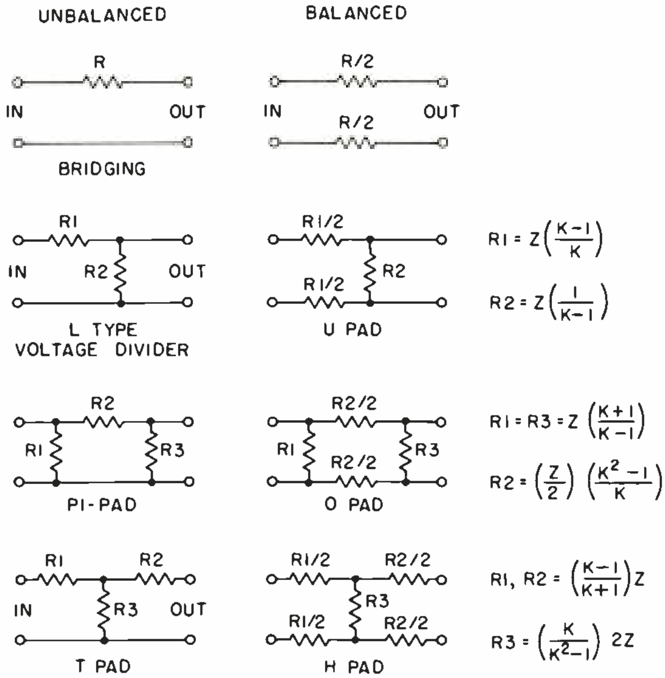

Figure 1. Equations for pad values of equal input and output impedance values.

of the pad is the input impedance of the load. It can be used only if impedances and voltage levels are defined and termination of the source and load is not required.

The second type of pad is the same voltage divider pad but with the shunt element in the form of an external resistor.

The third type is the so-called pi-type pad. It consists of three resistors. Two outside ones shunt or terminate the load and the source; one in series determines attenuation of the pad or insertion loss.

The fourth is the T-type pad. It may be viewed as part of a twin voltage divider with a second shunt element being the input resistance of the load. These are the four basic types of pads. One can spend all his life engineering audio circuits and systems just using these basic pad forms.

In order to make any one of these pads balanced, the value of the series or bridging element should be divided by two. One thing should be remembered. Using a balanced pad in an unbalanced circuit is more dangerous than an unbalanced pad in balanced circuits. Unbalancing a balanced line usually doesn't introduce as much noise and pickup as loosing the ground in unbalanced circuit by inserting a balanced pad.

Many textbooks have detailed formulas for figuring pads with accuracy to the n-th degree. In practice we are mostly interested in 5 per cent accuracy— in certain cases more precision is required. This tells us that with standard tolerance resistors we can construct any pad we need. It is hard to remember all the formulas for different pads. Sometimes there is no time to look it up in the book and even if you do the effort may not be justified.

You may have a problem of just shifting the range of level adjustment about 20 dB without any regard to accuracy. For these problems I always carry with me a copy of a dB scale superimposed on a voltmeter scale. This instantly gives me the conversion factor from dB to voltage ratio and backwards. Since most of the pads are voltage dividers this gives me the impedance ratio of the pad. Here are the most commonly needed points:

dB	Voltage ratio
0	1
2	1.25
4	1.6
6	2
8	2.5
10	3.15
12	4
14	5
17	7
20	10
26	20
30	31.5
40	100
50	315
60	1000
80	10000

This chart is also useful when amplifier gain is specified as a voltage amplification to convert it to the dB equivalent (approximately). When accuracy is not required and attenuation is 20 dB or more use of formulas is not needed. Only T pad which allows us to maintain lowest circuit impedance without loading excessively the source or the load, is easier to calcu-

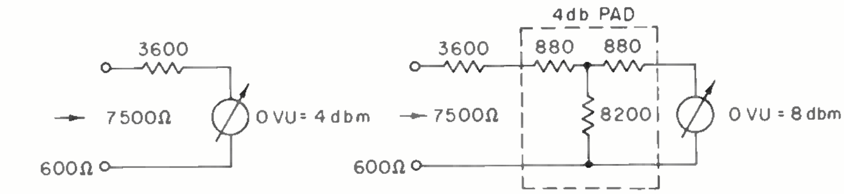

Figure 2. Meter pads.

late using formula.

Pads which are designed to produce more than 20 dB of attenuation (and accuracy is not of prime importance) values for the resistors are chosen as follows: If source requires termination then a resistor slightly higher than the required terminating value is selected and used shunted across the source. For instance, if a circuit is 600 ohms and you want to drop level 20 dB, the shunt resistor can be 620 ohms (5% standard value) or 680 ohms (10% value).

Voltage ratio for 20 dB attenuation is 10/1. If load impedance is 10 k ohms but we want to keep the impedance of the pad and interconnecting line low, we should select a shunt resistor for the load to be (let us say) 620 ohms. This value in parallel with the 10 k ohm load impedance will give us a combined resistance of 585 ohms. In order to achieve 20 dB attenuation a series element of the pad should be nine times the value of the shunt resistance. Since the value in this case turns out to be 5250 ohms, the closest standard value is 5600 ohms. This is 7 per cent off the attenuation target—or a little more than ½ a dB.

If you feel you can get better results with other standard values try changing the shunt resistor to 680 ohms. Total shunt resistance will be then 37 ohms. The series resistor then becomes 5720 ohms. Again, using the same 5600 ohm resistor we are only 2½ per cent off—less than 0.2 of a dB. This is all fine, but we have used 10 per cent resistors and we can be as much as 11 per cent off (10 per cent for the shunt value and 1 per cent as an effect of 10 per cent value variation of the series element). But don't get discouraged. Today's resistors are so good that chances are that you won't be more than a couple of per cent off in the worst case. And you can select resistors if you have enough on hand.

Almost all of today's amplifiers have variable gain so error in the pad can be easily compensated by the gain control. However, when it comes to splitting and combining signals, closer tolerance resistors are needed if accuracy is imperative. One interesting aspect of today's mass-produced resistors is that deviation in resistance value is common to a particular lot and if we start testing (for instance) 10 per cent resistors from the same batch you will find most of them being of the same value within 1 or 2 per cent, but all of them several per cent away from the nominal value.

Should you have to make the same pad balanced, divide the 5600-ohm resistor into two (actually consider 5720 as true value). Dividing, we get a value of 2860 ohms in each leg of the balanced pad. Closest standard value is 2700 ohms. If we use this value we will be 9 per cent off. If we substitute the 680 ohms shunt by a 620 ohm resistor, error will be less than 1.5 per cent.

Let us see what happened to the loading of the source. If we have used 620 ohmms parallel with 6200 ohms of pad resistance, total loading is 565 ohms which is too low. Use 680 ohms, then total resistance will be 618 ohms or 2.5 per cent off.

The higher the attenuation the easier it is to find the value for the pad using this method. If, on the other hand, attenuation has to be low (6 dB isolation pad, for example) either use the text book formula or an approximation method when certain conditions exist. Take, for instance, a transitorized line amplifier, class AB with an output source impedance of a few ohms (some, with heavy feedback can have 0 ohms source impedance). Termination of such an output is not needed (every load will be bridging anyhow). All you must have is 6 dB isolation from the load which is to see a 600 ohm source. Well what can be simpler—just put a 600-ohm resistor (two 1200-ohm resistors in parallel) in series. With this, all conditions will be satisfied.

If you require a balanced pad divide 600 ohms into two resistors. But the closest standard value is 330 ohms, a 5 per cent value is available in 300 ohms. But what if you don't have it and your dealer doesn't stock them. Then either use 270 ohms in series with 33 ohms, 220 ohms with 82 ohms, 180 with 120 ohms or four 1200-ohms resistors in parallel.

Let us look at a slightly different kind of pad—a vu meter pad. As you know, standard vu meters are designed to have a 0 vu reading when a series resistor is 3600 ohms and the voltage level is 4 dB above the 1 milliwatt


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level in a 600 ohm circuit—or 1.228 V. Meter resistance is 3900 ohms and with the external multiplier 7500 ohms. Ballistics of the meter are requiring that the meter see a 3600-ohm circuit for proper damping. If we want our vu to read 8 dBm instead of 4 dBm at 0 vu, our pad should be constructed as follows:

First of all we have to lose 4 dB of voltage in a 600-ohm circuit. We can make an L pad with a 680-ohm shunt value and a 470-ohm series value, thereby obtaining correct loading for the meter damping and proper voltage for indication. But there is one thing wrong—1000 ohms load for just the meter circuit may be too much for some amplifiers. Unless we have power to burn we have to try to design bridging pads.

Usually, a T pad is the most appropriate pad form in this case. Formula for calculating a T pad is

$$R_1, R_2 = \left(\frac{K-1}{K+1}\right) Z$$

$$R_3 = \left(\frac{K}{K^2-1}\right) 2Z$$

Where R1 and R2 are series resistors
R3 is a shunt resistor

Z is input and output impedance.

K is the ratio of current, voltage and power corresponding to a given value of attenuation expressed in dB.

If we insert this T pad between the vu meter and on external 3600 ohm multiplier resistor, input and output impedances become identical and the pad is easier to calculate. If you put a pad between the amplifier or line to be measured and the multiplier, then a different formula for T pad has to be used—the so-called tapered. Since input and output impedances differ

$$R_1 = Z_1\left(\frac{K^2+1}{K^2-1}\right) - 2\sqrt{Z_1, Z_2}\left(\frac{K}{K^2-1}\right)$$

$$R_2 = Z_2\left(\frac{K^2+1}{K^2-1}\right) - 2\sqrt{Z_1 Z_2}\left(\frac{K}{K^2-1}\right)$$

$$R_3 = 2\sqrt{Z_1 Z_2}\left(\frac{K}{K^2-1}\right)$$

Values of K expressions can be obtained from the tables by looking up the desired attenuation in dB and corresponding value of K. Substituting into first formula for T pad values of

$$\left(\frac{K-1}{K+1}\right) = .226 \text{ for 4 dB and Z of 3900}$$

$$\text{and } \left(\frac{K}{K^2-1}\right) = 1.048$$

we get for R1 and R2 value of 880 ohms and for the shunt resistor 8200 ohms. Since we have a series resistor of 3600 ohms, adding 880 ohms it makes it become 4480 ohms. We have a choice of connecting in 4700 ohms and correct the attenuation by trimming other values or taking two resistors in series, 3300 and 1200 ohms. Resistor of 880 ohms can be simulated using 560 and 330 ohms in series (only little more than 0.1 per cent off).

The most precise calculations can become meaningless if one can not obtain correspondingly accurately resistance values. Accuracy is meaningful to a point—there is no sense to adjust meter resistors to 0.1 per cent accuracy if meter movements can be as much as 5 per cent off. One might say the tolerances add up. They do, but they can also cancel each other out. If the meter reads low and the pad doesn't attenuate enough, the results may be almost perfect reading. It is our responsibility to use our ingenuity and practical know-how to use this disadvantage to our purpose by learning about shortcuts in designing pads—adjusting their tolerances by using standard resistor values and achieving precision results. ■

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John M. Woram

THE SYNC TRACK

• Earlier this year, two meetings on microphones were held in the New York area. The first was a talk by Electro-Voice's Lou Burroughs to the Society of Broadcast Engineers, held at the WQXR Theater in midtown Manhattan. For many years, Lou has been giving informal talks on the uses, and abuses, of microphones. "Doctor Burroughs' Medicine Show"—it has been irreverently called, usually by people who wish they had thought of it first.

"Doctor" Burroughs offers no universal cure-all for the microphone user. He just gets up and tells everyone what they already know, or maybe it's what everyone *should* know, or, *used* to know. Whatever it is, it amounts to an hour or two of good old horse sense, and if his medicine show comes your way, make sure you get to hear it. Lou is just now finishing a book on microphones which should be ready by the end of this year. Judging from his talks, it should be a valuable addition to the literature now available.

It's perhaps a rare session when someone doesn't break at least one of the rules of good microphone usage. Of course, rules were made to be broken, but if you know what the rules are, at least you can break them with authority, and if you really know the rules, you may find an even better way of doing things.

For instance, have you ever split a small group into two sections, facing each other? Then have you put two cardioid microphones more or less back to back—one facing each section of the group to give you more control. Congratulations, you've just built yourself a fine omnidirectional microphone! To explain—some three-pattern microphones contain two diaphragms. For a cardioid pattern, only one diaphragm is activated. For the omnidirectional pattern, the diaphragms are connected in phase, and for a figure-8 pattern, the same two diaphragms are connected together out-of-phase. So, two cardioid mics placed back-to-back will give you just about the same thing. If they're in phase with each other, you can forget about 90 degrees off-axis rejection—there isn't any. Just don't be surprised later on if you have a lot less separation than you expected. However, if the mics are out of phase with each other, the net result will be more as a figure-8 pattern. Off-axis sounds will be minimized, yet you still have some control over each mic. For best results, the microphones should be closely matched, so that cancellation remains as constant as possible over the entire frequency range. Also, a minimal amount of gain riding should be attempted, since the combined polar pattern will change as the relative balance between mics is varied. If you really want to get clever, try using only one mic, but in a figure-8 pattern. Which brings up another of Mr. Burrough's points — never use two microphones when one will do the job better.

Consider another case of two microphones feeding the same track. Providing they are some distance from each other, yet fairly close to their respective sound sources, an out-of-phase connection may help minimize leakage, yet not affect the primary sounds. The leakage, arriving from relatively far away, is substantially the same at both microphones and is therefore cancelled. The primary sounds, originating just in front of their respective mics, are largely unaffected by the phase reversal.

Two dissimilar microphones, wired out-of-phase and placed close to each other will yield a severely equalized sound that might be useful for a special effect. The equalization will be a function of the difference between the frequncy responses of two microphones as they are combined.

That's just three examples of what can be happening between two microphones. Have you taken all three of them into account every time you do a session? Probably not, if you're like most others.

Lou is also a little suspicious of gobos, or flats. While intended to give some additional separation between instruments, they often do little or no good. Putting a gobo between, say, an acoustic guitar and the other instruments in the studio may do more harm than good. If a cardioid microphone is being used, the off-axis rejection of the mic itself is more effective than any flat. Chances are, the leakage you hear is getting into the front of the microphone, via reflections from surrounding surfaces. If you must use a gobo, try putting it behind the guitar player. You may be surprised to find out it cuts down more leakage there than it did up front. In other words, think about what you are doing. Obviously since the microphone has even less brains than your least favorite producer, it may need your help in discriminating between sounds arriving from in front. Since off-axis rejection has been built in, you get it whether you want it or not. As you know, this means the musician should be standing in front of the mic, and should not move around much. It also means that you probably don't need flats off-axis. Think about it.

For musicians who *must* move around, an omni-directional microphone may be best, particularly for singers. Unless the singer is sprawled across the floor (it happens) the mic-to-source distance is continually changing. At least with an omni pattern, you can eliminate the varying response due to proximity effect or singing into the mic off-axis. Working the omni mic in a little closer will cut down on the leakage. Even in an isolation booth, it may be a good idea to work an omni mic in close. Often, the trouble with a singer in a booth is that she sounds just like a singer in a booth. The isolation booth replaces the sound of leakage with the sound of being in a booth. Moving in close may help get rid of that "music-in-a-casket" sound.

If Mr. Burroughs puts a fraction of his good advice into his book, it will be a must for everyone who uses microphones. In the meantime, try to get to one of his "medicine shows."

A few weeks after Lou's talk, the New York section of the A.E.S. invites Andrew Brakhan of AKG and Peter Giddings of the Beyer Division of Revox to discuss new developments in condenser and ribbon microphones, and to participate in a subjective comparison of the three well-known types of mics: condenser, ribbon, and dynamic. As moderator of this meeting, I thought it would be interesting to find out just how readily identifiable are the much-discussed sounds of the various categories of microphones.

Before the meeting, a tape was prepared of several instruments. In front of each instrument, six microphones were set up. Some were ribbons, others condensers, still others dynamics. Each mic was recorded on a separate track. Later on at the meeting, the audience was invited to identify the individual microphones by type. Hardly what one would call an objective scientific experiment.

However, we all know by now that the published specs on any mic tell only part of the story. We've heard about the *condenser sound*, the *warmth* of a ribbon, transient response, plus a variety of subjective terms by which most people describe their favorite microphones. Much of this information is just not transferable into scientifically measurable parameters.

So, we thought we'd see just how much of the subjective could be recorded and transmitted to a large audience. Judging by the comments, listeners found it far easier to attach a subjective value (*warm, crisp, mellow, bright*) than to accurately identify the particular type of mic. Some concluded that the mellow sounds were produced by ribbons and the crisp mics were condensers. There were perhaps 225 people in the audience, and at no time was there a clear majority accurately identifying the various types.

Most experiments are supposed to have a conclusion, and even this one, despite its lack of scientific pretension, had one. It was—Don't come to any conclusions without listening First.

Anyone who thinks the ribbon microphone is dead should call the Beyer people fast. The old RCA 44 was (and is) a superb mic, despite its fragility and size. However, it is not the last thing that has happened in ribbon missing out on something.

The case for condensers hardly needs pleading here. Most studios have at least a few different types around and are familiar with their advantages. The AKG 451 series used in our tests is but one of many excellent condensers currently available. How it will sound in your application can only be answered by you, the user. However, for those who are considering a condenser purchase, I can unhesitatingly recommend it on several counts having nothing to do with its electrical specs: it is quite small, and an impressive collection of accessories, such as extension tubes and right angle swivel joints are available, as are interchangeable capsules. It's worth a listen.

* * *

On my way out to the Midwest Acoustics Conference last month (more about this in the future) I stopped in at the Electro-Voice factory in Buchanan, Michigan. Electro-Voice's Alan Watson is doing some interesting work which he describes in a paper at the 42nd AES Convention. The paper is entitled, "Time Average Holography." To quote the convention program, "Basic relationships between diaphragm vibration patterns and response variations are reviewed". In another paper, Thomas Lininger, also of Electro-Voice, discusses "Microphone Transient Response Measurement." Again quoting the program, "Transient response in relation to acceptance of a microphone has been observed and an attempt at correlating these observations with analysis of human ear transient response was made." Perhaps in time we shall have it on paper why we like certain microphones. In any case, interested readers are directed to the Audio Engineering Society for further information. ■

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Norman H. Crowhurst

THEORY AND PRACTICE

● The other day I received a religious publication, in which the editor has an article on situation ethics. He points out that, while in general it would be wrong to break into your neighbor's house, if a situation occurred whereby you could save his life only by breaking into his house, then it would be wrong not to break in.

Reading this made me realize how given people are to one or another form of absolutism. And we in audio are by no means exempt from this. How many articles have been written about crossovers and phase problems? This is still an area where theory and practice are poorly related, by many. I read an article on this subject recently, in which the author called attention to the need for optimizing a network with the actual loudspeaker impedances, rather than using values based on a hypothetical resistance load.

I think that was the first time I had read that, when I was not proof reading one of my own articles. So I was pleased to see that the author gave me credit in the bibliography he appended to the article. Now comes a letter from a reader, asking about the use of electronic crossovers in conjunction with mixing. What he has in mind is the use of a single channel to handle frequencies below 100 Hz, with the usual separated stereo above that frequency.

He wants to know if the below 100 Hz outputs from two stereo channels of electronic crossovers can be mixed into a single channel for below 100 Hz. He adds that much has been said and indicated by so-called authorities that mixing out of such a source can be woefully derogatory of signal, its total output, its phasing with the upper channels, and phase cancelling within the mixed section.

He does not name his authorities. I do not recall reading that kind of thing anywhere, but that does not say it has not been printed. It could be that some fairly technicaly informed salesman started it, because you can sell more bass units that way. But here was a case where so many variables are involved that no simple answer is possible. However, I would have to take issue with the basic import of his prior information.

First let's take the points mentioned. The problem of getting phasing correct between each upper frequency stereo output and the common bass unit. In a multiway system on a single channel, the usual practice is to have the units close together and phase them so that frequencies in the vicinity of crossover —in this case 100 Hz—emerge in phase. You can use either theory or practice or a combination of both, to achieve this result.

Now, the argument seems to be, if you separate the bass unit from each upper-range unit or units, then this emergence-together feature is no longer possible. That's true, until you realize that the crossover frequency represents a wavelength of 11 feet. Presumably one reason for wanting to combine the bass is limitation of space—it is for use in a small room, such as an apartment. So how far apart will the speaker units be?

Probably 4 or 5 feet at most. This is fairly good proximity, when you are talking about a crossover at this frequency. In a small room, if you had full range units for both stereo channels, and they were connected out of phase, or so the frequencies below 100 Hz get reproduced out of phase, the error would show in two ways. Program intended to come from center, and thus having equal signal from both channels, would suffer from what has been called the *dissociation effect* —it gets lost and sounds more as reverberation than original source sound.

Also, due to the fact that the room is relatively small, there will be a loss of bass frequencies, below 100 Hz particularly. This is because one unit will be blowing when the other is sucking, producing little or no resultant sound pressure in the room at these low frequencies. Put them in phase, and the bass is good.

So the phase cancelling that can occur within the mixed section can also occur in the room where separate channels are used, perhaps not quite to the same degree, but enough to deteriorate performance quite seriously.

One wonders whether these so-called authorities were around when CBS astounded the world with their demonstration of the "isophonic" principle. This used a system in which the bass, below 250 Hz (so it must work for below 100 Hz) came from a concealed bass unit, under the davenport (or whatever you call that piece of furniture locally), while left and right stereo units of quite small size handle the rest of the range. Nobody could tell that the string bass did not come from where it was supposed to come from.

This is because, as earlier experiments had demonstrated, our sense of direction is based on the initial parts of sounds—initial transients. Our sense of location was based (and still is, as far as I know) on the pluck or bow sounds at the beginning, and our hearing could not unfool us about where the rest of note came from. So, especially for small rooms, such a system works fine, and saves quite a bit of space.

There is another factor here, also related to room size. Someone may object that it is possible to locate low frequency sounds aurally, and quote the fog horn as an example. But one detects the direction of a fog horn in a location out over the open sea, with no walls to create reflections, much less to contain the low-frequency sound. When we listen in any confined space, such as a room, our hearing faculty, which includes the interpretive faculty of the brain, as well as a pair of ears, registers the characteristics of the room in which we hear the sounds.

Generally, this faculty gets us to ignore the room characteristics, while concentrating on the content of the sound that comes with it. So the low frequencies are "mussed up" by the room, and our hearing faculty goes to work on information from the other frequencies, while appreciating the bass note information *en masse*. Outdoors, it may be different, but this condition is imposed on us by listening in a relatively small space.

There is one possibility we should mention, related to program content, which can never be ignored. For what we may term normal stereo, where everything is left, right, or somewhere in between—no problem. Frequencies below 100 Hz are either in phase or so close to in phase that the resultant will not be materially affected (on a dB scale) by electrically mixing. But some recordings use another device to get special effects. They put some components of program deliberately out of phase, on the two channels, to use a dissociation effect. If these parts happen to include bass elements, then these will get cancelled.

But this is not such a serious loss as it first appears, especially if you are listening in a small room, as we pointed out earlier, acoustic cancellation will occur at these frequencies, even where separate channels are used. So, either way, when the recording does this, you are going to suffer a bass loss, if you listen in a small room.

That about covers the thing from a system point of view. Now a little attention to the nuts and bolts. How do you connect the two below-100-Hz outputs together—just parallel them? We would suggest that, as protection against improper operation of the electronic crossover units should any component frequencies from different channels want to cancel one another, each ouput from the crossover units be fed through a series resistor to the common-channel input for below 100 Hz.

The value of this resistor could be about one fifth of the nominal impedance of the circuit. If the circuit is high impedance, meaning 100 k, then a resistor of the order of 20 k in each lead will serve. It is not critical.

As with any stereo system, the whole system should be correctly phased. This may be a little more difficult to do when the units are not all close together. Getting each channel's multiway units correctly in phase within that channel is one thing, and phasing left and right is something else, in a normal system, with two full-range channels. The common-bass system involves an element that fits neither of those jobs exactly.

Probably the best way to check for correct phasing between the common-bass channel and the separate higher-range channels, is to put all the speakers in close proximity, either before or after you check for proper stereo effect over the upper range, with the speakers separated. ■

moving

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NEW PRODUCTS AND SERVICES

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Mfr: International Technical Industries.
Circle 60 on Reader Service Card.

TRANSFORMER

● Owners of medium impedance input tape recorders can now use the model A97A line matching transformer. This permits the direct use of low-impedance microphones. The microphone side of the transformer has a three-pin balanced Cannon xlr-type connector. At the tape recorder, an amphenol screw-type unbalanced connector is used. The transformer is 2½-inches long and ¾-inch diameter.
Mfr: Shure Bros., Inc.
Price: $21.00
Circle 52 on Reader Service Card.

OP-AMPS

● A new series of economical integrated-circuit op-amp components is now available for professional audio. Among the main highlights is an amp/preamp limiter. As part of an input circuit, the limiting facility can automatically prevent overload, eliminating the need for mic-input pads. There are now a total of twenty-four basic processing cards in the series, designated Integra III. In addition to the amp/preamp limiter, the series includes line distribution and mic amplifiers, automation oscillator and filter circuits, tape and phono amplifier-equalizers, summing or mixing amplifiers, and remote stereo gain control circuits. All cards are 2½-inch high for 3½-inch files and come with pc connectors.
Mfr: Fairchild Sound
Price: $30 to $115 depending on circuits.
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ENDLESS-LOOP CASSETTE

● These are continuous-play, endless-loop tape cassettes that are designed to be directly usable in most cassette recorders and players. Accidental activation of a machine's rewind or reverse mode will not damage the cassette. It also contains an automatic brake that is activated when the cassette is out of a machine; it locks the tape in place, preventing fouling. Special, durable tape backing and Super Dynamic magnetic coatings are used. Three lengths are offered: one minute, three minutes, and six minutes. Longer running times are available on special order.
Mfr: TDK
Price: EC-1—$4.75; EC-3—$5.00; EC-6—$5.50
Circle 55 on Reader Service Card.

A Note from the Publisher

We are gratified to offer a very unusual and extremely worthwhile new service to our readers; a service that is a means of achieving a goal we've had from our very beginning.

We stated in our initial editorial in November 1967:

> "In a field as broad as professional audio has become, the dissemination of information necessary to the improvement of performance becomes vital. However, with the exception of the highly respected professional societies, no one has yet come forth to provide such service. **db's** emphasis, therefore will be to complement the theoretical orientation of existing publications with a focus on practical new concepts.
>
> "Our purpose is to establish an intra-industry dialogue —an exchange of ideas—with immediate relevance to everyday applications."

Starting this month, we are offering **db** readers a way to achieve this interchange of ideas that is simple and efficient—a way to build on the ideas in our articles with other involved readers—without the time and expense of travel. It is called **db TeleSessions**—group telephone conferences held in connection with selected **db** articles. You will be able to dial into one or more hour-long, freewheeling discussions in connection with an article you find especially provocative. You can exchange experiences and ideas with a small group of other readers as concerned and involved with that area as you are.

A key ingredient in these **TeleSessions** is the other readers who participate. You will find that each of them is involved and experienced in activities where success often depends on empirical results. This is a chance for you to learn from other experiences instead of from expensive trial and error.

Before launching this joint project with **TeleSession Company** (which has run over 700 sessions), we reviewed the results of prior sessions. Participants were from very different job levels and functions. They found the **TeleSessions** not only stimulating, but a low-cost form of consulting that gave immediate answers.

On the inside of the back cover, you will find a complete description of the way **TeleSessions** work, including the ways in which they are different from conference calls and how *you* can participate.

For example, after reading this month's article AUTOMATING THE AUDIO CONTROL FUNCTION, you can discuss ideas with other readers who are affected by these developments. Your session might explore the ways to use this technology in your particular field, or it might focus upon an exchange of ideas and information about new marketing problems, or how this technique can be adapted to specific needs.

On the other hand, readers most interested in building their own consoles or mixers will be dialing into the **TeleSession** based on ideas within the article A STEREO CONSOLE YOU CAN BUILD.

We have invited the authors as well as other experts to join our readers in **TeleSessions** and become involved in the discussions. These are just a few examples of the content of your **db TeleSessions**, for of course the actual direction of each discussion depends on the unique questions and contributions each participant offers.

We will select one or more articles or columns from each issue in the coming months to be the basis for **TeleSessions**.

We urge you participate in one or both of these electronic meetings.

Robert Bach
Publisher

RICHARD FACTOR and STEPHEN KATZ

The Digital Audio Delay Line

All electronic control of delay and reverberation is a reality today, with products of several manufacturers already available. The authors examine the function and value of digital systems.

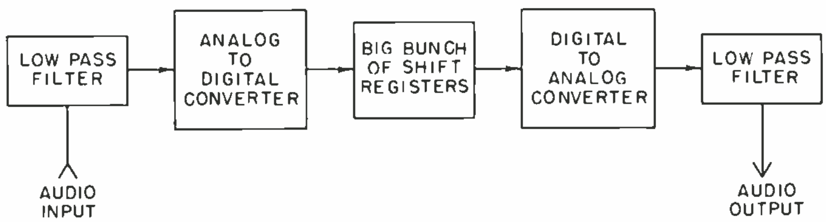

Figure 1. A simplified block diagram of a digital audio delay line.

THE AUDIO PROFESSIONAL recently has become aware of a new audio processing device, the so-called delay line. Various claims are made for its efficacy. It is alleged to multiply strings and voices, make possible greatly improved p.a. setups, and generate more realistic echos.

All of the above claims, and more, are fully justified. In addition to the uses listed above, the delay line has other characteristics which are not so nearly universally applicable, but are nonetheless interesting for occasional use and difficult or impossible to obtain by conventional methods. Devices such as the *comb filter* and effects such as continuous repetition of segments of signals without degradation can be achieved with a suitably configured delay line.

Before discussing in greater detail what can be done with the delay line, let's find out what it is and how it works. The reference[1] goes into greater detail technically, so I'll just draw a few pictures. FIGURE 1 is the greatly simplified block diagram of a digital delay line. The diagram is purely functional—it leaves out such essentials as sample/hold circuits, power supplies, and flashing lights. Following the flow of the signal, you can see that it is filtered and converted into digital form, and then sent to a big bunch of shift registers. This is where the delay takes place. After the signal has traveled through the shift registers, some tens or hundreds of milliseconds later, it is converted back into audio.

SHIFT REGISTERS

The heart of the delay line is the shift register. The top half of FIGURE 2 shows a three bit shift register built from conventional logic IC's. A logical 1 on the input terminal is transferred to output 1 after one clock pulse.. It is likewise transferred to 2 on the second pulse and to the final output on the third. Thus, it requires three clock periods to transfer information from the input to the output. In a serial-bit delay line, the typical clock period will be about 2-3 microseconds, so to get a multi-millisecond delay, lots of shift registers are required.

In commercial practice, mos ic's are used as shift registers. The mos process allows very compact circuitry and low power drain, thus permitting many bits of delay on one chip. A typical chip will contain 1024 bits. Data is stored in the inherent capacitance of the mos transistors rather than as a d.c. level, so power dissipation is a function of operating speed. Storage requirements are so great that a typical delay line could easily require over 100 of these devices. The bottom half of Figure 2 shows a typical storage cell in a mos shift register.

Since the shift registers are all strung together in series, it is possible to "tap" the chain at any point to control the amount of delay (FIGURE 3). It is also possible to connect

Richard Factor and Stephen Katz are both vice presidents of Eventide Clock Works, Inc. of New York City. The company manufactures a digital audio delay system, as well as other products.

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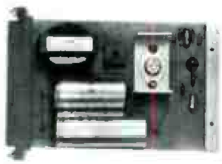
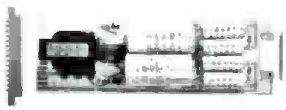

(e)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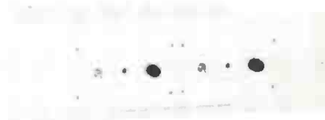
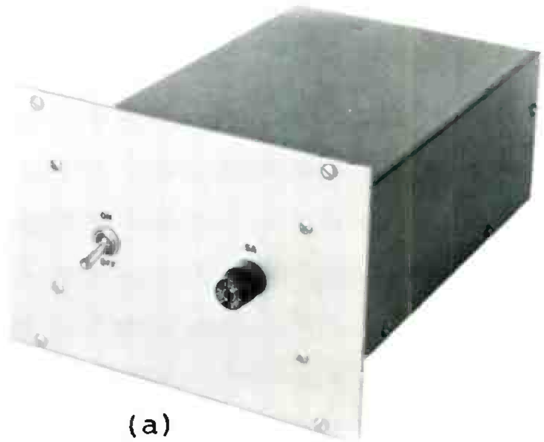

(g)


(d)


(c)


(b)


(a)

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624	(g)	Note : 624 is a combination of 667/II and 667D/II on a common chassis.										2	19x5¼x10"	Rack
667/II	(b)	117	--	12-30 (G)	18	2.0	1.0	0.05	0.3	Yes	2.1	1	6x5¼x10"	662RM
667AA	(a)	117	6.3	18	--	0.35	0.1	0.3	1.0	Yes	0.4	--	6x5¼x11"	662RM
667AA/24	(a)	117	6.3	24	--	0.35	0.1	0.3	1.0	Yes	0.4	--	6x5¼x11"	662RM
667AA/27	(a)	117	6.3	27	--	0.15	0.1	0.1	0.5	Yes	0.16	--	6x5¼x11"	662RM
667B/18	(d)	117	6.3	18	--	0.1	0.1	0.1	1.0	Yes	0.12	--	3x5¼x8-3/4"	662RM
667B/24	(d)	117	6.3	24	--	0.1	0.1	0.1	1.0	Yes	0.12	--	3x5¼x8-3/4"	662RM
667D/II	(c)	117	--	4-6.5 (G)	6.3	3.0	1.0	0.1	2.0	Yes	3.1	1	6x5¼x10"	662RM
667T/15	(b)	117	--	±12-24 (F)	15.0	1.0	0.1	0.1	1.0	Yes	1.1	1	6x5¼x10"	662RM
667T/24	(b)	117	--	±12-24 (F)	24.0	1.0	0.1	0.1	2.0	Yes	1.1	1	6x5¼x10"	662RM
692PS/6.3	(e)	117	--	6.3	--	0.5	0.2	0.07	3.0	--	0.51	--	3½x5" (D)	692SCH or 692RM
692PS/24	(e)	117	--	24	--	0.18	0.2	0.07	0.4	--	0.2	--	3½x5" (D)	692SCH or 692RM
725BPS	(f)	117	--	±15 (E)	--	0.05	0.1	0.5	0.5	--	0.06	--	2½x7½ (D)	725CF or 725SCH

NOTES:

(A) at full load - line variation from 90V to 120V. (B) m V rms. (D) Printed circuit board with gold plated contacts and mating connector. (E) Bi-polar 30 V max. (F) Bi-polar 48 V max. continuously variable. (G) Continuously variable.

ARCHITECTS AND ENGINEERS SPECIFICATIONS: POWER SUPPLY

The power supply shall be a high quality unit, featuring extremely low ripple, short circuit protection, overload protection, remote sensing and compact solid state design. The power supply shall operate from a source voltage of 117 V AC and provide power outputs of __V dc at __amp., __V dc at __amp. and/or __V ac at __amp. (See tabulation above).

The power supply shall have voltage regulation of __% for line voltage variation from __ to __V at full load. Ripple shall be __mV or less. It shall have short circuit protection when load exceeds __amps.

The power supply shall have dimensions of __H x __W x __D. Rack mounting models shall be capable of mounting on standard rack mounting frames. Plug-in models shall be capable of being mounted in appropriate card files or single card holders. Plug-in models shall be equipped with gold-plated contacts and shall include mating connectors. The power supply shall be FAIRCHILD Model ___. (Obtain type number from chart above).

more than one output section to the chain, to produce several signals with differing delays. One problem which arises with this concept is that more steps of delay variation are desired than can be handled with a convenient switching arrangement. If, for instance, 100 ms. of total delay in 1 ms. steps were desired at several outputs, 100 delay line taps and several hundred switch contacts would be needed. A method of avoiding this difficulty is shown in Figure 4. With this scheme, each output section becomes a short delay line which can be switched in small increments, while the long chain, which represents a large portion of the system cost, is shared by all the outputs.

SYSTEM COST

The main criterion of system cost is the amount of delay required. Each system will have one input section and one-to-a-few output sections whose cost will remain fixed regardless of delay length. However, as the available delay increases above about 100 ms, the bulk of the cost will be that of the ic's. Shift registers are manufactured by most of the major semiconductor houses. A typical register will have about one thousand bits of delay and will cost between 0.5 and 1¢ per bit. Add to this printed-circuit board space, driving and power supply circuitry, and the cost increases to about 2¢ per bit. (These prices assume quantity purchase). The num-

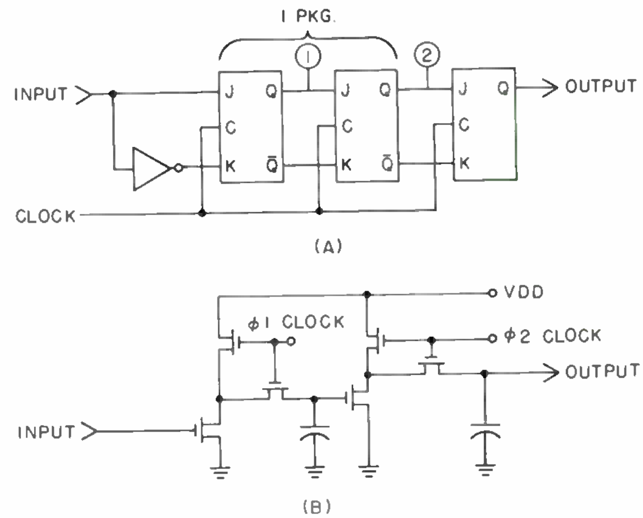


Figure 2. Shift register operation. At (A) is a shift register made from JK flip-flops. Each package provides a two-bit delay. At (B) can be seen a mos dynamic shift register. Data is stored in device capacitance. One chip contains up to 2048 bits of information. This information is from TI bulletin CC402, page 40.

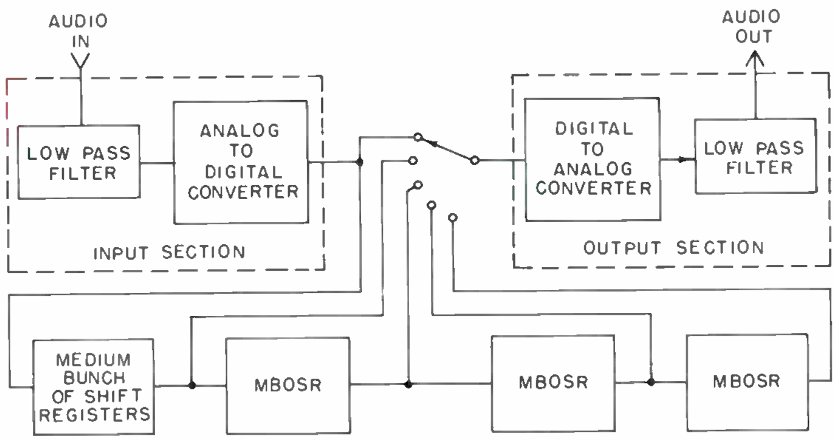
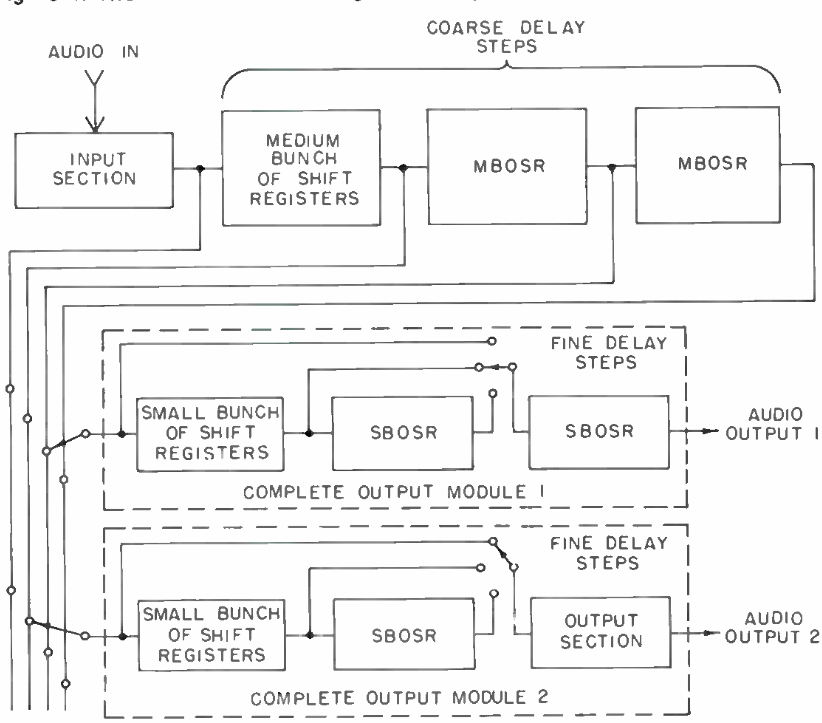


Figure 3. A simplified block diagram of a coarsely tapped delay line.

Figure 4. The methods of obtaining fine delay steps with multiple outputs.



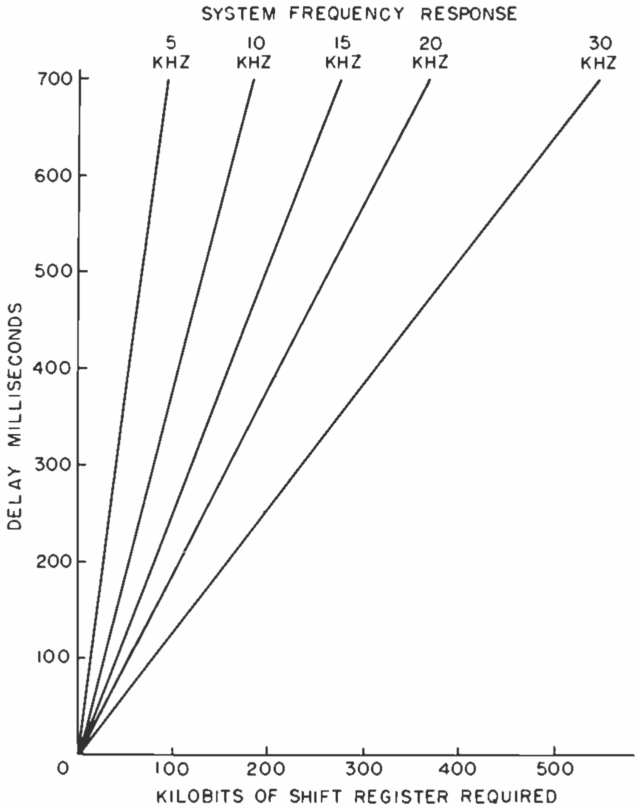


Figure 5. Shift register requirements for given length of delay and system frequency response.

ber of bits required is related mathematically to the system frequency response and the delay length. The graph (FIGURE 5) shows the approximate relationship. When specifying a custom delay line system, it is best to limit the frequency response to a realistic value as cost increases just as rapidly with frequency as it does with delay length.

It is anticipated that the price of shift registers will drop, although not drastically and not rapidly. Until they do. it is unlikely that there will be much application for really long delay lines, such as replacements for the tape-loop rig used by radio stations to avoid broadcasting nasty words on phone-in shows. There is no technical reason why long delay lines can't be built—a unit has been built to simulate communication circuits utilizing multiple satellite "bounces." It has a total delay of 2.5 seconds variable in 1 millisecond steps.

APPLICATIONS

We now have a basic idea of what goes on electronically in a delay line. Let's discuss how we can use what comes out of it. Contemplate if you will, two identical violinists playing identical tunes (on identical violins, of course). Violinist A is standing on a stage 100 feet from you, Violinist B is ten feet further away. Listening to the pair playing, it will not sound like one violinist, despite their outputs being identical. This is due to the differential time of arrival of the two signals (about 10 ms). A delay line can also generate such an effect, saving lots of money on identical violinists. A multiple output delay line attached to a few pan pots can generate a spatially valid string section from one player.

P.A. SYSTEMS

The typical p.a. system consists of an amplifier or group of parallel amplifiers driving speakers focussed over the area to be covered. Depending upon where the listener is standing. he may hear the stage speakers or side or rear speakers more loudly. If, for instance. the side speaker seems loudest, it's a good bet that he is closer to it than to the front. This will give the subjective effect that the sound is in fact originating from the side and the listener may turn to face the loudspeaker. Doing so. he will fail to see the gesticulations of the performer and some of the information will be lost.

Even if attention remains up front, it is artistically dissatisfying to have the sound and the performance separated. Enter the delay line: If the sound piped to the side and rear speakers is suitably delayed. a phenomenon known as the Haas effect comes into play. If a sound arrives from two or more directions, the direction from which it first arrives will appear to be the source. This is true, within limits, even if the later-arriving sound is louder! So, by delaying the reinforced signal, the sound will appear to originate in the front even though most of the amplitude is arriving from the side.

ECHO

Studies have shown that natural room echoes consist of a relatively sharp return some milliseconds after the original impulse due to the initial reflection from the room's dominant surface(s), followed by a more generalized reverberation from multiple secondary signal paths and re-reflections. A

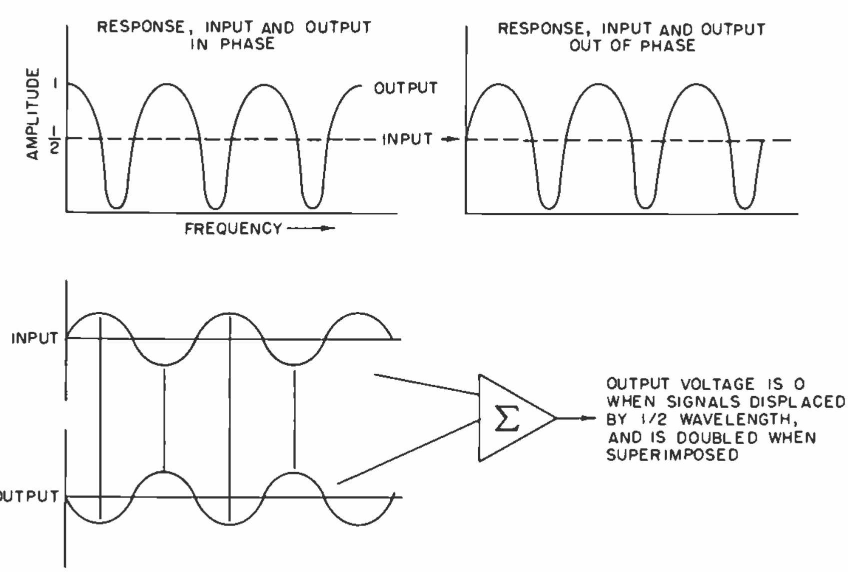


Figure 6. Comb filter synthesis and characteristics.

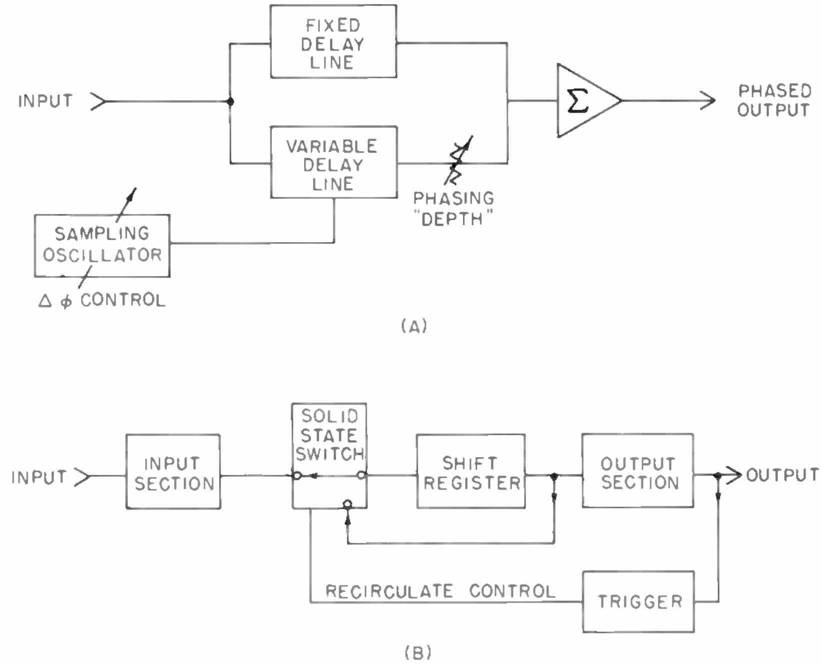


Figure 7. At (A) a delay line setup for phasing. (B) shows signal capture and repetition.

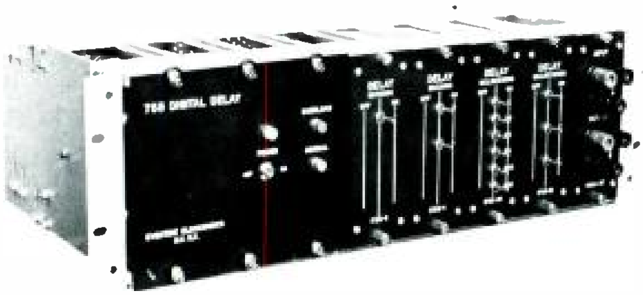
reverb unit synthesizes these generalized reflections, but cannot simulate the initial return. By feeding the reverb unit with a delay line, and summing the delay line output and the reverb output, a more realistic room echo can be obtained.

THE COMB FILTER

A comb filter has a response similar to that of many bandpass (or band reject) filters evenly spaced throughout the frequency range of interest. Such a filter can be synthesized using a delay line. If the delay line input and output are summed together, cancellation of signals whose half-period is equal to the delay setting is accomplished (FIGURE 6). (Signals whose full period is equal to the delay setting are reinforced.) Cancellation will also occur when the delay is equal to an integral number of full periods + ½ period, and reinforcement on whole period multiples. For instance, with a delay of 10 milliseconds, nulls will occur at 50 Hz, 150 Hz, 250 Hz, etc., and peaks at 100 Hz, 200 Hz, 300 Hz, etc. By summing the output out of phase, the reverse occurs: Nulls, 10 Hz, 200 Hz, 300 Hz; Peaks: 50 Hz, 150 Hz, 250 Hz.

Here's an example of what to do with this. Set the delay to 16.66 milliseconds and feed the output back out of phase. Nulls will now occur at 60 Hz, 120 Hz, 180 Hz, etc. If you have a poorly recorded tape with tremendous amounts of hum and hum harmonics, it can be processed through the comb filter and the hum components will be completely removed. Since the remainder of the tape presumably contains no coherent frequencies which remain fixed, the processed tape will be quite usable.

This digital audio delay unit has switchable delays from 1 millisecond to 2.5 seconds.



PSEUDO-STEREO

If a mono signal is fed into a pair of interleaved comb filters (one with in-phase output summing, one out of phase), the program material is divided into two channels, providing some very interesting pseudo-stereo effects. Signals of a non-coherent nature will appear to have a centralized spatial orientation while pitched instruments will appear to wander from side to side as they are played.

Adding together several outputs from different delay taps with similar or different amplitudes can produce a large class of mathematically interesting filters, including comb filters with extremely sharp notches and otherwise flat response. Such filters could be used to realize better interference rejection than the simple example given for 60 Hz rejection above. Full discussion of the mathematics of such filters is beyond the scope of this article.

PHASING

The *phasing* or *flanging* effect can be produced with a pair of equal delay lines, one of which is continuously variable. Summing the outputs of the two lines (FIGURE 7) produces a comb filter in which the combs slide up and down in frequency. The effect produced is identical to that produced using the standard two-tape-recorder-32-patch-cord hookup, but is neater and more readily controllable.

OTHE APPLICATIONS

Certain other uses suggest themselves. One possibility is in the repetition of brief signal segments, either for effect or for spectrum analysis. A signal can be captured in a delay line in the following manner. (FIGURE 7, bottom). A signal is entered into the delay line. As soon as this signal starts to appear at the output, a trigger disconnects the input converter and connects the shift register output back to its input, causing the signal to recirculate indefinitely, without degradation. The signal segment can now be examined by a single swept filter, which is incremented in frequency for each recirculation.

The above compilation of delay line applications is by no means exhaustive. It is a representative sampling of uses for this remarkable new processing tool. ■

REFERENCE

1. Blesser and Lee, *An Audio Delay System Using Digital Technology*. Journal of the Audio Engineering Society, Vol. 19 #5, p. 393.

ELLWOOD W. LIPPENCOTT

A Transmitter Control Panel

Here's an important aid to the third phone operator/ announcer that station engineer management may want to incorporate.

THE THIRD PHONE operator/announcer is with us to stay. When the complete broadcasting plant is under one roof, the transmitter control panel to be described can be invaluable for the third phone man. This is particularly true if the transmitter is out of view of the operator/announcer. There are no frills, no extras (maybe one—the 'scope), a minimum of instruction for correct operation and with full facilities for the third phone man to do his job well and to comply with FCC regulations.

The control panel outlined was designed for an FM stereo station and contains three meters, three lever switches, one push-button switch and an over-modulation indicator light as essentials. The "extra" is a small 'scope. The layout of such a control panel is shown in FIGURE 1.

This panel should be in front of the operator/announcer at all times while he performs his on-the-air duties. The three meters (from left to right) will normally read left channel audio, modulation, and right channel audio. The lever switches directly under the meters have spring return to center or normal. Moving each lever switch to the right or left will connect the associated meter to either of two other desired circuits for momentary reading of transmitter operating parameters.

As an example, the right-hand meter and level switch, when the switch is moved to the left, will read transmitter plate voltage and when moved to the right will read transmitter plate current. When the switch is released it will return to center (normal) and again read right channel audio. Connections for this switch are shown in FIGURE 2.

The center meter is somewhat different. When the switch is in the center position the meter will read over-all modulation (both left and right channel modulation combined with the pilot modulation). Moving the switch to either right or left will read carrier frequency deviation. A carrier deviation meter is a center-zero indicating device and automatically reads carrier deviation at a glance, either plus or minus. Since the meter we are using indicates zero at the extreme left we must have a method of reversing the meter connections to get a plus or minus reading. This is accomplished by connecting the associated lever switch as shown in FIGURE 3.

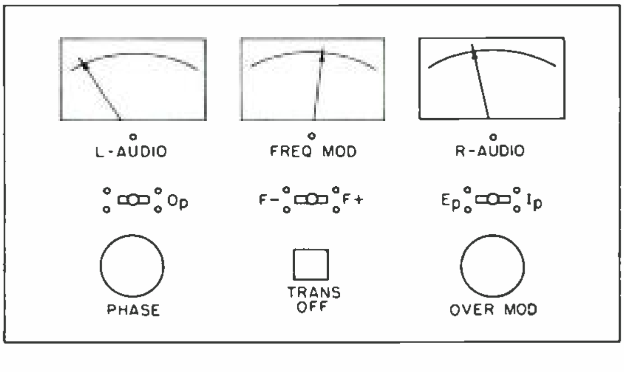
The left hand meter and its associated switch will read left channel audio when the switch is in the center position. Moving the switch to the right will indicate power output or antenna current while moving it to the left will indicate any other desired parameter such as line voltage (primary power), tower lights, etc. Here again we use the circuit configuration of FIGURE 2.

Thus, with the combination of three meters and three lever switches the operator/announcer can monitor a total of eight circuits. The switches used for this purpose are Switchcraft TELEVER 60324, non-locking, with four form C pileups on each side.

A word about meters. My experience has been primarily with a 200 microampere d.c. movement supplied by Hewlett-Packard as an extension of the modulation meter in their 35-B frequency/modulation monitor. This meter was used without change in the circuits as an extension of the modulation indicating meter in the monitor. A diode and resistor in series in each audio channel allowed comparison of the right and left audio channels for proper and equal setting of the levels when switched to right or left.

The right and left meters of the described control panel, when used as audio level meters, do not read the vu or dBm or anything else about the actual audio level in the circuit. Their readings are only an indication of proper

Figure 1. The layout of the control panel.



Ellwood W. Lippencott has written on broadcast engineering topics before. He is based in Riviera, Arizona.

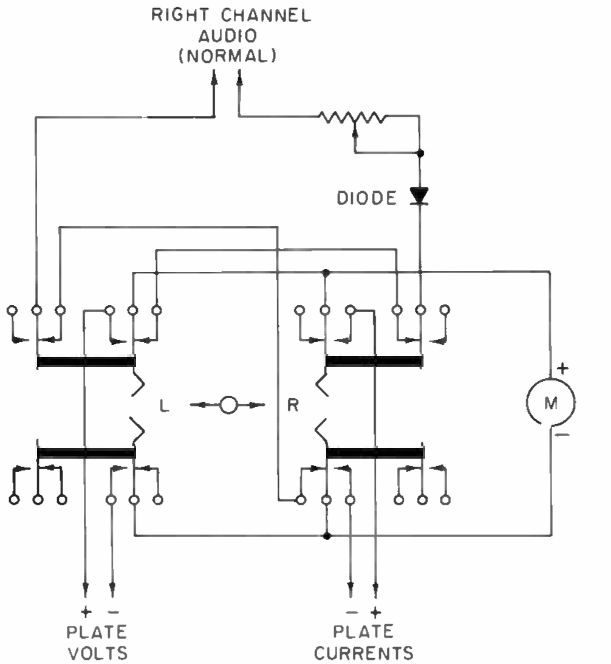


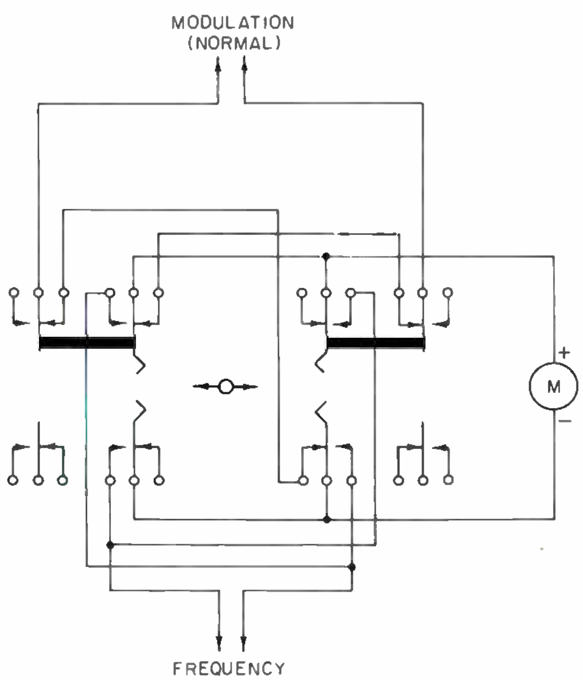
Figure 2. The three-way switch discussed in the text. It chooses between right channel audio, plate volts, and plate current.

level (not actual level) and equal level between the two stereo channels.

In the control panel outlined preference would be to use a meter with a scale reading of 0 to 100 for the right and left meters. A 100 microampere movement would prove most satisfactory. Multiplier resistors will be required to make the meters read the desired parameters correctly.

With such meters the multiplying factor should be a permanent part of the printed transmitter log sheet and the actual reading of the meter should be logged. The sample log sheet as shown in FIGURE 4, should be acceptable to any FCC inspector. If the scale reading is 87 it could

Figure 3. The modulation switch. Centered it reads overall modulation; moving the switch either right or left indicates carrier frequency deviation.



TRANSMITTER LOG
WXYZ

DATE ______ OPERATOR ______ ON ___ OFF ___
SHEET NO. ______ OPERATOR ______ ON ___ OFF ___
CARRIER ON ______ OPERATOR ______ ON ___ OFF ___
CARRIER OFF ______ OPERATOR ______ ON ___ OFF ___

Ep X 100	Ip X .01	OUT X 20KW	FREQ	REMARKS
60	73	20	+85~	

Figure 4. A sample log sheet that should be acceptable to any FCC inspector.

represent 87, 870 or 8,700 volts in the Ep column. In the Ip column it would represent 87 or 870 milliamperes or 8.7 amperes. The power output or antenna current meter would read in watts or RF amperes. A log, reading such as shown in FIGURE 4, would indicate readings of 6,000 plate volts, 730 plate milliamperes and 4 kW output with an operating frequency 85 Hz above the assigned frequency.

The center meter should be an exact duplicate of the meter used in the modulation monitor and purchased from the monitor manufacturer.

There is no need of going into the method of making connections between the transmitter and control panel for monitoring the various circuits. This has been covered many times in the past by several of the technical magazines. Furthermore, today's transmitters are all designed for remote control and in most cases it will only be necessary to run a shielded pair of wires between the transmitter and control panel for each circuit to be monitored.

Below the meters and level switches (left to right) are a small 'scope, a transmitter *off* button switch and an over-modulation flashed light. The 'scope will continuously display the phase relationship between the right and left audio channels. It can be fed from the audio console or from the stereo modulation monitor. The over-modulation light is extended from the modulation monitor.

Some readers may wonder why other *on* and *off* switches and other controls are not included in this control panel. It is my honest opinion that the operator/announcer should go to the transmitter position to put the station on the air or to make power adjustments if and when necessary. To extend other switches and controls to the panel would only add confusion for non-technical personnel. With modern transmitters and the reliability built into them there is very little for the operator/announcer to do that might be called technical—other than read meters and log the readings. If something in the way of a malfunction or a catastrophe exists it is the operator's duty to shut down the equipment and call the engineer. This he can do with only a transmitter *off* switch and a telephone.

To review: the control panel outlined will give a continual, visual display of transmitter over-all modulation, right channel audio level, left channel audio level, phase relationship between right and left audio channels, and an indication of over-modulation (when it occurs) by the flashing light. Momentarily operating the three lever switches to either the right or left position will give all the transmitter operating parameters required for logging as well as one other parameter that may be desired. In case of an emergency or malfunction the third phone operator/announcer can shut the station down as required. Further, any individual can be taught to operate the panel in minutes. ■

WALTER JUNG

Automating the Audio Control Function, Part 2

Last month served as an introduction to the concept that begin to be developed in this installment—which discusses the using of the digitally programmable switch (or dps).

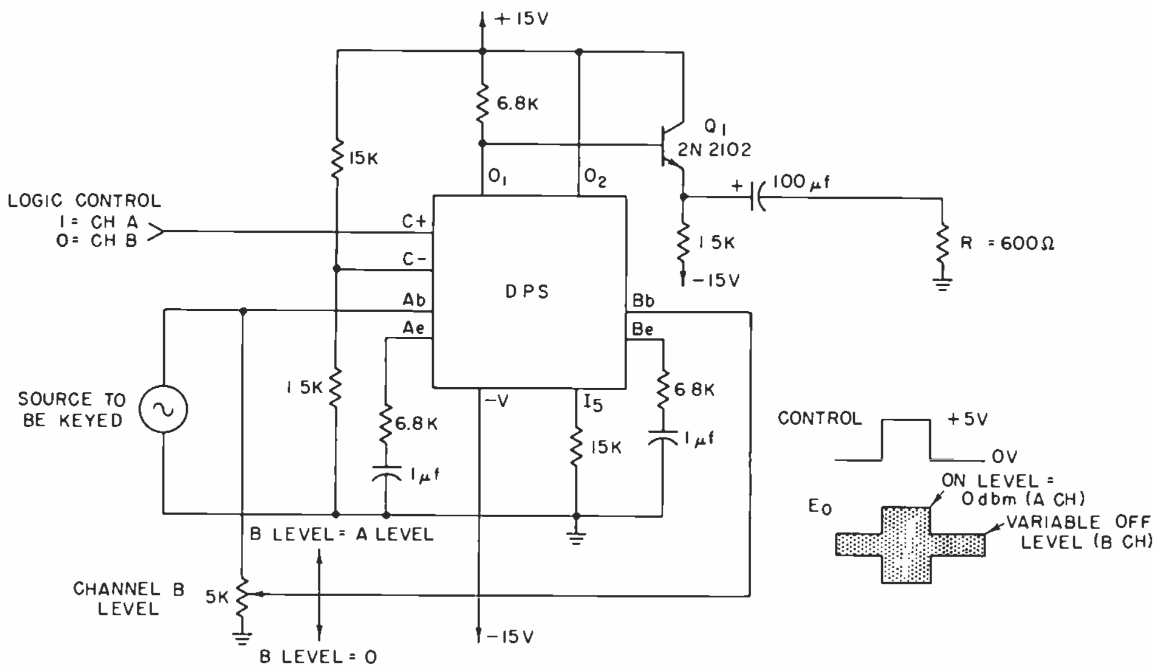


Figure 1. A remote controlled on/off keyer with variable off level, 0 dBm signal handling.

LAST MONTH we developed some *black box* parameters for the monolithic balanced modulator used as a Digitally Programmable Switching element (dps). At this point it might serve us well to reflect on what we have at this juncture before departing into circuitry exploiting this tool.

The dps as introduced last month is a flexible functional switching block useful in a wide variety of audio processing functions. Its inherent structure is equivalent to a dpdt switch. However, since the mechanism is an active one (as opposed to conventional bilateral passive devices), it allows the additional options of insertion gain or loss for the *on* state, and also the choice of inverting or non-inverting phase relation in the signal transmission path. These factors, coupled with the digital programmability of the device can result in some remarkably useful audio gadgetry—which is the subject of what is now to follow. We'll start off the discussion in the most elementary fashion and describe how the dps can be used as a simple, remotely controlled *on/off* audio switch. From this we will progress into more elaborate schemes using a variety of the configurations possible with the device .And along the way we'll make suggestions how the various circuits can be interconnected to form specialized systems.

APPLICATIONS, BASIC

Perhaps the most simple application of the dps is as a

remotely controlled *on/off* keyer. This might be used as switchable test oscillator for console checkout or a time tone generator for on the hour tones or slate countdown. FIGURE 1 illustrates a circuit useful for this. Here a continuous 0 dBm level tone at the dps A_{in} input is gated on whenever a logic one is applied to the control input. In the on state the dps passes the tone with unity voltage gain and a 180 degree phase reversal to O_1 where it is buffered by Q_1, which in turn drives the external 600-ohm load. For a simple on/off keyer the B_{in} input would be grounded, resulting in a direct on or off type of switching. But by making the B channel level a *variable* fraction of the signal input you'll get a 0 dBm level from the A channel when *on* plus a variable level up to 0 dBm from the B channel when it is *on* (FIGURE 3).

You can increase the usefulness of this simple keyer by varying the manner in which you implement its logic control. By driving the logic control input with a variable length timer such as the one in FIGURE 2 you can generate tone bursts of a desired length, either manually by push button S_1 or keyed by an external time base trigger pulse. For instance, 1 second ticks applied to OS from a crystal or 60-Hz line derived (where accuracy permits) timing chain would trigger variable length tone bursts at the output of FIGURE 1. Two possible schemes for these time bases are shown in FIGURE 4.

A GENERAL PURPOSE DPS SWITCH

The relatively simple circuit of FIGURE 1 illustrates the basic idea of switching one of two signals to a common

Current Switching and Summing With the Dps

FIGURE A (A) illustrates the classical op-amp summing amplifier. With S_1 closed the gain of the stage is set by R_f and R_{in} according to the equations shown. For optimum d.c. stability A's plus input is grounded through R_{eq}, the d.c. equivalent resistance of R_f and R_{in}. Due to the inherently low input impedance at the summing junction, as many additional inputs as desired may be connected in a manner similar to R_{in} and S_1 with no interaction. With S_1 an electronically operated ideal switch, these inputs may be remotely controlled.

FIGURE A (B) illustrates this same op amp configuration interfaced with the dps output structure, where Q_1 represents the dps output stage. The dps output circuit is essentially a common base stage or current generator. A d.c. return to V+ (R_1) satisfies the bias requirement of the dps, and C_c is an a.c. coupling capacitor coupling the signal current from the dps into the summing junction. The a.c. signal current from the dps is $\frac{E_{in}}{RE}$, and all of this current flows into the summing junction of amplifier A. Since this input current is essentially the same as that of the classical case, E_{in} and R_{in} of FIGURE A(A), the dps structure satisfies the requirement of FIGURE A (A)'s ideal switch, switching back and forth between channels A and B which generate their respective input currents in A. The output voltage will be of the form shown in the equation.

It also logically follows from FIGURE A (B) that additional dps switches may be connected in parallel with Q_1 without degrading performance as long as R_1 remains high in relation to the summing impedance.

FIGURE B illustrates how the dps current summing concept may be extended further by placing two or more dps's in series. The current output of the first set (Q_1 and any additional switches, connected as in

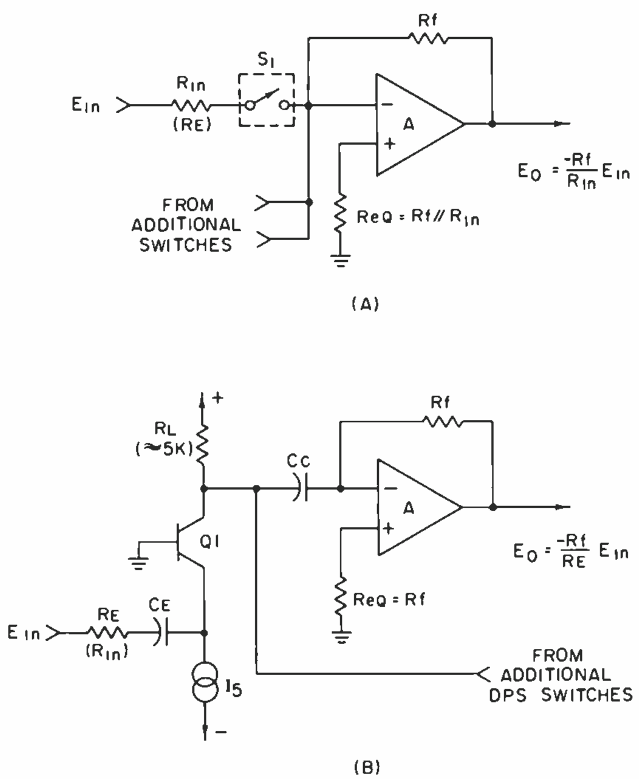


Figure A.

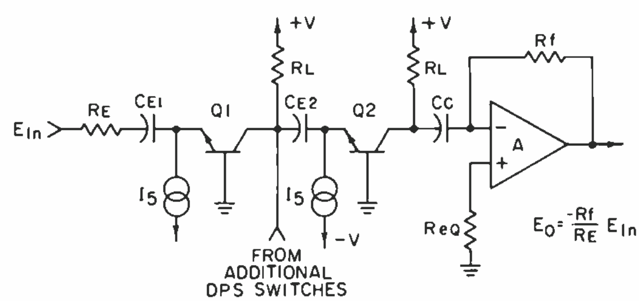


Figure B.

FIGURE A (B)) is fed through a blocking capacitor CE_2 directly into the AE or BE input of a second dps, Q_2. The AE and BE inputs are current summing junctions, similar to the op amp summing node of A. Thus Q_2 performs a function similar to A of FIGURE A (A), receiving the current from the dps switching bank. In the second dps, the composite signal current is switched and applied to A in a manner similar to FIGURE A (B).

With near unity current transfer from one stage to the next, the input current generated by E_{in} and RE appears ultimately at FIGURE A (A) and produces an output signal across R_f, just as was true in the case of FIGURE A (B).

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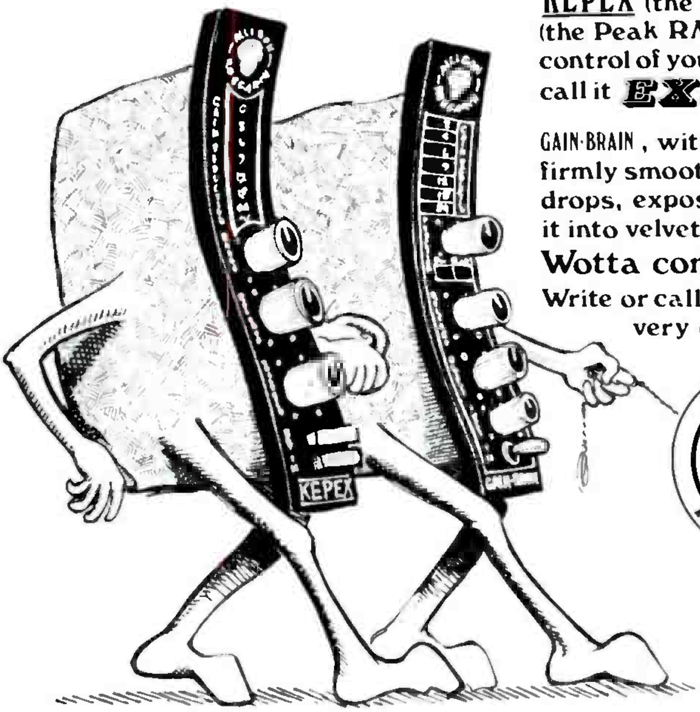

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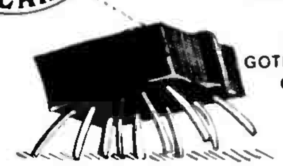

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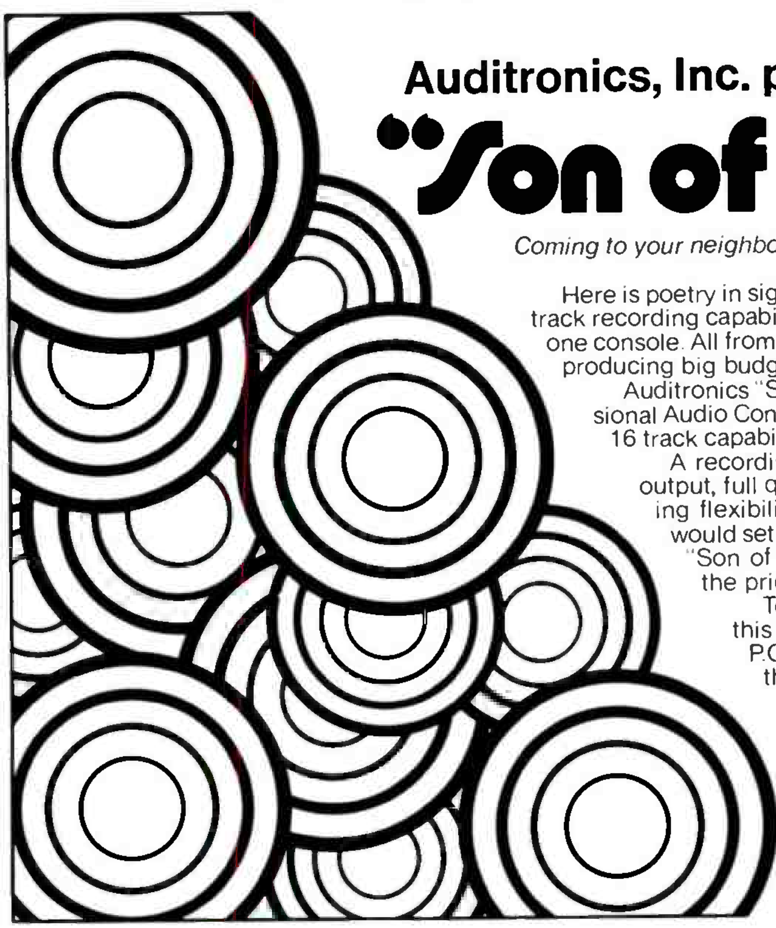

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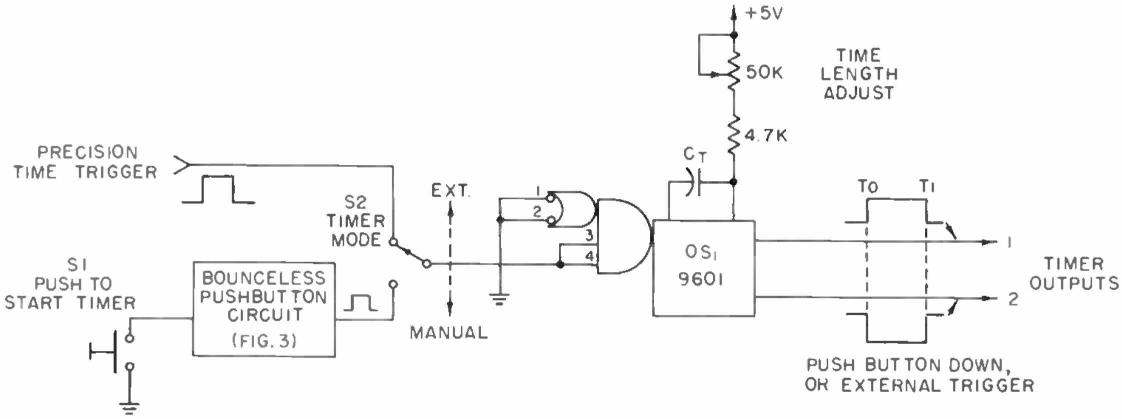


Figure 2. A variable range timer, manual or triggered operation. Notes: O_{s1} = 9601C type one shot multivibrator such as Fairchild's 9601C or Motorola's MC8601P. The time ranges for C_t are as follows: 500 μF—1 to 10 seconds; 50 μF—0.1 to 1 second; 5 μF—10 to 100 ms.; 0.5 μF— 1 to 10 ms.; 0.05 μF—100 to 1000 μsec.; 0.005 μF—10 to 100 μsec.

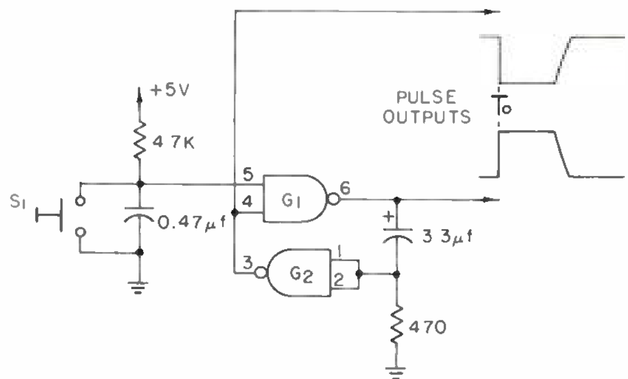
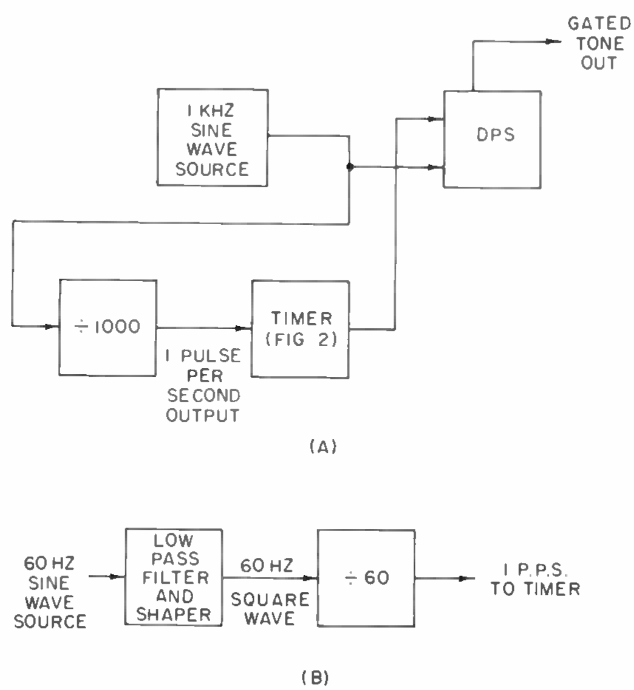


Figure 3. A bounceless pushbutton trigger circuit. Notes: 1. Output is a single pulse each time S_1 is depressed. Negative output pulse width $\cong$ 1.5 ms. Positive pulse width dependent on S_1 down period. 2. S_1 = spst pushbutton. 3. G_1 and G_2 are sections of a 7400 series Quad 2 input Nand gate available as SN7400N from TI; MC7400P from Motorola; N7400A from Signetics; and 9N00/7400 from Fairchild.

Figure 4. (A) A synchronously keyed 1 kHz timer. At (B), 60 Hz power line derived 1 pulse per second trigger source.



output. We can modify this circuit slightly to give it greater dynamic range and thus make it generally more useful. This is FIGURE 5.

In this circuit the dps is used as a common base amplifier at both A and B inputs. The output from O_1 is a.c. coupled into an op amp current-to-voltage converter, A_1.* Since all of the signal nodes (A_e, B_e, and O_1) in this circuit are current summing junctions, there are no appreciable voltage swings on the dps itself, so by virtue of this the dynamic range at both and output is maximized. This circuit can handle a +20 dBm signal at either input, and, given an adequate power output op amp for A_1, supply a + 20dBm signal into a 600-ohm load. All of this with a distortion level of less than 0.1 per cent thd and on-to-off isolation approaching 100 dB! Note also that the input impedance of 20 k is moderately high, even though the common base amplifier is generally thought of as being a low input impedance stage (which of course it is when

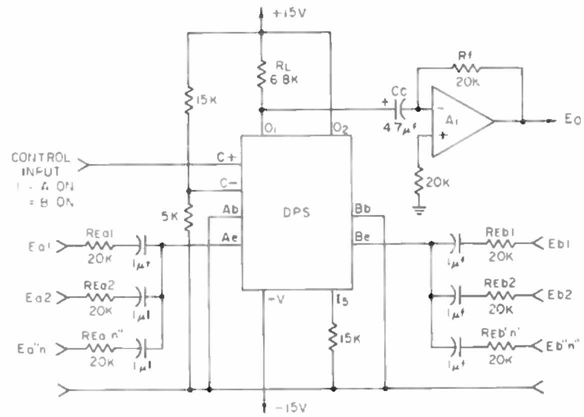


Figure 5. A general purpose common base dps. +20 dBm input and output capability with input summing options. Notes: 1. Gain from any on input to output is R_f/R_e. *2. Input summing optional at both inputs. 3.* $E_o = E_{a1} + E_{a2} + E_{a''n''}$ *or* $E_{b1} + E_{b2} + E_{b''n''}$.

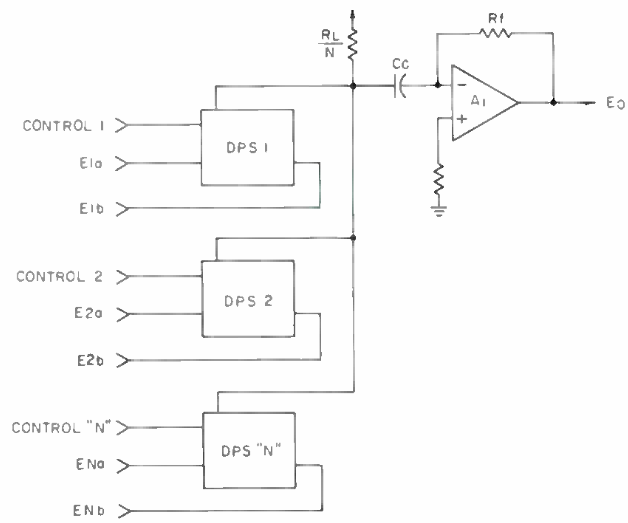
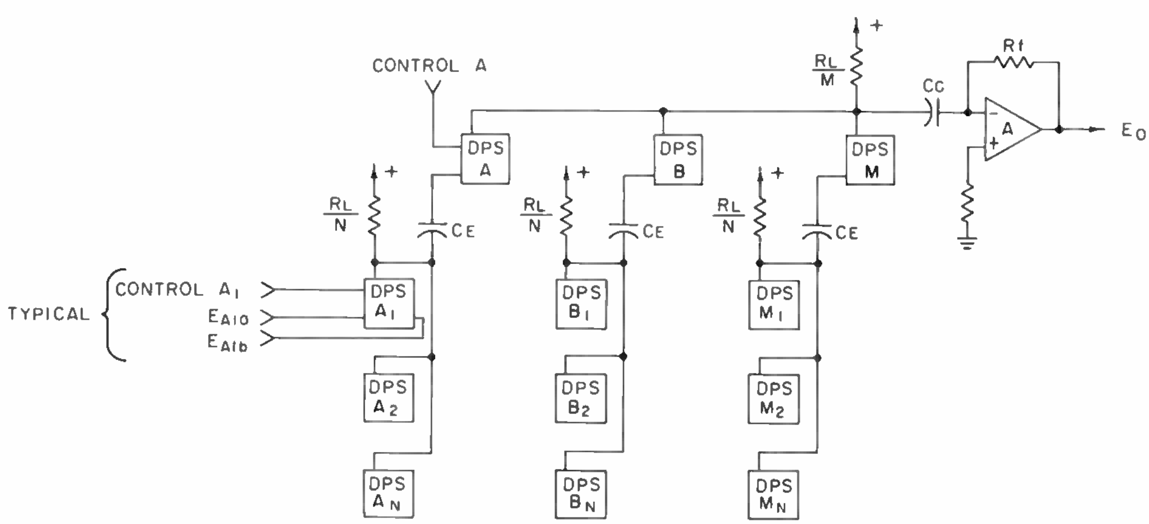


Figure 6. A general purpose summed output dps switching matrix, 1 through "N" inputs. Notes: input modes for the dps stages not shown, may be common base, common emitter, or combinations as required. 2. Gain from any on input to output = R_f/R_e. *3.* $E_o = E_{1a}$ *or* $E_{1b} + E_{2a}$ *or* $E_{2b} + E_{na}$ *or* E_{nb}.

Figure 7. Single master bus dps switching matrix with A through "M" submaster busses each with 1 through "N" inputs. The notes are the same as in Figure 6 except for the formulas. Those are: $E_o = E_a + E_b + E''_m{}''$ *where* $E_a = E_{a1a}$ *or* $E_{a1b} + E_{a2a}$ *or* $E_{a2b} + E_{ana}$ *or* E_{anb} *and* $E_b = E_{b1a}$ *or* $E_{b1b} + E_{b2a}$ *or* $E_{b2b} + E_{bna}$ *or* E_{bnb} *and* $E''_m{}'' = E_{m1a} + E_{m1b} + E_{m2a}$ *or* $E_{m2b} + E_{mna}$ *or* E_{mnb}.



looking directly into the emitter). For the best general all around switching configuration using the dps. this common base configuration is preferred. You can build on it in a variety of ways. One variation for instance is to sum a number of inputs at either A_e or B_e (or both) as discussed last month. Another is to take advantage of the current summing properties at the output of a dps such as is shown in FIGURE 6.

*See boxed section

MATRIX APPLICATIONS

Since the output signal current from an individual dps

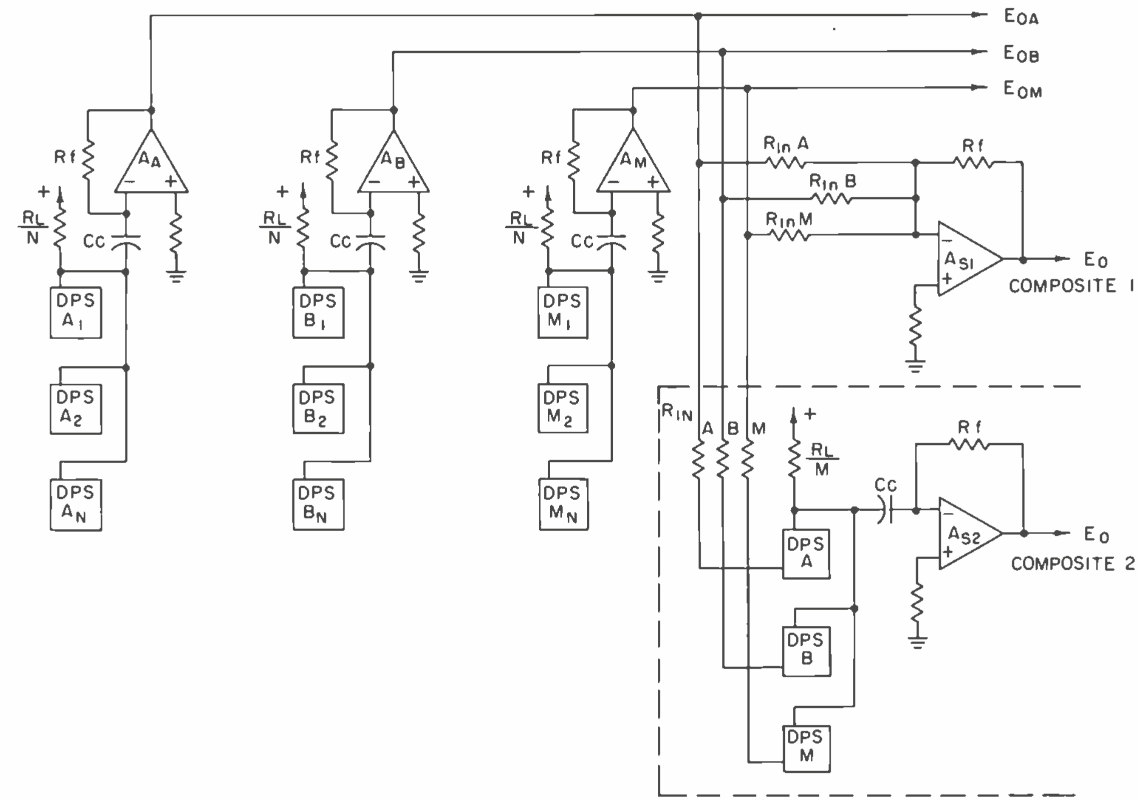
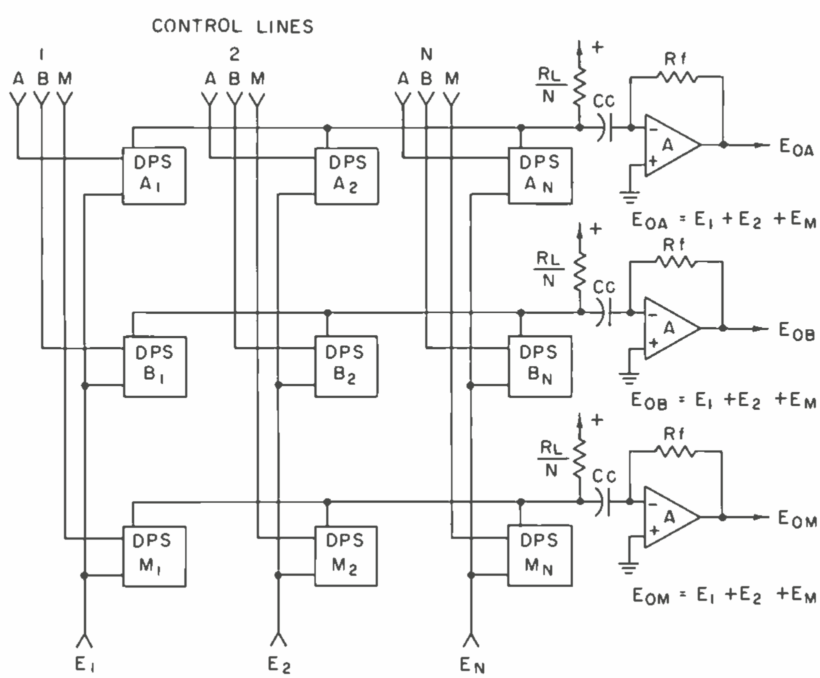


Figure 8. Master and submaster dps. Matrix "M" busses with "N" inputs, with both switched and summed master outputs and submaster outputs. Notes: 1. Individual signal and control inputs not shown for clarity. 2 E_{oa1}, E_{ob} and E_{om} are individual bus outputs after Figure 5. $E_o = E_{oa} + E_{ob} + E_{om}$ where $R_f/R_{in} = 1$. E_o composite$_2$ is a switched composite output after Figure 6.

Figure 9. A dps assignment switcher. 1 through "N" inputs switched to A through "M" output busses.



will all flow directly into the summing junction of the current summing op amp due to its naturally low input impedance, it logically follows that additional stages may be bussed together to form a multiple input switcher. The only requirement is that d.c. load resistor R_L be selected to maintain the d.c. bias on the dps outputs at the proper level. The correct value for R_L is R_L/N, where R_L is the value for a single dps (such as FIGURE 5) and N is the total number of dps stages paralleled. Since only one output op-amp is used, this switcher actually gets less expensive per stage the more stages you add on. You can even intermix input configurations or working gains of the individual channels by making the appropriate component selection at those points.

Now suppose you carry the concept of current summing at the output a step further, and apply it to *another level* of switching using another dps. Here we replace the op-amp input we had in FIGURE 6 with a dps common base input (A_E or B_E), since this also may be used as a summing junction. This gives us a hookup such as FIGURE 7. Here the individual summing busses from dps groups A through "M" are fed to the AE inputs of dps A through "M".

The example shown in FIGURE 7 is arranged as a "M" x "N" input, single output switching matrix. The output will consist of all of the inputs that are on at a single instant, with the gain ratios determined by the individual dps input scale factor(s). The A-"M" submaster banks are switched in groups by master bus dps's A-"M".

To obtain individual bus outputs from dps groups A, B, etc. we modify this switching configuration slightly and add an op-amp to each bus, as in FIGURE 8. Now we have the outputs of busses A, B, etc. available individually as well as simultaneously. A further variation is available in the manner in which the busses can be combined. An op-amp summer such as A_N, can mix the busses with minimal interaction and scale the gains of the submaster busses to the master output if desired. Or, if *switched* bus summing is desired, additional dps elements can be inserted in series with the summing resistors (shown in the boxed section).

Another useful variation in switching matrices is an assignment switcher or patch bay, where any number of input lines are assigned and routed to an additional number of output busses. FIGURE 9 illustrates how this is accomplished. Here input lines 1 through "N" feed a series of dps elements, one dps for each input line and each output bus (A-M). Thus any input line (or lines) may be connected to any (or all) of the output busses, allowing unlimited flexibility in signal routing wth no sacrifice in performance. A side benefit of an all electronic switching system such as this is the dramatic reduction it provides in console panel space required and associated wiring. For "N" inputs and "M" outputs only "NxM" wires from the assignment switcher are required to activate the matrix, while the actual audio wiring can stay in the recesses of the console. And the digitally programmable switching matrix has the obvious benefit of being addressable in computer language, allowing subsequent instantaneous re-programming to duplicate a desired state when recalled from memory.

All of the above matrix applications have utilized the dps in an *on/off* type of switching, where only 1 of the 2 inputs are normally used. Although there are certainly more than a few large-scale switching applications which could utilize both signal inputs, some more common uses are within one stage where a single dps is used to change the state of a signal processing function. In the next installment we'll go into these "smaller scale" dps applications as we look at more circuitry. ■

Martin Dickstein

SOUND WITH IMAGES

Instant Replay

• Ever wonder, while watching a football or baseball game, how some small piece of the action that happened just a few seconds ago can be shown again, and at full speed or still frame or any speed in between? Sure, it's instant replay, but perhaps you might be interested in a few of the details of the system that costs about $100,000 per unit.

Back in 1967, only 5 years ago, Ampex developed this special device that has since been bought by broadcast networks, local stations, commercial production houses, and other establishments that could afford the unit to the total tune of more than $10,000,000. At first thought, it would seem that this company, which is so deeply involved in the tape field, would have come up with a tape unit for the instant replay device, but it turns out that this was not the way to go at all.

One of the capabilities built into tape recorders, video or audio, provides the user with a means to proceed quickly to a portion of the recording anwhere in the tape. This allows playing any part of the tape whether it is ahead or behind the point at which the tape has been stopped. Getting to the spot which is to be replayed, however, is the trick. On a video tape recorder, it may seem like the rewind and fast forward might be really fast, but it takes longer, and is not as accurate as is demanded by the requirements of broadcasting networks or stations where replaying a scene quickly is "instant" replay. This means introducing the replay, cuing up the action desired, playing the scene (at slow, normal or inbetween speeds or with mixed speeds and even freeze-frame) all in a time out or before the next play starts. That has to be "instant," not just quick replay. As good as tape is for many applications, in this case a new method had to be devised.

The process was developed using a magnetic disc. Hi-fi fans know how quickly and easily they can move a pickup head to any part of the disc and how simply they can replay a portion of the record they had just heard. It is this same flexibility which is used in the magnetic disc system with several obvious advantages. Since there is no cutting into the surface of the disc, the same one can be used many, many times without deterioration of signal. Also, the magnetic disc has the advantage of magnetic tape in that there is no processing necessary to regain the information put on the surface, as with an audio disc. Other advantages of the magnetic disc over other media, even magnetic tape, are fewer drop outs than tape, longer life, steadier picture, and the ability to play in reverse at any speed, immediately after recording the desired material.

The disc is made of aluminum and is optically smooth and flat. Then the magnetic material is added. This is a thin layer of a magnetic cobalt alloy which is electroplated on. To protect this surface, a layer of rhodium, a few micro-inches thick, is plated on. The mirror-like surface is now protected against corrosion and permits smooth movement of the magnetic head across the disc.

Before discussing the operation of the disc system, it might be well to see how an instant replay is produced for a broadcast. More than one unit is used when possible in order to allow for complete coverage and flexibility of operation. Each of the record units can be fed from any of the cameras, usually on a preset basis. As the action takes place, the director of the show can select which camera will be recorded by which machine and in most cases, the experienced director can anticipate the course the action will take. A word on the intercommunicating headphone/mouthpiece to the replay unit operator and the repeat is ready to go. Each play is recorded at the different angles and perspectives of the cameras, and with three recorders at his disposal, the good director has the odds with him that the desired action will be there within seconds for repeat performance. (This recording from the cameras in no way has any effect on the camera output singals which are still being fed live to the viewing audience as the action is taking place in real time.)

There are primarily two problems associated with instant replay at slower-than-normal speed. For the image on the screen to be steady during slow-motion replay and for a complete picture to be seen, it is necessary to change only the speed at which the information is fed to the screen, but the scanning rate of lines, fields ,and frames must not be disturbed. In color transmission, it is similarly necessary to retain the original hues and not disturb the color demodulation process in the receiver at home. Thus, the video signal must still be at the normal bandwidth, with the normal scanning rate and interlacing, and containing the proper color components in proper phase and amplitude even though the picture is moving at slower than normal speed.

To see how these problems were overcome, consider first that a frame is made of two separate fields, in a 2-to-1 interlace system, the fields being scanned individually 1/60th of a second apart. The first field is scanned and is reproduced on the screen in alternate lines, leaving the in-between lines for the second scanning to fill in with the next field, thus making up a full frame. Each field can, therefore, be considered to be displaced from the other by ½ line, the displacement of the horizontal sync pulse taking place during the horizontal blanking period. However, considering the vertical sync pulse interval, both fields are the same.

It might seem a simple matter to produce a slow-motion image by repeating each frame a predetermined number of times. For example, if an image (one frame, or two fields) were repeated twice before the next image was projected onto the screen, this would show up as half-speed action. However, this type of reproduction provides very unsatisfactory results. This can be realized when consideration is given to the fact that a moving object is displaced a definite distance in the 1/60th of a second between fields, and the faster the object is moving the greater the displacement. When reproduced in slow motion, the effect would appear to be a double exposure on the picture screen. To avoid this distortion, it is necessary to reproduce each frame made up artificially of two identical fields thus avoiding the displacement of moving objects in adjacent fields. To repeat the same field twice while retaining the normal 2/1 interlace pattern, it is necessary to introduce a half-line displacement into the second scan of the original field. Since the visual information in the two reproduced fields is the same, introduction of the half-line displacement provides a full frame, with proper interlace format, and without the distortion of movement.

With slow-motion reproduction of color, the problem is again related to fact that there are two scanned fields in each frame, but this time it is a matter of proper phasing. In the first frame of the color image, the first field (odd lines) has each line begin with 0-degree phase. The second field (even lines) has each line begin with a chroma phase of 180 degrees. In the second frame, the first field has odd lines begin with a chroma phase of 180 degrees while the even lines of the second field begin with 0 degrees. Notice what happens when the slow-motion image is made up of repeats of just one field of each frame.

The first frame has the first field beginning with a chroma phase of 0-degrees. If this field is repeated (displaced by ½ line, of course), the interlacing lines of the second field of the new first (artificial) frame would also have the second line beginning with 0-degree chroma phase. The requirements of the properly reproduced image are that the adjacent lines be of opposite chroma phase. To accomplish this, a chroma phase inverter is introduced in the electronics of the system.

The inverter operates by removing the chroma information from the video signal, inverting its phase, and recombining it with the luminance portion of the signal. Proper color reproduction requirements are then satisfied.

The chroma inverter is activated during the even fields of the artificial frames of slow-motion replays, along with the half-line delay circuits. However, during freeze-frame operation, the introduction of the delay and inverter systems is slightly different. For still-framing, the first field has no delay and no chroma inversion. The second field has both delay and inversion. The third field (first field of the second frame) consists of a repeat of the first field again. Thus, the inversion must remain active while the delay must be removed. In the fourth field (second field of the second frame) the delay is again introduced but the inversion is removed as this is again a repeat of the same original field. Etc., etc., until the action starts again.

The disc recorder unit, at least the Ampex HS-100, utilizes four discs, 16 inches in diameter, mounted one above the other on a common rotating shaft, spinning at 60 rev./sec. The rotation speed is kept constant by a vertical sync reference signal. One complete revolution of the discs corresponds, therefore, precisely to one image field. Each field begins and ends during the vertical blanking interval.

One head is provided for each surface of each disc. This allows four consecutive fields to be recorded (one by each head). Each head does its own erasing, recording and playing back. Each of the heads is mounted in such a manner as to be capable of being moved radially by a stepping motor independently of the other heads. Thus, during every four-frame operation of the system, each head is doing something independently of the other three.

Following the operation of each head sequence, the first head begins at the outside edge by recording the first field, during the first complete revolution of the discs. During the second field, the first head moves one step toward the center of the disc. In the third field, the first head moves one more step toward the center. For the fourth revolution, the same head goes into the erase mode and wipes whatever might have been recorded previously in that track on the disc. This first head records the fifth field, two tracks away from the first field it recorded. This cycle is repeated for each sequence of five frames.

The second head, during the first field, erases what might have been put down previously in its first track. It then records the second field in this first track. During the third and fourth fields, the second head moves a step each time toward the center of the disc. It is now ready to repeat its previous cycle.

Head number three begins during the first field by moving a step and then erases its previously recorded material during the second field. It records new material in this track during the third field and moves a step inward for the fourth field.

The sequence of the fourth head consists of two steps inward during the first and second fields, erasing during the third field and recording the fourth field. As each track is 7.5 mils wide and each move radially is a distance of 10 mils, it is simple to determine the distance center-to-center of the adjacent recorded tracks of each head, as they move toward the center.

When the heads have moved toward the disc's center and reach their farthest position, a sensing mechanism reverses their direction of motion and the heads continue their sequence while moving toward the outer edge of the discs. Recording is done in those tracks previously left open during the inward movement. The same reversal takes place again when the heads reach their outermost positions.

Playback is accomplished by turning off the erase/record mode and switching to the playback mode as in tape units. Freeze-frame is achieved by arresting the radial movement of the heads at any point. Slow motion is accomplished by a combination of the freeze-frame operation and the normal speed. By reversing the motion of the heads at any time, the action can be shown backwards.

All of this information refers primarily to a rather expensive unit which broadcasters used almost exclusively, but simpler and less expensive units can be used for special effects by video production houses, for slow-motion study by industry and schools, and in many other applications. Build-it-yourself kits for the home? Not yet! ■

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PEOPLE, PLACES, HAPPENINGS

• Make a special note on your calendar for the **Sixth Annual Audio/ Recording Seminar** to be given in **Brigham Young University** in **Provo, Utah.** It's scheduled for June 20 to the 24th, and will feature four days of specialized classes in the campus located at the base of the Rockies. A few of the participants in this year's program will be **Bert Whyte** who will examine the evolution of four channel systems into a coherent whole and chart their future. **Milton T. Putnam** will examine quadriphonic recording hardware and discuss the use of delay devices and demonstrate the value of the Hass and Madison effects. **Jim Cunningham** will discuss the problems of dubbing multi-track to realistic quadriphonic sound.

In addition **Jerry Feree** will present fundamentals of disc mastering as they apply to preparing tapes for mastering houses and discs in quad playback. A special highlight will be a session of control room engineering that will be conducted by **William Robinson.** This will be held in Los Angeles and a chartered jet will take students from Provo to L.A. Write the **Brigham Young University Department of Special Courses and Conferences, 242 HRCB, Provo, Utah 84601.**

• Engineers and musicians in the northeast will be particularly interested in the **Electronic Music Workshop** to be held on June 5th through the 16th in the Boston area. Workshops and lab sessions will be given and there will be studio time available. Contact **Robert Ceely, Boston Experimental Electronicmusic Projects, 35 Elm Street, Brookline, Mass. 02146.**

• Dealers have been appointed to sell and service 3M professional audio recorders/reproducers and accessories to the recording studio market. They will sell and service all 3M brand professional audio recorders and will inventory a wide assortment of parts for rapid servicing of customers. The new dealers are: **Westlake Audio** in Los Angeles, **Daniel Flickenger & Associates** in Hudson Ohio, **Gill Custom House** in Palos Hills, Illinois, **Automated Processes** in Farmingdale, N.Y., and **Telephase Electrosystems** in Memphis, Tennessee.

• **Frederick C. Zimmer** has been appointed to the post of director of engineering for **DuKane Corporation.** He brings to DuKane his 25 years of experience in product and system design for both commercial and aerospace markets. He has held positions at **Control Instrument Corporation, Sperry Gyroscope,** and **Whitehouse Products, Inc.** His most recent position has been that of head of design at the AIL Division of **Cutler-Hammer.**

• **Dictaphone Corporation** has named **Donald King** to be the western regional sales manager for its recording automation group. The group comprised the Scully and Metrotech divisions. He comes to Dictaphone from Custom Fidelity Company of Los Angeles. Prior to this he has had a long career in broadcasting, including assignments as a station manager, and chief engineer.

• **R. A. Moog, Inc.** pioneer in the development of the electronic music synthesizer announces the fact that they have merged with **MuSonics, Inc.** manufacturers of educational-use synthesizers. The new firm will be known as **Moog Music, Inc. William L. Waytena,** formerly chairman of Musonics, Inc. is chairman of the board and chief executive officer and **Robert A. Moog** formerly president of R. A. Moog, Inc. becomes president of the new company. Headquarters and manufacturing facilities of the new company are located in suburban Williamsville, N. Y. just two miles north of the Buffalo International Airport. It is stated that all product lines of the two former companies will make up the broadened product lines of Moog Music, Inc.

• **Tomona Tani,** president of **TEAC Corporation of America** has announced the appointment of **Sachima Tani** to the post of executive vice president and general manager. He brings to TEAC many years of extensive experience with the corporation from its very beginnings. TEAC Corporation of America distributes the TEAC line of consuimer and professional tape and high fidelity products. Mr. S. Tani is the brother of Mr. **K. Tani,** president of TEAC Corporation (Japan) and Mr. T. Tani, president of the U. S. Corp.

• **Peter Horseman** has joined the professional equipment division of **James B. Lansing Sound, Inc.** as an applications engineer. He will be based at JBL's Los Angeles facility and will work closely with audio contractors in developing sound system designs and specifications for commercial installation. Prior to his JBL appointment he spent nine years as a systems engineer with **Hannon Engineering.**

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