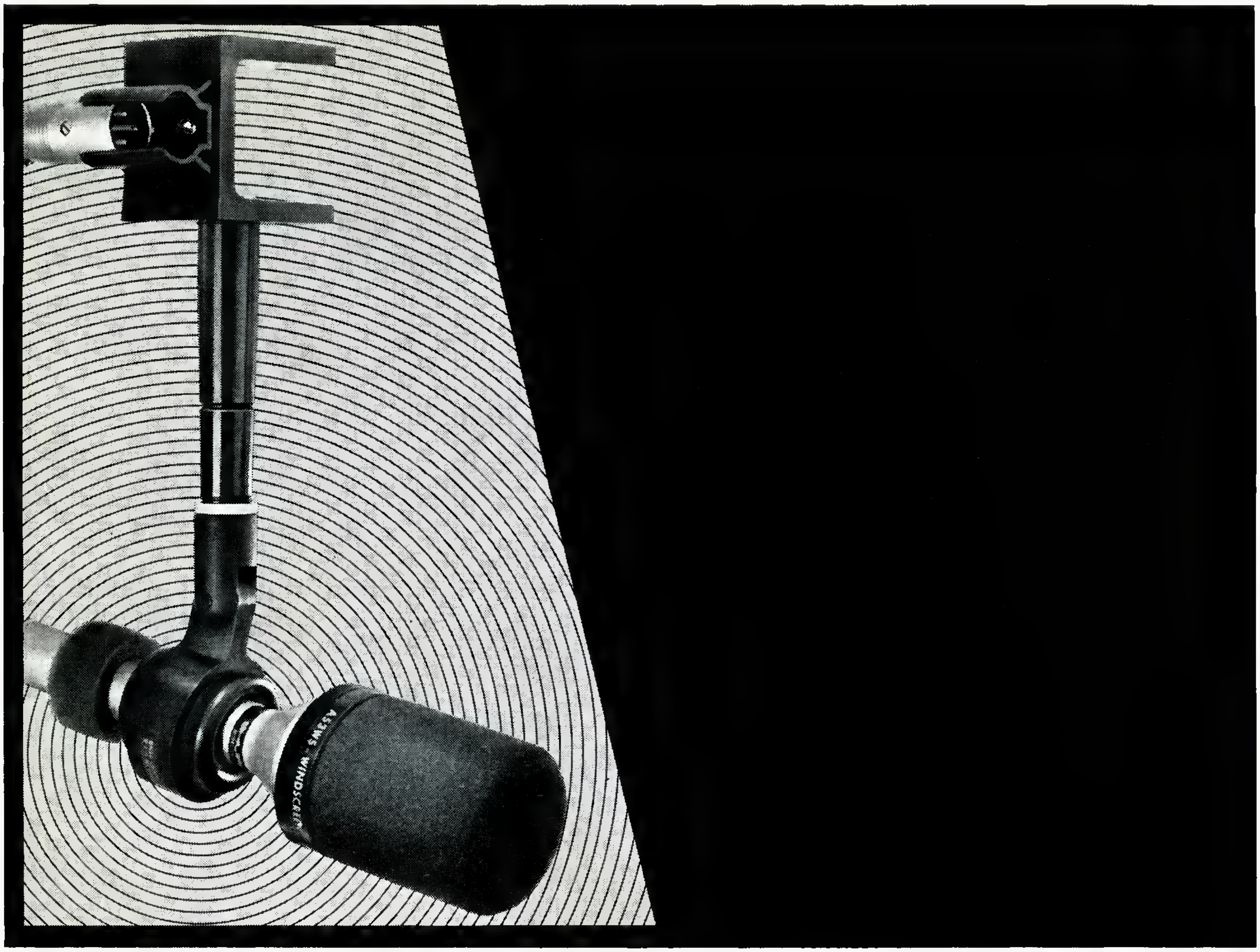
db

THE SOUND ENGINEERING MAGAZINE

AUGUST / SEPTEMBER 1972 $1.00



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SPECIAL NOTICE

● It's easy enough to see that this is a big, big issue. The biggest one we have yet published, in fact. You will also notice that this is a combined August-September issue. We decided to combine the August and September issues into one grand one for two reasons. One is—it is a grand thing to do. But the compelling one was the devastating floods that happened earlier this summer in Harrisburg, Pennsylvania. Harrisburg is the location of our printer. He had up to four feet of water in his press room at the crest of the floods. He might have been back in operation four days later if it were up to him, but the utility company could not turn on the gas supply (vital to press operation) for several days more. The resultant delays forced our June issue to be extremely late and pushed back the issues to follow accordingly. We could not have caught up in time for the AES Convention without this combined issue.

ABOUT THE COVER

● Digital Equipment Corporation is again the supplier of our cover (they were on April's cover). One of their small computors is about to be inserted into a Neve console—highlighting the fact that this issue has several articles dealing with the subject of automation in audio. They begin on page 38.

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Robert Bach PUBLISHER	**Larry Zide** EDITOR
Bob Laurie ART DIRECTOR	**John Woram** ASSOCIATE EDITOR
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letters

The Editor:

Confusing *phase* with *polarity* seems to be a trait among writers on the subject of audio.

Two loudspeakers can have the same *polarity* regardless of their locations. But two loudspeakers located several feet apart can never be "in phase" except for a listener equidistant from the two speakers.

The late W. B. Snow in his monumental paper of 1953[1] wrote "It is good practice to observe a poling convention throughout all channels, including the microphones and loudspeakers, although the channel spacings are so wide that only very low frequencies can be considered at other than random phase in one channel compared to another."

Note the distinction between polarity and phase. The polarity must be 0 or 180 degrees; for a situation where a listener is 10 feet from one loudspeaker, 15 feet from another, and a frequency of 1000 Hz is being radiated from each, the phase shift would be approximately.

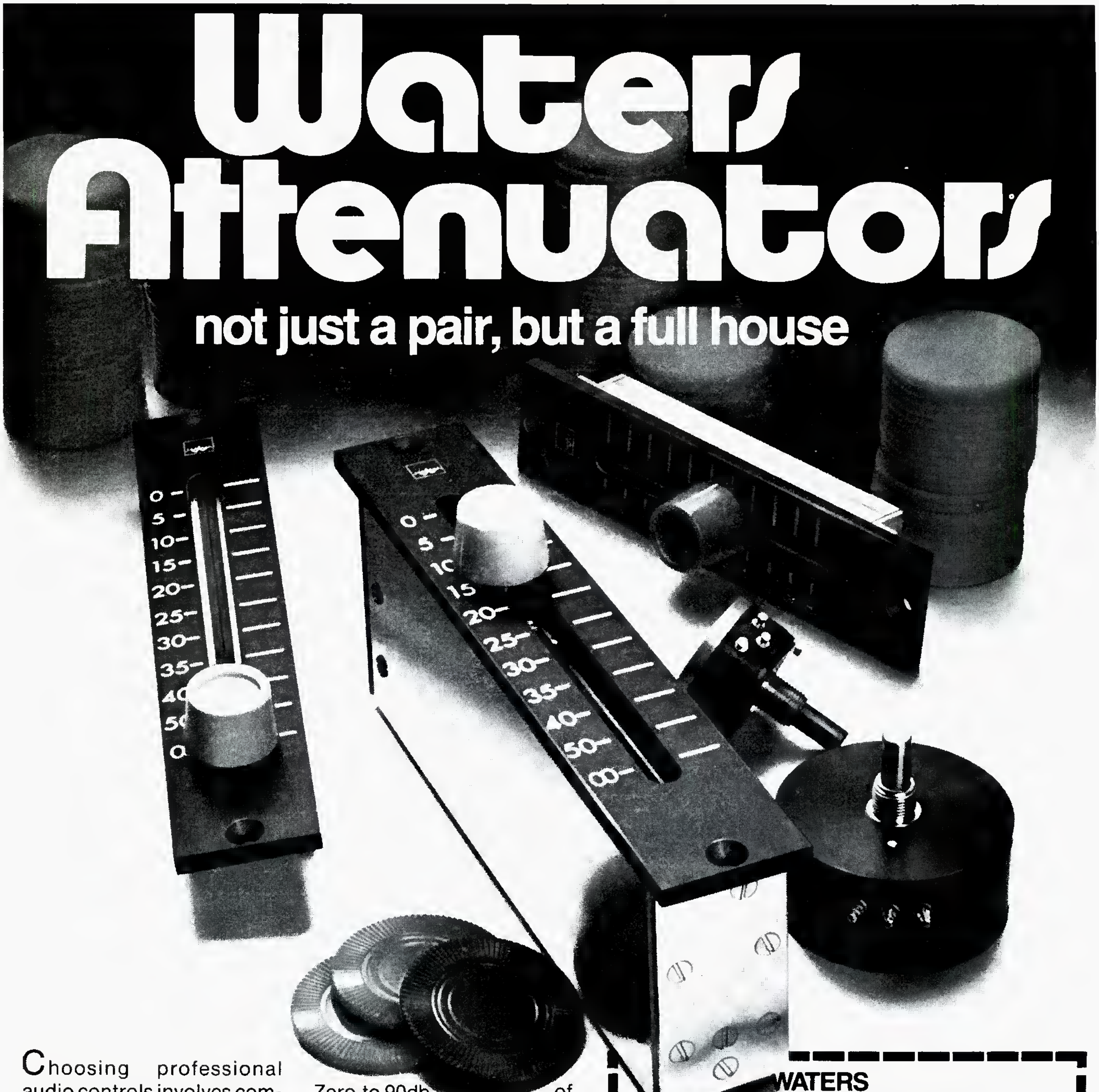
$\phi = 5 \times 1.124 \times 2\pi$ radians or more than 2000 degrees

For very high frequencies, moving one's head a matter of a mere 6 inches would cause a phase shift of more than 1000 degrees.

All this is material with which readers of **db** are well aware, but may need to be reminded of occasionally. And it might be well to be reminded to distrust writers who are careless on this subject. They just might be careless on other subjects also.

This writer's early work in stereo involved recordings made with only two microphones. Polarity didn't seem to be of prime importance since speakers were 25 feet apart. Introduction of a bridged center speaker didn't alter the fact that polarity wasn't critical; the center speaker could accept

1. W. B. Snow, "Basic Principles of Stereophonic Sound", Jour. of Soc. Mot. Pictures and Television Engineers, Vol. 61, No. 11, Nov. 1953, pp 567-589. The quote is on page 582, Col. 2.



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L + R or L − R about equally well. But when a third microphone was used to focus central stage events, thereby adding a strong monophonic component to each stereo channel, polarity of the center speaker had to be L + R. If the L − R polarity was used, the soloist almost disappeared; whatever central stage events were to be heard had to be combined in one's hearing system out of what came from the flanking speakers. And it should be remembered that the sounds reaching one's ear from the flanking speakers are in random phase for all but the lowest frequencies. This means that one can not ignore polarity; Snow's remark still holds; one should maintain proper polarity, even if it is impossible to achieve "correct phase."

But remember L + R or L — R is not phase but *polarity;* one can achieve one or the other with a polarity reversal of one or the other channel. Phase is a general concept that includes all phase angles from 0 to 360 degrees and multiples thereof. Polarity has to do with the special case where precisely 180 degrees is introduced.

Paul W. Klipsch
Klipsch and Associates
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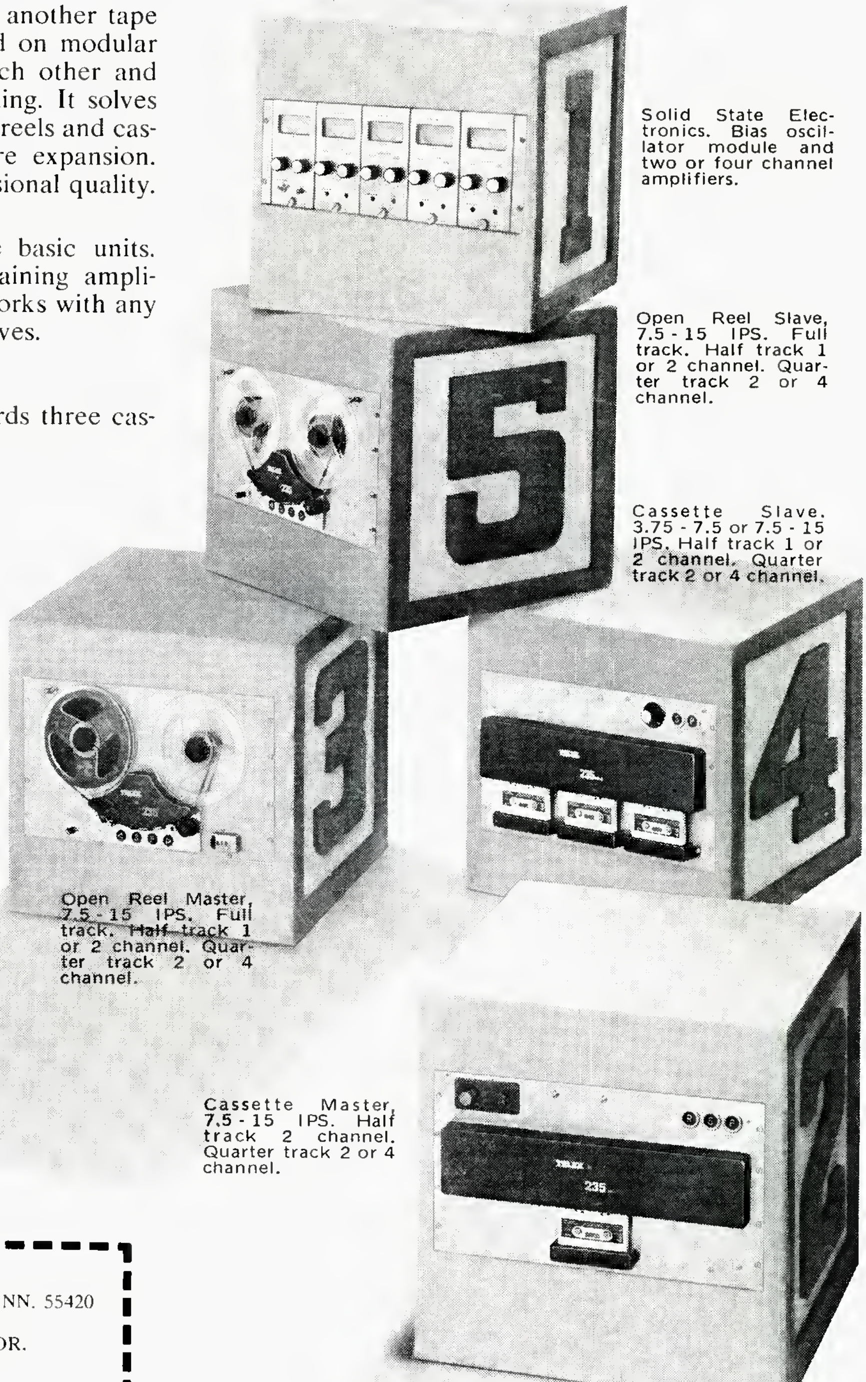

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George Alexandrovich

THE AUDIO ENGINEER'S HANDBOOK

● In our every day work we are used to dealing with audio lines of short length which don't introduce any noticeable ill effects to our signals. Inductance of the wire is low. So is the capacitance to ground and between conductors. D.c. resistance is usually on the order of few ohms and is generally disregarded.

However, in radio and television work audio crews are faced with a problem of transmission over telephone lines—which may be providing an audio path from the local sports arena or from a political convention hall all the way across the country. The telephone company sees to it that signals traveling a long distance are boosted from time to time to compensate for the losses occuring along the way. So-called *repeaters* provide needed amplification. But broadcasting audio signals have to be of wider frequency range than those just for speech. In a long line, high frequencies are attenuated because of capacity between wires and ground. Low frequencies are lost because of series inductance of the wire. Midrange frequencies are affected by resistive losses as well as inductive and capacitive effects to a lesser degree. Response of a typical telephone line has a 6 dB slope with the knee at approximately 2-3 kHz, attenuating 10 kHz more than 10 dB. The low end has similar rolloff characteristics which will vary from line to line—but not so much that we can not help correct it to some degree.

As I said before the telephone company sees to that that deterioration of the signal quality doesn't reach the point where it can not be boosted or restored without too much noise. The telephone company is also willing to lease equalized lines to broadcasters—and even make up the equalization losses. Depending on the amount of equalization required, losses may run as high as 30 dB. However, equalized lines are generally available as unidirectional wires permitting communications in one direction only. But to obtain an equalized line may present some difficulties these days with telephone crews very busy with maintenance work. If you need a good line in a hurry you may as well forget it. The only thing left for you to do is to help yourself. Have an equalizer handy so that it can be inserted into the line any time you need it.

The simplest equalizer for the line consists of rlc resonant circuit adjusted to the frequencies of about 8-9 kHz and connected across the line as shown in FIGURE 1. The resistive part of the circuit adjusts the effect of the circuit on the response of the line. If the r is minimal, the effect is greatest but also the loss introduced by the equalizer is the greatest. If r is made large effect is small. Typical values for the circuit are:

l 200 mH with q of 50 or more
c 2000 pF
r 0-1000 ohms

Because the gain of the line changes with the increased effect of the high-frequency equalizer, it may be advisable to use a double ganged potentiometer so that the second section adjusts the output level as the amount of equalization is varied. See FIGURE 2. Resister R3 has to be adjusted so that gain in the midrange remains constant as the potentiometer is rotated. What this circuit does is simply keep the losses constant. It should be kept in mind that output level from such an equalizer should be fed into the medium level input in the console where additional amplification is available. Normal telephone levels range between −20 to −10 dBm level. If we introduce a loss of another 10-15 dB because of equalization equalizer output level will be more like −30 or −35 dBm. If you are concerned with a permanent line there is no need for adjustable equalization.

But now let us assume that you require to boost low frequencies as

Figure 1. The simplest telephone equalizer.

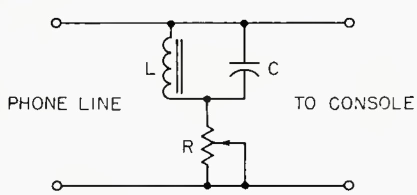
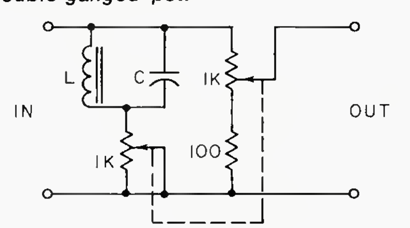


Figure 2. It may be advisable to use a double-ganged pot.



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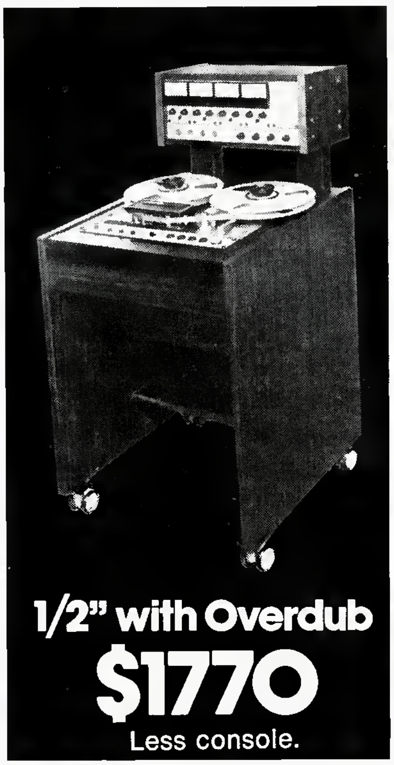

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And if all that seems like an awfully good buy for your money, here's what $1,770 for our Series 70 Recorder / Reproducer buys you:

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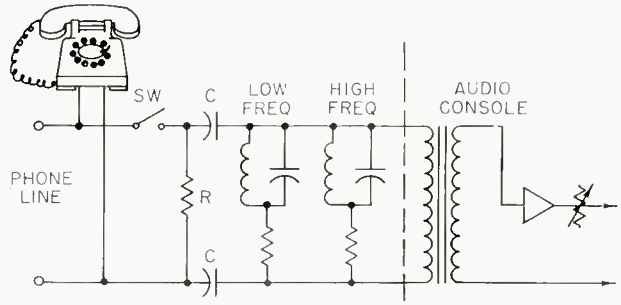


Figure 3. Tap the phone line as shown to feed the phone line into the audio system.

well. Simply erect the same type of a circuit as for high frequencies except with different values. If you want the circuit to resonate at 100 Hz you would need either a 2 henry choke and 1 μF capacitor, or a 1 henry inductor with 2 μF capacitor—or any other suitable lc combination producing desired low-frequency resonance (consult a reactance-frequency chart). Connected across the line in a similar way, it will attenuate all frequencies outside the resonant frequency. Again, resistor in series will limit the effect of the circuit.

Needless to say, a conventional telephone line has d.c. potentional and has to be decoupled by a capacitor—and then a fixed resistor has to be connected across the line to act as a holding circuit for the relay at the telephone exchange. This refers only to the two-way telephone line with dial and ring capabilities. Let us assume that you use a conventional telephone receiver for the incoming line. As the outside call is received, contact is initially established by lifting the head set. In order to feed the signal into your audio system you have to tap the phone line as shown on Figure 3. The disconnect switch is for the purpose of breaking the circuit after the transmission is completed. After the initial contact is established through the use of telephone set this switch is turned on and the receiver is put back on the hook. Then the holding circuit takes over the function of keeping the line open.

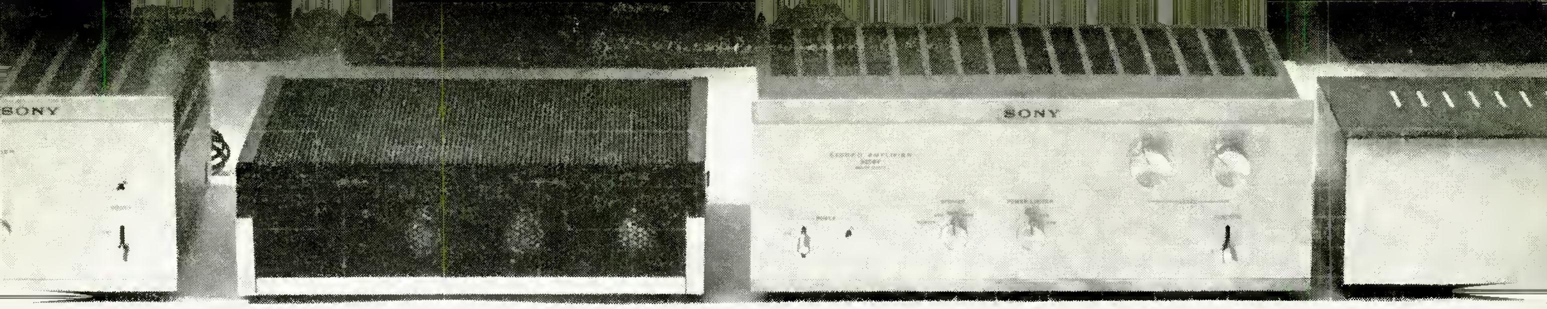
It is quite obvious that very few people in this field of audio have time to build and experiment with their own circuits. Simplest thing is to use available program equalizers for the job. Inserting an already designed unit into the telephone line may prove to be the simplest thing.

Now let us examine the problem of line equalization from the radio station to the transmitter site. Although many stations depend on the telephone company, just as many string their own lines—at times for several miles. These stations should consider themselves lucky. They can pump as much level into the lines as they want to—knowing that signal will not be clipped, as in the case when commercial telephone lines are used. Those stations can and should expect better signal-to-noise ratio at their transmitters, as well as better frequency response.

The transmission line for the signal can be fed from a source impedance of almost zero ohms without a transformer, with ample voltage to insert the equalizer on the other end—and still have enough level to drive a transmitter. Only the telephone company requires a 600-ohm pad on the output of the console. Impedance of the line as measured at the transmitter will consist of impedance of the wires only, therefore losses will be minimal. If the impedance measured is only a couple of hundred of ohms at midrange, load doesn't have to match it—let it be bridging (and balanced, of course).

In this case of a super private line you can pre-emphasize the signal before feeding it into the line—just as it is done in tape recorders or disc recording. Boost the most troublesome frequencies (highs and lows) and then de-emphasize them at the other end if required. This may also hold true for leased lines if there is no appreciable restriction on the level feeding the line. Usually there are transformers in the line which restrict you from feeding high enough levels, also the telephone company is concerned about the crosstalk between adjacent channels or lines. If the signal strength in one pair of wires is much higher than certain acceptable level it could be heard in other lines.

As long as transmission of signals will be done through wires, problems of losses and equalization will be with us. New techniques of signal transmission are emerging, starting with scrambling using digital and analog signal conversion which tend to improve the existing conditions. But even with these techniques response of the line is very important because it limits the transmission rate and its quality. ■



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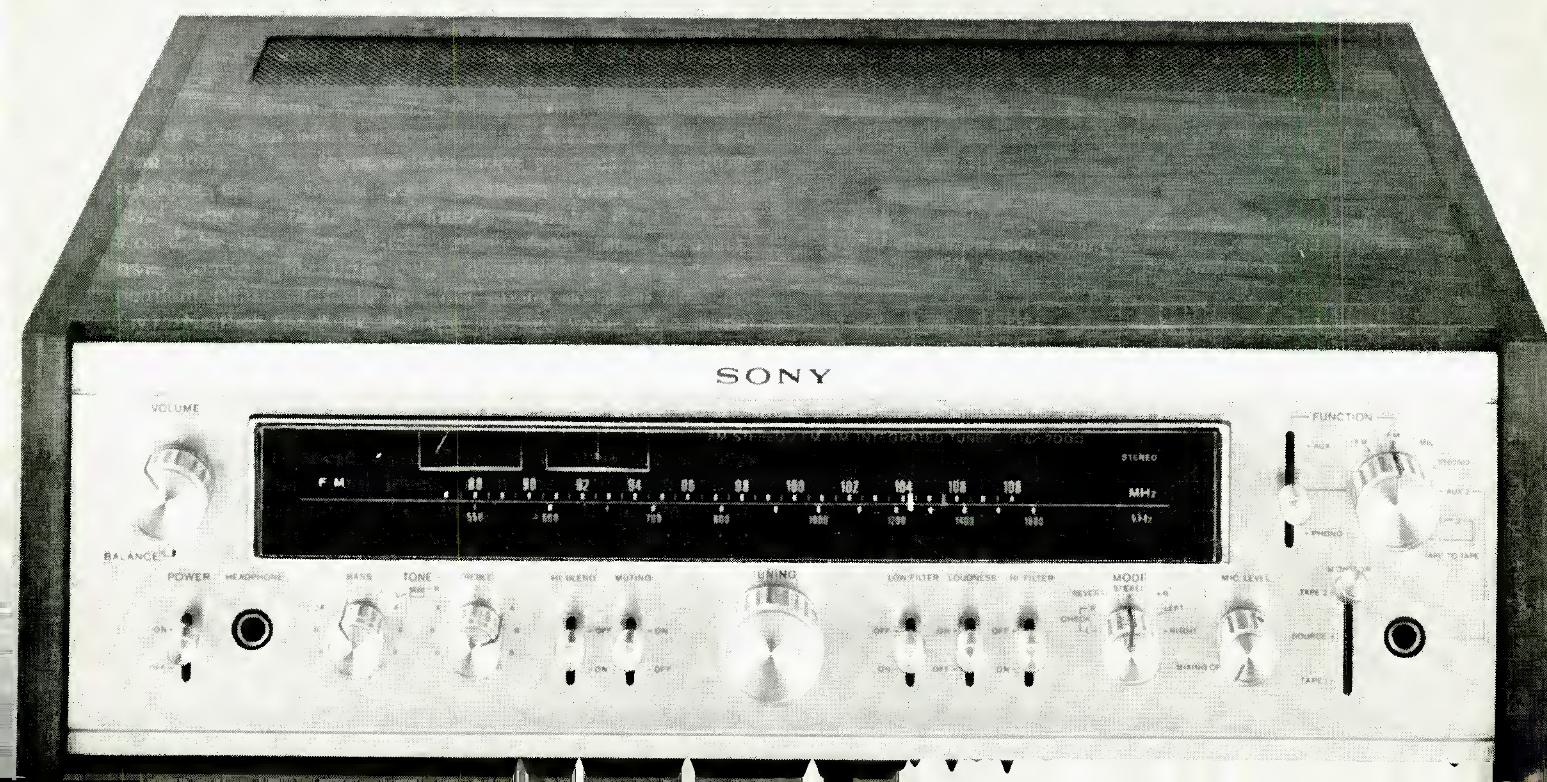


Norman H. Crowhurst

THEORY AND PRACTICE

● Often, in analyzing a situation, there are different ways of looking at the same thing. For example, a composite audio signal can be viewed as a waveform with various squiggles in, through time, or it can be analyzed into a spectrum of audio frequencies, that also vary with time. In typical introductory books on sound, or on music, waveforms will be shown for a violin, a flute, and perhaps some other instruments, with the notion that these show why the instruments sound different.

Unless you know how to interpret waveforms in terms of their frequency content, such pictures do not help much. The uninitiated, for example, would look at the two waveforms in FIGURE 1 and conclude they were quite different when, in fact, they each represent a fundamental with the same percentage of third harmonic and therefore would be indistinguishable from one another in listening.

So, you conclude, the important thing about audio signals is not their precise waveform, but their precise frequency content. To you, as a listener, this is probably true. It expresses what your hearing depends on to give you the sensation that you experience far more accurately than any other statement in such concise terms.

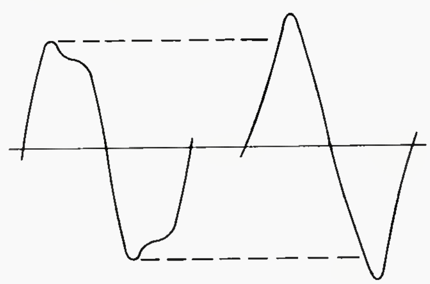
But if you are an audio man, concerned with handling such signals in electronic equipment, the picture changes. That statement is no longer true. For many purposes, an important factor is the peak amplitude of the signal, and how it fits within the handling capability of the system. Thus, although the two waveforms of FIGURE 1 represent the same signal level, with merely a phase shift in the harmonic, relative to the fundamental, their peak levels are different, which means an amplifier or similar piece of equipment might handle them differently.

To a lesser degree, this is true at other points on the waveform than just the peaks. Although most treatments of feedback networks analyze their performance with frequency as a reference—the Nyquist plot and other forms of stability diagram are usually plotted with frequency as the referent, whether or not frequency appears on the chart as such. But the system really handles waveforms on an instant-to-instant basis.

This is why, when some element within the chain runs into a condition such that its parameters change, such as clipping, or crossover distortion, the way feedback behaves is affected at the point on the waveform where this effect occurs, not at some specific frequency, although the frequency of signal being handled at the moment may also affect the result.

This fact explains why feedback can sometimes cause an amplifier to clip more severely than it would without feedback, or with a lesser amount of feedback. The purpose of feedback is to reduce distortion (among other

Figure 1. Although these waveforms look different, they have identical components of fundamental and third harmonic. Their rms values are the same, they would sound the same, in quality and level, but their peak values are quite different.



purposes). So if an amplifier component clips quite positively and nothing will make it deliver more current, voltage, or whatever, feedback will correct the input to try to eliminate the deficiency.

Then it's like the irresistable force (produced by the feedback), meeting the immovable object (the component that clips): catastrophe! The impact is worse than if the clipping was less severe, or the feedback less forceful. This is something that the theory, based on frequency analysis, does not show. Waveform is modified on an instant-to-instant basis.

Here's another example. Suppose that a clipping effect acts as short-circuit to signal, beyond a certain level, and that it is sensitive not to absolute level from a reference zero (FIGURE 2) but to changing signal. Then when signal exceeds a certain level in one direction, the output waveform will suddenly square off beyond that point.

If the squaring off referred to absolute signal, the flattening would cease only when the level of input came back down to the same point. But if the circuit responds to changing signal, as soon as the signal *change* reverses direction, the flattening ceases, which means it hits the opposite clipping point a little sooner.

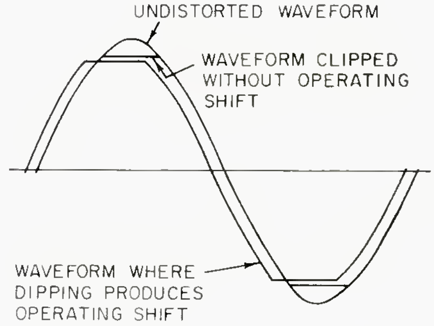


Figure 2. Some waveforms to show different possible effects of clipping, at the same nominal level (a slight difference in clipping level is shown, only to avoid confusion between the two shapes).

FIGURE 2 shows this distinction.

Another example of a signal that can be viewed differently is a frequency modulated radio-frequency carrier. From the viewpoint of the modulating signal—the audio—the central carrier frequency is modified by the modulation, without changing its amplitude. This is a fairly simple concept, and one that proves useful in examining modulator and demodulator circuit behavior. You consider the amplitude and frequency of a waveform that, at every point, looks like a single, pure sinusoid.

Provided the frequency, during its fluctuation, does not deviate by more than the prescribed 75 kHz from center frequency, this will, in general, ensure that the transmitted intelligence stays within the band allowed to that carrier. But that is not quite enough, from the viewpoint of determining whether any energy *splashes* into an adjoining channel.

A radio frequency, frequency modulated by a specific modulating frequency, can also be regarded as generating a whole family of sidebands, distributed across the spectrum like lines separated by intervals equal to the modulating frequency.

The magnitude and distribution of these side bands, for any one modulating frequency, depends on the exact relationship between the modulating frequency and the so-called deviation ratio. And when, as happens on practical signal, many modulating frequencies are applied simultaneously, the changing assortment of sidebands generated becomes very complex, although the composite wave, at all times, looks like a single, simple sine wave, whose frequency is varying.

If a few tiny stray sidebands spill over into an adjoining channel, and are then trimmed off by a filter that removes everything above 75 kHz deviation from central carrier, how does this affect the modulation trans-

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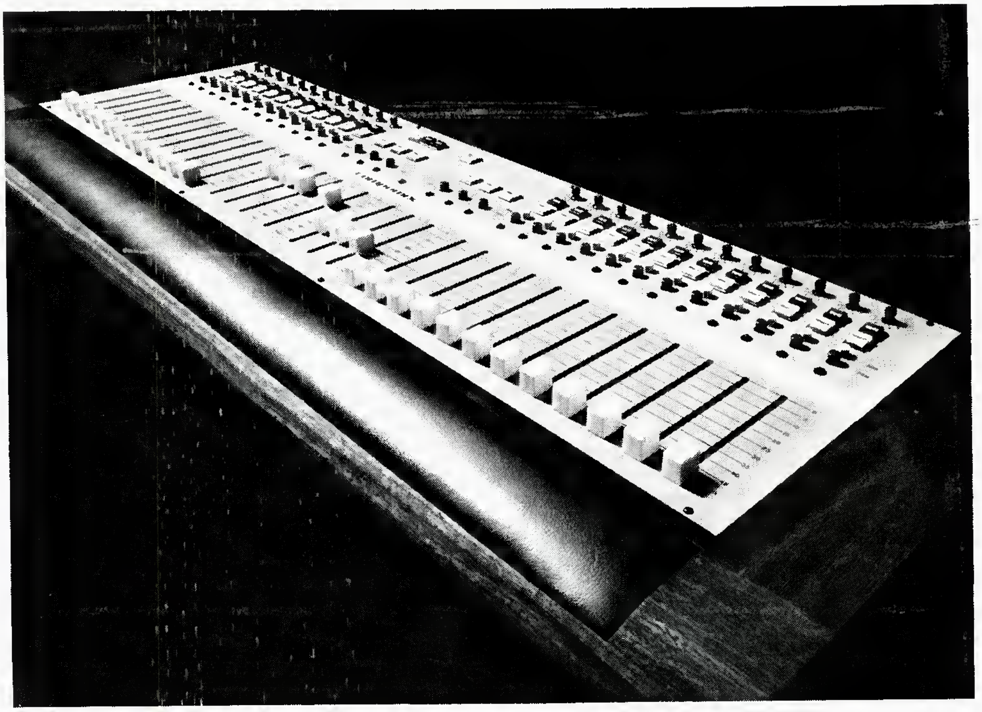

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mitted? Removal of those sidebands, without changing anything within the allowed band, would probably cause the modulated carrier either to vary slightly in amplitude, or else to fail to follow the input modulation faithfully, or perhaps both.

But when feedback is applied over the whole system, so that, with all sidebands beyond 75 kHz of central carrier completely removed, the demodulated signal modifies the input, the remaining sidebands, within the allowed range, are modified so that very little fluctuation in amplitude occurs, and the output modulation closely follows the input.

With amplitude modulation, all those frequencies are present. True the signal can also be regarded as a carrier whose amplitude is modulated without its frequency being changed. But those sideband frequencies are inseparably linked with the modulation frequencies they represent. If the sidebands are lopped off at 4.5 kHz on either side of the carrier, the transmitted modulation is lopped off at 4.5 kHz, period.

With frequency modulation, the link is not so definite. In both cases, what influences the received signal is how the effective carrier varies, one in amplitude and the other in frequency, with the opposite quantity regarded as constant in each case. But in frequency modulation, whether frequencies more than 75 kHz from the carrier are represented in the composite synthesis does not directly affect the modulation, as it does in amplitude modulation.

Of course, that statement has limits too. If it didn't, fm would find a use on medium frequency band, where it is definitely not applicable: you cannot have a fractional deviation ratio. If the modulating frequency is 9 kHz, the nearest sidebands it can generate, am *or* fm, will be 9 kHz on either side of the carrier.

Maybe the notion I receive in some readers' letters, that you can have your cake and eat it too, derives from listening to too many campaigning politicians, who make all kinds of promises. As adults, we should not be so naïve, although I am not exempt from such foolishness.

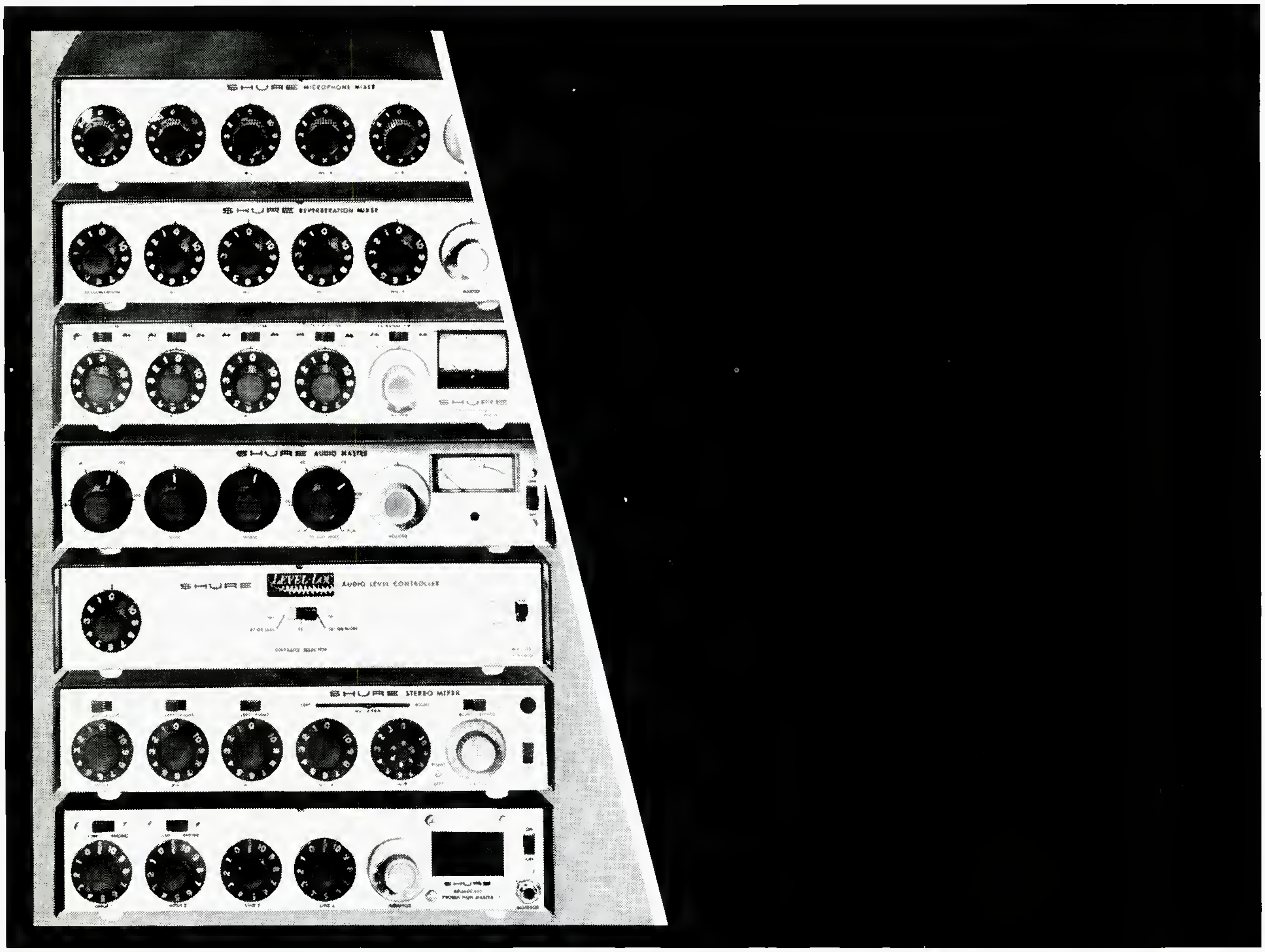
Earlier this year, I had heard several complaints about the Truth in Lending procedures, from both borrowers and lenders in various categories. So I studied the matter. Foolishly, I presumed that Congress had designed this legislation for the mutual benefit or protection of both borrowers and lenders and, because a Senator from our state is on the Bank and Finance Committee, I wrote to him about it.

Using my mathematical ability, I had worked out a more satisfactory procedure, that would involve less work and achieve what I thought the object was, which I sent to him in a 3-page letter. Can you guess what happened? He passed my letter to the real architects of the bill, who replied to me.

Congress (Senate or Representative House) did not design that bill. The Federal Reserve Board did. And from their reply—also a 3-page letter—that my Senator forwarded to me, the reasons they give for rejecting my suggestions make it very plain that the object of the legislation was not what I had foolishly imagined. Rather it is to help every American to get himself further into debt, thereby giving more control to various levels of creditors and most control to the super-creditors of all: the Federal Reserve System.

Why do I mention this? Because it is an election year. Maybe the fact that the people we elect get their directives, not from us, but from various government agencies, will explain why there is so little choice between Democratic, Republican, or Independent. They all follow bureaucrats' orders. And in a sense, it's our fault. We do not give them anything else to do.

If we would just stop listening to campaign promises that they have not a prayer of fulfilling, and look for candidates who want to represent us and implement what we want done, thereby reversing roles back to what the Constitution laid down, instead have the servants boss the masters, this country could change in a hurry. Why not pass the word? ■



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John M. Woram

THE SYNC TRACK

Systems Reliability

● As this is being written, I have just returned from the sixth annual Audio/Recording Seminar at Brigham Young University where I had been invited to conduct the session on Control Room Engineering.

At a later date, **db** will have a detailed description of the seminar. In this month's column, I'd like to dwell on at least one subject that came up during the seminar; *equipment;* or to put it more broadly, *systems reliability.* The subject came up almost as an afterthought towards the end of the class, and so we didn't have too much time to get deeply involved in the pros and cons. Unfortunately reliability often comes up only as an afterthought, as Ralph Nader has been noting for so long.

Anyway, a day after the seminar ended, I was flying back to Fun City, thinking of all sorts of brilliant things to say, now that it was too late. Perhaps it was the altitude—at 30,000 feet, systems reliability is a very important subject. But even on the ground, a little more thought wouldn't hurt.

Which brings us back to the studio. (Well, what did you expect?) I suppose it an oversimplification to say that recording gear should have built-in reliability, *but* reliability costs money, and may not show up on the spec sheets in measurable terms. In fact, reliability judgments can only be made after the passage of time. If the equipment doesn't blow out (or up) we can eventually say it's reliable. Of course, with the obvious exception of new products, most equipment has been on the market long enough for some general consensus of reliability to have been formed.

you write it

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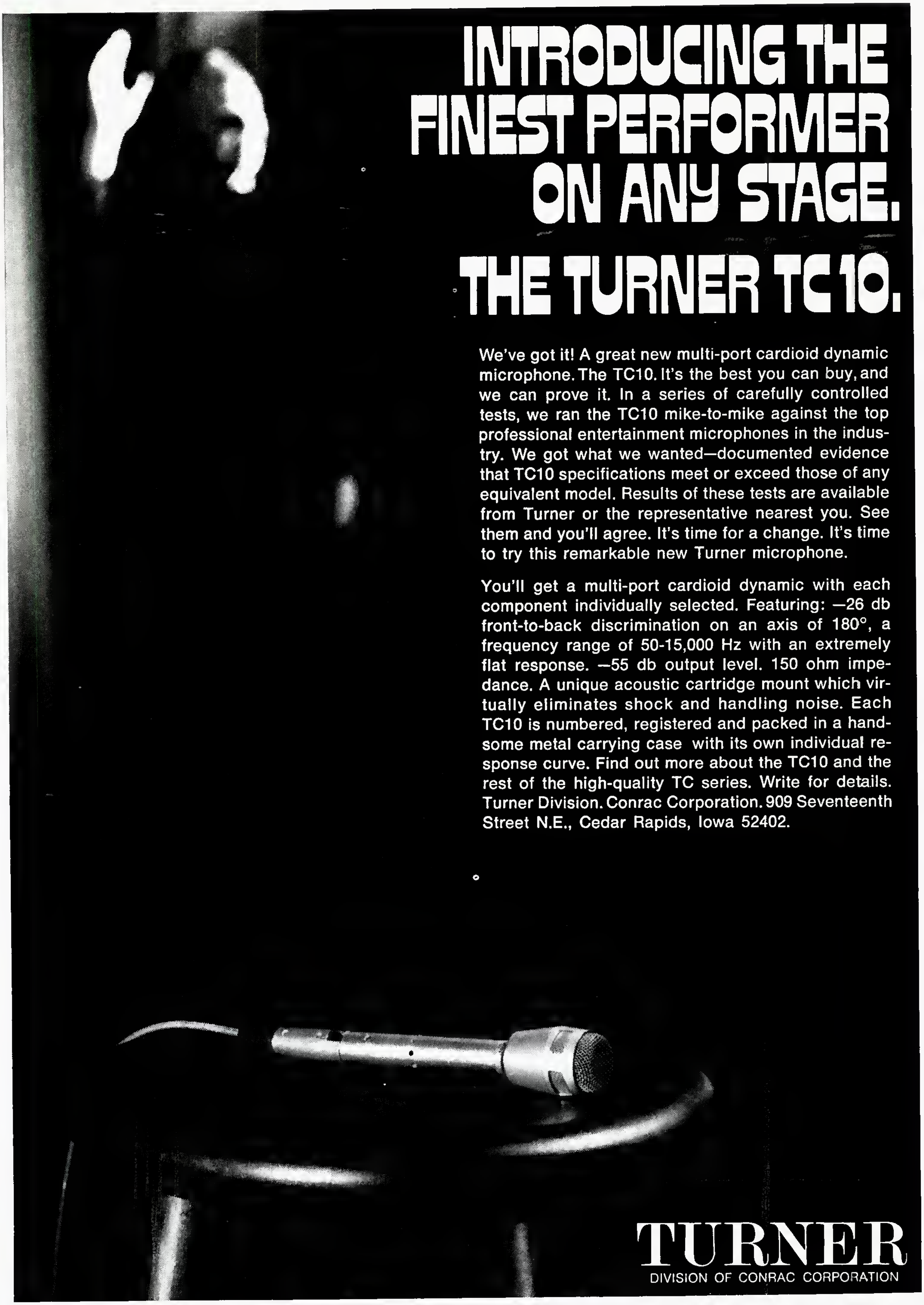

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For example, some time ago we solved one of our little reliability problems with monitor amplifiers. We had a quantity of very inexpensive amps with very impressive specs. And, in practice, the amps fully met their published ratings. Just one little problem though—they had an uncomfortable way of dying when one least expected it. Also, lifting an input or output was a virtual guarantee that they'd go up in smoke.

So, we installed a bunch of Crown DC-300 power amps. End of problem. The darn things are indestructable, and Lord knows we've tried hard to do them in. They also cost about three times as much as the old unreliables. But, it doesn't take too long to realize there's no savings in buying unreliable amps. Trouble-shooting any system, you proceed by eliminating—one by one—the probable failure points. The fewer there are, the quicker and easier the trouble shooting.

A month or two ago, I mentioned how sealed relays had cut down our failure rate. On a larger scale, the DC-300 amps have reduced our monitoring problems to zero. That's worth a lot of money. In fact, it seems we can easily afford the Crowns. It's the others that we can't afford.

Another frequent reliability problem is headphones. In any studio, phones are bound to get a lot of abuse. There's just no way to prevent them from getting dropped, kicked, or just plain stepped on.

Sometime ago, I talked to the Koss people about the need for a "studio proof" headphone. They were interested in working on this problem, and we began by buying a small quantity of their *red-devil* phones (model KRD-711). Then every time we managed to break one, we'd send it back for analysis. In a very short time, they came up with the phones we now use—a modified version of the 711. The phone uses a steel-cased Switchcraft plug, and an extra-durable coil cord with improved strain relief where the cord enters the ear piece. We've been using them now for over six months and have had no failures. No doubt they will last a lot longer, but from past experience I know that six months of studio use is a long time.

Studios interested in trying these modifiied phones should contact Joe Kotowski at the Koss factory in Milwaukee, Wisconsin for price and delivery information.

While on the subject of headphones, it seems that every studio has its own method of supplying a headphone feed to the musicians. Our system is not ultra-sophisticated by any means, but it works well and is *reliable.*

Stereo phones generally use three conductors, common, left, and right. Since we supply a mono feed to our two cue lines, we use the left and right conductors only. The ground is unused, and so, its only function is to connect the left and right earpieces together—in series—thereby doubling the total impedance of the phone. Since the Koss phones mentioned above have an impedance of about 250 ohms for each earpiece, this makes the impedance 500 ohms per phone. Even with twenty phones connected in parallel, the total impedance is a safe 25 ohms. Note that the phones themselves do not require rewiring. The only modification required is at the amplifier. Simply connect the left and right conductors to the amps' two output terminals and you're in business.

As a safety measure, we've inserted a 10-ohm power resistor in one of the conductors, so that momentary shorts—as phones are plugged in and out—do not affect the amp. We use a stereo amp, the Crown D-40, with each side of the amp driving its own phone feed. The sight of a Crown amplifier driving a headphone system raises an occasional eyebrow on visiting engineers; but again, the old reliability factor must be considered. The combination of the Koss phones and the Crown amp has reduced our cue line problems to zero. What's that worth?

Let's get back to the subject of relays for a minute or two. Sealed units are more reliable than the non-sealed variety. Enough said. But, what about the reliability of the circuit in which the relay functions? It's a common practice to connect one side of a relay to the power supply, with the other side connected—via a length of wire—to a switch. Closing the switch grounds the length of wire and actuates the relay. The logic (?) of this convention is that if the length of wire is accidentally grounded anywhere along its length—for example, a short circuit—the power supply will not be damaged. The only thing that will happen is that the relay will become activated. This assumes that the power supply is more valuable than whatever it is that the relay is doing.

But, is it? What if the relay is in the *record* circuit, and the accidental short has just erased the string track which was recorded a few weeks ago? If I had a choice, I'd prefer a situation where a failure would put a machine *out* of the record mode, rather than into it.

To me, a system that may destroy something more valuable instead of itself is not reliable. What does *your* tape machine do? ■

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Martin Dickstein

SOUND WITH IMAGES

Figure 1. The benefits of sound and the detriments of noise are illustrated. The demonstrator uses a variety of noisemaking devices including the giant mosquito, a sound cannon, and a household motor.

Think quiet

● Speaking of ecology, and who isn't these days, a recent public exhibit gave visitors brief lessons in sound, provided interesting facts and ideas for better mental and physical health, visually demonstrated and offered helpful advice on the subject of what to do about a most serious form of pollution—noise.

Noise itself is not of recent origin, of course, nor is it strictly American. In fact, according to information available at this display, Julius Caesar was so bothered by noise that he outlawed chariot driving at night; Charles De Gaulle banned by law transistor radios in public places; the hearing ability of some of the young rock musicians of today has deteriorated to the level of that of a 75-year old man; and a tribe in the quiet regions of the Sudan has been found which has 75-year old men with hearing capability of a 25-year-old American man.

The temporary exhibit, "Think Quiet: The Sound Show," was set up by the Owens-Corning Fiberglas Corp. at their New York headquarters for the purpose of presenting to the public the benefits of sound and the detriment of noise. To do this, live demonstrations, visual exhibits and "hands-on" displays provided to give the visitors an understanding of some of the principles of sound, the meaning and dynamics of noise, and some possible means of controlling this rapidly growing menace.

First, people were invited to take a hearing test. Different tones at various levels of loudness were reproduced from tapes played on three LaBelle Repeater units. A brief explanation of how the ear works was given, and a short description of what sound is and how it is measured was also presented, with the introduction of the term dB. A booklet gave a short discussion on sound terminology and the characteristics of loudness. It was also shown that high-pitched sounds were far more noticeable than low-pitched ones at the same decibel level giving rise to a relative weight difference described by the term dbA.

A film was then shown at the next display which gave the viewer a three-minute wrap-up of a 24-hour period recorded and filmed on the corner outside the building in which the exhibit was taking place. The intersection is an extremely busy one and is intensely noisy during the normal work day, but the sound level drops dramatically during the night, although never quite down to the level of quiet. The display then discusses the physiological and psychological dangers inherent in this type of disturbance. Aside from hearing loss, the body reacts to noise in other ways. A sound of even moderate level and short duration can cause the brain's blood vessels to dilate while the exact opposite is true in other parts of the body. Blood pressure rises, heart rhythm changes, pupils of the eyes dilate and even the stomach changes its rate of acid secretion. Excessive noise has been related to stress-caused ailments such as heart disease, ulcers and indigestion. Steady noise has been found to bring on tension, nervousness and irritability which in turn may cause fatigue, depression and inefficiency.

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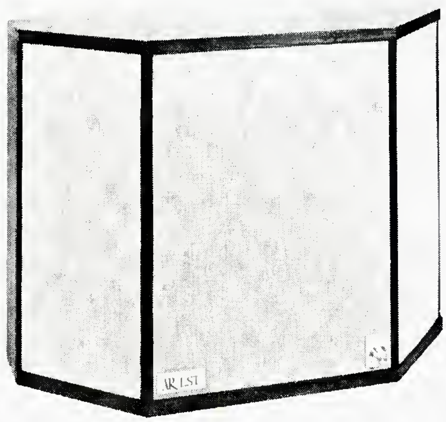

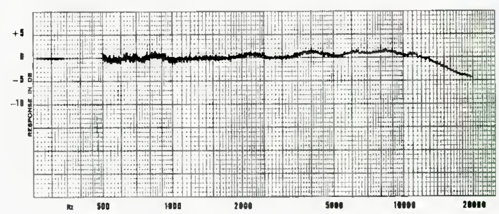

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Figure 2. A guest listens to noise as heard in different locations and is told what is being done to control it.

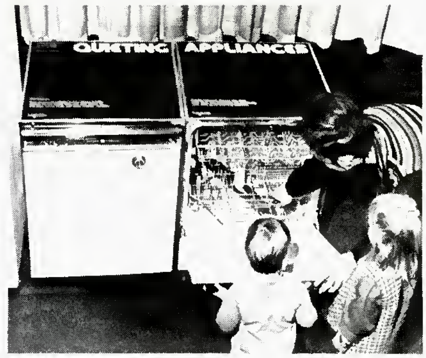

Figure 3. The difference between a noise treated dishwasher and a conventional one is illustrated.

The frustration of failing hearing was aptly demonstrated in small telephone-type booths. The listener was told a joke beginning at a normal level of 60 dB. With each succeeding line, the level was dropped 10 dB. Some visitors found that defective hearing is no laughing matter. The full joke was repeated at full volume for those who missed the punchline, but the point of the exhibit had been made.

Farther into the exhibit, a live demonstration gave a more detailed explanation of the ear and its function with a large plastic model. The motion of sound waves through the air was shown with a large "sound cannon," a drum which, when struck with a mallet, caused a row of streamers hanging at some distance away to ripple. An "electronic Caruso," a single-pitched voice on tape, was played loudly enough to shatter a wine goblet and thus illustrate the destructive power of sound energy. A screaming baby, also on tape, was played at four levels of intensity to demonstrate how disturbing this sound can become in the close living conditions of a large city apartment house built without the proper sound-deadening precautions. In a similar vein, another display illustrated simply with a doll walking on a roof of a house, how this would sound with and without sound isolation. A microphone was used to pick up the noise and feed to an amplifier and speaker to enhance the effect.

In a contemporary appliance-filled kitchen exhibit, different electrical household devices could be activated by the visitors with the relative sound levels indicated on a screen. It was shown quite forcefully that with all the labor-saving devices working simultaneously the noisiest room in the house reaches the level of a factory.

Photos, tapes and slides indicated the types and levels of noise that are developed each day in the office and on the road, at construction sites and in plants and factories, and in various means of transportation such as subways and airplanes. Methods for cutting down on some of the noise were given and demonstrated. Practical and inexpensive suggestions such as how to quiet an ordinary dishwasher for about $2.00 and a garage door for about $5.00 were offered.

In another part of the display, four slide projectors, each with drums containing 80 slides, were used in rear projection among photographs and charts to illustrate how noise is created and what can be done to control it. Simple solutions from quieter garbage cans in the street to use of carpeting, as well as wall and ceiling

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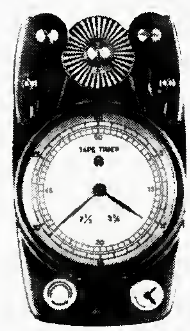
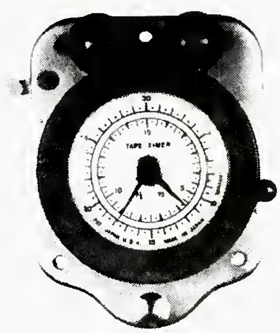


The Standard Model is calibrated for 7.5 and 3.75 ips and can also be used at 15 ips. Dimensions are 1⅞" x 3¼". It is priced at only $49.95.



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insulation materials in the home, to the more complex ones which utilized rubber wheels on subway cars in Montreal and Mexico City, were presented. Means to quiet household devices, the use by the construction industry of quieter hydraulic pavement breakers at some building sites and the application of sound absorbing materials in the construction of the latest jet engines so that they are ten times as quiet as some smaller aircraft were also mentioned.

Some of the interesting and vital facts depicted at the show indicate that the noise level in the United States is doubling every ten years; an estimated 16-million people in the country suffer from some degree of hearing loss directly caused by noise; and that industry is paying $4 billion a year in accidents, absenteeism, inefficiency, and compensation claims directly attributable to noise in the office and factory environments. Recently, state and federal government agencies have begun acting to try to control noise. The first state-wide noise control legislation was signed into law in January, 1972, in New Jersey. This law allows the setting of noise level standards for mechanical devices from construction equipment to automobiles, with violations liable to fines up to $3,000.

The main purpose of the exhibit was to help each individual do something to help in the fight against noise pollution. Four steps, generally, were indicated as possible ways of alleviating some of the noise. These are to move the source, absorb the noise, block it, or reduce it. Noise from typewriters, washing machines, dryers and dishwashers can be cut down by rubber mounting and insulation. In addition to drapes and carpeting in the home, moderate level in the playing of the t.v. and hi-fi can also help. Noisy cars might be quieted by tuning the engine or fixing the muffler. Each individual can also try to get codes and regulations passed by the various governments and then enforced wherever possible to help fight noise.

The entire exhibit was interesting in the facts that have been made available to visitors and also in the simple ways in which visuals of different kinds were used to demonstrate an aural phenomenon. Hopefully, the public came away with some understanding of sound and noise and the effects on the hearing of all of us, and a desire to do something about this form of pollution before it gets completely out of hand. Among the units of equipment used in the display were reel-to-reel Wollensak tape recorders, Bogen 100-watt amplifiers, JBL speakers, Kodak slide projectors and Triangle 16-mm film projectors.

In line with the above demonstration of noise and its possible harmful effects, it might be interesting to mention that the federal government and NASA are looking into the problem of airplane noise during take-off and landing and have been for some time. Recently, a 300-representative conference took place in which the aircraft industry was informed of the latest results of the experimental quiet engine being developed by NASA, for use by commercial jetliners of the future.

Tests were made with microphones arranged in a semi-circle 150 feet from the engine during static running and the noise generated was measured. The total of four such engines mounted on DC 8/707 type aircraft would only be 90 EPNdB, compared with the present 116 EPNdB during take-off and 118 EPNdB during landing. The jet noise was reduced primarily by special construction in the engine and the fan to allow the exhaust air to exit with less noise being created than before. Noise was further reduced by the addition of Fiberglas sound absorbing material to the nacelle surrounding the engine. The

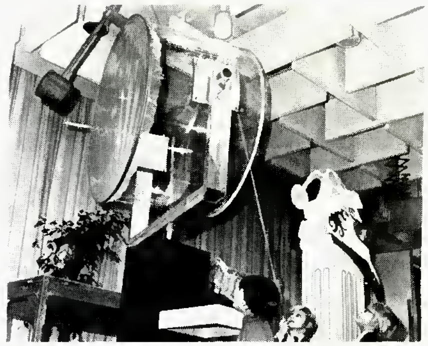
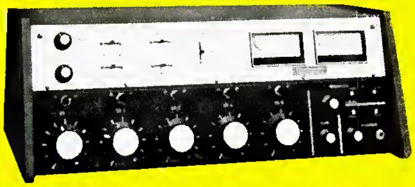


Figure 4. The sound cannon. The rope when pulled activates the mallet. It hits the drum sending sound shock waves through the air and rippling streamers hung a short distance away.

nacelle is designed to take care of the frequencies in the 2,000 to 4,000 Hz range. These have been found to be the sounds most easily perceived by the human ear and are therefore the most annoying as noise in this frequency range. The term EPNdB has, therefore, been adopted to describe these *effective perceived noise decibels* and is presently being used to measure the intensity of the noise in this limited frequency range, thus taking into account the characteristics of the normal ear. Maybe there's some hope for our hearing after all. Tomorrow the airplanes, the following day the rock music and transistor radios. Then the kitchen. ■

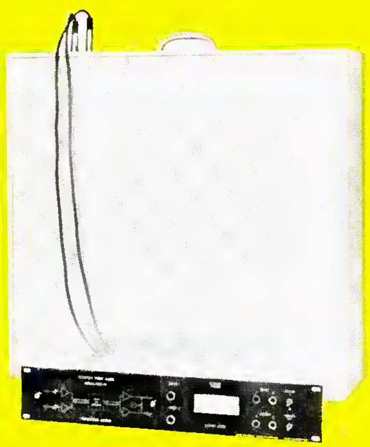
NEW PRODUCTS AND SERVICES

CONSOLES



● A complete line of audio consoles, primarily for broadcast work, are available. They include mono and stereo models with five or eight channel inputs. All console mixing inputs are supplied with preamplifiers and with provision for use in low or high level application. Electronic f.e.t. switching of input channels to program and audition buses is standard.
Mfr: Spotmaster-Broadcast Electronics, Inc.
Price: $775 to $2950.
Circle 72 on Reader Service Card.

DELAY SYSTEM



● The Cooper Time Cube is a dual acoustical delay system with two electronically independent delays of 16 and 14 milliseconds. Frequency response is ±2 dB 40-10,000 Hz, distortion is less than 1 per cent at all program levels up to ±4 dBm output. Signal-to-noise is greater than 70 dB. Applications are particularly appropriate for synthesizing quad tapes and records from two-channel originals, loudness or spacial enhancement of program, delaying feed to reverb chambers or devices for added dimension, and improving optical film recording by delaying audio to light valve or galvanometer (applicable to Westrex, RCA, Maurer, or other film recording systems).
Mfr: UREI
Circle 69 on Reader Service Card.

GRAPHIC LEVEL RECORDER

● This compact unit, the Justi-Meter III, has been designed to permit recording of the frequency-response characteristic of audio equipment on a chart. The linear range is 40 dB and the chart is frequency calibrated. The unit covers a sweep range from 20 to 20,000 Hz obtained from a (supplied) B & K Q-R-2009 test record. The sweep and plotting take 50 seconds. The chart starts automatically from the test record signal, and measures the response of phono pickups directly, and with external connections that of amplifiers, filters, and tape recorders. Two writings speeds, over 40 dB/sec. and approximately 12 dB/sec. are provided. A powered condenser mic input is also provided.
Mfr: Justi-Meter Corp.
Price: $595.00
Circle 60 on Reader Service Card.

PEAK PROGRAM METER

● The M-900 is a relatively inexpensive peak reading meter with a linear dB scale from +9 to −36 dB. It makes possible an ability to operate close to the overload level without creating distortion. Important specifications include: voltage requirement is 24 V d.c.; current consumption is approximately 20 mA; input impedance is 10 k ohms unbalanced; indication linearity is ±1 dB; integration time is approximately 10 ms; and fall back time is 2 seconds full scale (adjustable).
Mfr: NTP (Audiotechniques)
Price: $110 (single units).
Circle 65 on Reader Service Card.

SERVO-CONTROLLED RECORDER

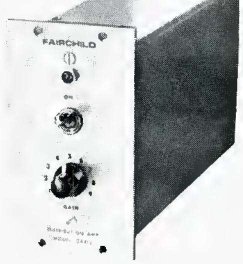
● The new 79 series audio recorders use a d.c. servo capstan system incorporated in the Isoloop differential drive system, enabling operation of the unit at variable speeds ranging from 5 to 45 in/sec—in addition to fixed speeds of 7 1/2, 15, and 30. Eight, sixteen, and twenty-four track configurations are standard with readily accomplished conversion upward of eight- and sixteen-track units. Space exists in the cabinet for noise-reduction units, remote transport and electronic controls are incorporated in the recorder's cabinet but may be removed for use at the console.
Mfr: 3M Company
Price: (approximate) $15,000 8-track; $20,000 16-track; $29,000 24-track.
Circle 62 on Reader Service Card.

1/4-INCH, 4-CHANNEL

● Four track recording and play with Simul-Sync capability on each channel is offered on this new A-3340 model. It takes 10 ½-inch reels and operates at 7½-15 in/sec. A mixable mic and line input exists for each channel, making this unit a single-box field-recording system. Separate switches permit each channel to be activated for record, and switches on the head cover permit each channel to be switched from normal to Simul-Sync. In this latter position, the record head of the switched channel is used for playback, eliminating normal record-play head delays—and permitting synchronized additions to other channels. Important specs of the unit include speed accuracy of 0.5 per cent, three-motor operation with a dual speed hysteresis-synchronous capstan motor, flutter of 0.04 per cent at 15 in/sec with ±3 dB response at that speed 30-20,000 Hz. Signal-to-noise is 60 dB harmonic distortion is 1 per cent crosstalk is 60 dB, and stereo separation is 50 dB—all at 1 kHz. Rewind time for 1200 feet is 90 seconds.
Mfr: TEAC
Price: $849.00.
Circle 67 on Reader Service Card.

DISTRIBUTION AMPLIFIERS
Models No. 725DA5/T, 725LA, 725LA/T, 725LAD, 725LAD/T2, DA415-8, DA 1520

FAIRCHILD SOUND EQUIPMENT CORP.
A ROBINS INDUSTRIES CORPORATION
75 AUSTIN BLVD., COMMACK, N.Y. 11725
(516) 543-5200



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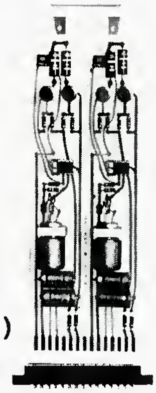
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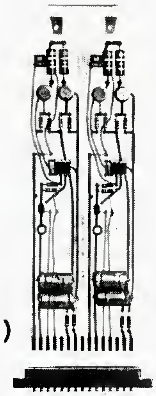
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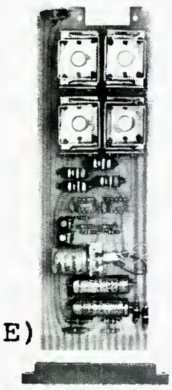
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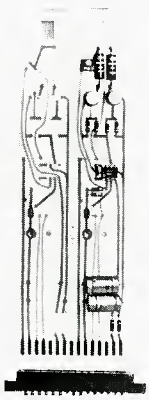
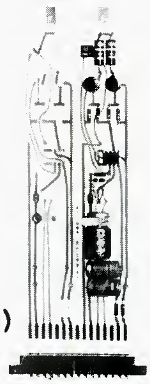
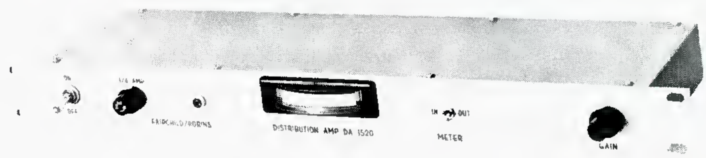

(A)


(B)

(F)


State of the art distribution amplifiers for use in broadcast and recording studios, PA systems, telephone networks, schools, etc. Features include card-file and rack mounting modularity and power supply compatability with all other Fairchild/Robins circuitry. Low noise, high gain* transistor or IC op-amp circuits offer high stability, extremely low distortion, flat frequency response and overload protection, and almost nil crosstalk between channels.

These reliable economical units use heavy duty components for 20 year continuous-duty design life with low power consumption. Epoxy printed circuit boards with gold-plated contacts and mating P.C. connectors are supplied.

8 CHANNEL, Model DA415-8 (Illust. G) which operates on 117VAC consists of 8 separate Class A emitter follower isolation amplifiers whose inputs and outputs are accessible individually through a blue ribbon connector allowing for a variety of interconnections, including the distribution of a signal into 8 separate lines. Output circuits are used with reactive loads driving 600 ohm lines, which match transformers normally used in transmission lines, and telephone links. Unity gain enables direct input connections from high level sources such as consoles, tape machines, mixers and telephone lines.

The output stage has very low output impedance to work into resistive or inductive loads (important when feeding telephone lines through a transformer so that the Telco line impedance is bridging the output of the DA415-8).

The self-contained power supply has active filter and voltage regulator circuits, overload and short circuit protection, and very low ripple voltage. The gain control is used for common input to all amplifiers.

15 CHANNEL, Model DA 1520 (Illust. F) distributes an audio signal into 15 balanced lines. It mounts into a 19″ rack, taking only 1¾″ of vertical space and has a self-contained power supply for plug-in to 117 VAC line.

Output circuit of the amplifier consists of two sets of complementary symmetry current drivers with heavy negative feedback, assuring low distortion and low output impedance needed for channel isolation.

VU meter monitoring of input and output signals and gain is provided. All input and output connections are made through screw type terminals on a barrier strip, allowing for a variety of applications. Output level of the amplifier remains constant, regardless of the number of lines being fed by it. Input to the amplifier is balanced and isolated by an input transformer.

2 to 75 CHANNEL, Model 725DA5/T (Illust. E) is a high level, high power distribution amplifier which can also be used as a power amplifier to drive small cue speakers or as a line amplifier, etc. Each of two separate sections delivers 26 dbm signal into 600 ohm load. Combined power output is 36 dbm into lower impedance load. Delivers balanced output without use of transformer.

25 CHANNEL, Model 725LA (Illust. A) is a versatile basic modular amplifier which can be used as a line amplifier or distribution amplifier, delivers up to 27 dbm into 8 ohm load unbalanced.

25 CHANNEL, Model 725LA/T (Illust. B) is the same as 725LA but with matching input transformer.

50 CHANNEL, Model 725LAD (Illust. C) Two line amplifiers on a single card. Each amplifier similar to 725LA. Separate decoupling for each amplifier. Delivers balanced output without use of transformers.

50 CHANNEL, Model 725LAD/T2 (Illust. D) Same as 725LAD except with matching input transformers.

*except DA415-8, unity gain.

DISTRIBUTION AMPLIFIERS Models No. 725DA5/T, 725LA, 725LA/T, 725LAD, 725LAD/T2, DA415-8, DA 1520

FAIRCHILD/ROBINS
FAIRCHILD SOUND EQUIPMENT CORP.
A ROBINS INDUSTRIES CORPORATION
75 AUSTIN BLVD., COMMACK, N.Y. 11725
CABLE: ROBINSOVERSEAS COMMACKNY
(516) 543-5200

NAB

FAIRCHILD/ROBINS DISTRIBUTION AMPLIFIERS

Model No.	Illus.	No. of Ampl-ifiers	Gain db	Input Imp. ohms	Output Imp. ohms	Max.Out-Put Level dbm	Load Imp. ohms	S/Noise db(0dbm out) Gain	S/N	Dist. Max.%	Freq. Resp. db± (a)	Interch. Crosstalk db min.	POWER REQ. VDC	@ ma	Rec. Power Supply	L x H x D Dimensions	Assoc. Mtg. Hardware
725DA5/T	E	1	5-40	10K	0.5	36	8&up	40	85	0.2	0.5	70	±15	20/120 (e)	667T/15	2½x7½(b)	725SCH or 725CF
725LA	A	1	0-25	100K	0.1	27	8min.	35	70	0.2	0.5	--	±15	120	667T/15	2½x7½(b)	725SCH or 725CF
725LA/T	B	1	20-45	200-600	0.1	27	8min.	55	71	0.2	0.5	--	±15	120	667T/15	2½x7½(b)	725SCH or 725CF
725LAD	C	2	0-25	100K	0.1	27	8min.	35	70	0.2	0.5	70	±15	120(f)	667T/15	2½x7½(b)	725SCH or 725CF
725LAD/T2	D	2	20-45	200-600	0.1	27	8min.	55	71	0.2	0.5	70	±15	120(f)	667T/15	2½x7½(b)	725SCH or 725CF
DA415-8	G	8	0	100(d)	10	15	600	0	75	0.3	1.0	70	117(AC)	45	--	3½x5½x10	662RM
DA 1520	F	1	0-40	600	600(c)	20	600	35	70	0.2	0.5	70	117(AC)	40	--	19x1-3/8x6	19"rack

(a) 20-20K Hz (b) Printed circuit board with gold-plated contacts and mating connector (c) All 15 outputs (d) Each amplifier, 12K ohms all inputs paralleled (e) 20 as line amplifiers, 120 as power amplifiers (f) each amplifier

ARCHITECTS AND ENGINEERS SPECIFICATIONS: DISTRIBUTION AMPLIFIER

The Distribution Amplifier shall employ transistor or Integrated Circuit design. Vacuum tube circuitry will not be acceptable.

The transistors, Integrated Circuits and associated circuitry shall be assembled on glass epoxy circuit boards which shall have gold-plated contacts, and the entire circuit board shall be removable from its housinq, if provided, for immediate inspection and service.

The frequency response of the Distribution Amplifier shall be __ to __ cycles, plus or minus __ db. The gain of the preamp shall be __ db.

The amplifier shall be capable of a power output of __ dbm with distortion under __% at __ cycles and __% at 1KC. Input impedance of the Distribution Amplifier shall be __ ohms. A transformer may be used at the input for lower impedance requirements.

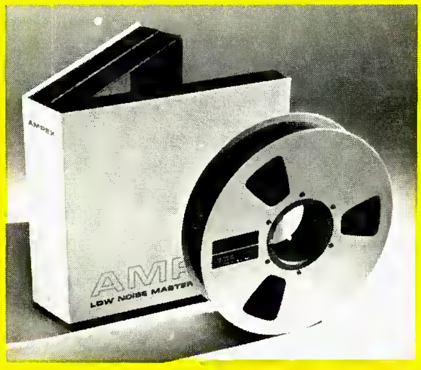
The output of the device shall be __ ohms. The preamp shall operate from an __V. power supply at __ma.

The physical dimensions shall be __" x __" x __".

The Distribution Amplifier shall be the Fairchild/Robins Model ____.

MASTERING TAPE

● A new low-noise mastering tape with a backcoating has been made available. The new oxide formula and binding system are stated to provide improved recording and play characteristics and greater durability. The 1-mil 407 series and 1.5-mil 406 series replaces the 444 and 434 series mastering tapes. Extremely fine ferric oxide particles are used on the new formulation to achieve a higher packing density, resulting in a 3.5 dB improvement in undistorted output capability over previous tape from this company. The tape is offered on reel sizes from 7-inch to 14-inch diameter, and on standard NAB hubs. Widths of ¼, ½, 1, and 2 inches are available.
Mfr: Ampex Corp.
Circle 66 on Reader Service Card.



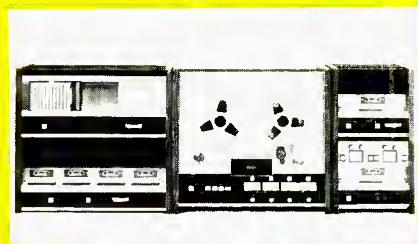
OSCILLATOR

● Model 700 is a plug in externally adjustable audio oscillator integrated circuit. It draws ±24 V at ±20 mA and will deliver ±10 dBm at 0.5 per cent harmonic distortion. By changing resistance values to the add-on circuit (which requires two resistors plus an output pot of 1k) frequencies of 16 Hz to 15 kHz may be obtained.
Mfr: Opamp Labs Inc.
Price: $35.00
Circle 61 on Reader Service Card.

MIXER/PREAMP

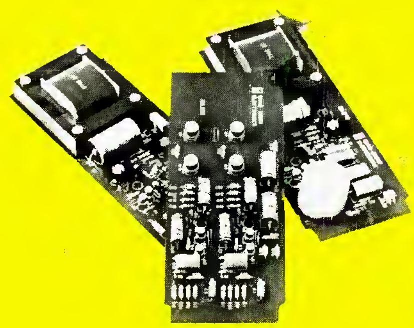
● This stereo unit is designed for clubs, small broadcast stations, school and background systems (with paging), and advanced music reproduction systems. It has two pairs of low-level RIAA-equalized inputs, two pairs of high level inputs, two mic/line inputs (mono) switchable to left, right, or both output channels. Used direct, the mic inputs are high impedance; with an A-1002C input card and CMA-481 input transformer is balanced low impedance. Line outputs are at ±18 dBm each channel unbalanced. Each channel has its own equalization controls; there is a master gain control. A separate monitor output is available with its own level control. It is switchable to program or cue any of the four stereo inputs.
Mfr: Bozak
Circle 71 on Reader Service Card.

CASSETTE DUPLICATOR



● High speed cassette duplication is accomplished by the System 200. It used modular units for maximum flexibility in meeting user needs. Open reel, cassette, and endless-loop masters may be used. Open reel and/or cassette slaves are options. There are fully automatic cassette slaves for unattended operation. New ferrite heads are used for long life and reduced cleaning maintenance. Other features include automatic rewind and cuing, defective cassette stop and indication, plug-in electronics with both digital and linear i.c.'s. All major tape formats—cassette, open reel, cartridge—can be duplicated.
Mfr: Infonics
Circle 63 on Reader Service Card.

AMPLIFIER CARDS



● The CA series are 2¾-inch by 7-inch printed-circuit cards designed for upgrading existing consoles. They include the CA 127, an operational amplifier input, +30 dBm transformer output, unity to 40 dB amplification; the CA 227 transformer input and output (+30 dBm), 30-60 dB of amplification; and the CA 272—dual amplifiers, op-amp inputs, single ended outputs (+24 dBm). All three boards have 150 per cent over-voltage and short circuit protection, distortion of 0.2 per cent harmonic from 20 Hz to 20 kHz, and low noise of −129 dBm for the CA 227, 110 dBm for the other two.
Mfr: Quad-Eight
Circle 68 on Reader Service Card.

POWER AMPLIFIER

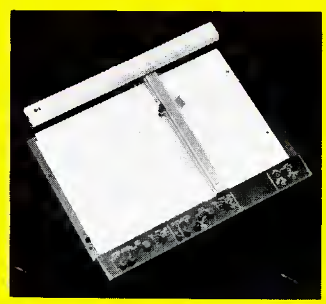
● The model 240 is a stereo power amplifier with 125-watts continuous power per channel over the 20-20,000 Hz range—with both channels driven simultaneously. Relay operated protective circuitry has been designed to prevent damage to output transistors, power supply, or speakers from excessive levels of subsonic frequency. Rack adapters or a walnut cabinet are available for a variety of appearance requirements.
Mfr: Marantz Co. Inc.
Price: $395.
Circle 70 on Reader Service Card.

TAPE MAINTENANCE KIT

● QM-3 is a kit designed for maintenance of tape machines. It contains products designed to provide a preventive maintenance program to a recorder. Included are: QM-102, 2 oz. bottle of liquid head cleaner; QM-301, splicer for ¼-inch tape; QM-501, 150-inch roll of Mylar splicing tape; QM-201 wand head demagnetizer; and QM-502, 6-inch cotton tipped sticks.
Mfr: Nortronics Co. Inc.
Circle 74 on Reader Service Card.

X-Y RECORDERS

● Models 7044A and 7045A are two new general purpose chart recorders. Model 7044A is a medium speed recorder with a slewing speed of 20 inches per second, the model 7045A is faster at 30 inches per second. Input range of both instruments is from 0.5 mV to 10 V, floating 500 V d.c. or peak a.c. The writing area is 10 by 15 inches. Standard paper is used.
Mfr: Hewlett-Packard
Price: 7044A-$1350; 7045A-$1675.
Circle 64 on Reader Service Card.



SPEECH COMPRESSOR/EXPANDER

● The varispeech permits speech recorded on tapes to be played back at any speed between half and two and one half time original speed without affecting pitch or speech identity. It uses a standard cassette operating at variable playback speed. Signal processing is electronic, using no moving parts other than the normal tape motion.
Mfr: Lexicon, Inc.
Circle 73 on Reader Service Card.

JOSEPH MARRONE

How About Some Automation?

db TeleSessions

You're invited to use a brand new communications medium to build on the ideas in this article with a group of people as involved in the subject as you. By means of a new "electronic meeting place" called db TeleSessions, you'll be able to dial into a group telephone conference with other db readers across the country. You can reserve your spot in the discussion now by turning to the inside back cover for details and information on how to participate.

Automation in varying forms is already available in the audio profession. More is coming. But how is it applicable to your needs? Do you need it? What might it cost? This article covers it all.

PROBABLY THE MOST significant event in the history of audio engineering was the marriage of electronics to music reproduction. The application of vacuum tubes to the manufacturing of records and the broadcasting of programs over the airwaves in the late 1920's gave the infant electronics industry tremendous momentum. The vast consumer market potential for radios, phonographs, and ultimately television, spurred the industry on and created a demand for better methods of producing and reproducing sound. Of necessity, the equipment used to do this became more and more sophisticated, especially within the recording studio and transmitter room. In parallel developments, both the audio and broadcasting segments of the industry moved forward in attempting to close the gap between real sound and artificially produced sound. With each technological advance in the science of music reproduction, the industry moved asymptomatically closer to that elusive thing called ultimate fidelity. Once a high degree of fidelity had been achieved, the industry reached out in new directions and made music reproduction into an art form.

As all this was taking place, the type of equipment used to record and reproduce sound became more complex at an ever accelerating pace. By the 1960's, the highly perfected electronic equipment needed for accurate sound production in recording studios had become orders of magnitude more complex and more difficult to use than its ancestors.

Many additional skilled people were required to operate the newer generations of equipment. Entire staffs were needed to record a single performance, and to perform the work necessary to mix down the original material into a mass reproducible form. And somewhere in the never-ending attempt at perfection in sound reproduction, the time-honored techniques and machinery started to run into a man-machine interface problem. The proliferation of equipment and people, all of which had to work in perfect synchronization and harmony, reached, or nearly reached, the breaking point. The audio industry had built up a system that, by its very nature, relied too heavily on the man, and taxed his capabilities to the point where he became the weakest component in the system. Man was being called upon less as a creative entity within the system, and more as an automaton just to keep the system functioning.

During the years that electronic equipment in the audio/broadcasting field was being perfected for the highly specialized tasks that are unique to this industry, there were many parallel developments going on in other fields that involved electronics. Mass production techniques in automobile manufacturing became electrified, chemical processes were being controlled automatically by electronic systems, electronic guidance systems employing highly advanced feedback techniques were enabling us to shoot rockets to the moon, and electronic computers were being developed to calculate our pay checks, our electric bills, and even monitor our heartbeats when all this madness drove us into a hospital. Indeed, electronics was being applied to virtually every field of endeavor in the arts, the sciences, in industry, and in the home.

Perhaps at this point, we should examine some of these other developments that have tremendously benefited other industries, and ask, "Can any of these technological advances in electronics be of any benefit to audio engineering and specifically, can they make the man-machine interface problem less burdensome to the man and free him to be more creative?" I believe the answer to this question is yes. Man-machine interface problems are not unique to audio engineering. They exist in all phases of industry and manufacturing. They exist in airplanes, in submarines, and in the manned space program. The general method of attacking this class of problem is to automate the machine as much as possible without taking ultimate control away from man. Automation in this sense means taking the "dog work" away from the operator, thereby, giving him more freedom to operate the system to its fullest capacity. A side benefit of this kind of semi-automation is the elimination of subjective human errors that usually occur in the repetitive and lower level system functions. Now that we have come this far, it seems fair to state that the audio industry is ready for another marriage to the electronics industry. This time, the major concern will not be on better sound reproduction, but on better utilization of existing equipment and manpower. The benefits of such a marriage will not be from a purely technical standpoint, but rather from a business standpoint, because if some form of automation is feasible, then increased efficiency and throughput should be possible, and this will directly affect profits.

Joseph Marrone is a sales engineer with Digital Equipment Corporation.

When we think of automation, or more properly of semi-automation, there are several ways to proceed. Everyone dreams of the day when everything he must do can be accomplished by the push of a button. But in the context of our discussion, that represents an impractical form of utopia. Hence, it is necessary to define some goal to which we can address ourselves; some readily obtainable objective that can be accomplished with a minimum amount of effort, a very high probability of success, and no ill effects on the present operation of the system. One of the dangers with proceeding too fast in the area of automation is that one tends to bite off more than one can chew: leading a noted philosopher to utter the unforgettable phrase "Most vast projects are usually started with half-vast ideas." So for openers, it is important to define the minimum identifiable task consistent with the three guidelines stated above. Instead of trying to automate the entire operation all at once, with a 1 per cent probability of success, it makes much more sense to select a small, innocuous project that has a 90 per cent probability of success. In order of increasing complexity, here are some typical projects which lend themselves to automation.

Studio Control. Provide a better method for playing tape cartridges. Use of a conductive band on the tape would enable accurate timing of the upbeat, and provide a means of cutting microphones in and out at the proper time.

Tape Recorder Control. Provide a means for automatically starting several tape recorders that must be accurately synchronized. This will be of benefit for dubbing, punching in, mixing, etc.

Automatic Transmitter Switchboard. Provide a system that monitors transmitters A and B, and automatically switch one to full power if the other fails.

Generation of FCC Log. Provide a means to monitor all important transmitter parameters automatically, and print out the FCC log without operator intervention.

Improvement of the Mix-Down. Provide a means to '"remember" the fader, echo send, pan pot, and other parameters associated with the mix-down process so that the operator doesn't have to commit everything to memory.

Network Control. Provide a system that stores a day's programming and automatically switches the outgoing audio (and video) continuities at the proper times and performs the functions of pre-roll, wipe, cut and dissolve.

In each case, the important concept is to place actions of a purely mechanical nature under automatic control, while actions involving judgment should remain under control of an operator. To automate a system requires us to have full knowledge of the normal as well as the abnormal sequences of operation of the system. Once defined, we can design some kind of black box to perform this sequence of operations. Analog techniques, digital techniques, or some combination of the two will permit us to design what we need. Analog techniques are familiar to those who have had experience in feedback control such as is commonly found in operational amplifiers, servomechanisms, etc. Digital techniques were first employed using relays, switches, and timers. Today, most digital systems are of solid-state design. The digital system possesses several characteristics that make it ideal for automation purposes.

Decision-making Ability. Digital circuits can make elementary decisions about a system based on the inputs, and perform some action that depends on the decision. The basic decision making element is referred to as a *gate*, and gates are combined to form logic circuits that make higher level decisions.

Memory. Digital circuits can be made to remember some past event or data. This means that the system can look back and see a "snapshot" of some portion of past history.

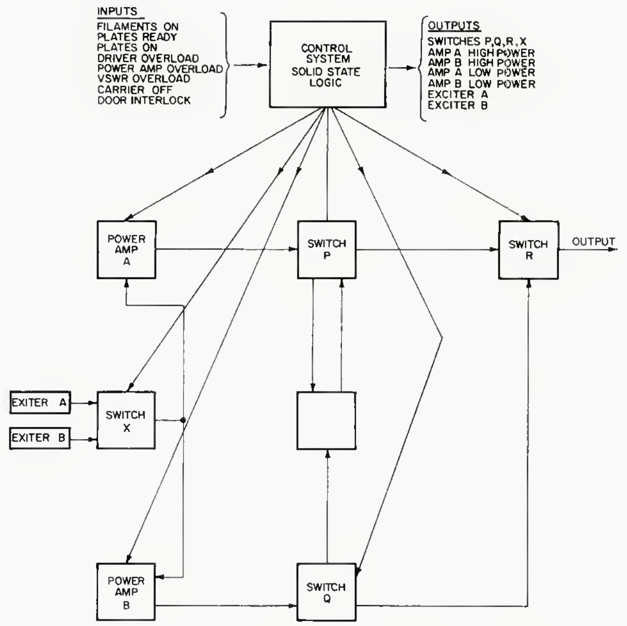


Figure 1. An automated transmitter switchover system.

Sequential Operation. Digital techniques make it easy to perform sequences of operations whereby the termination of one event causes the next event to start. Digital counters, shift registers, and one-shot multivibrators are all used for this purpose.

The proper combination of logic circuits, memory, and sequential circuits enable the designer to build digital systems that can perform a given set of tasks. The more flexible and changeable we make the system requirements, the more complex the logic becomes. As systems increase in complexity, it soon becomes obvious that a digital system must have some means of easy modification so it can be made to do any number of different jobs. This is precisely what led to the development of the small general-purpose computer. A computer is nothing more than a collection of general-purpose logic circuits and memory combined in such a way that one can program the job to be done by putting certain commands in memory. During execution of the program, these commands are interpreted by the logic circuits, resulting in a sequence of events similar to that achieved with ordinary digital systems. The big difference is one of flexibility, and the decision to use a straight digital circuit or a computer involves a trade-off on the cost of implementation *versus* the flexibility of the resulting system.

Let's examine some of the relative costs of specially designed digital systems (often referred to as hard-wired logic systems) and small computers. While each application must be evaluated on its own merit, there is a general rule of thumb that hard-wired systems fit into the under $5,000 category. The availability of small computers in the $5,000 price range make it impracitcal, for the most part, to build special purpose digital systems that cost more than this, when for the same price, or slightly more, one can purchase a small computer with much greater flexibility. On the other hand, a programmable computer does require a programming effort that costs money, and this must be taken into account when estimating the total cost of a job. The important thing to remember is that programming changes do not require us to cut wires, add parts, or change schematics; all of which must be done when trying to modify a hard-wired logic system. So there is a trade-off between the ease of

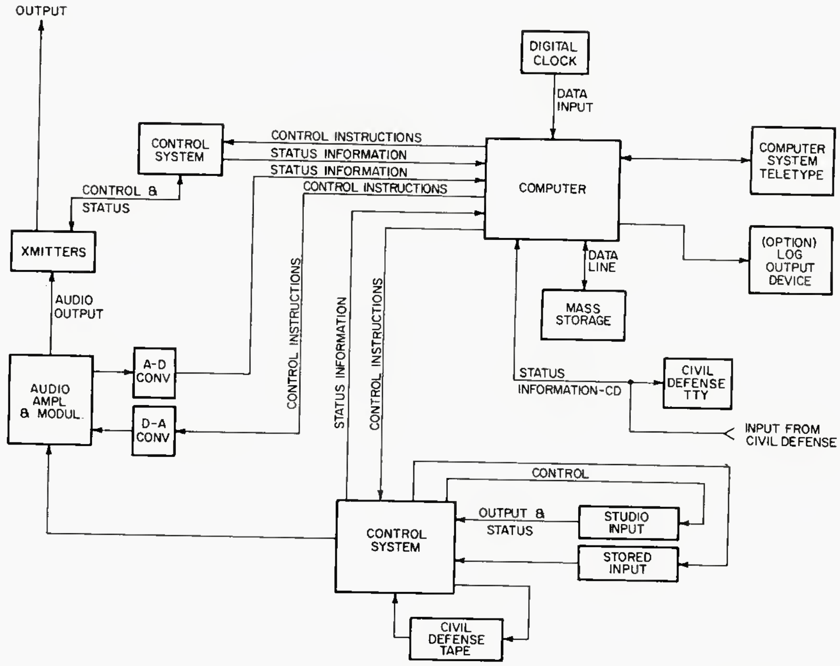


Figure 2. A small computer could control a complete radio station.

modification and cost of modification. The price of a small computer system is affected by several factors. Some of these are as follows:

The size of the memory required to do the job.

The requirement for additional bulk memory (often called mass storage). This is available in several forms such as magnetic tape, magnetic disks, and magnetic drums.

The need for special circuitry to interface the computer signals to the audio equipment being monitored or controlled. Much interface hardware, such as analog-to-digital converters, digital-to-analog converters, and on/off devices such as relays are available off-the-shelf and make most interfacing jobs fairly easy.

The need for special operators' consoles to operate the computer system and give it commands.

Putting all these factors together, we can talk about a minimum practical price of a small computer with memory and some interface equipment of about $7500. Systems in the range of $7500 to $20,000 might be typical for a majority of applications in audio engineering, while for certain specialized tasks price could range as high as $50,000 to $100,000. These prices are all subject to radical change as electronic technology progresses, but the relative magnitudes will stay fairly constant.

Let's examine two typical systems to see how both hard-wired logic and small computers can be applied to specific tasks. FIGURE 1 shows a typical transmitter system in which amplifiers A and B normally operate at half power. If either A or B should fail, the switches P, Q, and R would eliminate the bad amplifier from the system and leave the good one on the air. Then the good amplifier would be boosted to full power. This can all be accomplished automatically by monitoring the status of each transmitter. Such parameters as plates on, filaments on, plates ready, driver overload, etc. would be inputs to the logic system, which monitors them on a continuous basis. The detection of a failure in any of these conditions would be interpreted by the control logic, and the appropriate action taken, such as shutting down the bad transmitter and bringing the other up to full power. The same type of monitoring can be used on the two exciters, and provision made to switch the back-up exciter onto the system should the primary exciter fail. The entire control system to do this type of job can be built using off-the-shelf plug-in modules. Eeverything including the logic gates, power supplies, mounting hardware, a.c. and d.c. input and output adaptors are available today at reasonable prices. Very little special design, if any, is necessary to build such a system. Some people refer to this equipmenf as an "electronic erector set." Once the logic circuit is designed, it can be built from standard printed-circuit cards, which plug into sockets that have been wired for the particular job. This is a very modular approach and allows relatively easy field modification of the equipment. Also, time to repair will be low because relatively few types of modules will have to be kept in spare parts inventory.

The inherent flexibility of a small computer in forms of control operation make it a natural choice for many forms of audio automation. Returning to our example of a radio station, while a hardware logic controller would be useful for such a simple operation as monitoring the performance of transmitters and adjusting them to satisfy particular conditions, how about the more general problem of station operations? Obviously, the problems involved for such a function—essentially, augmenting the work of the station engineer—are considerably more sophisticated than those for the more straightforward one of transmission control. Here we require a system that is able to make rather sophisticated judgments, that is reasonably flexible about how it is going to achieve its ends, and that is "smart enough" to be able to call for help when it runs into a problem beyond its capacities.

Such a system will have to monitor and correct transmitter power output, check for frequency drift and correct for any deviations beyond the established limits, monitor and be prepared to accept any emergency inputs such as Civil Defense information, check component performance to determine whether any malfunctions have occurred that would affect output performance (*e.g.* an overmodulated signal), and keep a record of station performance.

Such a system would be built around a small computer. And here we can see how a small computer can act in a partnership with simpler logic systems, for the subsystems used for regulating the transmitters would be virtually identical to the system described above. The only differences would be that there would be monitoring outputs from the control unit to the computer, and control outputs from the computer to the system. Other aspects of the system would not be quite the same as the control circuit for the transmitters; a diagram of such a system is shown in FIGURE 2.

In addition to the control system for the transmitters, a similar control system would be devised to control the program source (*i.e.*, whether information is to come from stored tapes, from live studio operations, or from the Civil Defense announcer). In the control system, other parameters, such as relative audio levels, degree of source modulation, and the like, could also be monitored and corrected if necessary.

For the audio amplifier, its performance could be monitored by the computer through an analog-to-digital interface. Corrections, if required, would be supplied by the computer through control signals transmitted via a digital-to-analog converter.

Program schedules could be stored in the computer's mass storage device, which could be either magnetic disk or tape; standby alternatives could be kept in reserve for program emergencies. Also, data could be stored on this mass storage device for later analysis—either directly, or as a printout. By correlating actual events such as a time standard or internal clock, this stored information could form the basis for a comprehensive station log.

And finally, the line from the Civil Defense network could be monitored for possible use. If the correct information was received by the computer through a communication line, it would activate the special unit that would broadcast the alert and instructions.

And because of the sophistication of the programming, the minicomputer could signal the engineer through some input-output device, most probably the unit's Teletype.

The computer is ideally suited to remember all those operating procedures that one must now commit to memory. Again, there is a great amount of standard equipment available today to do these jobs, and the customized interfacing between computer and system can be kept to a minimum. And although it might be possible to construct a similar device out of logic gates and memory, thus making it a hard-wired logic system, hard-wired systems of this magnitude tend to be horrendously expensive. Also, such systems are difficult to justify unless a convenient means has been provided for checking them out if they malfunction. Another important justification for using a computer in this application is the ease of checkout by means of special programs called "diagnostics." These programs can help pin-point problems in the system, and most important of all, could be incorporated into the regular computer routine to do periodic checks on the status of the equipment. This feature is fairly easy to do with a programmable computer, and extremely difficult to achieve in hard-wired logic. In addition, changes of the system's performance characteristics are much easier to implement by developing and reading in new programs as opposed to having to make physical hardware changes. To be sure, we have only touched the surface of sudio-engineering automation. Obviously, even the simplest of projects requires a great deal of thought to determine the best way of doing the job. The intention here has been simply to indicate typical systems-level approaches that can be employed. Solid state logic, minicomputers, and interfaces are new tools available to audio engineers and technicians. They are here today, offering off-the-shelf solutions to many problems. So perhaps in the near future you will discover that your pet project has a natural—and inexpensive—solution through automation techniques. ■

ACKNOWLEDGEMENTS

Eric Small, chief engineer—WOR-FM
Irv Joel, chief engineer—A & R Recording Studio

RICHARD FACTOR and STEVEN KATZ

Automatic Tape Transport Control

Several automatic tape search units are now available. One such unit, available from Ampex, is described. The applications of this sort of device may be endless.

It is 2:30 in the morning and you've been in session since 3:00 the previous afternoon. The producer looks into your glassy eyes and says sincerely, "OK . . . let me hear the beginning of take 20, the bridge of take 7, and the ending of take 17." Only 2500 feet of tape separates you from a good night's sleep.

In prehistoric days, (*i.e.,* before 1970) finding the requisite selections would have involved searching for each slate, finding it, and then waiting for the appropriate section. You *can* skip ahead to find the section, but are you *sure* you're still in the right take?

Within the last three years, the major professional tape recorder manufacturers have introduced various electronic devices to simplify and speed up this time consuming task. This article will discuss the method of operation and applications of these devices.

Basically, the new devices are digital counters which are coupled to the tape by electro-mechanical sensors. They give an indication of how much tape has passed the heads from the feed to the take-up reel. The least complex of these units counts the number of tape turntable revolutions and gives an index number proportional to the number of revolutions. Other units are capable of automatically shuttling the tape transport back and forth to reach a number which can be preset from the front panel.

At least one of these units, the Ampex Tape Search and Control unit, is capable of measuring the elapsed time of the tape in minutes and seconds and searching for any location on the tape to the nearest second. Using such a device in the session alluded to enables you to find takes 20, 7, and 17 just by dialing in the starting time of the take and pressing a button. The engineer is now free to fill in his take sheet, move a microphone, readjust levels, or what have you, while the tape transport automatically cues itself.

HOW IT WORKS

The heart of the tape search and control unit is the counter/memory which determines the current tape location, (see FIGURE 1). This circuit is comprised of several integrated circuits which accept input pulses and count up or down depending upon the direction of tape travel. The total number of pulses (up counts minus the down counts) is stored and sent to the display and comparison circuitry. It should be noted that one reason these search units have come out so recently is that the circuitry involved in just this counting/storage function would previously have required a rack cabinet full of equipment.

The authors were responsible for a digital delay line article that appeared in May. Steve Katz is chief engineer of Sound Exchange Studios in New York City. Richard Factor and Mr. Katz are both v.p's of Eventide Clockworks, Inc.

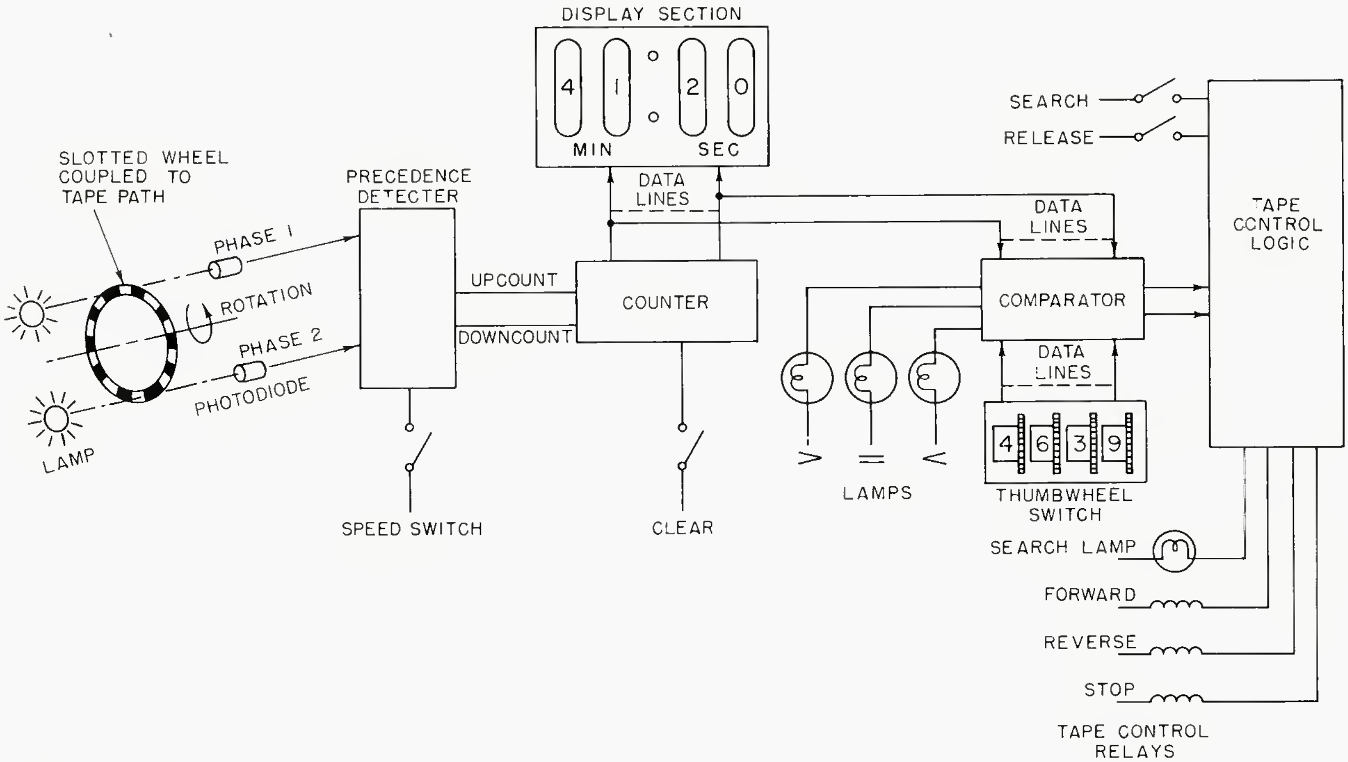


Figure 1. A block diagram of the Ampex tape search and control unit.

The counter receives up/down pulses from a tachometer located on the tape deck. To derive an index number, rotation of the tape turntables is sensed. To determine actual elapsed time, the tachometer must be in the tape path, and its rotation must have one-to-one correspondence with the amount of tape passing over it. The advantage of this approach, in addition to giving precise time intervals is that the tape may be transferred from machine to machine, or have leader added to the beginning or end without changing the timing. A turntable rotation sensor is sensitive to tape slippage caused by air packing in the reel, and to the amount of tape on the reel.

The tachometer produces two signals which differ in phase so that direction may be determined. By sensing which of the phases comes first, the other can be locked out and the appropriate up count or down count derived. The frequency of the dual phase signal is proportional to the tape speed. This signal can be generated magnetically or photo-electrically.

HOW THIS IS USED

The stored position information is conducted to the display unit. This information is in binary-coded decimal form. Since the Ampex Search and Control Unit uses Nixie type readouts, the bcd information must be converted into decimal form. A single integrated circuit, replacing many transistors, performs the conversion for each digit. Thus, the display gives a continuous indication of the current tape time.

The bcd information going to the display also goes to a circuit known as a comparator. The other information going to the comparator comes from the front panel thumbwheel switch, with which the engineer can select the desired location. The comparator takes both sets of data and determines which is numerically larger, or if they are both equal. Determining which is larger is all that is necessary to decide which direction of tape travel is necessary to obtain equality.

SEARCH SEQUENCE

You have just finished take 5; the display is reading 20 minutes and 33 seconds. You want to go back to the beginning of take 2, at 4 minutes and 3 seconds. You set the thumbwheel switches at 0403 and depress the search button. The comparator determines that the tape must travel in a reverse direction to reach the desired time, and the reverse relay closes. The tape speeds up and zips back to 4:03. Suddenly the comparator sees that the times are equal, and then the forward relay closes, and the tape slows down, stops, and begins going in the forward direction. As it passes the "equals point," it again slows down and reverses. This process repeats several times with progressively decreasing tape travel until finally the tape reverses direction during the one second interval during which the display time and the thumbwheel switch-set time are equal. When this occurs, both the forward and reverse relays are released, and the stop relay is activated for a short period.

Thus, the tape is shuttled back and forth in decreasing lengths of travel and then finally stopped at the precise time selected. One consequence that may not be immediately apparent from the above description is that the tape search unit also acts as a motion-sensing unit. Since it stops the tape at the instant of reversal, the brakes are activated and the machine comes to a halt *only when the tape is not moving.* Thus, all starting, shuttling, and stopping is accomplished with the torque motors only, and brakes are never used. This means that operation of the unit is not dependent on a built-in motion-sensing unit, and that the control unit may be employed not only with the MM1000 and MM1100 16-track machines, but also with Ampex and other make ¼-inch tape recorders. It also means that tape handling is substantially more gentle than when a mere human runs the machine.

CONTROLS

In addition to the display and thumbwheel switch referred to above, there are several additional controls on the unit. The search button initiates the searching operation; the release button terminates the search before completion if for some reason the engineer changes his mind. The clear button is used to reset the counter/memory

Figure 2. The search unit.

either at turn on or the beginning of a tape. This, in effect, can be used to set a reference point as described later. The 15/30 or 7½/15 in/sec switch is set to the current speed of the tape transport, to assure that the timing information will be correct.

Several indicator lamps other than the display are provided, two to indicate in which direction the tape must travel to reach equality, one to indicate when the times are equal, and one to indicate when an actual search operation is occuring.

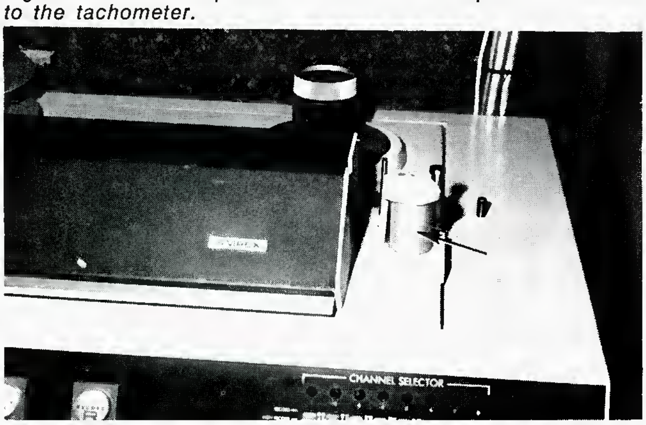
Since the tape search and control unit can be used with virtually any professional tape recorder, a panoply of applications suggest themselves. In addition to uses in recording studios, there are uses in broadcasting, cutting studios, film work, and others. We will take these in order and give some suggestions and applications in each field. After that, we will discuss some of the future possibilities in automatic tape control, using both search units and other devices and techniques.

RECORDING STUDIO APPLICATIONS

In a typical recording session, the search unit would be used for timing and control functions. After breaking out a new reel the tape would be advanced about one minute in and a grease pencil or splicing tape mark would be made. This mark will henceforth and for all time be the permanent reference for the tape. All time markings written down for permanent record will be elapsed time from this mark. This mark is centered over the play head and the "CLEAR" button on the tape search/control unit is depressed, setting the time to 00:00.

As the session progresses, each take beginning is

Figure 3. On an Ampex MM-1000 the arrow points to the tachometer.



marked on the box or take sheet. If it is necessary to go back to any take, this time record gives the thumbwheel switch setting necessary to go to the top. Likewise the time within the take can be noted for each musical feature. Thus if a return to the first bridge is desired, it can be precisely returned to, with no danger that it's actually the second bridge. If it is desired to work with one take, the zero can be reset to its beginning, and a little mental arithmetic (adding the starting time to the time into the take) can be obviated.

BROADCASTING APPLICATIONS

The broadcast industry is more accustomed to automation than the recording industry. Frequently entire stations' air operations are automated, from music selection to logging. And yet, the day-to-day operation of stations in production, news gathering, air checking, etc., is almost inevitably manual.

A common procedure in station news rooms is to provide a tape to record telephone conversations, both for broadcast and legal records. Typically a day's worth of conversations will be recorded, and the news editor will later abstract one section of one or more conversations. Marking the tape time of each cell or event will allow instant location. Trying to wind back to the appropriate flag more often than not just leaves a problem for the janitor (and in a small station, the engineer *is* the janitor). Locating spots in an air check is another possible application, when the dj starts a segment, he can put a tape up and record the show from then on, logging the starting time of the tape on the box. If it becomes necessary to locate a spot for a sponsor (or a violation for the FCC), it is just necessary to put the tape back on and run it to the proper time. For operations with little cohesiveness in programming (top 40 rock, for instance), where almost every segment is under three minutes, and most are only seconds, it saves a lot of time to find things right away.

Having bi-directional search capability with thumbwheel switch control allows one to select a given time even if the tape has not already been at that location, facilitating this task.

OTHER APPLICATIONS

The timing function alone is of inestimable value in an operation such as a cutting room, in which tape timing is necessary. Say an album is sent over with no timing information. The engineer must know how long the cuts are to determine the level and pitch at which the disc must be cut. Unfortunately, timing normally takes just as long as cutting, raising the cost of the operation intolerably. With the tape search unit in the timing mode, determining the length of the tape takes no longer than running it fast forward to the end and reading the display number. Also, if the tape is unleadered, one knows when to spiral by timing information alone.

So much for the current applications of the tape search unit. The list is not exhaustive, but it does show some of the things that you can apply it to in your field.

POSSIBILITIES IN AN AUGMENTED SEARCH SYSTEM

None of the devices to be suggested are offered for sale by any manufacturer, but they are all possible with modifications to the basic unit, and do not require any particular advance in the state of the art before becoming feasible.

A simple auxiliary circuit would enable one to store location information in a random access memory. It would be possible to, say, log the location of each selection directly in the search unit, simply by pressing a key corresponding to the selection number and simultaneously depressing the "store" button. Depressing the number key and the "retrieve" button will automatically send the tape back to the top of that selection. This has applications in recording, broadcast automation, etc. . . . For tape interchangeability, a small magnetic card reader could store and transfer this time data from one search unit to another. The magnetic card would go in a little pocket on the tape reel or in the box.

Adding a keyboard and a slightly more complicated memory would enable one to partially automate the mixdown function. For instance, if it is desired to mute a track because of noise during some period, one would enter in the *time out* and *time in* for that track, along with the track number or some identifying code. This application is obviously not limited to muting tracks—anything that must be done during the tape playback can be programmed into the memory. The amount of data stored is not limited by the capacity of any code recorded on the tape, as, indeed, no code is recorded, thus saving the track for musical information. Also, there is no leakage problem between an encoded track and a recorded track.

One might think that a one-second resolution would be inadequate for this application. However, the tachometer on the tape deck actually puts out 48 pulses per second (at 15 in/sec), thus dividing the entire tape into approximately 20 millisecond increments. Any one of these may be selected for an event to occur.

One slight disadvantage of this system when compared with a tape-encoding scheme is the possibility of tape slippage. The tachometer is normally in perfect contact with the tape and no slippage occurs. However, in fast forward or rewind, a small error is possible, especially if there is paper leader in the tape. This error is rarely more than a second, which is irrelevant in normal operation, but critical in a mixdown application. If it is necessary to rewind to a precise spot, a small piece of aluminum foil can be placed at the precise beginning reference and this be either found photoelectrically or used as the start reference.

Sometimes it is desirable to synchronize two or more tape machines to allow additional tracks to be recorded. The normal procedure for doing this is to record both tapes with the SMPTE time code and have two time code readers compare the tapes and adjust one to the other. This works very nicely, but costs about ten thousand dollars. An alternative is to use two search units and a photoelectric strip as described above to start the tapes simultaneously. If the two tape machines have d.c. servo control, they can be locked by running both machines from one control track. A completely electronic positioning system not requiring a time code is under development.

Certain accessories to enhance tape search unit operation are possible. For instance, remote readout units can be employed to indicate to the performer and producer the elapsed time from the beginning of the tape. Such a remote unit can work in several modes—it can be slaved to the main unit and read the same time, or it can be reset independently and provide an elapsed time from the beginning of a take while the master unit will preserve the absolute time.

We have described some of the applications for automatic tape search and control equipment. Low cost, versatile equipment of this nature has recently been rendered possible by the decreased cost of complex integrated circuits. As this trend continues, we anticipate the introduction of digital equipment with built in calculation or computation functions, as well as other more elaborate functions yet unanticipated. ■

WALTER JUNG

Automating the Audio Control Function, part 3

Ongoing applications of digitally programmed switches (dps) - a subject particularly of value to the audio pro that is beginning to become involved with some aspects of automation.

LAST INSTALLMENT we began to apply the dps to switching and control of audio signals, and explored a number of configurations. Since we left off on switching matrices, this would be a natural point of departure into the next series of applications.

It turns out that the factors* that make the dps switching structure a natural one for switching bus and matrix applications also lend it well to related applications—as digitally programmable attenuators and amplifiers.

FIGURE 1(A) is a sketch of the basic scheme in which the dps can be used as a programmable attenuator. Switches S1-S"N" are the digitally-controlled switching elements, here shown as ideal switches. Each switch controls an input current which will be in proportion to Ein and its respective input resistance. For any combination of S1-S"N" on conditions the net gain will be $\frac{Rf}{Req}$, where Rf is the feedback resistor and Req is the equivalent parallel resistance of the switching network. In effect this stage is a special case of a switching summing amplifier, where instead of a number of different audio lines being switched and mixed, we have a single audio line being switched through parallel paths, with different *weights* for each switch combination, thus changing the net gain.

The configuration is a flexible one, as any desired gain range and gain per step may be selected by choosing the appropriate resistance ratios and number of steps. FIGURE 1(B) illustrates how this would be accomplished using a number of dps switching elements, 1-"N." The total range of gain change will be:

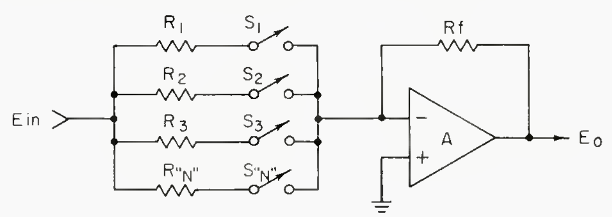
ΔAv total $=$ ΔAv per dps xN
where N $=$ number of switches,
ΔAv $=$ gain change per switch and ΔAv total $=$ total gain change

Thus, for instance, a gain change of 60 dB would require 6 switches of 10 dB steps or 10 switches of 6 dB steps—depending on how fine you want your steps. The beauty of this system is that you can readily manipulate it to suit differing requirements. An example is FIGURE 2.

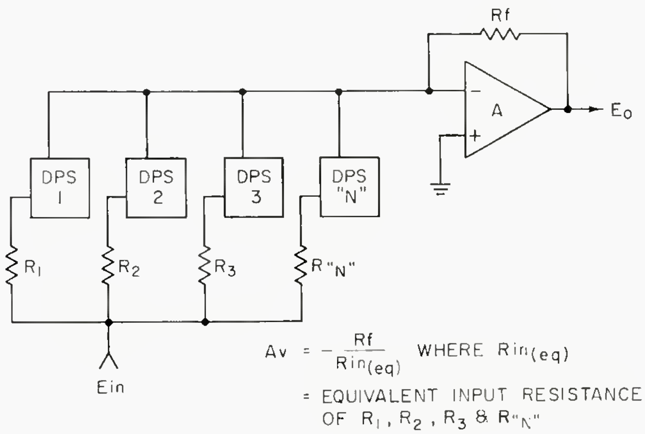
It is quite interesting to note what you can do with a programmable attenuator such as this. When fed a digital word such as might be generated during an automated mixdown process, a digital attenuator can directly perform the attenuation function without the intermediate hardware steps of digital to analog conversion and voltage controlled amplification. An interesting and useful side benefit of this approach would be a direct readout of attenuation in dB via l.e.d. indicators driven from the storage register driving the digital attenuator. A block diagram of this system is shown in FIGURE 3.

Dps elements may be used within an amplifying feedback loop as well as at the input as described in FIGURE 1(B). FIGURE 4 illustrates how this is done in another gain ranging circuit. In effect all we have done is reconnect the input of the dps bank to the amplifier output, and feed our input signal to the summing junction via a single fixed resistor. The effective feedback path is varied by the circuit of FIGURE 2. The end effect is quite different however, as in this case gain is raised as the dps's open, the direct opposite of FIGURE 2. This then is a programmable gain amplifier.

Figure 1. (A) is a programmable attenuator. (B) is the dps programmable attenuator.



(A)



(B)

*see appendix of "Automating" Part 2

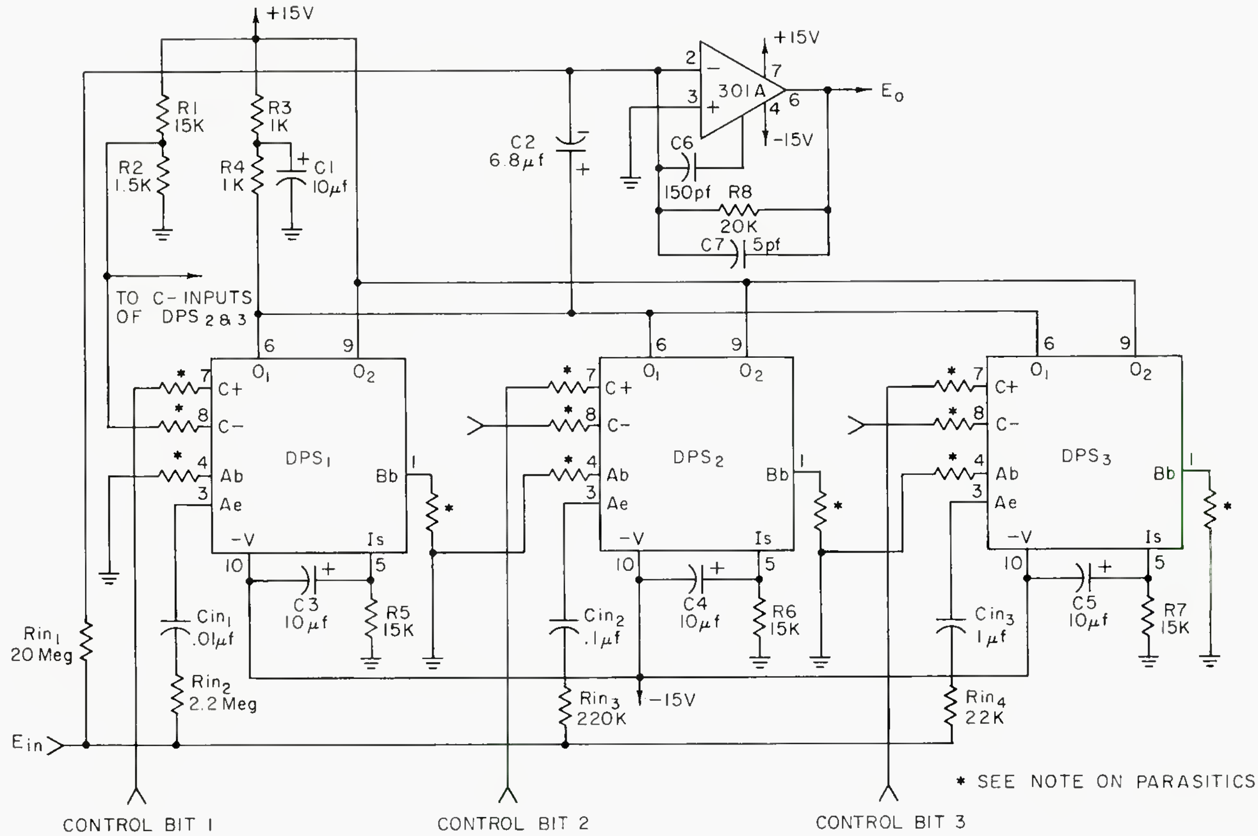


Figure 2. A programmable attenuator. A practical example of the attenuation technique we've been discussing is shown in this figure. Here three dps switching elements are set up for a total gain change range of 60 dB, ranging from 0 dB insertion gain (all switches on) down to —60 dB gain (all switches off). Signal handling capability is +20 dBm at both input and output, distortion less than 0.05 per cent, and with a low-noise op amp for A1, s/n can be in excess of 90 dB at max output. The gain state of the attenuator is under control of the three digital lines (1 per switch) with control logic as in Table I.

Gain is set by the ratio of the combination of R_{in} resistors to R_f (in this case RB). With all three dps elements off, gain is $\frac{R8}{R_{in}1}$ or $\frac{20K}{20M}$ which is $\frac{1}{1000}$ or —60 dB. Switching on DPS1, changes this to —40 dB, with DPS1 +DPS2 it becomes —20 dB, and with all three dps elements on gain is 0 dB.

Several features of the circuit are worthy of note for those who might want to duplicate or modify it to suit special needs. One of them is the decoupling from the +15-volt supply line, provided by R3-R4, the split d.c. load return resistors. The decoupling is absolutely necessary unless the supply used has virtually *zero* impedance at signal frequencies, otherwise the off signal at each switch (which appears at 02) will develop a signal across this impedance and feed it into the op-amp summing junction. This will ruin the otherwise excellent on-off isolation of the switches. It might be well to remember that we're talking here of a switch isolation capability of 100 dB—you just can't take liberties and realize this kind of performance!

Another point is a side effect brought about by the busing of dps outputs. Since a dps output is in actuality the output of an amplifier stage, (this is true regardless of the state of the switch) it follows that it does generate noise components. This is a point to be considered when adding units together in bus fashion, as the total system noise increases each time an additional dps is added in parallel. Noise from a dps is caused primarily by the reference current generator which is fed by I_s. Therefore for low-noise applications, bypassing the I_s terminal (C3, C4, C5) to —V is recommended to minimize this current noise, as well as operating at minimum possible I_s. By using this technique in the circuit of Figure 2 a noise level of —80 dBm was achieved.

Table 1. Control states of Figure 2 attenuation characteristics.

	Control Bits		
Gain	1	2	3
—60dB	0	0	0
—40dB	1	0	0
—20dB	1	1	0
0dB	1	1	1

A good application for this technique is as a "distortionless" agc system, shown in FIGURE 4. Here a number of dps elements are arranged to switch the negative feedback applied around amplifier A1. The number of switches would correspond to the agc range desired and gain change per step (as in FIGURE 2). Gain change points are determined by a number of level comparators which sense the relation between the input level and a series of reference voltages. The input signal is level detected by an appropriate detector (peak, rms or whatever form of control response is desired) and fed to the comparator bank. As the input rises, it will progressively

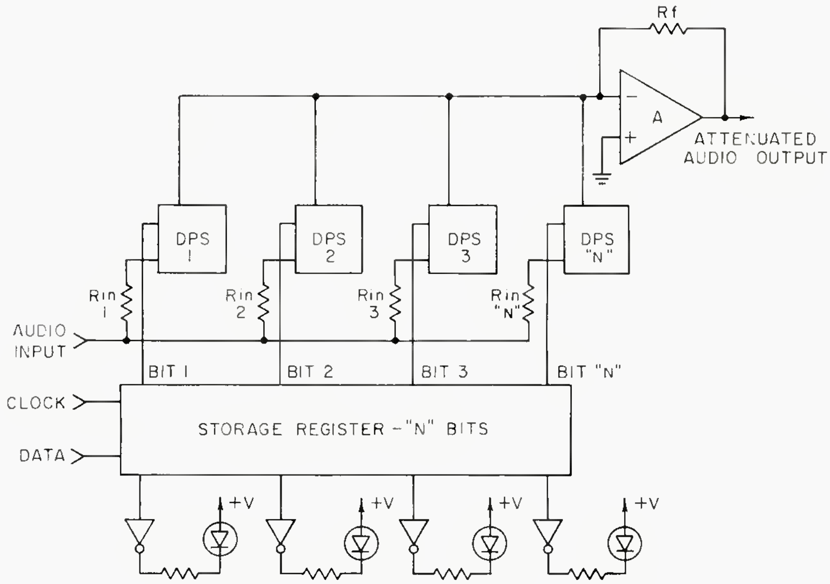


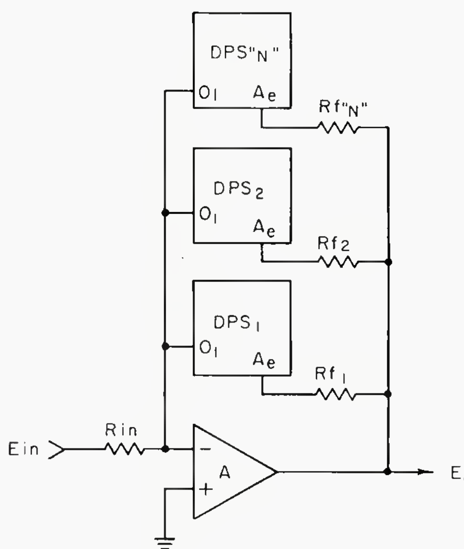
Figure 3. A digital attenuator with an led readout.

become greater than reference voltages E1-E"n." as it passes each level, the comparator outputs change state, activating the dps's and reducing the gain. So, although the input may be constantly changing, the output will stay constant (as long as the input stays within the switching dynamic range of the system). A useful auxiliary feature would be a sequential l.e.d. gain-reduction readout driven from the control logic as shown.

An interesting variation on the level sensing gain ranging amplifier is a level sensitive *gating* circuit used to key program material on or off above a predetermined threshold. A block diagram of this is shown in FIGURE 5(A). The system is similar to FIGURE 4, except for the fact that it uses a single gate function and a variable gain preamplifier preceding the level detector. FIGURE 5(B) is a working circuit of such a system, with a detailed description of the individual elements.

Obviously, the uses you can find for a gating system like this are numerous. Careful control settings will allow undesired background noise to be keyed off while allowing desired material to pass. Or you can slave a number of gates to a common master which can switch the others on or off by appropriate choice of control logic. A single device may be used for either expansion or compression by driving the auxiliary input with a higher (expansion) or lower (compression) level signal. You can key in a special effects signal on the auxiliary input to get some spectacular effects as the unit switches into the auxiliary channel and keys on the "doctored" signal.

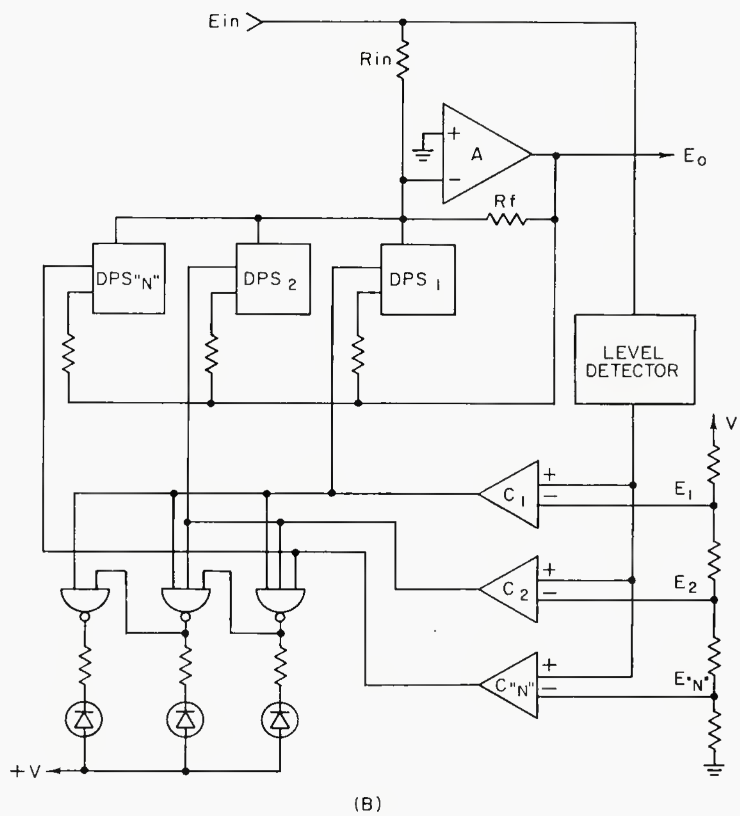
Generally the circuits we have been discussing in this installment have centered around digitally-controlled amplifying circuits. We have seen how they may be implemented in a number of ways. A natural extension of this



$AV = -\frac{Rf(eq)}{Rin}$ WHERE Rf(eq)

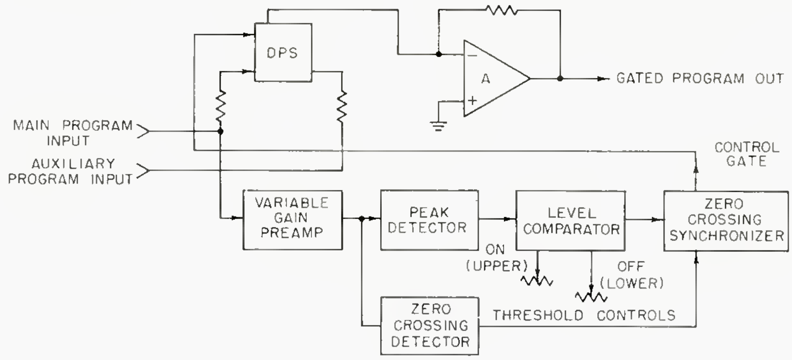
= EQUIVALENT FEEDBACK RESISTANCE OF Rf_1, Rf_2 & Rf"N"

(A)

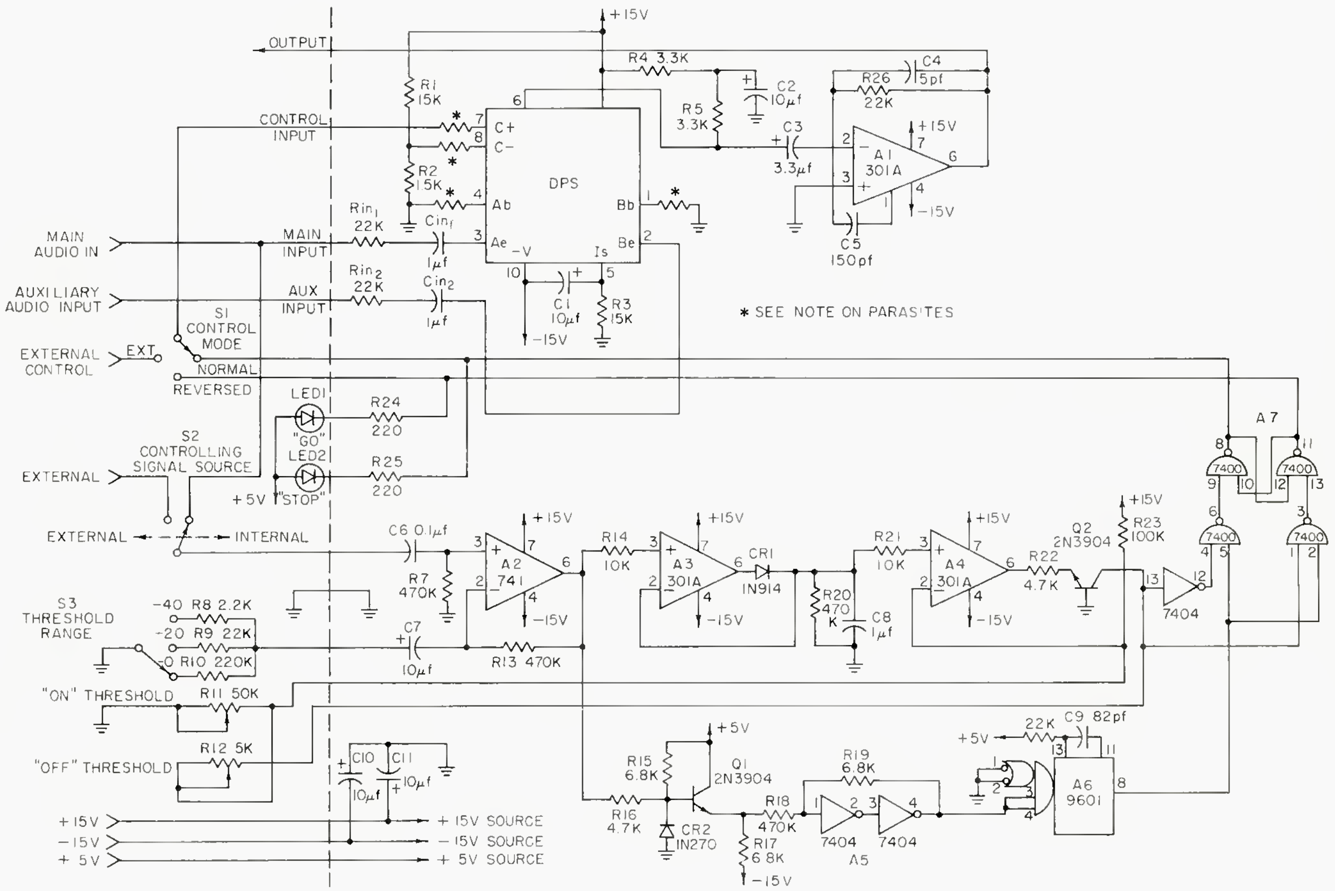


(B)

Figure 4. At (A) a programmable gain amplifier using dps elements in the feedback loop. At (B) is a distortionless agc system with led gain reduction readout.



(A)



(B)

Figure 5. A level-controlled synchronous audio gating system. The level-controlled audio gate system shown in (A) and (B) is a good example of automated signal processing in a couple of senses. Basically, this current is a on-off switch using a dps. However the key to its versatility is in *how* it accomplishes the switching. This it does in quite a sophisticated manner.

Regarding (B), the circuit can be segregated in two main sections. The upper half, consisting of the dps and A1, comprises a switched input op-amp. The entire lower portion of the circuit is used for control signal processing in such a manner that the on-off switching points are not only under precise control, but they may also be remotely programmed.

Look at the circuit, the first function under our control is the controlling signal source selection (S2). This determines whether the switching control signal is to be derived from the signal itself (internal), or from another source (external).

Once inside the circuit, this signal is amplified by A2, where gain is determined by the switch selected feedback resistors R8, R9 and R10. Note that the gain is set by a contact closure to ground—a natural configuration for programmability.

The output of A2 drives two paths, A3 and its circuitry and Q1 and the circuit following. A3 is an operational peak detector which reads signal peaks from A2 and stores a corresponding d.c. level on C8. The level across C8 is then fed to A4, which compares the audio sample to a d.c. reference voltage developed by R23 and R11. When the signal becomes larger than the reference, A4's output swings minus, driving Q2 on and switching R12 in parallel with R11. A new reference level is now established, set by R23 and R12, much lower than the first. When the audio signal drops to this new level A4 flips positive once again, resetting the circuit to its original state. The positive and negative comparison levels just described are the "on" and "off" points of this threshold switch. They are independently variable, and may be set with up of 20 dB of separation between switch points.

The logic compatible signal from Q2 is fed thru a zero axis sychronizing circuit to a control flip flop, A7. Zero crossing pulses are derived by A1 and A5-A6 is the switched two-input amplifier scheme, which allows a number of new possibilities.

As we have discussed above on applications of the level controlled gate, the two inputs of a single dps may be used to remotely switch a pair of signals. These signals may be unrelated, but the uses which first come to mind are the in-out transfer of a normal signal and a processed one, such as eq, reverb or delayed, matrixed or non-matrixed etc. Others may be for 2 unrelated inputs such as remote selection of 2 test frequencies, 2 different program lines; Telco or NET, etc.

The list of uses for a general purpose versatile switching device such as this could go on and on. Among the uses would be programmable filters, multiplexers, choppers, fsk Modems, function generators with programmability, fm stereo modulator or demodulator, frequency scanner and so on. It will be interesting to see to what degree the dps technology is exploited in solving the audio challenges of today and tomorrow.

Future installments of this series will explore other variations of audio automation and programming technology. ■

to ensure that switching occurs at a point of zero potential to minimize switching transients. Both states of the control flip flop are fed out to an external switch which selects whether the signal is to be normally off and thresholded on or vice versa.

Now, with the basics of the circuit in mind, it may be appreciated how the circuit can be used to automate control of audio signals. Signal threshold point switching is variable over a total variation of greater than 50 dB using the variable gain A2 pre amp to move the threshold levels of A4 with respect to the audio signal being switched. Further, the individual on and off switching points are independently variable. All of these functions may also be remotely programmed if desired.

Both control mode and control source selections are remotely programmable. Although simple selection switches are shown, these functions can be automated also by using a dps for S2 and a digital multiplexer for S1. An auxiliary input provides access to the normally off channel. A signal applied to this input will be keyed on during the off periods of the main input.

Component designations and values are provided for those who wish to duplicate the unit. A circuit board and component kit will be available in the near future.

Specification for Figure 5B

Attack Time: 300 μsec worst case, measured with 0 dB tone burst with 0 dBm threshold. Significantly improved with lower thresholds.

Release Time: Approximately 100 msec per dB of difference between *on* and *off* threshold settings.

Insertion Loss: In on state of either channel insertion loss is nominally 0 dB. Off state attenuation ≥ 80 dB.

Frequency response: ±1.0 dB, 20 Hz-20kHZ.

Distortion: Less than 0.1% THD at any signal level within rating.

Output level: 22 volts p-p into load of 2k or greater.

Input Level: +20 dBm max, Z in 22K.

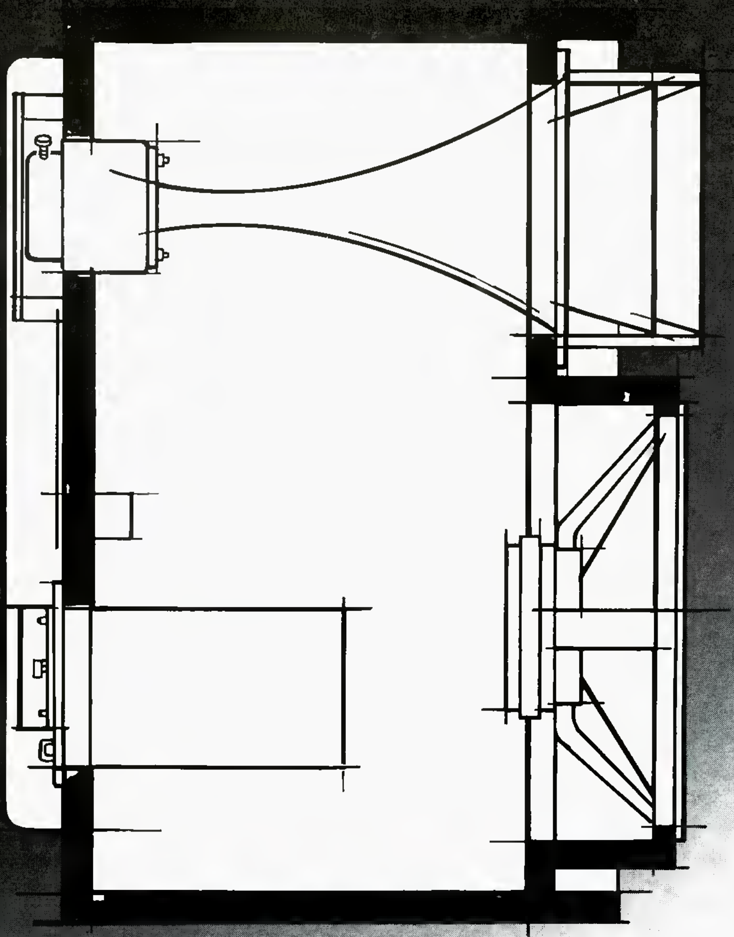
Signal to noise: 90 dB or greater below maximum output.

Control logc: A ttl "1" level at control input connects main input to output. A "0" connects auxiliary input to output. Switching time for either transition ≤ 100 ns.

Control Outputs: 2 supplied; a "1" level ttl signal when control signal input is above threshold plus the complement of this signal. Current drive for led indicators also supplied.

PARASITIC SUPPRESSION

As a practical matter, it is worthy to note the reason behind the otherwise inexplicable resistors which we note on the valued circuit schematics. We refer here to the low value resistances in the base leads of the dps chips. They are parasitic suppression resistances, necessary due to the extremely high bandwith of the 1496/1596 transistors. These speedy devices can oscillate at VHF frequencies and you may never even see a trace of it on the output waveform! The indication that this is happening is the appearance of mysterious d.c. conditions; an output appearing when it shouldn't, d.c. levels wandering, and so forth. You can usually confirm the presence of an oscillation by probing the suspected point with a d.c. probe (or even your finger) while watching the output. If the device is oscillating, this will stop it, and the output will behave. A junction susceptible to this phenomenon can be tamed down to normal behavior by the addition of the "base stopper" resistors which serve as Q spoilers. Its a good idea to include these resistances in any circuit using these chips to insure stability. There are four required, one for each base lead, and they should be placed as close as is physically possible to the respective leads (C+, C−, A_b, and B_b). Yes, in case you are wondering, it is just like a similar problem with old beam power tubes!



Maximum acoustical output:	112 db SPL @ 4 ft., equalized for flat response (pink noise), 40 to 15,000 Hz, in a free field.
Dispersion:	40° vertical x 90° horizontal.
Biamplifier:	Altec Model 771 B. Input sensitivity: .5 VRMS-60 Kohm unbalanced input for full output. .1 VRMS-600 ohm balanced input for full output.
Crossover:	500 Hz, 12 db/octave.
Noise (both sections):	80 db below rated output.
LF section:	60 watts, continuous sine wave power, 0.3% THD.
HF section:	30 watts, continuous sine wave power, 0.3% THD.
Dimensions:	26.5″ wide, 31″ high, 23.5″ deep.
Weight:	112 lbs.
Finish:	Grey epoxy with black grille fabric.

The New Altec 9846 Biamplified Studio Monitor Speaker System

If stereo disk cutters go down to 30 Hz, shouldn't the monitors?

Yes. Because the music goes down that far.

You should hear everything when you're laying down tracks or you might not really know what you've got. Why wait till you play a test cut to find out what's at the low end? You don't have to with our new monitors.

What have we done differently to get that low end? First of all we're using sealed boxes. No more ports or bass reflex cabinets. We've gone to a very high-compliance speaker with a big magnet structure. It's well damped so that it responds accurately to signals down to 30 Hz. There's very little distortion. You can't get this with ported boxes.

Above 500 Hz things are different too. There's far more smoothness than in previous designs. Notice the proximity of the woofer and the HF horn. They're within inches of each other. This creates a smooth transition from one source to the other as you go up the spectrum.

Have you ever wondered why good monitor systems are two-way systems? It's because they guarantee transient accuracy. They don't have the inevitable problem of source displacement that occurs in systems with separate tweeters. And they don't burn out, as tweeters in most 3-way systems are prone to do, with the super highs present in much of today's music. Few drivers are capable of such a wide frequency range. Because so few speaker builders use the kind of phasing plug which makes this possible.

To make the system work, power has to be delivered. Biamping is the only solution. So we built a biamp right into the enclosure. It provides more than enough power to make the components produce more than enough sound. Depending on the music, the 90 watts of available power may be equivalent to three or four times that amount when compared to single amplifier systems.

This is truly a new recording tool.

Write for further information on the 9846 Monitor Speaker system.
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PROFESSIONAL STUDIO PRODUCTS

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DAVID E. BLACKMER

A Wide Dynamic Range Noise-Reduction System

Noise in tape recording is examined and a system for reduction that is now in use is described.

TAPE RECORDING has been one of the major sources of noise in professional studio systems .The almost universal application of multitrack techniques has accentuated the problem. The widely-used Dolby system gives just enough noise reduction to offset the 12 dB noise buildup inherent in sixteen-track recording. The continuing demand for improvement in quality delivered to the consumer makes further evolution in noise-reduction systems mandatory.

Several forms of noise reduction have been developed which selectively restrict the frequency response of the playback system alone. Signal components 40 or even 50 dB below program level form a vital part of perceived sounds. These are buried in the background noise and cannot be separated from it without extremely elaborate systems hence program quality is degraded. We have therefore restricted our interest specifically to those systems capable of complementary coding and decoding over a wide dynamic range.

A number of possible techniques exist for signal coding before recording and decoding on playback. The most practical appears to be the form which uses the signal itself to carry the requisite information for decoding. Such systems are equivalent to compression before recording and expansion on playback. Many of these have been used extensively. The Dolby A system is a four band compressor-expander. Several recording companies have used 1:2 compression factor companders for tape masters. Oppenheim and Stockham at MIT's Lincoln Laboratory devised an ingenious linear decibel compressor-expander using logarithmic amplifiers. Burwen has described a 3:1 compression factor system using peak signal detection capable of 110 dB range between peak signal and background noise. And of course, long distance telephone circuits have been equipped with companders for more than 30 years.

The quality and usefulness of a noise-reduction system for professional studio aplications can be judged on dynamic range coverage, accuracy of dynamic tracking, especially on rapid rise transients, and ability to handle the imperfections inherent in the tape recording process gracefully.

It was our objective in this work to develop a system capable of recording the entire useful auditory dynamic range with excellent transient tracking. It must operate with existing recorders and be producable for a practical price.

David E. Blackmer is president of DBX Inc., of Harvard, Mass.

DESIGN EVOLUTION

A linear decibel compressor-expander may be accomplished in the manner shown in FIGURE 1. Voltage controlled amplifiers control the input and output gain. Input and output level sensors control gain with an appropriate control polarity to produce compression in section A and expansion in Section B. Such a system might seem practical as a noise-reduction system as it can multiply the recorder dynamic range by the compression factor. A 60 dB recorder with a 2:1 factor system could yield a 120 dB signal-to-noise ratio. Unfortunately, the instantaneous background noise level varies with the signal envelope. The human ear can tune out a steady hiss to the point where the listener is only subliminally conscious of it. A varying noise envelope has a much higher auditory annoyance factor. Consequently this system is quite unsatisfactory for critical applications although its performance looks outstanding on paper.

TAPE NOISE

It is important to understand the behavior of the noise sources in tape recording. We are all familiar with the sound of the background noise in the absence of signal. It typically has the spectral distribution shown by the lower curve in FIGURE 2. When a signal is recorded the noise increases. The other curves show the resulting spectral distribution when a 180 Hz tone is recorded at various levels. Note that the noise in the entire spectrum is increased with increasing signal. The increase is greater near the tone frequency. The broadband noise increase is due to the increased number of magnetized domains passing by the playback gap. The increase near the tone is due to inhomogenity in magnetic particle distribution in the tape coating and to surface asperities which lift the tape away from the gap as they pass through. Both these mechanisms cause an amplitude modulation of the signal by random functions which of course results in noise sidebands. The upper curve shows the background noise accompanying a +10 dB record level corresponding to the recording level on program peaks. The entire high frequency noise spectrum has risen. Note that the instantaneous signal-to-noise ratio of tape recording is nearly constant above −10 dB recording level. This causes an apparent increase in the high-frequency content of tape recorded signals.

With most audio signals, masking effects cover up noise effects within several octaves of the frequency band containing the dominant energy, but background hiss which is 50 dB under a strong low frequency signal can be plainly heard.

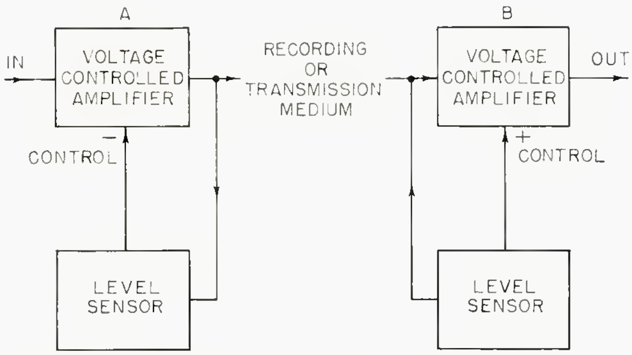


Figure 1. A linear compressor-expander system using voltage controlled amplifiers.

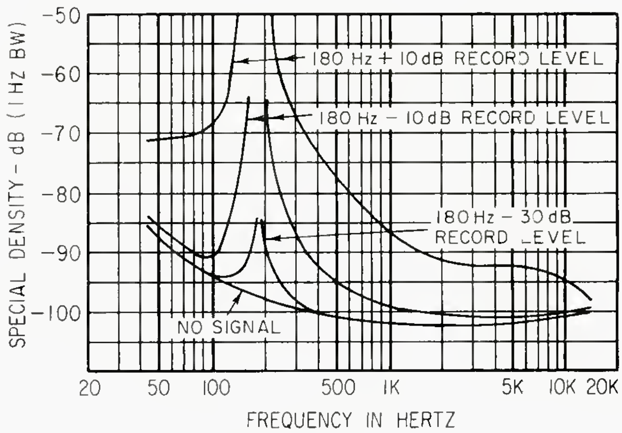


Figure 2. Noise in tape recording.

HIGH-FREQUENCY PRE-EMPHASIS

The high-frequency hiss changes accompanying signal envelope level changes may be rendered virtually inaudible with high-frequency pre-emphasis. If a 12 dB weighting is used from 400 Hz to 1600 Hz as shown in FIGURE 3, then the high-frequency hiss accompanying a strong low-frequency signal is reduced by nearly 12 dB. This will also cause a sharp increase in susceptibility to high-frequency self erasure which is already a problem without this pre-emphasis with many types of signals commonly recorded in contemporary music.

PRE-EMPHASIS OF LEVEL SENSOR SIGNAL

This problem may be resolved with the use of high-frequency pre-emphasis before the level sensing circuit. Any desired frequency shaping function may be applied to such a compressor-expander system level sensor as long as both record and play level sensor pre-emphasis curves are identical. We have chosen a 20 dB pre-emphasis starting at 1600 Hz which produces the sine-wave response curve shown in FIGURE 3. Note that this is the single frequency sine-wave response curve of the system. The complex wave weighting remains that shown in FIGURE 2 at all times.

TRANSIENT RESPONSE

A system such as this is only useful if it has adequate transient response tracking properties. Peak responsive level sensors seem at first glance ideal. Unfortunately, peaks 24 dB above vu meter reading are common, especially in mixdown. Tape recording levels are commonly run in a region where such peaks are flattened. This peak truncation is free from sudden slope changes, but serious tracking errors would occur between record and play level sensors. Equally obviously, average level sensors would be too slow. Furthermore, most tape recorders have considerable time delay dispersion (frequency dependant phase shift changes not giving constant time delay). Neither the peak nor the average value of a signal remains unchanged with non-constant time delay. The one function which is unchanged is the rms value of the signal. The rms value is the sum of the energy of all frequency components present without regard to their phase relationships. If there were a means to sum the squares of instantaneous signal levels, then the signal rms value could be determined. Fortunately recent advances in analog computation techniques have made this possible. The circuit shown in FIGURE 4 squares the signal by doubling the logarithm then taking the antilog. This squared signal is averaged and used to derive the control signal.

The final system design is shown in FIGURE 5. The input pre-emphasis network boosts the high-frequency content of the signal. The voltage-controlled amplifier controls signal gain in response to the output of the level sensor with a pre-emphasis network feeding the level sensors. The compression factor is set at 2:1. The unity gain point is set at +4 dBm at 1 kHz.

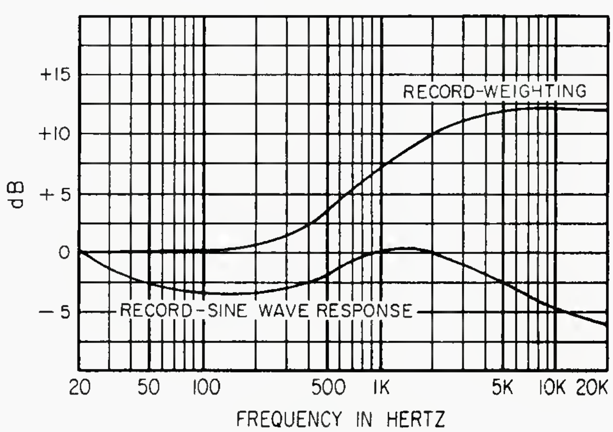
The decoder has an identical level sensing circuit which will therefore give the same control signal as that used in the input compressor. The control polarity to the vca is reverse. The over-all system gain remains unchanged as the two vca's give complementary gain changes. A de-emphasis network on the output restores frequency response.

SYSTEM PERFORMANCE

This noise-reduction system when used with a studio tape recorder has an equivalent "A" weighted background noise under −90 dBm without audible imperfections. A difference in apparent high-frequency response can be heard as the noise reduction system is alternately switched in and bypassed while recording and playing simultaneously with a tape recorder, but carfeul comparison of the original signal with both recorded signals shows that the noise-reduced signal sounds like the original and that the cause of this difference was the increase of tape hiss with record level as shown in FIGURE 2.

There are other barely audible imperfections due to asperity noise with some types of very clear signals but these noise components lie in the same frequency region as the dominant signal energy band hence we must look to improvements in tape surfaces for their reduction regardless of the noise-reduction system used. Incidentally,

Figure 3. High-frequency hiss changes accompanying signal envelope level changes may be rendered virtually inaudible with high-frequency pre-emphasis.



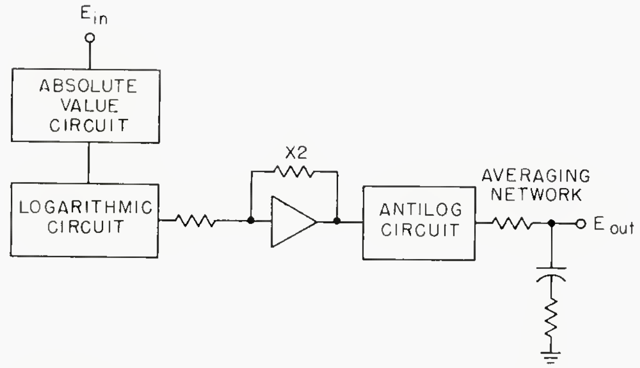


Figure 4. This circuit squares the signal by doubling the logarithm, then taking the antilog.

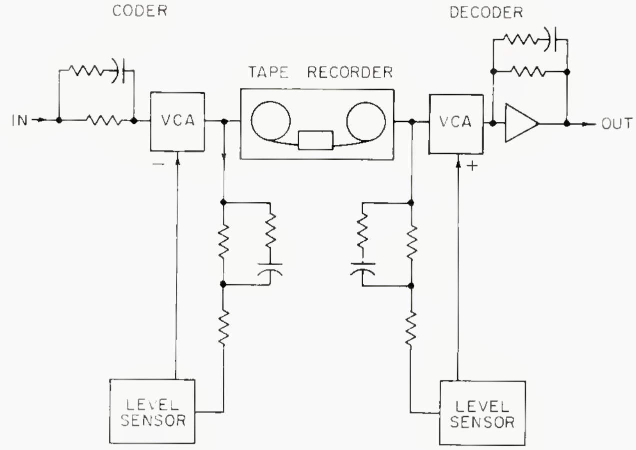


Figure 5. The final system design.

these components can be masked almost completely with a steady hiss. The required level of this "noise perfume" is about −65 dB. It is no coincidence that noise-reduction systems which claim no audible effect have this residual noise level present in the output.

The DBX noise-reduction system handles very well in studio use. No level matching is required for accurate transient response. The complete absence of audible background noise requires some rethinking of recording techniques. Gain riding is unnecessary in all original session recording. There is more latitude for post equalization so that most equalization can be left for the mixdown session. Limiting and compression are only used where required for effect. A number of generations of re-recording may be used without excessive noise buildup.

The signal quality improvement due to this system can give the studio master tape recording a better sound and dynamic range than the best disc recording, thereby significantly improving the quality of audio delivered to the growing number of critical listeners. Only when it is gone do you realize how pervasive background noise has been in recorded sound. ■

Entente Cordiale

A friendly understanding.

A phrase that perfectly describes the relationship between Revox tape recorders and Beyer microphones. One fine piece of equipment uniquely complementing another.

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For example, take the new Revox A77 Mk III. It's the successor to the critically acclaimed Revox A77, the machine that moved Stereo Review to comment, "We have never seen a recorder that could match the performance of the Revox A77 in all respects, and very few that even come close."

Or take the new Beyer M500 microphone. You've never used anything quite like it. It combines the sharp attack of a condenser and the sturdy reliability of a moving coil with the unduplicatable warmth of a ribbon.

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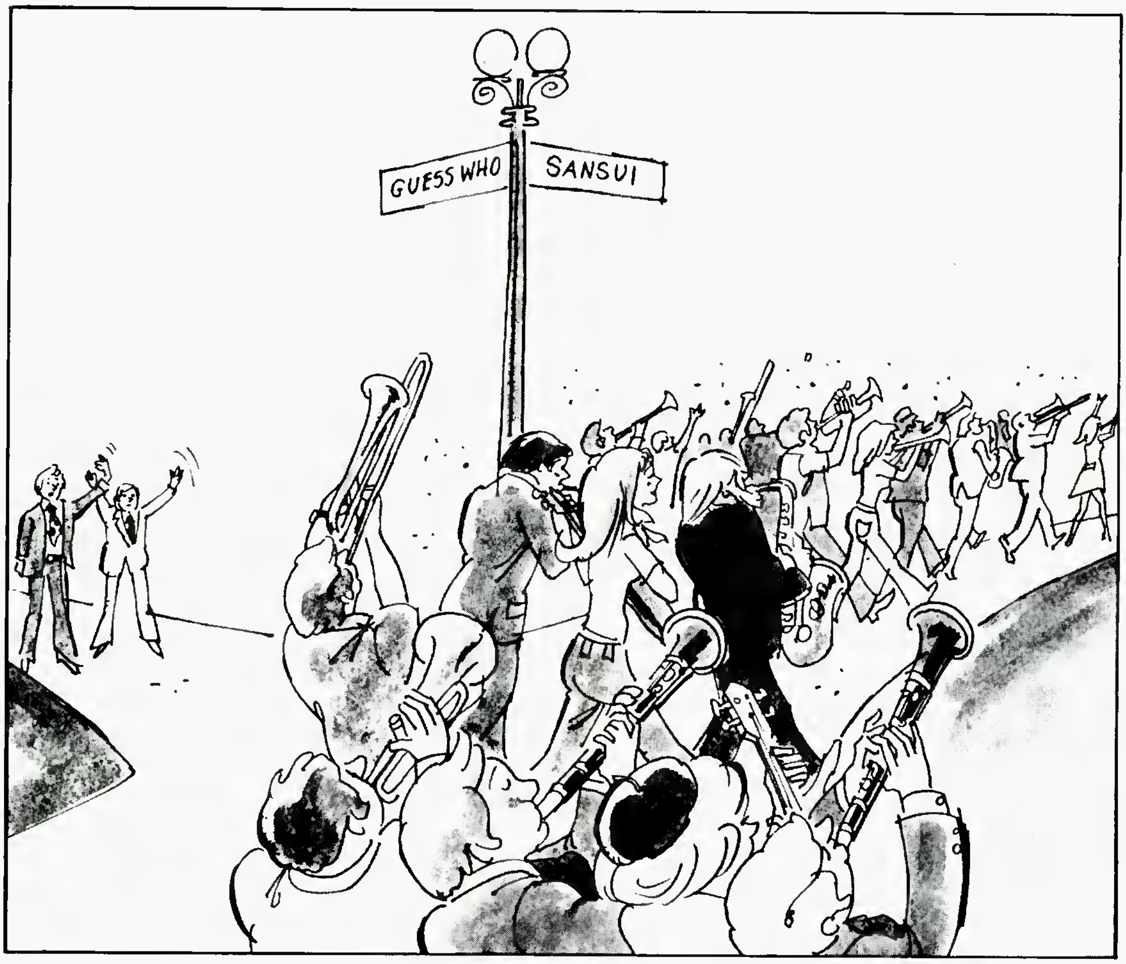
Circle 47 on Reader Service Card

look who's going



You may have heard it already—or you may not have—but more and more recording artists, more and more recording labels, and more and more engineers and producers are using Sansui's QS four-channel technique to encode their records in the four-channel mode. The list, growing from week to week, is quite impressive. But more important than the list itself is the fact that all these independent artists and companies have conducted extensive testing and actual use procedures before they made up their minds. What speaks for Sansui's QS System is not the pressure of a major software company nor exaggerated statements and promises; it's simply the performance.

Whether it is such an outstanding artist as Carole King or Joan Baez; or perhaps eminent musicians and producers like Enoch Light or Dick Schory add to the list such a beloved figure as B. B. King — they all are going Sansui's QS way because they think it is the best way. Some of these artists have actually produced with other matrix four-channel encoding methods, but have found that the most satisfactory results, in terms of the freedom of the producer and the artistic results, are in the one and only balanced and symmetrical system—that is, Sansui's QS method.

the QS way—and why

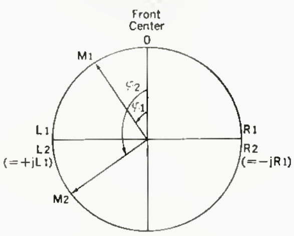
(Report on the Sansui QS Coding System)

There are now almost three hundred — yes, three hundred — Sansui-type matrixed four-channel record albums available all over the world, most of them encoded with Sansui QS encoders. In this country alone, almost 30 albums are already on the market. And we hate to hold anything back from you, but there are a number of artists who will, in the very near future, be on the market with their QS four-channel recordings. Also, major labels.

While we do not know the particular reasons each of these artists and producers selected Sansui's QS System, we know that it could be any or all of those enumerated here. These are the qualifications that make Sansui's QS matrix system uniquely efficient and effective and musically satisfying. In fact, we believe Sansui's QS System is the only matrix system that can claim that it has no known major drawbacks, that are not subject to refinement. These are the features of Sansui's QS system:

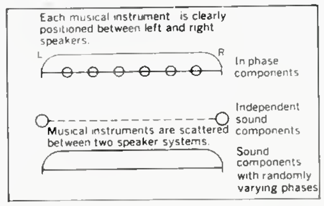
TOTAL, ACCURATE SOUND-SOURCE LOCALIZATION in every direction and at any point *inside* the sound field. No dropouts, cancellations or irritating shifts in position. The "overhead" effect, with a performer in the dead center, is readily achieved. This means there are no problems about having to place performers in some positions and avoiding other areas. It means that the total acoustic perspective is the same as that for discrete tape.

Recording and Localization of Sound Images in 360°



TOTAL COMPATIBILITY. Sansui's QS Coding System is compatible with two-channel stereo playback of encoded recordings. With four-channel playback (ambience synthesis) of conventional two-channel recordings. With other matrix decoders. With all existing home hardware and with all existing professional equipment. With present broadcast standards and equipment.

Location of Various Sound Components in Left and Right Channels



There are many important implications in this comprehensive situation. For example, when QS-encoded material is played back in conventional two-channel stereo, it produces an entirely correct stereo perspective. The rear-channel information serves to produce a broadened and enhanced stereo perspective instead of jamming rear-channel information unnaturally into the wrong places to confuse directionality and obscure the stereo effect.

We believe that it is in the interest of the entire industry that the very best system be selected, regardless of politics, regardless of cross-currents and undercurrents, regardless of alliances and pride or even some dent in someone's reputation.

We have no other ax to grind than to play the fiddle that will make the best music for the industry. If you care to know more about Sansui's QS System, please contact our New York office for a demonstration and materials. It may interest you that the RIAJ (Record Industry Association of Japan) has adopted the Sansui system under the name of Regular Matrix to be the standard for recordings in Japan. An application for acceptance of our standards is now with the Recording Industry Association of America.

It's no wonder, then, that everybody who is anybody in the four-channel medium is going the QS way. Why not join the trend? The QS encoder is very simple to adjust, easy to use and reliable. Try it. Check out our claims with your own material, in your own way. Learn for yourself what the present members of the Sansui QS bandwagon have already discovered.

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43rd AES Convention

THE EXHIBITS

Dates: *September 12-15*
Place: *New York's Waldorf-Astoria*
Exhibition Hours:
Tuesday—*5 to 10 p.m.*
Wednesday—*11 a.m. to 10 p.m.*
Thursday—*11 a.m. to 5 p.m.*
Friday—*11 a.m. to 5 p.m.*
Banquet: *Thursday, September 14th.*
Cocktails— *7 p.m., Gold Room*
Dinner—*8 p.m., Starlight Roof*

THE PAPERS

Tuesday
A. 9:30 Quadrasonics. *John Woram,* Vanguard Recording Society, Inc.
B. 2:00 Broadcast Engineering. *Eric Small,* WOR-FM.
C. 2:00 Transducers. *Mahlon D. Burkhard,* Industrial Research Products, Inc.
D. 7:30 A Forum on the State of the Art. *John Woram.*

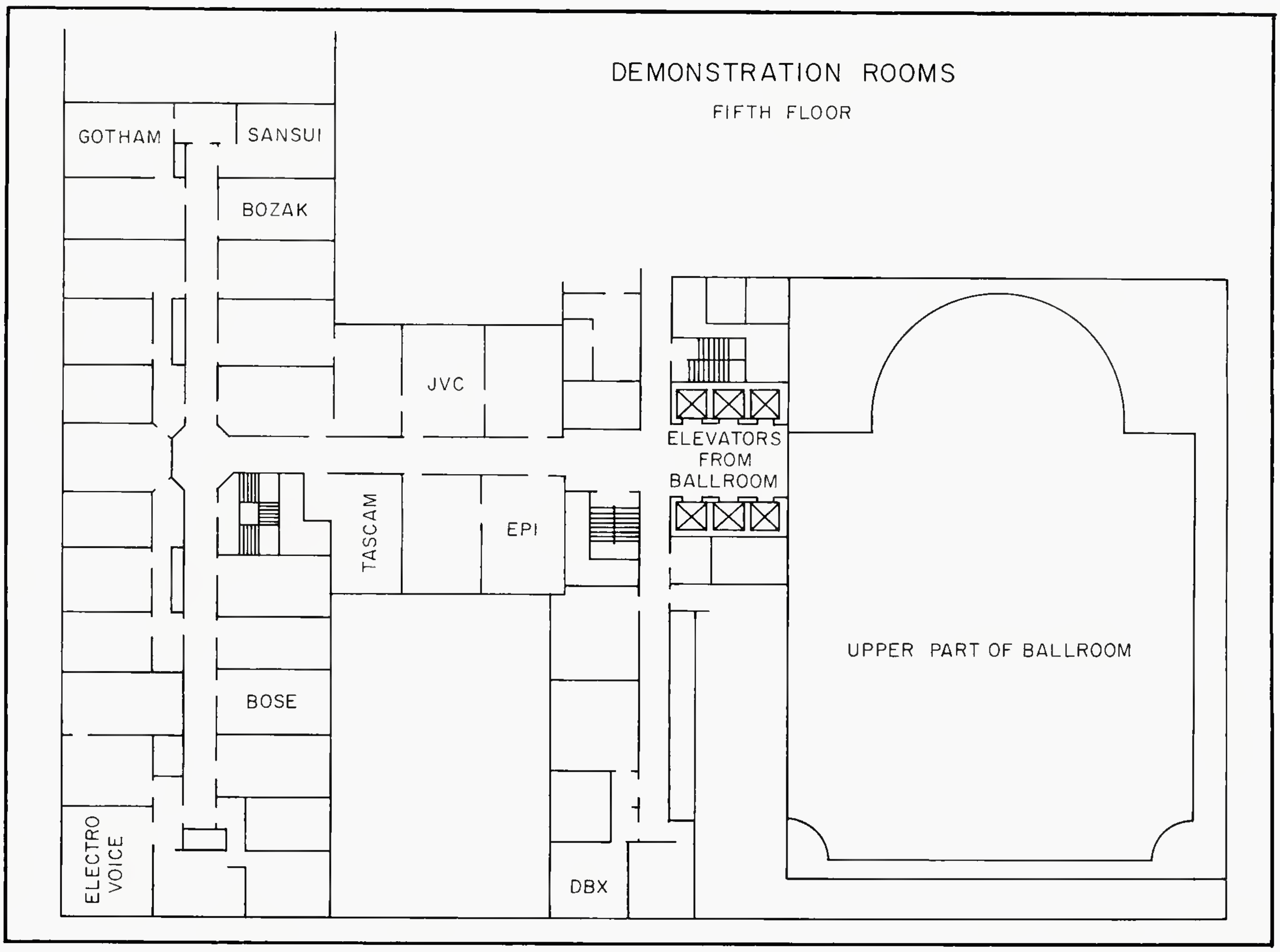
Wednesday
E. 9:30 Recorders. *Arthur E. Gruber,* AEG Associates
F. 2:00 Signal Processing. *Albert E. Grundy,* Institute of Audio Research.
G. 7:30 Tape Duplication Seminar, *Ray Dolby,* Dolby Laboratories, Inc.

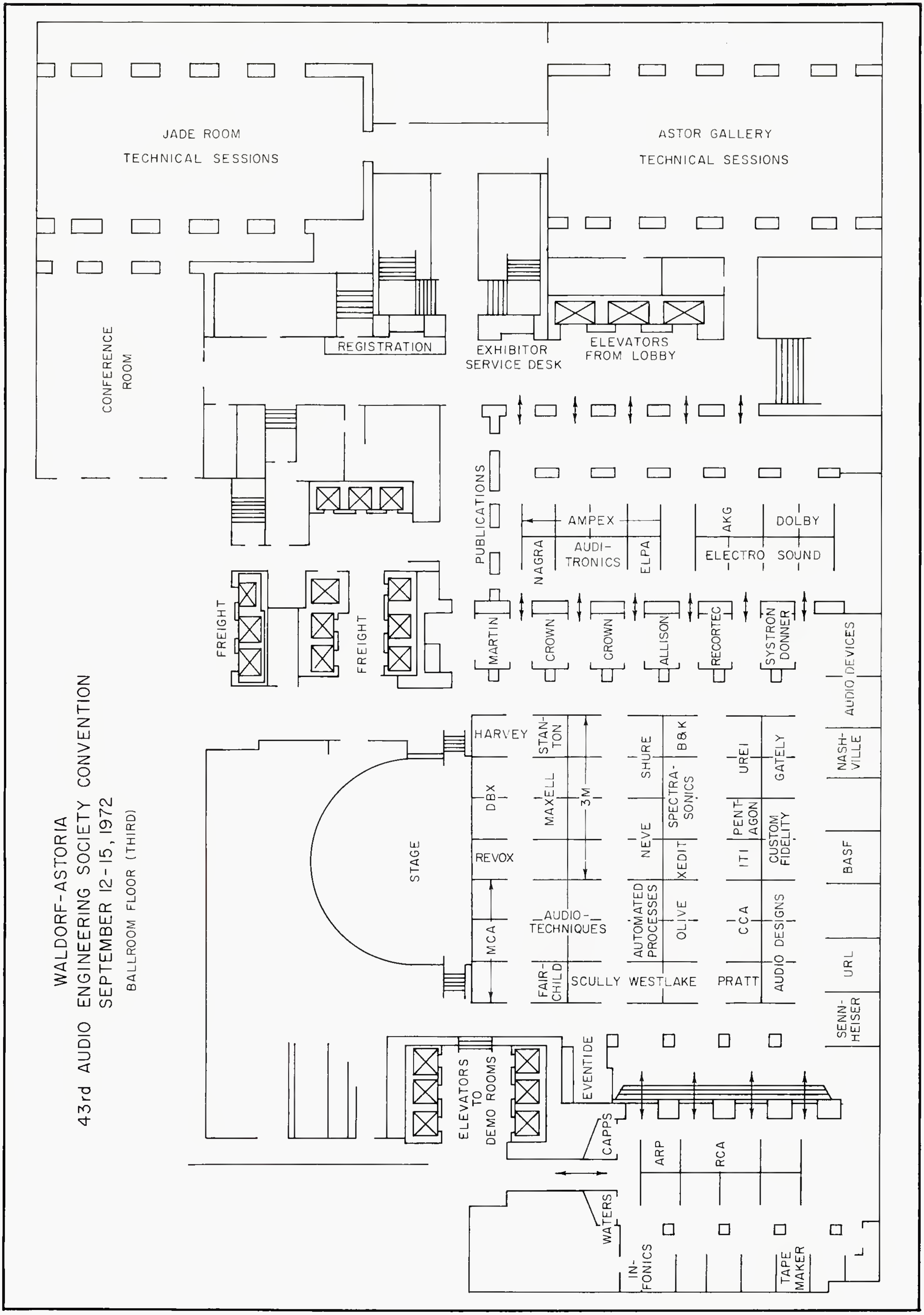
Thursday
H. 9:30 Disc Recording. *John J. Bubbers,* Stanton Magnetics, Inc.
J. 9:30 Sound Reinforcement and Architectural Acoustics. *David L. Klepper,* KMK Associates.
K. 2:00 Audio and Medicine. *Philip Kantrowitz,* Queensborough Community College.
L. 2:00 Audio Instrumentation. *E. E. Gross,* General Radio Company.

Friday
M. 9:30 Audio in Musical Education. *Max V. Mathews,* Bell Labs.
N. 2:00 Electronic Musical Instruments. *Daniel W. Martin,* D. H. Baldwin Company

Technical Sessions B, J, L will be in the Astor Gallery. All others will be held in the Jade Room.

On following pages detailed maps of the exhibits being shown at the convention. Last minute changes after our press time may create some minor changes.

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43rd AUDIO ENGINEERING SOCIETY CONVENTION
SEPTEMBER 12-15, 1972
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FREIGHT
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BOOKCASE

20. The Audio Cyclopedia (2nd ed.). *Dr. Howard M. Tremaine.* New and updated, here is the complete audio reference library in a single volume. It provides the most comprehensive information on every aspect of the audio art. This new edition includes the latest audio developments including the most recent solid-state systems and integrated circuits. It covers all subjects in the fields of acoustics, recording, and reproduction with more than 3400 related topics. Each topic can be instantly located by a unique index and reference system. More than 1600 illustrations and schematics help make complicated topics masterpieces of clarity. 1760 pages; 6½ x 9⅜ hardbound. **$29.95**

1. The Technique of the Sound Studio. *Alec Nisbett.* This is a handbook on radio and recording techniques, but the principles described are equally applicable to film and television sound. 264 pages; 60 diagrams; glossary; indexed; 5½ x 8½; clothbound. **$14.50**

7. Acoustical Tests and Measurements. *Don Davis.* Provides solid understanding of the entire subject of acoustical measurements; based on actual field test work, using commercial equipment. 192 pages; 5½ x 8½; hardbound. **$6.95**

8. Handbook of Electronic Tables & Formulas, (3rd edition). A one-stop source for all charts, tables, formulas, laws, symbols, and standards used in electronics. Includes an 8-page, full-color fold-out chart showing latest FCC allocations for the entire frequency spectrum. 232 pages; 5½ x 8½; hardbound. **$5.50**

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24. Basic Electronic Instrument Handbook. *Edited by Clyde F. Coombs, Jr. Hewlett-Packard Co.* A basic reference background for all instruments. Offers saving in time and effort by having complete information in one volume on how to get the most benefit from available devices, how to buy the best instrument for specific needs. Reduces chances of costly errors. Ideal reference book, it is an excellent source for the beginner, technician, the non-electrical engineering man, or general non-engineering scientific and technical personnel. 800 pages. Hardbound. **$19.50**

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31. Solid-State Electronics. *Hibberd.* A Basic Course for Engineers and Technicians. An extremely practical reference book for anyone who wants to acquire a good but general understanding of semiconductor principles. Features questions and answers, problems to solve. 1968. 169 pp. **$9.95**

32. Circuit Design for Audio, AM/FM, and TV. *Texas Instruments.* Texas Instruments Electronics Series. Discusses the latest advances in design and application which represent the results of several years research and development by TI communications applications engineers. Emphasizes time- and cost-saving procedures. 1967. 352 pp. **$14.50**

35. An Alphabetical Guide to Motion Picture, Television, and Videotape Productions. *Levitan.* This all-inclusive, authoritative, and profusely illustrated encyclopedia is a practical source of information about techniques of all kinds used for making and processing film and TV presentations. Gives full technical information on materials and equipment, processes and techniques, lighting, color balance, special effects, animation procedures, lenses and filters, high-speed photography, etc. 1970. 480 pp. **$24.50**

40. Radio Transmitters. *Gray and Graham.* Provides, in a logical, easy-to-understand manner, a working knowledge of radio transmitters for quick solution of problems in operation and maintenance. 1961. 462 pp. **$16.00**

23. Wide Screen Cinema & Stereophonic Sound. *M.Z. Wystozky.* First published in USSR in 1965 this excellent English translation covers wide gauge films, panoramic films, circular panoramic cinematography; technical fundamentals of stero sound recording for film, as well as details of the Soviet systems now in use. 284 pages. **$15.00**

33. Noise Reduction. *Beranek.* Designed for the engineer with no special training in acoustics, this practical text on noise control treats the nature of sound and its measurement, fundamentals of noise control, criteria, and case histories. Covers advanced topics in the field. 1960. 752 pp. **$19.50**

27. Noise & Vibration Control. *Edit. by Leo L. Beranek.* Practical design and regulatory information; formulas, choice of materials and structures, city codes and hearing protection; indispensable for design engineers, public officials who prepare regulations for noise control, safety and environmental engineers involved in noise and vibration controls. Covers data analysis, transmission of sound, psychophysiological design criteria, hearing damage risk, etc: Wealth of detail, comprehensive index and concise appendices. 650 pages. **$29.50**

28. Environmental Acoustics. *Leslie L. Doelle.* Applied acoustics for those in environmental noise control who lack specialized acoustical training. Basic information in comprehensible and practical form for solving straightforward problems. Explains fundamental concepts; pure theory minimized. Practical applications stressed, acoustical properties of materials and construction listed, actual installations with photos and drawings. Appendixes illustrate details of 53 wall types and 32 floor plans and other useful data. 246 pgs. **$18.50**

21. Acoustics—Room Design and Noise Control. *Michael Rettinger. 1968.* The enormous problems and hazards presented by noise are dealt within an orderly and practical manner. With many charts, graphs, and practical examples, the text covers the physics of sound, room acoustics, and design, noise and noise reduction. 392 pages. **$17.50**

22. Acoustics of Studios and Auditoria. *V.S. Mankovsky.* Basic theory plus a mass of design data covers the field with special reference to studios and places of public performance. For acoustical designers and specialists in sound transmission in cinema and broadcasting. Features exhaustive treatment of studio acoustics by the statistical, geometric and wave methods in parallel. 416 pgs. **$15.00**

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PEOPLE, PLACES, HAPPENINGS

● Word from **Rupert Neve Inc.** announces the opening of a new office in Hollywood, California, and the appointment of **John Marston** as Los Angeles sales manager. The office is located at 1800 North Highland Avenue, Suite 616.

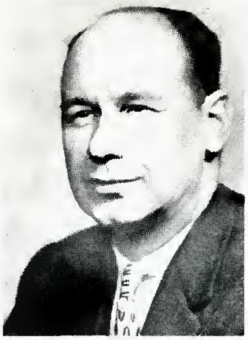
In another announcement from Neve we learn that **Arthur A. Schubert Jr.** has joined the U.S. division of the Cambridge, England-based company as chief engineer. He will be responsible for the technical management of the corporation. Mr. Schubert comes to Neve from **CBS** where he was a senior project engineer.

● Two appointments at **Westrex: Ken Tanzawa** has been appointed vice-president, general-manager of the Westrex division of **Litton Industries.** With his headquarters in Tokyo, he will be responsible for all Westrex activities in the far east.

Peter Buck has been appointed executive vice-president. He will direct all Westrex activities octside the United States. Since joining Westrex in 1946 as a recording engineer, he has served as general manager and director for several regional offices in the far east, Australia, and Europe. He is presently headquartered in London.

Clark

● **Robert J. Brown,** general sales manager from the **Mincom Division of 3M** has announced the appointment of **Paul B. Clark Jr.** as western area manager. Mr. Clark has served as a sales engineer and supervisor for the division's instrumentation, professional audio, electron beam, and video products. He succeeds **Bob R. Boatman,** who has been named area manager of a newly-established southeastern sales region with headquarters in Dallas, Texas. Mr. Clark will headquarter in Camarillo.



Kramer

● **Karl Kramer** has recently joined **Midwest Audio Corporation** of Chicago as manager, manufacturing. His responsibilities at Midwest will include engineering, procurement, and all phases of manufacturing. Midwest is a designer and producer of specialized sound systems for the transportation industry. Mr. Kramer comes to the company from **Jensen Manufacturing** where he held top managerial and technical positions for a number of years.

● **Donald F. Smith** has been named director of sales for the recording automation group of **Dictaphone Corporation,** comprising the **Scully** and **Metrotech** divisions. He replaces **Edward Ittner** who has left Dictaphone to join another company. Mr. Smith comes to Dictaphone from **Ampex** where he held several executive sales positions.

● The **Fourteenth Acoustical Training School,** conducted by **Michael J. Kodaras** and **Robert Lindahl,** acoustical consultants, will be held at the **Dearborn Inn,** Dearborn, Michigan on October 30, 31, and November 1, 1972. The three-day session will be devoted to acoustics, lighting, and air distribution of interior systems as well as general principles of architectural acoustics, sound transmission loss, acoustical correction of rooms, industrial noise, and noise control of air conditioning. Numerous workshop sessions will permit those attending to obtain first-hand experience in solving basic problems. The number of registrants is limited; previous sessions have been oversubscribed. Information on the sessions can be had from Robert Lindahl, P.E., 2261 Winthrop Road, Trenton, Michigan 48183 or Michael J. Kodaras, 75-02 51st Avenue, Elmhurst, N.Y. 11373.

● Late word just in: Fire of accidental origin swept the studios of **Sound Exchange** at 265 West 54th Street in New York City. Damage was extensive. What the fire itself didn't do, smoke, and water trying to put it out did. It looks awful. But just as the legendary Phoenix, we look to Sound Exchange to rise from the ashes more beautiful and better equipped than before.

● Word from **Bill Putnam,** chairman and president of the **URC companies** is that **Don Sears,** founder and president of **Sound Recorders, Inc.** and **Seco Laboratories,** both in Omaha, Nebraska has been named vice-president and general manager of all studio operations of the **United Recording Corporation,** headquartered in Hollywood, California. In making the announcement, Mr. Putnam pointed out that this includes all subsidiary studio operations at **Western Recorders** in Hollywood, and **Coast Recorders** in San Francisco. Mr. Sears in turn has also made an announcement —**Ron Ubel,** former second in command at the Omaha facilities will now be a vice-president and general manager there.

Lambrecht

● **Heinz Lambrecht** has been made national sales manager of broadcast and commercial products for **Telex.** He will supervise dealer as well as OEM sales and will report directly to **Sidney T. Kitrell,** director of marketing for aircraft/broadcast and industrial products for the company. Mr. Lambrecht comes from twelve years experience in commercial sound and communications. He was formerly with **Webster Electric Company** in Racine, Wisconsin, as design engineer, field sales engineer, and finally, sales manager.

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