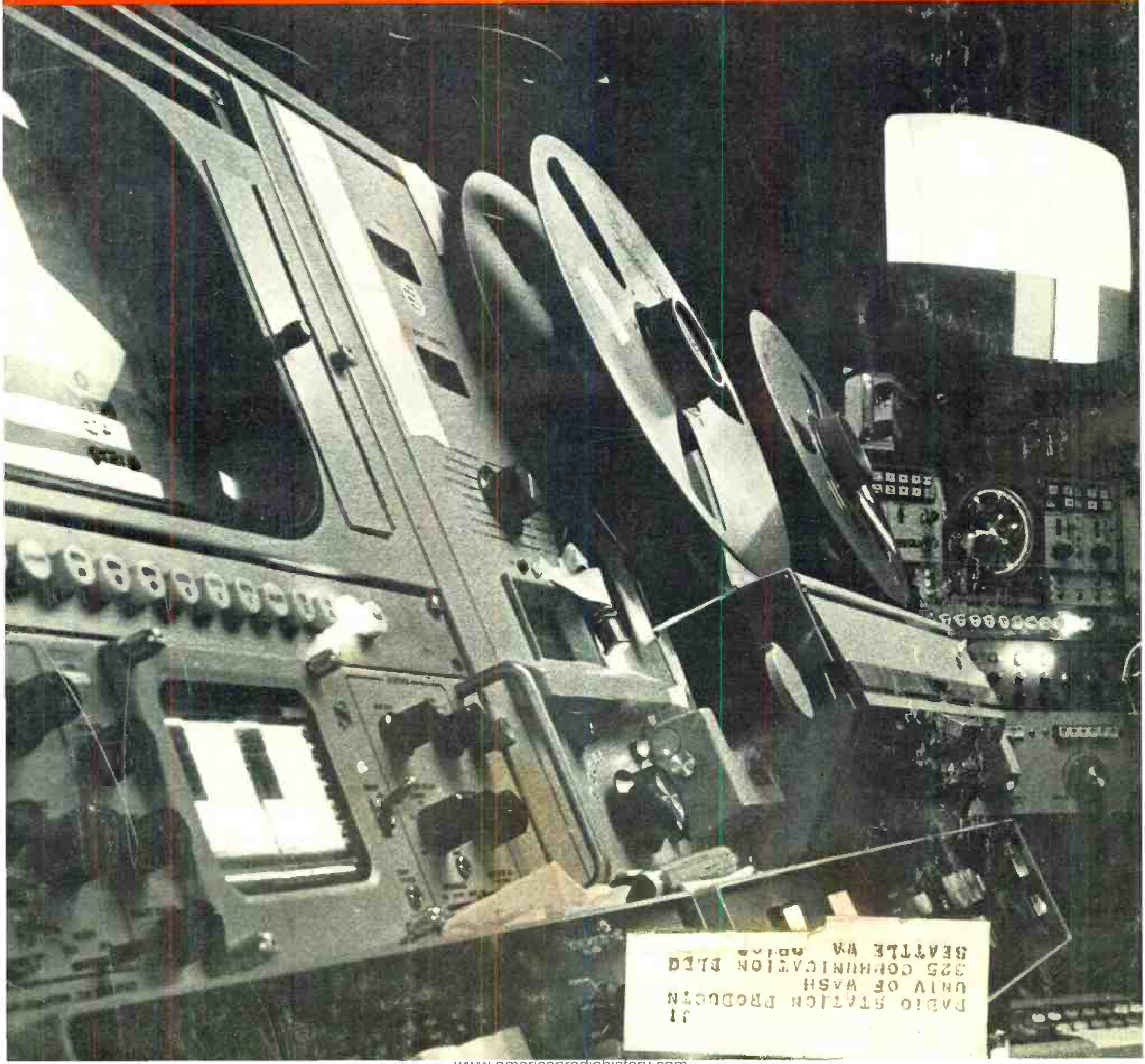
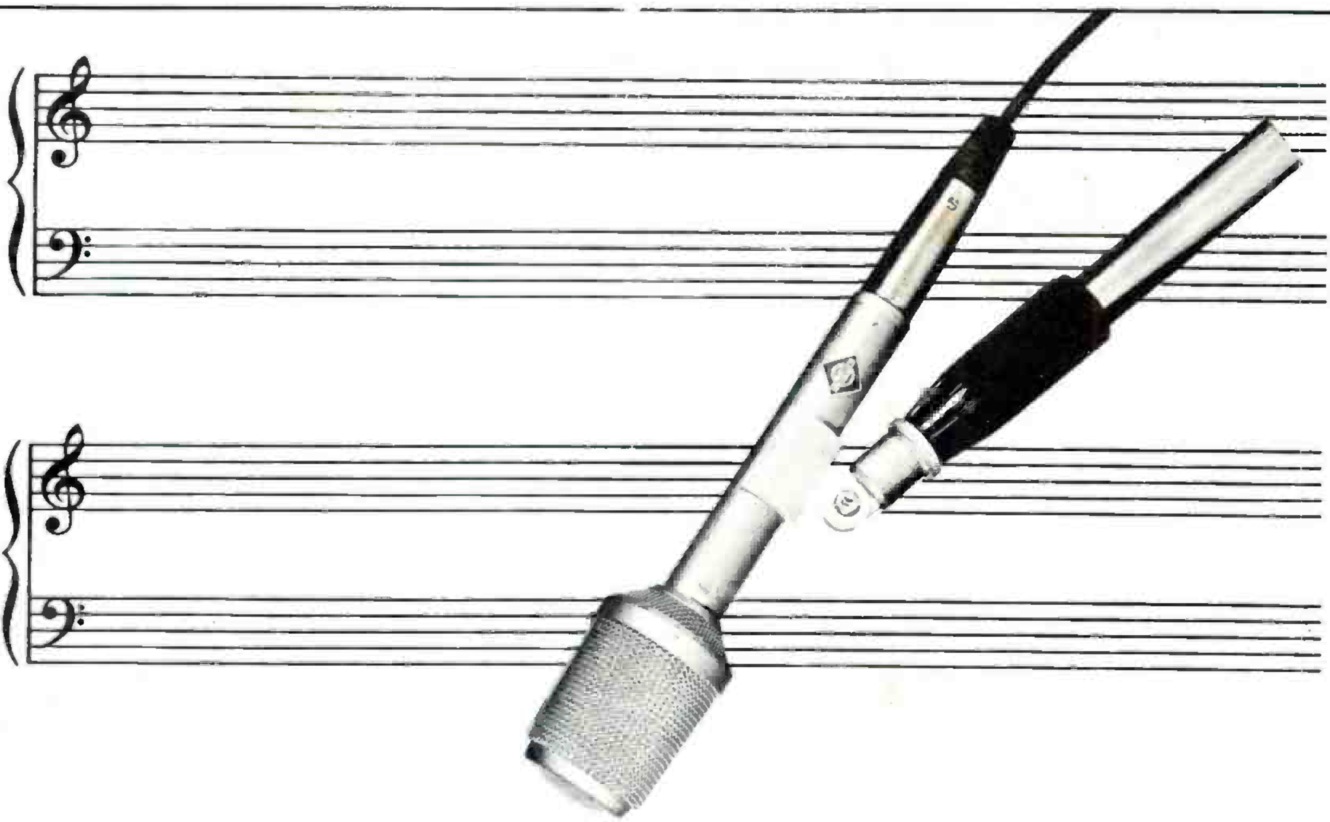
db

THE SOUND ENGINEERING MAGAZINE

SEPTEMBER 1971 $1.00





COMING NEXT MONTH

● Marshall King concludes his work on television and audio synchronizing when separate video and tape recorders are used. Everyone who is, was, or might be in t.v. audio will want to be sure to have read all three parts.

In LIVE ROCK: HOW IT IS, R. H. Coddington reviews the public-address equipment requirements of rock music groups, including the sound-pressure level demands. He then offers some important suggestions on a responsibility that sound men may not be shouldering.

The 41st Audio Engineering Society meeting takes place at the Hotel New Yorker, starting on October 5. So we will have a listing of the papers, the exhibitors and their spaces—and we will be there too.

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

ABOUT THE COVER

● A high band video tape recorder graces our cover as seen from an interesting angle. Could it be that this is the awesome view a t.v. sound man gets? Television audio man and author Marshall King took the photo.

db

THE SOUND ENGINEERING MAGAZINE

SEPTEMBER 1971 VOLUME 5, NUMBER 9

db is listed in **Current Contents: Engineering and Technology,**

db, the Sound Engineering Magazine is published monthly by Sagamore Publishing Company, Inc. Entire contents copyright © 1971 by Sagamore Publishing Co., Inc., 980 Old Country Road, Plainview, L.I., N.Y. 11803. Telephone (516) 433 6530. db is published for those individuals and firms in professional audio-recording, broadcast, audio-visual, sound reinforcement, consultants, video recording, film sound, etc. Application should be made on the subscription form in the rear of each issue. Subscriptions are $6.00 per year ($7.00 per year outside U. S. Possessions, Canada, and Mexico) in U. S. funds. Single copies are $1.00 each. Controlled Circulation postage paid at Harrisburg, Pa. 17105. Editorial, Publishing, and Sales Offices: 980 Old Country Road, Plainview, New York 11803. Postmaster: Form 3579 should be sent to above address.

letters

The Editor:
Some years ago, when I was contemplating the, to me, momentous decision to tool up for cutting stereo masters, Fairchild sent me on memo a specimen of the 641 stereo cutting system. Although I never got to the point of putting it into service, I think I can supply some of the reasons why it was not effectively competitive with other systems in use.

I doubt seriously that the Fairchild's kiss of death was its "compatibilty". I am more inclined to think that other factors were more important. For one, the cutter was very large and heavy, and its shape was such that it could not readily be mounted on many lathes, including my own. Being so massive, the cutter required an advance ball, which, aside from scoring the lacquers, did not allow sophisticated depth control systems. Functionally speaking, I do not find reasonable the built-in vertical attenuation as a presumed asset. Recording engineers have been always too willing to tailor response curves

advertisers index

db

THE SOUND ENGINEERING MAGAZINE

SALES OFFICES

New York
Specifax Media Sales, Inc.
Manfred W. Meisels
603 East 82nd Street
New York, N. Y. 10028
212-249-7768

Boston
Specifax Media Sales, Inc.
William H. Fields III
49 Gray Street
Boston, Mass. 02116
617-426-5315

Denver
Roy McDonald Associates, Inc.
846 Lincoln Street
Denver, Colorado 80203
303-825-3325

Houston
Roy McDonald Associates, Inc.
3130 Southwest Freeway
Houston, Texas 77006
713-529-6711

Dallas
Roy McDonald Associates, Inc.
Semmons Tower West
Suite 714
Dallas, Texas 75207
214-637-2444

San Francisco
Roy McDonald Associates, Inc.
625 Market Street
San Francisco, California 94105
415-397-5377

Los Angeles
Roy McDonald Associates, Inc.
1313 West 8th Street
Los Angeles, California 90018
213-483-1304

Portland
Roy McDonald Associates, Inc.
2305 S. W. 58th Avenue
Portland, Oregon 97221
503-292-8521

Pennsylvania
Specifax Media Sales, Inc.
Andrew N. McCreight
Box 199
Chester Springs, Pa. 19425
215-363-7355

IF YOU DO ANYTHING WITH 1/4" TAPE, YOU CAN DO IT BETTER WITH REVOX.

Automated broadcast operations
Scientific analysis
On location mastering
Tone or time changing
Audio tape quality control
Electronic music synthesis
Noise analysis
Film synchronization
Radio telescopy
Language laboratory
Machine tool control
Phonetic analysis
Radio telemetry
Industrial research
Information retrieval
Electrocardiography
Making calibration tapes
Tape mastering with SELFSYNC
Data storage from digital computers

And that's a simple statement of fact.

From the moment it was introduced, the Revox A77 was hailed as a recording instrument of unique quality and outstanding performance. The magazines were unanimous in their praise. Stereo Review summed it all up by saying, "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."

So much for critical opinion.

Of equal significance, is the fact that the Revox A77 rapidly found its way into many professional recording studios.

But what really fascinates us, is that the A77 has been singled out to perform some unusual and highly prestigious jobs in government and industry. The kinds of jobs that require a high order of accuracy and extreme reliability.

Take NATO (the North Atlantic Treaty Organization) for example. When they wanted a machine to standardize on, a machine that would lend itself to use in a wide variety of circumstances. And most importantly, a machine that was simple to use, the logical choice was the Revox A77.

Or take the governmental agency that wanted an unfailingly reliable tape machine to register and record satellite bleeps. The choice? Revox.

Or the medical centers that use specially adapted A77's for electrocardiographic recording.

We could go on and on (see accompanying list), but by now you probably get the point.

No other ¼" tape machine combines the multi-functioned practicability, unfailing reliability, and outstanding performance of a Revox.

If you have a special recording problem that involves the use of ¼" tape, write to us. We'll be happy to help you with it.

And if all you want is the best and most versatile recorder for home use, we'll be glad to tell you more about that too.

REVOX DELIVERS WHAT ALL THE REST ONLY PROMISE.

Revox Corporation
155 Michael Drive, Syosset, N. Y. 11791
3637 Cahuenga Blvd., West, Hollywood, Calif. 90068
In Canada: Tri-Tel Associates, Ltd., Toronto, Canada
Lamb House, Church Street, Chiswick, London W4 2PB

Circle 15 on Reader Service Card

for their own convenience, and then proudly describe a limitation as an improvement. There were also rumors of instability in the r.f. feedback of the 641.

Although, in theory, the matrixing of a vertical-lateral system can be justified, in practice it does not always work out. To be successful, such a system must be absolutely symmetrical in frequency response, and must have zero phase-shift between the two sections; otherwise there are problems with bleeding between channels, especially at high frequencies. I doubt if the 641 fulfilled these requirements. As for the "advantages" of vertical attenuation at low frequencies, any such practice is usually carried on for the convenience of the disc-cutting engineer, to avoid momentary groove discontinuities which not only make embarrassing noises, but sometimes do not track. I know of no evidence that stereo reproducers have any difficulty tracking low-frequency vertical excursions, nor has any excess wear been demonstrated.

Certainly any statement that the 641 is "superior to all its competitors" leaves its author far out on a fragile limb. One of its competitors, which I employ in my own work, is the Ortofon system. The cutter itself has about the volume of a pack of cigarettes, has drive and feedback coils on a rigid almuinum T-bar in the configuration described by Blumlein, is flat plus or minus ½ dB throughout the entire audible range, and has no notable deficiencies—although a little more attention could have been devoted to its protection (*e.g.*, failure of one tube in the feedback loop can cause destruction of the armature, requiring expensive and rarely speedy repair in Denmark). This cutter is an example of European finesse. (No, I do not sell them.)

On re-reading Mr. Schulze's article, I find at least one theoretical point on which to quibble. Mr. Schulze shares with many people, especially Europeans making "compatible" tapes, that the *directional* effects represent the sum total of the stereo experience. My own senses tell me that there is another component of stereo, which I might describe as *spatial*, which seems to depend on the phase relationship between the channels. There is no evidence that this last component is not to some extent affected by attenuation of vertical low frequencies. And, if these frequencies become troublesome for the cutting engineer, I personally prefer a dB or two of vertical limiting, which would occur only during those rare moments when needed, rather than a fixed attenuation present at all times. In any case, a properly-made tape, with careful attention to phasing and economical employment of microphones, has in-phase information increasing inversely with frequency, so that at extreme low frequencies the information is almost entirely lateral.

David B. Hancock
Recording Engineer
New York, N. Y.

The Editor:

This letter is prompted by the nonsense written by Richard Schultze (The Fairchild 641 revisited, July 1971) about the effects of bass rolloff in the difference or vertical signal of a stereo disc. This is one place where intuition completely fails most engineers in the industry, so it is very useful to examine this phenomonon from a rigorous mathematical viewpoint. Indeed, the results that pop out lead to certain exciting new possibilities for creating systems, as we shall see below.

Mr. Schultze's downfall came in his failure to recognize the difference between the sum of the *powers* in the left and right stereo channels and the sum of the *voltages* in these channels. It is the sum of the powers that we perceive when we are listening to a stereo recording in stereo; it is the sum of the voltages that we hear when we add the channels electrically and listen in mono. Mr. Schultze's elaborate demonstration with vectors served only to show that if we roll off the bass in the electrical difference (L-R), that the electrical sum (L+R) is unaffected. To me, knowing that these signals are orthogonal on the stereo disc, this fact seems obvious. However, power is proportional to the *square* of the voltage. In Mr. Schulze's

(continued on page 29)

Use this coupon to order your Miida MC/800 Personal Electronic

Calculator

...the engineers' electronic slide rule

TIMEKEEPER
P.O. Box 835, Great Neck, N.Y. 11021
Please send check or money order.
N.Y. residents add 6% sales tax.

Name______________________________

Address____________________________

City____________ State________ Zip________

Beautifully styled, the MC/800 is ideal for office, shop, school, laboratory or home use. It adds, subtracts, multiplies and divides at a calculating speed of 0.3m sec. Compact and silent in operation, the MC/800 is a reliable, high-quality instrument that sells for considerably less than calculators of inferior quality. We have checked this product carefully and can recommend it without hesitation.

8-digit entries and answers for all operations / floating decimal point / chain and mixed calculations / overflow indicator / 110 VAC, 50-60 HZ (conversion to 220 V can be made) / 2½" h x 6¾" w x 9" d / weight 3.3 lbs / One year warranty—factory service available in New York and Los Angeles.

only
$199.50
(includes shipping)

Circle 18 on Reader Service Card

total tape duplicating with "BUILDING BLOCK" simplicity

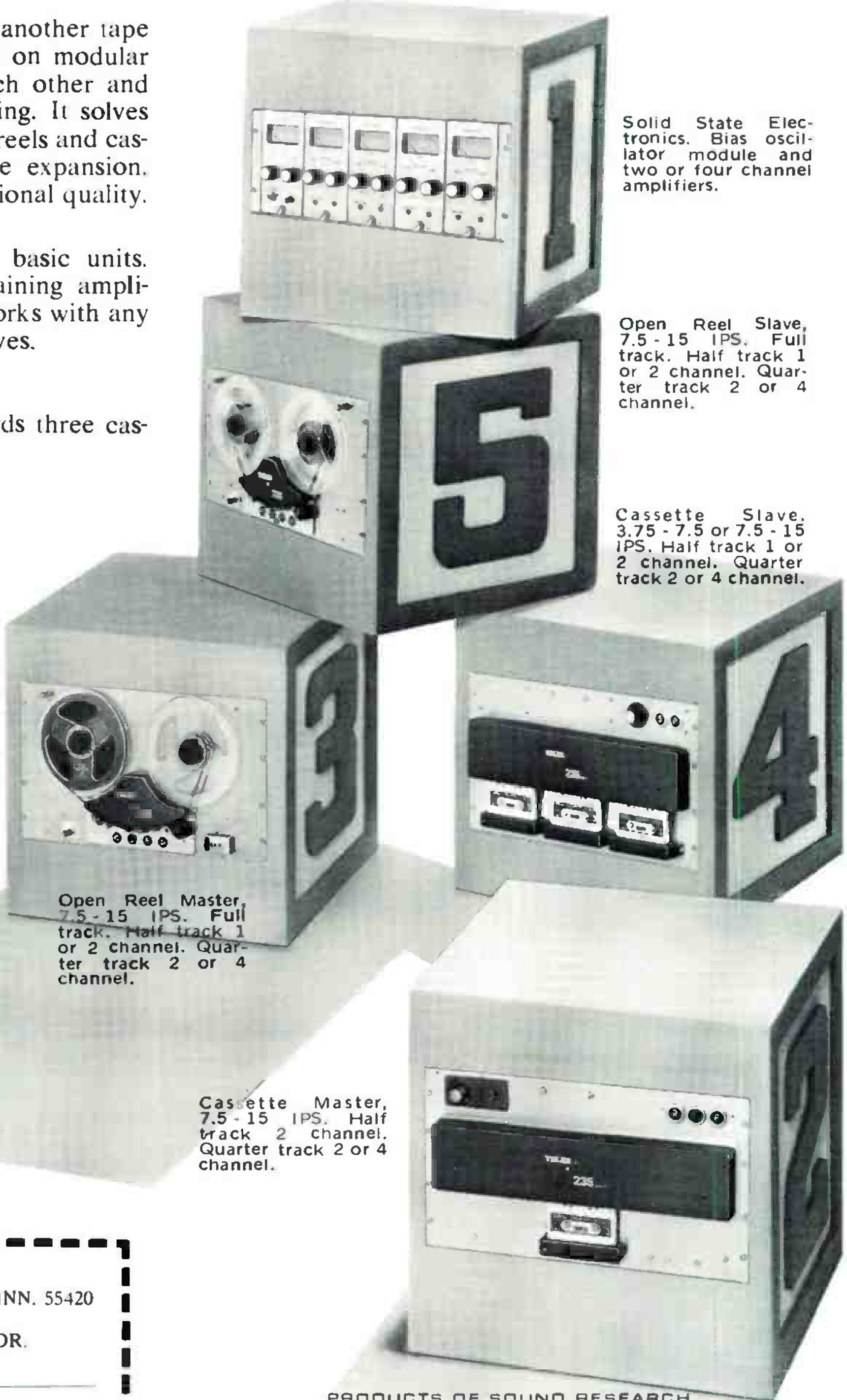
CASSETTE TO CASSETTE, REEL TO CASSETTE, REEL TO REEL

The Telex series 235-1 is more than just another tape duplicating system. It is a concept based on modular "building blocks" which complement each other and provide total flexibility for tape duplicating. It solves the problems of interfacing between open reels and cassettes. It is a system designed for future expansion. Engineered to make tapes of true, professional quality. And it's priced within your budget.

The Telex system consists of only five basic units.

1. Solid state modular electronics containing amplifiers, meters and controls. This unit works with any combination of ten cassette or reel slaves.
2. Cassette master play transport.
3. Open-reel master play transport.
4. Cassette slave record transport. Records three cassettes simultaneously.
5. Open-reel slave record transport.

The five units are totally compatible. Intermix cassette and open-reel master or slave transports to suit your duplicating requirements; cassette to cassette, reel to cassette, reel to reel, or even cassette to reel. All units fit into table top consoles of uniform size so when your requirements change, you just add more units. It's that simple. Telex series 235-1 is heavy duty equipment with hysteresis synchronous motor tape drives, momentary push button controls and time delay circuits for smooth, positive tape handling. Selected premium grade duplicator heads provide long life and excellent frequency response. And fail safe, automatic features enable non-technical personnel to operate the system efficiently. Telex "building blocks" make a totally flexible and complete duplicating system. It's the sensible approach, designed to meet your needs today, next month and in the years to come. Made in the U.S. to professional standards.



TELEX COMMUNICATIONS DIVISION
9600 ALDRICH AVE. SO. • MINNEAPOLIS, MINN. 55420
DBS
PLEASE SEND INFORMATION ON DUPLICATOR.

NAME ______________________

TITLE ______________________

INSTITUTION ______________________

ADDRESS ______________________

CITY ____________ STATE ________ ZIP ________

PRODUCTS OF SOUND RESEARCH
TELEX®
COMMUNICATIONS DIVISION
9600 ALDRICH AVENUE SOUTH • MINNEAPOLIS, MINNESOTA 55420

CANADA: DOUBLE DIAMOND ELECTRONICS, LTD., 34 Progress Avenue, Scarborough 4, Ontario
EXPORT: ROYAL SOUND COMPANY, INC., 409 North Main Street, Freeport, N.Y. 11520

Circle 16 on Reader Service Card

Norman H. Crowhurst

THEORY AND PRACTICE

● A commonly recurring question in my mail concerns the design of crossovers for some or other choice of speakers, not always specified in any detail—not that I could always help much more if they were! Some specify that they would like 12 dB/octave type, or 24 dB/octave type, and some say where they would like the crossover frequency.

But from many of them, one way or other, I get a clue that they have made such decision without any satisfactory basis. True, I have written more articles than I can count about choosing the best slope and circuit configuration. And there are plenty of articles about winding inductors (in the constructor-type magazines), but somehow these readers are left without the whole story.

A recent letter questions whether there is any way that an 8-ohm woofer can be combined with 16-ohm midrange and tweeter units, for use with a transistor amplifier. From that, I would guess that the output is transformerless, although it may be provided with 4-, 8- and 16-ohm output terminals. Or it may not even have that provision, just a pair of terminals for all impedances.

If more than the two terminals are provided, to differentiate for various load impedances, the probability is that resistance padding is used. For example, connecting a 4-ohm speaker might seriously overload the output transistors, unless a protective resistor is included to limit output current. But there is no transformer to match different impedances in the classical sense, which would allow different impedance units to be combined with different crossover sections (FIGURE 1).

In that instance, which was possible with most tube amplifiers, because they did have multiple-tap output transformers, part of the crossover could be designed for 8 ohms and part for 16 ohms, and the transformer would put the whole together so correct matching prevailed through the frequency range.

Without such a transformer internal to the amplifier, this method will not work, but an external transformer can be used instead. However, the better way is probably to put the transformer on the output side of the crossover network, and design the circuit for one impedance (FIGURE 2). This obviates the need for a single, wide-range transformer, which is somewhat expensive, but came as an integral part of tube amplifiers.

With only one output transformed to the impedance used for all, the transformer has to handle only the frequencies used in that loudspeaker unit. Which is the better unit to transform? This is a good question, to which the answer may not be universal. Probably a good question to help make the decision is, which impedance really suits the amplifier best?

Transistor amplifiers have outputs designed to span a number of impedances, and it may be that an amplifier truly serves each load value equally well (or at least two of them). It may also be that one load better matches, so that higher output, or cleaner output, is achieved with that value. This can only be checked for the individual amplifier in question. Having decided which of the two impedances you are going to use suits the amplifier best, you can then decide to transform the other one.

The only problem so far is that of finding out which impedance suits best, which may not be too easy, with modern, high-powered amplifiers. It may not be very critical, in that case, either.

While a transformer that auto-transforms 8 ohms up to 16 can also transform 16 ohms down to 8, merely by reversing connections, the frequency range for which the transformer is better suited may have an influence. If the transformer is too small, it will not likely handle the lower frequencies needed by the woofer without distortion. If it is a big one, designed to match woofers nicely, then it may not handle the highs too well, and thus should be used for the woofer.

Another factor to the problem may be that the right transformer for the job, either way, just does not seem to be around, and you do not have the facilities for getting one specially wound. Perhaps you want to convert from 8 ohms to 16, or *vice versa*, and the best transformer you can find will only provide a 4 to 8 conversion. What then?

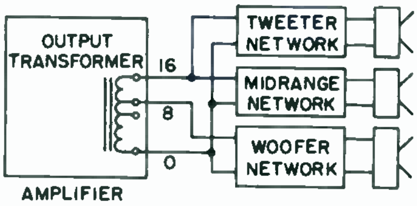
If the impedance *ratio* is the same (in this case, each is 2:1), then the transformer may be pressed into service, with some careful observations. If a transformer designed for 4 to 8 ohms is used for 8 to 16, the lowest frequencies it is intended to handle will be distorted. Suppose it is designed to handle down to 50 Hz without distortion, at 4 to 8 ohms. On 8 to 16 ohms, it may distort at 100 Hz.

Under this circumstance, it would be best to design the crossover for 8 ohms, then transform the output up to 16 ohms. The crossover to divide between the mid-range and tweeter should be designed for 16 ohms, and operated on the output side of the transformer (FIGURE 3).

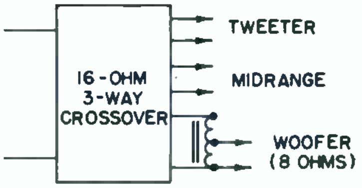
Now comes the problem of deciding on crossover frequency. One way is to accept the rated frequency range of the speaker units involved, where these are designed as parts of a multi-way systm. If a unit, either the mid-range or tweeter, is of the horn type, the crossover frequency should not be lower than the horn cut-off frequency. Otherwise, limits of frequency range are not usually critical.

One thing that can be useful is to use the low-frequency (woofer) unit's own voice coil inductance as part of the filter. Assume that 12 dB/octave filtering is desired, then the only way to achieve this is to employ series connected filters for this crossover. The one between mid-range and tweeter can be parallel-connected (FIGURE 4).

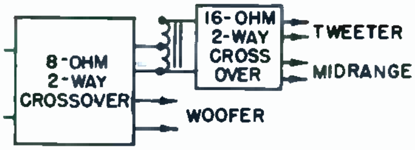
To find the frequency for the low



1. How different speaker impedances can be fed through appropriate crossover filters, when the amplifier has an output transformer.



2. A way to achieve matching when the amplifier does not have an output transformer.



3. If the transformer is inadequate for handling woofer frequencies, this alternative to Figure 2 may be necessary.

to midrange crossover, connect the unit to the output of an amplifier fed from an audio oscillator (FIGURE 5) with a small (about 1-ohm) resistance in series, and connect this to a scope so a lissajou loop is displayed (FIGURE 6).

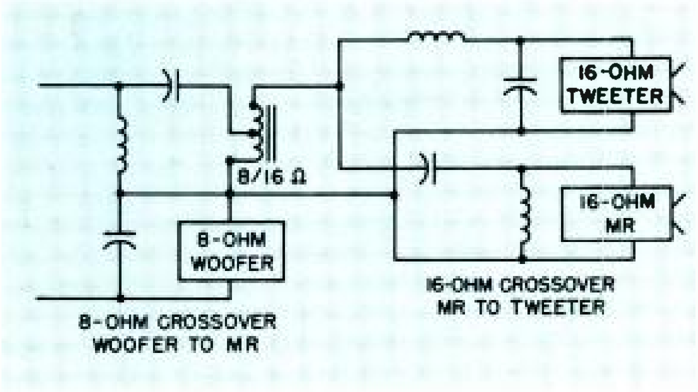
As the unit is a woofer, it will have a low-frequency resonance, at which point its impedance will be higher than rated value, represented by a steeper line. As frequency is raised, this line will open out into a loop and close to a line again at a less steep slope, which will be about its true impedance.

A well-designed unit will then run through two or three closures to a line, with slight loops in between, before the trace begins to open out again into a steeper loop. The point you want is where the inductive component is 0.707 of the resistive component. This results in an impedance of 1.225 times nominal (where the lines closed). As the height of the line is almost fixed (1 ohm being much smaller than the voice-coil impedance) by amplifier output voltage, the length of the trace horizontally is what changes, due to changing current in the 1-ohm resistor.

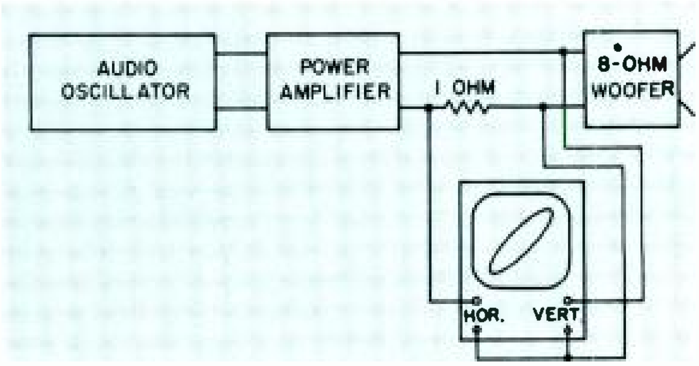
The trace that will identify the presence of a reactive component 0.707 of the resistive component will be very slightly over 0.8 of the width, and the points where the loop crosses the vertical will be 0.57 of the height (also shown on FIGURE 6). When you have found the frequency that most closely approximates these conditions (they may not exactly coincide) this is a good frequency for the woofer to midrange crossover.

Assume this comes at 900 Hz. The voice-coil impedance is 8 ohms. You need two capacitors whose reactance is 0.707 of 8 ohms at 900 Hz. 0.707 of 8 ohms is 5.65 ohms. At 900 Hz, omega is 6.28 x 900 = 5,650. The capacitance figures close to 32 mFd. The inductance of 5.65 ohms at 900 Hz will be 1 millihenry.

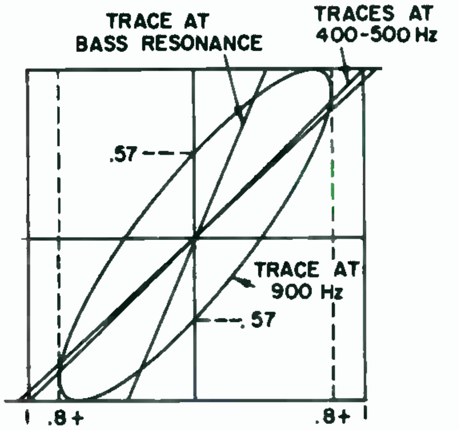
We will pursue the design of this crossover system in the next issue. ■



4. A circuit that allows the voice-coil inductance of the woofer to be used as an element of the crossover network.



5. Set-up for checking the impedance characteristic of the woofer, to find a suitable crossover frequency applicable to the circuit of Figure 4.



6. Traces relevant to the test circuit of Figure 5.

Circle 21 on Reader Service Card

John M. Woram

THE SYNC TRACK

● About a year ago, this column rambled on about four-channel sound. Written just after the Consumer Electronics Show, there was, at that time, much speculation about the future of the new gimmick. Now, so much later, the future of four channel is still being debated. Presuming it does not get talked to death, (which appears unlikely at times), we seem to be heading toward an interval during which the encoder-decoder system will prevail. The interval will end when, and *if*, someone invents a method of recording and retrieving four discrete channels from a disc as efficiently as we now do two.

In the meantime, several manufacturers are marketing encoder-decoder systems. At the studio, a discrete four-channel program is encoded onto a two-channel tape, from which a conventional stereo type record may be produced. When played back at home through a decoder—*voila!*—four channel sound. Almost. True, there is signal coming out of four speakers, but it is not completely predictable.

Before my phone starts ringing, I should explain my definition of predictability. If we listen to a standard two-channel tape, we can predict what the eventual pressing will sound like. Providing all the ground rules have been observed, it will sound pretty much like the tape. Of course, there are those ground rules. For example, the tape should not exceed the state-of-the-art limitations for disc cutting. (Bass response, dynamic range, over-all level, transient response, and what not.) By now, we've come to know our limitations, and we never exceed them. Well, hardly ever. Hence, the predictability of the disc.

But now, on to four channel. Continuing our tape-to-disc train of thought, our decoded four-channel program will sound just like the original four-channel discrete studio tape, provided this tape was made within the limitations of the encoder-decoder system.

At the moment, this is no simple matter, since these systems use matrices, phase inverters, phase shifters and so on. All of which may be elegantly defined on paper. Who would argue with, as one manufacturer puts it, "$L'_F = L_{COS}\,\theta + R_{SIN}\,\theta = L_F + 2R_F\mathrm{SIN}\,\theta\cos\theta + L_R\cos 2\theta$." It looks very impressive. I wonder what it sounds like? Whatever it sounds like, it's a long way from the day when each (decoded) speaker will contain independently variable, and predictable, information. In the meantime, another equipment manufacturer tells us that his system is often the equal of, and sometimes better than, discrete four-channel sound. Which means that the random phase shifts on a well produced two-channel recording may very well be more pleasing than an ill-conceived discrete four-channel spectacular. However, a well-done four-channel recording should certainly come off even better.

An early rear-channel system was simply an additional speaker wired across the positive terminals of the two front speakers, thus producing a difference signal at the rear. If the front speakers were reproducing a mono signal, there would be no difference signal. However, if the program was 180 degrees out of phase at one front speaker, there would be a maximum difference signal produced at the rear. For stereo, the rear speaker provides a continuous output, yet always completely dependent on the front program. And since each half of a conventional stereo program is a very complex signal, the only practical way to determine what will come out at the rear is to turn the system on and listen. Knowing that Rear = R = / L−R / is not too much help.

Of course, the latest four-channel systems are more sophisticated, yet the most effective way of making a four-channel master is to simultaneously encode and decode, listening to the decoded program as the encoded tape is being made. In this way it will be immediately apparent when the limitations of the encoder-decoder system have been exceded.

There still seems to be a definite limit to the amount of encoded information that can be stored in a two-track format. So, one system favors left-to-right-separation, at the expense of front-to-back. In another system, the opposite appears to be true. Similarly, some systems will permit more diagonal panning, while others are more effective with adjacent speaker panning. Placing a soloist on all four tracks often tends to be almost impossible. And rear-center soloists have a way of disappearing when played back in the conventional stereo mode.

Which brings up another point—compatibility, or the lowest common denominator. How should a four-channel product sound when played back on less than four speakers?

At the risk of offending a lot of people, I am about as interested in what four-channel tape will sound like on a mono playback as I am in limiting my impressions of Michelangelo's Pieta to what I can gather from looking at a photograph of it. If Michelangelo had painted his statue in contrasting colors, no doubt we should have more interesting photographs available. Mercifully, this didn't occur to him. So, today we have a masterpiece. You can study a photograph of it, buy a small reproduction, or go to Rome and see the real thing. It's up to you. Chances are, if you went to Rome and found the statue painted up for better photo reproduction, you'd feel cheated.

Well, the analogy may be a little strained, but why get overly concerned about mono compatibility? In a good four-channel program, rear-speaker information is often created simply because additional speakers are available. The same information when heard up front may very well get in the way. In the ideal four-channel system, subtle balances can be established, both left-to-right and front-to-back. Combined into mono, the balances are destroyed. In order to hear everything in mono, many balances must be over done, thus marring their effect in four channel. To a lesser degree, the same holds true for stereo.

In short, you can't make an ideal four-channel tape that will also be an ideal stereo and mono tape. Why try? Yes, I know about the F.C.C. and the kiddies who support us all by buying 45 singles after listening to the mono radio. In the case of the former, perhaps record companies might consider dual releases—one in stereo, and (more-or-less) mono compatible; the other a no-compromise four-channel product. We all lived through an era of mono and stereo releases, and if the consumer is to be expected to rush out and buy more hardware, he should at least be entitled to the best software available.

As for the singles market, I rather suspect that four channel will find its greatest popularity in album sales anyway. Singles might continue to be made as they are now, with the four-channel version available only on the lp album. This might even boost album sales in the long run.

Hopefully, it will not be too much longer before encoder-decoder parameters become more-or-less stabilized. Even in the present state of confusion, programs encoded on one system may often sound satisfactory when played back on another system's decoder.

At the Vanguard Records studio, we are setting up a special facility that will contain most of the current encoders and decoders. With a little experimentation, we hope to be able to improve the quality of the encoded product. And as we become more familiar with the characteristics of the various units, we may be able to apply what we learn to our present recording and mixing techniques for even further improvement. Good four-channel sound means more than just assigning the various tracks of a multi-track tape to four speakers instead of two. Phase and time delays, to mention two obvious variables, take on a new significance. As we learn more about what we can, and cannot, expect from four-channel sound, we can look forward to even further improvements in the state of our art. ■

ARNOLD SCHWARTZ

THE FEEDBACK LOOP

● The word linearity excites good mental images and non-linearity is something we try to avoid. These two concepts seem to connote some element of good and evil. As we usually avoid thinking and studying evil, non-linearity is often shunned by the engineering mind. Yet there is good reason to study non-linearities, if only for the reason that understanding non-linear mechanisms can help our understanding of linear systems.

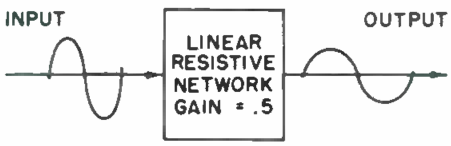
LINEAR SYSTEMS

What is linearity? A system is linear when the output is some constant proportionality of the input. A resistive attenuation network is one of the simplest forms of linear systems. If this network has a 6 dB loss, then the output would be half the input signal. That is the proportionality constant is 0.5, and we can multiply any input signal by this factor and predict the output signal. If we consider each point on the sine wave of FIGURE 1 and multiply it by 0.5, we wind up with the obvious, a signal with the identical waveform but with exactly half the amplitude of the input. A linear device can have gain, and in this case the proportionality is greater than 1. For example, an ideal amplifier can have a gain of 6 dB, so that each point on the waveform is multiplied by 2, and the resulting output is identical in form but twice the amplitude of the input.

Linear systems can be described graphically. If we plot the output signal as a function of the input signal for three linear systems, unity gain, 6 dB loss, and 6 dB gain we find that we have a straight line in each case (see FIGURE 2).

However, the slope of each line is different. The curves plotted in FIGURE 2 tell us at a glance what kind of device we are considering. The fact that it is a straight line indicates its linearity; the slope of the line indicates the gain (or loss).

Figure 1. A linear device



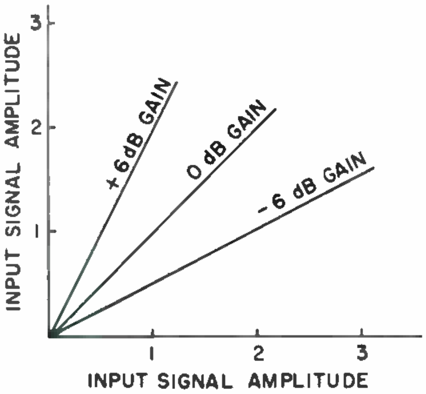


Figure 2. Linear system input-output function.

NON-LINEAR SYSTEMS

What is non-linearity? A system is non linear when the output is *not* a constant proportionality of the input. We can describe such a system in the following way. Let us say that the gain of a system at 1.0 volt input is 2, at 2.0 volts input the gain is 1.5, at 3.0 volt input the gain is 1.2, and so forth. If we plot the output as a function of the input signal (see FIGURE 3), we arrive at a curve that is not a straight line. If we were to feed a sine wave with a 1-volt peak amplitude into this system and multiply the amplitude by the appropriate gain, the output waveform would *not* be identical to the input waveform. Unfortunately for the audio engineer, the world abounds in devices that have to some degree, this characteristic. For example, when we approach the saturation level of a tape we find that as the signal is increased the gain of the system (i.e. the magnetization of the tape) does not increase proportionally. Non-linear systems, in contrast to linear systems, do not have the simple situation of a number of curves differing in slope only. Theoretically, the curves can have any shape and slope. All these curves describe systems where the output amplitude is a function of the input amplitude, and the output waveform is distorted.

Figure 3. Non-linear system input output function.

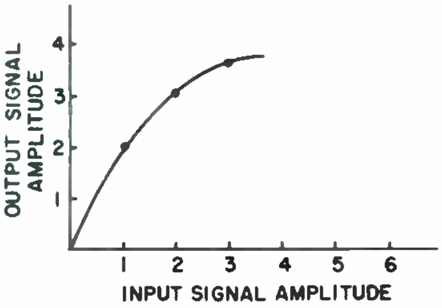


Figure 4. A linear system.

An alternate view of distorted waveforms is provided by a harmonic analysis. The output signal is displayed as a fundamental (the input waveform) and a series of harmonics; when the harmonics are combined with the fundamental the composite output waveform is produced. Each type of non-linear system will produce different sets of harmonics, and in different proportions.

GENERATION OF NON LINEARITY

The description of distorted waveforms by harmonic analysis has its counterpart in the description of the functioning of the non-linear device itself. The non-linear device under consideration can be viewed as a set of parallel black boxes each generating a different harmonic. In the case of the linear system this simplifies down to a single black box as shown in FIGURE 4. Here the linear system is merely a "k" describing the proportionality of input to output signal. A relatively simple non-linear system is shown as two parallel black boxes (FIGURE 5). Each black box is fed by the input signal, and the outputs are added to produce the composite waveform. One element of this system produces the linear component or the fundamental, and the other element called a square generator produces the second harmonic. This generator is

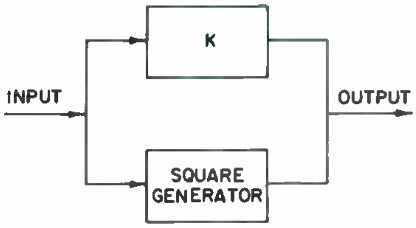


Figure 5. A non-linear system.

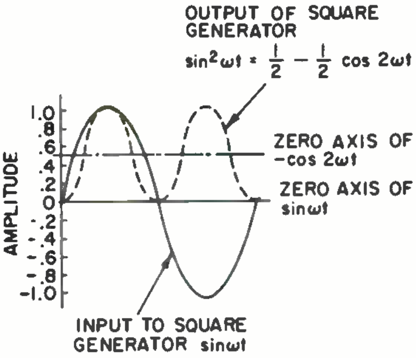


Figure 6. The graphic illustration of $\sin^2 \omega t$.

described as a square generator because that is precisely the function that produces second harmonic distortion.

How does it do this? If we go into the trigonometric functions we find that: $\sin^2 x = \frac{1}{2} - \frac{1}{2} \cos 2x$. If we let x be the function sin ωt; where ω is the angular frequency, and we multiply (sin ωt) (sin ωt) we come out with a d.-c. component (½) and a second harmonic shifted 90 degrees (-cos 2 ωt).

Generation of the second harmonic by squaring a sine wave can be illustrated graphically as it is in FIGURE 6. If we take each point on a sine wave and square the magnitude we will find that for all positive values we get a corresponding positive value. However, negative values when squared equal a positive number and hence we get a polarity reversal on the negative half of the cycle. It is interesting to note that the square generator is a rectifier circuit, and if we were to discard the second harmonic (ripple) we would have a d.-c. output.

By either the trigonometric function, or by plotting the square of each point on the waveform, we can see that second harmonic is generated by a "square function" generator. The relative amplitude of the second harmonic as compared to the fundamental is determined by the gain (used here in the linear sense) of the square generator. In the example of FIGURE 6 we assumed unity gain.

A higher order harmonic generation can be represented in similar fashion; third harmonic, for example, by a "cube function" generator. ■

Martin Dickstein

SOUND WITH IMAGES

Follow-Ups

● In further attempts to keep our readers up to date on the latest happenings in the visual fields, we should like to look back on some past developments of equipment and ideas and perhaps provide, for those that like to look ahead, a preview of what they might keep their eyes open for in the future.

Let's first go back to March of 1970. At that time, there was a total eclipse of the sun. We discussed the dangers of looking directly into the sun during this period in a column. Shortly after that time, an item in the news caught our attention. A young girl had lost her sight due to looking directly at the sun during the period of totality for a period of 4 minutes. There was no pain, so the girl did not realize the danger involved. As it turned out, there is a happy ending to this story. After more than a year, through what appears to be a miracle, or at least unexplainable so far to medical science, the girl seems to have recovered her sight sufficiently to see well enough to return to high school, and although her sight blurs occasionally, she can read without the use of glasses. Subsequent news reports showed that 133 other reported cases showed up in which people suffered eye damage for the same reason. Interestingly, of the more than 30 states in which these reports originated, California had the most cases, 22, and the eclipse was only 35 per cent total in that state.

Speaking of the eclipse, a civilian version of the video tube developed for the military permitted direct coverage of the solar spectacle for home viewing on t.v. The Tivicon, developed by Texas Instrument, uses a solid state light-sensitive silicon wafer in place of the chemical film found in most camera tubes. This silicon slice, measuring only ⅞ths of an inch in diameter, contains 750 rows of 1,000 light-sensitive diodes which are photochemically formed. The unit can not be burned out and is claimed to be as much as ten times as sensitive as normal black-and-white tubes.

Another development in video equipment came as a result of a failure in the Apollo 12 mission in November of 1969. Inadvertently, one of the moon-walking astronauts pointed the camera they were using directly at the sun and the vidicon burned out. No further reception from the moon was possible. For Apollo 15, RCA developed a "sun-proof" camera which could not be damaged even by direct facing toward the sun. This unit was demonstrated in June of last year and was, of course, very successful in transmitting color pictures from the moon including the take-off of the lander vehicle. This camera, based on the CBS Laboratories' color t.v. system of using a field-sequential color wheel, had nearly 400,000 tiny silicon diodes on the imaging surface. The t.v. system, of which the 10-pound camera was a part, was designated Ground Commanded Television Assembly (GCTA). It was remotely controlled, very remotely—from 250,000 miles away. The difficulty here, of course, that between the pushing of the control button and seeing the results there is a lapse of almost 4 seconds. This makes for a lot of overshoot unless carefully controlled. It was as was excellently demonstrated during the moon-walk. As it might not have been obvious to many people viewing the proceedings on the moon, communications with the astronauts was by radio, as was all the telemetry including such things as heartbeat, etc., but the t.v. camera was connected to the rover vehicle by cable after which, of course, the signal was mixed with all the telemetry and voice and transmitted to earth.

Among other developments in the video field as the result of moon-mission requirements was the ability to convert the slow scan of the astronauts' camera to the broadcast standards of the U.S. and Europe. Imaging tubes capable of this operation were used by NASA, and similar units have now been developed by General Electrodynamics. Called Omnicon, the tube can perform the scan conversion operation as well as store the input material for up to 15 minutes or more. Thus, this kind of tube can hold transient information for readout purposes, catch non-repetitive 'scope traces or hold motion for recovering on video tape or reading on a t.v. monitor.

When it comes to space television, there are projects now in the works which will really take the records for long distance transmission. For example, there are now out in space, two space ships heading for the vicinity of Mars. On both of these 287 million mile voyages, t.v. will play an important part in gathering of information for relaying to earth for evaluation. The objectives of the Mariners are to study the surface and atmosphere of Mars in detail. To accomplish these, one craft will map 70 per cent of the planet while the other will repeatedly study selected areas to observe changes on the surface and in the atmosphere. Recurring phenomena such as dust storms, clouds, and seasonal changes will be studied at close range. To do this, both vehicles will carry identical payloads including a television camera with a wide angle lens and one with a telephoto lens. The 70 per cent mapping will be done with medium resolution, while 5 per cent will be done with high resolution. Mariner H (Mission A), the first of the two vehicles, is scheduled to arrive on November 14, while the second, Mariner I (Mission B) should arrive ten days later. The total trip will have taken 190 days.

Although detection of life forms on Mars is beyond the resolution capabilities of the cameras, the wide angle unit will be able to receive features on the surface of about 3,280 feet in length, while the narrow angle camera will pick up features about 328 feet in length. The sequences of approach photos will begin at ranges in excess of one million miles. One interesting feature of the wide-angle camera is that it is equipped with eight filters, the choice of which are under the command control of earth. The cameras will also act as bore-sights for the infrared radiometer experiments. Further on, in the 1975-1976 Viking voyages to Mars, two more ships will be paired with cameras, only this time there will be a landing. Maybe at that time, the question of whether there is life on Mars may be answered. The trip at that time will take over 300 days and will be 460 million miles long.

Here's what 8 FM station engineers said about the Bang & Olufsen SP-12 cartridge:

WVCG/WYOR Coral Gables, Fla.
...this excellent cartridge is ideally suited for professional applications. SP-12 would be a good choice for the new quad-4 channel-stereo discs.

KBUC San Antonio, Texas
The cartridge is without a doubt the "Rolls-Royce" of the broadcasting industry!

KRBE Houston, Texas
Low's and hi's came through very impressively over entire audio range. The SP-12 is an excellent cartridge surpassing both the Shure V-15 and the Stanton 681EE in all respects in my tests.

WKJF-FM Pittsburgh, Pa.
Tracking, so far, has been excellent. SP-12 has been used "on air" 7 hours a day since received and not stuck or skipped yet.

WEMP Milwaukee, Wis.
We appreciate the wide-range response without the harsh "edge" that so many cartridges add to the sound.

KDIG La Jolla, Calif.
An excellent cartridge, none better on the market today.

KBAY San Jose, Calif.
Up 'til now the Shure V-15 type II has been our favorite for critical listening. After installing the B & O cartridge in the shell the Shure cartridge was in, we've left it there. It sounds great!
Exceptionally clean, undistorted, pure sound. One London Phase Four recording in particular has always broken up during a highly modulated passage, we assumed the record was over-modulated, until we played it using the B & O cartridge.

KMND Mesa, Ariz.
If there could be any comment at all, it would have to be that the cartridge seemed to display a very smooth and pleasing sound, a very flat and very clean, clear and brilliant response. The separation is very good and both channels are quite consistent on response.

Write for a report of FM Station Engineer Evaluation

SP-12 Cartridge
$69.95

Bang & Olufsen of America, Inc.

525 East Montrose Ave., Wood Dale, Illinois • In Canada: Musimart Ltd.

Circle 24 on Reader Service Card

NEW PRODUCTS AND SERVICES

PROGRAM EQUALIZER

● The frequency range of 200 Hz to 7500 Hz is covered by this model PEQ-4 program equalizer and filter. Six selectable low frequency points for each peak and dip position permit simultaneous boost or attenuate from 200 to 1000 Hz. An additional seven positions each for mid-frequency dip and boost perform the same function for the range from 1250 to 7500 Hz. Output is up to +24 dBm sine, and distortion is below 0.1 per cent. Signal to noise is 89 dB below +4.
Mfr: Lang Electronics
Circle 68 on Reader Service Card

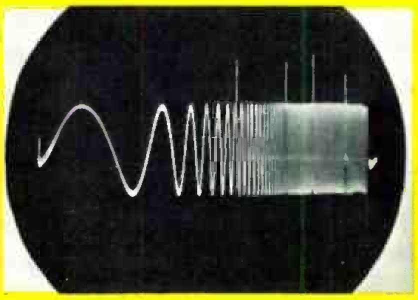
DUPLICATORS

● The new model AD-15 permits duplication and on-line tailoring of audio programs in any 150-mil or ¼-inch tape format and is modularly expandable, allowing use of from one to three slave units. A simple change of plug-in heads, guides and tailoring assemblies allows reproduction directly onto mono, stereo, and quarter-track open reels, two- and four-track cassettes and four- and eight-track cartridges. Proper settings for maximum performance are locked into each head stack so there is no need to make head adjustments when changing formats. The AD-15 master and slaves employ the widely used Ampex ABR-15 professional audio transport which accepts 150-mil and ¼-inch tape on pancakes and reels ranging from 5-inches to 15-inches in diameter. Reel and capstan systems on the AD-15 are servo controlled and referenced to a special bridging oscillator for consistent tape speed within ±0.08 per cent from beginning to end of reel. Solid-state electronics contribute to reliable, high quality performance. Signal-to-noise ratio on copies is within 3 dB of the master. Flutter and wow does not exceed 0.1 per cent at top duplicating speed of 30 in/sec. The AD-15 slaves may be temporarily converted for use as master production recorders when required. The system, capable of producing up to 168 1,200-foot copies in one 8-hour shift, is available with from one to three slaves and with any configuration of heads and tailoring devices. This enables users to match current needs, then gradually update for future requirements.
Mfr: Ampex Corp.
Price: From $5000
Circle 65 on Reader Service Card

B-TYPE NOISE REDUCTION

● The 360 Series are single-channel B-type processors which can be used for recording or playback. Model 361 with remote changeover facilities is suited to multitrack work. A new test extender will be introduced for easy performance verification and servicing of 360 Series noise-reduction modules. The Model 320 encodes Stereo f.m. broadcasts as well as masters for cassette tape duplication.
Mfr: Dolby Labs
Circle 52 on Reader Service Card

SWEEP FREQUENCY TEST TAPE



● The No. 113T 8 track cartridge test tape offers an instantaneous method of checking frequency response on all types of stereophonic equipment. This tape is designed with all necessary correction factors included in it, therefore, no charts or graphs are needed for instant response measurement. All that is needed is a cathode ray oscilloscope and a sweep frequency tape for instantaneous response measurements. Only a few quick adjustments on the equalizer circuits and the job is done.

The audio range of the No. 113T 8 track cartridge is from 60 Hz to 15 kHz with marker pulses at 1, 3, 5, 10, and 15 kHz.
Mfr: Pacific Transducer Co.
Price: $7.00
Circle 66 on Reader Service Card

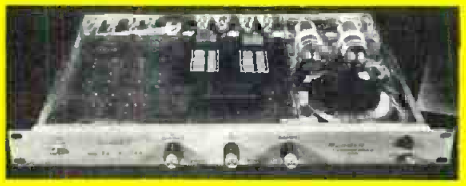
ELECTRET CONDENSER MIC

• This hyper-cardioid condenser microphone performs with maximum noise cancellation. It utilizes a field-effect transistor preamplifier built into the microphone housing which is powered by a 1.5 volt penlight battery and is adjustable to 600 ohms or 10 k ohms output impedance. The MC-201 covers an 8½ octave sound field and operates over a 100 dB dynamic range (The "Electret" is supplied with a specially designed wind-screen which is extremely effective. It is also unaffected by external vibration and is ideally suited to stage use. The power consumption is exceedingly low and normal battery life is approximately one year. The MC-201 comes equipped with a sturdy die-cast stand and adaptor for floor stand or boom.

Mfr: TEAC
Price: $50.00
Circle 63 on Reader Service Card

DYNAMIC NOISE FILTER

• The model 1000 dynamic noise filter reduces noise when playing master tapes, pre-recorded tapes, records, cassettes, or f.m. By varying its bandwidth automatically in response to the music it is able to reduce noise with negligible audible effect on the program content. For low levels attenuation is 25 dB at 30 Hz and 22 dB at 10 kHz. At high levels frequency response is flat within 0.2 dB from 20 Hz to 20 kHz and harmonic distortion is typically .01 per cent at +18 dBm. The chassis accommodates one, two, three, or four channels ganged in pairs for stereo and the number of channels can be varied by plugging in epoxy encapsulated modules. The filter uses an active transformer at its input.



Mfr: Burwen Labs
Circle 63 on Reader Service Card

MONITOR SPEAKERS

• The availability of new loudspeaker systems is announced. The Model 66 is a 3-way, 12-inch woofer, system and has an l-c crossover and two special controls. The company claims its loudspeaker systems provide a legitimate performance *vs.* cost improvement over other systems for two reasons: low costs as a result of modern semi-automated production techniques, and high quality due to sophisticated quality control equipment and procedures. The new loudspeaker systems were designed by Edward M. Long formerly with Audio Dynamics, CTS of Paducah and Ampex Corp.

Mfr: Quadraflex Industries, Inc.
Price: $139.95
Circle 57 on Reader Service Card

G. R. Thurmond

Bass Response in Speakers, Enclosures, and Rooms

Is room equalization really necessary, or will flat components obviate this need? The investigations of this article answer this question.

The contention is that flat components, properly used, will produce a system with flat response.

WITH THE AVAILABILITY of numerous equalization techniques, the response of any sound reinforcing or playback system can be smoothed and flattened. Many people have taken advantage of an equalization technique to improve the response of their system, but others eschew their use. The usual reason given for such a refusal seems to be that equalization is not needed for really good systems. The contention is that flat components, properly used, will produce a system with a flat response.

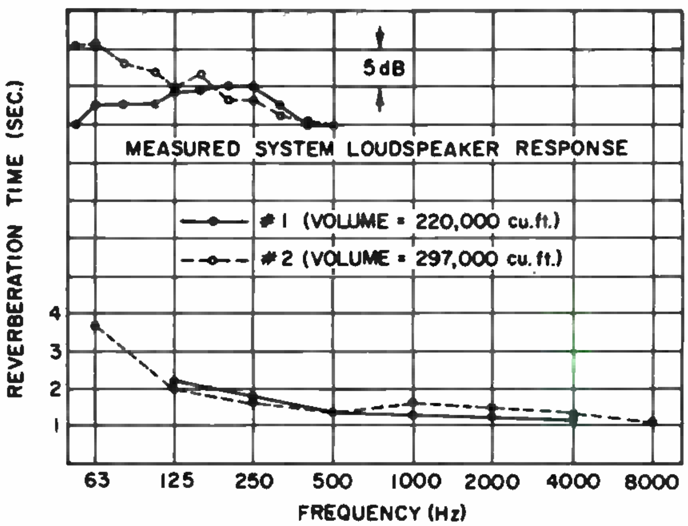
Is this really true? In all our years of experience in the equalization of sound reinforcing systems, we have never discovered one with a flat response.

Granted: in many cases the components are at fault. High frequency horns are anything but flat in response; they are used mainly because the levels which they can generate are mandatory. The response of column type loudspeakers is usually worse. Few microphones are really flat.

But these shortcomings are much more prevalent in the treble region than the bass. It is relatively easy to find microphones and loudspeakers which measure flat in the bass region. Yet, the systems we have measured exhibit strong irregularities in this region, too. Furthermore, these irregularities seem to vary considerably, even between systems with identical components. We decided to examine the responses of systems we have data on to see if a pattern could be detected. If so, this would be a great help in understanding and controlling the actual response.

G. R. Thurmond is an independent acoustical consultant, formerly with McCandless Consultants, Inc., and operating from Austin, Texas.

Figure 1. Measurements of two rather large churches.



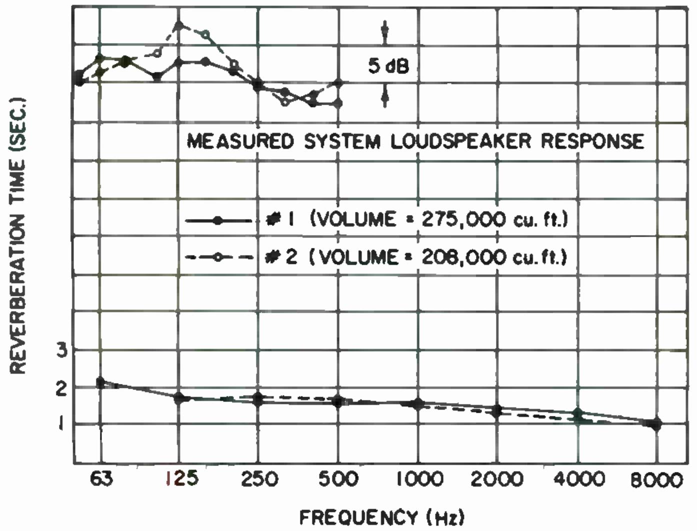


Figure 2. Two medium-sized auditorias.

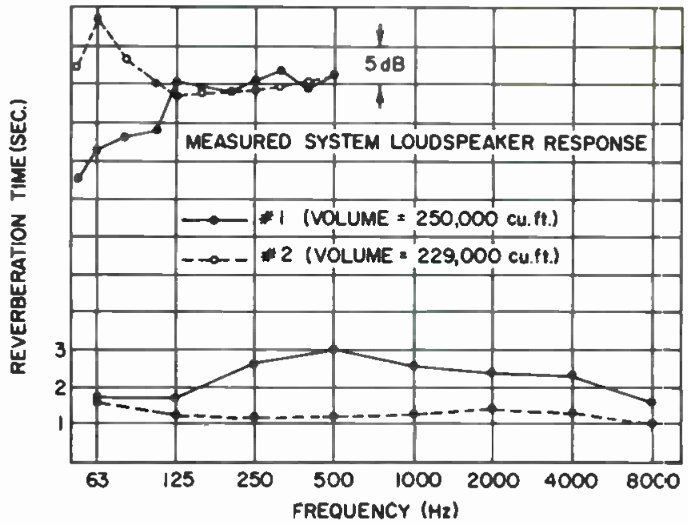


Figure 3. Two nearly identically-sized churches that vary considerably in reverberation time.

One way to search for a pattern in seemingly diverse data is to eliminate as many variables as possible. Following this technique, we confined our examinations to the bass response of systems which used, insofar as possible, identical components, similarly mounted in similar rooms. Furthermore, all systems selected were measured with microphones whose response was well known, so their effects could be corrected out. Only a few systems met all these criteria, and they are all included here. No attempt was made to evaluate loudspeaker sensitivity; all curves are normalized in the 250-500 Hz region.

First, let us consider two rather large churches (FIGURE 1). These two rooms are strikingly similar in appearance and characteristics. Even the loudspeaker systems are almost identical. The same model low-frequency speakers are used in both, installed and mounted almost identically. The only real difference is that the smaller church has two bass speakers, while the larger has four. A comparison of the responses shows a strong similarity with the four-speaker system having a better extreme low end. This is just what we might expect; perhaps we are on to something.

Let us look further, though. We find two medium-sized auditoria which are also quite similar (FIGURE 2). Here, the loudspeakers are not exactly alike, however. The larger auditorium has two bass speakers (the same model as in the churches) in a box hung in a ceiling opening, with no attempt to close the space around the box (again like the churches). The smaller auditorium had four bass speakers (of a different make but similar design) driving a large exponential horn which is formed as part of the ceiling.

Here again we find a difference in the response, but not as much as we might expect. In fact, the difference is quite a bit less than in the churches, especially on the extreme low end. The difference in enclosures certainly should accentuate the difference in responses here. Apparently something else is offsetting this difference. A difference in loudspeaker characteristics? Possible, but it does not seem likely; the designs are too similar. The rooms? We don't measure any significant difference here. But let's see what effect room reverbration might have on the bass response.

The next two churches shown in FIGURE 3 are almost identical in size, but are very different in reverberation (and, of course, design). They each have one low frequency loudspeaker, both the same as in the first two churches. The enclosures are similar, but located differently within the rooms. But look what happens to the bass response! The difference does not seem to correspond at all to the difference in room characteristics. Or is there some significant characteristic of which we are unaware? Could the loudspeaker enclosure or mounting be that important?

Let us look at another comparison (FIGURE 4). Here we have three large gymnasiums of similar size and reverberation. All have exactly the same bass loudspeakers; in this case, a large hornloaded unit. The largest room has two bass units, the middle one has four, and the smallest (ironically) has six. In each case, the speakers are mounted near the center of the room to provide 360-degree coverage. No additional baffles were used in any case.

The similarity in the responses suggests that we may have eliminated all significant variables in this comparison. (Gym #2 has a lay-in ceiling above the speakers, while the others do not; this may account for the variation at the extreme low end.) The loudspeaker type and mounting must be the really significant variables; when they are held constant, everything comes out the same. It is obvious that the room has a powerful effect, (note the free-field response of these bass units) but this seems to be consistent and predictable. In all the previous examples, variations in loudspeakers and mountings could well account for the response differences.

But before we write *q.e.d.* let us think back a little. In the first example of two churches, the loudspeaker types and mountings were essentially identical (as were the rooms), yet there is a significant difference in the responses. If the number of loudspeakers causes this difference, why did it not cause a similar difference in the case of the three gyms? Furthermore, why is there such a strong difference in the responses of the second two churches? Are the loudspeaker mountings really that different? Or is there some factor we are completely overlooking?

It would be nice if we had two rooms where all factors seem identical, or nearly so. The responses should be the same in such a case. If not, then there must be some factor which we have not perceived.

We have such a case shown in FIGURE 5. Here we have two large auditoria of very similar size and design. Both had new loudspeaker clusters installed within the last year, which are essentially identical. The same bass units were used in each (and the same as in the gyms), installed the same way. Our measurements of these systems were unusually thorough. They were made with a Bruel

It would be nice if we had two rooms where all factors seem identical, or nearly so.

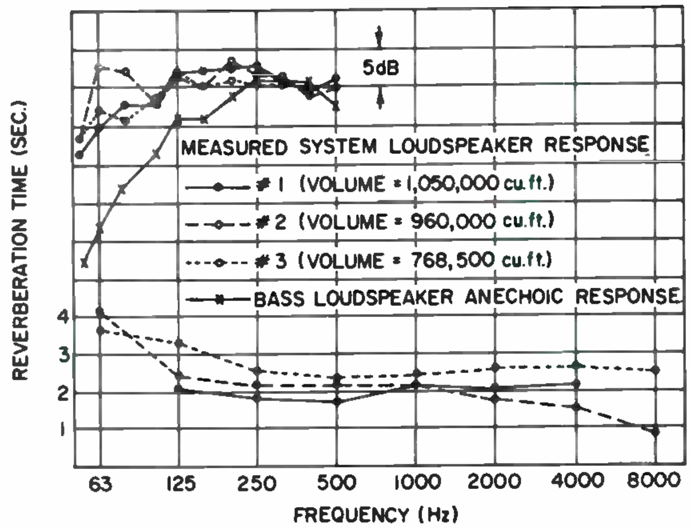


Figure 4. In this comparison of three large gyms, size and reverberation are similar.

and Kjaer condenser microphone, a continuously-swept third-octave filter, and automatically plotted on a graphic level recorder. More than a dozen measurement locations were taken in each case. The over-all averages are plotted out here. And what do we find?

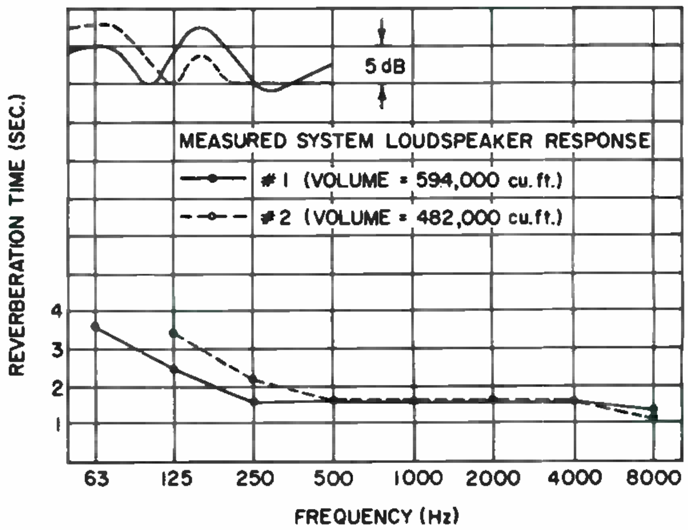
We should expect a very close correspondence between these two systems, for, once again, the significant variables have been eliminated. Even the room affects should be minimal. But it isn't so! To be sure, the two curves bear a family resemblance, but more like cousins than twins.

There is only one possible explanation: the room has more effect than we expected. Furthermore, this effect does not appear to be consistent or predictable. That means that the only way to be sure of a good system response is to use good components, properly installed, and then to measure and correct the final result. And that, of course, means equalization.

But just how necessary is this equalization? All the responses shown are "flat" ±5 dB. This is often accepted as satisfactory, especially in the bass region. But such variations are highly audible; in fact, they account for most of the characteristic sound of each particular installation.

Furthermore, these curves represent almost a "best case." All the troublesome irregularities of treble speakers, crossovers, and microphones were eliminated. Also, small rooms are typically plagued with standing wave irregularities which do not show up in the large rooms we have looked at. All these effects together really mess up a response. No wonder equalization helps! ■

Figure 5. Two large auditoria in which all the criteria seem to be the same.



NEW PRO-4AA

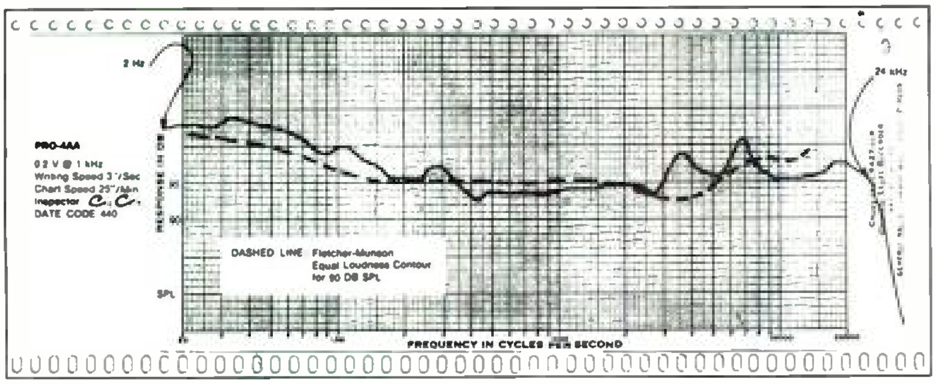
The good sound you monitor is the Sound of KOSS

Better than speakers

Here is a new, rugged, wide-range dynamic headset with good sensitivity that delivers precision monitoring far beyond that obtained by speakers. You exclude outside noise; you eliminate disturbing room reflections. Freedom from distortion and flatness of response are exceeded only by the famous Koss electrostatic units. You'll find that the "Sound of Koss" is the *good* sound.

Price $60.00

TYPICAL RESPONSE CURVE OF PRO-4AA



WHAT THE SPECS DON'T TELL YOU—

EXTENDED BASS RESPONSE TO 2 CYCLES/SECOND—The compliance of the diaphragm is large, the seal is good against the ear and the physical design permits very long excursions to give you lo-lo bass at full level—and the PRO-4AA does it with liquid-filled cushions for perfect seal and superlative comfort, whether you wear glasses or not. This bass range response *exceeds by 4 octaves* that needed to insure good low-end monitoring.

TAILORED CURVE SHAPE FOR FATIGUE-FREE LISTENING — Characteristically the PRO-4AA runs ± 3 dB except below 150 Hz and for a rise at 6.5 kHz. As long as we must have *some* variation, we put the belly of the curve in the mid-frequencies to follow closely the Fletcher-Munson equal loudness contour for 90 db SPL at 1 kHz. This keeps fatigue low in long, intense sessions because you don't reach for the extremes of the range as you tire and your ears become less sensitive. Last, but not least, the music sounds *good.*

HIGHER HIGHS THAN YOU CAN USE—The 24 kHz point on the chart is about where the "2" is in the figure 1521—the PRO-4AA needs longer chart paper than this standard charting equipment provides. Even though you can't hear these frequencies, the capability promotes good transient response in the range you *do* hear.

WHY DOES KOSS RAISE THE BOTTOM?—The wide-range coupler used for Koss measurements effects a perfect seal to promote high bass level almost to dc. An air leak to the sealed cavity lowers the bass response level. Koss feels that modern side-burns and luxuriant hair are good for a full 1-second leak or 4 dB less. This makes the PRO-4AA the flattest dynamic headset we know how to design at this time!

See and hear the PRO-4AA at your dealer today or write the factory for a 16-page catalog on "How to Choose Stereophones."

ELECTRICAL SPECIFICATIONS

Frequency Response Range, Typical: 0-20,000 Hz ■ **Efficiency:** Medium ■ **Total Harmonic Distortion:** Negligible at 95 dB SPL. ■ **Source Impedance:** Designed to work direct from 4-16 ohm amplifier outputs. When used with headphone jacks where series resistors are employed, response should not be measurably affected, but slightly higher volume settings will be required. ■ **Power Handling Capability:** Maximum continuous program material should not exceed 5 volts as read by an ac VTVM (Ballantine meter 310B or equal) with average indicating circuitry and rms calibrated scale; provides for transient peaks 14 dB beyond the continuous level of 5 volts.

PHYSICAL SPECIFICATIONS

Cushions: Fluid-filled for high ambient noise isolation averaging 40 dB throughout the audible range. ■ **Headband:** Extendable, stainless steel with self-adjusting, pivoting yokes; conforms to any head size. ■ **Boom Mount for Microphone:** Knurled, anodized, aluminum knob on left cup with threaded shaft and 2 compressible rubber washers; accepts all standard booms. ■ **Headset Cable:** Flexible, 4 conductor coiled cord, 3 feet coiled, 10 feet extended. ■ **Plug:** Standard tip, ring and sleeve phone plug. ■ **Element:** One inch voice coil virtually "blow-out proof"; takes surges up to 20 times rated maximum power levels. Has 4 square inches of radiating area from 2 mil thick mylar diaphragm. ■ **Weight of Headset Only:** 19 ounces.

KOSS Stereophones

Koss Electronics, Inc., 4129 N. Port Washington Rd., Milwaukee, Wis 53212 • Koss Electronics S.r.l., Via Vallorta, 21 20127, Milan Italy

Circle 30 on Reader Service Card

Michael Rettinger

Directional Response of Some Multi Transducers

Multi speaker and multi microphone setups respond to specific laws of acoustics. Failure to acknowledge this causes problems in both recording and in auditorium acoustic control. This article shows the directional characteristics of some common combination-system tranducers.

The response of multiple transducers is subject to the vectorial composition of the sound rays at the point of interest.

MULTIPLE LOUDSPEAKERS and multiple microphones are presently employed for a wide variety of purposes. In the case of multiple sound radiators, we have (for instance) two-way and three-way loudspeaker systems in which separate emitters are employed for the reproduction of the different frequency bands; we have the three- and five-loudspeaker systems behind certain motion-picture screens for the reproduction of stereophonic sound, as in Cinemascope and Cinerama, etc.; and we have the column or columnar loudspeakers which embody several identical emitters. For the recording of stereophonic sound and, to some extent, for the recording of multiple sound tracks, we have an array of similar microphones energized by a single sound source.

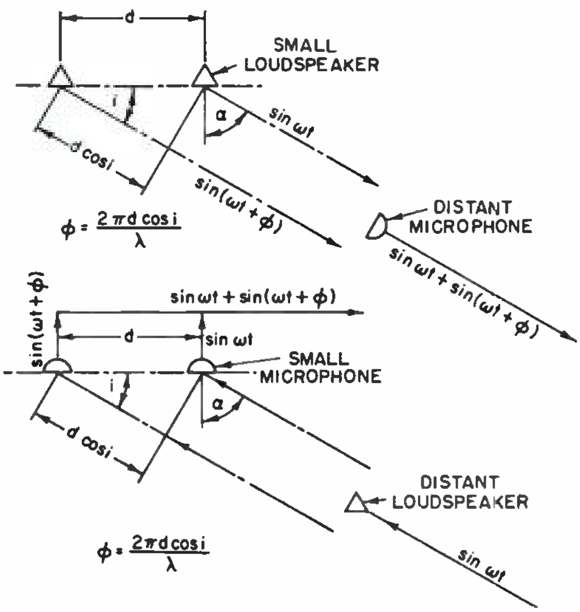
The response of multiple transducers is subject to the vectorial composition of the sound rays at the point of interest. It is the purpose of the following to illustrate the directional characteristic of some comomn multiple acoustic transducers.

The top FIGURE 1 shows two loudspeakers spaced a distance d apart, and issuing sound at the radiation angle a. In the following it is assumed that the loudspeakers are small (theortically point sources), that they are emitting the same frequency (which for a two-way loudspeaker system corresponds to the frequency of the crossover network), that the signals are of equal amplitude, that the emitters are vibrating in phase, and that the receiving microphone is sufficiently distant to assure plane-wave reception.

Michael Rettinger has appeared in our pages before. He is an independent consultant on acoustics based in Encino, California.

The lower part of FIGURE 1 shows two small, equally sensitive, non-directional microphones separated a distance d and receiving plane-wave sound coming to them at a reception angle a. As in the case of the two loudspeakers shown on top of the illustration, the combined signal is given by:

Figure 1. Two loudspeakers spaced a distance apart issuing sound at an angle. Below the speaker drawing, a similar situation exists except microphones are substituted.



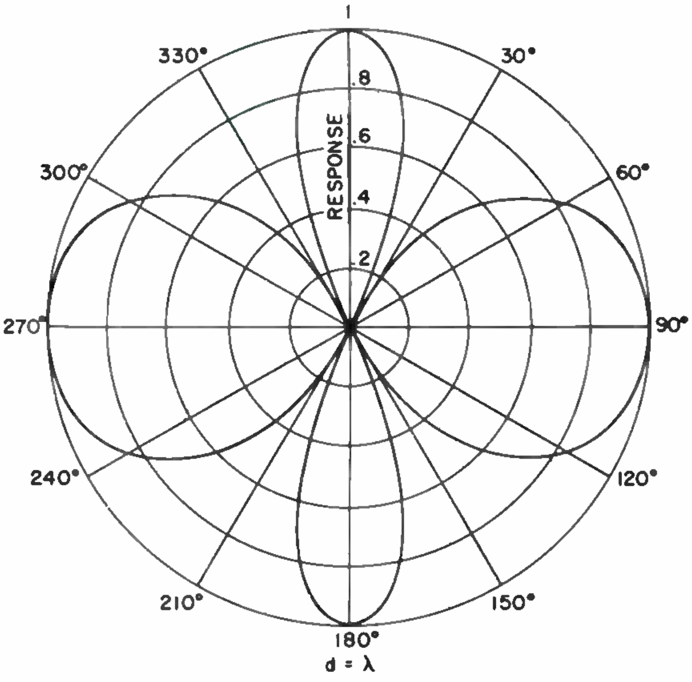


Figure 2. The directional characteristic of two separated equally-sensitive, phased, transducers with the distance between them equal to one wavelength.

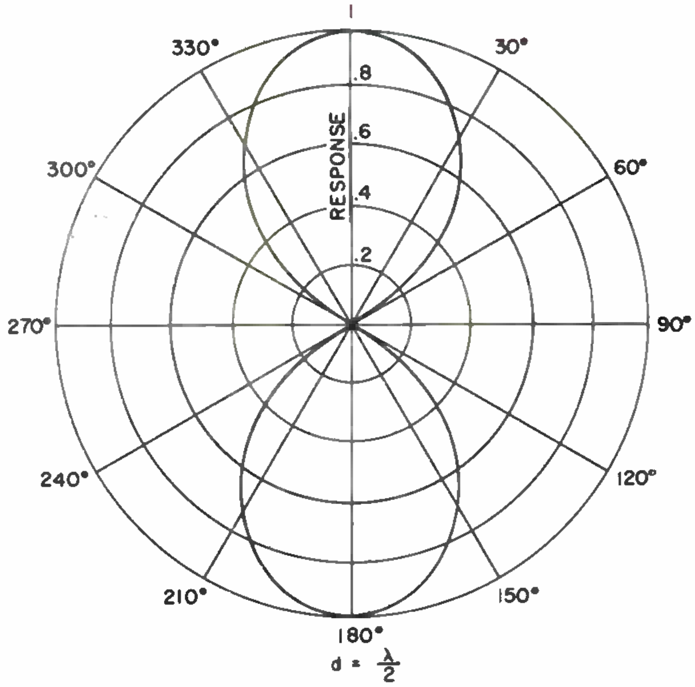


Figure 3. The directional characteristic of two separated transducers, this time spaced one-half wavelength apart.

$$R = \sin \omega t + \sin (\omega t + \varphi) \qquad (1)$$

where $\omega = 2\pi/\lambda$

λ = wavelength

t = time

φ = phase

$$= \frac{2\pi d \cos i}{\lambda}$$

$$= \frac{2\pi d \sin (90-i)}{\lambda}$$

$$= \frac{2\pi d \sin a}{\lambda}$$

Note that in equation (1) the amplitude of the signal ($\sin \omega t$) was taken as unity for the simple determination of the pertinent directional response.

To be able to plot the directional characteristic of the two separated, equally sensitive, omni-directional transducers operating in phase as a function of the ratio of the distance between them and the wavelength, equation (1) must be transformed into another expression. This rather lengthy procedure has been done elsewhere[1] and shall not be repeated here. The result is

$$R = 2 \cos \left(\frac{\pi d \cos i}{\lambda}\right) \sin \left(\omega t + \frac{\varphi}{B}\right) \qquad (2)$$

The cosine factor before the sine term in the above expression may be considered the amplitude of the resultant signal.

Another expression for the directional characteristic of multiple in-line transducers is[2]:

$$R = \frac{\sin nK}{n \sin K}$$

where n = number of transducers arranged in a line

$$K = \frac{\pi d \sin a}{\lambda}$$

Depending on the relationship between source separation and wavelength, the polar pattern of the transducer can become quite complicated.

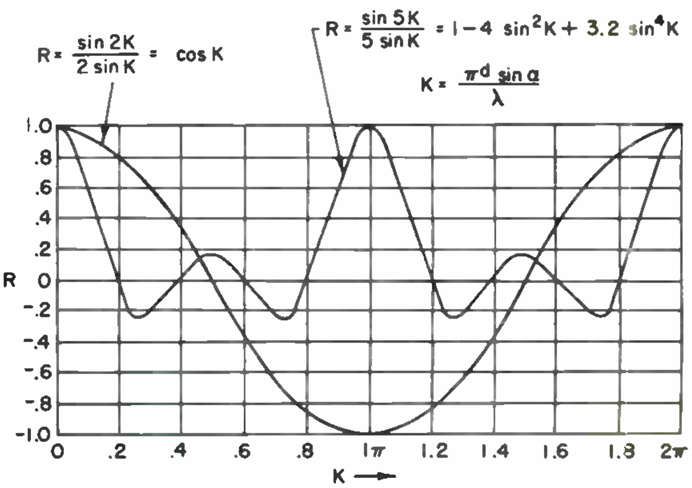
FIGURE 2 shows the directional characteristic of two separated, equally sensitive, phased, small transducers when the distance between them is equal to a wavelength, while FIGURE 3 presents the directional characteristic of such transducers when the distance between them is equal to half a wavelength.

Depending on the relationship between source separation and wavelength, the polar pattern of the transducer can become quite complicated. Thus, for $d = 5\lambda/2$, the pattern exhibits ten lobes (instead of the four shown on FIGURE 2).

In the case of two-way loudspeaker systems, with the units either 1 ft. or 2 ft. apart, FIGURES 2 and 3 show what happens to the frequency response of such a system at a certain radiation angle when the crossover network frequency is 1000 hertz, respectively 500 hertz, as it often is. Co-axial two-way loudspeaker systems are less likely to demonstrate such an objectionable response.

Co-axial two-way loudspeakers are less likely to demonstrate . . . objectionable response.

Figure 4. The directional characteristic of two small transducers in terms of positive and negative response.



... one must genuinely wonder why the reproduced sound is often so intelligible ...

In FIGURES 2 and 3, the response of the transducers is plotted in what might be called absolute terms, that is, without respect to a positive or negative sign associated with the response, because this is the conventional way of plotting such polar patterns.

FIGURE 4 shows the directional characteristic of the two small transducers in terms of a positive and negative response. The negative response means that a phase shift of 180 degrees has taken place on part of the sound pressure at the position of observation.

FIGURE 4 also shows the directional characteristic of an in-line array of five small loudspeakers separated a uniform distance *d* from each other, according to the equation

$$R = \frac{\sin 5K}{5 \sin K} = 1 - 4 \sin^2 K + 3.2 \sin^4 K \qquad (3)$$

point sources, the directional characteristics of multiple in-line emitters, plotted in three dimensions, are surfaces of revolution about the line as an axis.

Considering the fact that two-way loudspeaker systems exhibit marked lobes in their directional pattern at the frequency of the crossover network, that three and five-loudspeaker systems (as behind some motion-picture screens) display as shown above a distinct or characteristic polar pattern of their own, that the loudspeaker frequency response of even one emitter system measured in the open on its axis generally leaves something to be desired in uniformity, that in the theater numerous reflected sound rays add their own phase and amplitude distortion to the signal, one must genuinely wonder why the reproduced sound is often so intelligible—as if the ear could synthesize order out of disorder. ■

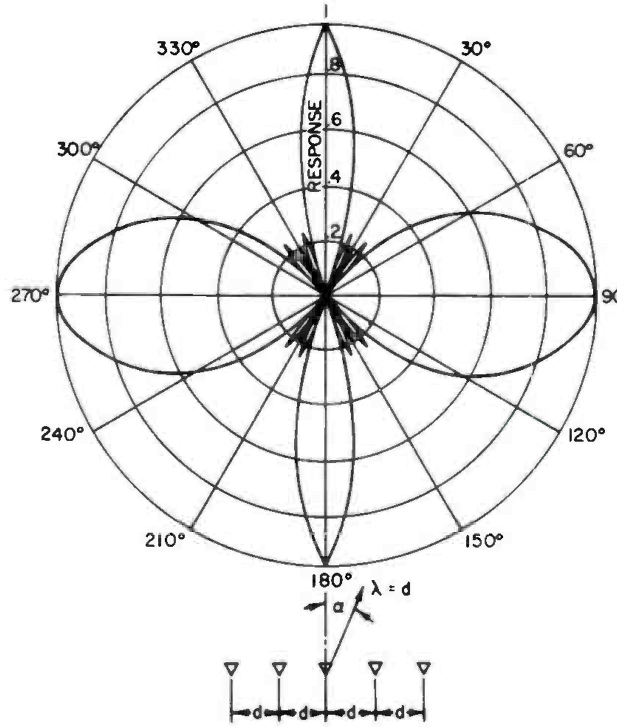


Figure 5. The polar pattern of the formula given in the text as equation (3).

REFERENCES

[1] *Acoustics—Room Design and Noise Control*, by M. Rettinger, page 87, Chemical Publishing Co., 200 Park Ave. South, New York, N.Y.

[2] *Acoustical Engineering*, by H. F. Olson, page 35, D. Van Nostrand Company Inc., Princeton, N.J.

Marshall King

The Sync Pulse in t.v. Audio, Part 2

The author continues in his exploration of the methods and techniques of synchronizing video tape machines to audio tape machines.

Many of us have discovered, the hard way, that using half of one method with half of another simply will not work.

NOW THAT we have examined some ways whereby sync pulses from the video tape machines can be used in audio, and how these same machines can be used as "resolvers" in maintaining synchronous sound in double-system recording, let's look closer at the general nature of *resolving* and at other ways it may be achieved.

Our goal is to maintain a synchronous relationship between picture and sound during playback when the two are recorded on separate machines. Since there are several ways to do this, it is not difficult to become thoroughly confused in trying to keep one method clear in mind from the others. Many of us have discovered, the hard way, that using half of one method with half of another simply will not work. What we do, as we sit down to lick our wounds, is try to think of one or two "infallible guides" that will keep us out of trouble in any chosen technique. *Are* there any simple truths that will apply to all methods of handling synchronous sound? If so, let's search for them here, then see if the systems we are about to investigate conform to those truth.

At the outset it may be well to remind ourselves what the fuss is all about. In cases where we record picture on one tape machine and sound on another, why *won't* they play back together if only we can get them started together?[1] There are at least two reasons, and it is these which necessitate the use of sync pulses. First, since the tape is moved across the heads as a result of contact pressure between the tape and a rotating shaft, rather than on sprocket holes and gears as in film, there can be slippage between tape travel and the motor that drives it. Since slippage can scarcely be relied upon to be constant, it is a strong negative factor which wipes out our assurance of proper and consistent playback times. Second, the power that drives our machine motors may vary either in amplitude or frequency. Batteries do run down, and a.-c. frequencies do change from one geographic location to another. The solution to these and other undesirable variables is to focus our attention on, not the speed consistency of various kinds of drive motors, but on the rate of tape travel, *regardless what the motor is doing*. A convenient way to do this is to record reference marks (sync pulses) on our tape, and to control the rate at which these go by a magnetic head during playback, *regardless what the motor is doing*. Under such control, the playback speed of a machine may or may not be steady, but it *will* accurately track the speed of its companion. In other words, we are putting the variables under our control.

Thus, our attention is properly directed to the tape up on the top floor, not on the motor down in the basement. In strict film operation, on the other hand, all attention is given to the motor downstairs, for, with gears and sprocket holes, the film up above must follow exactly.

To keep our thinking from being cluttered by endless variations, we are going to say that we have only four sync-handling machines to consider. Film technicians give

[1] We shall assume throughout this discussion that we have means for having two tapes arrive at a common beginning point of a program at their proper operating speeds.

Batteries do run down, and a.-c. frequencies do change from one geographic location to another.

. . . our concern is not whether the playback time of the program is greater or lesser than the record time, but whether the picture and sound stay together during such playback.

them one set of names and v.t.r. (video tape) people give them another, but regardless of name, their individual four functions are to *record* or *play back* either *picture* or *sound*, as indicated in FIGURE 1.

It can be seen from this that the film technician has a vocabulary somewhat more indicative, if not more precise, when he designates his four machines as *camera*, *recorder*, *projector*, and *reproducer*. In video tape we have somehow settled for inferences, knowing that both the video tape and audio tape machines can perform the dual functions of recording *and* playing back. It is not sleight-of-hand, but an inevitable ramification that we have ignored the television camera as one of our important machines. Since it neither records nor plays back, we can dismiss it out of hand.

With regard to the differences between film and tape, it must be admitted that the dividing line between the two is becoming more scanty as time goes on, and it is increasingly important that we share concepts and terminology. Since we now see sprocketed film equipment in the inventory of many television studios, and since we see television gear on many movie lots, we can no longer say that the tools and techniques of the two media are mutually exclusive. Such differences, as they have existed, have often been due more to the avoidance of boundary violations in trade jurisdiction than to any lack of interest in common goals.

Let's consider some terms, many of which are invented for this discussion.

1. *Single-system recording* is where both a picture and its corresponding sound are recorded on a single machine, such as the single-system film camera, in which both picture and sound are recorded within the camera on film. This practice, developed in 1928 at Fox Studios, is no longer a widespread means of recording sound for motion pictures, now that magnetic tape has been developed and proven superior. A more important example of single-system recording today is video tape operation as it is followed in the majority of cases.

2. *Double-system recording* is where a picture and its corresponding sound are recorded on separate machines, as in the common motion-picture technique of recording the picture on film within the camera and sound on a tape recorder. It is the possible need for the adaptation of this practice in television that is being investigated here.

3. *Companion* is the relationship of one machine to another when they are engaged in recording or playing back a picture and its corresponding sound.

4. A *carrier* is a strip of material which contains program information and concommitant control cues. This information can be optical (film), magnetic (tape), or mechanical (sprocket holes). A *prime carrier* is one in which the recorded information can be made to control the motion of a second, or *slave carrier*.

Figure 1. Film technicians give them one name, and video tape another, but the functions are clear.

	FILM		TAPE	
	RECORD	PLAYBACK	RECORD	PLAYBACK
PICTURE	CAMERA	PROJECTOR	VIDEOTAPE MACHINE	VIDEOTAPE MACHINE
SOUND	RECORDER	REPRODUCER	AUDIOTAPE MACHINE	AUDIOTAPE MACHINE

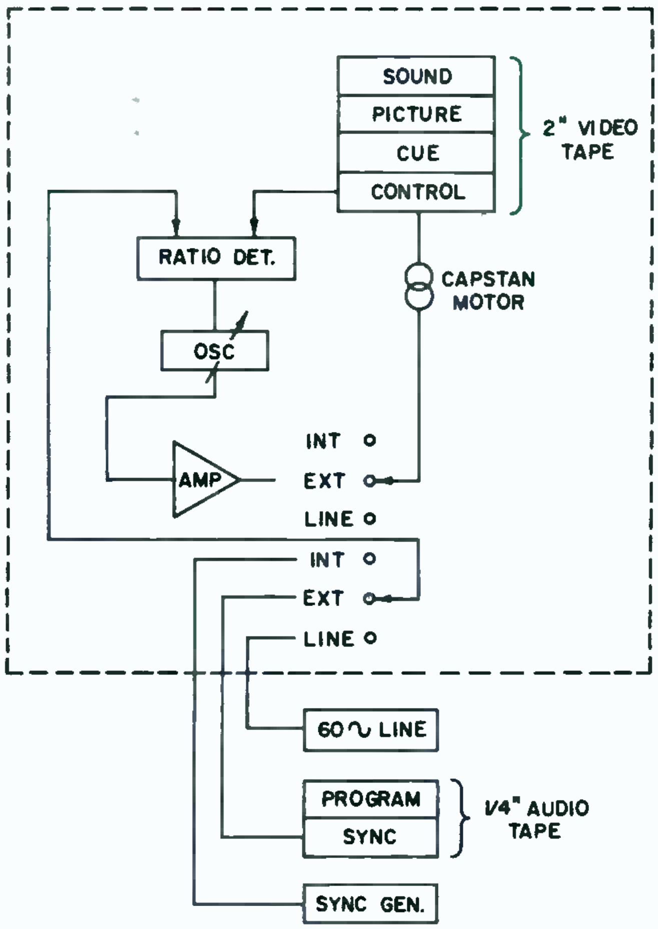


Figure 2. This video tape machine has a servo circuit that uses a ratio detector and a variable oscillator.

5. *Slippage* is a form of inefficiency between interconnected moving parts whereby the coupling at one or more points is not positive enough to maintain a linear ratio of 1:1. For example, even though a gear having 100 teeth drives a gear having 10 teeth so that the gear ratio is 10:1, it is intended that the *linear* speed on one circumference against the other will be 1:1. The chance of slippage is even greater at couplings where contact pressure, rather than sprockets and gears, is relied upon for positive drive. Such a case is in tape equipment where a rubber idler wheel presses the tape against a rotating capstan shaft in order to get carrier drive.

6. *Carrier stretch* is the condition whereby a magnetic tape or other carrier has stretched from its original length due to stress, handling, aging, heat, or other cause. This is a particularly serious defect requiring strict monitoring by a servo circuit during *all* playback operations; otherwise, no matter how constant the motor speed in both *record* and *playback operations*, there will be a poor, or even unusable, relation between the two operations. The deleterious effects of slippage and stretch represent the biggest differences between tape and film operation. Film solves the problem of slippage by the use of gears and sprocket holes; tape solves its slippage and stretch problems by the use of servo circuits. Part of our inquiry here is to see how the two methods can work together on the same project.

7. *Slave operation* is where, during playback, the reference marks on one carrier passing its playback head causes corresponding marks on its companion to pass its playback head at the same time.

8. *Autonomous operation* is where, after starting together during playback, the reference marks on both a carrier and its companion refer to the constant frequencies from which they were derived.

Slowly, the picture slows down to something below normal as the sound speeds up to something above normal . . .

9. A *frame* is the smallest amount of picture information that constitutes a complete instantaneous view of a scene. Common frame rates of movement are 24 frames per second in films and 30 frames per second in television.

10. *Sync* is the coded instruction laid down on a carrier which can be used during playback to control the speed of the carrier.

11. *Resolving* is the method whereby the sync is extracted from a carrier and used to control its speed during playback.

12. A *resolver* is a device used to compare two frequencies by electronic sampling and to bring one in step with the other by applying feedback information between the two.

It is not our purpose here to give a technically-detailed schematic view of the inner workings of a resolver, but rather, to show in broad terms the method whereby resolving is accomplished. Since an integral part of a video tape machine is a circuit which is also the heart of a resolver, let's use this as our model. It's a servo circuit which consists essentially of a ratio detector and a variable oscillator, as indicated in FIGURE 2. Later we may show this in a single block labelled *servo*. As shown, the ratio detector is receiving sync pulses from two sources. Coming from the right is a signal from the tachometer playback head (tach head)[2] which reproduces the recorded sync pulse. The input on the left comes from any selectable source; it could be the 60-hertz line frequency, the sync pulse from another tape machine (such as an audio tape), or the output of a sync generator. Whatever it is, it is the frequency with which the v.t.r. machine is asked to stay in step during playback. This is brought about as follows:

The rate at which the recorded sync pulses go by the tach head in the v.t.r. machine is compared in the ratio detector to (in this case) the sync output of an audio tape machine in the playback mode. If the former is higher than the latter, the detector sends a positive voltage to the variable oscillator. If it is lower, it sends a negative voltage. These small correction voltages affect the frequency-determining components of the oscillator so that the capstan motor is either speeded up or slowed down, as required, in order to stay in step with the incoming frequency it is being compared to. Here, one tape machine is being made to follow another, which is called *slave* operation. It could as well be made to follow the constant-frequency generator which supplied the sync pulses during recording, and this would be *autonomous* operation. These two will be analyzed in a moment. In either case, the picture stays in step with the sound.

Again, the v.t.r. machine is represented here in the broadest terms. No effort is being made to show designations as they may actually appear on the machine. The video tape operator does have selectable inputs to assign to the drive motor, but they are not labelled as shown here. We are calling the selections *internal*, *external*, and *line* merely for convenience of illustration. Regarding circuitry and numbers, the only v.t.r. detail we shall note is that, for reasons pertaining to the electronic handling of the color vectors of a video picture, the capstan motor of a video tape machine is driven at 59.95 Hz rather than

[2] A tachometer head is a playback head in the tape deck which, instead of picking up program audio, picks up the sync pulses which have been recorded on the tape.

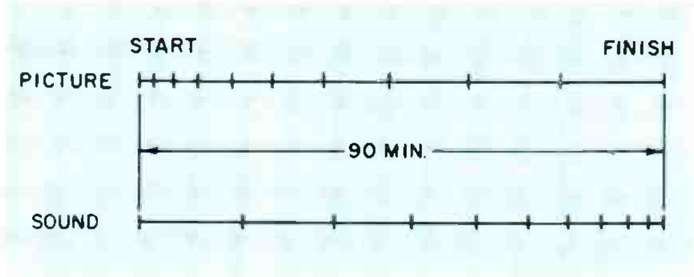


Figure 3. In a given program, the start and finish may be together with the sync still far off in between.

60 Hz as might otherwise be expected. A form of this frequency[3] is also recorded on the *control track* of the video tape. This is tantamount to putting sprocket holes in film stock. In other words, these recorded sync pulses may be considered to be electronic sprocket holes, and are put to work in the servo circuit as indicated in Figure 2.

In order to prove what it takes to maintain synchronous playback between picture and sound, we must first define what synchronous playback is. With regard to one being out of sync with the other, we must first acknowledge that there is a lower limit of discrepancy, a lesser value of which will be found to be acceptable. We'll define this lower limit arbitrarily on the basis of experience.

To illustrate, our concern is not whether the playback time of a program is greater or lesser than the record time, but whether the picture and sound stay together during such playback. Sponsors and network executives may be upset if a 60-minute program runs 56 minutes, but as viewers we are not apt to be aware of it. Therefore, we might be tempted to say that our requirements are met when the picture and sound start together and finish together.

But this, too, will never do for our definition, for in a 90-minute program where picture and sound start and finish together, the sync could be very far off at all intermediate points, as illustrated in Figure 3. The segments shown on the two horizontal lines represent portions of the program which took equal amounts of real time to shoot. Obviously, during the playback circumstances shown, the two start together but the picture goes by much faster than the corresponding sound. Slowly, the picture slows down to something below normal as the sound speeds up to something above normal, and the two arrive in step once more as the 90th minute goes by.

Clearly, we must insist that the two stay together consistently during shorter intervals, down to the point where we are not aware of a discrepancy. It matters not that the picture and sound may wander in and out of sync exactly as shown in Figure 3 providing that the time interval of such wandering before "catching up" is not 90 minutes, but rather, something on the order of, say, a tenth of a second. Perhaps we can say that this is the maximum interval within which any sync discrepancy cannot be discerned by our senses. Or, *synchronous operation is where picture and sound are not out of step with each other for more than a tenth of a second at any time in a continuous playback*, and our goal is to achieve this situation. Next month, I will get to a group of rules that will be nothing more than what was left over after discarding one bad practice after another in the many years of doing it the hard way. ■

[3] Strictly speaking, only the v.t.r. capstan motor is driven at 59.95 Hz, while four times this frequency, or 239.80 Hz, is generated for the control track (also for reasons pertaining to color accuracy). But it is only one-fourth of the 239.80 on the control track that is observed by the ratio detector and used for sampling motor speed. So for our purposes here we show a single sync generator supplying 59.95 Hz for both motor drive and control track, as well as having a third output to feed this common sync to the audio tape.

letters

(continued from page 4)

example, the powers on his redistributed (A' and B') channels become:

power in A' = $0.75^2 = 0.563$

power in B' = $0.25^2 = 0.063$

The sum of these powers is only 0.625, as opposed to the original power in channel A of 1.0. We therefore have a loss of 10 $\log_{10}$ (0.625) or −2.04 dB. This power loss is caused by a loss of power in the (L−R) component, as I will demonstrate generally. To do this, it is necessary to use the algebra of complex quantities, or 2-dimensional vectors. However, the generality of the results is worth the mathematical complication. *Definitions*: given some complex vector A. A has a real part, Re A, and an imaginary part, Im A. $A = \text{Re}\,A + j\,\text{Im}A$ where j is $\sqrt{-1}$. A has a complex conjugate, A^*, defined as $A^* \triangleq \text{Re}A - j\text{Im}A$. A may be expressed in *polar form* as $|A| \exp(j\,\theta)$, where $\theta \triangleq \arctan \frac{\text{Im}A}{\text{Re}A}$ and the magnitude of A, $|A| \triangleq (\text{Re}^2 A + \text{Im}^2 A)^{1/2}$. $|A|$ corresponds to the length of the vector A, and θ to its angle from the origin. The power in A, $|A|^2 \triangleq AA^*$

Sum and Difference Relationships: given L, the left channel, and R, the right channel. Define

the sum $S \triangleq L + R$ (1)

the difference $D = L - R$ (2)

Let us now find the sum of the powers in the left and right, which is what we hear when we listen in stereo.

Adding (1) and (2) $\quad 2L = S + D$

Subtracting (1) and (2) $\quad 2R = S - D$

$$4L^2 = (S + D)(S^* + D^*) = SS^* + DS^* + SD^* + DD^* \quad (3)$$

$$4|R|^2 = (S - D)(S^* - D^*) = SS^* - DS^* - SD^* + DD^* \quad (4)$$

Adding (3) and (4)

$$4(|L|^2 + |R|^2) = 2(SS^* + DD^*) = 2(|S|^2 + |D|^2) \quad (5)$$

This means that the sum of the powers in the left and right channels is proportional to the sum of the powers in the sum and difference signals. These powers depend only of the *length* of the vector, and *not on its phase*. This means that *we can shift the phase of the sum and/or difference signal without affecting the sum of the powers in the left and right channels*. This is a very powerful result. We note, however, that if we modify the frequency response of either the sum or difference channel, as Mr. Schulze proposed, then we will affect the stereo reproduction.

The first useful application of eq. (5) comes in the field of f.m. broadcast. Broadcasters have always been disturbed by "phase distortion swish" in mono reproduction from material recorded on stereo cartridge tape. However, if we record the sum, S, on one stereo channel, and the difference, D, on the other, and then add and subtract after playback, eq (5) tells us that the stereo reproduction will not be degraded in terms of frequency response. Instead, the phase shift between the two gaps of the stereo head will result in stereo placement errors at higher frequencies, which is a degradation of second-order importance compared to the huge nulls in the frequency response of the mono sum caused by phase cancellation when the standard right/left recording technique is used.

The Orban/Parasound Stereo Synthesizer uses eq. (5) to generate pseudo-stereo with phase shift networks, in which the frequency response of the pseudo-stereo is identical to the original mono. (ref. 1)

Given certain constraints, eq. (5) also points towards a correct solution for several of the problems Mr. Schulze originally stated, notably "the work is better divided between the two speakers." Most of the power requirements for the reproduction of music are in the bass range. The capabilities of the average consumer stereo amplifier are far better utilized if the bass energy is equally divided between the channels, so both channels can contribute power. Provided that the bass is in phase on both channels (a requirement satisfied if the recording was mixed through panpots), we will demonstrate below a technique for moving the bass equally to both channels without affecting the total power, and without affecting the mono sum signal, either.

Let the component of some bass frequency on the left channel have amplitude of 1 and let us choose our 0 degree phase reference to coincide with the phase of this vector. Let the component due to this frequncy on the right channel have k times the length of the left component, and the same phase as the left component. Then

$$L = 1\,e^{j0} \quad (6)$$

$$R = k\,e^{j0} \quad (7)$$

Adding (6) and (7)

$$S = L + R = (1 + k)\,e^{j0} \quad (8)$$

Subtracting (6) and (7)

$$D = L - R = (1 - k)\,e^{j0} \quad (9)$$

We will now shift the phase of D by 90°.

$$D' = (1-k)\,e^{j90^\circ} = j(1-k) \quad (10)$$

We now derive our new left and right vectors:

Adding (8) and (10)

$$2L' = (1 + k) + j(1 - k) = (1 + 2k + k^2 + 1 - 2k + k^2)^{1/2} \exp\left(j \arctan \frac{1-k}{1+k}\right) = \sqrt{2}(1 + k^2)^{1/2} \exp\left(j \arctan \frac{1-k}{1+k}\right) \quad (11)$$

Subtracting (10) from (8)

$$2R' = (1 + k) - j(1 - k) = \sqrt{2}(1 + k^2)^{1/2} \exp\left(j \arctan \frac{k-1}{1+k}\right) \quad (12)$$

We observe that the magnitudes of our new left and right components are the same, as desired. From eq.(5), we have not affected the total power in the left and right channels. Since we have not touched the sum signal, that too remains the same, so the channels still add up in mono as they did originally. We have not touched the magnitude of the difference signal, so the degree of vertical modulation is still the same as it was before. Any attempt to reduce the vertical modulation while keeping the sum of the left and right channel powers constant must change the sum signal, and thereby the frequency balance between stereo and mono reproduction. The best we can do to improve the situation without degrading either the stereo or the mono sum is to introduce a phase shift in the difference channel as we have done above. Best musical and mechanical stereo/mono compatibility in stereo discs are mutually exclusive, and all we can do is trade one off against the other. An interesting tradeoff between the two has been made in the Orban/Parasound Stereo Matrix, which has full musical compatibility, and a typical vertical/lateral ratio of 0.65/1. Interested readers are referred to ref. (1).

It is hoped that this discussion has shed some light on the commonly misunderstood phenomonon of difference signal bass rolloff.

[1] Robert Orban, *A Rational Technique for Synthesizing Pseudo-Stereo From Monophonic Sources*, J. Audio Eng. Soc., 18, 157, (1970)

Robert Orban
Parasound, Inc.
San Francisco, Cal.

BOOKCASE

As a service to our readers we are pleased to offer these books from prominent technical publishers. All prices listed are the publishers' net. Shipping charges are included.
Use the coupon at the bottom of the page or give the complete title, author and coupon number. Be sure to indicate quantity on the special instructions line if more than one copy of a title is wanted. Full payment must accompany your order. We cannot ship c.o.d. Checks or money orders should be made payable to Sagamore Publishing Company, Inc. Because of the time required to process orders, allow several weeks for the receipt of books.

Radio Engineering Handbook, 5th Ed.
Henney. Prepared by a staff of specialists, this working manual of radio science provides information on each of the branches of radio engineering, with emphasis on working practice, final working formulas, dimensions, and actual useable circuits.
1959. 1,800 pp. **$31.50**
Circle 36 on Coupon Below

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**
Circle 40 on Coupon Below

Electronic and Radio Engineering, 4th Ed.
Terman. A thorough coverage, in easy-to-understand terms, of those principles and techniques which are the basic tools of the electronic and radio engineer. 1955. 1,078 pp. **$19.50**
Circle 39 on Coupon Below

The Technique of the Sound Studio **NEW EDITION**
by 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. **$13.50**
Circle 1 on Coupon Below

Radio Broadcasting
edited by Robert L. Hilliard. Each of the five chapters has been written by a prominent educator with an extensive background of practical experience in commercial and educational broadcasting. 190 pages: 6¼ x 9½; indexed; clothbound. **$6.95**
Circle 2 on Coupon Below

Transistors for Audio Frequency (Audio-Frequency Amplification)
by Guy Fontaine. 1967. This systematic and detailed treatment of the application of transistors in audio-frequency amplifiers shows how the published transistor characteristics are related to the principles of design. An ideal textbook or reference on the subject for engineers and advanced technicians. 384 pages; 5½ x 8; illus.; clothbound. **$7.95**
Circle 12 on Coupon Below

Closed-Circuit Television Handbook
by Leon Wortman. Gives comprehensive detailed information about the field in an easy-to-understand presentation, suited to those who plan to use, install, and service cctv. 288 pages; 5½ x 8½; clothbound. **$5.95**
Circle 18 on Coupon Below

An Alphabetical Guide to Motion Picture, Television, and Videotape Production
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, and much more. 1970. 480 pp. **$24.50**
Circle 35 on Coupon Below

Integrated Circuits
Motorola. This complete, authoritative volume of design and construction techniques for modern integrated circuits covers theory, thin-film techniques, diodes, transistors, thermal design, and other vital aspects.
1965. 373 pp. **$13.75**
Circle 30 on Coupon Below

Basic Bibliography of Science and Technology
McGraw-Hill. Important to the researcher, librarian, or student, this reference book lists and describes over 8,000 outstanding books in every scientific and technical field. A topical index organizes all subject headings into about 100 general categories enabling the reader to locate quickly the listings which pertain to his field of interest.
1966. 738 pp. **$19.50**
Circle 34 on Coupon Below

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**
Circle 33 on Coupon Below

The Audio Cyclopedia (2nd ed.)
by 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 9-3/8; hardbound. **$29.95**
Circle 20 on Coupon Below

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

Acoustics—Room Design and Noise Control
by 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; hardbound. **$17.50**
Circle 21 on Coupon Below

Sagamore Publishing Company, Inc.
980 Old Country Road, Plainview, N. Y. 11803
Please send me the books I have circled below. My full remittance in the amount of $.......... is enclosed. N.Y. State residents add 6% sales tax.
1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17
18 19 20 21 22 23 24 25 26 27 28 29 30 31 32
33 34 35 36 37 38 39 40

Name ....

Street Address ....

City .... State .... Zip .....

Special Instructions ....

Canada and foreign: Add $1.00 per book

Acoustical Tests and Measurements
by Don Davis. Provide. a 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**
Circle 7 on Coupon Below

Television Broadcasting
Chinn. A practical. compact technical guide to the equipment. systems. facilities. good engineering practices. and operating techniques of television broadcasting. 1953. 695 pp. **$17.00**
Circle 37 on Coupon Below

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**
Circle 8 on Coupon Below

Handbook of Broadcasting, 4th Ed.
Abbott and Rider. This non-technical handbook sets forth clearly and thoroughly the fundamentals of radio and television broadcasting. Over 100 illustrations amplify the text.
1957. 531 pp. **$9.50**
Circle 38 on Coupon Below

Electronic Musical Instruments
by Richard H. Dorf. Now in its third edition and sixth printing since its first appearance in 1954. this is considered the authority on electronic organs. 393 pages: 239 diagrams and photographs. **$10.00**
Circle 19 on Coupon Below

Intermodulation and Harmonic Distortion Handbook
by Howard M. Tremaine. A complete reference guidebook on audio signal intermodulation and harmonic distortion. 172 pages: 5½ x 8½: softbound. **$4.25**
Circle 9 on Coupon Below

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 semi-conductor principles. Features questions to answer, problems to solve. 1968. 169 pp. **$9.95**
Circle 31 on Coupon Below

Practical PA Guidebook: How to Install, Operate and Service Public Address Systems
by Norman H. Crowhurst. 1967. This book gives all the basics needed to become a successful PA operator, in any situation where the reinforcement, relay, or distribution of sound can provide a service. 136 pages; 6 x 9; illus.; softbound. **$4.50**
Circle 15 on Coupon Below

CLASSIFIED

Closing date is the fifteenth of the second month preceding the date of issue.
Send copy to: Classified Ad Dept.
db THE SOUND ENGINEERING MAGAZINE
980 Old Country Road, Plainview, New York 11803

Rates are 50¢ a word for commercial advertisements. Non-commercial and employment offered or wanted placements are accepted at 25¢ per word.

FOR SALE

STUDIO CLEARANCE SALE #3. Ampex AG-300-8-C Solid State 8 track with sync. panels, console cabinet, using AG-350 electronics $5995.00. Ampex AG-350 full track recorder with two track stereo playback through McIntosh C-22 rack mount preamp. $1095.00. Ampex 300-2 two track stereo recorder with 350 electronics and new heads $1095.00. CROWN SS-722 with case and extras $695.00. Teletronix LA-2A leveling amplifiers $225.00 ea. Altec 1567A mixers with 3 transformers and meter $195.00 ea. Pultec MAVEC preamplifier equalizers $185.00 ea. A.K.G. Norelco C-60 System $135.00. Ampeg B-18 Bass amplifier $169.00. Ampeg Gemini II $169.00. 16 position solid state 8 track recording and remix console system consisting of Melcor and Altec components. $4395.00 For additional information write or call **Wiegand Audio Laboratories, R.D. 3., Middleburg, Pa. 17842. Phone (717) 837-1444**

FERROGRAPH TAPE RECORDER, latest model 7½, 3¾, two-track stereo tape deck in sealed carton: $390.00. **(212) 463-1222.**

ONE STOP FOR ALL your professional audio requirements. Bottom line oriented. **F.T.C. Brewer Company, P.O. Box 8057, Pensacola, Florida 32505.**

AMERICA'S LARGEST SELECTION of new and used broadcast and recording equipment! Latest bulletins available. **The Maze Corporation, P.O. Box 6636, Birmingham, Ala. 35210.**

WHATEVER YOUR EQUIPMENT NEEDS —new or used—check us first. Trade your used equipment for new. Write for our complete listings. **Broadcast Equipment & Supply Co., Box 3141, Bristol, Tenn. 37620.**

REEL-TO-REEL cassette 2-track duplicator. Infonics master and one slave. Twelve copies per run: $3000.00. **Rodel Audio, 1028 33rd Street, N.W., Washington, D. C. 20007. (202) 338-0770.**

HAECO announces complete repair service and overhaul for all Westrex cutterheads. Conversions of 3D-II and older models to higher performance standards and reliability. Helium cooling systems and hi-temp coils can protect your investment. Repair insurance program available. Rapid service. Lower cost. **HAECO, 14110 Aetna, Van Nuys, California 91401.**

SOLID-STATE AUDIO PLUG-IN OCTAL (1" Dia. x 2" H) modules. Mic preamps, disc & tape preamp-equalizers, tape bias osc & record ampl power amps & power supplies. Send for free catalog and audio applications. **Opamp Labs., 172 So. Alta Vista Blvd., Los Angeles, California 90036.**

NEW YORK'S LEADING supplier of professional recording equipment and hi-fi stereo components. All major brands in stock. Call for quote—sales—service —leasing—trade-ins. **Martin Audio, 320 West 46th Street, New York, N.Y. 10036. Telephone: (212) 265-6470.**

TECHNICAL CONSULTATION—recording/broadcasting studios, and performing arts. Thirty audio lines—consoles, microphones, equalizers, monitoring systems. Call or write—quotations, specifications. Listings. **Frankford/Wayne Laboratories, 212 N. 12th, Philadelphia, Pa. 19107. (215) 561-1794.**

EMPLOYMENT

PROFESSIONAL RECORDING PERSONNEL SPECIALISTS. A service for employers and job seekers. Call today! **Smith's Personnel Service, 1457 Broadway, N.Y.C. 10036. Alayne Spertell 212 Wi 7-3806.**

ENGINEER/MIXER: M.A. degree. Experienced. $20/hour. **(212) 691-8754.**

PEOPLE, PLACES, HAPPENINGS

• **Fairchild Sound Equipment Corporation** has been acquired by **Robins Industries Corp. of College Point, N.Y.** Terms of the transaction were not disclosed. Fairchild operations will be headed by **George Alexandrovich,** vice president and **David Bain,** sales manager, who, along with other engineering, production and marketing people continue along with the company. In the announcement by **Herman D. Post,** president of Robins, it was stated that research, development and production will continue without pause. "We are confident that the corporate strength we can inject into Fairchild Sound Equipment will enhance its position in the market, and enable it to provide more customer service . . ."

George Alexandrovich's monthly column is absent this month because of the transfers and moves from Fairchild Sound's old quarters to new ones. He'll be back.

• Magnetic tape heads on data processing and video recording equipment could be made to function more effectively if there were a better understanding of exactly what causes them to wear. This is the hypothesis of researchers at **Battelle Memorial Institute** who recently proposed a study to investigate the wear phenomenon of magnetic tape heads—a problem which leads to lost data in computer storage and poor quality video reproductions. Battelle presented details of the proposed study in a meeting at its Columbus Laboratories to representatives of tape, tape head, instrumentation, recorder and computer manufacturers. The results of the study are expected to lead to practical recommendations for improved head and tape materials; new design criteria; and faster, more reliable methods of evaluating tape abrasivity.

• From **CCA Electronics Corp.'s** president, **Bernard Wise** we learn of the following appointments.

Robert Sidwell has been appointed to the position of the newly-created position of v-p broadcast equipment sales. He has been with CCA as sales manager since 1967 and prior to that owned and operated a radio station in Florida.

• **Robert Badger** becomes v-p broadcast activities. He comes to the parent company from **WABY,** CCA's broadcast subsidiary in Albany, N.Y.

Bruce Emonson, president of CCA's Canadian subsidary, **Caldwell A/V Equipment Company, Ltd.,** has been elected a corporate vice president of CCA. He had served as president of the subsidiary prior to its acquisition in 1970 and continues in that capacity.

Joseph J. Fox who joined CCA as controller several months ago has been appointed to the position of treasurer.

• **Irish Tape** has introduced a new line of chromium dioxide blank cassettes. Both 60 and 90 minute cassettes will be offered in the line, called the Irish 263 series. The tapes are specifically designed for machines with built-in bias switches, or properly equipped duplicators.

• **Richard Stover** has returned to **Superscope, Inc.** according to the announcement by **Joseph S. Tushinsky,** president. His new title will be general manager of the new Superscope education products division. Mr. Stover originally came to Superscope several years ago to help organize the tape duplicating division and its various departments for the creation of music on tape and custom tape duplicating. He had left Superscope about one year ago to join another tape duplicating firm.

• **Hewlett-Packard** has begun marketing instructional videotapes on electronic subjects for use by scientific and technical organizations, hospitals, medical schools, colleges, etc. One of the first offerings is called **Practical Transistors.** It is a nine hour, fifteen-tape series. Along with it, offerings in tutorial, operational, and maintenance subjects will be given. The tapes are available in Sony one-inch, Sony one-half inch (EIA-J compatible) and Ampex one-inch monochrome formats. Prices range from $65.00 to $150.00 depending on length and format.

• A recent announcement from **Collins Radio Company** tells of a stock purchase agreement with **North American Rockwell Corporation** which will make a 35 million dollar cash investment into Collins. As a result of the agreement, the Collins board will consist of thirteen members, seven designated by North American and six by Collins. **Arthur A. Collins** continues as president, principal executive officer, and director with **W. F. Rockwell, Jr.** (chairman and chief executive officer of North American) as chairman of the Collins board.

• **Robert E. Brockway,** Class of '46, Hofstra University, president, **CBS Electronic Video Recording division,** has been named this year's recipient of the tenth annual **George M. Estabrook Distinguished Service Award.** The award is based on service in alumni activity or achievement in chosen field or both. Mr. and Mrs. Brockway have both been members of the association's board of directors. Mrs. Brockway was also graduated from Hofstra. Mr. Brockway joined CBS in April 1968 as president of the then newly formed Electronic Video Recording division, after key posts with **Manhattan Cable Television** and **Sylvania Electric.**

Condenser microphones have long been known for their sound... and their cost, and their complexity. Now Electro-Voice introduces a series of genuine condenser microphones that provide sound embarrassingly close to the most expensive studio models, without the high cost and complexity.

A big problem with conventional condenser microphones has been the need for a high voltage power supply to polarize the diaphragm. E-V has eliminated it completely with its new *electret* condensers. We've found a way to permanently trap this voltage right on the surface of the diaphragm, thus doing away with the need for bulky, expensive power supplies.

How do the new Electro-Voice electrets sound? Response is clean, flat, and transparent, with very high output for full recording volume. It's just what you would expect from condenser microphones costing much more, and by far the best sound-per-dollar you've ever heard.

A simple FET circuit inside each E-V electret microphone matches both professional and home tape recorder inputs with equal quality. This low-noise, high-output circuit operates from a single "AA" penlite battery for as long as 1200 hours of use.

Choose either omnidirectional or Single-D cardioid types. The chart shows the prices, and some of the reasons for the difference in cost. Whichever model you choose will give you excellent transient response, high sensitivity, and uniform polar response. Our "second-generation" electret design offers vastly improved protection against extremes of humidity and temperature. And the ruggedness of E-V electret condensers is rivalled only by E-V dynamic models. All-in-all, new E-V condensers are a significant improvement over less sophisticated condenser microphones (electret or otherwise).

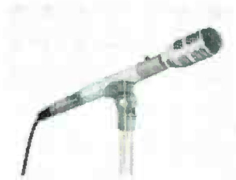
If your goal is to record natural sound, or natural music — try an E-V electret. Or in the PA field where condensers have never been sufficiently reliable — try an E-V electret. But don't tell your listeners how much you paid for your new microphones. They'll never believe you!

ELECTRO-VOICE, INC., Dept. 912BD
686 Cecil Street, Buchanan, Michigen 49107
In Europe: Electro-Voice, S.A., Lyss-Strasse 55, 2560 Nidau, Switzerland

MODEL 1710
Omnidirectional
80-13,000 Hz response
Unbalanced 150 ohm output
—50 dB output level*
$39.75

MODEL 1750
Single-D Cardioid
80-13,000 Hz response
Unbalanced 150 ohm output
—43 dB output level*
$45.00

MODEL 1711
Omnidirectional
60-15,000 Hz response
Balanced 150 ohm output
Professional cable connector
—50 dB output level*
$59.70



MODEL 1751
Single-D Cardioid
60-15,000 Hz response
Balanced 150 ohm output
Professional cable connector
—43 dB output level*
$75.00

*Output level ref. to 1 mw/10 dynes/cm². The smaller the number, the better. Prices shown are suggested retail.

Electro-Voice®
a **GULTON** subsidiary

Circle 12 on Reader Service Card