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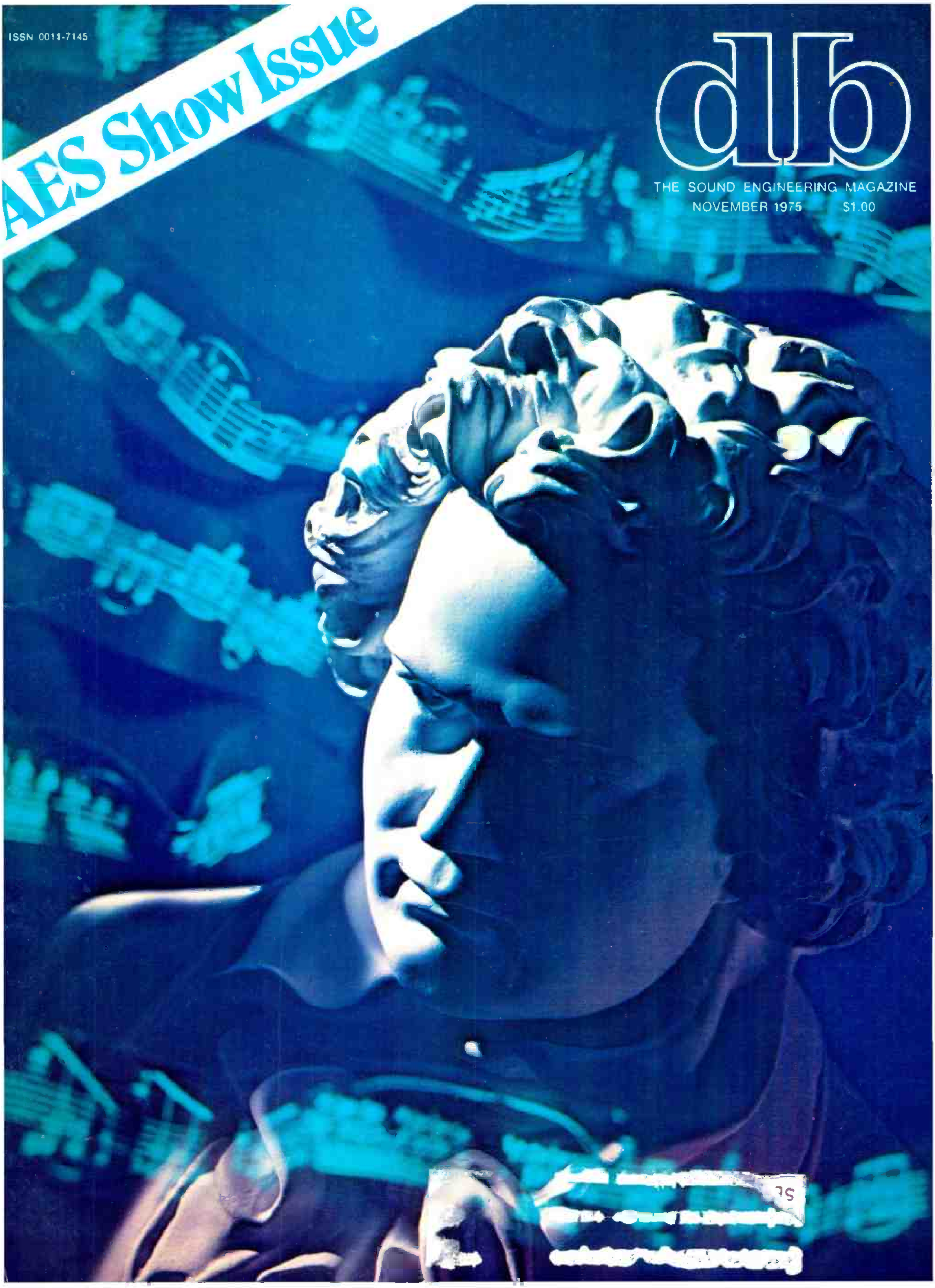
THE SOUND ENGINEERING MAGAZINE

NOVEMBER 1975 $1.00



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- Acoustical management of outdoor performances, to keep the audience and the neighbors happy, is revealed in Michael Rettinger's SOUND CONTROL BY BARRIERS.

- Editor Larry Zide has been travelling again, this time for a fascinating VISIT TO THE CBS TECHNOLOGY CENTER in Stamford, Connecticut.

- The columnists will be on board . . . we hope. December will also bring a complete index of articles and authors for the years 1974 and 1975.

about the cover

- Ludwig von Beethoven makes the big time with a psychedelic interpretation by photographer H. Armstrong Roberts.

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THE SOUND ENGINEERING MAGAZINE

NOVEMBER 1975, VOLUME 9, NUMBER 11

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

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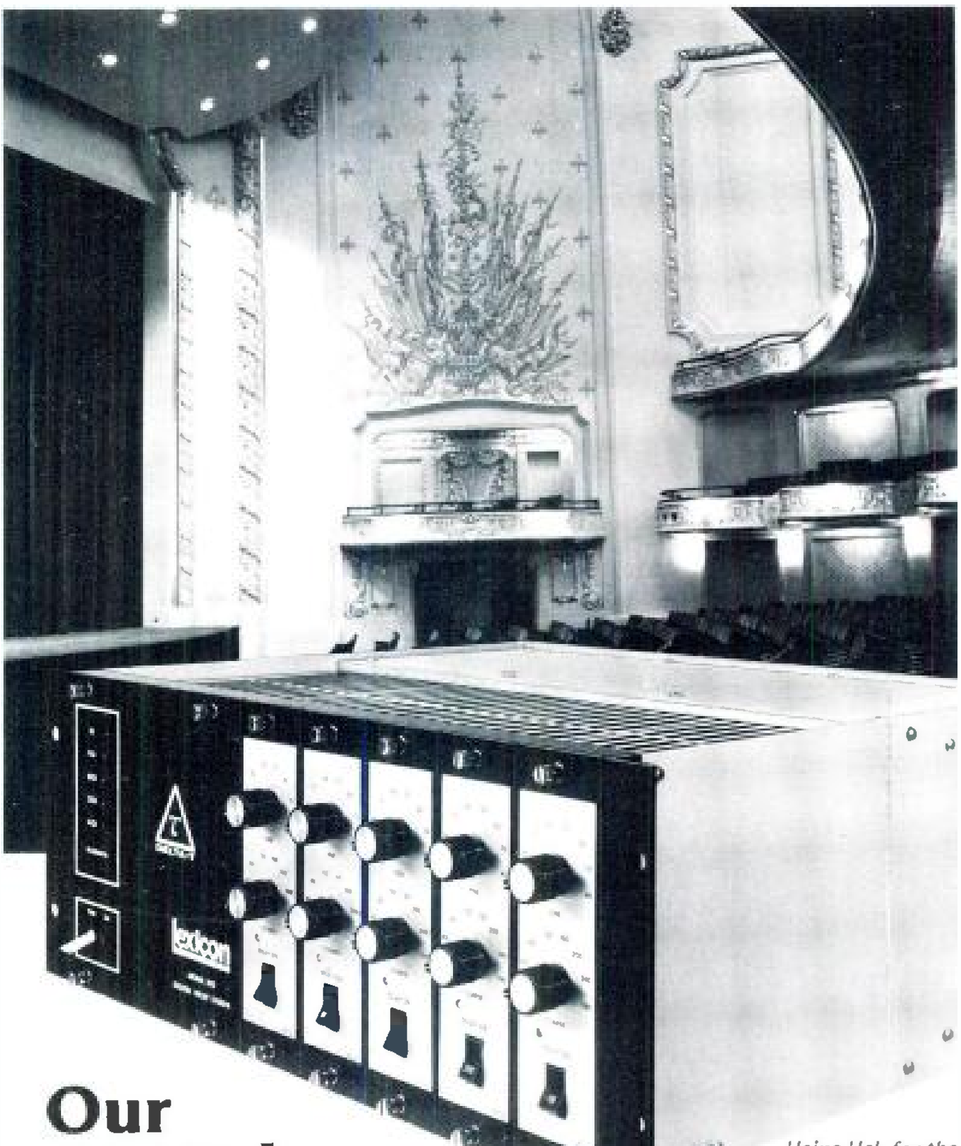

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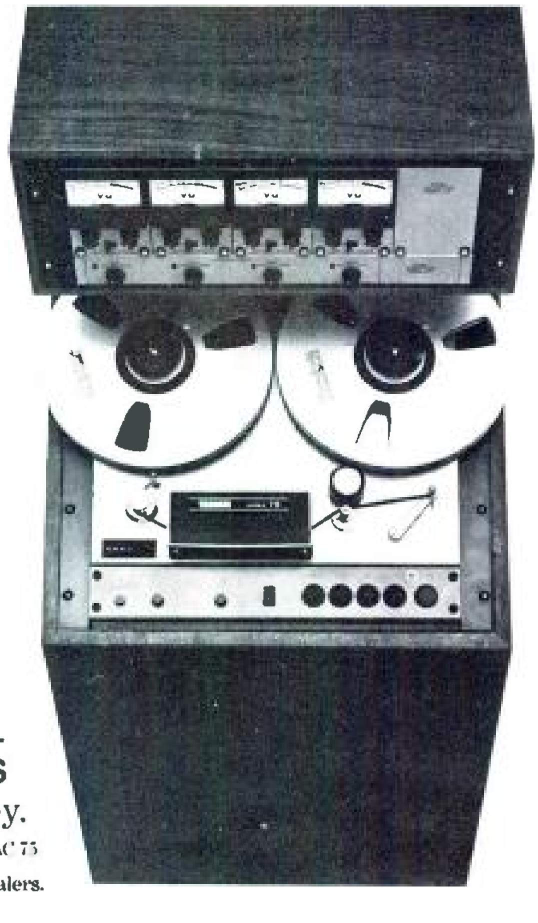

db letters

THE EDITOR:

Now you have gone and done it! What am I talking about? Well, it was the article entitled *Handy Black Boxes* by Don Davis. Now this is the kind of article that many of us would like to see more of. They are so useful. How about some articles on the equalization of buildings, both the engineering and the method of doing it? How about some articles that discuss the pros and cons of various makes of that type of equipment? While we are at it, I'd like to know about various types of equipment that are simple, inexpensive, easily made, and solve sometimes knotty problems one meets up with—not necessarily in the larger systems. I have in mind the ever-present problem of rf or a.c. interference in lines and similar problems. Practical articles are most useful.

Additionally, I liked the article about the f.e.t. mixer, which was in one of your past issues. In the same issue as the article by Mr. Davis was the one on *Electret Microphones* by Basil Lane, an interesting article, keeping us up to date on the latest in that particular field.

WILLIAM C. POLLARD
db-Video Systems
Manchaca, Texas

THE EDITOR:

In your August 1975 issue there was an article entitled "In the CD-4 Groove." As a relatively new reader, it took me several hours to figure out why the published photographs were confusing to me. The photos are printed in a a way that makes it appear that the pertinent information is carved into the sides of mountain ridges sticking up out of the surface plain. I had to turn everything upside down in order to be "In the CD-4 Groove." I wonder whether other readers suffered from the same illusion.

EUGENE F. KATONA
Cincinnati, Ohio

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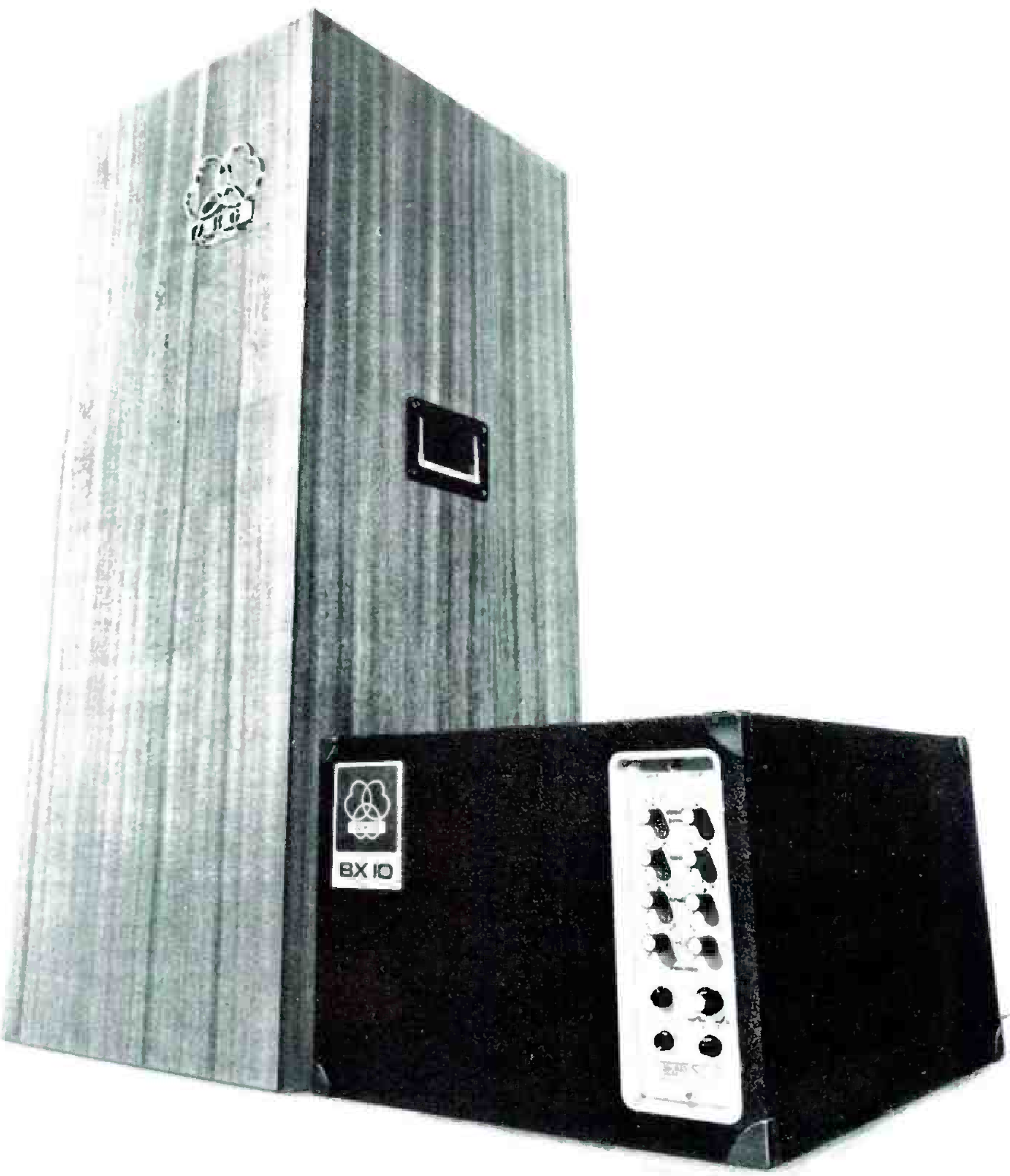

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FREE LITERATURE

ELECTRONIC KITS

The new catalog from this manufacturer lists their extensive line of electronic kits, including two new 5 MHz oscilloscopes. Mfr: Heathkit.

Circle No. 85 on R.S. Card.

MAGNETOSTRICTIVE ALLOYS

Nickel-iron and nickel-cobalt alloys in 0.0005 inch strip form are described in this bulletin. Mfr: Magnetics. *Circle 86 on R. S. Card.*

SPEAKERS

A variety of speakers, with numerout different applications, are described in this 10-page booklet. Mfr: CTS of Paducah, Inc.

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DIGITAL DELAY NOTES

A detailed application note dedescribes and diagrams various loudspeaker setups, with very specific instructions. Also included is a bibliography. 14 pages. Mfr: Lexicon, Inc.

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INVENTORS

Of special interest to the independent inventor is this booklet entitled "The Link Between the Inventor and Industry." It contains hints on how to market ideas to industry. Mfr: Lawrence Peska Associates, Inc.

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CERMET TRIMMERS

A two-page data sheet on model 63 cermet trimmers shows seven configurations and contains complete technical specifications. Mfr: Spectrol Electronics Corp.

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REPLACEMENT STYLI

A complete listing of replacement styli for this manufacturer's cartridges, including keying information for matching the proper stylus to the appropriate cartridge is available in this 15-page booklet. Mfr: Shure Bros.

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MICROWAVE AMPLIFIERS

Technical information regarding their solid state miniaturized amplifiers are offered in this brochure, designated IFA-2102. Mfr: RHG Electtronics Laboratory, Inc.

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AMPKLIP SEMICONDUCTOR FUSES

A lineup on their SF 60C series 600 volt ampklip semiconductor fuses is featured in data sheet No. PD-8.005. Mfr: International Rectifier.

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DA-6RS/E	Rack mount. 1 in/6 out stereo or 2 in/12 out mono.	$229
DA-16BR/E	Rack mount. 1 in/8 out stereo or 2 in/16 out mono. Individual output level controls, selectable metering & headphone monitoring.	$287
DA-2080/E	Rack mount main frame with protected power supply, metering & headphone monitor. Will accept up to 10 slide in modules. Each module has 2 inputs & 8 outputs. Individual output level controls & selectable meter switch. Up to 20 in/80 out.	
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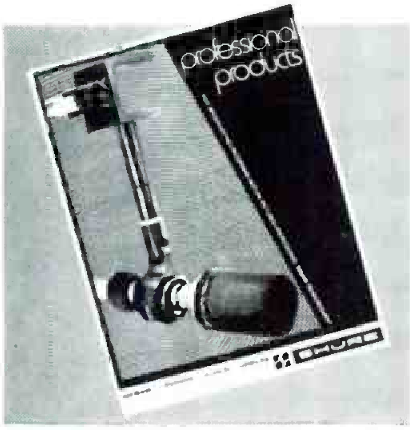
Studio equipment home shopping guide



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db theory&practice

NORMAN H. CROWHURST

• At the Brigham Young University Workshop this year, one of the special pleasures was the opportunity to meet Bill Putnam again. Bill had been responsible for introducing me to BYU, but we had not met in many years. When we got together to discuss old times, as inevitably happens on such occasions, we discovered that we first met back in 1959 when both of us received the award of a Fellowship in the Audio Engineering Society at the same banquet.

But we have much more than that in common. At this year's workshop, Bill opened his remarks by reading from something he wrote in 1962, which he followed with a 1969 update. This was a recital of the qualifications required to be a successful sound mixer. I will quote them here, verbatim.

1962 ORIGINAL

The qualifications of a successful sound mixer represent a unique and rare combination of skills. He must be adequately technically oriented to understand and evaluate the performance of the various electronic and acoustical devices with which he works. He must have sufficient musical aptitude to interpret the wishes of the arranger-conductor; he must be creatively artistic, imaginative, have a flair for showmanship, willing to try the impossible, and have the ambidexterity of an octopus. He must have the unique talent of being able to communicate with artists and directors at any artistic level, be able to perform his functions deftly under extreme pressure and. above all, must have the patience of Job.

1969 UPDATE

His countenance, appearance, wardrobe, vernacular and externalized image must be appropriately compatible with the environment and clime established by the group of performers with whom he's involved. He must be contemporarily knowledgeable and articulate regarding the latest pot pourri *of narrative dissertae of the underground periodical community. He must be subtly esoteric and effuse an aura of non-committal aplomb while espousing an extemporaneous and original glossary of quasi-technical linguistic exercises to instill the confidence of those involved. He must have a threshold of pain in excess of130 decibels, be expertly capable of "doing his thing" in light levels below 0.02 lumens, while maintaining his normal respiratory functions in an atmosphere of less than one part oxygen to four parts nitrogen. and wonder why Job was such a square.*

APPLICABLE TO 1975-76?

Bill questioned whether it is not time he produced another update. While little of his 1969 statement is obsolete, there are new features on the scene that should be included. Just before Bill came to Provo, he had recorded a new album for Bing Crosby. When Bing had asked Bill to do this, Bill responded, "If you can still sing, I can still mix."

The essence of Bill's remarks, speaking as a man who has been responsible for the mixing of more records than any other man of our generation, was to stress the importance of the kind of nitty-gritty things the workshop had been covering for its participants. Being able to produce something that will be in demand in a competitive world will always require creativity. never be subject to setting down by numbers, like a recipe book.

OF DOERS AND TALKERS

This tied in well with the experience earlier in the workshop that I told about last month—concerning the people who wanted formulas for everything. Implicit in Bill's listing of qualifications is the ability to do, rather than to talk about it.

When a musician or an artist approaches the mixer to explain what he wants, the mixer must wear his own artistic hat, yet understand what it is the performer wants. He makes the translation of how to achieve it in his own head without giving the performer a dissertation on the technicalities of the subject. And he just does it!

But to produce what is needed effectively, he must know all that technical stuff. If all he knows are the appropriate formulas—even if he can keep them straight and use the right formulas in the right places—he will not have the creativity needed to in-

terpret the performer's needs.

OF PERFORMERS AND TEACHERS

As we have commented before, the sizeable element of truth in the adage, "Those who can, do; those who can't, teach" is responsible for many of our problems in education today. This points up a difference between working for a performer and working for educational purposes, in many instances at least.

Unfortunately, the tendency is all too often almost exactly the reverse of what it ought to be. When we work with a true professional—a performer—he or she knows what he wants, he knows his work, and assume you know yours, so he leaves the technical outworking to you. Perhaps too much so, at times, when a little communication might produce better results.

But when audio/visual experts work with most educators, they often adopt the attitude that they know precisely what is needed and will often tell you how to do your job, even if they know nothing about it.

Performers are different from teachers in another way—the basic role they fill, or should fill.

Any performer has primary control of the performance—and should have. A performer will modify his performance from time to time to adapt to what pleases his public. But what he does is, and should be, his decision. The public is there to enjoy what he does, because he does it.

In education, the purpose is, or should be, quite different. A teacher should not be a performer, in this sense, although many of them act as if they are. A teacher's concern should be that the *students* learn. That should be the primary control, not the aggrandizement of the teacher.

Too often, teachers are frustrated would-be performers, as well as dropouts from whatever subject it is they aspire to teach. Such a situation can call for a great deal of tact on your part when offering mediated a/v materials, perhaps even more than Bill suggested in his "qualifications of a mixer" in order to get across in finished result something far more than a good rendition of the teacher talking to his class!

It is useless to tell a teacher something like, "Your objective should be to have every student in your class learn," especially if the teacher's attitude is the average, "I teach it the best way I know how, so if they don't learn it, it isn't my fault." Translated into mediated instruction, this be-

comes, "That is the best mediated instruction possible. So any student who cannot learn from it must be very dumb!"

With that as the teacher's philosophy, you are not going to change it by telling the teacher his or her philosophy is wrong. Presumably the teacher does expect some students, at least, to learn from the material being put together. So you must work from there. Put yourself in the place of the student. Could *you* learn from the lesson? Would you have difficulty? If so, why? Are you average? Better than average? Less than average? Should you be able to learn the material, or isn't it for the likes of you?

In other words, play just a little bit dumb, instead of acting authoritarian. One authoritarian in the room is usually enough, if not more than enough! Use leading questions, like those in the previous paragraph, to get the teacher, hopefully, to question whether the instruction being put together really is adequate.

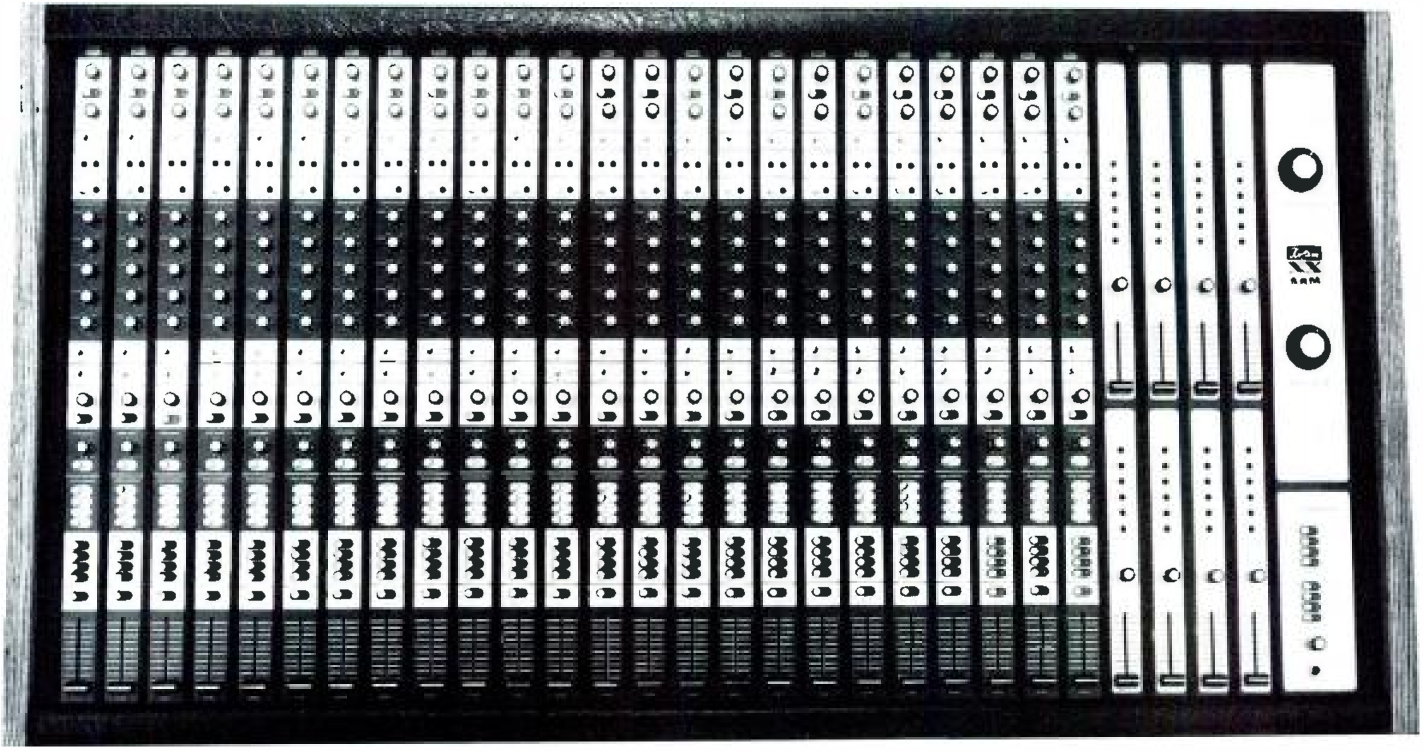
You have to be careful to simulate just the right amount of "dumbness." If you act too dumb, the teacher is likely to say, "Just mediate what I give you, and don't bother trying to understand it." Act bright enough so that teacher will think that you ought to be able to get it, but are having troubles with it.

This approach may not be enough to bring about the desired change. Many teachers don't have the creativity to view the learning situation from the perspective of the student. So do some puzzling through on your own. Try to think of a better way to convey the information, if it seems foggy to you the way it is. Then ask if some approach you think up wouldn't be a way to explain it. Don't suggest it as if you think you've a better way than teacher's, but just as if you want to make sure you have the right idea—would that be another way to explain it?

You might get a surprise. More than one instructor we have worked with has been turned on by this kind of attitude. The fact that you show yourself interested is a help. And once you have gotten more than one method into the picture, many teachers will open up to discuss the merits of different approaches, and you may finish up working together on something that's better than the ideas either of you started with.

WHAT PART DOES AUDIO PLAY?

In the world of performance/entertainment, audio can fill different roles,



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and part of your job is to see that it fills the right role on the job for which you are engaged. In radio, or sound recording, audio is everything. In t.v. or motion pictures, its role may change. Sometimes it is merely supporting. Sometimes the audio is the main thing, and the picture is supporting. And sometimes the two blend together so you can hardly tell which is which.

That is the kind of skill about which Bill's qualifications were concerned—being able to do just what is needed.

What role does audio play in educational material? Most often, probably, it will fill the role of an instructor, leading, telling, hinting. But it will also carry sounds that are part of the instruction. If you are using any kind of audio visual—printed page, pictures, projected or even in motion—you need to correlate audio and visual for maximum effect, according to what you are trying to do.

It helps, in building educational material and coordinating its parts, to think about the degree of initiative it expects of the student. Good, creative educational material has the student doing virtually all of his own thinking. The material doesn't tell him anything, just gives him hints and other guidance to lead him to think it all through for himself, also making sure that if he cannot quite make it on his own, the material can come to his rescue.

Today, few students are ready for that, straight off. They've been too long conditioned to the "instructional" type of teaching/learning, where they are told everything, requiring very little initiative. So you need to *build up* to good material, or you will leave the mass of poor students behind.

One more thing. If you have a good relationship with your subject-matter expert, this comes easily. If not, it requires a great deal of diplomacy.

Too much of today's education is shrouded in mysticism. Do what teacher tells you, and presto, magic, eventually, you know how to do something. So, you may find yourself able to solve equations, for example, without having the vaguest idea what an equation is!

My little book, "Taking the Mysticism from Mathematics" addresses this situation relative to mathematics. But it is a far more universal problem. Since most teachers seem committed to perpetuating it, changing it must come from outside the teaching profession.

One device is taking the student into your confidence. This can have a great many advantages, beside making learning ever so much easier. Probably the biggest plus is that the student himself knows whether he is learning or not. So he can pace himself, correct himself, and thus continuously become a better learner. With well designed educational material, a student should never have to ask a teacher "How am I doing?" He should know.

Of course, that doesn't mean that sometimes he would not like teacher's evaluation as well. But he should not be almost totally dependent on that, as is the situation most of the time today.

Another very important advantage of taking the student into your confidence, is that measurement becomes much easier. When taking a test yourself, how often have you asked, "Now I wonder what's the point of that question?" That is a sure indication that you were not taken into the confidence of the material. But if you know the point of the question, and still cannot answer it, you know exactly what your problem is.

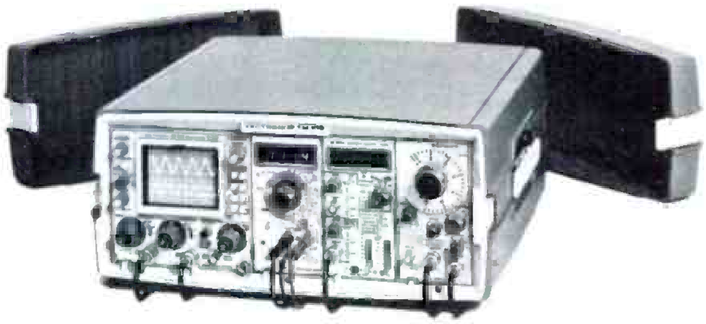
That is the way your material should make it clear for the student. ■

TEKTRONIX ANNOUNCES

A new concept in portable instrumentation

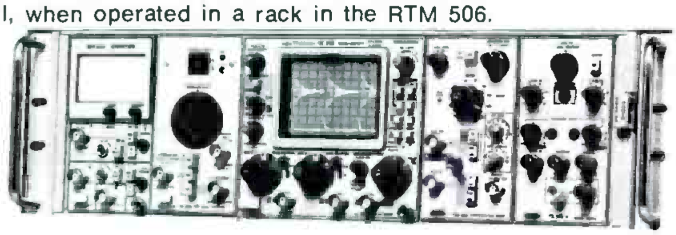
The TEKTRONIX TM 515 Traveler Mainframe looks like fashionable flight luggage, compact and easy to carry, or slide under an aircraft seat. In reality, it's a five-compartment power module/mainframe that provides power and interface connections for TM 500 plug-in modular instrumentation. Plug in the new (two-wide) SC 502 15-MHz dual-channel oscilloscope, and you have the beginnings of a powerful take-along instrumentation system.

You can optimize a TM 500 system to your needs by selecting from more than 30 plug-in modular instruments. With the TM 515 Traveler Mainframe and SC 502 Oscilloscope as a nucleus, select from DMM's, counters, generators, power supplies, signal processors, and even blank plug-ins for your "home-built" circuits. Intended applications include areas from digital field service to medical, from audio/communicatons to on-site industrial controls maintenance.

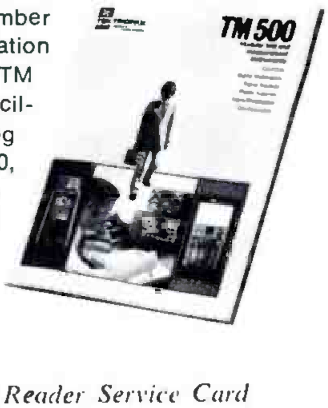


The SC 502 is Tektronix quality, featuring clean triggering characteristics, delay line input, trigger view, trigger holdoff, 1 mV sensitivity, and the capability of working through the rear interface circuit board with other TM 500 instruments. It features a specially brilliant crt designed and built by Tektronix for use in areas of high ambient light. Include a DD 501 Digital Delay alongside the SC 502 and gain the capability of delay-by-events—you can then obtain stable digital displays from electromechanical sources like disc drives that would otherwise be too jittery for accurate viewing on any conventional oscilloscope. Include the DC 505A Universal Counter and DM 502 Digital Multimeter to complete your TM 515 package, and discover the benefits of simultaneous counter and DMM capability with trigger level readout at the touch of a push button.

The TM 500 concept lets you take along on field servicing trips the same instruments you use in the lab or for production testing, thereby enabling you to maintain the same standards on the "outside". The SC 502 Oscilloscope, for example, may be used as a bench instrument in any multiple-compartment TM 500 mainframe, and it offers unique systems capabilities, as well, when operated in a rack in the RTM 506.



Contact your local Tektronix Field Engineer or circle the appropriate reader service number for a demonstration of TM 500 instrumentation or additional technical information on the TM 515 Traveler Mainframe and SC 502 Oscilloscope. For an up-to-date TM 500 Catalog write to Tektronix, Inc., P. O. Box 500, Beaverton, Oregon 97077. In Europe write Tektronix Limited, P. O. Box 36, St. Peter Port, Guernsey, Channel Islands.



TEKTRONIX®

committed to technical excellence

Circle 30 on Reader Service Card

Underneath the blankets: 36 inputs, 32 equalizers, 16 outputs, 6 limiters, 3 patch bays, and some other stuff.

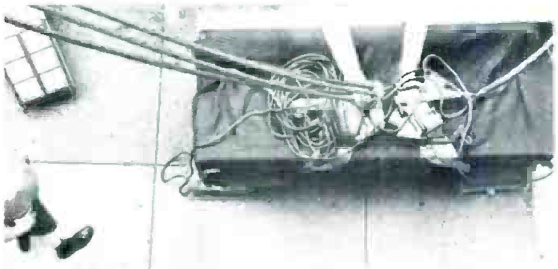
Up Goes the Console

Disregard the title of the column for this month: there is no Sync Track column this month. What with finishing the last few pages of my first-born book (an excerpted chapter is elsewhere in this issue) there just hasn't been time.

As if that wasn't enough, there were other distractions such as a new baby at the Institute of Audio Research. Junior weighs about 1800 pounds and the delivery took a bit longer than expected.

The new baby is a 36-in/16-out console that proved too big for the elevator and too heavy for the stairs, so it had to be "flown" in courtesy of Conventry Movers, midwives at many a recording studio delivery.

The pictures tell the story. Did Junior survive the delivery? Tune in next month. ■



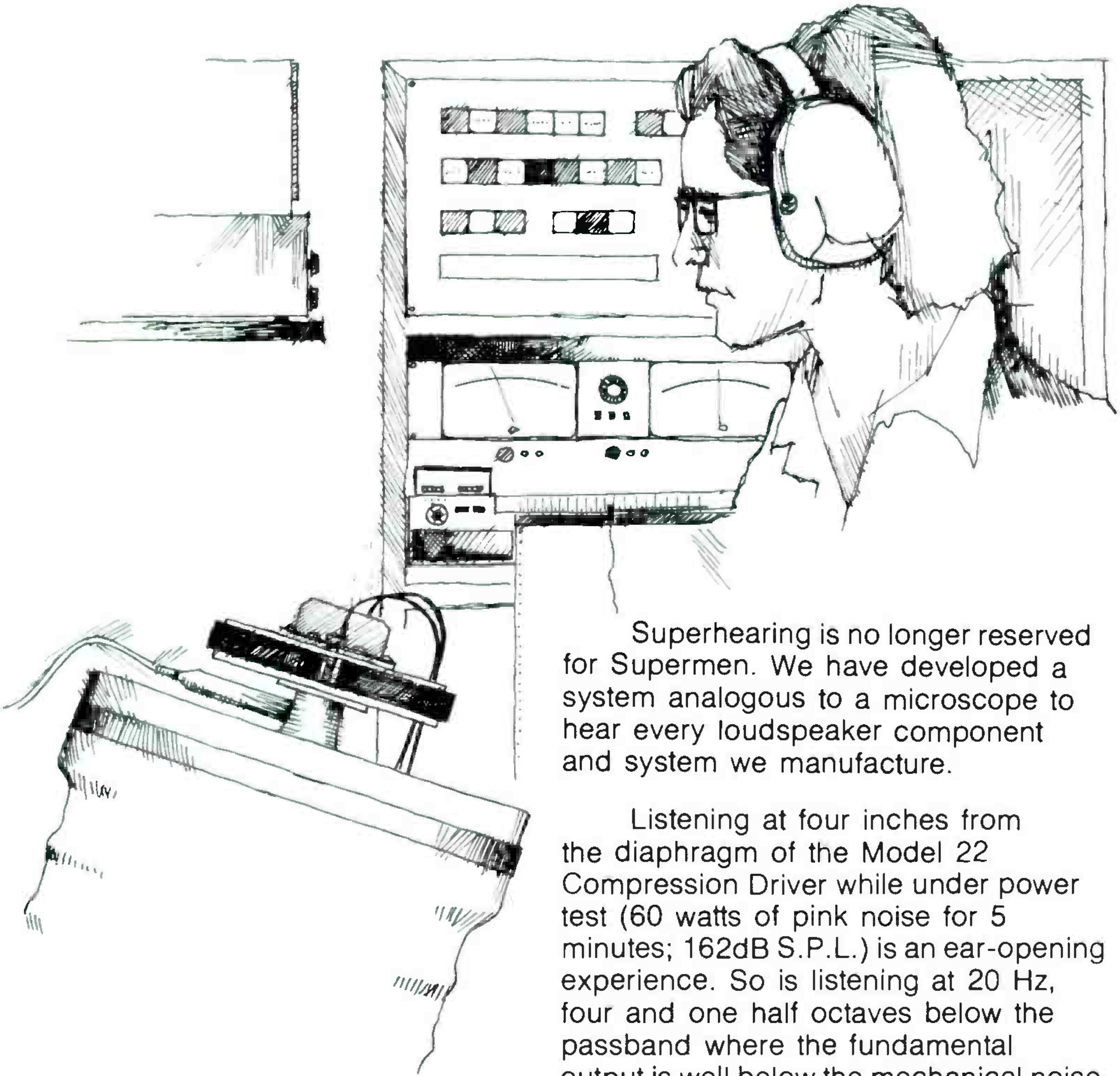
Relax, we've never lost one yet!

We've cleared the supermarket okay, keep going.

One small step for the console; one big ulcer for the Institute of Audio Research.

A little hands-on experience in the recording industry as students learn how to recalibrate a wall.

Going in.
Some students just can't wait.



db sound with images

MARTIN DICKSTEIN

● Just five years ago, almost to the month, **db** carried a three-part story on the then latest development in the video field—the video disc. It wasn't until this year that the device finally made its appearance on the market, but only in Europe so far. It just so happened, that on two consecutive days in March of this year, two new disc developments were announced. These, supposedly, will be ready for the market in 1976 or 1977, although there has been some comment that it might even be 1978. Anyway, this seems to be the year the public gets its chance to decide which method of producing video they prefer. Let's take a quick look, with only some detail, at the differences, since not one process is compatible with either of the other two.

The whole thing started back in 1927 with the first attempts to put pictures on discs. At 78 rpm, with a bandwidth of 5 kHz, the invention was able to reproduce 30 frames a second of a 30-line transmission. It wasn't until 1965, however, that further development took place, and five years after that, in 1970, Teldec came out. It was only one of several which were in the experimental stage, but this one, after a great deal of difficulty, finally made it.

TELDEC

The principal of operation is similar to that of an audio record player, but with some major differences. The disc, made of a very flexible polyvinyl material, is placed on the turntable, but instead of there being just a single center hub, there are three pins to hold the record more firmly in place. With a rotational speed of 1800 rpm, it was found necessary to lift the disc slightly off the table itself. An air pressure was built up under the disc to lift it slightly while spinning.

The arm, actually two parallel rods, extends across the radius of the disc and supports a sliding pickup head. The cartridge moves in toward the center by sliding along the rods a distance of 0.008mm, the width of each groove, with each revolution. Although the grooves contain the video (and audio) information, the stylus does not ride within the cut but along the hills-and-dales and (by pressure) reads the images and sound for reproduction on a t.v. set. The specially designed stylus and the extremely thin, closely packed grooves, combined with the high speed of rotation, results in a 3.0 MHz band width, as required for video reproduction. (Compare this to the standard 15 kHz needed for audio only. With audio only on the video disc, the frequency response would be somewhere around 70,000 or 80,000 Hz with a multiple channel capability.)

Although there is a limited time on each disc—just 15 minutes for a 12 in. disc—and the record is cut on only one side, the records can be reproduced (stamped, not pressed) very quickly, in quantity, inexpensively, and they are practically indestructible. The records are capable of being played more than 1,000 times with no noticeable signal deterioration, and can be handled with no fear of finger marks hurting the quality.

The obvious feature of quick cueing up as with an audio disc, the freeze-frame capability and quick replay of any desired portion of the record, as well as the simplicity of operation make this a strong contender for the home market where video tape has not seemed to be able to make strong inroads.

SELECTAVISION

There were other contenders in the field at the time Teldec first demonstrated in this country, but for one reason or another they faded out and went to black (as they say in television). Remember Cartrivision, which was a tape system? Then there was CBS' EVR which was a film system. Maybe it's better not to recall these, but there was another one called SelectaVision by RCA which was also around then, only not in the same form as today.

The RCA system is also a grooved disc/arm-and-stylus method of video reproduction, but very different from the Teldec operation. The record master is first cut mechanically with a spiral groove similar to that on an audio disc, but with much greater concentration. The record is aluminum with a copper coating and another coating sensitive to an electron beam. A modified scanning electron microscope is then used to aim a beam modulated by the video signal onto the sensitive coating in the grooves. The result is a series of transverse slots varying in width and spacing at the bottom of the previously cut groove. The recording is done in a vacuum to prevent scattering of the electrons during cutting.

After the master is made, the vinyl records are pressed, covered with a metal coating, styrene, and a thin layer of oil. The first two are for the purpose of enhancing the variations in capacitance, the latter for lubrication to preserve the life of the record and the stylus.

During playback, the sapphire stylus rides in the groove but does not vibrate as in an audio pickup. Instead, it is made to read capacitance variations in the indentations of the grooves. These are then converted to video signals and fed to the usual t.v. screen. Audio is included both in the recording and playback. Chrominance, luminance and audio are recorded in a composite f.m. signal; bandwidth is 3 MHz. (As many as 125,000 records can be made from one master, it is estimated.)

The record spins at 450 rpm, in contrast to the 1800 by Teldec. This means that four t.v. frames per revolution have to be reproduced (compared to one in the other process) and this prevents the capability of stop-action or slow motion that other systems have. The slower speed, however, makes for a simpler mechanism and the automatic tracking of the stylus in the groove eliminates the necessity for complex control systems. Automatic drop-out compensation is included in the system, and an "arm-stretcher" (no kidding) is incorporated to control the relative speed of the record and stylus to account for variations in the event of uneven disc rotation. The turntable itself is run by a synchronous motor locked to the power line.

The record, 12 in. in diameter and 0.07 in. thick, which will contain 30 minutes of video in more than 5,500 grooves on each side, has a video signal-to-noise ratio of better than 40 dB, and an audio bandwidth of 15 kHz. More than 500 plays are considered possible with no loss of signal, and the stylus is expected to last between 300 and 500 hours. Replacement of the inexpensive cartridge is supposed to be almost as simple as that of the home audio unit. It is expected that this disc system—closest in design to the present hi-fi turntable—because of the low cost of the player and the discs, the ease of handling, the ruggedness of the records, and the greater simplicity and slower speed than that of previous versions, will prove to have strong inducements for the consumer.

MCA-PHILIPS

The other very strong contender in the home video field represents a combination of designs developed by MCA and Philips. Both companies came out

sound with images (cont.)

with units which were compatible, so they combined their talents and have developed an optical video-disc system. Here the disc turns at 1,800 rpm, but instead of its having grooves as do the other records, the video signal is impressed in minute indentations. These individual tiny marks, made by a laser beam, carry both video and audio information. The width of the indentation is about 1/25,000 of an inch and varies in length from 0.8 to 2.5 microns. The depth is about 0.1 microns. (A micron is a millionth of a meter, or 1/25,400 of an inch.) Although there are no distinct tracks as such, the indentations do spiral; the spacing is abouttwo microns between the marks.

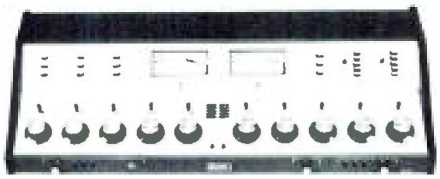
One difference is evident in the playback, unique to the MCA-Philips unit. The other video-discs use arm/stylus pickups, but in this device an optical reader uses a laser (blue light) to read the information. The laser, with mirrors, is mounted internally in the player, and reads the bottom side of the disc rather than the top as do the others. Accuracy of reading is controlled by a servo mechanism and a split beam to keep the main beam focused at the center of the marks. The beam bouncing off the indentation is read by a light-sensitive cell and reconverted to video and audio information. (The 1,800 rpm speed permits one frame per revolution. This means that by holding the reflecting mirror stationary, a stop-action image is possible. Similarly, it is also possible to crawl forward, watch in reverse, or scan quickly through the disc to the desired portion without any delay.)

The cost of the player and discs, which will play up to an hour, will be close to that of the RCA units. The records will be coated so that handling will not mar the quality of their reproduction. With the MCA-Philips system, the records are literally longer lasting because there is absolutely no wear at all during play since only a beam of light hits the surface. The laser is said to last well over a thousand hours and can be replaced easily.

It is widely agreed that there is a place for a video disc system for the home. Whether there will be a market for more than one, none compatible with any other, is something else again. Since Americans spend nearly two billion dollars a year for audio records (sound only), what will the video-plus-sound record bring? The year of the bicentennial may show-and-tell. ■

db new products & services

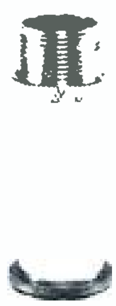
MONO CONSOLE



• Capability for mixing 28 sources (10 microphone, 18 line level) through 10 mixers is present in model 3310 monaural console. Switching is noiseless d.c. controlled. All mixers have remote turn on and can be activated by an audio or video source. A bridging input for Mixer 8 accommodates up to 5 tape cartridge playbacks without interaction. The 10 microphone level inputs appear at Mixers 1-4; line level inputs in Mixers 5-10 include two mixers which are switch selectable for five sources each. Audio from either bus can be sent back down any of the five remote lines or can assign the selected remote line to a particular mixer. Other features include 5 watt intercom, precision step attenuators with cue detent, and full dual channel operation with identical program and audition metered busses.

Mfr: Sparta Electronics Corp. (Cetec)
Price: $2,600.
Circle 50 on Reader Service Card

SNAP-ON MIC STAND ACCESSORY



• Flexibility in placing mics in different situations is the aim of US01 snap-on, snap-off microphone stand accessory. The two piece unit consisting of a male and female attachment eliminates the need for threading and unthreading mic holders and baby boom attachments from floor stands.

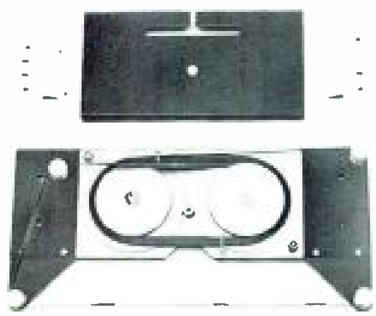
Mfr: University Sound
Circle 51 on Reader Service Card

AUDIO DUPLICATING HEADS

• Improved core design for extended bandwidth, reduced crosstalk and reduced bias and signal power requirements are claimed for a new line of glass-bonded ferrite duplicating heads. The heads are available in 8-track cartridge and 4-track cassette formats for direct replacement of metal or ferrite heads used on Cetec (Gauss), Electrosound and Ampex high speed duplicating systems. Precision grinding eliminates the need for mechanical azimuth and tilt adjustments. The heads feature a newly designed roller guide which is said to improve flutter characteristics and help maintain the proper tape wrap angle across the heads.

Mfr: Saki Magnetics, Inc.
Circle 52 on Reader Service Card

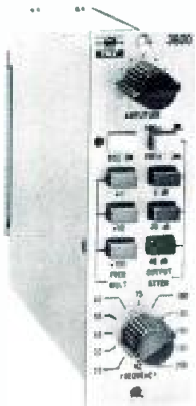
DELAY/ECHO ACCESSORY



• Designed for use with the Revox A77 tape deck, this delay/echo unit incorporates an endless loop cassette that can be attached to the deck by removing the deck plate and reel turntables and affixing a cassette mounting plate. The device can be used for automatic message repeating continuous short-term program monitoring, time delay of programmable machinery, and tape echo. In the echo application, the variable speed control unit can be used to extend the range of available time delay.

Mfr: Revox Corp.
Price: (Including tape) $187.
Circle 53 on Reader Service Card

PRECISION OSCILLATOR



● Performance testing of audio equipment accepting input levels from −80 dBm to +18 dBm can be achieved with model 3600 oscillator. The unit is equipped with its own output transformer and attenuator for normal testing and an unbalanced low impedance output for use in slate/tone systems. The unit features stable output level of ± 0.25 dB, 33 calibrated frequencies from 20 Hz to 20 kHz, selectable output impedance of either 600 or 150 ohms, transformer coupled output and calibrated output attenuator.
Mfr: Modular Audio Products
Circle 54 on Reader Service Card

DOLBY F.M. COMPENSATOR

● F.m. time-constant compensation, an important adjunct to Dolby f.m. broadcasts, is provided by model 622P1. The new model compensates for the reduced transmitter pre-emphasis in Dolby f.m. use, offering somewhat more flexibility than the model produced previously by the same manufacturer. Model 622P1 is a 2-channel device which can be used for receiving Dolbyized f.m. and playing Dolby recorded tapes, as well as operating when connected in-line for listening to regular f.m. broadcasts. Dolby noise reduction circuitry may be in a free-standing unit or included as part of a tape recorder. The unit contains no amplification, power supply or power lead. It can be hooked up with two processor noise reduction units simply to decode Dolby or in two alternate ways which provide recording without changing hook up.
Mfr: Switchcraft
Price: $24.95.
Circle 55 on Reader Service Card

MODULAR CONSOLE

● A unit which eliminates the need for jury-rigged consoles to handle 8-track recording and can be expanded to full 16-track capability is a practical choice for the budget setup. Grandson II, model 110-8 is a free standing console expandable to 24 mixing positions—48 inputs—in 50 in. width, applicable to recording/re-mixing/on-air control. It offers complete metering, two echo send/receive channels, talkback communications, separate control room and studio monitoring, either 8- or 16-channel monitor matrix, test oscillator, simultaneous stereo and 8 channel outputs, two independent headphone cue fold-back circuits, and a full line of matching accessories, including a plug-in patch bay. There are 8 program output channels and 8 vu meters. Monitoring functions and muting circuitry are TTL logic controlled and may be individually programmed. Solo function with l.e.d. indicator on each input may be used without interrupting a program.
Mfr: Auditronics, Inc.
Circle 56 on Reader Service Card

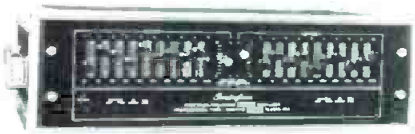
STUDIO CONSOLE

● A system called MIXLOG is built into console model Eclipse, which logs settings of faders, echo sends and pan pots for future reference. The unit is equipped with 20 inputs and 16 output-mix channels, plus quad, stereo, and mono mixdown. Other functions include four echo and two cue channels. The patching system is provided with a custom-fitted vu screen monitor, closed circuit t.v., tape machine remotes and digital timer. The console includes extensive graphic equalization on all inputs, assignable group submasters, presettable input muting and discrete operational amplifier circuits.
Mfr: Sphere Electronics
Circle 57 on Reader Service Card

SINE/SQUARE WAVE GENERATOR

• LAG-26 generator produces a fast rising square wave for testing transient response and a low distortion sine wave. It has a sine wave output range of 20 Hz-200 kHz at zero to 5V rms with ± 1dB flatness and distortion less than 0.5 percent below 20 kHz. The square wave output is 20 Hz to 20 kHz in the zero to 10V p-p voltage range with 0.5 μs rise time. The unit synchronizes signals from an external source and has a calibration accuracy of ±3 percent. The output impedance is 600Ω unbalanced and the frequency range is in 4-decade bands.
Mfr: Leader Instruments Corp.
Price: $139.95.
Circle 58 on Reader Service Card

TWIN-GRAPHIC EQUALIZER

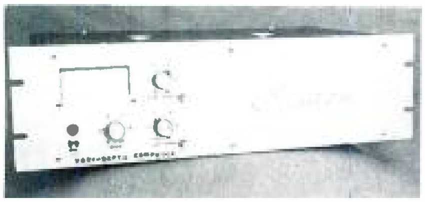


• Two completely independent ten-octave equalizers with separate equalization panels are contained in the TG2209-600. The unit handles tape-to-disc transfers, radio and t.v. production, p.a. feedback suppression, environmental equalization, and sound reinforcement. Front panel pushbuttons select either equalized or unequalized output, low and/or high shelving, and zero-gain lights on or off. There are separate terminations for input and output of sections A and B. Four l.e.d.s provide visual front panel display for balancing input to output signal ratios. The equalization panels have ± 12 dB boost and cut provided individually for each octave. Separate equalized signal zero-gain controls are used for each channel, enabling exact balancing of input to output with a +6dB and −12dB range. L.e.d.s are used in conjunction with the zero gain level controls.
Mfr: Soundcraftsmen
Price: $550.00
Circle 59 on Reader Service Card

REEL DEGAUSSER

• Model R24024 demagnetizes reels up to 1 in. wide and 17 in. in diameter. Operating on 115V a.c., 50-60 Hz, 18 amps, the unit is capable of degaussing 100 reels of ½ in. tape in approximately 15 to 30 minutes. One-inch reels are handled by doing one side and then flipping over the reel to repeat the erasing operation through the action of turning on the main power switch and slowly rotating the reel on the spindle for several revolutions. The device has a forced air cooling system; it may be operated for 40 minutes continuously before requiring a cooling off period. An automatic overheat phase is indicated by a panel light; during this phase the magnetic field is automatically shut off but a cooling fan continues. Included are a three-conductor power cord, fuses, and pilot light.
Mfr: Robins Industries Corp.
Price: $240.00.
Circle 60 on Reader Service Card

VARI-DEPTH COMPUTER



• Designed to operate in conjunction with the manufacturer's vari-pitch computer, for disc-mastering, the new vari-depth unit utilizes state-of-the-art low power cmos logic elements to digitize the incoming audio and transfers this information to the master disc. The operator may program the system to deepen on lateral as well as on vertical information. This feature, when used with the vari-pitch system, permits high level cutting at nominal pitches above 500 g.p.i. The system can be programmed for automatic deepening during lead-in, banding, expand and finish operations. The complete system can be installed quickly on any Scully or Neumann lathe and is compatible with all Westrex, Neumann and HAECO solid state systems.

Mfr: Capps & Co. Inc.

Circle 61 on Reader Service Card

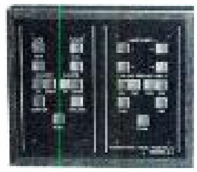
CABLE BOX

• An ingenious arrangement of connectors eliminates running cables from a mixing console to the stage in live performance concerts. The multi-connection cable panel has 15 female and 5 male cannon XLR connectors, two of which are wired in parallel. From this panel placed on the stage, a single 200 ft. multi conductor cable runs back to the mixer console, where all patching cables can be plugged into another cable panel on the side of the cable box. It is possible to use two of the cable extension boxes, left and right of stage for multiple microphone applications. The mother unit has a handle on the side which is used to wind in the cable at the end of the performance and can be removed and placed inside the panel box.

Mfr: Lamb Laboratories, Inc. (RSE)

Circle 62 on Reader Service Card

WIRELESS REMOTE CONTROL



• Maximum communication distance with minimum antenna and power requirements are claimed by the manufacturer of the WR 420. The system is based on a building block approach to audio visual system planning. The receiver and transmitter can be expanded by additional control modules in ten-function increments. The controls, which are suitable for remote operation of projectors, tape equipment, sound systems, lighting, etc. can be placed in any location. They are able to operate through walls or any obstruction because of a coded matrix system which does not permit interference with the command. Any number of pieces of equipment can be controlled simultaneously with a touch of the appropriate button on the remote control.

Mfr: International Visual Products,

Circle 63 on Reader Service Card

NORMAN CROWHURST

Being Practical About Feedback, part 2

Where phase is involved, gain is modified by something more complex. Most simply, phase is associated with low- and high-frequency roll-offs—the latter unavoidable.

As we showed in part 1, feedback takes care of variations in gain due to a variety of causes, as long as they do not involve phase. Whether the gain changes at different parts of a signal waveform or is due to variation in the parameters of various components, negative feedback reduces the amount by which gain changes, by the factor we introduced, called the feedback factor, $1 + A\beta$.

From the purely mathematical view, this still applies, even when phase enters the picture, but it is complicated by the fact that $1 + A\beta$ then becomes a complex quantity, and thus gain is modified by something other than the simple numeric we discussed in part 1.

PHASE—HIGH AND LOW ROLL-OFFS

While phase can enter the picture in some quite complicated forms, the simplest, and quite unavoidable ones, are associated with high and low roll-offs. Direct coupling avoids low frequency roll-offs and thus simplifies feedback design for the low frequency end. But anywhere that you put in either capacitor or inductive coupling, you are inserting a low frequency roll-off that will need consideration if it is included in a feedback loop.

Although low frequency roll-offs can be avoided, high frequency roll-offs are unavoidable. Using direct coupling extends response down to zero frequency—d.c.—but there is no way to extend high frequency response up to infinity! There has to be a high frequency roll-off somewhere.

Any roll-off can be represented by a simple r-c combination, or a simple r-l combination (Figure 1). While avoiding the use of inductors will keep the l out of it, the r-c variety, or its equivalent, due to time effects in active devices, cannot be avoided.

Figure 1. Basic configurations of reactance elements that contribute to each kind of roll-off.

Any single stage, high frequency roll-off will be characterized by a 3 dB point, at which the phase delay is 45 degrees. This occurs, when an actual r-c combination is involved, where the reactance of the c is equal to the r value it shunts. For design purposes, the frequency where this occurs is a good point to which to normalize the response. (Figure 2).

Thus at double this frequency, reactance will be half the value of the associated resistance, and phase will have risen to about 63.5 degrees. At half this frequency, reactance will be double the value of the associated resistance, and phase will be about 26.5 degrees. Figure 2 shows the response as a polar diagram, with frequency plotted along the curve. If we plot the same thing out as an amplitude and phase response, it looks as shown at Figure 3.

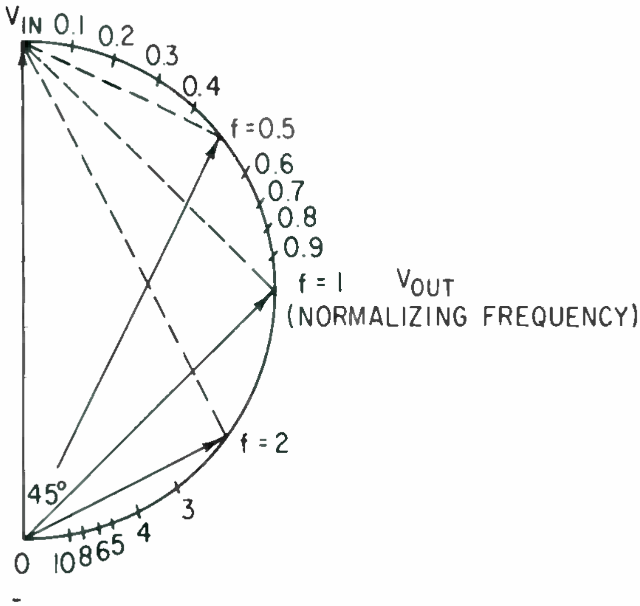


Figure 2—A polar plot of a single high frequency roll-off.

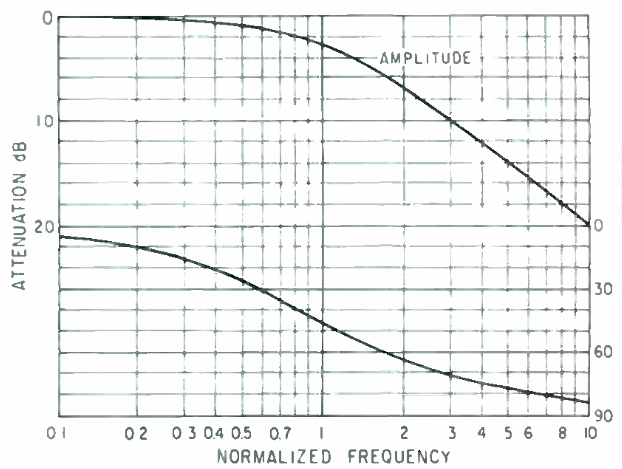


Figure 3—The same response as that shown at Fig. 2, plotted as dB and phase angle, against frequency, normalized.

An advantage of plotting it out as at FIGURE 3, is that when you have several stages, you can add the gains, as well as the dB amplitude, and the phase responses, to get the overall response all the way through. For the simple case where all roll-offs are identical, FIGURE 4 shows the response of two stages, and FIGURE 5 the response of three stages.

Note that in FIGURE 2 the curve never crosses the reactance axis to enter what we may call the positive zone. We will see what this means for feedback presently. In FIGURE 4 the curve does cross the reactance axis, going on to curves into the zero point, almost at the 180 degree phase angle. But the fact is, it only reaches the full 180 degrees, which constitutes positive feedback, when the amplitude gets down to zero—too late for it to cause oscillation.

In FIGURE 5, the curve goes further around the zero point as it turns in and thus crosses the 180 degree line, where its amplitude has dropped to 1/8th of its starting amplitude. What this means is that if the $A\beta$ product is eight or greater, its value will reverse its phase from negative to positive feedback, or from positive to negative in Formula 1: [$A_F = \dfrac{A}{1 + A\beta}$ (A_F is the gain with feedback)] It will still have an amplitude of one or greater in the positive direction.

So an amplifier with three identical roll-offs within a feedback loop will become unstable when the loop gain reaches eight, which is 18 dB. The value of $1 + A\beta$ for which this happens is nine, before phase angle starts in, representing a dB feedback of just over 19 dB.

FIGURES 2, 4, and 5 are what form the basis of Nyquist plots, which are plots for simple systems with one, two or three identical roll-offs. It is possible to make such a plot for any system, or loop. So let us generalize, to see what we could learn from such a plot.

NYQUIST PLOTS

Let us assume that the polar plot we make is for the value of $A\beta$. The usual way is have zero at the right and

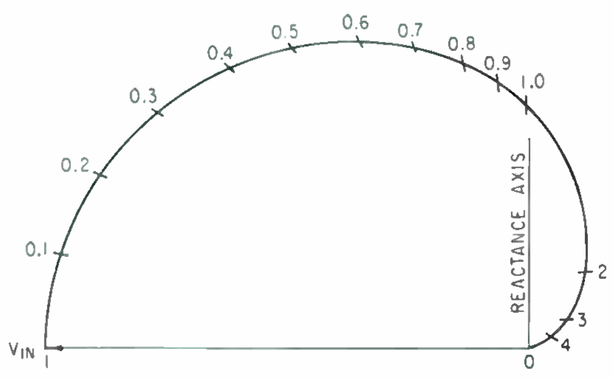


Figure 4. A polar plot for two identical high frequency roll-offs.

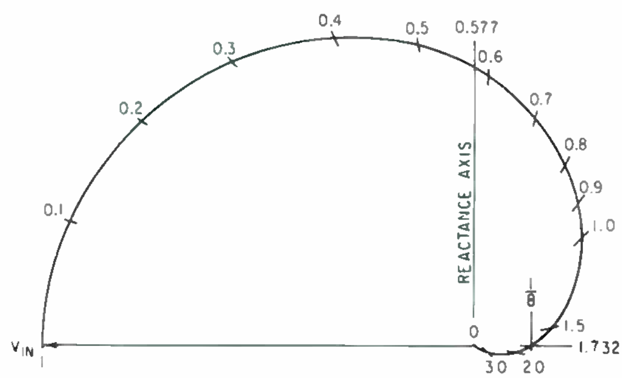


Figure 5. A polar plot for three identical high frequency roll-offs.

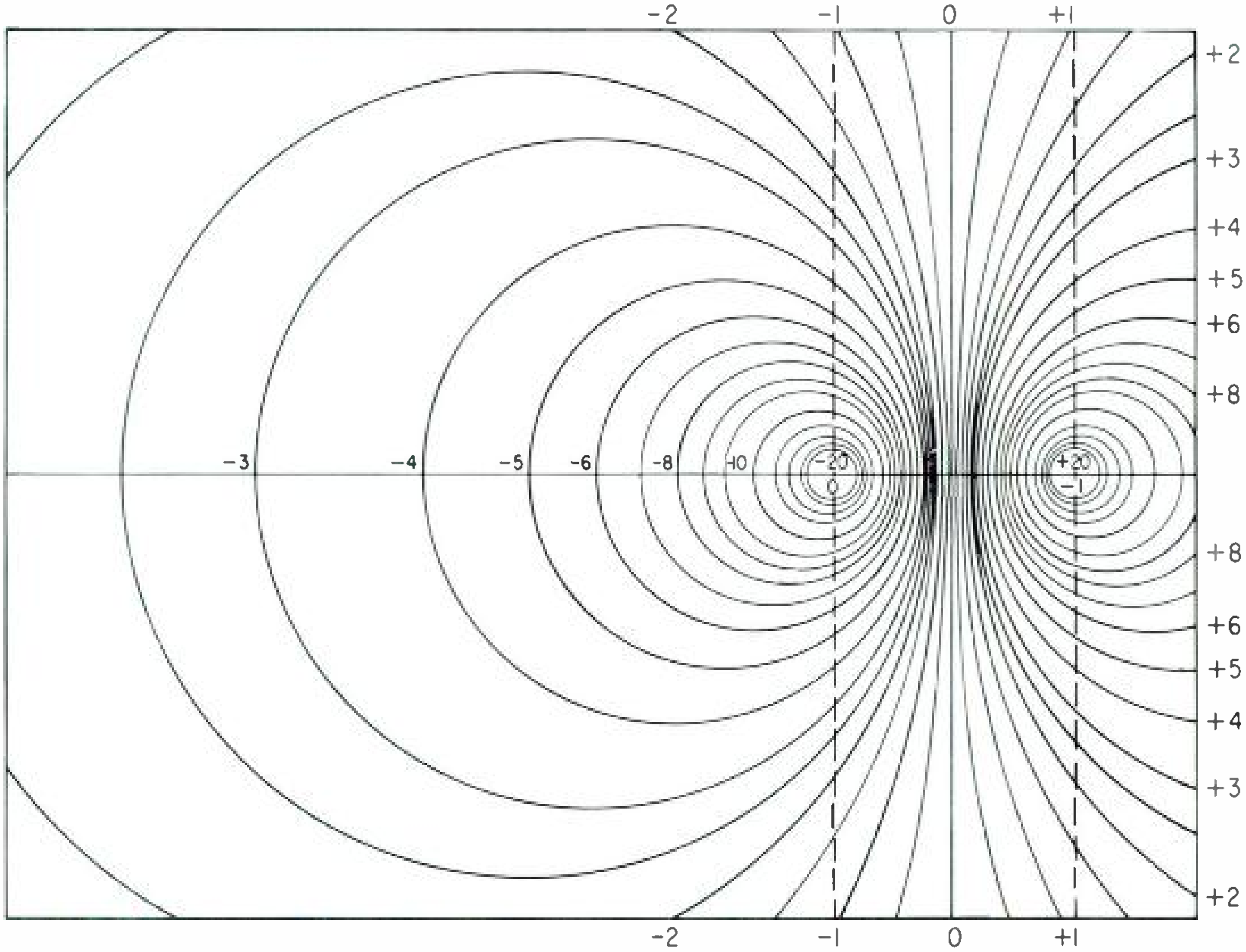


Figure 8. A completed set of rulings, using the construction of Fig. 7.

measure the unphase-shifted vector to the left. Now if (FIGURE 6), we measure to the same scale we used to measure off $A\beta$, to the left, we measure off one to the right, the new point can be labeled -1. And as the distance from zero to the point of the vector represents $A\beta$, the distance from the -1 point to the same end of the vector will represent $1 + A\beta$. A useful result!

Further, if we use as the origin, the first zero point, against which to plot a polar curve of $A\beta$, the same curve, using the -1 point as origin, will also be a polar curve of $1 + A\beta$.

PROVIDING A SCALE

Now, in most feedback systems (except where feedback is used to modify response, which we will deal with in part 3), β is a constant, a fraction. So all the phase shift and amplitude change is in A. The curve representing the locus of $A\beta$ can be used to represent the locus of vector A by merely dividing by β. Since β is always fractional, the result is that A is always bigger than $A\beta$.

Gain with feedback, from Formula 1, is given by A divided by the feedback factor $1 + A\beta$. Now the quantity $A\beta/(1 + A\beta)$ is the ratio between the two vectors, one with its origin at zero and the other with its origin at -1. So the gain with feedback is this same ratio, divided by β which, as we have assumed for the moment, is a constant.

This means that if we can provide a background scale of lines that represent constant ratios of these two vectors, we will be able to see what happens to response with feedback, at a glance. Let's take the case where $A\beta$ is half of $1 + A\beta$ (FIGURE 7).

When $A\beta$ is $-1/3$, $1 + A\beta$ is $+ 2/3$. And when

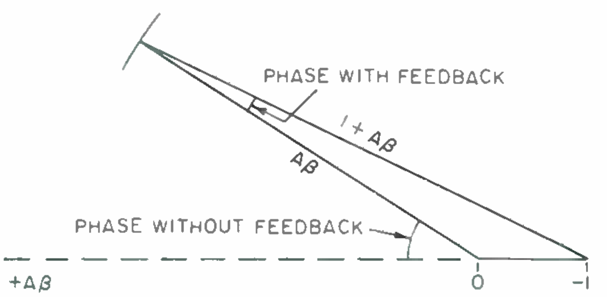


Figure 6. The construction for the Nyquist diagram for criterion of stability.

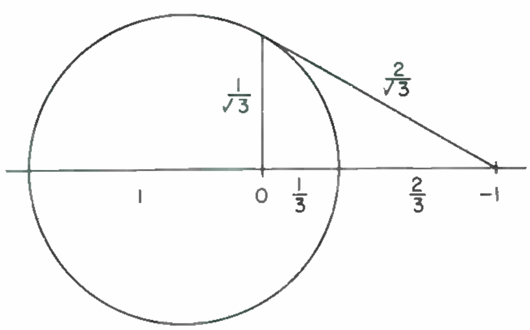


Figure 7. Construction for the circular scale rulings that enable the Nyquist plot to show response with feedback.

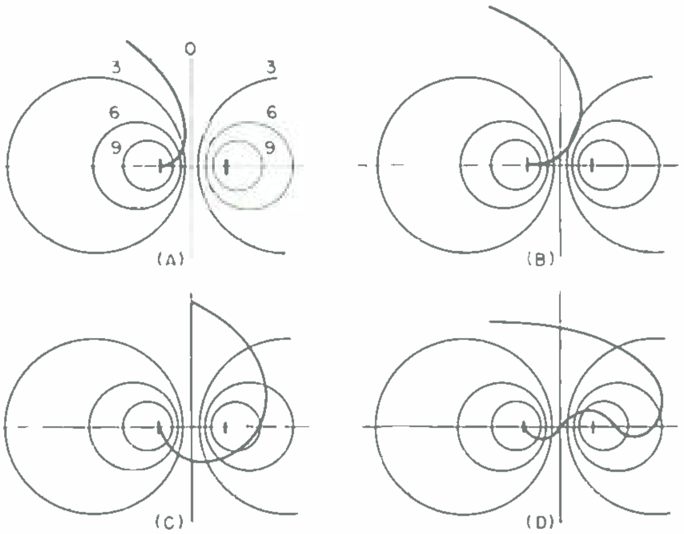


Figure 9. Some of the kinds of thing that a Nyquist plot can show: (a) a progressive roll-off with no peaking; (b) a response that has peaking before cut-off; (c) a response that will be positively unstable; (d) a theoretical response. shown in some textbooks to explain conditional stability.

$A\beta$ is $+1$, $1 + A\beta$ is $+2$. The locus of all points so that $A\beta$ is half of $1 + A\beta$ is a circle passing through the points $-1/3$ and $+1$. If you are not sure of that, take a point on the circle that is easy to figure, such as where $A\beta$ is at 90 degrees. Here its value will be $1/\sqrt{3}$, which is approximately 0.577. Applying your right triangle geometry, $1 + A\beta$ is $2/\sqrt{3}$, so it is true here.

Actually, all the loci for points at which $A\beta$ and $1 + A\beta$ have the same ratio are circles, except for where $A\beta = 1 + A\beta$, the locus for which is a straight line. If we draw these circles at, say, dB intervals, and then plot the curve for $A\beta$ using it as background, the scale provided by the circles will show the response of the amplifier with feedback. These scales are shown at FIGURE 8.

That takes care of amplitude response, in dB. For phase angle, we already have phase angles for gain without feedback, $A\beta$, and for the feedback factor, $1 + A\beta$, leaving the phase of the amplifier with feedback as the angle between the other two.

Notice that, for large amounts of feedback, the angle between the two vectors $A\beta$ and $1 + A\beta$ is much smaller than the angle of $A\beta$ or $1 + A\beta$ individually. So feedback reduces phase shift in a manner similar to the way it reduces gain.

Now if we plot the locus of vectors for $A\beta$ against frequency using this scale of circular rulings as background, we can see what kind of response we get with feedback (FIGURE 9). If the curve moves uniformly inwards toward the zero point, the response with feedback is a modified roll-off. If the curve moves outward over the circles, possibly even into the area where the circles curve around the -1 point instead of the zero point, then the response with feedback peaks, but is still stable. If you want to, you can read the amount of peak from the circles it crosses.

Finally, if the curve encircles the -1 point, the amplifier with feedback is unstable, because at the frequency where phase shifts 180 degrees for loop gain ($A\beta$), the

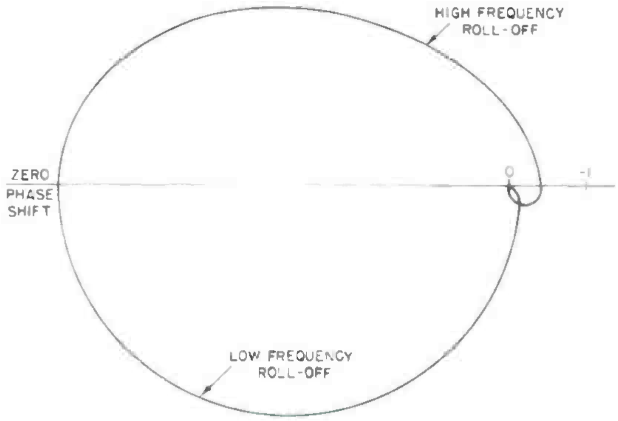


Figure 10. The complete response for an amplifier will include both high frequency and low frequency roll-off characteristics, where there is a low frequency roll-off (i.e. where direct coupling is not used throughout).

gain is more than 1, and thus it will oscillate. The textbooks usually give one more condition, called *conditional stability*, and illustrate it by the last kind of curve shown at FIGURE 9.

Here, technically, the curve does not encircle the —1 point, but returns to zero without actually going round the —1 point. According to the theory postulated by the textbooks, such an amplifier will be stable, until shocked into a condition of oscillation, at which the loop gain is more than one, but is averaged to one by distorting the waveform so that, for part of the waveform it has a value on the curve beyond —1, and for part of it the gain is zero, because the amplifier is saturated.

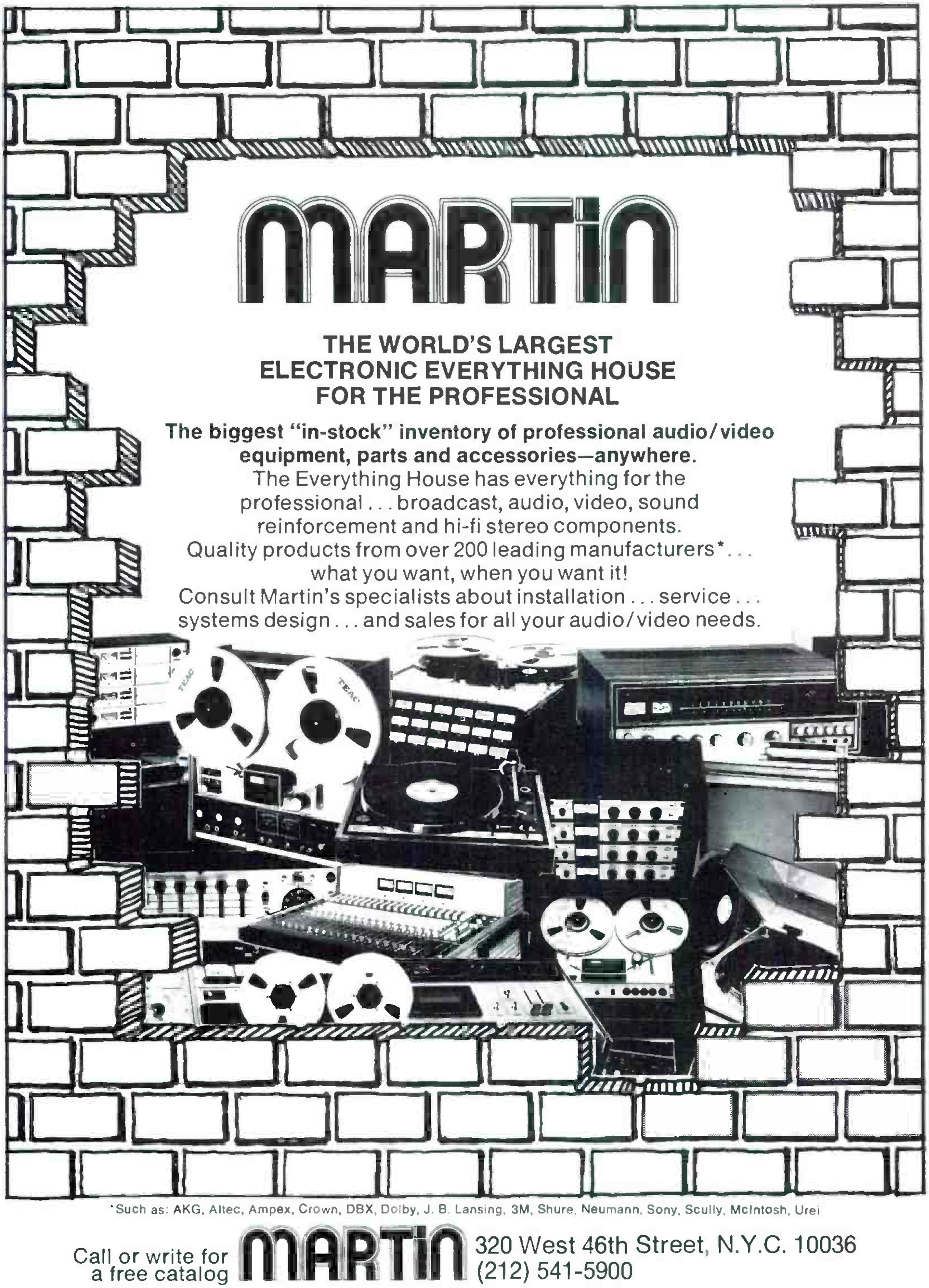
Possibly an amplifier could be constructed that would fulfill that condition. I have never made one, but I have encountered conditional stability, about which we will have more to say in part 3.

NUMBER OF ROLL-OFFS

The characteristics we have been discussing have related to high frequency roll-offs. Where r-c or r-l couplings are also used between stages, there will also be low frequency roll-offs, that will rotate the other way toward the zero point (FIGURE 10).

While it should be fairly obvious that the more stages you have inside a feedback loop, the greater the ultimate phase shift, in its multiples of 90 degrees, and thus the greater will be the possibility of running into either peaking or instability. This means that juggling values to make the amplifier stable becomes more difficult.

In this part, we have given the basics of the theory relating to phase in feedback circuits. In the final part, we will apply that theory to some practical criteria that can aid in feedback design, take a look at practical ways that feedback can be used in audio circuits, discuss some of the things that can happen that the simplified theory we have discussed so far does not cover, including some causes of conditional stability met in practice. ■



Circle 49 on Reader Service Card

JOHN WORAM

Echo and Reverb

The author provides a clear description of delay, doubling, and reverberation and decay. The benefits of each are detailed.

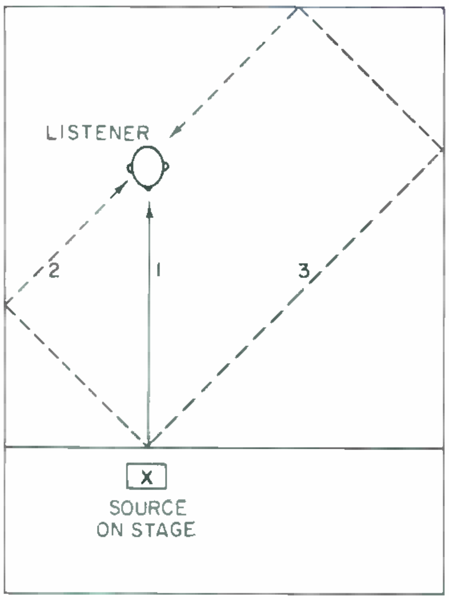


Figure 1. Echo and reverberation within a room. The listener hears a combination of (1) direct sound (2) early reflections [echo(es)] (3) later reflections [reverberation].

Herewith an excerpted chapter from John Woram's soon-to-be-published book on studio techniques and operation. When published, this chapter will contain more information and illustrations than our pages presently permit. When is the book due? Watch these pages for future announcements.

WITH THE EXCEPTION of the anechoic chamber, all monitoring environments have some effect on what the listener hears. The perceived "sound" of a musical instrument will vary from one room to another, and an acoustically dead studio will never be confused with a concert hall with superb natural acoustics.

Since the sound of a musical instrument radiates in all directions, the listener hears something quite different than the direct output of the instrument. Although some sound certainly reaches the ear via a direct path from the instrument, other signals are also present, as the sound radiates from its source to the various room surfaces; walls, ceiling, floor. Each of these surfaces absorbs some portion of the signal and reflects other portions back into the room. Consequently, the listener hears an incredibly complex mixture of direct and reflected information. The proportion varies according to where he is seated. If very close to the sound source, he will hear mostly the direct signal. On the other hand, the listener at the rear of a large concert hall may hear a signal that is almost totally reflected sounds.

The character of the reflected sound is influenced by the construction of the room surfaces and the room's volume. Some materials are more reflective than others, and all have some influence on the frequency response of the reflected signal. Carpeting for example, tends to absorb higher frequencies, while glass may efficiently reflect these same frequencies.

In the older concert halls, one often finds mirrored walls, panelling, thick plaster surfaces and perhaps parqueted wood floors—all of which reflect sound and contribute to that illusive "concert hall realism."

On the other hand, the surfaces of a well-designed rock studio may reflect very little sound back into the room. And, since so much recording is done with close microphone placement, the microphone "hears" a signal that is, for all practical purposes, only the direct sound of the instrument.

Whether the primarily direct pickup is due to close miking, a dead studio, or a combination of both, the resultant sound is usually described subjectively as *dry, tight*, or by some other descriptive term that suggests the absence of reflected information.

As a corrective measure, some sort of signal processing is often required to simulate a more natural sound. Or, for a special effect, an apparently greater than normal amount of reflected information may be sought. The engineer has several tools at his disposal to either simulate the ambience of the concert hall, or, create a unique effect that would otherwise be unattainable.

Before describing these methods, a few definitions, and a brief discussion of room acoustics, are required.

TERM	GENERAL DEFINITION	SPECIFIC DEFINITION
Echo	A repetition of a sound.	One (or a few at most) repetitions of an audio signal.
Reverberation	A re-echoed sound.	Many repetitions, becoming more closely spaced (denser) with time.
Delay	To postpone to a later date.	The time interval between a direct signal and its echo(es).
Decay	Progressive decline.	The time it takes for the echo(es) and reverberation to die away.

ROOM ACOUSTICS

The total energy present within any listening environment is a mixture of three components: the original, or direct sound, the early reflections, and the later, more diffuse reflections or *reverberation*. These three components are illustrated in FIGURE 1. Although there is, of course, only one direct path from sound source to listener, there will probably be a few early reflections and very many later reflections within the typical music listening room. In fact, a pictorial representation of an actual sound field might appear as shown in FIGURE 2.

At first, the direct sound, is heard. After a very short interval, (1), a repetition is heard as the direct sound, reflected from a nearby surface, reaches the listener's ear. Still later, (t2), another repetition is heard, and then (t3), perhaps another, as the sound reflects from other nearby surfaces. As time passes, more and more repetitions occur as the sound waves spread through the listening environment, striking and being reflected from the myriad surfaces within the room. As we reach (t4), (t5), (t6) . . . the reflections have become so closely spaced they fall almost on top of one another and the listener is no longer aware of their individual identities.

As a matter of fact, even the very earliest reflections, (t1), (t2), (t3), are seldom consciously noted as such. Instead, their presence is sensed rather than distinctly heard. However, in some cases, an early reflection may become audibly noticeable, and perhaps distracting. In certain concert halls, for example, the space under a balcony may produce an undesirable echo that is heard as an acoustic "slap-back" a moment after the direct sound. However, in a well designed room, echo and reverberation will not distract from the direct sound, yet their presence will add a richness that is conspicuously lacking in a deadened rock studio.

Referring again to FIGURE 1, we may label four variables within the listening environment. After some *delay*, there is an *echo* (or echoes), then after further delay, there are many closely spaced echoes, or *reverberation*. In time, the reverberation will die away, or *decay*.

In order to simulate this natural condition, the engineer

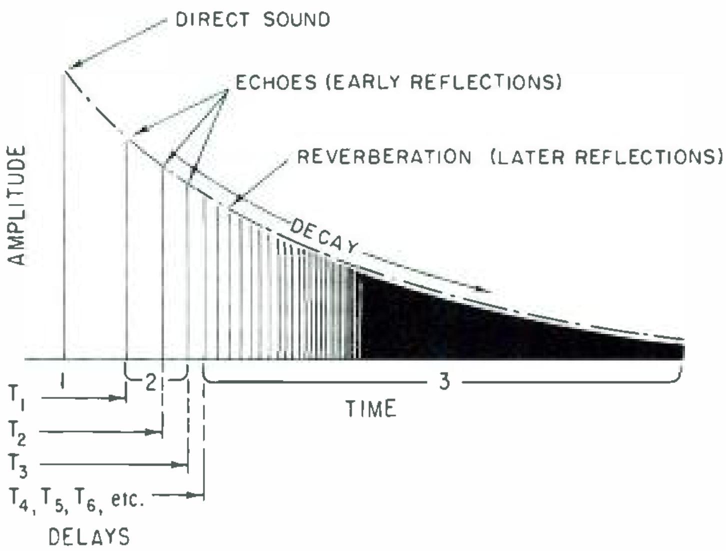


Figure 2. Pictorial representation of a sound field, showing direct sound, early reflections, and reverberation.

may be called upon to produce artificially one or more of these variables.

THE TAPE DELAY SYSTEM

To produce an echo, some sort of signal delaying process is required. Until recently, this delay was accomplished with an auxiliary tape recorder. In addition to the normal recording process, the signal to be treated would also be fed to this extra machine. Depending on tape speed and the distance between the record and playback heads, the output would be delayed by some fraction of a second. FIGURE 3 illustrates the tape delay process, and lists the delays available when the heads are about two inches apart, which is typical of many professional tape recorders.

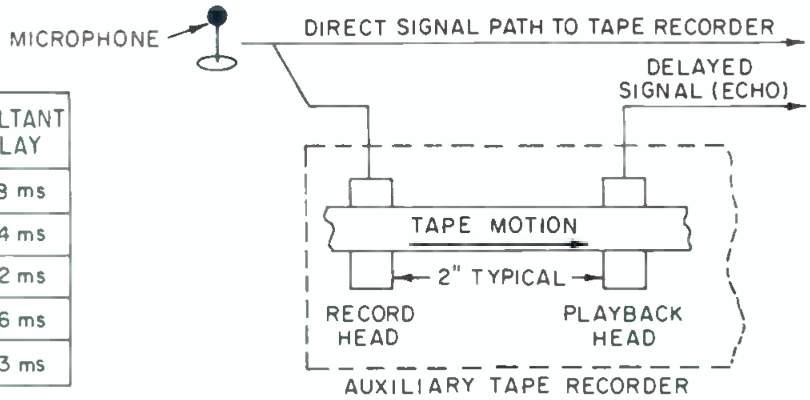
If it is possible to continuously vary the speed of the machine over a wide range, additional delays become available. But although the tape delay system is certainly usable, it may be inconvenient, since the echo tape must be rewound frequently, and at the slower speeds, the record/playback responses may not be equal to the demands of the session.

THE DIGITAL DELAY LINE

With the introduction of digital technology to audio, the so-called *digital delay line* may take over the task of the tape delay system. The advantage of the digital delay line is that the delay is provided electronically, with no moving parts in the system. And, the delays may be continuously variable over a very wide range (0-200 milliseconds, typical), with no effect on the frequency response. A modern digital delay line may also have two or more outputs, so that the input signal may produce two or more echoes, each of which is delayed by a different amount. To create more than one echo using tape delay, each output would have to feed an additional input on the auxiliary tape recorder. Consequently, each additional delay would add one more generation of tape noise.

ACOUSTIC DELAY LINE

Another recent development in delay lines is the Cooper Time Cube, developed by Dr. Duane H. Cooper of the University of Illinois. The device uses acoustical delay lines in the form of coiled tubes, through which the signal



TAPE SPEED	RESULTANT DELAY
3 3/4 IN/SEC	528 ms
7 1/2 IN/SEC	264 ms
15 IN/SEC	132 ms
30 IN/SEC	66 ms
60 IN/SEC	33 ms

ms = MILLISECONDS (THOUSANDTHS OF A SECOND)

Figure 3. Artificial echo, produced by a tape delay system.

to be delayed travels. A pickup hears the sound after it has travelled the length of the tube. Two tubes are provided, giving delays of 14 and 16 milliseconds. The two may be combined in series to produce a delay of 30 milliseconds: however no other delays are available without physically altering the lengths of tubing. Of course, some care must be taken to protect the units from mechanical vibration, which would be transmitted to, and then through, the tubes. The electronic circuitry is less complex than the digital delay line, and it is consequently less expensive—though admittedly not quite so flexible as devices using digital technology.

DOUBLING

It is of course possible to produce a delay so short that the ear cannot recognize it as such, even if its level equals that of the direct sound. Generally, as the delay is increased between 20 and 40 milliseconds, there comes a point within that range at which the echo separates from the direct signal and becomes clearly audible. The actual time value at which this happens depends on the nature of the signal in question. The echo of a drum figure might be clearly heard at 20 milliseconds, while a legato string line may require twice that delay or more before the echo is heard.

These very short delays may often be used to simulate the effect of, say, a larger string section. In a large string section, the ensemble is never precisely synchronized. In fact, the very slight imprecision of attack within the ensemble gives the listener an aural clue that the group is large. (Of course, once the imprecision becomes overdone, the listener will object to the sloppy playing.)

Now, by selecting a delay that is too short to create an audible echo, the engineer can apparently double the size of the string section. The almost instantaneous repetition of the signal simulates that very slight imprecision which is characteristic of a large group. The result may be particularly effective if the delayed signal is panned somewhat away from the direct signal, to create the illusion that the ensemble is spread out over a wider area.

Once sufficiently delayed signals have been produced, they may be mixed with the direct signal to simulate the early reflections (echoes) of a concert hall. Or, for a special effect, these echoes may be combined out of proportion to the direct sound. However, the natural reverberation content of the concert hall cannot be satisfactorily created with a time delay system.

REVERBERATION AND DECAY

Reverberation was described earlier as a series of closely spaced reflections—so closely spaced that they are not perceived as discrete echoes. Rather, their cumulative effect creates an impression of room liveness.

The time intervals between the multiple reflections are entirely random, and to create such an irregular pattern with a series of delay lines would require a rather involved—and expensive—system. At this time, it is more practical to use an artifical reverberation device, which simulates the random multiple echo pattern of natural reverberation without actually producing discrete echoes. Basically, an elastic medium is set into motion by an applied audio signal, and the to-and-from vibration is sensed by a special type of contact microphone attached to the vibrating medium. When the applied signal is removed, the vibration gradually diminishes, simulating the natural decay of room reverberation.

THE REVERBERATION PLATE

In professional studios, perhaps the most widely used device is the artificial reverberation plate. This unit consists of a steel plate, protectively suspended within a wooden cabinet. Attached to the plate is a driver element, not unlike a very small loudspeaker. The element is driven by an audio signal, and sets the plate in motion. As the plate flexes back and forth, the motion is sensed by two contact pickups, also attached to the plate. The audio signal produced by these pickups simulates the reverberant field of a large concert hall. When the applied signal ceases, inertia keeps the plate in motion for several seconds, although via a mechanical damping system, this decay time may be reduced as desired.

SPRING REVERBERATION SYSTEMS

Although the steel plate produces a remarkably good simulation of reverberant sound, it is rather large and quite expensive. Later generation plates, although considerably smaller, are no less costly and may be somewhat beyond the reach of studios with a modest equipment budget.

Reverberation devices using coiled springs are available over a wide price range, and although the least expensive ones certainly are not the equal of the steel plate, they are easily within the financial reach of the smallest studio operation.

Although specific construction details vary from one manufacturer to another, this type of system uses a long coiled spring, suspended between a driver and a pickup element. The applied signal sets the spring in motion, and this motion is sensed by the pickup element.

The simplest spring units are easily recognized by a characteristic metallic sound quality which is unconvincing as a simulation of natural reverberation. However, as a special effect on say, an electric guitar, the springy sound may create a unique sound texture not otherwise attainable. Inexpensive spring units have been built into a wide variety of guitar amplifiers and electric keyboard instruments with considerable success, and the well equipped studio may have a few available for use as the occasion demands.

The more sophisticated spring systems have decay time adjustments, and a sound quality that many find comparable in quality to the steel plate. In addition, the decay time may be varied while the unit is in use, since this adjustment does not produce any mechanical vibration which might be "heard" by the pickup element.

ACOUSTIC REVERBERATION CHAMBERS

The most elegant reverberation device is the acoustic reverberation chamber, actually a highly reverberant room in which all surfaces have been treated for maximum reflectivity. A loudspeaker placed within the room transmits the signal to be processed, and a microphone placed some distance away picks up the multiplicity of echoes produced as the signal is reflected again and again within the room.

Although the room is unsatisfactory for normal listening, the overly reverberant sound, combined with the direct unprocessed signal, creates the illusion that the recording was made in a very large room with normal reverberant characteristics.

The acoustic reverberation chamber is a luxury many studios cannot afford. In order to function adequately, the room must be reasonably large, and of course carefully isolated from extraneous noise sources. Especially in high rent districts, it may be difficult to justify setting aside an appreciable area solely for the creation of reverberation. And, decay time may not be varied with ease, as on the steel plate or high quality spring system. To significantly alter the room's characteristics, some acoustic padding, perhaps in the form of draperies, may be added or removed. Although this is certainly effective, it can be a time consuming process to tailor the reverberant sound to fit the immediate needs of the session. However, the well designed chamber does produce a most pleasant reverberant field, and is perhaps the closest approximation of natural concert hall reverberation that is available at this time.

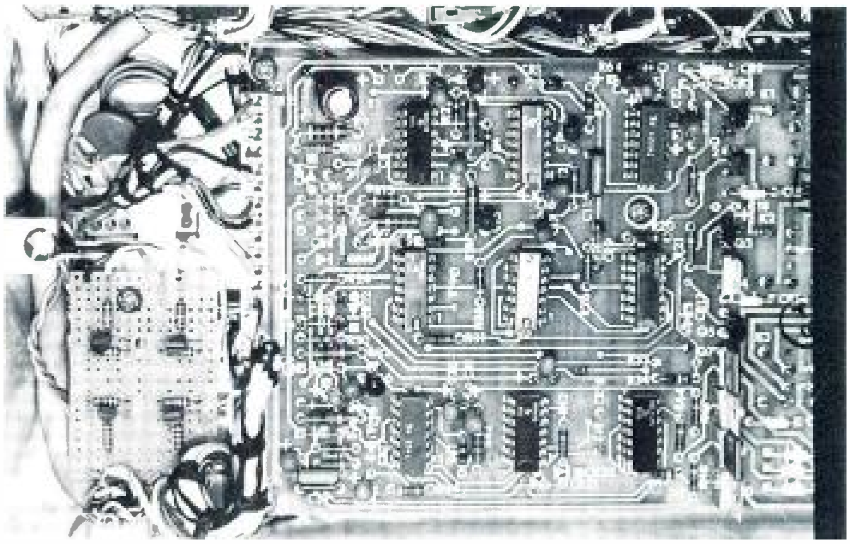
STEREO REVERBERATION

By its nature, natural reverberation is diffuse, and the listener is scarcely conscious of a recognizable location from which the reverberation is coming. Rather, the reverberation envelops the listener from all sides.

But, regardless of construction details—steel plate, spring, or room—a reverberation device with only one output is limited in its effectiveness to simulate the natural condition. A single output, panned to some specific point, will be just that; a point source of sound, reverberant in quality but not diffuse.

Consequently, the definitive stereo reverberation system should have two outputs, both derived from the same input. For example, the steel plate mentioned earlier has two pickup devices, located at different distances from the driver element. As the plate flexes, the instantaneous deflection is randomly dis-similar at the two pickup locations. However, if the two outputs were compared, one at a time, they would sound practically identical since they are, of course, reacting to the same input. But, if one output is routed to the left and the other to the right, the subtle variations in phase will create an overall ambient sound field that more closely resembles natural reverberation.

In the acoustic reverberation chamber, two microphones may be set up within the room to create the same effect. ■



Inside view of chassis, showing location of the added circuit board, lower left. The cable along the left side brings the remote control and light functions from the local buttons to the enlarged remote control plug at the lower left.

MITCHELL D. TANENBAUM

Modifying a Scully 280B

Several ingenious changes in the transport electronics of the Scully 280B improve remote operation, motion sense response, and editing capability.

UPON INSTALLING AND TESTING four new Scully 280Bs at our studios, we found certain lacks in the transport control electronics. Pushbuttons did not light and no provision was made for lighting them. This means that an operator using the machine by remote control had no indication of the status of the transport. The *stop* button was lit when power was applied, regardless of the position of the tape-break arm.

The automatic tape lifter was defeated by depressing one of the remote fast wind buttons. We have always preferred separate remote and defeat buttons. In addition, the transport would often remain in an interlocked state due to the motion sense even after tape motion ceased. Also, we wanted an edit mode similar to Ampex' *play-edit*.

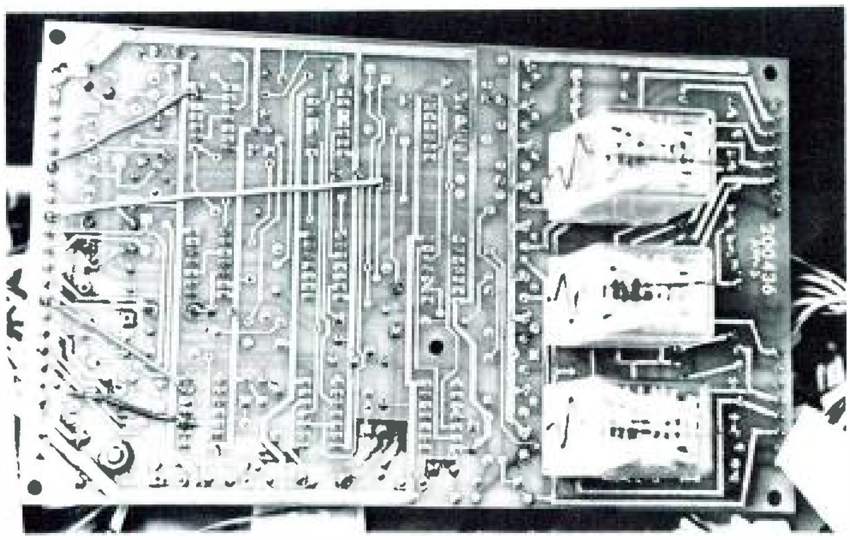
We decided that with a little ingenuity, we could alter the electronics to fit our needs. Because of the flexible nature of digital circuitry, these modifications are straightforward and can be made with few alterations to the logic board. Fortunately, there are a sufficient number of redundant contacts on the logic board on which to bring out the extra functions. One small additional circuit board must be added to switch the lights. This board can mount conveniently to the left of the logic board.

LIGHTING THE PUSHBUTTONS

In order to light the pushbuttons, logic signals are taken from key points in the circuit and fed to a Darlington configuration light driver. The high input impedance and gain of this circuit allows several lights to be driven without exceeding the fan-out capabilities of the ttl circuitry. Shunt resistor R2 keeps a small warming current through the bulb when it is off to prevent a current surge during cold turn-on. Four of these circuits are used to light the *play*, *fast forward*, *rewind*, and *edit* pushbuttons.

The following flip-flops will provide a Logic 1 when the transport is in the designated mode:

Mitchell D. Tanenbaum is technical director of station WBFO, Buffalo, N.Y.



View of the under-side of the circuit board showing the added jumpers and broken foil between contacts. Some unused holes in the board may be used for at least one of the jumpers.

Play	U5, Pin 10
Fast Forward	U1, Pin 10
Rewind	U1, Pin 13
Edit	U7, Pin 13

These logic levels are used to drive the Darlington circuits.

In order to have the stop light indicate the status of the tape-break arm, the arm switch and its connectors must be rewired, with the wiper to ground, the normally closed contact to U4, Pin 5, and the normally open contact to the low side of the stop light. Remove the hard-wired ground to the stop light.

An open ttl input is at Logic 1. However, Scully has tied all the inputs to the logic board low, using 180-ohm resistors.

Therefore, R19 must be removed and R17 replaced with a jumper. C20 will still serve to slow the rise and fall times of the input, preventing transients from affecting the transport mode. When the tape-break arm is down, U4 Pin 5 is held at Logic 0 and the stop light has no ground return. When the arm is raised, U4 Pin 5 is opened, raising it to Logic 1. Simultaneously, a return is provided for the stop light.

SEPARATE REMOTE ATL DEFEAT

In order to provide a separate remote atl defeat, CR4, CR12, and R33 must be removed. Delay capacitor C18 may be removed to provide instant response on the atl defeat. The application of 5V to R34 will defeat the atl.

Replacement of R29 (470k) with 150k will improve the response time of the motion sensing.

Removal of CR20 will allow the transport to enter the edit mode from play.

Following are step-by-step instructions for modification of the logic board:

Remove: R19, R33, C-18, CR4, CR12, CR20
Remove: R17 (47) and replace with Jumper.
Remove: R29 (470k) and replace with 150k.
Break foil: J202 between Pins 2 and 3.
J205 between Pins 6 and 7, and Pins 9 and 10.

Jumper From	*To*	*Function*
U5 Pin 10	J205 Pin 10	Play Status
U2 Pin 12	R33 (J202 side)	F.F. Status
U2 Pin 13	J202 Pin 3	Rew. Status
U7 Pin 13	J205 Pin 7	Edit Status

The following changes in J202 have now been made:

Pin 3: Rewind status—Pin 7: F.F. status.

The following changes in J205 have now been made:

Pin 7: Edit status—Pin 10: Play Status—Pin 11: atl defeat.

MODIFICATION OF THE CHASSIS

Remove the 9 Pin remote connector (J111) and replace with a 15 Pin Molex connector. The hole will have to be enlarged slightly. The following is the pin configuration we used:

PIN NO.:	1	2	3	4	5	6	7	8	9	10
FUNCTION:	Rec	Rec Lite	Play	Play Lite	Rew.	Rew. Lite	F.F.	F.F. Lite	Stop	Stop Lite

PIN NO.:	11	12	13	14	15
FUNCTION:	atl def	+5V	+24V	Ground	

The remote control buttons should be paralleled with the local buttons. The +5V line is common to all buttons.

The remote lights should be paralleled with the local lights, and the +24V line should be bussed to all the lights. The low side of each light should go to the collector of its driver.

The tape-break arm must be rewired. The wiper is grounded, the n.c. contact to J202 Pin 1, and the n.o. contact to the low side of the *stop* light. Remove the ground, which is presently connected to the low side of the bulb.

These modifications will be beneficial to remote operation of the Scully 280B and will give improved motion sense response and edit capability. Also, the status outputs from the logic board can be easily interfaced with larger digital control systems.

Special function sequences can be easily predetermined and controlled for production flexibility. Several machines can interact. ■

REFERENCE

For more detailed diagrams of the 280B, see *Instructional and Maintenance Manual for Scully 280B Series Recorders/Rproducers,* Scully-Metrotech Div., Dictaphone Corp., 475 Ellis St., Mountain View, Ca. 94040.

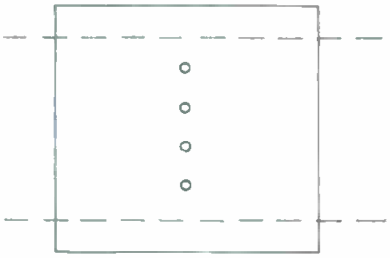


Figure 1. The gaps in a headstack.

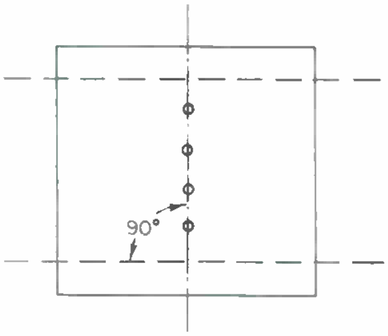


Figure 2. A perpendicular line drawn through the points gives the impression of perfect alignment.

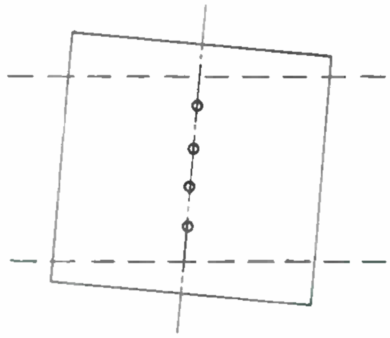


Figure 3. Closer scrutiny of the gap points.

JIM GRIMSHAW

Multi-Track Phasing Errors

Tracing a mysterious misalignment in mono multi-track recording discovered a mechanical gap which can be corrected with a little care. This article may prove basic to some but vital to others and it is so offered.

THERE EXISTS in the innards of every multi-track machine certain problems. The limits of frequency response, distortion, noise and the like are common irritations. However, one very important parameter has been omitted from our ever-growing list of limits and specifications. That missing link is the adjustable mechanical phasing error. The phasing error I'm talking about is not the mechanical error inherent in every headstack, nor the electrical phasing involved in recorder electronics, although these are not to be overlooked. Rather, the consideration here is with the illustration of a third error which can, without proper maintenance and care, literally destroy a multi-track session.

Since the beginning of time, recorder manufacturers have played down the significance of recorder phasing problems. Their well-worn theory is that "there are too many acoustical phasing problems and incompatibilities present in the original program material for anyone to be concerned with the minimal problems the recorder itself may introduce." I contend that most recorders in use today are misaligned, and probably came from the factory that way.

Jim Grimshaw is associated with Fred Arthur Productions, Ltd. in Denver, Colorado.

MULTI-TRACK IS THE CULPRIT

During a multi-track session a few years back, I began to get the feeling that what I was hearing during playback was not quite what I had heard coming straight through the board. Now taking into consideration the fact that we all miss things during the heat of recording, I decided to run a mono mix on the next take to see which was deceiving me—my ears or my memory. Later, while alone, I had a little time to go back and make some rather detailed a/b comparisons of the master and the quarter-inch mono mix. To my surprise, I discovered that the multi-track did indeed appear to have some very serious problems which the mono mix did not. The mono mix sounded just as I remembered the session to have sounded. The multi-track, on the other hand, seemed to have lost something in the mixdown. I did my best to duplicate the conditions under which I had made the first mono mix, but regardless of what I did I just could not make the master sound like the quarter inch. It didn't take a genius at this point to guess that the multi-track machine itself was the culprit.

That assumption, although later to be proven correct, did appear to have some hidden ambiguities. The machine appeared for the most part to be a reliable unit and had a good overall sound. Bench tests proved the unit to be meeting specifications and a poll of other engineers using the machine proved nothing. The only thing left to do was

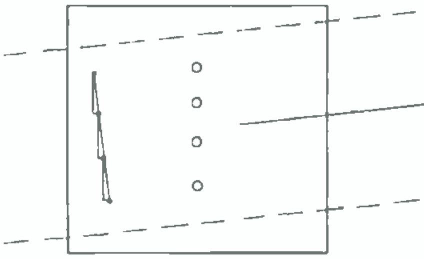


Figure 4. A perfectly recorded tape being played on a misaligned head.

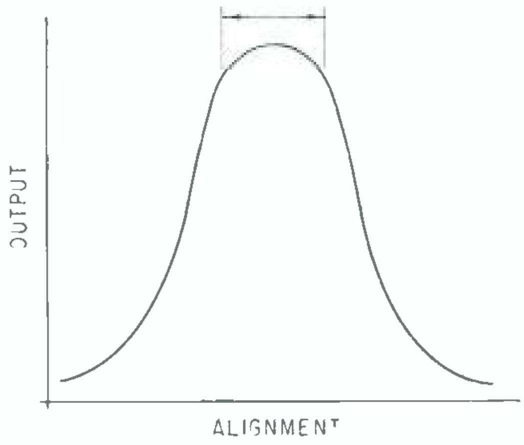


Figure 5. Usual output-versus-azimuth of a playback head.

to test under studio conditions and to keep a close eye on the unit.

During the next session, a lot of machines were kept rolling: the machine in question, a backup unit, a two-track, and a mono machine. Everything worked fine. I decided to keep this testing procedure in operation for the next several sessions in the hope that somewhere along the line the multi-track machine would foul up again and I'd be able to catch the trouble.

A couple of weeks went by and still the little beauty was performing to beat the band. Then late one afternoon I received a request to record a drum set in stereo. Well, you can guess the rest. Playback in stereo was fine, but put those drums in mono—it'll never happen.

Although I did not know the technical reason for this problem at the time, I did come to realize what the difference between the fouled up sessions and good sessions had been. As long as no attempt was made to record anything in stereo or as long as leakage was kept low, the machine worked fine. But high leakage or stereo splitting was *verboten*. Well, even a first year technical student could tell you that you've got a phasing problem somewhere along the line.

A quick consultation with the manufacturer proved of little value, so the machine went back on the bench again, this time for its most thorough checkout ever. Amplifiers were tested first. Phase shift comparisons were almost unreadable, as the manufacturer guaranteed. All wiring from and throughout the machine proved correct. As a last ditch effort to prove what I was beginning to believe was an erroneous diagnosis, I fed a test tone into the machine and began comparing tracks. Well, there it was as plain as day. Even though I was putting in a highly compatible signal, what I was getting out was definitely not compatible. A little research and study revealed to me what I am now about to tell you.

The machine in question was a well-known manufacturer's four-track. The test procedure I initiated was to inject a 15 kHz line level signal from a single oscillator into all record inputs simultaneously. This assured a compatible input. Next, I set the machine in record and monitored the playback, using a 'scope adjusted for vectoral analysis. I rigged a switching network to allow me to compare all possible combinations of track pairs. I thus observed varying degrees of compatibility, depending upon the pairs selected.

ADJUSTMENT ERROR

From this and a good deal more testing, I derived the following theory: our phasing error can be defined as the mechanical gap or adjustmental error which produces a leading or lagging characteristic between any two or more headstacks when a uniphase, unifrequency source is passed across the head gaps or coils, or *vice versa*.

Let us *assume* for the moment that the gaps of any headstack can be thought of as perfect points. (FIGURE 1). Now let's consider the dotted line in FIGURE 1 as the tape edges. Passing a line through the points, as in FIGURE 2, gives us a representation of perfect alignment. All gaps are in perfect alignment with one another. Gaps are conglomerately perpendicular to the tape edges. In considering FIGURE 3, we see a situation somewhat less than perfect. Gap points still appear theoretically aligned with one another, but are they? Actually their group alignment is highly dependent on their alignment with the tape. FIGURE 4 gives us a representation of a perfectly recorded tape being played on a misaligned head. As can easily be seen, the gap conglomerate-to-tape misalignment has caused an apparent gap-to-gap misalignment. In this case, track 4 is leading track 3, track 3 is leading track 2 and so on.

So now you're saying to yourself, "This guy is nuts" because any fool can see the tape would be off azimuth. Okay, I agree, but this is so only in the most radical cases. Remember when we assumed the gaps to be represented by single points? Well we all know that it just isn't so in real life. Giving the gaps back their length, we must also give them back their width, and herein lies the problem. Any head, even a spanking new one, has an output-versus-azimuth graph resembling that of FIGURE 5.

It's in this flat portion of the curve that we are all fooled. Trying to adjust out multi-track headstacks with an a.c. vtvm, or machine vu meter is like the proverbial needle in the haystack hunt. New heads will give a narrower peak than old worn heads, but adjustment with anything other than a 'scope or phasing meter will not produce accurate alignment.

But you still say you're not convinced that a headstack can be giving maximum output and be out of phase with its neighbor. Let's take a look at the numbers. Taking a standard two-track machine operating at 7.5 in/sec and using a test tone of 15 kHz we find that one electromagnetic wavelength is equivalent to only 0.0005 in. That being the case one degree is equivalent to 0.00000139 in. Not much, huh? In adjustment, these dimensions are represented by almost imperceptible movements of the adjustment screws.

TRANSPORT SPEED

Many factors affect the problem under discussion. Among these is transport speed. Higher transport speeds reduce the phasing error. For example, taking our statistical two-track mentioned before, and upping its transport speed to 15 in/sec. at the same test frequency, we find that one wavelength is now equivalent to 0.001 in. Conversely, dropping the transport speed to 3.75 in sec., we arrive at the very short wavelength of 0.00025.

To test and adjust any multi-track machine for phasing error, set up as you usually would. You'll need your test tapes, preferably new ones—older tapes have a tendency to develop a phasing error distortion. You should also have good quality oscillator and a 'scope adjusted for vectoral analysis. After the usual cleaning and demagnetization, put on an alignment tape. Adjust your nominal output at 700 Hz or 1 kHz to zero for all tracks. Now jump up to 15 kHz and trim all outputs for zero out. You can go back later and set these where you usually would.

While reproducing 15 kHz, make comparisons of pairs of tracks, feeding one of each pair to the horizontal input and the other to the vertical input of your 'scope. All possible combinations of pairs must be checked. Example: 1 & 2, 1 & 3, 1 & 4, 2 & 3, 2 & 4, 3 & 4, for a four-track unit. If any pairs of tracks appear to be in anything other than a compatible relationship, adjustments are necessary. A good procedure to follow is to start with the outside tracks and make minute adjustments on the azimuth screw until these become compatible. Make sure as you do that you do not lose your peak output.

Now check again all possible pairs to determine if all combinations are in phase. As with all things, compromises must be made. No two machines are alike; some will adjust out well and others will not. If you are unable to get all possible pairs to achieve a less than 90-degree relationship, check your basic alignment. Check for skew, tracking, tape tension, etc. As a last resort, try a new alignment tape if available. If all else fails, you may be in need of new heads. Remember even new heads can be defective.

If you are able to align the playback headstack for compatible phase relationship between all pairs of tracks, without losing output peak, at 15 kHz, run your alignment tape down to 7.5 kHz. Now check all possible pairs, as you've done before. It is very possible to be in good phase relationships at 15 kHz, and be out of phase at 7.5 kHz. After that, check the next sub multiple—3.75 kHz or the nearest tone you have. If this checks out, you're probably home free as far as the playback head is concerned. Remember, as the frequency goes down, the wavelength gets longer. The longer the wavelength, the less error is possible.

Now let's turn our attention toward the record head. Feed in a 700 Hz or 1 kHz tone and adjust the record gain while recording for a zero output, at the same time monitoring the playback. Then jump up to 15 kHz and trim the outputs for zero out. Again while recording, follow the same procedure as you used for aligning the reproduce head. Compare all possible combinations of tracks and make slight adjustments in the record head azimuth until all tracks appear compatible with one another. Remember, don't lose the peak output in the process and don't touch the reproduce head or you'll have to start all over. Once you've got the tracks compatible at 15 kHz, sweep your oscillator down to about 1 kHz while comparing each pair of tracks.

If the alignment is correct, phase relationships will become more compatible as you sweep down. If any phase reversals occur, you must start again. Obviously the procedure becomes more time consuming and difficult as the tracks increase in number.

In conclusion, I should state that phasing error, as defined here, is only critical when program material of common source is applied to two or more tracks on a multi-track machine. Examples of this would be the recording of anything in stereo or in quad, such as drums or brass. Possibly the most important area is in the two-or four-track quad mix where many common sources are combined together. We must all realize that mono is still with us and multi-track compatible air plays are necessary to reproduce the sound with the greatest fidelity possible. ■

GEORGE ALEXANDROVICH

Balanced and Unbalanced Lines

You must know what the advantages and disadvantages of each are before you can use either effectively.

Reprinted from db July 1969

How many times have you asked yourself, "Should I use balanced or unbalanced lines?" Before we get into any further discussions about the use of the lines, let us look at balanced and unbalanced lines in general.

Transfer of electrical signals from one location to another can be accomplished only by means of a closed loop, meaning that for energy being sent out, an equal amount of energy with the reverse polarity should be received back. If we talk about electron flow, the amount of electrons being sent into the line should be replenished through the incoming line, thereby producing the current flow. We talk here about two wires; one, as being used to send signal out, and the other which acts as a return path for the same signal.

Signal currents in the audio-frequency range changes its direction many times per second, but the direction of current flow in two conductors is always opposite to each other.

All audio transmission lines work on the same principle. In the case of field telephones for instance, one wire is omitted and ground is used as a return path (FIGURE 1). Therefore, basically only one wire (hot) has to be strung. This single conductor acts pretty much as an antenna, picking up random electrical fields surrounding it and mixing them with the original signal being sent through the line. The ability of this transmission line to pick up extraneous harmful fields causes electrical interferences. Since one side of such a transmission line is at ground potential and the other is not, we call such lines unbalanced (FIGURE 2). Unbalanced it

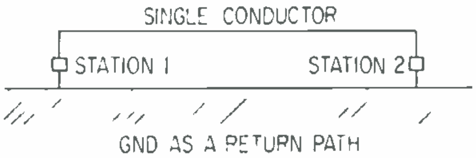


Figure 1. An unbalanced line such as is used by field telephones.

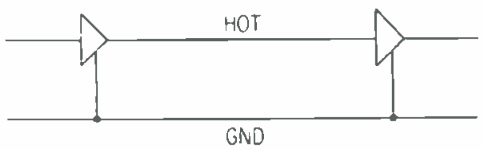


Figure 2. The unbalanced interconnection of amplifier stages.

is, with respect to ground and external fields. One way to protect such a line from picking up external interferences is to isolate it from external fields. This can be accomplished through shielding. In order to minimize the effect of external fields, insulated transmission wires are kept close together with flexible metal shield in the form of a tube surrounding them. This shield is normally grounded at either one or many points.

If this shield were near 100 per cent effective, it would have to be made of high-quality magnetic shielding material with an ability to block all electrostatic and magnetic fields as well. Shields made out of aluminum, copper, or other non-magnetic (non-ferrous) materials stop *almost* all electrical interferences caused by electrostatic and radio-frequency fields but don't protect the wires from picking up low-frequency magnetic fields from a.c. transformers, power lines, motors, solenoids, and other sources).

Since there are no practical shields which protect the unbalanced transmission lines completely for all types of interferences (and in order to accomplish transmission of low-level signals economically over long lines) a method of balancing transmission lines at both ends is being used. This method of balancing has been known for decades, the largest user of balanced lines are telephone companies. Almost all telephone installations and exchanges use balanced lines which allow use of unshielded wires with relatively low losses (FIGURE. 3).

In order to balance the line, transformers must be used at each end of the line. Transformer windings connected to the balanced line are made with the center tap normally grounded, at least at one end. The idea behind this hookup is to keep both wires of the balanced line at the same potential with respect to the ground. External fields affecting these lines, obviously affect both sides of the line equally at the same time, although both sides of the line may be saturated with external fields. Signals arriving at the terminating transformer cancel each other out.

The effectiveness of a balanced line depends on two factors. *First*, on the transformers and their symmetry of windings; *second*, on the transmission lines and the symmetry of the interference signals induced in both sides of the line.

Before we start saying when we

Figure 3. The use of a transformer for the conversion of a line to and from unbalanced to balanced.

Figure 4. A shielded unbalanced line using a double conductor within a shielded shell.

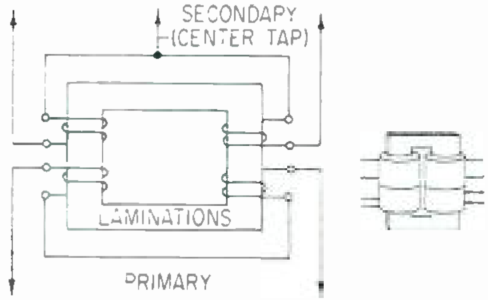


Figure 5. A balanced transformer.

should use balanced and when unbalanced lines, we ought to examine the advantages and disadvantages of each.

Balanced lines are obviously less sensitive to the external interference but require transformers which are costly, bulky, and can pick up external fields themselves. Unbalanced lines are more economical to use over shorter distances but require much more care in connecting two pieces of equipment in order not to cause any harmful ground loops. Unbalanced lines also are more susceptible to external interference. Balanced lines are cheaper for longer transmission hauls because they do not require shielding. Their interwire capacitance as well as capacitance to ground can be kept small (if space is not premium) but they require good transformers which impose limitations of their own on frequency response, level, noise, distortion, and phase.

Unbalanced lines are very costly on long hauls but very economical when used within the system. They are very convenient when used with transformer-less transistorized equipment and do not present limitations on electrical performance found in balanced circuits. A disadvantage (but many may think of it as an advantage) is the ability of a balanced line to have its phase inverted 180 degrees, something that is harder to do with unbalanced lines unless a transformer is used or a phase inverting amplifier.

We come to the point where we should decide on an arbitrary set of rules guided by the state of technology in the field of audio, regarding the time and place for the use of balanced and unbalanced lines. (Let us just add before any final deductions are made that in both cases the amount of interference picked up by the line is proportional to the length of it. Other factors may have an influence on the amount of pickup, such as the curvature of the lines which may have some cancelling effect, or ground capacitance which would have tendency to attenuate any rf type of interference. These factors will be neglected at present.)

The set of rules we have to establish are to be developed based on the experimentation and experience of many audio specialists and engineers in this field. Let us start with microphone lines.

Don't ever attempt to use unbalanced mic-input lines unless they are shorter than a few inches. This includes the mic for talkback system within a console, announcer mics on goosenecks with the mic wires going directly into the preamp input, or any microphone with built-in amplifier (fet condenser microphones which may have output levels far exceeding levels of conventional microphones). Although it is possible to run some mic lines unbalanced, no self-respecting professional would dare to run these lines any other way than balanced.

In the cases where inter-connection of several pieces of equipment with unknown ground potentials and different power supplies is expected, balanced lines are mandatory. This refers to patch bays in particular. You never know what equipment you will be called upon to patch in and the only sure way to prevent ground loops, noises, and melted patch bays, amplifiers, and power supplies is to use balanced lines isolated with transformers.

The trends of today's designs are to process the signal from the source to the final destination with the least amount of deterioration of quality. In this age of economical awareness, miniaturization, and high performance standards, the transformer is the most objectionable part of the system, so a tendency exists to eliminate it from as many circuits as possible. I think this trend against the *indiscriminate* use of transformers is a healthy one, but I also think that we will never get to the point where transformers can be entirely eliminated. Let us consider recording or broadcast consoles.

The only place experience dictates us we should use transformers for isolation and balanced lines is on all inputs and outputs from the console. (Except for the direct monitor speaker lines not going through the patch bays). All the rest of the circuits within the console can be unbalanced and there is no other reason than having access to parts of the circuitry balanced, to do otherwise (FIGURE 4).

I doubt very much that you will find any patch bays used for access to the individual components within the system in recently designed consoles. Most of the circuits use switches rather than patch bays, eliminating the need for transformers.

It is sad to note that there has been no successful attempt made over the years to improve transformers except for some minor improvements in alloys for laminations. While other components such as capacitors, resistors, switches and transducers have undergone radical changes in their designs and performance, transformers from today are little better than from the 1930's. There are no balanced mic transformers of quality readily available off the shelf which do not have to rely on triple magnetic shields to eliminate hum pickup, rather than having their both halves of primary and secondary windings physically positioned out-of-phase so that all external fields induced into the transformer would cancel out (FIGURE 5).

Foreign countries use such transformers extensively. We don't. Nor do we have any transformers which have their coils completely surrounded with laminations acting as a magnetic shield? Nor do we have laminations, without the gap (except for expensive toroids). It looks like we have to do some catching up with the rest of the world. Perhaps we should learn to be more perceptive to better ideas, designs, and practices.

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STUDIO SOUND—Europe's leading professional magazine. Back issues available from June, '73. $1 each, postpaid. **3P Recording, P.O. 99569, San Francisco, Ca. 94109.**

PROFESSIONAL CASSETTE DUPLICATION, any length, stereo or mono.. any master acceptable.. Dolby. **Cantor Productions, 26 W. Nottingham Rd., Dayton, Ohio 45405. (513) 6571.**

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MODERN RECORDING TECHNIQUES by Robert E. Runstein. The only book covering all aspects of multi-track pop music recording from microphones through disc cutting. For engineers, producers, and musicians. $9.95 prepaid. **Robert E. Runstein, 44 Dinsmore Ave. Apt. 610, Framingham, Mass. 01701.**

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WANTED: Educational public broadcast station needs surplus studio equipment, particularly magnetic tape and mixing equipment. Contact **Boulder Community Broadcast Association, 885 Arapahoe, Boulder, Colorado 80302.**

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Copies of all issues of **db—The Sound Engineering Magazine** starting with the November 1967 issue are now available on 35 mm. microfilm. For further information or to place your order please write directly to: **University Microfilm, Inc.
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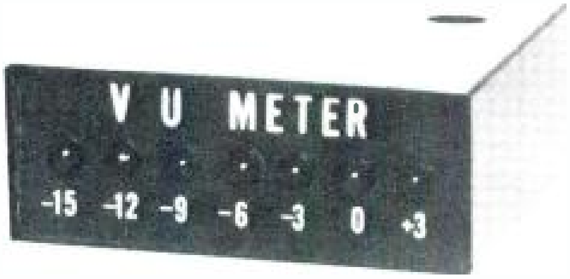

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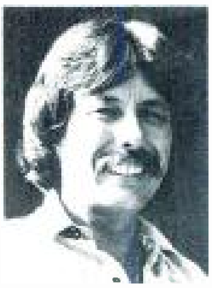

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• **Michael J. Noakes** has been named national sales manager in the United States and Canada by **Revox Corp.** of Syosset, N.Y. Mr. Noakes had previously served as national sales manager for the firm in the United Kingdom. His responsibilities will include Revox sales in the United States and Bib products in the U.S. and Canada.

• **Russ Jones** has joined **Oberheim Electronics** of Santa Monica, California as a director of the corporation and vice-president of marketing. Mr. Jones will set up domestic and international distribution for the product line. **He** comes to Oberheim from **Acoustic Control.**

• Exclusive distribution rights to international markets, excluding Brazil, has been granted by **Audio Designs and Manufacturing, Inc.** to the **Ampex Corporation**, of Redwood City, California. The agreement is for a two-year period. ADM will continue to market its products in the United States from its Detroit facility.

• **Sudden Rush Music**, 750 Kappock Street, Bronx, N.Y. 10463, is now releasing a regular monthly column of songwriter leads designed to fill requests for material coming in from various sources. The column is being published in **Musicians Classified,** a newspaper which provides New York area musicians with specific leads for employment and other relevant information.

• An example of the ultimate ambition in studios has recently opened at the **Casino Montreux** in Montreux, Switzerland, focalized by **Alex Grob.** The facility has a control room and studio interfaced with a full-sized concert hall capable of holding a symphony orchestra. They use a Neve desk with 32 inputs and 24 outputs, two Studer 16/24-track tape machines, one Studer 4-track and two Studer 2-track recorders, Dolby 24-track noise reduction, two Tannoy monitors and 4 Westlake monitors. The quad control room was designed and built by **Westlake Audio.** In addition to Mr. Grob, the management staff consists of **Anita Kerr** and **John Timperley.** The interfacing was done by **Mercury Electronics** and **Scenic Sounds** of London, England.

• Two new appointments have been announced by the **Cetec Corporation** of N. Hollywood, Ca. **Jules L. Sack** has been named as director of marketing and sales and **Wesley M. Fujii** as vice-president and general manager of the Cetec Audio Division. Mr. Fujii comes to Cetec from **Electro Sound, Inc.** Mr. Sack was formerly associated with **Stereodyne, Inc.**

• A new position, that of national sales manager, professional products, at **Capitol Magnetic Products** of Los Angeles, will be filled by **Larry C. Hockemeyer.** Mr. Hockemeyer will be responsible for all domestic sales of studio and duplicator products, as well as the educational cassette line. Mr. Hockemeyer was previously with **Data Packaging Corp.**

• Four major promotions at **James B. Lansing Sound** of Los Angeles have been announced. **Arnold Wolf,** president of JBL since 1970 has been promoted to the position of chairman of the board. Mr. Wolf has been replaced as president by **Sterling Sander. I. B. Stern** takes on the job of vice president of Harman International, expanding his role to include U.S. marketing for JBL, Harman/Kardon, and Tannoy. **Rod Bell** has moved up to the post of vice president, marketing for the professional, consumer, and export divisions.

• Convention sites through 1980 have been selected by the National Association of Broadcasters. Next year's confab will be in Chicago, March 21-24. In 1977, the convention is scheduled for Washington, D.C., 1978, Las Vegas, Dallas in 1979, and New Orleans projected for 1980. Other NAB news includes several elections. **Wilson C. Wearn,** president of **Multimedia Broadcasting Co.,** Greenville, S.C. has been unanimously elected as chairman of the Board of Directors. **Harold R. Krelstein,** board chairman of the **Plough Broadcasting Co.,** Memphis, Tenn., was re-elected chairman of the radio board of directors, with **V. Kay Melia,** general manager of **Station KLOE,** Goodland, Kansas, elected vice-chairman. **Walter E. Bartlett,** senior vice president for television of the **AVCO Broadcasting Co.,** Cincinnati, Ohio, was elected chairman of the television board of directors and **Robert D. Gordon,** vice-president and general manager of **Station WCPO-TV** of Cincinnati as vice-chairman of the t.v. board.

• **TRW,** of Los Angeles, has announced four new organizational changes. **James J. Jensen** has been appointed to a newly created Electronic Components Division management position, focusing on the independent telephone market. **Karl E. Heller** and **Richard E. Sostek** will be reassigned to functions relating to domestic operations. Mr. Heller will be responsible for active and passive components and Mr. Sostek responsible for electromechanical devices. In the same division, **James J. Hermacinski** has been promoted from the position of national sales manager for TRW/Cinch divisions to the new position of ECD director, field sales operations.

• Two sales management personnel changes have been made at Switchcraft, Inc., Chicago. **Scotty Wallace** has been named eastern regional sales manger. **Wally Wheaton** has taken Mr. Wallace's position as district sales manager for the Chicago area. Mr. Wallace was associated with TAB Books before joining Switchcraft in 1973. Mr. Wheaton was at one time associated with **International Register Co. and Shure Bros.**

• **Christine M. Davis,** recently released from active duty as an Air Force First Lt., has joined **Sparta Electronics,** of Sacramento, Ca., as personnel and administrative supervisor. Also at Sparta, **Rigoverto Felix** was promoted to the post of customer service manager. He had been serving as a checkout supervisor in transmitter engineering.


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