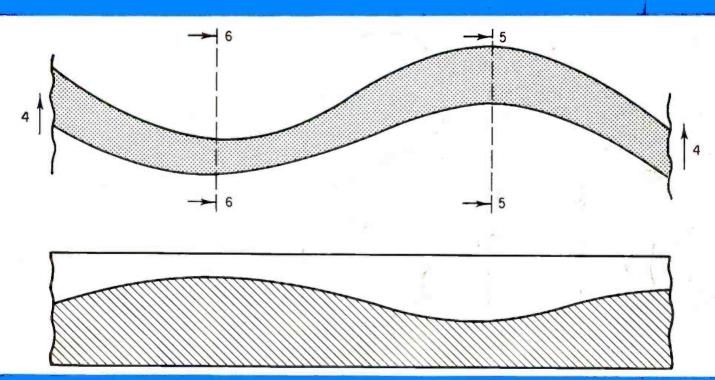


THE SOUND ENGINEERING MAGAZINE JULY/AUGUST 1968 75c

The Compatible Disc:
Playback
Recording
Tracing Distortion



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Coming Next Month

In September—Melvin C. Sprinkle has prepared a definitive study of Frequency Response Measurement. If you think you know all there is to know about this important aspect of audio, you will be subject to some awakenings.

Due last month but crowded out for lack of space, David Hancock's article on a new ultra-high-quality ribbon microphone will appear. Mr. Hancock makes a case for the continued belief in ribbon mics as a means to highest quality reproduction.

Electronic Video Recordings—rumor runs wild on this new system proposal that is expected to revolutionize audio/visual communications. Edward Tatnall Canby has prepared a report that brings things into focus—consider it a state-of-the-rumors report.

And there will be our regular monthly columnists: George Alexandrovich, John A. McCulloch, Norman H. Crowhurst, and Martin Dickstein.

Next month in db, the Sound Engineering Magazine.

About the Cover

This patent drawing depicts a groove with lateral and vertical modulation—certainly in keeping with modern stereo cutting practice. The patent date is 1918! See John J. Bubbers article beginning on page 16.



July/August 1968 • Volume 2, Number 7

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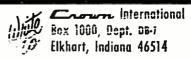
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_etters

The Editor:

I just received your May 1968 issue and read with interest Martin Dickstein's report on rear projection (in his column Sound with IMAGES). We are in the business of providing audio-visual consultation, design, and engineering services to architects and others who are planning space for audio-visual communication facilities. We happen to be strong proponents of the rear-projection technique, having designed numerous systems with up to twenty-foot wide glass permanently installed r.p. screens, and are thus well aware of the advantages and limitations so well expressed in Mr. Dickstein's column.

We would like to add one fact, and that is the phenomenon that allows the eye to function more efficiently in a lighted room, and thus focus on smaller images. This, therefore, allows for the use of smaller screens and image sizes. . . thus the set of ground rules for image size and viewing distance as applied to rear projection do not match the old standards for front projection (nominally maximum viewing distance to be 6W or 6 times width of screen). With properly designed r.p. systems 8W to 12W standards can be employed making for more efficient use of the space, better screen illumination, and permitting more ambient light in audience areas for the obvious advantages this will provide.

> Hal Guzofsky Audio Visual Consultants Denver. Colorado

more on Symbol Standards

The Editor:

In reference to your inquiry about caps and lower case letters on electrical symbols—it has always been a great nuisance to have caps in symbols. I doubt if many engineers when writing notes for their own use, ever use anything but lower case, such as **db**, **cps** (or **hz**) etc. Down with caps!

Bernard J. Koetting KSGM AM-FM Perryville, Mo.

The Edition

In res, mse to your April Editorial, my preferences are: Frequency—cycles per mad; other units—lower case letters

irnin Pa.

The Editor:

I welcome the opportunity to make some comments regarding audio terminology. As yet, I have no quarrel with the terms you listed. I would, however, like to make some suggestions on some other ones.

I believe that ad writers, and others, over use the prefix pre. One may buy blank discs, package discs with music upon, blank tape, and tape already recorded. Since we do not call a phono album a pre-recorded disc, why call tape music pre-recorded? Why are kit speaker cabinets listed as: pre-cut, pre-drilled, and pre-sanded? If they were cut, drilled, and sanded would it not be the same?

I would also like to see serious audio publications go back to the terms agreed upon by most engineers: *Monaural* for single-channel headphones; *binaural* for two-channel phones; *monophonic* for single-channel speakers; and *sterco-phonic* for two-channel speakers (naturally I include the entire audio chain).

There should be a limit to the use of the word professional, or professional type. This should be limited to equipment that would ordinarily be used by professional studios. True, some studios may use lesser equipment where utmost quality is not important, but you could hardly call a Wilcox-Gay disc cutter a professional piece of gear as compared to a Studer or Scully lathe.

Robert F. McDonald Temple Records Lafayette, California

Mr. McDonald has many good points to make. Editorially, we do not allow the use of the phrase pre-recorded when what is meant is commercially-recorded tapes. A tape is blank (or perhaps virgin), it might be recorded, and it may carry commercially-recorded information. But pre-recorded? Never.

We don't quite see cye to cye with Mr. McDonald on the mono/stereo nomenclature. We prefer to use mono (or monophonic) for single-channel material, stereo (or stereophonic) for two-channel material. More than two channels are simply identified as three-channel, eight-channel, twenty-four-channel, or what have you.

As for the use of the label professional we can only lament the mis-use of the word just as we lament the abuse of the phrase high fidelity. We believe in the real meanings of words; to us, professional refers to a product that is used by professionals. It may be a product that has other value as well, but it must have a primary purpose in the profession for it to be labelled as such. Thus, as an example, Audio Devices' 1251 tape is professional tape even though we venture to guess that more of it is sold to the hi-fi consumer, than strictly to the trade. Ed.

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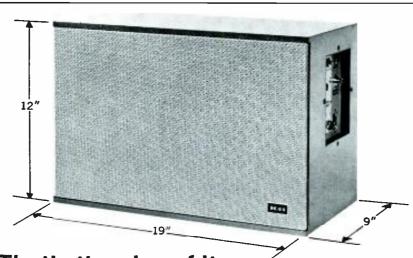
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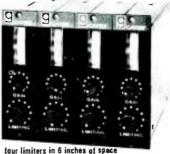
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Feedback Loop

JOHN A. McCULLOCH

• While in Las Vegas on a recent business trip. I had the opportunity to talk with Bill Porter of United Recording of Nevada. Along the way I had learned that one of his recent efforts in the pop field had been marked as being made from an exceptionally well cut master lacquer. In the course of conversation this subject came up, and I asked Bill about his method of disc mastering. His reply was, "I work with the same materials and processes as anyone else in the industry, but I pay particular attention to the alignment of my equipment and its operating levels. If each step is carefully done the resulting master must be good."

Let's look at the process, and the areas wherein we might expect trouble. This review will be nothing spectacularly new, but might assist those to whom disc mastering is not an every day occurrence. Also, as this is just a general review, we shall limit the discussion to the mono system; ignoring for the most part special requirements of stereo recording.

One of the first requirements for obtaining a good recording on disc is that some means of reproducing the cut be available. Because the quality of the cut will be judged by this playback, it must be reproduced on a calibrated system of the highest order. Just as in magnetic tape recording there is a standard alignment tape, disc recording has a standard playback disc. Recently the relative level in cutting, referred to as the zero level, has been changed. A new standard disc available from the NAB supplants the earlier RCA standard. However the frequency response tests on the mono RCA 12-5-49 disc are still very valid, and may be used to check out the response of your playback system.

As the "new" zero level is based upon 7 cm./sec. peak recorded velocity, and the zero level on the RCA disc is based upon 5.5 cm./sec. p.r.v., there is a difference of slightly more than 2 dB between the two zeros. Using the RCA disc, it is possible to calibrate a system (within a few tenths of a dB) by setting the RCA zero to -2.1 dB on the

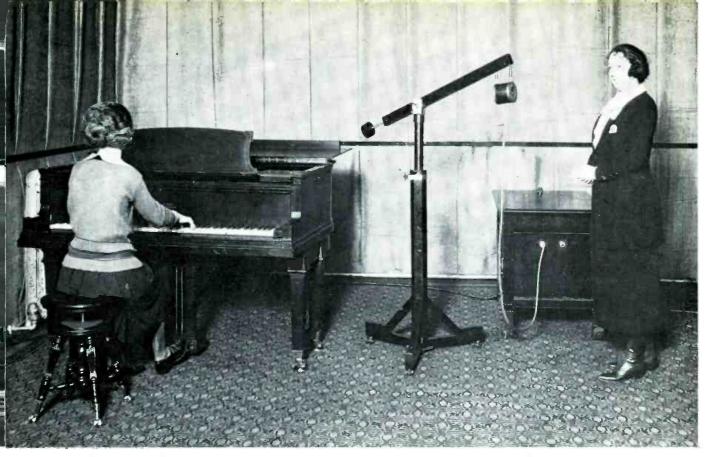
calibrating meter. The zero on the calibrating meter now should represent the zero set by the 7 cm./sec. p.r.v. reference.

It is preferred that a playback equalizer having variable high- and lowfrequency equalization points be used, and aligned with the test record in a manner similar to the alignment of the normal tape reproduction system. Thus, theoretically, we now have a flat playback system. Practicality enters the picture and the relative quality of the reproducing cartridge and stylus impose a limit upon the system as to just how flat we may make the reproduction of the test tones. In general use, if the system is within a dB or two of the ideal curve in the range of 100 Hz to 10kHz, the system is acceptable. Each cartridge and stylus change used will require a different calibration to maintain optimum results.

As in most mechanical systems there may be a point of resonance which places a sharp hole in the response. Such holes may be tolerated if they are no more than 5 to 10 per cent of the center frequency in width. However, a change of cartridge and/or stylus might be suggested to try and remove this hole.

The elliptical stylus is suggested as it more nearly conforms to the cutting stylus. For disc to tape transfer, an increase of about 20 per cent over the normal stylus force may improve the tracking of the disc. At all times the angle of the stylus must conform to the specifications of the particular recording being played back. (Monophonic is approximately 90° and stereophonic discs are 105° as measured from the record surface.) The exact vertical alignment of the stylus in the groove, as seen from the end position of the arm is also important.

Calibrating the recording system is more difficult, but a simple procedure might be as follows: Take an oscillator of low distortion and feed signals of selected frequencies into the recording system at a reduced level (typically -14dB for the 100Hz to 10kHz spectrum). Keep this level constant into



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Frequency Response: 20-20,000 Hz (± 2.5 db 30-18,000 Hz). Directional Characteristics: Uni-directional cardioid (axis variable from 0° to 90°). Output Impedance: 50, 250 or 600 ohms balanced. Output Level: —50 db @ 250 ohms where 0 db = 1 volt/10 microbar. Noise Level: 24 db SL where 0 db = 2 x 10-4 microbar. Dynamic Range: 110 db.

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the system, and record a disc starting with the higher frequencies downwards in steps. With a parallel light source focused across the disc, the grooves should form a Christmas tree pattern, with the high frequencies forming the base of the tree at the outer edge of the disc. For proper results it is important to keep the test at the outside diameters of the disc, therefore a new disc should be used for each test. If the frequency steps were selected to fall on the RIAA record equalization curve at 2dB intervals the pattern should be quite evident. Any deviation from these regular steps means a discrepancy in proper recording.

A less accurate method for testing the newly recorded material is to playback the tones on the calibrated playback system. Why less accurate? As mentioned hefore when we were calibrating the playback system, a response within a dB or two of the RIAA curve was acceptable, but with the light system, deviations of 1dB may be measured, A change in length of the wave pattern displayed by the light reflections by a factor of 2 (twice as long, or half the length) is equal to a change in level of 6dB. Again, the zero level corresponds to 7 cm./sec. p.r.v. At 33 r.p.m. a display of 1.6 inches is zero, and at 45 r.p.m. it is 1.2 inches. (To find the relative level multiply the display length by 4.4 for 33 r.p.m., and 6.0 for 45 r.p.m., and compare with the zero level. Both factors are now in cm./sec. p.r.v.

If the recording is not flat, check the drive amplifier for frequency response by disabling the recording equalization circuitry, and cutting a disc. This time the light display should be even for all frequencies, and errors introduced by the amplifier or the cutting head may be discerned. Should this pattern be smooth and straight, then the proper recording equalization is not being supplied, and that section of the system should be repaired or substituted.

If the pattern is uneven, then check the driver amplifier by substituting the load and external feedback circuitry required as given by the manufacturer. Check all normal characteristics of the amplifier. In particular make sure that the amplifier is capable of delivering its full rated output power, as the recording equalization makes very heavy demands upon the amplifier when high frequencies are to be recorded. 10kHz requires a boost of 13.7 dB—almost twenty-four times the power required at 1kHz.

In checking the head, look for shorted or partially shorted coils, poor connections, and shorts to ground. In the mechanical sense check for misalignment of the stylus holder with the armature or its associated mechanism. The stylus should be seated firmly in the bottom of the holder, but do not force it or chip

the cutting edges. The stylus must be sharp and clean, and look like a chisel to the record surface. (typically 2° or 3° from vertical in monophonic cutting) The stylus should face normally to the rotation of the disc, and should be from a reputable manufacturer. Check that the stylus, as viewed in line with the grooves, is vertical with respect to the record's surface. (Assuming a smooth and flat disc!)

If the system is still deviating from the desired response (RIAA in most cases) after all checks have been made, correction to the driver amplifier's recording equalization, or addition of passive equalization before the system, should be considered.

The groove depth and recorded level seems to be the most widely varying factors in recordings I purchase, or have shown to me. Bill Porter reports using an unmodulated groove width of $2\frac{1}{4}$ to. 3 mils for lps, and 4 to 5 mils for 45s This is in line with the recommended practices, and most processors are in this range. Assuming the use of a calibrated microscope it is easy to set these widths, thus determining the depth of cut. If your microscope is not calibrated, look at the lead in, banding, or lead out grooves on a high-quality commercial pressing for an approximate calibration. Compare several!

I have heard of cutting with levels as high as +14dB on 45s and +10dB on lp. Whether this was using the new standards or the older was not referenced, but even so, Mr. Porter recommends and uses +3dB or +4dB for his lps, and +6dB to +7dB for 45s. On some lps zero to +2dB is used consistent with program frequency content and the length of the material. The low frequencies give the greatest excursion, and while it is possible and practical to cut extreme levels in high frequencies, the playback stylus will not track these frequencies to the extent that they may be cut into the original recording. The result is blamed as a poor cutting.

Bill Robinson formerly of Capitol Records reports using lp levels of OdB. and 45 levels of +6dB. His zero reference is also referred to 7 cm./sec. p.r.v.

As an assist in maintaining top quality in the cutting, particularly where super power amplifiers (200 watt variety) are not used in the driving of the cutting head, keep the recorded material as far outward on the surface of the disc (commensurate with length and other factors) as possible. This will avoid the use, or excessive use of innerdiameter equalization. Of course groove spacing, groove depth and width, level of cutting, and equalization are all interrelated factors in determining the final quality of the disc. Some areas must give a bit that more may be gained in another area, but that is left as the mastering engineer's decision.

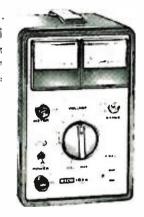
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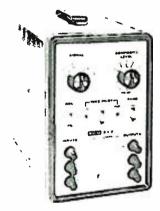
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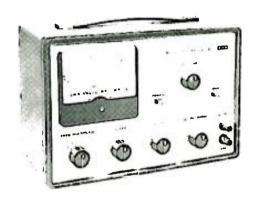
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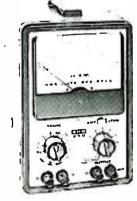
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db July/August 1968

The Audio ngineer's Handbook

GEORGE ALEXANDROVICH

• In past months I have discussed system operation, alignment procedures, testing, and maintenance. It seems appropriate at this time to discuss a subject that is related to almost every facet of audio and every branch of electronics-switching.

It is hard to imagine any practical electric circuit without some sort of a switch, be it in the form of an on/off switch or any other device acting as a gate for current flow.

It is generally understood that the word switch describes a device with mechanical contacts and a mechanical others.

In the field of moving-contact switches, new devices called reed switches (actuated by a magnetic field) have found great popularity. Just as with the solidstate switch, this new type offers extreme long life with high reliability

activator designed to control current in a circuit. Present-day technology offers us a long line of new devices capable of performing switching functions. Many are solid-state such as transistors. silicon-controlled rectifiers (scr's), photodiodes, light-sensitive resistors, voltage-dependent resistors (vdr's), and

DUMMY LOAD SPEAKER (B) (A) RIGHT CH. LEFT CH RM PROGRAM CHANNEL LEFT RM PROGRAM CHANNEL RIGHT PROGRAM FEED RM ECHO CHANNEL LEFT ECHO FEED RM ECHO CHANNEL RIGHT (D)

Figure 1. (A) single-pole, single-throw power switch; (B) double-pole, single-throw power switch; (C) single-pole, double-throw switch used to switch a speaker; and (D) four-pole, triple-throw key for the delegation of mic-channel signals in a stereo mixing console.

even under unfavorable environmental conditions.

In the computer field, solid-state devices (flip-flops) have almost forced mechanical switches out of the field. There is still a dependence on basic mechanical manually-activated switches to control the more sophisticated circuits. It is possible to construct systems using solid-state switching exclusively, but cost (for one thing) makes such systems impractical-for the present at least.

MECHANICAL SWITCHES

Basically, a mechanical switch consists of two or more contact surfaces, some of which are mounted on flexible or movable conductors. The physical form of the contact structure and activator mechanism classifies the switch. The usual classifications include rotaries, toggles, keys, push-action switches, sliders, and rockers.

The time to consider the many other factors in a switch besides its losses or convenience of operation is when new equipment or circuits are being designed. Before a switch is specified a few basic facts about the circuit in which it will operate must be established.

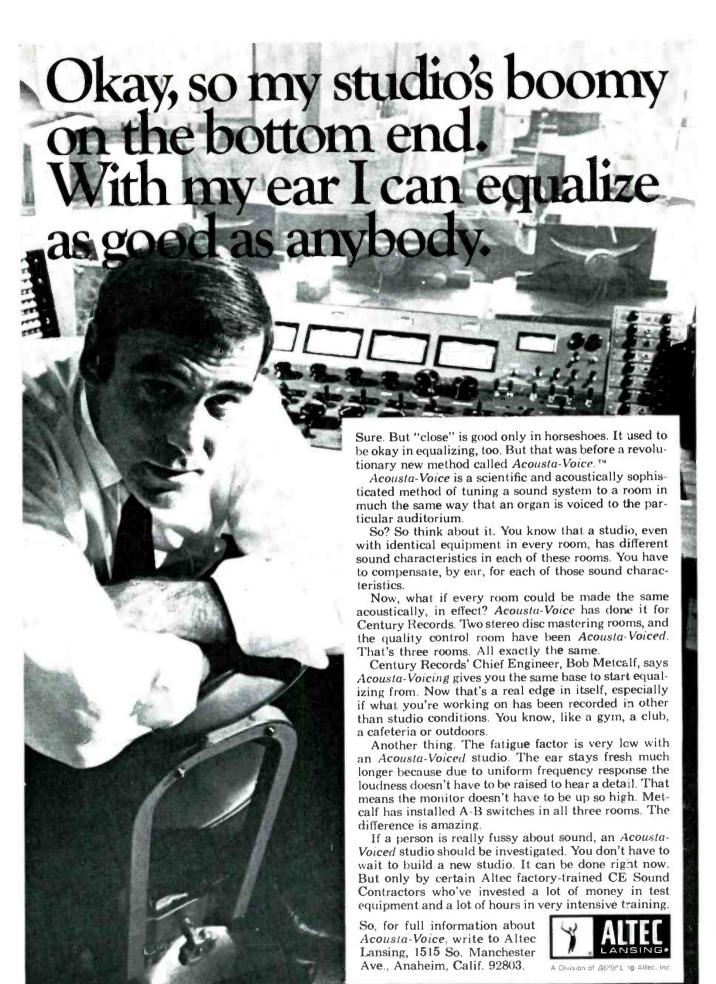
The fundamental factors to consider are-contact configuration; current carrying capacity; maximum circuit voltage; contact life, space and environmental conditions. Also consider additional special requirements such as low mechanical noise during switching, illuminated activators, and ease of maintenance.

Usually, after completing specification listing it will be found that the search for a desired item has narrowed to one or two types, preferably readily available from the market. (Remember that the more complex switch contacts become, the less chance there will be to find such a product off-the-shelf. There are many occasions when practical compromises must be made in the interests of reasonable economy and availability.)

Subdivide all switches used in audio into categories by function. You will end up with:

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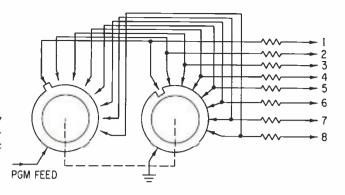


Figure 2. A rotary switch delegating a signal to one of eight busses.

Power switches. They turn equipment on and off. Usually they are single- or double-pole, single-throw switches. Contacts for voltages of 120 a.c. minimum are available in current-handling capabilities for most equipment.

High power audio switches. These switch speaker circuits, cutterhead lines, etc. The switch must have contacts rated for heavy currents with voltage ratings up to 100 or more volts if it is to be used to switch 70.7-volt lines. Contacts may have to be double-throw type to switch in dummy loads in place of the normal load.

Switches for moderate audio levels. These are used to control linelevel signals consisting of several mA of current. Contact resistance should be moderately low along with high reliability factors, since much line-level switching is done frequently. Voltage and current ratings need not be high. Contact configurations may range from simple single-throw, single-pole (spst) contacts to multi-pole, multi-position switches. A good example of such a complex switch would be one that is used in switching a mic channels to different programs and echo busses. Aside from delegating program and echo signals into one of the available channels, dummy loads are switched into the rest of the channels to maintain constant the impedance of the mixing network.

Microphone level switching. This requires well-shielded, low contact-resistance switches. In most cases, two-pole types will be used because mic lines are usually balanced (except when the mic is located next to the preamp), so both sides of the line should be switched simultaneously. With the present-day practice of supplying power to condenser mics through the audio lines, one should be extremely careful in designing mic-level switching or patching. A presence of d.c. on the line will inevitably cause problems if design and installation care is not taken.

In FIGURE 2 a sample of a rotary switch is shown. This is designed to delegate a signal into one of eight

mixing busses when the signal source impedance is very low (and can be substituted by a short to ground). If echo signal is also to be switched, an additional section must be added.

If the source impedance is 150 or 600 ohms a substituted dummy load may have to be a resistor equal to the source impedance. In this case switching will be more complicated, requiring additional sections on a rotary switch. For simplification, a push switch such as is shown in Figure 3 could be used.

POWER SWITCHING

I often wonder why most professionals like to use heavy-duty expensive switches yet the moment they hear a click in the audio lines when switching power circuits they suspect the power switch first. There is nothing wrong with using the best switches available with

safety factors ranging well beyond requirements. But there is a danger of not providing proper operating conditions for the switch so that there is premature contact failure or, at least, undesirable side effects due to contact deterioration. The best power switches offer reliable mechanical construction, good contact materials, and longer life—under the condition that elementary arc and noise-suppression circuits are being used across the switch.

Since we deal with extremely low noise levels in audio, we are concerned with any source of disturbance affecting the signal-to-noise ratio of the system. It needs reminding that unless audio circuits in the vicinity of a power line are properly shielded or grounded, noise will still be detected due to line transients. This is true even if careful suppression has been applied to the switch.

Each time the contacts of a switch begin to open, contact resistance increases. This increase produces an increased voltage drop with a consequent temperature rise. When switching heavy currents, contact temperature can rise to where the metal surfaces in actual contact begin to vaporize. This vaporization, together with ionized air between the contacts provides the path for current—creating an arc.

When a power switch is turned on, contact bounce caused by momentary separation between the contacts after they hit, will also cause interference.

Yet another potential for problems is any reactive element in the current

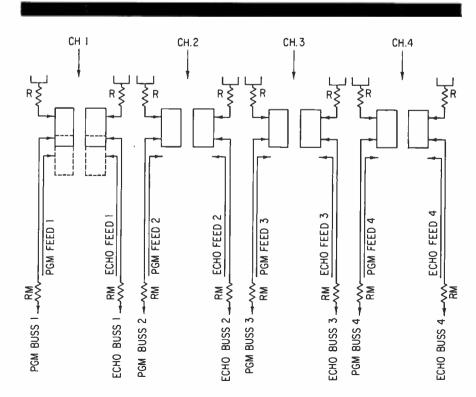


Figure 3. A four-station, push switch, interlocked dpdt for a four-channel console capable of switching program and echo simultaneously.

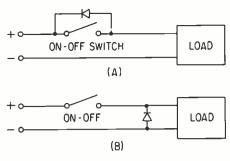


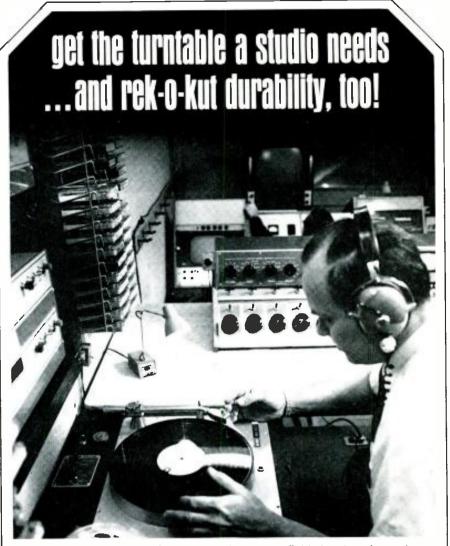
Figure 4. At (A) current is diverted through the source. If this is undesirable the circuit at (B) isolates this part of the circuit. Note that in both cases, the diode's cathode faces the positive (+) side of the voltage.

path being switched. Stored electrical energy can be discharged across the switch contacts during the time the switch is turned off. The simplest method for diverting this discharge current created by inductive elements is by placing a capacitor across the switch. However, a capacitor left alone with the inductance of the circuit (for instance, an electric motor or transformer) will produce dying oscillations at a frequency determined by the values of inductance of the load and the capacitor across the switch.

A resistor inserted in series with the capacitor will reduce the Q of the circuit—cutting down on the amplitude of the oscillations. Just remember that you can minimize oscillations; you can never eliminate them completely. Usually, the values for the capacitor and resistor are determined experimentally, but generally a resistor that approximates the impedance value of the load will generally do the job.

D.C. SWITCHING

With d.c., different methods of arc and transient suppression are used. A diode will usually solve the problem. Connecting a diode with its cathode facing the positive pole will short out secondary currents due to circuit inductance. By placing the diode across the switch (FIGURE 4(A)) we depend on diverting the current through the source. If it is undesirable to pass transients through the source, which may be tied to other audio circuits, the diode can be connected across the load, isolating this part of the circuitry (FIGURE 4(B)). In either case, the diode should have a piv rating equal to, or exceeding, the source voltage. Peak current ratings should not be less than expected inductive currents. Fortunately, we are concerned only with single peak currents-thus qualifying most of the small silicon diodes with low continuous current rating but high single peak ratings.



There's hardly an engineer in the broadcast field that hasn't used a Rek-O-Kut turntable in his career. Rek-O-Kut has been building studio turntables for over a quarter century. So you are assured of buying top quality sound reproduction along with the ruggedness and durability that boradcast and commercial installations demand. The rim drive Rek-O-Kut B-12H by Koss Electronics permits slip cueing without sacrificing fidelity. And your Koss/Rek-O-Kut will last and last and last with a minimum of maintenance and repair. Write for complete details on the popular Model B-12H or the 16" studio B-16H today.

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Theory and Practice

NORMAN H. CROWHURST

• A reader from Fort Wayne, Indiana writes about something that has puzzled him for years, and his question fits right into this column. While he says he hasn't A-B-ed public-address amplifiers that do and do not use lashings of feedback, he has the impression that those with much feedback may produce the acoustic variety more readily than those without feedback. I imagine he's not alone in this impression.

So he formalizes his question, "Does the incorporation of large amounts of inverse feedback in public address amplifiers degrade the amount of acoustic gain and phase margin that can be used in practice?" To explain what he thinks might happen, he further questions, "Does the feedback internal to the amplifier in effect 'use up' some of the external or acoustic gain and phase margin and thereby limit the usable gain?"

Before I answer that question, I'll deal with an earlier one, also related to the effect of feedback, although the reasons are quite different — at least, I think they are. This case happened some years ago, and I was called in professionally because theory and practice didn't seem to agree.

The caller was a manufacturer of guitar and other musical instrument amplifiers. What prompted the call was a rather embarrassing situation in which a customer had been A-B-ing a couple of his products. At the time, he made a 15-watt amplifier with no feedback, and a 50-watt amplifier with plenty of feedback; of course, at a higher price.

The customer had just wanted convincing that the 50-watt unit was worth the extra price. So my client rigged up an A-B comparison on the spot, convinced that the greater loudness of the 50-watt unit would give a conclusive demonstration.

Instead, to his embarrassment, there was no doubt about it, the 15-watt amplifier appeared to give much more power. Being a musician himself, he knew he couldn't convince his customer

the 50-watter was really giving more watts. The musician believes his ears!

By the time I arrived on the scene, he'd put the two amplifiers on the bench and checked their output ratings, suspecting something "screwy." But they both performed according to specifications. The 15-watt unit wouldn't give anywhere near 50 watts, with any amount of distortion, and the 50-watt unit gave its rated power with ease.

He'd measured this into a resistive load, so he thought maybe using the rather complicated reactive load provided by a loudspeaker might cause the difference. And it's true, this could have been the explanation. So they put earplugs in their ears, to enable them to bear the steady tone at 15 and 50 watts, and turned up the power on each amplifier in turn.

The results with the loudspeaker weren't significantly different from the dummy load, apart from the fact (obvious only to engineering students!) that both amplifiers were now delivering a reactive VA that may not have amounted to real live rated watts.

So what was making the difference? The answer to that question was related to the reason a guitar amp needs so much power. The average output from a guitar amp is probably not more than 2 to 5 watts, even when you use a big amplifier. That's why you need earplugs to stand even 15 watts of steady tone. But those pluck tones momentarily require a real surge of power that goes over the top.

The 15-watt, non-feedback amplifier allowed the pluck tones to go over the top, just chopping them off, and then played the rest of the note normally (Figure 1). By listening carefully, you could hear it chopping, but it was so short, and not too different from the pluck sound anyway, so that sound was really quite passable.

On the other hand, the 50-watt job used class AB. Up to 50 watts, the output was lovely. It would even give 60 watts, with not too much distortion, on a steady tone. But a little beyond that

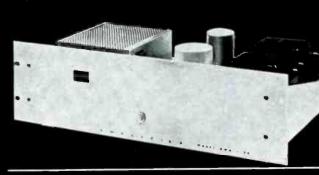
db July/August 1968

"Name three reasons why BOZAK Sound Systems are preferred wherever quality counts"



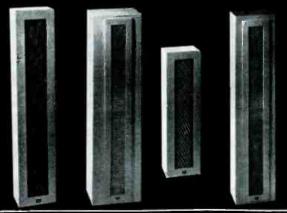
"1.BOZAK Mixers"

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it suddenly fell apart.

The output stage reached saturation at this point, and feedback momentarily disappeared so that the input level, where feedback should have offset it, was suddenly some 26 dB higher than usual. This really clobbered the input stages, biasing them to cut-off, from which they recovered only after a second or so.

So on guitar music, before the gain was as high on this amplifier as it was on the 15-watter while still sounding good, the pluck sound momentarily pushed the 50-watter into saturation. It didn't recover for almost a second, and as it recovered, it rectified signal that the 15-watter reproduced undistorted (FIGURE 2)

Turning the gain down just a whisker, so the pluck stays below the 60 watt level (120 watt peak) this amplifier reproduces undistorted. But its level is then less than half that in Figure 1. This explained the impression.

Of course, this led to a redesign of this manufacturer's 50-watt unit. But the point here is that practice didn't appear to agree with theory or measurement — until the reason was found.

Now to our reader's question. As anyone who has worked on p.a. knows, the acoustic feedback point can be quite critical. To sustain a howl, the gain must be just a fraction of a dB above the critical point. Of course, one frequency usually rings first, determined by a resonance in the microphone, the speaker system, the auditorium, or a combination of them all. And if one frequency doesn't ring, another soon will.

But as you slowly turn up the gain, before ringing starts, you can hear it being stimulated by certain syllables or frequencies in the program material. It will ring momentarily and quit. If the auditorium is a difficult one, you may even have to operate this close to ring-

ing, to make the volume loud enough for the majority of the audience to hear.

Now, of course, our theorists will already have answered our reader's question, by saying there is no connection between the two kinds of feedback. One is purely electrical, in which an amplifier is either stable or unstable as an entity, and won't be influenced by what the input or output signals consist of, whether these be program or acoustic feedback.

Which is true. And the internal feed-back doesn't *directly* affect the external, or acoustic feedback. So, unless you examine what happens a little more closely, theory assures us that a feedback amplifier can work just as close to acoustic feedback as a non-feedback amplifier, if both have the same over-all gain, frequency response, etc.

So what could make a difference? Just this: one thing internal feedback does is to hold the internal gain rock steady. The non-feedback amplifier may allow its gain to fluctuate a fraction of a dB during the waveform, particularly of louder signals, such as when the person at the microphone raises his voice. Did you ever notice how that can promote acoustic feedback?

Somewhere on the larger waveform, the gain is just a little higher than average, and this promotes ringing, particularly if, at that instant, the program also contains the ringing note. maybe as a sibilant. But the average gain is not quite enough to sustain an acoustic howl, so it dies away, and only shows when the person at the mic shouts certain words.

The amplifier with a lot of feedback isn't so tolerant. True it may not howl until something in the program sets it off. If you turn either kind of amplifier up in a quiet room, you may be able to get a little beyond the howl point before a howl actually begins.

This is because the howl needs a geometric standing-wave pattern to propagate, and it may take the introduction of the critical note in the program material to start the howl going. But once started, with a feedback amplifier, it refuses to stop until some definite step is taken, such as turning down the gain or moving the mic.

It would be difficult to correlate this as using up the margins, because the two margins have quite different nature. The electrical margins to maintain stability and a level frequency response are usually a long way from oscillating at any frequency. If they aren't, it won't take acoustic feedback to make the amplifier sound horrible!

The operating acoustic-gain margin is often quite small at certain frequencies. With reference to acoustic feedback, we can hardly refer to phase margin in the same sense. Of course, the fed back wave must reach the microphone in the correct phase to augment sound coming from the loudspeaker at the same instant.

But the travel time from loudspeaker to microphone is invariably several complete waves. In a feedback amplifier, even phase reversal can't occur, while the gain remains steady. Considerable attenuation must come before very much phase shift is allowed to occur.

With acoustic feedback, many wavelengths, complete revolutions of phase, occur in the sound wave. The higher the frequency, the greater the number of wavelengths from speaker to microphone.

In theory, the two forms of feedback are quite different and apparently unrelated, beyond the broadest possible concept of the word *feedback* itself. Yet an effect of electrical feedback on the internal working of an amplifier *can* make acoustic feedback more difficult to handle with that amplifier.

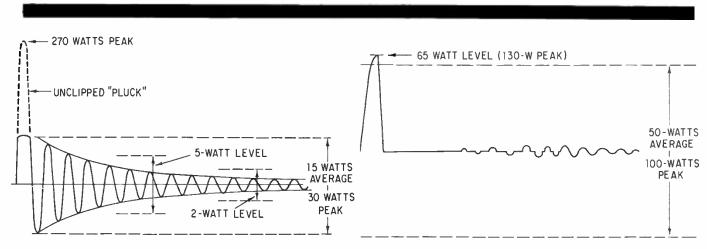


Figure 1. A 15-watt, non-feedback amplifier allows pluck tones to go over the top, just chopping them off, and then playing the rest of the note normally.

Figure 2. This 50-watt amplifier has been pushed into saturation. As it recovers it rectifies the signal.

Editorial

THE STEREO DISC has come of age. The necessary duplication of stereo and mono discs that has so confused the consumer market will soon give way to one record — the *compatible* stereo disc.

And then you will play a stereo disc even if you are broadcasting a mono signal.

The truly compatible disc does not (and may never) exist, but a mix down of two channels of information with minimal loss in quality can be achieved now. Success depends on the engineer's understanding of certain basics and his method of handling some inherent quality-limiting factors.

In this issue of **db** we offer several articles on the problems of compatible use. John Bubbers explains the necessity of using a proper cartridge — for either mono or stereo broadcasting. In Tracing Distortion, Arnold Schwartz details the problems created as the stylus attempts to trace the intricate paths placed by the cutter. But it is left to a recording engineer, David Greene, to pin down the problem — the disc itself.

Too many stereo discs marketed today cannot be satisfactorily played in mono. It is sometimes argued that these discs are made only to *sell*. And the engineer/artistic people involved have played with balances, phase, and gain in order to make what they consider a more saleable product. We certainly favor a record that can be merchandised, for unless the record sells, its artistic merits — whatever they may be — become lost. We do not agree, however, that a useful purpose is achieved if these alterations destroy the disc's mono playability.

There are many mono-only stations and there are many people who, for one reason or another, wish to play records via a single channel. Even in instances of stereo broadcasting, there will always be large numbers of mono listeners.

These groups must not be told that they simply have to settle for what they can get.

It is a clear responsibility of the industry to ensure that the final product will be as enjoyable to the mono listener as to the stereo listener. Certainly, there is every reason to expect that the sound from professionally cut stereo discs will be undistorted under any circumstances.

L.Z.

Compatible Disc Playback

JOHN J. BUBBERS

Many broadcasting stations are faced with the problem of playing stereophonic discs even though their transmission is mono. Good sound quality along with non-destructive play of these discs is possible.

PPROXIMATELY TEN YEARS after the introduction of the stereophonic disc the question of compatibility has once again come to occupy the advertising and promotional literature. The technical community has been somewhat less occupied with this problem, especially since the term has come to mean one of several concepts over the decade of stereo. The two most prevalent implied meanings are the ability to play stereo records with monophonic pickups (the vertical component having been modified in the recording process), and the ability to play a stereophonic disc monophonically without a change in the musical balance or intent. It is the purpose, here, to cover only the first implied meaning or use of the compatibility concept.

In the beginning, monophonic recordings presented a single set of problems in that the recorded information was limited to one plane of motion, either lateral or vertical. Certain obscure systems had been proposed from time to time to record at some intermediate angle, particularly in the early days of phonograph recording, since even then the battle waged over lateral *versus* vertical. The intermediate or skew angles were proposed as universal playback methods, but these suffered from the inherent inefficiency of the vectorial loss of output during the recovery of only the vector component, instead of the resultant present in the plane of mo-

tion. As a further limitation these devices were acoustical and lacked electronic amplification to overcome their low sensitivity; hence, were only an academic consideration and never a large factor in the development of large-scale home and studio devices.

The introduction of electrical recording cutters and the resultant possibility of stereophonic recording was most definitively outlined in a patent issued to Alan D. Blumlein in 1933 in England (and later in the United States in essentially the same form). While the earlier proponents of stereophonic recording had related the use of two channels to a vertical and lateral channel, Blumlein described in great detail the advantages of the 45/45 recording system since he further recognized the difficulty of maintaining uniform characteristics in a system that had two dissimilar channels. vertical and lateral, as their main vectors. The later Keller patent in the U.S. (1938) rounds out the basic principles of the stereophonic recording systems as we now know them. The effect of removing the low-frequency vertical component has been described in this early literature and in essence has only been rediscovered by the more contemporary recording practitioners. However, nowhere in the original teachings on stereophonic disc recording does the word compatibility appear. One can only assume that this concept has come about as a refinement, as in the case of monophonic playback where a balance change is effected, or as something not intended as in the case of pickups intended for lateral-only use. It is this latter problem which will be treated here.

BASIC MONO PICKUP DESIGN CONSIDERATIONS

The designer of monophonic pickups formerly was concerned with a moving systems which responded optimally in one plane, either lateral or vertical. His specific chore, then as

John J. Bubbers is vice president for field engineering at Pickering and Company of Plainview, New York. Prior to his joining Pickering in 1962 he had been a vice president and chief engineer of B'C Recording for nine years. From 1950 to 1953 he was chief engineer of radio station WLIB of New York City.

now, was to arrange the mechanical and electrical parameters to achieve uniform frequency response, low distortion, and a degree of ruggedness; all to permit field use of the device. The three parameters which the pickup designer could vary were mass, compliance, and resistance (this last-named frequently is called damping). Classical mechanics demonstrated that mass and compliance were frequency-sensitive functions, but while the mass was analogous to electrical inductance. compliance could be simulated by a capacitance, as a first consideration. The fact that certain geometric configurations whose weights were identical, but by arrangement of pivot points had different effective masses, was frequently used to the advantage of the designer for the most meaningful exploitation of the parameters within which he worked. The compliance was frequently changed to satisfy the requirements of the arm-mass-pickup compliance relationship to provide a pick-up-arm resonance which would not be subject to the periodic disturbances present in any normal playback system, such as disc eccentricity or normal warp. The damping, or resistive comonent of these pick-ups was the most elusive parameter, since it introduced mass and (unlike its electrical counterpart) was the most difficult to measure directly. The magnitude of damping required could however be determined by several classical means borrowed from the equivalent differential expressions or from electrical analogues.

The result of continued work over the years lead to many (Continued on page 21)

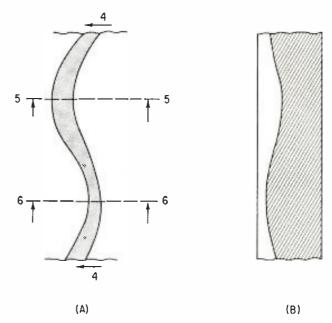


Figure 1. These illustrations are from U. S. Patent Number 1283903 (Reynard, November 5, 1918). They show an oblique-cut groove that allows reproduction to take place with a vertical- or lateral-response pickup. (A) shows the top view of the groove with the lateral component illustrated and (B) the corresponding vertical component.

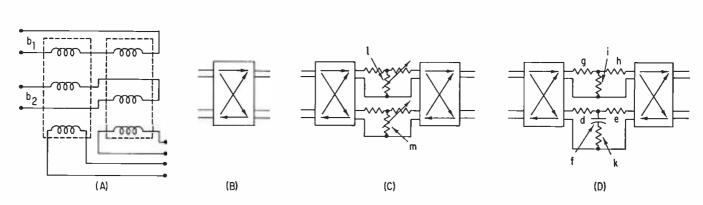
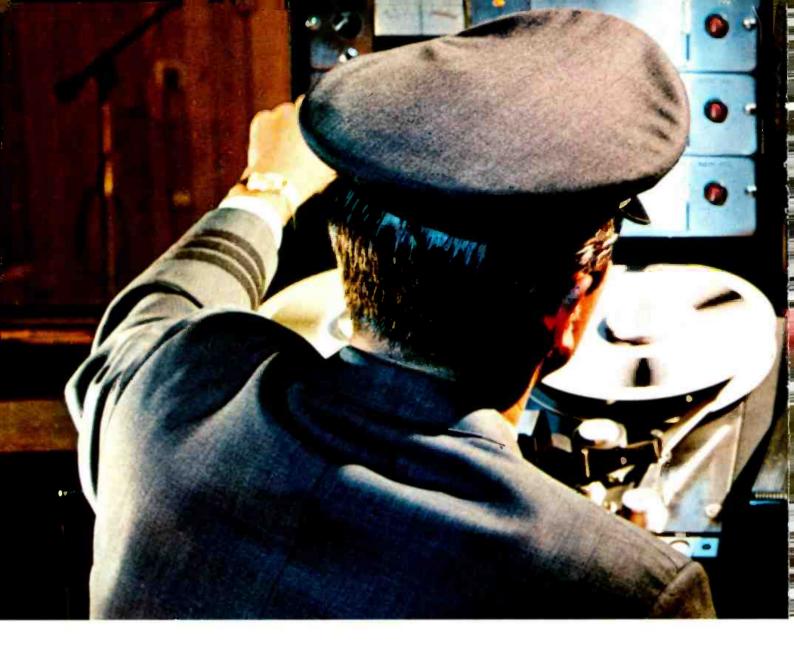


Figure 2. These are portions of British Patent Number 394325 (Alan D. Blumlein, 1933). The patent illustrates the various ways of making sum and difference channels for the purpose of converting a vertical-lateral system to 45/45. In (A) the area within the dashed lines is a common three-coil hybrid transformer; (B) is the diagramatic representation of (A); (C) shows a double matrix in series with a lossy gain control in each channel — its advantage lies in the fact that the sum or difference vectors can be modified by the gain controls and the resultant sum and difference rematrixed in the right-hand matrix; at (D) a frequency-sensitive gain control is indicated in the difference channel — vertical as derived from a 45/45 system — and it indicates the simplest kind of frequency-sensitive reduction of vertical component.



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excellent monophonic pickups, whose frequency response exceeded 20,000 Hz, with low distortion while being well damped. In this way no excessive resonant amplitude rises existed and hence the transient response was of equally high order. Important to this review is that the designs concerned themselves with monophonic reproduction, and in only some instances were these devices designed as universal pickups (vertical-lateral for broadcast use). With the freedom of having the motion limited to one plane, the designer frequently did not concern himself with the mechanical impedance in the unwanted plane of motion. In several designs the motion of the unwanted plane was actually limited, since it was felt that this restriction would further remove undesirable commotion of the tone arm. As a general consideration, the mechanical impedance of the pickup varied with the plane of motion, and was most frequently at a minimum in the design plane.

STEREOPHONIC PICKUPS

With the arrival of stereophonic recordings, the pickup designer was now forced to change his approach. The mechanical impedance ideally was equal in all planes, that is lateral, vertical, and all planes between. Several pickups had horizontally positioned armatures during the monophonic era, however, the vertical mechanical impedance was far greater than the horizontal impedance, Some other designs employed the vertically-oriented stylus with a verticallycompliant bearing, which plane was electrically sensed, and the output then matrixed to give a resultant 45/45 pickup. The general trend in all of these designs, however, was directed to a horizontally positioned moving member in a universally movable bearing. In magnetic devices the moving member is a part of the magnetic-voltage generating system while the piezo electric devices use an intermediate member to couple the motion to the appropriate voltage-generating elements. (Other designs have been employed such as movingcoil, in which there is not necessarily a horizontally-oriented member but for this discussion we will limit it to the most popular current design.)

WEAR OF DISCS

Many studies have shown that wear incurred during the playback of discs is related to the tip area, tip configuration, tracking force, frequency of recording and amplitude of recording (these last two are inter-related since they represent the acceleration applied to the playback tip and mechanical impedance of the moving system.) In those planes where the mechanical impedance is at a maximum the wear will be greatest; conversely in those planes with low mechanical impedance the rate-of-wear will be at a minimum. The actual rate-of-wear in general terms is a function of the motional impedance presented by the stylus, in an ideal stereo pickup the motional impedance should be, or at least approach, zero. Actual modern designs under ideal conditions have shown that this condition can actually be approached.

DISTORTION PRODUCTS

One important consequence of the use of styli with varying motional impedances relative to their design plane, is the unpredictability of the effect of the motion on the moving assembly when constrained. Under the worst case, a pickup having no vertical compliance is called upon to reproduce a vertical signal. The best that one can hope for is no signal at all. This, however, is frequently not the case. In fact, the stylus or the discs plastic now deforms in an unpredictable fashion and generates unwanted signals; this is more generally termed distortion. During the deformation process, which is frequently in excess of the elastic limit of the record material, the rate of wear is at its greatest. The magnitude of the distortion products will generally be a function of the mechanical impedance and the extent to which the linear characteristics of the stylus-plastic relationship is exceeded.

EQUALIZATION

Early systems frequently utilized passive networks for playback equalization. While this may have been state-of-the-art some thirty years ago, it is my opinion and observation that the investment in one of the newer solid-state equalized playback preamplifiers will round out the disc reproducing chain, making possible reproduction well within NAB playback standards. Further, the signal-to-noise ratio will frequently be much improved over the old passive networks, since they were designed for pickups whose output was some 15 dB higher than the current crop.

CONCLUSION

The use of a monophonic pickup to achieve playback of a stereo disc will generally lead to wear greater than if the record is played back with a pickup whose motional impedance is uniform and small in all planes. Further, the use of a pickup with high motional impedance in one plane will result in large, unpredictable distortion products in that plane. The magnitude of these effects is a function of the motion in the planes of high impedance. While they may be reduced to a lesser magnitude by various recording devices, they are nevertheless present whenever the motional impedance and plastic effects either exceed their linear range or secondary effects take place.

When one considers the liklihood of damage from playing a stereo disc with a monophonic record, in light of the experience and studies conducted over the last decade, then the continued use of any but the best playback pickups is an expensive and possibly destructive attitude. The cost of even the best pickups does not exceed the cost of a dozen of the best LP's. When one equates these two factors, the cost of the pickup versus the cost of a dozen discs, there can be no question as to the reasonable and logical conclusion for the professional user. The improved results combined with the increased wear far exceed the risky continuation of using a monophonic pickup for stereophonic reproduction.

db July/August 1968

The 1968 Midwest Acoustics Conference

ROBERT B. SCHULEIN

In his brief report, Mr. Schulein outlines the most prominent audio-interest facts of this recent conference.

N AN EFFORT TO ADVISE industry of the acoustical activities of universities in the midwest, the second annual Midwest Acoustics Conference was hosted by Northwestern University in Evanston, Illinois, on April 11, 1968. Last year, the first Midwest Acoustics Conference was held at the University of Illinois, Chicago Circle Campus and, in 1969, the University of Wisconsin at Madison will participate. Although none of the presentations dealt directly with the field of professional audio, several of the talks were of indirect interest.

In the presentation by W. O. Olsen, Performance Characteristics of Hearing Aids, some of the subjective aspects of various forms of distortion were discussed in regard to speech intelligibility. Distortion was introduced by symmetrical and asymmetrical clipping over a range of about 2 per cent to 50 per cent while subjects listened to voice communication over a 200 Hz to 4 kHz bandwidth. The presentation was concluded with a tape-recorded demonstration of the subjective nature of the various forms of distortion investigated. Of particular interest was a presentation by Karl Reimann entitled Piezoelectric Film. This paper concerned the details of an acoustic transducer which consisted of a vacuumdeposited piezoelectric cadmium sulfide-field-effect transistor on a high-temperature plastic film. Such a device, if further refined, could be the basis for a small, high-quality microphone.

The effect of noise on binaural hearing was discussed in a paper by Z. G. Schoeny. In his presentation, Dr. Schoeny investigated the masking ability of binaural noise on the perception of pure tones of various frequencies as a function of the amplitude and phase characteristics of the binaural noise signal.

Three particular areas of sound reproduction research were discussed by Dr. Harry F. Olson in his after-dinner speech *Trends in Sound Reproduction*. The areas discussed were non-

linear distortion reduction, subjective aspects of sound reproduction, and the reduction of the bandwidth required to transmit speech information. Two specific examples of nonlinear distortion reduction by means of complementary distortion were described, one dealing with the reduction of tracing distortion in phonograph records, and the other with the reduction of loudspeaker distortion due to non-linearities in the magnetic field, and compliance of the cone edge suspensions. In regard to the subjective aspects of sound reproduction, a proposed playback system was outlined, consisting of six separate speaker systems surrounding the listener. Three of the speakers were used to provide the primary signal output and maintain the auditory perspective, and three other speakers were used to simulate the reverberation characteristics of the room in which the recording was made.

The final topic of his discussion was a speech bandwidth reducing system capable of a 1000-to-1 bandwidth reduction by making use of the recognition of speech elements such as syllables, syblets, and phonemes.



Figure 1. Dr. Harry F. Olson delivering his after-dinner talk Trends in Sound Reproduction.

Tracing Distortion

ARNOLD SCHWARTZ

A thorough understanding of tracing distortion will enable both the final user and the cutter of records to extract the lowest possible distortion from the disc.

AN A DISTORTION-FREE phonograph pickup ever be designed? There are many sources of pickup distortion including mistracking, non-linear transducing system, and deviation of the playback angle from the recorded angle. Proper design and manufacture can minimize. and hopefully eliminate these types of distortion. Tracing distortion, however, is inherent in the geometry of the present disc recording-playback system. This type of distortion is noticed mainly at the inner diameter grooves where break up and shatter become evident at high frequencies and high recorded amplitudes.

The pickup generates tracing distortion because the play-back stylus does not have the same shape as the recording stylus. The recording stylus is a wedge-shaped cutting tool, while the playback stylus is spherical. This problem is further accentuated by the fact that the inner diameter linear groove velocity is less than half the velocity of the outer grooves. We will discuss the distortion generating process and then separate the factors which determine the magnitude of tracing distortion. We will also discuss current approaches to the reduction of tracing distortion.

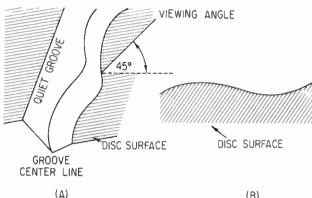


Figure 1. (A) is a stereophonic groove with one wall modulated. (B) gives a view of the modulated channel 45° to the disc surface and perpendicular to the quiet groove.

LINE CONNECTING SUCCESSIVE POSITIONS SUCCESSIVE OF BALL CENTER, Cs POSITIONS OF BALL (A) (B) (B) Figure 2. In (A) a playback tip is shown in a cross section of the record groove, while in (B) the same view as Figure 1 (B) has the recording and playback styli in the groove.

TIP RADIUS

DISC SURFACE

TRACING DISTORTION IN STEREOPHONIC RECORDING

Tracing distortion in stereophonic recording is easier to visualize and is more of a problem than in lateral or monophonic recording. FIGURE 1(A) shows a recorded groove with one wall modulated sinusoidally. By viewing this modulation 45° to the recorded surface and perpendicular to the quiet channel (see arrow FIGURE 1(A), the modulated channel will be seen as a hill and dale or vertical recording as shown in FIGURE 1(B). The playback stylus is conical with a spherical tip, and is shown riding in the groove in Figure 2(A). The recording stylus, which is shown cutting the groove, is wedgeshaped. Since only the spherical portion of the playback stylus is in contact with the groove, it can be represented by a sphere; see Figure 2(B). We can also imagine that the groove is stationary and that the stylus ball rolls along the groove. In the following analysis, we will be using a stereophonic amplitude-responsive pickup, where the output of each channel is proportional to the groove modulation amplitude. A velocity responsive pickup would serve as well, but the waveform would be differentiated and make visualization of tracing distortion more difficult. The electrical output of an ideal linear-amplitude pickup is an electrical replica of the mechanical motion of the pickup stylus. The stylus motion will be the curve connecting the successive positions of the center of the ball as it rolls along the groove contour; see Figure 2(B).

Arnold Schwartz is president of Micro-Point, Inc., manufacturers of recording styli. He was previously associated with CBS Labs, where he was instrumental in the design of their series of test records.

An exact analysis of tracing distortion can become very complex, and since we are only interested here in the concept. we will make a simplifying assumption; let us assume that a sine wave is a series of semicircles alternating convex and concave. Figure 3 shows a modulated groove comprised of semicircles with R_g representing the radius of the modulation. and the playback stylus is shown with R, representing its radius. We will make $R_g = R_g$, a condition that can occur frequently in actual playback conditions. The playback stylus rolls along the groove in the direction of the arrow, and the dotted line represents the successive positions of the center of the playback stylus ball. These positions in turn are represented by the dotted circles. When the ball travels on the convex portion of the curve, the ball center describes an arc that is a longer distance than the corresponding groove contour. When the ball travels on the concave portion of the curve, the ball center describes an arc that is a shorter distance than the corresponding groove contour. In fact when the playback ball is centered in the convex segment the centers of the modulation and playback balls coincide and the playback ball motion goes to zero for an instant. The resulting pickup output is a distorted version of the groove contour. The tracing distortion thus generated will consist mainly of second and third harmonics. FIGURE 4 displays the fundamental, second, and third harmonics (in proportional magnitudes) derived from the playback of an actual sine wave. By combining all three wave forms, a composite wave form is derived that reflects the tracing distortion process described above.

RELATIVE MOTION OF RECORD AND PLAYBACK STYLI

CENTER OF BALL WEDGE-SHAPED

CUTTING STYLUS

FACTORS AFFECTING TRACING DISTORTION

The amount of tracing distortion generated is dependent upon the following factors:

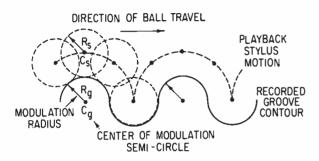


Figure 3. Semi-circle modulation where Rs = Rg.

Tip Radius: Tracing distortion is directly proportional to the tip radius. As the tip radius increases, the difference between the distances the ball center travels on the convex and concave segments of the groove contour increases and thus increases the distortion. Figure 5 shows two identical grooves; in Figure 5(A) it is played with a relatively small tip radius, and in Figure 5(B) with a large tip whose radius equals the modulation radius. The distortion is negligible in Figure 5(A) compared to the high distortion of Figure 5(B).

Modulation amplitude: Tracing distortion is directly proportional to the modulation amplitude. As the modulation amplitude increases, the modulation radius decreases, see Figures 6(A) and 6(B), which makes the playback tip radius appear larger by comparison and thereby increasing distortion. Figure 6(B) shows a modulated groove which is the same as in Figure 6(A), but with a larger modulation amplitude. The

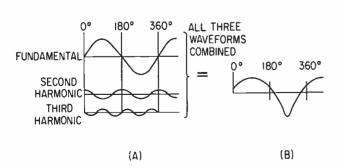


Figure 4. The tracing-distortion components of sine-wave modulation. At (A) are the fundamental and harmonics generated by tracing distortion. At (B) the output waveform for sine-wave modulation is shown.

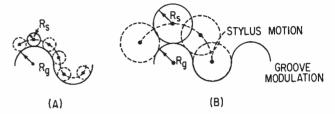


Figure 5. The effect of tip radius on tracing distortion. At (A) is a small tip radius while at (B) the effects of a large tip radius may be seen.

playback distortion from the groove in Figure 6(B) is greater than that from Figure 6(A). Tracing distortion is therefore greater at higher recorded levels.

Modulating frequency: Tracing distortion is proportional to the frequency. As the frequency increases, the radius of modulation decreases and by comparison the playback tip radius appears larger—resulting in increased distortion. Figure 7(B) shows a modulated groove which is the same as that in Figure 7(A), but with a higher modulating frequency. The playback distortion from the groove in Figure 7(B) is greater than that from Figure 7(A). Tracing distortion is more severe at higher frequencies.

Recorded diameter: Tracing distortion increases as recorded diameter decreases. Figure 7(A) and 7(B) also represent two identical recorded grooves, but with Figure 7(A) at a larger diameter with little or no distortion, and Figure 7(B) at a smaller diameter with higher distortion. Tracing distortion is thus more pronounced at inner diameters.

A pattern emerges from the above discussion of the effect of tip radius, amplitude, frequency, and diameter. We can make a general rule which states that the distortion generated is proportional to the ratio of the playback tip radius (R_{\bullet}) to the radius of modulation (R_{\bullet}). Put into a formula; tracing distortion $\sim \frac{R_{\bullet}}{R_{\star}}$. The modulation amplitude, modulation frequency, and recorded diameter in combination determine the modulation radius. In the analysis of an actual sine wave, a quantity called the radius of curvature is used which is the second derivative of the equation describing the amplitude of modulation. When the ratio $\frac{R_{\bullet}}{R_{\star}} = 1$ the second harmonic distortion is 25 per cent, and the third harmonic distortion is $12\frac{1}{2}$ per cent. As the ratio $\frac{R_{\bullet}}{R_{\star}}$ decreases the decrease in second harmonic is directly proportional, the decrease in third harmonic, however, is more rapid. The effect of the ratio

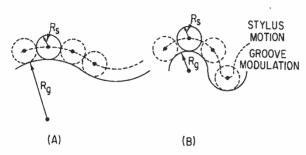


Figure 6. The effect of modulation level on tracing distortion. (A) low-level modulation and (B) high-level modulation.

(A) (B)

Figure 7. The effect of frequency or diameter on tracing distortion.
(A) is a low frequency or outer diameter, (B) is a high frequency or

inner diameter.

STYLUS MOTION

 $\frac{R_{\bullet}}{R_{\pi}}$ on harmonic distortion is shown in Figure 8.

When the playback stylus is tracing a complex wave, as in the case of music, intermodulation tones are generated. When $\frac{R_{\bullet}}{R_{\epsilon}} = 1$ the effect is severe and the familiar break up or groove shatter will be heard.

STYLUS WEAR

The effect of a worn playback stylus can now be related to tracing distortion. Figure 9(A) shows a normal unworn stylus viewed 45° to the recorded surface (same view as in Figure 1). Figure 9(B) shows the same tip after it has been worn. The stylus now appears to have a larger radius and the ratio $\frac{R_s}{R_g}$ for any set of recorded conditions will be larger, and the distortion proportionally greater. To give the reader some feeling for the magnitudes involved, we list three sets of conditions for $\frac{R_s}{R_g} = 1$, and where second harmonic distortion is 25 per cent and third harmonic distortion is $12\frac{1}{2}$ per cent

ard stereophonic recording level of 5.0 cm/sec peak velocity.

Diameter 43/4" 51/2" 6"

Recorded level +6dB +3dB +10dB

Frequency 4 kHz 7 kHz 4 kHz

for a 0.7 mil tip radius. Recorded level is relative to the stand-

MONOPHONIC PLAYBACK

The discussion up to this point has been focused on stereophonic recording where each groove wall is independently modulated. What happens with lateral modulation when both groove walls are modulated in phase with identical signals? Figure 10 shows the phase and amplitude relationships of the harmonics generated by the pickup from each groove wall, and the resultant waveform when these are added to produce a lateral output. Second harmonic distortion is out of phase and is cancelled, while the fundamental and third harmonics

add. Note, that the percentage of third harmonic does not change since each wall has the same proportional distortion. The more significant second harmonic distortion does, however, cancel and clearly illustrates the additional problems created by the change from monophonic to stereophonic records.

The inherent distortion reduction of lateral recording as compared to stereo recording is analogous to the reduction of second harmonic distortion in an amplifier push-pull output stage as compared to a single-ended amplifier output stage. The two groove walls, in lateral recording, are also in push-pull.

TRACING DISTORTION REDUCTION

With the advent of stereophonic recording, with its inherent higher tracing distortion, various methods have been used to reduce it. The most obvious method has been the

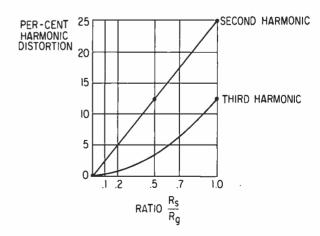


Figure 8. Tracing distortion as a function of $\frac{R_s}{R_a}$

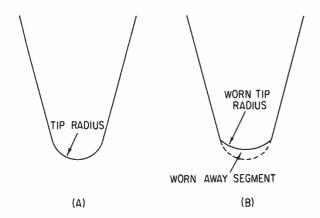


Figure 9. Stylus wear and tip radius. At (A) is an unworn stylus. At (B) the change of radius due to wear is clearly seen.

reduction of the tip radius. It is felt by some pickup manufacturers that a further reduction in tip size below 0.5 mil would decrease the bearing surface and cause undue high stress on the groove walls and permanently deform the record. To counteract this tendency, the elliptical tip was developed. This type of tip attempts to reduce the radius size in the tracing distortion direction to approximately 0.25 mil. An adequate bearing area is maintained by having a relatively large radius perpendicular to the small radius, where the large radius will not affect the magnitude of tracing distor-

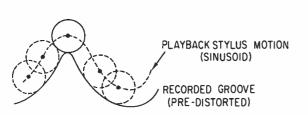


Figure 11. Tracing distortion correction by recorded pre-distortion.

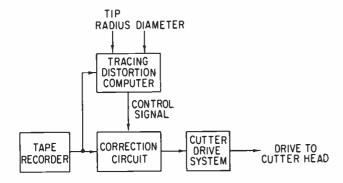


Figure 12. A block diagram of a tracing distortion correction system.

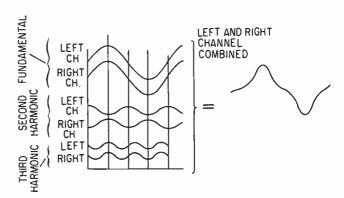


Figure 10. Tracing distortion in lateral-only recording.

tion. It is interesting to note that that change from 78 rpm to 33-1/3 rpm required a substantial reduction in tip radius to maintain low distortion levels.

A radically different approach to tracing distortion reduction is the method of pre-distortion. By this method, the recorded groove shape is distorted in such a fashion that the resultant playback distortion is substantially reduced. Figure 11 shows a distorted recorded wave form which when played back will yield an undistorted wave form. This method of tracing distortion reduction is complex. The amount of correction or pre-distortion required is constantly changing with frequency, amplitude, and diameter. The general system for this method of tracing distortion correction is shown in the block diagram of Figure 12. The output of the tape recorder is fed to circuitry which instantaneously computes the tracing distortion that would be present on playback. This circuitry

actually measures the ratio of $\frac{R_{\bullet}}{R_{\bullet}}$ by sensing frequency and

amplitude from the tape recorder output, while tip radius is preset, and diameter information is fed in from an external source. The tape recorder output is fed to circuitry which inserts the appropriate correction voltage and which is controlled by the computing circuitry. One of the difficulties of this method, aside from the complexities in generating and recording the appropriate correction, is that the correction is exact for only one size stylus. A stylus with a larger tip radius will be undercorrected and some distortion will remain. A pickup with a smaller tip radius will be overcorrected and will actually play back-some of the recorded distortion.

Tracing distortion correction by pre-distortion is used by RCA Victor in all their stereophonic recordings. The RCA device is called the Dynamic Recording Correlator and it corrects for a 0.7 mil playback tip. Playback distortion therefore reaches a null for this radius. The distortion will rise slowly for smaller or larger radii. RCA engineers find that by using the Correlator a net reduction in distortion is achieved over a reasonably wide range of tip radii.

db July/August 1908

Compatible Disc Recording

DAVID GREENE

Compatibility of the stereo disc with mono playback is dependent on the ability of the stereo channels to be summed. Care must be taken at the mastering end to ensure that this can happen without problems.

N RECENT YEARS there has been an increased emphasis in stereo by the record companies and by the f.m. broadcasters. The record companies desire to market a single record that will satisfy both the mono and stereo record user, thus decreasing inventory and a duplication of manufacture. The f.m. broadcaster has been faced with transmitting a single signal that will be acceptable to both the mono and stereo listener. Both have encountered a new group of problems that has made it necessary to re-evaluate the recording procedures and develop new tests to assure the best results under any conditions. This article deals with the nature of one of these problems and a method of evaluation.

First, let us look at the requirements of a compatible record. *Tracking* is of prime importance. A compatible record must be capable of being played on either a mono or stereo pickup without skipping. Also the *esthetic balance* of the program material must be maintained. And finally there should be no inherent *distortion* when played in either mode.

Mono records contain lateral information only. Stereo records contain information that results in vertical as well as lateral information. It is this vertical information that presents a serious problem to the playing of early stereo records on mono equipment. The inability of the playback stylus to properly track the groove results in audible tracking distortion and increased skipping. The poor vertical compliance of the playback system also causes serious damage to the record groove. As a result the record can no longer be played even in stereo.

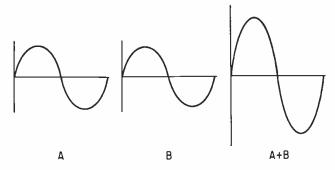


Figure 1. If channels A and B are in perfect phase, addition will result when they are combined.

However, in recent years cartridge manufacturers have been building cartridges with good vertical compliance. And the record manufacturer has learned how to concentrate that information that causes high displacement of a record groove into the lateral plane without causing an audible change in the stereo program material. Consequently, the stereo record of today is playable on much equipment manufactured after the early 1960's.

In today's commercial stereo recording the original recording is usually made on a tape recorder of either four or eight tracks. Each track contains distinct musical elements of the final product. Afterward, it is mixed down to two channels in a seperate remix session to obtain the stereo product. At this time the recorded elements are channeled so as to give the listener the apparent directions of left, right, and center. This center information is recorded equally on both the left and right channels of the stereo master. Let us investigate what happens when this stereo product is then summed to obtain a mono signal.

When the stereo channels are added, the left and right information combine in a simple manner. The center information, however, does not add in quite such a simple way. Classically we are aware of what happens when two identical signals are summed in-phase or out-of-phase. In an in-phase situation the signals simply add in amplitude as shown in Figure 1. If the signals are 180° out-of-phase we get a

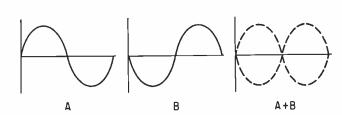


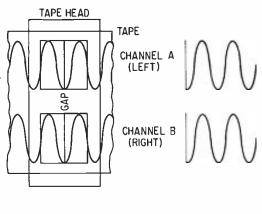
Figure 2. If the two channels are 180° out of phase, their addition will result in cancellation.

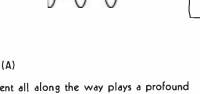
cancellation effect and the signals subtract as shown in Figure 2.

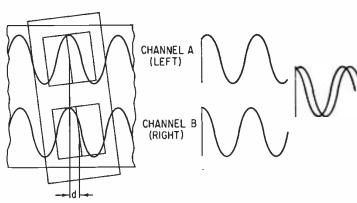
If a 180° phase shift is encountered it is simple to cure by reversing the phase of one channel with respect to the other. Since the center information on today's stereo products is identical, on both channels the laws of phase apply when the stereo is summed to derive the mono. Since all recordings are made on tape initially, let us look at what can happen on a stereo tape recording system.

It can be seen in Figure 3 that a slight azimuth misalignment can result in a phase shift of the center information. It should be noted that this can be caused by either the record head or the playback head in a recording system. It should also be noted that the shift is not 180° as described in Figure 2. Therefore a reversal of the phase of one channel with respect to the other will not eliminate this problem. The shift can be slight enough so that when the product is auditioned in stereo it is not noticeable. However, when the product is summed to derive the mono, this shift becomes a much more critical matter. When added the signals will add during some portions of the wave and subtract during others.

A minor phase shift between the channels causes selective cancellation of the upper frequencies in a mono listening mode. This results in a serious distortion of the product. This phase shift may come from improper azimuth alignment of any stereo head, or by equalizing one channel differently from the

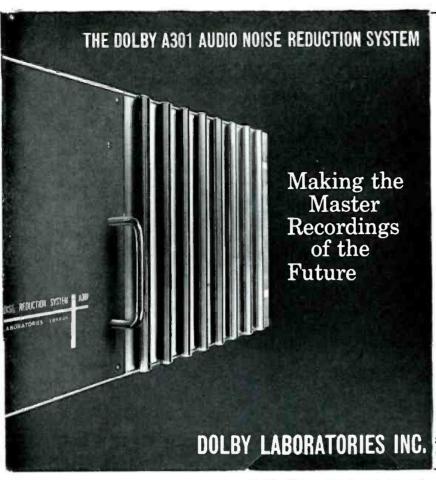






(B)

Figure 3. Tape head alignment all along the way plays a profound effect on the final result. In (A) the results of proper attention is shown in two channels that are in perfect phase relationship. At (B) is an exaggerated view of what happens when mis-alignment allows the phase relationship of one channel to lag the other.



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other along the mastering route. Of course, it can also be introduced by *inherent* phase shifts in either stereo mix channel. Once the stereo has been mixed, and at all subsequent points in its production, it must be checked by mono audition.

CENTER CHANNEL BUILDUP

Much concern has also been voiced about the problem of center-channel buildup. This is the situation where information on both stereo channels adds in a different proportion to the information on either one channel. This results in a change to the esthetic balance of the product when it is played mono.

This is not as serious a problem as most people believe, because this buildup occurs acoustically when the record is played stereophonically. The acoustical buildup of the center is not as much as the electrical buildup when played mono, but it nonetheless exists. If the balance is carefully analyzed at the time of the stereo recording, and this center buildup is esthetically objectionable in mono, careful listening to the stereo will reveal that the center information is too loud on the stereo to begin with.

In conclusion it can be said that a single record can be made that will be acceptable to both the mono and stereo disc user. All that is required is a careful understanding of the problems involved and thorough checking of the product at every step of the way; this will minimize those problems. Although a truly compatible record that will play *perfectly* in mono and stereo is not yet a reality, the stereo product of today comes considerably closer than before.

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Sound with Images

MARTIN DICKSTEIN

MORE ON CCTV LENSES

• The series of equations regarding lenses for c.c.t.v. which were given in June should really have a few more added to them to indicate further the relationships that exist in their parameters.

There may be lenses that do not conform to the following general relationship, but in most cases it is true that the diameter of a lens (D) can be found by dividing the focal length (FL) by the speed (f), or:

$$D = \frac{FL}{f}$$

(with D and FL in inches).

Another association that may come in handy in the choosing of the proper lens for a particular situation is that the

Table I If $L = f \times D$

Focal Length	Focal Length (inches)	Lens Barri Diameter (inckes)
, ,	,	(inches)
12.5	$\frac{1}{2}$	
25	1	
50	2	
75	3	1.2
100	4	1.6
150	6	2.4
225	9	3.6
300	12	4.8
350	14	5,6
500	20	8.0
600	24	9.6
750	30	12.0

1'' = 25.4 mm = 2.54 cmf = 2.5 in all above focal length (FL) is equal to half the vidicon raster width (or height or diagonal) divided by the tangent of half the total viewing angle (\emptyset) .

FL =
$$\frac{\frac{1}{2} \text{ w (or h or d)}}{\tan \frac{1}{2} \emptyset}$$

(FL and w in inches).

You may recall that the vidicon raster has a width of ½ inch with a height of ¾ inch (4 to 3 ratio). (The diagonal of the triangle can be calculated by taking the square root of the sum of the squares of the sides of the triangle.)

At this point it might be worth re-

stating another relationship which proves quite valuable: The lens-to-subject distance(s) in feet is equal to the focal length (FL) in inches times subject width (W) in feet divided by the vidicon target width (w) in inches.

$$s = \frac{FL \times W}{W}$$

It may be easy for some who carry their slide rules (with angle functions) to work out the necessary calculations in short order, but it will save time to use charts or graphs. The following are offered as time-savers for those who wish to use them:

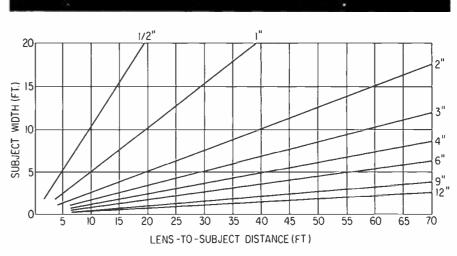


Figure 1. Subject width (W) versus lens-to-subject distance (s). Subject height is equal to three-quarters subject width.

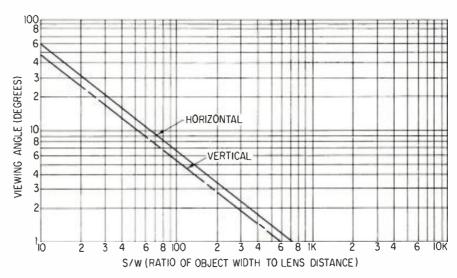


Figure 2. Viewing angle \emptyset versus object width-to-lens distance ratio.

Table II									
				– Lens (i	nches) -				
Angle (degrees)	1/2	1	2	.3	4	6	9	12	
Horizontal	53	28	14	10	7	5	3	2	
Vertical	41	21	11	7	5	4	2	1.8	
Diagonal	64	35	18	12	9	6	4	3	

Viewing Angle (Approx.) for Some Typical Lens.

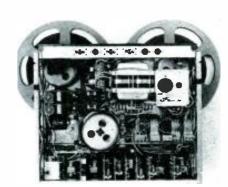
db July/August 1968

New Products and Services

TAPE RECORDER



• This all-transistor stereo recorder, Model A 77, offers a host of innovations. Most unusual is the electronically governed capstan motor. Thus, tape speed has been made independent of the a.c. frequency. It may be used with equal accuracy on 50 or 60 Hz current. Further, a low-current switch is used to electronically change the capstan speed to one of the two standard speeds. Normally 33/4 and 71/2 in./sec. is provided, but 71/2 and 15 in./sec. is available. Tape speed is specified to be within ± 0.2 per cent. Wow and flutter is 0.08 per cent at 71/2; distortion at 0 vu is 2 per cent and over-all frequency response is 30-20,000 Hz + 2, -3 dB both at 71/2. Other important specifications include standard NAB record equalization and switchable NAB/IEC



playback equalization; s/n is better than 58 dB at 7½; and stereo crosstalk is better than 45 dB. Each stereo channel accepts low- or high-impedance mics (switchable) and the output of the recorder is 2.5 v across 600Ω. Operation is fully controlled by solenoids, the transport uses three motors, and the unit accepts up to NAB standard 10½-inch reels. Versions of the transport are available in either two- or four-track stereo configurations and with or without built-in monitor amplifiers. A portable version includes small speakers.

Mfgr: ReVox Corporation

Price: \$499 (without amps), \$569 (with power amps)—both with a wood surround. The portable is \$590. A rackmount unit is \$499. Add \$100 for the 15 in./sec. version.

Circle 101 on Reader Service Card

SOUND/FILM RECORDER



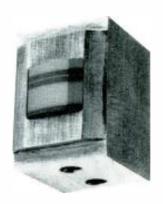
• This unit has been specifically engineered and designed for sound/film synchronization. The model 1000 is lightweight, at 7½ lbs, and compact at 11 x 9 x 3½ inches. It may be powered by a.c., or d.c. supplied from several types of battery source. The unit is equipped with a complete set of frequency-response curves and a calibration curve of its photoelectric automatic level control. Fully transistorized, and operating at a single speed of 7½ in./sec. with a full-track head configuration, the unit provides a guaranteed frequency response of 20-20,000 Hz

Mfgr: Uher-Martel Price: \$695.

Circle 105 on Reader Service Card

RECORDING HEAD

• A new duplicator head is for use with dual-track recording on equipment designed for duplicating stereo cassette tape via reel-to-reel duplication. Designated model 60-520, this duplicating head is glass-bonded to provide the user longer head life as compared with contemporary designs. Track width is 23.6 mils and gap length is 120 to 160 microinches. Dynamic crosstalk is less than -24 dB at the 2 to 5 kHz range and less than -20 dB at 32 kHz. Self inductance is 0.5 (±0.05) mH at 100 kHz and 100 mV, d.c. resistance is less than $4.4(\pm 0.5)\Omega$. The head is for dubbing equipment with a tape speed of 15 in./sec. and can be mounted on both sides for proper head track interleaving.



Mfgr: Ferroxcube Magnetic Recording Div. Circle 106 on Reader Service Card

CORRECTION

A listing in the AES Roundup pages of June showed an incorrect price for the Dolby Laboratories' remote changeover system. The correct price is \$425.



• This device, dubbed the Plierench functions equally well as a plier or wrench. Either way it allows for slipproof performance with its 10 to 1 gear ratio giving the user a powerful grip on the object. Its jaws are always parallel and are easily adjusted to use size by manipulating the handle. The tool is manufactured from forged steel and is heat treated. The standard jaw reaches deeply and securely to free difficult-to-grasp components. Two additional jaws may be interchanged. A pipe-jaw attachment will grip pipe up to 2 inches in outside diameter-and without thread damage. A large internal-external jaw grips tubes, pipes, or fittings from inside. It may be fitted into the Plierench from the inside as well as the outside.

Mfgr: Vaco Products Company Price: \$7.95 (8½ inch version). A smaller unit is also available. Circle 102 on Reader Service Card.

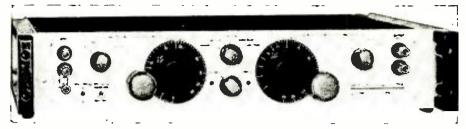
VTR

• The model 6402 video tape recorder uses full-field slant-track, two-head, frequency-modulated recording of the helical-scan type. The head assembly features all-electronic rotating elements; this is stated to eliminate one of the prime hard-to-clean areas of high wear. Tape of 1/2 inch width is used at a linear tape speed of 91/2 in./sec. providing up to 60 minutes from an 81/4 inch reel. Audio re-record capabilities make it possible to overdub the sound track without disturbing the video portion. Other features include: automatic audio and video level controls; built-in 2:1 sync generator; electronic editing capability; slow motion fixed at 1/12 normal speed, and stop motion for reproduction of any single field of video information. A slanted tape deck eliminates idler assemblies and inclined tape guides. This adds further mechanical simplicity to the unit by feeding the tape in direct alignment to the drum.

Mfgr: Craig Corporation

Price: \$1200

Circle 108 on Reader Service Card



• The new Type 1952 universal filter will perform as a low-pass, high-pass, band-pass, or band-reject filter at the turn of a panel switch. High-pass and low-pass filters provide fourth-order Chebyshev approximations with ± 0.1 dB pass-band ripple and an initial cutoff rate of at least 30 dB per octave at all frequencies from 4 Hz to 60 kHz. The low and high filters can be employed singly, in cascade, or in parallel; cut-off frequencies of the two filters can be ganged together to provide constantpercentage band-widths for band-pass or band-reject tuning. In addition to the usual signal-conditioning applications the unit also can act as part of a spectrum analyser or distortion meter and with a random-noise generator, produce controlled bands of noise as test signals. Minimum bandwidth in the band-pass mode is 25 per cent of the center frequency (approximately 1/3 octave) and a null (infinite attenuation) characteristic is provided in the band-reject mode. Maximum sine-wave input is 3 volts rms, or 30 volts rms with the input attenuator set at 20 dB. Input impedance is $100~\mathrm{k}\Omega$; output impedance is 6000

Mfgr: General Radio Corporation Price: \$950 Circle 104 on Reader Service Card

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The seminar will be held at 7:30 P.M. on October 22, 1968, in the Oriental Room of the Park-Sheraton Hotel in New York City.

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db July/August 1968

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edited by Robert L. Hilliard. Each of the five chapters has been written by a prominent educator with an extensive background of practical experience in commercial and educational broadcasting. The areas covered include: management and programming, operating and studio facilities, producing and directing, writing, performing. For those of you who want to, or must, operate on both sides of the control room, this is virtually required reading. 190 pages; 6½ x 9½; indexed; clothbound.

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by Patrick Finnegan. 1965. An informative guide for the planning, building, and operation of a small-market, UHF-TV station. Based on the author's lengthy experience in the technical operation of such a station, it explains equipment, layout, and building costs that apply to the small studio (one-man control room operation) with a heavy film schedule. A valuable aid for the station owner, manager, engineer and layman interested in this medium. 328 pages; 6 x 9; illus; clothbound

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by Alexis Badmaieff and Don Davis. A thorough and comprehensive "do-it-yourself" book providing a wealth of practical and theoretical information on the "whys" as well as the "hows" of constructing highquality, top-performance speaker enclosures. Contains detailed drawings and instructions for building the various basic enclosures. Includes infinite-baffle, bass-reflex, and horn projector types as well as several different combinations of these. The book covers both the advantages and the disadvantages of each enclosure type and includes a discussion of speaker drivers, crossover networks, and hints on the techniques of construction and testing. Written and compiled by two of the nation's leading authorities in the field of acoustical engineering. 144 pages; 5½ x 81/2; softbound.

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PRACTICAL PA GUIDEBOOK: HOW TO INSTALL, OPERATE AND SERVICE PUBLIC ADDRESS SYSTEMS

by Norman H. Crowhurst, 1967. This book gives all the basics needed to become a successful PA operator, in any situation where the reinforcement, relay, or distribution of sound can provide a service. It shows how to properly install, operate and service public address systems. All aspects of the subject, from survey to the selection of appropriate equipment, to installation, to routine operation and the maintenance of a finished system, are covered. Attention is given to solving problems encountered in providing successful service. The book's systematic and practical approach makes it highly useful to radio-TV servicemen, hobbyists, and PA equipment manufacturers. 136 pages; 6 x 9; illus; softbound.

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CLOSED CIRCUIT TV SYSTEM PLANNING

by M. A. Mayers and R. D. Chipp. 1957 This book discusses in detail the vitally important and rapidly expanding concept of closed circuit TV systems, its utility and functioning. This book is not an engineering or a technician's text—it is written for management. It explains and illustrates the kind of systems that are available and their applications. 264 pages; 8½ x 11; illus; clothbound. \$10.00 (\$11.95 in Canada)

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General Audio

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MICROPHONES

by A. E. Robertson, 1963. This book, primarily written as a training manual for technicians, will also prove valuable to all users of quality microphones whether for broadcasting, public address systems, or recording of all types. There is now an almost bewildering array of microphones of differing characteristics, and new designs are constantly being produced. The author makes no attempt to catalogue these but concentrates mainly on the principles of operation. He only describes actual microphones if they illustrate an important feature or have some historic significance. The book is intended for the user rather than the designer; mathematics have been omitted from the main body of the text. 359 pages; 6 x 9; illus.; clothbound.

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by Guy Fontaine. 1967. This systematic and detailed treatment of the application of transistors in audio-frequency amplifiers shows how the published transistor characteristics are related to the principles of design. To assure clarity, the figures are rendered in several colors and placed opposite the related text. Simple equations reinforce the lucid approach. An ideal textbook or reference on the subject for engineers and advanced technicians. 384 pages; 5½ x 8; illus.; clothbound.

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People, Places, Happenings

• Donald V. Kleffman has been named marketing manager of Ampex Corporation's video products division. In the announcement by Lawrence Weiland, general manager, it was announced that Mr. Kleffman has been with Ampex since 1959. In other marketing assignments, Richard Sirinsky has been appointed national sales manager and Frank B. Thompson has been named manager of video product management for the division. Both men also joined Ampex in 1959 and have held various other positions.



•Verne Linsley has been appointed district manager for the eleven western states for Perma-Power Corporation's Ampli-Vox division. According to James F. White, Ampli-Vox v.p., Mr. Linsley will work out of the company's new branch in Long Beach, California. His responsibilities will include all sales of Ampli-Vox cordless public address equipment in California. and the coordination of the activities of Perma-Power's sales representatives in the adjacent territories. This represents an in-company promotion for Mr. Linsley; he was previously an Ampli-Vox sales representative.

• Mervin L. Samuels has joined the broadcast division of Arrow Electronics, Inc., as sales engineer for the Metropolitan New York area. He will serve as Arrow's sales representative and technical adviser to broadcasting and recording companies. The broadcast division, although a relatively new adjunct to Arrow's operations, has increased the company's share of the electronic parts and audio markets significantly. The department was a contributing factor in Arrow's 1967 record sales of more than \$10 million. Mr. Samuels will operate for Arrow's Manhattan sales center and will act as liason with the broadcast division in the company's main facility in Farmingdale, New York.

has announced its schedule for the New York and San Francisco High Fidelity Music Shows. In New York, the show will be held on the second and third floors of the Statler Hilton Hotel. Dates are September 18th through September 22nd. The first day is dealers-only. The next two are reserved for dealers in the afternoon and consumers at night. Saturday and Sunday, the 21st and 22nd, are for the consumer. Admission for non-trade will be \$2.00.

San Francisco's show will follow a similar format. The place is the Civic Auditorium and the dates are October 30th through November 3rd.



• Anthony R. Pignoni, director of marketing for Philips Broadcast Equipment Corp., has announced the naming of Thomas R. O'Hara as a sales engineer for the northeast region. Mr. O'Hara will be headquartered at the company's main plant in Paramus, New Jersey. Before joining Philips, which markets under the trade name—Norelco, Mr. O'Hara was eastern regional manager for the GPL Division of General Precision Equipment Corp. Prior to this association, he was with the Rock International Corp., a division of Gates Radio Company.



In another announcement, Mr. Pignoni appointed E. J. Manzo as manager. commercial video systems. Mr. Manzo comes to Philips from Cohu Electronics, Inc. Prior to that he was broadcast product manager for CBS Laboratories.



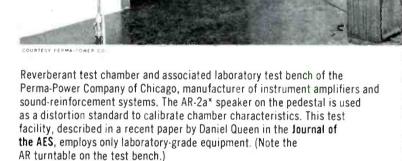
John H. Beaumont, a man who those of us in professional audio in New York had come to know and like well, has died suddenly at the age of 48. He was a Governor of the AES at his death and extremely active in the New York Chapter. We saw him at each Chapter meeting we attended. His help in arranging for our coverage of some of these meetings was valuable indeed and shall be missed.

He was a partner in Audio Techniques, Inc. of New York for six years. At the time of his death he was production superviser of Floyd L. Peterson, Inc., film producers. He was well known prior to these associations as a free-lancer in audio fabrication, recording, acoustic analysis, etc. As a charter member of the AES he held many offices both on a national and local level.



Saul J. White has died at the age of 64, a victim of cancer. At the time of his death he was chief acoustical engineer of Dyna-Empire, Inc. and Empire Scientific Corp., of Garden City, New York. Prior to that he was, for many years, chief engineer of the Audak Division of Rek-O-Kut Company. Prior to these titles, he had held positions with Racon Electric Co., and University Loudspeakers with whom he had been chief engineer for ten years. He was active in the AES as a charter member. In recent vears he had held several offices on both the national and local levels of that organization.

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