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db

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Coming Next Month

• Magnetic Tape is the topic.

The new year gets kicked off with articles on this vital part of the audio business. To begin, we look at metal particle tape as it may apply to the profession with its promise of upgraded analog performance. Is its promise the equal of digital? Find out next month.

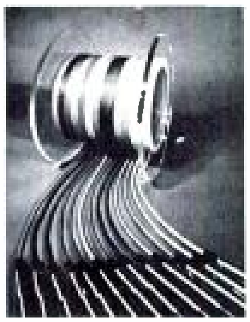
The article The Transfer Characteristics of Magnetic Tape takes a detailed look at some of the intracasies of tape performance.

Tape Speed versus Biasing is a practical discussion in our Application Notes format on the tradeoffs and interrelationships of these aspects of tape usage.

January will have a directory of raw tape specs; a roundup of the use of programmable pocket calculators as an audio tool (with programs); and a picture/text story on the recently concluded NRBA and AES Conventions.

Coming in **db, The Sound Engineering Magazine.**

About The Cover



• At the recently concluded AES Convention we were much taken by a display of wires and boxes at the Wireworks display booth. Accordingly we asked for a spool of their multi-colored cable-plus-connectors and photographer Robert Wolsch and art director Bob Laurie did the rest.

db

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db Letters

THE EDITOR:

I read with interest your announcement in the September issue that the October issue would be devoted to disc (sic) recording. Permit me to make some comments—some old, some new.

1. Now that the AES Journal has also seen the light and has changed its spelling of the word "disc" to its proper spelling, DISK, simply because as signatories of the IEC standards we must follow the international spelling of the word which is and always has been "disk," I wonder if you can't do the same.

2. It really amazes me to see that you can find it in your bag of tricks to write "When was the last time you saw a new disc cutting lathe? . . . and then simply ignore our new VMS 80 Lathe. I know that you already had done something at the time of the Hamburg AES, but it must be eminently clear to you that 90 per cent of your readers were not at that convention, for one thing, and really deserve a more in-depth coverage of the device rather than just a photo and a few words. Shortly, we will be delivering the first such lathe in the world to Sterling. I think that NOW is the time to write this up, rather than chasing after what must appear, even to you, a rather crude effort in a field in which *only* decades of experience make any difference!

3. The CBS system in no way allows any "more dB on your record." Anyone who claims that will have to prove it to me. There isn't any cutting system that will deliver any more dB, and that's the limiting factor.

4. How come you spelled Europa Disk with a "k"? It's certainly against your principles.

STEPHEN TEMMER
President, Gotham Audio
New York, N.Y.

db Replies:

1. Mr. Temmer's interest in international standards is of course well-known, and we appreciate his concern over the merits of "disc" vs. "DISK." We'll have to think about this one for awhile.

2. Some time ago, Gotham Audio Corp. announced that in 1979, the VMS 80 "will be reserved for sale only to previous Neumann disk equipment owners or to newcomers purchasing a complete tape-to-disk transfer system."

Without commenting on the propriety of such an announcement, it does seem to disqualify many—if not

(continued)

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most—db readers from purchasing such a system. No doubt, the VMS 80 is a significant improvement over earlier models, as we should expect from such a high-technology company as Neumann. However, until Gotham Audio decides to make the system available to the general public on a first-come, first-serve basis, we prefer to give priority coverage to equipment that is, or will be, available to all our readers. We trust that our readers will understand that, although we're delighted to learn that Sterling will be getting its own VMS 80 soon, we really do need something more than a delivery announcement to justify a feature article.

Since Mr. Temmer's letter was written before we published the Cybersonics Lathe story, he may have missed our point in featuring it in our October issue. It would appear that Cybersonics offers the prospective buyer some simplicity of design. We think it's a bit premature to label it a "crude effort."

3. We don't own a CBS DISComputer (DISKomputer?), so we can't prove a thing about "more dB on your record. However, technology does have a way of marching on, and it strikes us as reasonable that no one has a monopoly on improvements to the art of tape-to-disc transfer. Therefore, perhaps the CBS Technology Center has indeed figured out a way to coax a few more dB into the grooves. But if our feature story didn't prove that, why not ask them directly?

4. We didn't spell Europa Disk with a "k"—they did. And since it's their company name, it would be misleading for us to spell it any other way. Besides, then we'd have to figure out what to do with disco, discography, discotheque and all the others. It's just more than we can handle right now. We're having enough trouble giving letters to the editor the attention they deserve.

MORE ON BROADCAST AUDIO

THE EDITOR:

On page 2 of your September 1978 issue, there was published a letter from an anonymous Chief Engineer who complained about the terrible things done to audio in the name of "improved-on air sound" (audio processing).

I agree wtih him completely and, like him, am a "working stiff." However, I feel it is time that owners of stations, as well as engineers, speak out on important matters like this and not be fearful of criticism.

We at KRJB have set out on our own to create a totally "different" type of station. Not very many station own-

(continued)


This Auditronics 501
at Osmond Studios...

ers like to risk their capital doing something untried, but we did and it worked out great. We have concentrated on programming and offer our listeners everything from drama to opera and jazz. The response has been terrific: we have not spent *a cent* on publicity and although we are only 11 months old we have become very well known and gathered great public support for the style of programming we offer. "A welcome alternative to rock and elevator music" is the way one columnist put it. Individuality in programming is the key to a successful station. Being a copy-cat shows only that management of a station is unsure of itself. And that is why the public has suffered from lack of decent programming. And yes, we don't "process" our audio to the degree indicated by the disgruntled writer last month. Our listeners are happy, it seems, with the crisp clear sound of a solid state f.m. transmitter and recordings made by audio engineers. We are not as "powerful" as some stations, but we know we have the listeners by the ear lugs—and that's really all that counts.

Furthermore, our programs aren't interrupted by commercials. Commercials are run during commuter hours 6-9 a.m. and 4-6 p.m. and that leaves the rest of the day free to the listener to enjoy quality programs which other stations say they "can't afford."

Good luck to you, Mister Engineer. whoever you are! I hope you will continue to gripe about excessive audio processing, excessive regulation, excessive management interference in engineering matters *and* in programming decisions made purely on the basis of what management "thinks" the public wants.

If I wasn't sick and tired of the tripe offered by radio and the lack of alternatives, I wouldn't have embarked on putting KRJB on the air. It is a terrible thing when Radio—one of the most beautiful art forms—is relegated to nothing more than hyped-up juke box which plays the same tunes over and over again until one is sick of the stagnant mess.

I sign my name because I would not WANT to work any place else than where I am, and, judging by the desk drawers filled with application forms. KRJB would not have any difficulty finding someone to replace me. The dissatisfaction with the status quo in the broadcasting industry is rampant —and such public dissatisfaction may well destroy the industry altogeher unless it shapes up and takes note.

MIKE ERICKSON
Manager, Station KRJB
Monte Rio, CA

P.S. If management lets sales people make the programming decisions, it will also fail. Sales people are not en-

gineers and they are not capable of making realistic judgments on programming. Any salesman worth his salt claims he can "sell anything"—and yet if a station dares to be different, most salesmen claim "it won't work." Judging by the trash one hears on the airwaves of America most of those salesmen are proving their point. Whatever happened to salesmen who believed in their PRODUCT?

STILL MORE ON BROADCAST AUDIO

THE EDITOR:

On an otherwise typical day recently, I received the current issue of **dB,** which I usually find quite interesting. It is, however, written from a point of view quite removed from the situation of the medium market Chief Engineer. In particular I refer to the article on **The Sound of Broadcast Audio.** This article continues a line of reasoning which has become a de facto editorial line of dB.

The premise seems to be that the studio engineers are turning out a great recorded sound which radio stations set about to screw up by use of AGC and limiting devices. The assumption is that a wide dynamic range is necessary for the appreciation of music.

Speaking as a Program Director/Chief Engineer, let me put you in touch with a slightly different approach. We process audio for several reasons. The first of these relate to the items mentioned in the article, namely use of spectrum space, loudness, and FCC requirements. But we process audio for at least one more reason which is never mentioned in your magazine; processed audio sounds better. In fact it sounds a lot better. That's why we process it.

Audio comes to radio stations from various sources, each with their own problems. Records provide one of the biggest problems because they are made with a deliberately wide dynamic range. The result of this is that the music is always too loud for the system or too soft to hear. This is most true in noisy environments such as cars but is also true in most living rooms at anything other than neighbor-deafening volume on very expensive equipment. Modern recording techniques have made this problem worse. Years ago, recording engineers rode gain to bring various parts of the music to the listener at an acceptable volume. This applied, at least in popular, if not in classical, music recording. Where is it written in stone that a tap on a cymbal or triangle cannot be recorded at a high enough level so that it can be heard easily without bringing the system gain high enough to inflict pain when the whole orchestra comes in?

The next problem with recorded studio sound is that the NAB standard level itself is largely ignored or interpreted very differently from one disc to the next. In "improving" the sound of new records, old ones are made to sound as if there is no treble at all. 45 RPM records are recorded hotter than LP's.

To program a radio station we must assemble a wide range of audio sources, from records to networks, to mikes, to agency tapes, to cart machines, etc. Since all of these vary so widely in quality, dynamic equalization is necessary to give the listener a program that doesn't alternately blast and disappear.

An additional problem with the audio industry generally is that the "knob freaks" have taken over. These are people who are constantly fixing equipment that already works. I also include in this category the design engineers who seem to think up a new gimmick a week to obsolesce last week's gimmick. When are engineers finally going to admit that the audio amplifier has been invented, and need not be re-invented every month forever?

When broadcasters buy equipment.

(continued)

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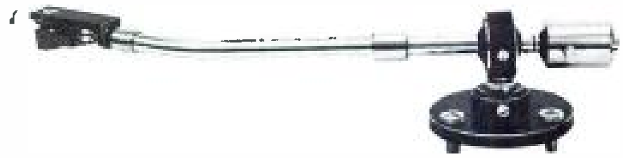

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it must last ten to twenty years for us to get our money back and make a profit. I learned recently that our idler-wheel turntables are now obsolete since the development of the new direct drive models. Why are they obsolete? Because rumble is now audible on soft passages of records. This is because the studio engineers are trying to widen dynamic range to amuse themselves in their very expensive studios. If they would learn to ride gain again, listeners would be able to hear the soft notes without risking ear damage when the rest of the band plays, and the "obsolete" turntables would sound fine again.

None of the above is intended to negate legitimate progress. But the progress should be directed to our needs rather than to the fantasies of the designers. Now that it has been decided, for instance, that cassette machines should be used for music instead of talk, we have poured extensive technology into the development of new tapes, new heads, Dolbys etc., and have almost ignored reel-to-reel recorders, which sound great. I expect shortly to learn that they are also obsolete now that the cassette has been proven to be such an excellent recorder for music.

Again, radio stations process audio because it sounds better. When I audition music on a good system in a relatively quiet office, I spend half my time fiddling with the volume to hear the music properly. Similarly, if I put a stack of records on a changer, (oops, guess that means I'm not a purist) at home and invite some people over, we usually give up on the music within fifteen to twenty minutes and turn on the radio.

Obviously audio processing can be abused, but if recording engineers would get in touch with the real world, and learn how to read a vu meter (oops I'm obsolete again), then we could set the gain at the NAB standard level, and just enjoy the music.

PAUL DUNN
WDBF Radio
Delray Beach, Florida

Anyone out there want to answer this one for us?—Ed.

AND STILL MORE

THE EDITOR:

I feel I must take issue with the letter from "A Chief Engineer," which was published in your September edition.

For the past fifteen years, I've been a radio programmer. Consequently, I've had ample time to observe the attitudes and performance of numerous chief engineers and I think your correspondent's letter admirably exempli-

(continued)

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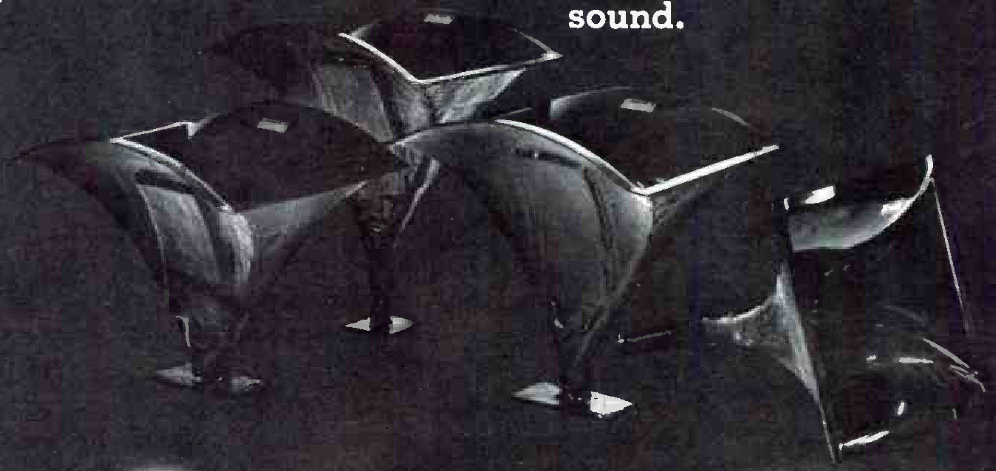

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fies those attitudes. So, hopefully, a little more concisely, I will attempt to respond to his marks from a programmer's viewpoint:

Dear Chief Engineer, if you would make even a cursory attempt to understand the business that pays your salary (and usually a damn good one), I doubt if you would level a bunch of snide remarks at the managers and programmers, as you did in your letter. I can speak for at least two other programmers (and who knows, maybe the bulk of the creative people in the business) and myself when I can do nothing but sadly shake my head after reading your letter. Your attitude is so typical. You fail to recognize a fundamental truth about our industry. We *are* SHOWBIZ . . . and we have COMPETITION.

Yeah, I would love to be able to boast about chiefing a station that offers response from d.c. to X-ray, totally flat, with 0.000001 per cent t.h.d., but I'm stuck with 250 watts at night. And being heard beyond the parking lot of the station is, surprisingly enough, important to me. What pays my salary, and yours, friend, is not a good proof, or "good engineering practice," but SALES. Like it or not, sales are dependent upon the number of listeners I can get, whether or not I'm in an ARB-rated market.

I read *your* trade publications; when was the last time you read one of mine? And I don't read them to get tips on what sort of processing "junk" to requisition. I just want to know, and *need* to know, what's going on in every area of this business.

You reveal your ignorance of the business of programming in the fifth paragraph of your letter. You assume that stations edit long music cuts and speed up turntables by 2 or 3 per cent simply to "pack in more records per hour." Why don't you ask the program director of your station why this is done? The answers will surprise you, believe me. You may even start to cultivate an interest in something about your station other than volts, current, deviation, and being sure the tech logs are filled out and signed.

I guess all I'm asking is for you to give us (the programmers) credit for a little common sense. We have ears. If we tell you we need 125 per cent on positive peaks, we're doing it for a reason. If the result is a horribly distorted signal, we *will* admit that our idea was dumb. RADIO'S GOAL IS LISTENERS . . . AND THEM THAT'S GOT THE MOST, GETS THE BUCKS! Then we all get rich.

But a bitch note should always contain a positive thought to make it palatable. On the plus side, you're the guy who keeps me on the air. You're the guy who drives twenty miles to the station in the middle of the night simply because some new jock who hasn't been properly trained is having a minor problem with a turntable that he could easily program around. You're the guy the production director hates because you've got his studio torn down, trying to isolate a problem that you've been working on for three solid hours.

The program director hates you because his 25-year-old QRK won't start in 1/16 of a turn. The manager can't stand the sight of you because you had the gall to ask him to sign a purchase order for 68¢ for a toggle switch. The manager's wife hates you because you haven't been able to locate a free replacement antenna for her Mercedes, and the salesmen hate you because you don't smile when they "Good morning, Bill," when your name is Steve. Especially after you've been up since 7 a.m. the preceding day because you're trying to make some antique meet proof.

We ALL (engineers, peddlers, and programmers) have the same goal . . . MONEY! Why can't we quit bitching, practice the admittedly difficult art of COMPROMISE, and GET ON WITH BUSINESS.

CHRIS EDWARDS
Bakersfield, CA

■

PATRICK S. FINNEGAN

db Broadcast Sound

System Grounding

• A good ground and shield system plays an important role in the quality of the audio a station produces. Although these do not contribute directly to the signal fidelity as do such factors as impedance, phase, and so forth, the grounds and shields operate to keep the audio signal free from noise, hum, and sundry interfering signals.

A REFERENCE

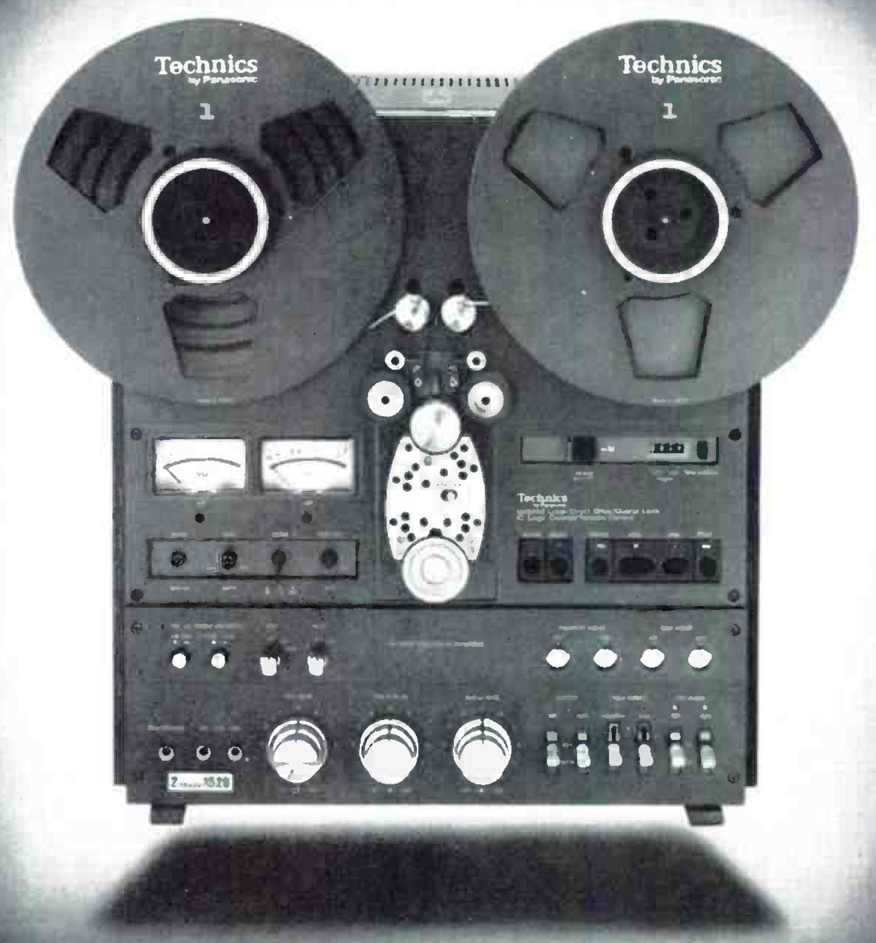
All the various signal and other voltage elements within the studio are somewhat relative to each other, yet all need some common reference point. The most generally accepted reference point is Mother Earth. Without this common reference, undesirable voltage potentials can develop between circuits, which can introduce hum and other extraneous signals into the audio circuit path.

Although accepted as the ultimate reference, earth ground is not always readily available. Neither is it practical nor desirable to directly connect each component unit of the system to earth. Isolation of some circuits from others in the system is often an important factor, so one or more sections of the system will have isolated grounds. Although separated from each other, these will eventually terminate on earth ground so as to maintain the common reference. Isolation of some grounds from others is similar to the practice used within amplifiers, for example, the signal ground separate from the power or chassis ground.

GROUND CURRENTS

A ground is not some "sink hole" into which we dump all undesirable signals and they just disappear. On the contrary, the ground becomes a part of the circuit which is attached to it. The undesirable signals or voltages which are routed to ground will cause currents to flow in the ground itself. When a ground bus has a variety of equipment units connected to it, a variety of signal elements are flowing in that ground. The combination of the currents caused by all these signal elements in the ground is essentially the sum of all the elements. The total of these currents will be higher than that of a single element, and in some cases much higher than that of an audio circuit attached to it. These ground currents create fields that can induce undesirable signals into sensi-

You were so impressed by the RS-1500 you asked for a studio version. Introducing Technics RS-1520.



The RS-1520 has all the performance of the award-winning RS-1500 plus the features you need in a studio deck. Like bias and equalization fine adjustments for each channel to optimize any tape formula. A 1kHz/10kHz test-tone oscillator for accurate equipment checks. The precision of ASA standard VU meters with a +10dB sensitivity selector. A Cue/Edit switch for quick, safe edits. And balanced, low-impedance, XLR-type output connectors to match other widely used broadcast and studio equipment.

To match the performance of its predecessor, the RS-1520 features the "Isolated Loop" tape transport with a quartz-locked, phase-controlled, direct-drive capstan. By minimizing tape tension, it virtually eliminates all signal dropout. While reducing modulation noise and wow and flutter to a point where they are barely measurable by conventional laboratory equipment.

Electronically too, Technics RS-1520 provides professional control. A separate microphone amplifier. Record amplifier. Mixing amplifier. An IEC standard playback equalization selector. While IC full-logic function permits absolute freedom in switching modes.

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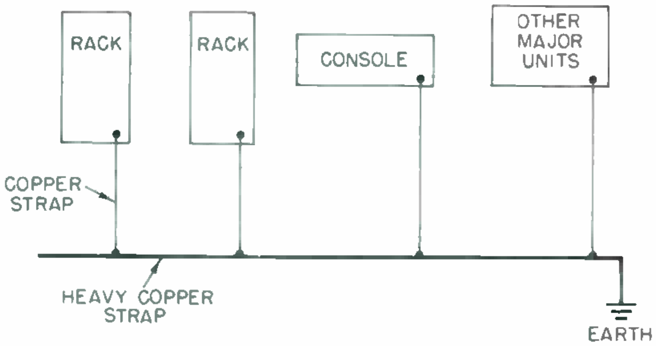

Figure 1. The building provides a controlled reference for the entire studio.

tive audio circuit paths. This is particularly true of an unbalanced audio circuit which makes use of the ground as one side of the circuit.

Because of the many currents which flow in a ground circuit or bus, it should be a very low resistance circuit. Should resistance develop because the conductor is too small, or where it joins other grounds, the ground currents will seek a lower resistance path which could be through the shield of an audio cable. Corrosion is a bad actor, and if corrosion should develop at a juncture of the ground bus, rectification or non-linear passage of the ground currents can occur. If a particular audio circuit makes use of this ground as one side of its audio path, then distortion and intermodulation distortion can occur in the audio itself.

BUILDING GROUND

A common *building ground* for the entire studio or station is the best approach to grounding. This will provide a controlled ground for the entire audio system within the station. This ground bus must be attached securely to earth ground at one or more points so as to maintain the earth ground reference. The incoming a.c. power is directly connected to earth ground, so unless the building ground is also attached to earth ground, voltage potentials between it and the power system can develop and induce hum, transients, and other noise factors into the audio system. But the building ground should attach to earth ground with its own direct connection and not through the power source connection. The power ground will carry heavy a.c. currents, transients, and noise that ride in on the power line. These should not be allowed any chance of entry to the building ground.

It is essential that the building ground be very, very low in resistance because of the many currents flowing through it. A heavy copper strap from 4 to 12 inches wide should be used, the width selected for the station situation. A large station with many equipment units and numerous cables would employ a wider strap. But even a smaller system that is in a strong rf field should have a strap wider than the minimum suggested.

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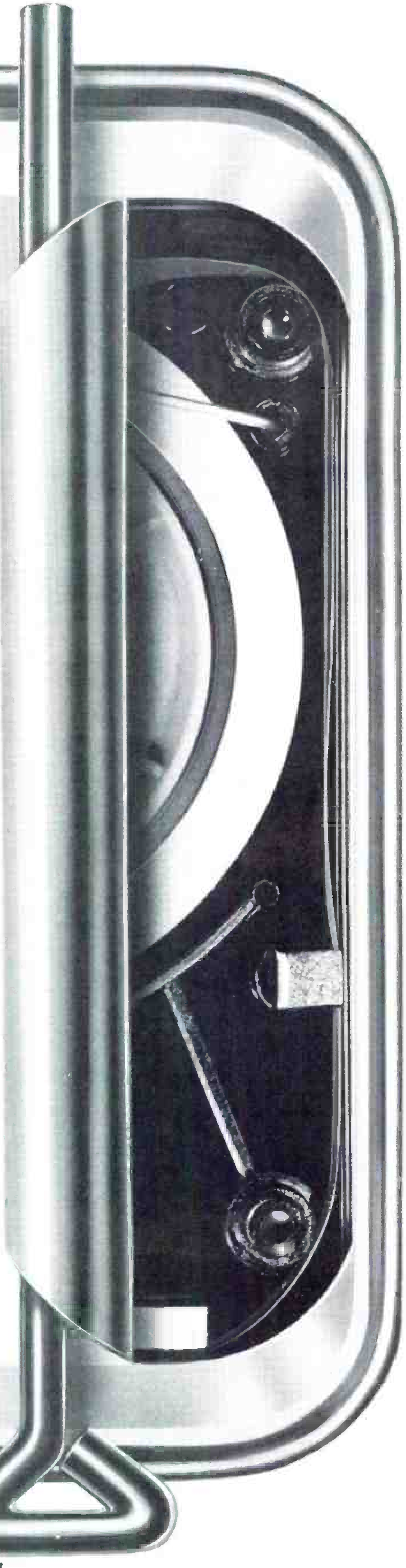

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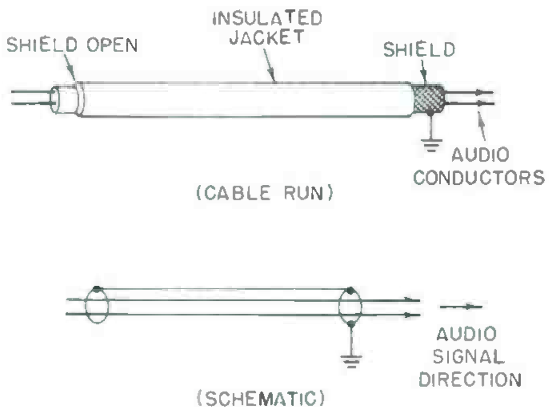

Figure 2. The controlled shield ground provides a drain to ground but an incomplete circuit for Circulating ground currents.

It is not practical to make the building ground with one single piece of copper strap, it would be too unwieldy to handle. But the smaller sections of strap should be mechanically and electrically bonded together so as to make one effectively continuous piece. Installation in a large room often makes possible a long piece of copper strap, but in areas where lightning occurs frequently, such a long ground can provide its own hazards. Transients from lightning surges, though brief in time duration, are extremely potent. The rapid transient is in the nature of an rf signal, so the strap length is important. The length of the strap can be inductive to the transient pulse, and an extremely high inductive voltage can develop which may flash over to audio units and destroy them. If a single section of strap must be over 20 feet in length, break up its inductive value by running one or more parallel straps, or by using a grid system.

Each major unit of the station system, such as an equipment rack, console, etc. should have its own ground strap that is attached to the building ground. This can be a 1 or 2 inch strap attached securely to the rack frame and the building ground. The individual units within the rack attach to this strap also. Besides mechanical connections, soldered electrical connections are an important consideration. In this manner, each individual major unit of the system has its own controlled ground bus.

SHIELDING

Important as the building ground and major units grounds are to the system, the audio signal circuits still need further protection. This is done by a controlled cable shield system for each circuit. In this system, each circuit is run in a shielded cable, the shield is insulated from other shields by a plastic or similar jacket, and the shield of a cable run is tied to the major ground at *only one end* of the cable run. By grounding the shield in this manner, there is no complete circuit for any currents that may be picked up on the shield. When installing the system, there should be consistency in method so as to avoid tying down both ends of a run which would allow ground currents to circulate in the shield. As a matter of practice, tie the shield to ground at the end of the run according to the audio signal direction.

(continued)

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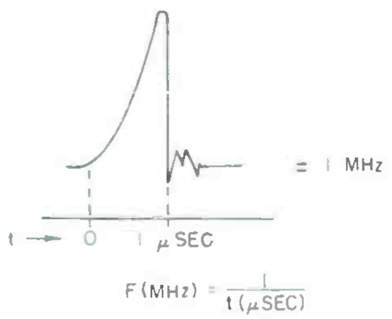

Figure 3. Transients and similar pulses are in rf regions of the spectrum even though they may be created by very low frequency devices. Shown is a 1 μsec transient.

In broadcast practice, a balanced audio system is used, this means that both sides of the audio circuit are above ground. The audio cable should have a pair of audio wires, a large coverage shield (preferably with a separate ground drain wire), and an insulated jacket. There will be occasions when a particular circuit must be run as an unbalanced circuit. But use the same type of cable. Although one side is grounded, it will be isolated somewhat from any shield currents that may be present along the run.

RFI AND EMI

The rf carrier from a nearby transmitting antenna, as well as the sundry other types of electromagnetic radiations, can enter the audio system and create a variety of noise and interference problems. Transients and similar noise bursts fall into this category and are really rf in nature. A transient or noise pulse, for example, which has a time duration of 1μsec (from onset to decay of voltage gradient) is a 1 MHz pulse. All signals of this type can enter the system by direct radiation or be carried on metallic conductors. In an environment where such signals are strong and prevalent, pay special attention to the shielding and grounding.

Enclosed equipment racks and cabinets are helpful against direct radiation. The covers and doors should make good metal-to-metal contact, and each unit bonded to the building ground. Avoid long ground leads on individual equipment units because these can become antennas to rf noise and pulses. If a ground lead must be long, shield it as you would an audio cable.

There is yet another facet to consider—metallic conductors have both a d.c. resistance and an rf resistance. This is due to "skin effect" in that the

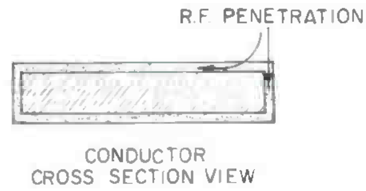

Figure 4. Rf currents only travel on or just below the surface of a conductor. This reduces the cross-section area and produces a higher resistance to the rf.

rf currents flow only in the surface of the conductor, presenting a much smaller cross-section and thus a much higher resistance than the d.c. resistance. This is why a wide copper strap is recommended for the building ground if strong rf fields are present.

CHECKOUT

When a new studio has been installed (or a new section), check out the grounding after its completion. Look for individual cables that have been grounded at both ends by mistake, cables which have been left ungrounded at both ends, connections that were left unsoldered (or cold joints), or shield grounds that may have broken off during construction of later circuits.

Visual inspection will generally

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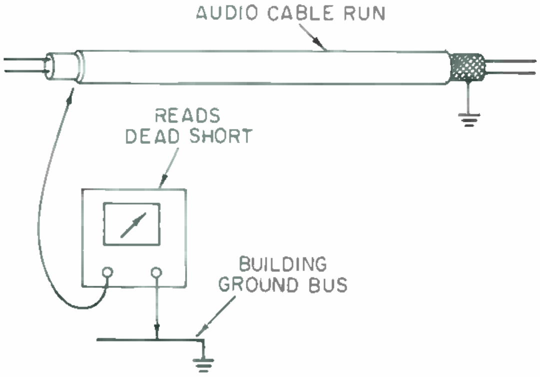

Figure 5. Use an ohmmeter to check for proper shield grounding.

show up shield grounds which have been left unsoldered, or what might appear to be a cold solder joint. An ohmmeter can also be helpful in the checkout of the shield grounding. Go to the end of the cable which is left ungrounded and measure the resistance between the shield and the building ground. There should be zero resistance on the lowest scale of the ohmmeter. If there is an indication of several ohms (unless an extremely long cable is used), there is a poor connection at the other end of the cable. But if the reading is infinity, then the other end wasn't grounded or the connection is not making an electrical connection.

LATENT PROBLEMS

Although a new system checks out okay, that doesn't mean it will remain that way or be as effective forever. There may have been no strong rf field present at the time of installation, but a year later, a new station comes on the air and locates its transmitter and antenna on a building adjacent to your studio! Other problems can develop, such as corrosion at a section in the building ground because a corrosive solder flux was used; shield wires may break off when new circuits are pulled in and so forth, just to name a few. When grounding problems develop, they can be a real headache to the audio quality and to the engineer trying to solve the problems.

RECAP

A good ground and shield system will improve the audio quality by keeping out hum noise, and other interfering signals. Construct a heavy, common building ground and use a controlled cable shield system. Make a careful installation and then completely check out the system after installation to insure that all connections are clean, tight, and securely soldered. ■

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db Theory & Practice

• I am still getting letters about damping factor, and how damping works. Some of them refer to the voltage generated by a voice coil, when it moves, as being a "feedback voltage," with the implication that it gets fed back to the input of the amplifier if it has a good damping factor. Well maybe that is a way of looking at it but it can be misleading.

The simple fact is that no connection is made from the loudspeaker voice coil to the amplifier input unless you have a system with what is known as "motional feedback." Years ago experiments were made with loudspeakers equipped with feedback voice coils. One way to do this was with the kind of voice coil that was double wound so it could be connected in series to make a 16-ohm coil or in parallel to make a 4-ohm coil. More sophisticated designs separated the two coils to avoid any electrical coupling between the coils themselves, but however it was done, the idea was the same: use one coil to drive the loudspeaker cone, and the other one to derive a voltage proportional to the movement, for use in feedback.

ELECTROMAGNETIC INDUCTION

Before we go any further, a mistake we often encounter springs from failure to understand basic electromagnetic induction. Such a voltage appears at the ends of a moving coil only when the coil moves. To verify this, you can connect a voice coil to a sensitive microammeter and move the cone manually. When the cone is stationary, there is no reading on the microammeter. When you move the coil, the microammeter deflects. But it deflects only while the coil is moving, and comes back to zero when the coil stops moving, whether that is at its position of rest, or somewhere else, such as fully forward, or fully back.

When the cone is moving forward, the microammeter shows a current—and, of course a voltage, to drive the current—in one direction, while when the cone moves backward, the current and voltage direction reverses. It is the direction of movement that controls the polarity of voltage or current—forward, one way, backward, the other way. Its position at an instant does not influence the polarity of voltage or current.

The magnitude of voltage and current depends on the rate of movement. If you move it forwards or backwards faster, it produces a bigger deflection on the microammeter. The reading does not depend on the distance you move it, but on how fast you move the cone. The faster it moves, the bigger the reading.

Thus, if it moves only a small distance, but at much higher frequency, the reading will be bigger. Reducing the frequency, for a given amplitude of movement, reduces the voltage induced by the movement and the peaks of voltage occur as the cone goes through its middle position, where it is moving faster, while they register zero at both ends of its travel.

ELECTRO-MOTIVE FORCE

Having got that straight, we hope, this voltage is generated in the coil, whenever it moves. If it moves due to current in the voice coil itself, supplied by the amplifier, then the movement produces just the same voltage. But in this instance, there are other voltages in the same circuit, that make this voltage due to movement, or back e.m.f. (electro-motive-force, in case you did not know what those letters stood for) not so easy to isolate, or measure. But it is always there, whatever else is also there.

When the amplifier feeds current into the voice coil, the coil resistance produces a voltage drop, sometimes called an "IR drop," because the voltage is found by multiplying current (I) times resistance (R). So now we have at least two votlages associated with the voice coil—one due to its movement, the other due to the current driving it and its resistance.

But a voice coil has inductance, as well as resistance, which produces yet another voltage. If the coil was just lying on a bench, not attached to a loudspeaker, and you fed the same current into it, you would find that at low frequencies the voltage and current would be in phase, and their relationship would be determined solely by the voice coil resistance.

However, as you run frequency up, the voltage and current would get out of phase, because of the reactance due to the voice coil's inductance. With the coil lying on a bench, not associated with the loudspeaker, it would be an air-cored inductance. But when you place it in the magnetic air gap where it works as a loudspeaker, the coil has magnetic material close to it, so it behaves at least partially like an iron-cored, or magnetic-cored, inductance.

This means that, even if you wedge the voice coil in the gap so it cannot move, the voltage produced by a given current, of a higher audio frequency, will be greater than it would be with the same current when the coil is lying on the bench. Now you have, at least approximately, the effective voice coil inductance when it is used as part of the loudspeaker.

MOTIONAL IMPEDANCE

But unwedge the voice coil, and things change again. Now it is behaving as a loudspeaker, and voice coil movement is producing voltages as well, as we described at the beginning. Assuming you have only one voice coil, which is still the usual arrangement, in spite of all the experiments with units having more than one coil, the relationship between voltage and current in the coil is the impedance of the loudspeaker, at least from the amplifier's viewpoint.

It should be obvious by now, that the relationship between voltage and current will change with frequency, due to both voice coil inductance and what is called "motional impedance," the fact that the coil moves, to drive the cone and produce the sounds we hear from it.

So motional impedance is, in some senses, the same as "back e.m.f.," using a somewhat different term of reference. Back e.m.f. is a measure of the voltage itself, while motional impedance relates that voltage to the current that causes the motion to produce the voltage.

Motional impedance also ties the action to something else. How much the cone moves depends on the acoustic impedance coupled to it by the design of the enclosure in which the loudspeaker is mounted. You can check this by connecting a loudspeaker unit to an amplifier delivering preferably a relatively low audio frequency, and changing the position of the unit, relative to its normal housing.

If you hold the unit out in space so the outside edge of the cone has nothing against it, the cone will move quite a lot, but you will not get much sound at low frequencies, because the air just shuffles round the edges rather than creating a sound wave. Now, if you put the unit in its mouting hole, or

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over the edge of a work-bench so as to stop some of the air-shuffling movement. the cone movement reduces, but you hear the sound a little more.

What happens to voice coil impedance when you do this? Movement reduces. so back e.m.f. must reduce. But presumably the amplifier is still producing the same output—as a voltage. If there is less back e.m.f., the voltage must be made up somehow else, either with IR voltage, or IX voltage—due to the coil's inductance. To increase either of those will require more current.

INCREASING CURRENT

So putting the opposition into the voice coil and cone movement acoustically increases the current drawn from the amplifier, at the same voltage and frequency. And it will increase the sound you hear, even more. You can play with the enclosure a lot more than that, of course. Whether the enclosure is of the infinite baffle type (totally sealed, except for where the cone goes) or of the reflex type (with vents somewhere else), opening and closing the back will make further differences.

But here we want to look at what damping does, and how it does it. You conduct all those experiments, using a steady frequency fed through the amplifier, and changing various other things to see the effect. While changing damping factor will change the effect of different current demands by the loudspeaker, that is not its main purpose.

DAMPING FACTOR

Damping factor is equivalent to a related internal resistance of the amplifier. If the load impedance is nominally 8 ohms, and the damping factor is 10, the internal impedance is 0.8 ohm. Now, you have the amplifier set so that with no load its output voltage is, say, 12 volts, at whatever frequency.

You connect an 8-ohm load, and that drops to about 10.9 volts. But if you connect a 4-ohm load, it will drop to about 10 volts. If the internal resistance is greater, which means the damping factor would be smaller, the change in voltage as impedance changes will be bigger.

As any loudspeaker has an impedance that changes with frequency, using different damping factors will also change the frequency response of the loudspeaker because it will change the voltage delivered at different frequencies. But that is not the important purpose of a damping factor.

Going back several decades, a principal was established that we seem to have successfully violated, that the bass response depends on the size box in which a loudspeaker is housed—a bigger box is needed for lower bass response. On this basis, a loudspeaker to reproduce down to 20 hertz would need to be as big as the average living room.

Loudspeaker designers have been successful in "cheating" on this principle by using resonance effects to hold the bass response up, using much smaller boxes. But this has its disadvantages. Once you get such a loudspeaker responding to a note in the lower octaves of its range, if the note stops abruptly, the loudspeaker tends to keep going. You have what is termed *transient distortion*. This is one thing that damping factor can stop.

Take a loudspeaker unit, not connected to an amplifier and tap its cone, or diaphragm. You will hear some kind of thump, produced by the movement of the cone in response to your tap. If the coil is not connected to anything, which electrically we call "open circuit," you will hear a little of the characteristic resonance of the system, used to help maintain bass response. The sound will be a real deep plop.

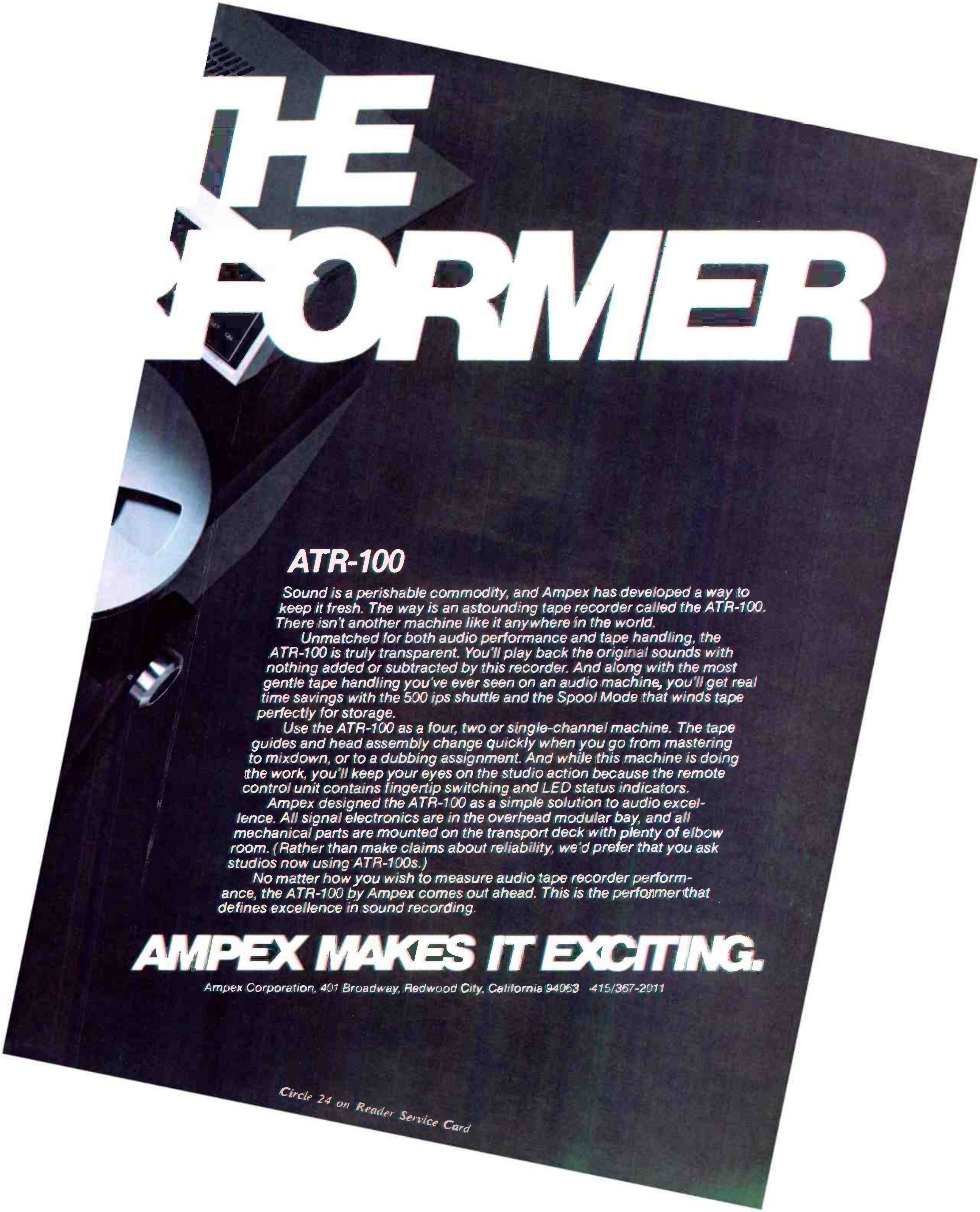
Have you ever thrown a large rock into a pool? If the pool is deep, like it is at the bottom of some waterfalls, the sound it makes tells the depth of the water. Shallower water makes a sound of higher pitch. When you hit the cone of a loudspeaker with its voice coil open circuit, you get the deep sound.

But now, strap the voice coil terminals with a piece of wire to short-circuit them. The sound changes. The cone sounds much "tighter," because the low resistance across the voice-coil terminals damps its movement. This is what a low internal resistance in an amplifier can do.

Thus, all the while the amplifier is producing the low frequency drive, the unit keeps resonating, according to its design, to reproduce that frequency. But when the frequency stops, the amplifier looks like that short-circuit strap across the loudspeaker terminals, to stop the movement quickly. How does it do that? Does it involve feedback? Perhaps that is a matter of viewpoint, or how you explain it.

But the fact that you have only a 2-wire connection between the amplifier and the loudspeaker means that there is no separate, external feedback loop, as there might be if you had a loudspeaker with two voice coils. ■


AMPEX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Editorial

Dear db Magazine:

I'm planning to build a new audio facility, so please tell me everything there is to know about consoles, microphones, tape recorders, monitor systems, et cetera."

Well, that's not *exactly* what the letter said, but it's not that far off, either. In an effort to oblige, we've covered some of these topics, and next year we'll get around to some of the others. Of course, we can't say *everything* but we hope we're covering at least a little something of interest in every issue.

However, what about a little something on *et cetera*?—you know; all those odds and ends that are so easy to ignore when planning the "super-studio." For example, ask anyone to name, in sequence, the major items in any audio signal path. You'll probably get an answer more or less like this. "Well, let's see now: there's the microphone, the mic preamplifier, an equalizer, fader, line amp, then the tape recorder, more amplifiers and finally the monitor system. That's about all there is to it, if you don't count signal-processing devices."

Now just a minute! Haven't you forgotten something? What's holding that microphone up? How does the signal get from the microphone to the console? What about direct boxes and such? AHA! You overlooked all that stuff, didn't you?

Well, it's no wonder that you did. Microphones and consoles are certainly more interesting than the plugs and cables that connect them. And so, these "nuts-'n-bolts" get taken for granted—all but forgotten, and lumped together under the heading of *et cetera.*

So, this month we make up for our past neglect, by taking a look at what we're usually guilty of overlooking. But first, we examine the mysteries of the PLL—or phase-locked loop—which seems to be cropping up all over the place. You've no doubt just read about the latest whiz-bang box which owes its success to an "all-new, phase-locked loop circuit." Very impressive. But just what *is* a PLL? After asking, and being asked, several times, we decided it was time to find out. Les Hadley at Signetics offers us an introduction to the PLL in this month's lead story, **The Integrated Phase Locked Loop.**

Later on, we'll be back with some more information about the application of PLL's in audio, but now we move on to examine **Audio Cable: The Neglected Component,** in which author Paul Miller provides some guidelines to selecting the right cable for the job. Do you know when to choose solid wire instead of stranded? How do you find pair #14 in a 27-pair cable? Are you using the right kind of microphone cable in your studio? You may be better able to answer these questions after learning something more about the neglected component.

Miller's story is followed by our brief application note which cross-references a handful of often-used audio cables. But, as most manufacturers are quick to point out, some cables are "more equivalent" than others, or make sure you check *all* the specs before ordering 10.000 feet of anything.

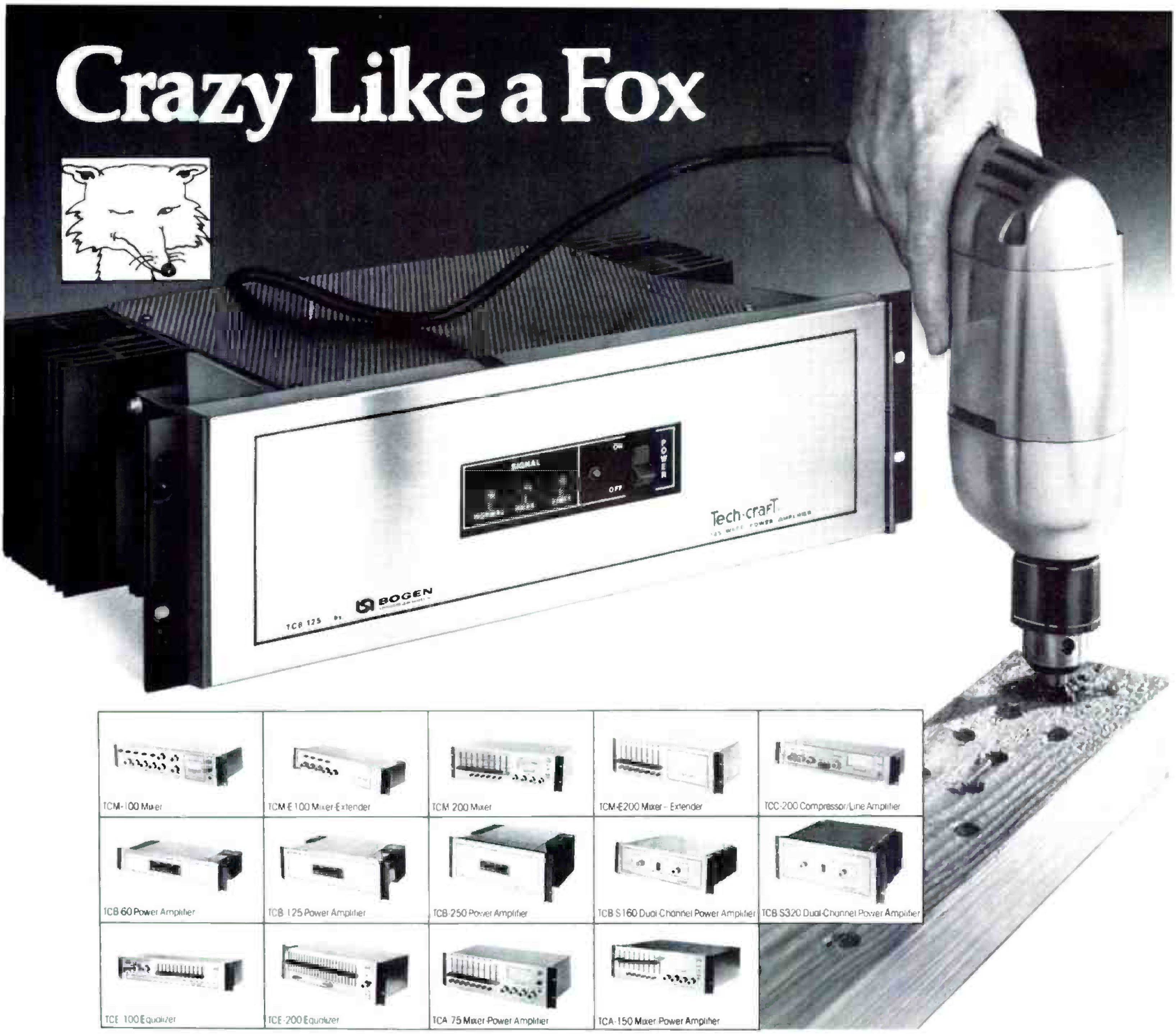
Next, Franklin Miller (no relation to Paul) offers us some help in selecting the right interface box. Whether building or buying. you'll find some useful information in his story.

Perhaps its ironic that. with all the fortunes that have been spent on record-making hardware. many studios seem to regard record-*playing* hardware more-or-less as an after-thought. If you're thinking about rectifying that oversight, read Larry Zide's review of the Technics model SL-1000Mk II turntable.

And finally. we conclude with John Woram's article. **Nuts-'n-Bolts**—a wrap-up of even more of the even-less spectacular items that can go such a long way toward pulling any studio operation together.

Before signing off. we can't help but remark about the interest our readers are showing in broadcast audio—check this month's **letters to the editor,** and you'll see what we mean. But most of these letters are from broadcasters. And that brings up the question; What does everyone else think about broadcast audio? There's at least one letter in this month's mail bag that should provoke some response from the recording engineers in the crowd. See if you can find it, and then let's hear from you.

J.M.W.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TCM-100 Mixer
TCM-E100 Mixer-Extender
TCM 200 Mixer
TCM-E200 Mixer – Extender
TCC-200 Compressor/Line Amplifier
TCB 60 Power Amplifier
TCB 125 Power Amplifier
TCB 250 Power Amplifier
TCB S160 Dual Channel Power Amplifier
TCB S320 Dual Channel Power Amplifier
TCE 100 Equalizer
TCE 200 Equalizer
TCA 75 Mixer-Power Amplifier
TCA 150 Mixer-Power Amplifier

LES HADLEY

The Integrated Phase Locked Loop

A system-in-a-chip, the integrated phase-locked loop does an efficient synchronization task in many applications.

THE INTEGRATED phase locked loop (or pll) can rightly be called the "system-in-a-chip." As one might expect, this system or block, may be used in a multitude of different applications, from the reception of both a.m. and f.m. radio signals to the decoding of telephone dialing codes—frequency shift keying (fsk) and pulse code modulation (pcm), as well as many more.

Although the concept behind the phase locked loop has been around since the early 1930's, its use had not been easily realized in system design until the late sixties, when the Signetics Corporation developed the first monolithic, integrated phase locked loop. Prior to this time, its usage was limited, due to the complexity and expense of designing discrete phase locked loop systems. With integration came devices which were predictable in operation, versatile, compact, reliable, and above all, economical.

Les Hadley is an applications engineer with the analog division of the Signetics Corporation.

PLL OPERATION

Let's look into this powerful "system-in-a-chip" and see what makes it tick. In FIGURE 1 you will see the basic block diagram of a phase locked loop.

The input signal comes into the phase comparator, where it is "mixed" with the output signal from an internal reference oscillator, vco (voltage controlled oscillator).

PHASE COHERENCE

It is the primary purpose of the phase comparator to determine the "coherence," or degree of synchronism, between the input signal and the vco. If you have difficulty visualizing the concept of phase coherence, it may help to think of the timing sequence in an automobile ignition system. The spark timing occurs in fixed synchronism, or timing relationship, from the engine. The point in time when the piston reaches top-dead-center is used as a reference, from which the distributor is forced to advance or to retard the occurrence of the spark. It is this precise timing—or *phasing*—relative to the mechanical piston cycle, which is analogous to electrical coherence or synchronism between two signals being compared in the phase comparator (see FIGURE 2).

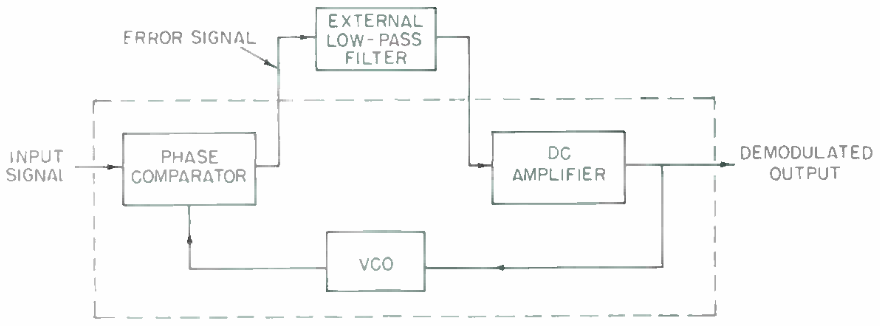


Figure 1. The phase locked loop block diagram.

CAPTURE AND LOCK

When the frequency of the input signal and the vco are equal, and the relative phase angle between them is constant, the phase locked loop is said to be *in-lock*. However, it is quite possible that the incoming signal may initially be different in frequency from the vco. This is where one of the unique characteristics of the pll comes into play, in the phenomenon called *capture*. When the incoming signal frequency closely approaches that of the vco (within what is called the capture range), an *error signal* is generated by the phase comparator, and fed through an external low-pass loop filter. The error voltage is initially a beat note equal to the frequency difference between the vco and the input signal. This varying signal modulates and drives the vco frequency toward the frequency of the incoming signal. The process continues to completion, at which point the beat-note error signal is reduced to a minimum and the vco frequency is equal to that of the incoming signal. Now the loop error signal is simply a dc voltage, proportional to the phase difference between the vco and the input signals. In the analog phase locked loop, once capture is complete, the system will follow slowly varying changes in the incoming signal, as in an f.m. system.

The vco is forced to run 90 degrees lagging with respect to the incoming signal when both signals are the same frequency. Another interesting feature of the pll is that the frequency range over which the system will remain in-lock is always greater than the capture range. Within certain limits, this capture range may be tailored to the needs of the designer, simply by changing the rolloff frequency of the loop filter.

THE EXTERNAL LOOP FILTER

At this point, loop filter design will not be discussed. Loop filters are external to the pll and are in themselves a complete design concept. Reference texts are available on filter design if needed. The combination of proper filter design techniques along with the characteristics of a pll combine to make a pll a very effective electronic filter. The filter design does permit some variations in the capture and lock characteristics of the pll. The ability to control the capture range enables the pll to selectively filter out unwanted signals which are outside the capture range and to latch on to those that are within. These principles are used for detection and recovery of various types of electronically coded information.

FIGURE 3 is a graphic representation of the capture and lock process, along with the respective phase comparator output signals.

In FIGURE 3(A), as the input frequency is swept toward the free-running frequency of the vco, ω_o, from a point below, capture occurs at ω_1. As the input continues to sweep upward, the vco is forced to run at higher and higher frequencies until a point is reached, ω_2 where the loop runs out of range (ability to track the incoming signal), breaks lock, and returns to the free-running frequency of the vco. In FIGURE 3(B), the conditions are reversed and the input sweeps down from a frequency above ω_o. The symmetry of the capture-and-lock range around the center, or free-running, frequency is characteristic of

Figure 2. The phase comparator determines the phase angle between a coherent input signal and the internal reference oscillator.

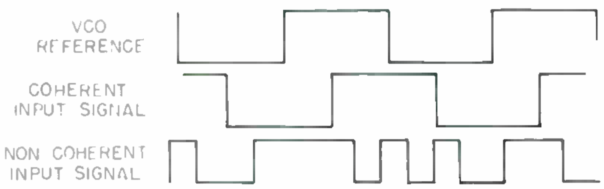
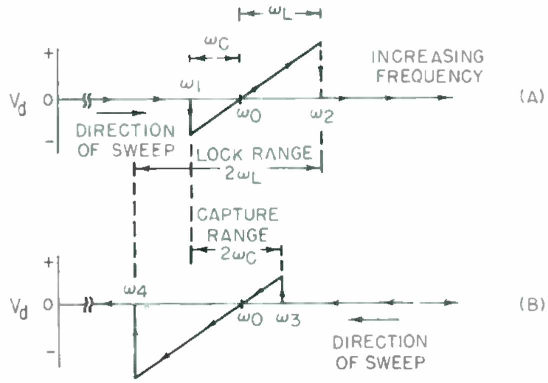


Figure 3. The capture-and-lock operation. Typical pll frequency-to-voltage transfer characteristics



ω_0 = Free-running frequency of the vco (internal reference oscillator.
ω_1 = Lower capture frequency.
ω_3 = Upper capture frequency.
ω_2 = Upper lock frequency.
ω_4 = Lower lock frequency.
$2\omega_C$ = Capture range.
$2\omega_L$ = Lock range.
V_D = Phase comparator output voltage.

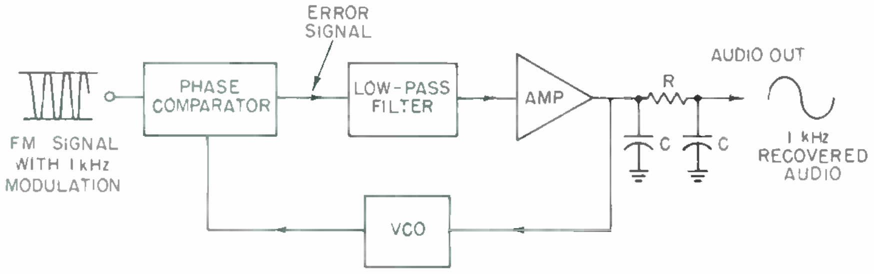


Figure 4. An f.m. demodulator block diagram.

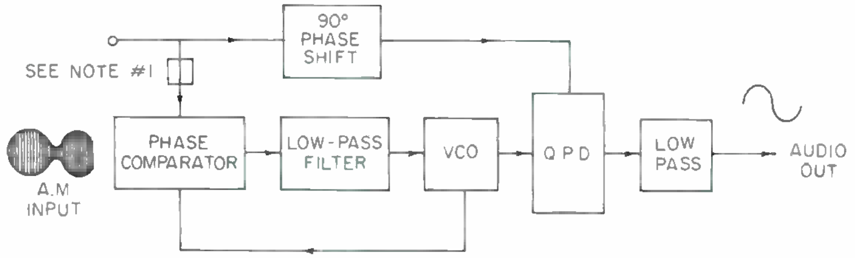


Figure 5. A pll a.m. synchronous detector block diagram. Note 1. The 90 degree phase shift network could be placed here instead (as in Figure 6).

an analog pll. Again, it should be remembered that the loop filter characteristics only affect the capture range and have no effect on the lock range. The actual phase difference between input signal and vco over the whole lock range can vary between 0 degrees and 180 degrees.

BASIC CHARACTERISTICS OF THE PLL

So far, we have covered in brief form some of the major operating principles of the pll. Before going into specific applications, let's summarize the basic characteristics of the pll.

1. The pll is a closed loop system which converts frequency and/or phase differences between an "unknown" incoming signal and a known reference into an error signal, which causes the *known reference frequency* to become synchronized to the unknown frequency.
2. Once capture has been achieved, the pll will follow slowly varying frequency changes in the unknown carrier instantaneously, thereby reproducing wave shapes with minimum distortion.
3. The pll can be controlled by external loop filter designs.
4. The pll is a stable system (single pole) and will not operate in an unstable mode unless poor system design techniques (specifically with regard to filter design) are employed.
5. The pll is an excellent building block for systems such as:
   a) FM demodulation
   b) AM demodulation
   c) FM Modulation
   d) Frequency synthesis
   e) Frequency shift keying (fsk) modems
   f) Pulse code modulation (pcm) techniques, etc., etc., et al.

SYNOPSIS OF AVAILABLE PRODUCTS & THEIR GENERAL USAGES

The following tables illustrate the wide range of pll's available, their general characteristics, and typical applications in which they will probably be found.

A.M. DEMODULATION

Less obvious is the use of the pll to demodulate amplitude modulated signals. This is the synchronous converter, or *synchrodyne,* first tried in the 1930's. Its use was more a theoretical novelty in the days of early vacuum tube technology, however, and the synchrodyne receiver was easily surpassed by the reliable *superhetrodyne.* The problem was stability. A local oscillator had to be precisely tuned to synchronize with the incoming a.m. carrier. The incoming signal was mixed and the product filtered to provide audio directly when the oscillator was exactly tuned. If the local oscillator was a little off, the output was a "hopeless garble."[1]

THE PLL AS AN A.M. DEMODULATOR

By making use of a 90 degree phase shift network and an extra quadrature phase detector (QPD), amplitude-modulated signals may be demodulated. Circuit simplicity makes this an attractive approach to certain signalling applications. It is not meant to be an alternative to the superhet receiver in broadcast reception, but it is ideal for low frequency signalling, such as WWVB reception, and carrier link tone detection.

THEORY

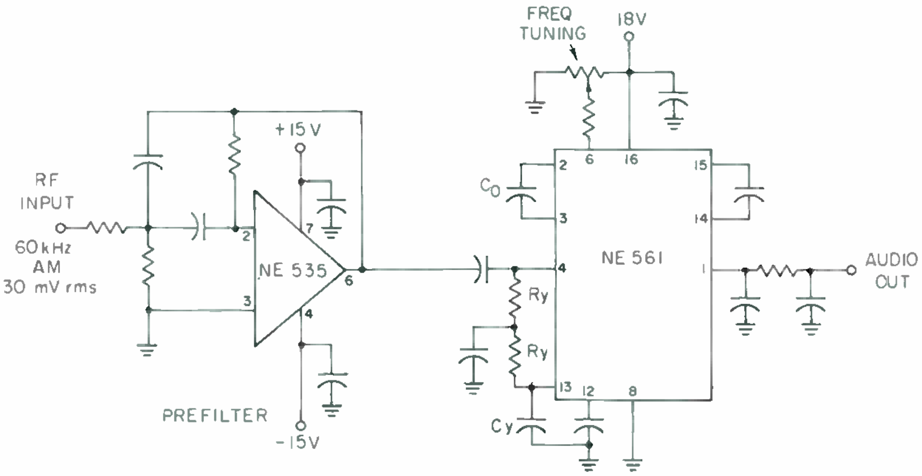
The pll a.m. detector differs in principle from the standard phase locked loop in that it requires two phase comparators. The second phase comparator is called the *quadrature detector* (or QPD) since the signal from the

Figure 6. A synchronous a.m. converter.

$$C_y = \frac{1.3 \times 10^{-4}}{F_0\,(H_z)}$$

$$F_y\,C_y = \frac{1}{2\pi F_0}$$

$$C_0 = \frac{300\text{pf}}{F_0\,(MH_z)}$$



a.m. input has been shifted 90 degrees relative to the main phase comparator. The block diagram shows the difference, compared to the standard pll, which detects f.m.

F.M. DEMODULATION

With the previously mentioned characteristics of the pll, let's apply the pll to an f.m. system, remembering that an f.m. signal consists of a carrier frequency with a modulating frequency superimposed (and this variation is small, less than 0.1 per cent).

In FIGURE 4, the following sequence occurs:

A. The f.m. carrier (10.7 MHz for an i.f. frequency) is compared to the internal reference oscillator. Since the known purpose of this particular pll is as an f.m. demodulator, the internal oscillator is set reasonably close to 10.7 MHz. The capture range of the pll compensates for variations due to temperature, component tolerance variations, voltage variations, etc.

B. Within a short period of time from the application of the f.m. signal (typically less than 10 cycles of the carrier or approximately 1 microsecond), the pll will lock onto the f.m.-i.f. carrier (10.7 MHz).

C. Once the pll has locked onto the carrier, any variation in the system will be tracked instantaneously (as long as the system is within its lock range). The audio-frequency variation becomes an "error signal," and is what appears at the system's output.

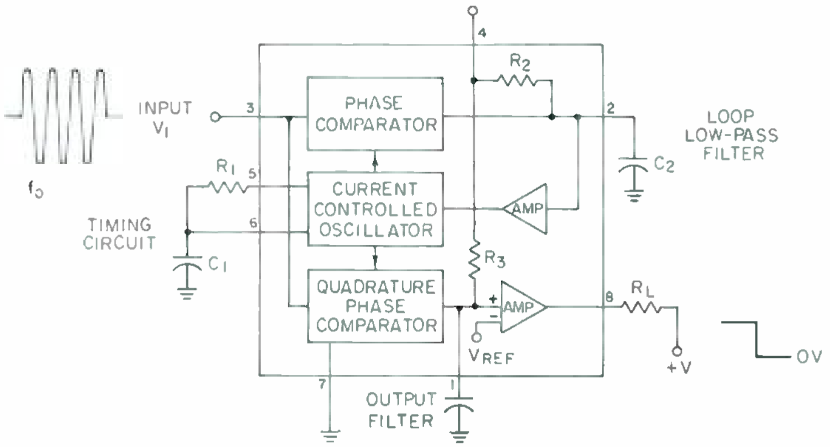
At this point mention should be made of the major difference between f.m. and a.m. demodulation. The f.m. demodulator operates as a function of the cosine of the error phase that is generated. The a.m. demodulator operates as a function of the sine of the error phase that is generated. Therefore, for f.m. demodulation, a 90 degree "error" angle is desirable, while a zero degree "error" angle is needed for an a.m. demodulation system. The additional phase comparator in the a.m. chip supplies the required phase shift (90 degrees).

CIRCUIT OPERATION

The circuit shown in FIGURE 6 is designed to operate at 60 kHz with an a.m. modulated signal. The NE 535 op amp is used to provide an MFB bandpass filter to improve selectivity and signal-to-noise. The signal from the filter feeds pin 4 of the NE 561 (the QPD), and in addition is phase-shifted 90 degrees and fed to the main phase comparator. Values of R_y and C_y may be calculated using the equations given. The center frequency is set by C_0 and fine tuning is accomplished by the 5k pot connected to pin 6. The capacitor across pins 14 and 15 is the loop filter. Its value determines the capture range, as discussed previously. The output pi filter at pin 1 is optimized for an audio bandpass of approximately 5 kHz.

A signal-to-noise ratio of better than 40 dB is possible for 30 mV rms input at 50 per cent modulation.

Figure 7. A pll tone decoder.



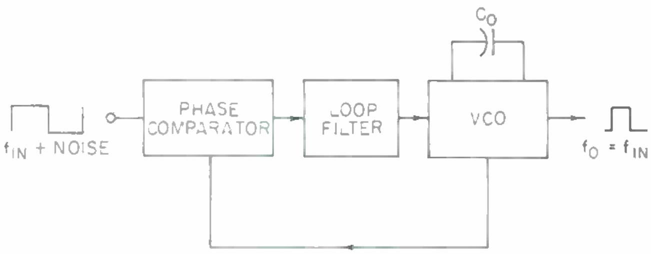


Figure 8. Signal regeneration and noise elimination.

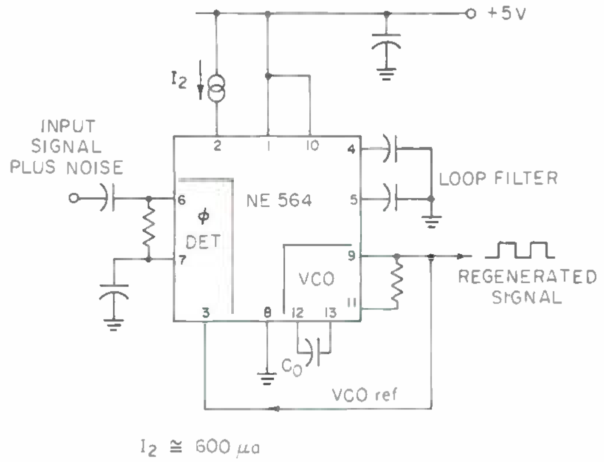


Figure 9. The noise filter circuit.

TONE DECODER

The tone decoder receives wide attention in the telephone industry, where it is valuable in all sorts of tone-signaling circuits. The output of the tone decoder is a d.c. logic level suitable for energizing relays or gates which automatically connect signal lines. The tone decoder uses two phase comparators in the loop to obtain lock information. Again, the second comparator is referred to as a *quadrature detector.* The tone decoder can be used as a tone burst a.m. detector. The linear range of the NE 567 tone decoder is limited and its functionality as anything other than a tone decoder is limited. An excellent application of this device is as a dual-tone, multiple-frequency (DTMF) decoder for use with telephone touch-tone signals. (*Touch-Tone* is a registered trade-mark of the Bell Telephone System.)

THE NE 567 TONE DECODING CIRCUIT

In the circuit shown in FIGURE 7, the NE 567 is used to detect tone signals within its capture range, which may be as narrow as ±5 per cent of the center frequency. Center frequency is determined by the vco free-running frequency. This in turn is set by the values of timing components R_1 and C_1. If a center frequency of 1 kHz were desired, for example, a value of C_1 equal to 0.1 μf would suffice. Note that the timing circuit (R_1C_1) is the heart of the circuit frequency accuracy and stability: therefore, quality components should be used. Capacitor C_1 should be a mylar or polystyrene device. The output is capable of sinking 100 milliamperes and goes low when a tone is received. Tone control links using a.m. or f.m. transmissions over radio, fibre optic or wire line are possible.

In essence, the normal pll vco output signal lags the incoming signal a nominal 90 degrees. The main phase comparator is referenced to this phase differential and is a cosine function; thus, its output is a null, or zero, when the pll is in-lock. In order for the QPD output to be a maximum when the pll is in-lock, an additional 90 degree phase shift is necessary, resulting in an output which is now directly proportional to input signal amplitude.

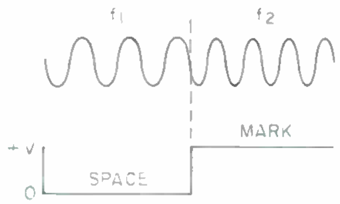


Figure 10. Frequency shift conversion. The carrier frequency is shifted down or up from its center frequency.

Figure 11. A pll with filter for fsk applications.

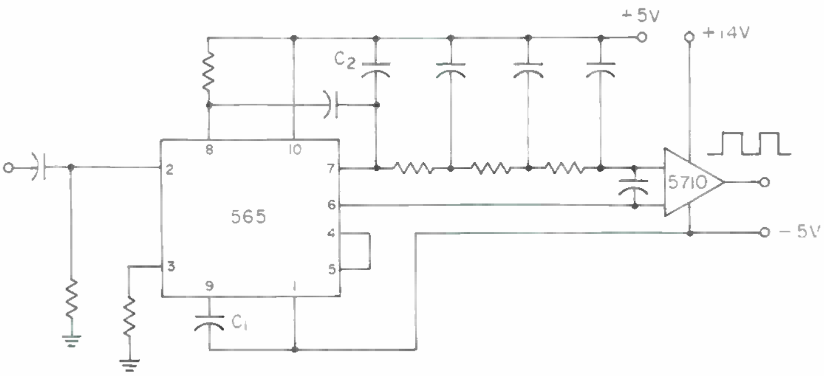
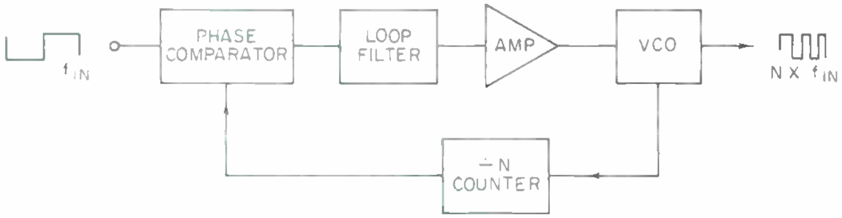


Figure 12. A pll frequency synthesizer block diagram.



Signetics Number	Upper Freq. (MHz)	Max. Lock Range (% F_0)	FM Distortion	Output Swing ±5% Deviation (Volts P.P.)	Center Frequency Stability (PPM/°C)	Frequency Drift/W Supply Voltage (%/V)	Input Resistance	AM Output Avail.	Typical Supply Current (mA)	Supply Voltage Range (Volts)
NE560	30	40%	.3%	1	±600	.3	2k**	No	9	+16 to +26
NE561	30	40%	.3%	1	±600	.3	2k**	Yes	10	+16 to +26
NE562	30	40%	.5%	1	±600	.3	2k**	No	12	+16 to +30
NE564	50	30+			±200++					+5 to +10V
NE565	.5	120%	.2%	.15	±200	.16	5k	No	8	±6 to ±12
SE565	.5	120%	.2%	.15	±100	.08	5k	No	8	±6 to ±12
NE567	.5	14%	5%*	.20	35±60	.7	20k**	Yes*	7	+4.75 to +9
SE567	.5	14%	5%*	.20	35±60	.5	20k**	Yes*	6	+4.75 to +9
NE566	.5	N/A	.2%	30%/V***	±200	.16			7	+12 to +26
SE566	.5	N/A	.2%	30%/V***	±100	.08			7	+12 to +26

*The 567 a.m. and f.m. outputs are available, but are not optimized for linear demodulation
**Input biased internally
***Figure shown is vco gain in percent deviation per volt
+With external control current
++@ 500 kHz

General Device Characteristics: High selectivity; noise immunity; bandwidth control; reference oscillator control; wide voltage operating range; wide frequency range; low power consumption.
These properties are accomplished with the use of very few peripheral components.

Table I. A quick-look guide to analog pll's.

NOTE—On Table I, don't overlook the General Device Characteristics listed below the tabulation.

NOISE FILTER

The pll may be used to lock on to a signal which is nearly buried in the noise. Once locked, the loop filter integrates out the noise pulses and the vco output provides a "cleaned up" version of the original CW signal (see Figure 8).

FREQUENCY SHIFT CONVERSION

Closely related to linear f.m. detection using the pll is the method of detecting digitally-encoded data signals which use fixed frequency shift keying or modulation techniques. Systems using this form of communication have been in use since vacuum tube days for the transmission of teletype signals. More recently, similar techniques are being used in computer modems.

The principle consists of having a center or carrier frequency which is frequency-shifted a fixed number of cycles above the center frequency for a *mark* and an equal number of cycles below f_0 for a *space*.

Early systems used selectively tuned L-C filters to detect fsk signals. Figure 11 shows a circuit incorporating the NE 565 as an fsk converter. Typically, the signal frequency is shifted ±5 to ±10 per cent for mark and space conditions with data rates of 300 BAUD (300 mark-space combinations per second, or 150 Hz). The signal is transmitted over wire line or radio link and fed into the NE 565 at a

Table II. Typical applications of phase locked loops.

Application	Device Type	Comments on Selection
Standard FM Demodulation	560, 562, 565	These are optimized for standard FM dev. of 75 kHz
AM Demodulation	561	The only PLL capable of synchronous operation
Low Freq. PLL Modems, etc.	565	Operates up to 500 KHz
High Freq. PLL CATV, Video, etc.	564	Operates up to 50 MHz
FSK (Telecomm, Modems)	564* 565	*Needs no external filtering
Modulator (Low Frequency)	566	This is a VCO
Modulator (High Frequency)	564, 562	The output of oscillator is available.
Single 5V Supply Operation (TTL Level Compatible)	564	The only one available
Freq. Synthesizers	564	The divider network can be TTL

level of 200 mV rms. The circuit is tuned to operate at a "mark" frequency of 1220 Hz and a "space" frequency of 1070 Hz. The demodualted signal is present at pin 7, in combination with phase comparator products of twice the center frequency. The purpose of the r-c ladder filter is to filter out those unwanted mixer products leaving the demodulated data signal. The low level signal is then fed into the 5710 comparator which produces usable output data.

During fsk reception, the pll remains in-lock at all times and its output signal may be explained by referencing back to the transfer function shown in FIGURE 3. Note that the greater the frequency deviation at mark or space conditions, the higher will be the peak-to-peak output signal and the better the signal-to-noise ratio of the converted signal.

THE NE 564 AS A FREQUENCY SYNTHESIZER

The monolithic phase locked loop has become very popular in frequency synthesis applications. The principle is shown in block form in FIGURE 12.

The vco signal is divided by the counter modulus and fed back to the phase detector. When lock is achieved, the input frequency will be multiplied by the modulus of the counter. If the divider number is changed, the vco output will track within the frequency range limits of the vco. Thus, if a reference frequency of 5 kHz is fed into the pll synthesizer, and the divider stepped in integral multiples, the output will equal (N) X (5 kHz), where N= the dividing modulus.

We have explored but a few of the uses of the modern phase locked loop. More exotic applications include the reception of radio signals from deep space and the coherent modulation of lasers. With a little practice, you too can be using the pll in your next design project. ■

REFERENCES

1. Gardner, Floyd, *Phaselock Techniques*, Wiley, 1966.
2. *Signetics Analog Data Manual*, 1977 Edition.
3. *Signetics Phase Locked Loop Manual*, Edited by J. A. Connelly.

PLL TERMINOLOGY

FREE-RUNNING FREQUENCY (f_0, ω_0)

Also called the center frequency, this is the frequency at which the loop vco operates when not locked to an input signal. The "prime" superscripts are used to distinguish the free-running frequency from f_0' and ω_0'which are used for the general oscillator frequency. (Many references use f_0' and ω_0' for both the free-running and general oscillator frequency and leave the proper choice for the reader to infer from the context.) The appropriate units for f_0' and ω_0' are Hz and radians per second respectively.

LOCK RANGE ($2f_L$, $2\omega_L$)

The range of frequencies over which the loop will remain in lock. Normally, the lock range is centered at the free-running frequency unless there is some nonlinearity in the system which limits the frequency deviation on one side of f_0'. The deviations from f_0' are referred to as the "Tracking Range" or "Hold-in Range." The tracking range is therefore one-half of the lock range.

CAPTURE RANGE ($2f_c$, $2\omega_c$)

Although the loop will remain in lock throughout its lock range, it may not be able to acquire lock at the tracking range extremes because of the selectivity afforded by the low-pass filter. The capture range also is centered at f_0' with the equal deviations called the "Lock-in" or "Pull-in Ranges." The capture range can never exceed the lock range.

LOCK-UP TIME (t_L)

The transient time required for a free-running loop to lock. This time depends principally upon the bandwidth selectivity designed into the loop with the low-pass filter. The lockup time is inversely proportional to the selectivity bandwidth. Also, lock-up time exhibits a statistical spreading due to random initial phase relationships between the input and oscillator phases.

PHASE COMPARATOR CONVERSION GAIN (k_d)

The conversion constant relating the phase comparator's output voltage to the phase difference between input and vco signals when the loop is locked. At low input signal levels, K_d is also a function of signal amplitude. K_d has units of volts per radian (V/rad).

VCO CONVERSION GAIN (K_o)

The conversion constant relating the oscillator's frequency shift from f_0' to the applied input voltage, K_o has units of radians per second per volt (rad/sec/volt). K_o is a linear function of ω_0' and must be obtained using a formula or graph provided or experimentally measured at the desired ω_0'.

LOOP GAIN (K_v)

The product of K_d, K_o, and the low-pass filters' gain at d.c. K_d is evaluated at the appropriate input signal level and K_o at the appropriate ω_0'. K_v has units of $(sec)^{-1}$.

CLOSED LOOP GAIN (CLG)

The output signal frequency and phase can be determined from a product of the CLG and the input signal where the CLG is given by $CLG = \frac{K_v}{1 + K_v}$

NATURAL FREQUENCY (ω_n)

The characteristic frequency of the loop, determined mathematically by the final pole positions in the complex plane or determined experimentally as the modulation frequency for which an underdamped loop gives the maximum frequency deviation from f_0' and at which the phase error swing is the greatest.

DAMPING FACTOR (Zeta)

The standard damping constant of a second order feedback system. For the pll, (zeta) refers to the ability of the loop to respond quickly to an input frequency step without excessive overshoot.

LOOP NOISE BANDWIDTH (B_L)

A loop property relating ω_n and T which describes the effective bandwidth of the received signal. Noise and signal components outside this bandwidth are greatly attenuated.

PAUL MILLER

Audio Cable: The Neglected Component

A guide through the labyrinth of audio cables— how to choose for insulation, flexibility, and ruggedness.

THE AVERAGE recording or broadacsting studio represents a sizable investment in such big-ticket items as multichannel audio consoles, preamps, mixers, tape decks, amplifiers, and assorted signal processing devices. Each of these components is chosen with great care, to make sure that the signal going to or from the console, or out on the airwaves, has the ultimate in fidelity.

But every bit as important to high-quality audio is a line of components that often doesn't get comparable consideration from engineers. These components are the electronic wires and cables that tie a system together.

Too frequently, electronic cable is taken for granted. While it may lack the glamour of a 20-channel console or a sophisticated tape deck, the myriad cables used in the modern studio must be carefully selected, or the cost of rectifying an improperly designed installation can be sizable.

Thumb through the typical electronic wire and cable catalog and you are faced with hundreds of different designs, each with diverse characteristics, including variations in conductor configuration, insulating materials, shielding, and jacketing. Selection may be confusing, but a cable design that is properly matched to an application can provide better performance, easier installation, and longer service life. With the trend toward miniaturization at all levels of hardware and componentry, a compact cable design may even permit tighter system packaging.

Analysis of various cable designs emphasizes the importance of cable construction elements. Shielding, for example, limits interference. In many signal-carrying applications, it is essential. And so is insulation; but the type of insulating material and its thickness in relation to the conductor determines capacitance. If capacitanceis too high, signal clarity will be jeopardized. The role of the outer jacketing, and its protective characteristic, is also important.

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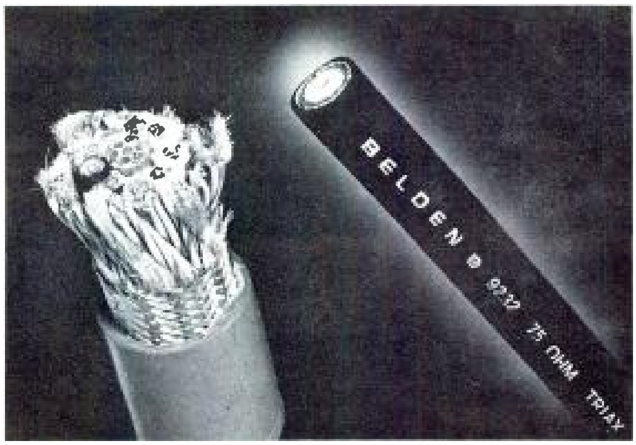
CABLE IS CHANGING

Before discussing some of the basics of wire and cable, it's interesting to note that significant changes have been taking place in cable design. Every year, wire and cable manufacturers come up with hundreds of new designs and design variations in response to the special requirements of broadcast networks, component and system manufacturers, recording studios, and other customers. Some designs eventually become part of the standard product line.

Probably one of the most dramatic examples of such change may be seen in t.v. camera cable. FIGURE 1 graphically illustrates how far we've come in developing improved camera cable. The t.v. color camera cable of fifteen years ago measured about 1½ in. in diameter, and contained 85 conductors. You can imagine the difficulty involved in dragging a cable of that size along a fairway to cover a golf tournament! Extra people were needed just as cable handlers, driving up production costs and limiting the degree of remote coverage in comparison to today. In addition to the problem of bulk, the camera cable of yesterday was comparatively costly; veteran engineers will

Figure 1. A comparison of t.v. camera cables—on the left, the 85-conductor cable of fifteen years ago. Next to it is today's triaxial configuration.



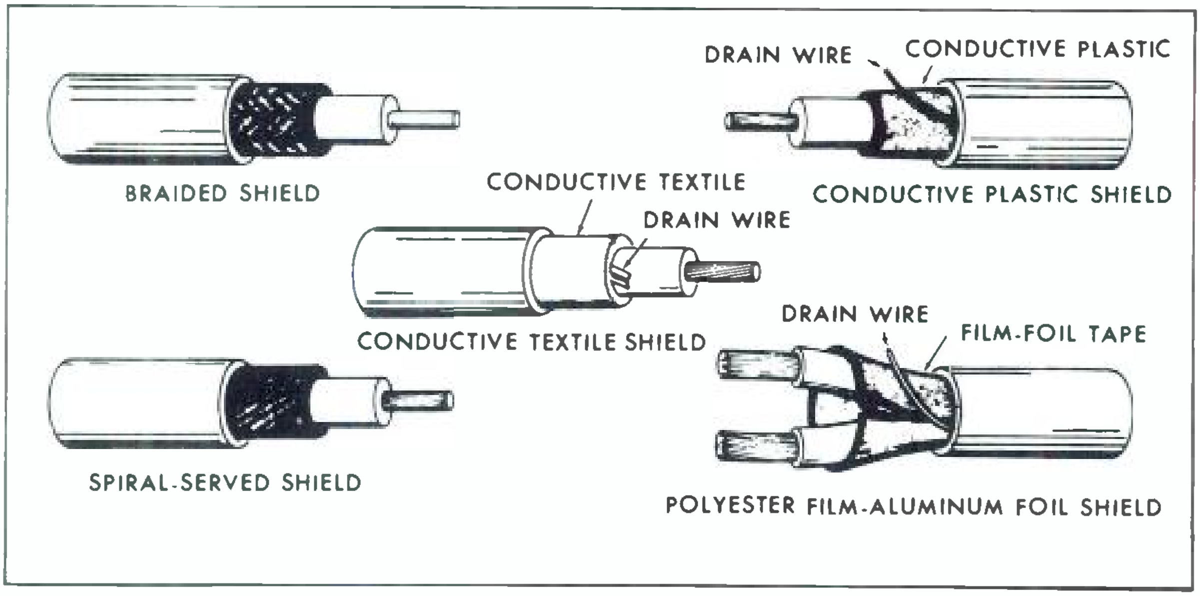


Figure 2. Various types of shields.

tell you it took an experienced man up to eight hours to attach the large (and expensive) multiple-pin connectors.

Today's camera cable is streamlined by comparison. It is a triaxial configuration, with two isolated braid shields and a center conductor. The functions are packed in a 0.360 outer diameter (o.d.) configuration—one-fifth the size of the old 85-conductor cable. Energy moves in both directions in this cable, with the braids serving as the path for power to the camera, and the core conductor along with the inner braid carrying the digital video signal. Triax is being used more and more by the networks and will become more common at the smaller independent stations.

On the audio side of the business, there have also been significant developments in cable, permitting longer runs without sacrificing signal integrity. One such development utilizes a new dielectric material. The dielectric, a cellular polyolefin called Datalene,™ offers lower capacitance than most other dielectrics. With it, longer mic and audio line runs can be obtained, without diminishing signal clarity.

BASIC ELEMENTS

Based on these examples, it is easy to see that more is going on in cable research and development than the addition of bright new jacketing colors. To understand the range of activity, here's a quick look at the basic elements that make up a cable, and how they affect performance:

Conductor—The signal-carrying conductor may utilize solid or stranded wire construction, depending on whether the cable will remain fixed in one place or will be moved periodically after it is installed. Solid conductors, though slightly less costly than stranded ones, tend to break after repeated flexing. As a result, solid is usually limited to rack and console wiring. One advantage of solid over stranded is that it does require less room—an important consideration inside a cabinet where space is at a premium.

Insulation—The primary purpose of insulation is to provide isolation between conductors. Insulating materials can be thermoplastic (for example vinyl, polypropylene, or polyethylene) or thermosetting (rubber, neoprene, and silicone rubber). For many audio applications, polypropylene has several advantages stemming from its superior chemical, electrical and physical properties. It requires less thickness to provide the same degree of protection as other materials. It also is tougher and more resistant to cut-through and crushing from rugged use..

Shielding—For many years, the primary defense against electro-static interference in studios was braided or spiral-wound wire strands wrapped around the insulation. Both shielding methods had drawbacks: They were costly, difficult to terminate, and heavy. About fifteen years ago, an alternative shield was developed for audio applications. Belden called this new aluminum foil/polyester laminate "Beldfoil." Beldfoil is much lighter than other shields, provides 100 per cent coverage, and is easy to terminate. using a drain wire running the full length of the conductor under the foil. FIGURE 2 and TABLE 1 compare different types of shielding and their effectiveness in repelling interference.

Jacketing—The outer element in cable construction is the jacket. Its function is to protect the inner elements of the cable. Like insulation, jacketing materials are either thermoplastic or thermosetting. Thermoplastic jacketing is lighter in weight, lower in cost, can be brightly colored, and generally is more water-resistant than thermosetting. Thermosetting or rubber jackets tend to produce cables that are limp, lie flat, and have better low-temperature characteristics—important for remote coverage in cold-weather climates.

MOST COMMONLY USED CABLES

Completing that very brief review of cable construction, we will proceed to discuss some of the cables commonly used in broadcast and recording studios. The most basic are the two-conductor shielded cables—such as seen in FIGURE 3. These designs, which have been around for fifteen years, are frequently the audio workhorses of the studio. One version has stranded conductors, which means it can take a great deal of flexing, and is used where components have to be moved frequently, such as in component racks. The conductors are twisted to reduce crosstalk, and are surrounded with a 100 per cent coverage shield. It has polypropylene insulation, which makes it smaller in diameter, and a vinyl jacket that is of light weight and low cost, with good electrical properties.

A variation of this cable, for use in fixed installations, is of exactly the same construction, except that it has solid conductors. Therefore, it is slightly smaller in diameter than its stranded counterpart.

In a third design within this family, the jacket and shield are bonded together, permitting the use of automatic stripping equipment.

BRAIDED SHIELD CABLE

Another common two-conductor shielded cable features a braided shield instead of foil. Braided shielding can take more abuse, making it ideal for microphone use in applications subject to considerable cross-traffic. It has stranded conductors, with vinyl insulation and jacketing. With an 86 per cent braid coverage, it is slightly more vulnerable to outside interference than a foil shield, and bulkiness makes it 0.075 in. larger in diameter than a comparable foil-shielded cable.

For use in fixed installations, a similar design with solid conductors may be preferred.

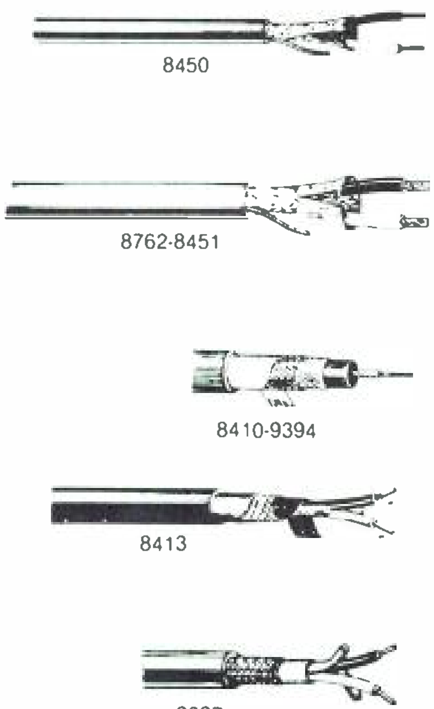
UNBALANCED LINES

In applications where unbalanced lines are acceptable, single-conductor constructions have been used by the broadcatst industry for some time. Belden's 8410 is a representative example, and is unusually durable, with seven elements comprising the cable: The conductor consists of three strands of copper, and four strands of copper-covered steel, which is surrounded with rubber insulation. Next comes a rayon braid, a tinner copper braid shield, cellulose yarn wrap, and finally, a black rubber jacket. The limpness of the construction enables it to lie flat so that it is less a hazard in the studio.

MINIATURE CABLES

Miniature mic cables, which are highly flexible and rugged, are being used by more and more studios. They normally feature a tinned cadmium bronze conductor, cotton serve, rubber insulation, 100 per cent coverage conductive textile wrap shield, tinned copper braid shield, and black rubber jacketing. Cables available with this construction include a two-conductor design with a nominal o.d. of 0.190 in., and a three-conductor construction, which has a nominal o.d. of 0.223 in.

Figure 3. Representative examples of two conductor-shielded cables.



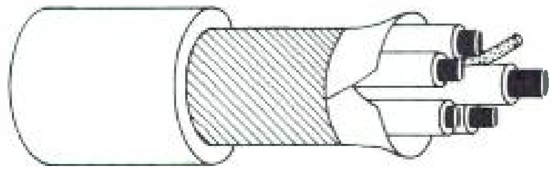


Figure 4. Extra shielding protection was designed into the special Star Quad cable developed for the Opryland Production Center.

COMPARING MIC CABLES

A comparison of several microphone cables may help point out that there is often more than one way to get the signal "from here to there," and that non-electrical specifications should not be overlooked either.

For example, the nominal capacitance, shield coverage, gauge, and stranding of Belden's 8413 and 9399 are the same. However, the insulation thickness of the 8413 is greater than that of 9399, while the outer diameter of the 9399 is somewhat larger, due to a greater jacket thickness. The 9399 is limper, yet tougher than the 8413, and its brown rubber jacket is often preferred for on-stage use, instead of the black-jacketed 8413. The tinned cadmium bronze conductors in the 8413 may be easier to solder, while the 9399's bare cadmium copper conductors are a contributing factor in the cable's limpness.

A third alternative is the vinyl-jacketed 9397, with the thinnest o.d. of the three. Although the conductor diameter is the same as the 8413 and 9399, it comprises 105 strands of 44-gauge wire, while the others have 45 strands of 40-gauge wire. The nominal capacitance of the 9397 is higher than the 8413 and 9399.

Which is the "right" microphone cable for your application? That depends on your particular requirements. The 9399 is built to take more abuse—in fact, it was originally custom-designed for a road tour through the Soviet Union by Tennessee Ernie Ford and Sandi Burnett. Billed as "The Nashville-to-Moscow Express," the artists (and the cable!) were to be subjected to a grueling 32-day marathon of public appearances, and total dependability was an absolute must.

On the other hand, for less-demanding applications, 8413 will serve just as well. Where a slightly higher cable capacitance presents no problem, the user may prefer the very-thin 9397.

MULTI-PAIR CABLES

In addition to these basic configurations, studios often use multi-pair cables. Utilizing the same basic designs, these cables may contain from 3 to 27 pairs of 22-gauge conductors. Each pair is twisted and individually Beldfoil-shielded, using the same basic construction as single-pair 8451. Polypropylene insulation gives the cable excellent audio and high frequency properties, and mechanical toughness. The outer jacketing is overall chrome vinyl.

As with most other cable designs, there are advantages and disadvantages to choosing a multi-pair cable over an equivalent number of single-pair lines. The multi-pair cable takes up less space, is easier to run through conduits, and is less of a physical obstacle for long runs across a stage or recording studio floor.

On the other hand, termination of a multi-pair cable is usually tedious, more unsightly and perhaps somewhat more fragile unless adequately protected. If such termination is enclosed with a good microphone box and therefore physically protected from abuse, the multi-pair cable is probably the best choice. But where space is not a premium, and Switchcraft or equivalent plugs are required at both ends, the engineer may prefer to work with single-pair cables instead.

When working with multi-pair cables, don't overlook the fact that each pair is color-coded—an important aid when trying to work with a single cable containing 27 pairs! Such a cable contains three layers of individually-shielded pairs. The outer layer contains 15 pairs: one red-shielded, one green-shielded, and 13 blue-shielded. Any blue-shielded pair within the outer layer may be easily located by noting its relative position in the cable, with respect to the green-shielded cable. Thus, blue-1 is right next to the green cable, and blue-13 is the last of the series, right next to the red one. The middle layer contains nine pairs (one red, one green and seven blue), and the inner layer contains three pairs (one each of red, green and blue).

A Cross-reference Guide to Selected Audio Cables

For the convenience of our readers, here is a cross-reference guide to some of the more popular audio cables in use today. It is IMPORTANT to note that all electrical specifications may not be identical when switching between cables from different manufacturers. Therefore, these catalog numbers should be understood to represent the nearest equivalents, and not necessarily the identical product.

Table 1. Properties of the basic types of shielding.

	SHIELD TYPE				
	BRAIDED SHIELDING	SERVED SHIELDING	FOIL SHIELDING	SEMICONDUCTIVE TEXTILE	SEMICONDUCTIVE PLASTIC
SHIELD EFFECTIVENESS AT AUDIO FREQUENCIES	Good	Good	Excellent	Fair	Good
SHIELD EFFECTIVENESS AT RADIO FREQUENCIES	Good*	Poor	Excellent	Poor	Poor
NORMAL PERCENT OF COVERAGE	40-95%	90-97%	100%	100%	100%
FATIGUE LIFE	Good	Fair	Good	Excellent	Good
TENSILE STRENGTH	Excellent	Fair	Good**	Good**	Poor
TERMINATION METHOD	Comb and Pigtail	Pigtail	Drain Wire	Drain Wire	Drain Wire

*Excellent when used with foil.
**Includes drain wire.

ALPHA	BELDEN	COLUMBIA	MANHATTAN	
2460	8450	02515	M 4325	all-purpose audio cable, solid conductors
2461	8451	02516	M 4326	all-purpose audio cable, stranded
		Multi-pair Cables		
6010	8777	06040	M 3103	3 pairs
6012	8778	06041	M 3106	6 pairs
6014	8774	06042	M 3109	9 pairs
6016	8775	06043	M 3111	11 pairs
6018	8776	06044	M 3115	15 pairs
n/a	9796	06060	n/a	17 pairs
6020	8769	06045	M 3119	19 pairs
6022	8773	06046	M 3127	27 pairs

Selected Audio Cables—partial specifications*

Catalog number	Number of conductors	Stranding	Insulation thickness (inches)	Jacket thickness (inches)	Nominal o.d. (inches)	Nominal capacitance between conductors (pf/ft.)	Notes
8406	3	45x40	0.019	0.025	0.223	35	A
8410	1	7x33	0.063	0.025	0.245	33	B
8413	2	45x40	0.019	0.025	0.190	30	C
8450	2	solid	0.006	0.018	0.118	40	D
8451	2	7x30	0.008	0.020	0.135	34	E
9397	2	105x44	0.012	0.031	0.176	40	F
9399	2	45x40	0.017	0.030	0.195	30	G
9451	2	7x30	0.008	0.020	0.135	34	H

NOTES

A 3-conductor equivalent of 8413.
B For unbalanced lines. Capacitance is between conductor and shield.
C Standard miniature microphone cable.
D Solid conductor equivalent of 8451, for fixed installations. Note slightly higher capacitance.
E All-purpose audio cable.
F Very thin microphone cable. Note slightly higher capacitance.
G Very durable microphone cable.
H Same as 8451, but jacket and shield are bonded for use with automatic stripping equipment.
*For complete specifications, consult manufacturer's catalog.

*Table 2. Selected Audio Cables—partial specifications.**

The conductors within each pair are also color-coded. Most manufacturers publish color charts. These are not yet standardized, so to avoid confusion, it's a good idea to select one such chart and stick to it.

SPECIAL PROBLEMS

These are the basic cables used in today's broadcast and recording studios. Sometimes, however, an application calls for a special-purpose cable not included in a manufacturer's standard product line. For example, Nashville's $15 million dollar Grand Ole Opry House had a severe problem with electromagnetic interference. Chief engineer David Hall recalls that a small radio station about half a mile away contributed to the problem by creating an rf field strength of 0.5 volts at the Opry House.

A two-part solution was required. First, the entire area was enclosed in a 2x2-inch welded wire mesh. Including the roof, floor and walls, this shield cost about 1 million dollars.

Part 2 of the solution was a custom-designed cable, which Belden calls "Star Quad." The cable consists of one 25-gauge conductor, surrounded by four 24-gauge conductors. The latter are individually shielded, and twisted to reduce cross-talk. Further protection is provided by an overall tinned copper-served shield with a drain wire. The star quad design has proven so effective that it is often used for microphone extension cords that must be placed in the vicinity of hum-producing fields. In fact, by the time the Opry House was finished, more than twenty miles of the specially-designed star quad cable had been used.

Although new developments in materials and cable construction take place continually, a resistance to change is not uncommon. In fact, some engineers tend to stay with the cable they've used for many years, simply because they know it does an adequate job. Designs that purportedly do the job better are viewed with suspicion. Some studios continue to use constructions that were introduced in the 30's and 40's. Unusual? Yes, but it offers a good example of the complexities involved in cable specification and selection. While manufacturers press forward with new product development, they also continue to provide the designs which over the years customers have come to know and trust.

But before ordering more of the cable you've been using all these years, why not check up on what's new? You may be pleasantly surprised. ■

For more information, contact the manufacturers directly. Most have regional sales offices.

Alpha Wire Corporation
711 Lidgerwood Avenue
Elizabeth, New Jersey 07207
(201) 925-8000

Belden Corporation
Box 1327
Richmond, Indiana 47374
(317) 966-6661

Columbia Electronic Cables
11 Cove Street
New Bedford, Massachusetts 02744
(617) 999-4451

Manhattan Electric Cable Corp.
1 Station Plaza
Rye, New York 10580
(800) 631-7742

FRANKLIN J. MILLER

Those Little Boxes

Using the correct interface device, whether commercial or homemade, will avoid these crises when what should be joined together—doesn't.

IN THE RESEARCH LABORATORY, equipment interface is rarely a problem. Although Murphy's Law states. "Matching devices, won't" there is usually sufficient time to prepare the proper cables, plugs and intermediate "black boxes" before turning on the equipment.

However, most of us don't do our daily chores under laboratory conditions, and rare is the engineer who has not at some time or other found himself trying to join two pieces of gear that seem to have been carefully designed to prevent any sort of interconnection whatever. In years past, the engineer was pretty much left on his own, and every studio had its distinctive collection of home-brew black boxes, ranging in quality and performance from great to gross.

Today, there are a number of commercially-available interface devices on the market, and now the same engineer may be faced wtih a different kind of problem—that of trying to decide which one is best-suited to his needs.

When buying any sort of interface device, the same principles apply as in acquiring the most expensive part of your audio system. You should know what you want out of the system, and how you're going to use it. A good method of doing this is by starting out with a block diagram of the desired end results. When doing this, keep in mind the functions you want within the system, as well as what possible dB levels will be at each point in the system. This can save you time and money when you're ready to buy.

With interface devices, higher specifications than needed are expensive, and unnecessary. For example, perhaps you need a matching transformer for your mixer-to-tape recorder hook-up. Recorded levels will average at +4 dB and peak at +18 dB. You want the best sound possible of course, but buying a matching transformer that will handle peaks at +30 dB will not improve your overall sound. On the other hand, don't sell yourself short either. If you know your levels will peak at +18 dB, don't get a matching transformer that quits at +4 dB.

An important point to remember is that whatever you plug into your system should be balanced at its input. An excellent safety measure would be to look for balanced input transformers.

BUILD IT YOURSELF

There are several interface devices that are not at all difficult to build by robbing the space parts bin. In fact, probably every engineer has at one time or another tried his hand at building a "direct box": either passive or active. (The direct box, not the engineer.)

PASSIVE DIRECT BOXES

The passive direct box is typically designed for electric guitars or other magnetic pickups. It will have a high input impedance (generally, about 60k ohms) and a low output impedance (150-160 ohms). It converts the guitar pickup output to mic line level and impedance, and allows the signal to flow on to the console. A ground-lift switch is often

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Franklin J. Miller is president of Sescom, Inc., of Gardena, CA.

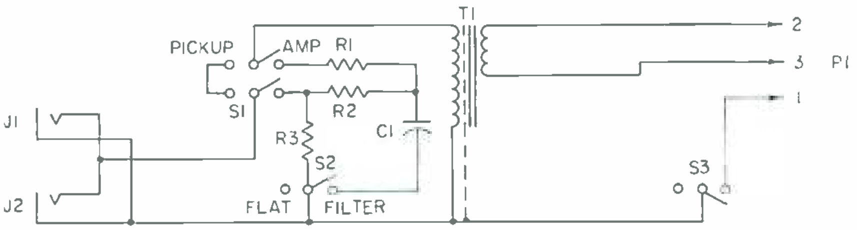


Figure 1. A passive direct box.

added at the output side to eliminate ground loops. A simple capacitor-filter may also be included.

The output impedance of the guitar pickup should be less than 60k ohms when using such a passive direct box. If it is greater, it will not be properly matched and there will probably be a deterioration of frequency response.

ACTIVE DIRECT BOXES

The active direct box is generally used for keyboards and synthesizers. As the name suggests, there is an active gain element in the units. The typical input impedance may be greater than 1M ohm, and even as high as 10M ohms. This is necessary since the output impedances of some electronic instruments may be greater than 100K ohms.

Most active direct boxes have little or no gain, because the high signal level from the instrument itself makes further amplification unnecessary.

CIRCUIT DIAGRAMS

Circuit diagrams and a brief description of some of the more popular interface boxes are given below.

FIGURE 1 shows a versatile passive direct box, which may be inserted between an electric guitar and its amp, or between the amp and the speaker. Note that the phone jacks are simply wired in parallel, to provide a direct connection regardless of where the box is inserted. The three resistors form a "T" pad for attenuation when the box is driven from the guitar amp. The output, P_1, is a three-conductor microphone plug, for connection to any studio mic line.

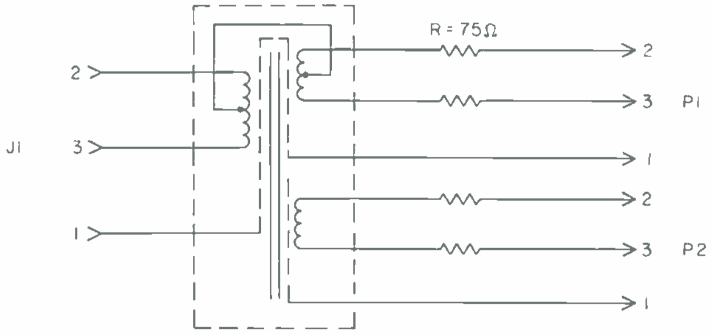
FIGURE 2 illustrates an active direct box with unity gain. It is designed for use between an electronic keyboard and its amplifier, and will drive a 600 ohm line into any low impedance mixer or tape recorder input.

MICROPHONE COMBINERS AND SPLITTERS

Combiners and splitters are transformers with multiple (two or more) inputs and a single output (or vice versa). In a typical application, a microphone splitter may be used to feed the output of a single microphone to several input devices. In sound reinforcement work, the microphone might feed the main console, plus an on-stage foldback mixer.

In FIGURE 3, note that two windings have their center

Figure 3. A microphone combiner or splitter.



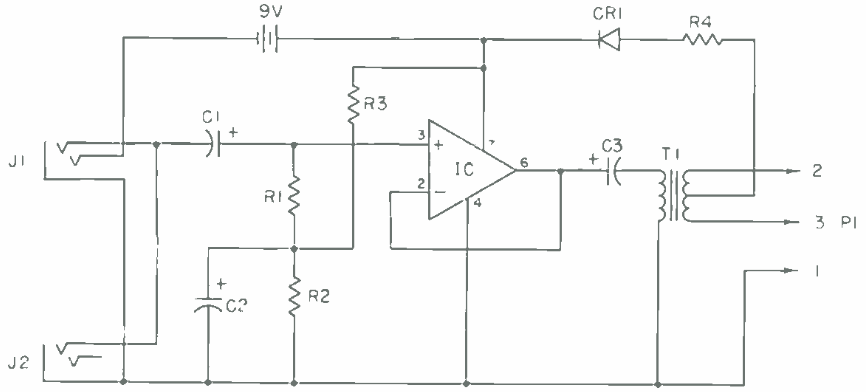


Figure 2. An active direct box.

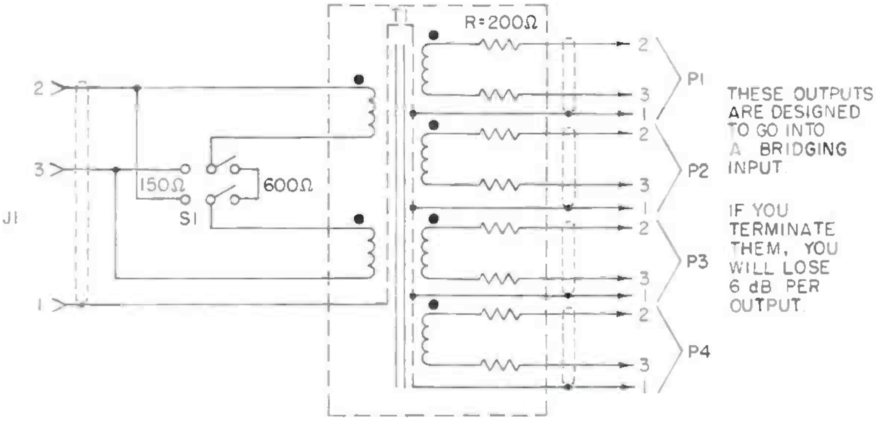


Figure 4. A line level splitter.

taps connected. This provides continuity for phantom-powered microphone systems. The console providing the phantom powering should be connected to P_1.

A microphone combiner reverses the procedure so that the user may combine several microphone outputs into one input. In this application, the resistors seen in FIGURE 3 are omitted.

LINE LEVEL SPLITTER

FIGURE 4 shows a line level splitter. The principle is the same as that of the microphone splitter, while the two-way switch allows the device to be fed from either a 150 ohm or a 600 ohm line. The four outputs may be used to maintain isolation and proper impedance feeds to four power amplifiers.

In summary, transformers are very useful devices, but are not to be considered as "cure-alls." In using audio transformers for whatever purpose, keep a few important points in mind for best results. Transformers are real devices with their own inherent problems, but when used properly they should give quite satisfactory results. Avoid abusing them; for example, a too-small transformer with improper (or not enough) core material will result in low frequency distortion. Never use a transformer that has less impedance than the source. If this happens, it will cause the transformer to under-terminate the source and cause a deterioration in frequency response.

If these rather basic facts are kept in mind when using transformers in systems interface work, you should encounter little or no problems wtih them. ■

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JOHN M. WORAM

Nuts-'n-Bolts

Here are some of the "forgotten" components without which no studio will function.

If you've been db-watching lately, you may have noticed that each month we've been taking a close look at what's happened in such interesting areas as microphones, broadcast audio, disc mastering and digital technology.

What you may not have noticed is what's happening with microphone stands, plugs, patchcords and equipment racks—the "nuts-'n-bolts" that tie all that other stuff together. After all, what can be said about such things anyway, other than that they exist? Let's face it: its much more interesting to talk about (and play with) digital tape recorders and the latest computerized console, and all that other fun stuff.

But, the nuts-'n-bolts really shouldn't be totally ignored either, for these things are important too, as you'll quickly discover if you overlook them when planning your new studio. In this little feature, we'll briefly cover a few of the odds and ends that didn't quite make it to the feature-story category.

MICROPHONE STANDS

When was the last time someone in the studio tripped over the tripod legs on the guitar mic stand, sending your brand new condenser microphone crashing to the floor? If it was less than ten minutes ago, why not consider going back to the old round-based floor stands, and then adding one of the new easy-to-use boom arms on top of it? For example, an Atlas MS-4 or MS-20 floor stand, with a small boom arm from AKG, Beyer or Keith Monks, gives you the best of both worlds: a tip-proof base (well, almost), and an easily-adjustable boom.

When buying microphone stands and boom arms, make sure the threads agree. Most microphones sold in this country come with a stand adapter with a ⅝-inch 27 TPI thread. Many European microphone stands are threaded to some other standard, and a thread adapter is therefore required. Keith Monks manufactures some eighteen different styles of adapters, so they should have whatever you need. It may not be a bad idea to glue the thread adapter to the microphone stand. Otherwise, they have an annoying habit of lodging themselves securely in the microphone's own stand adapter. Later on, it gets to be crazy time, as you try to guess which microphone is concealing which thread adapter.

Where the tripod leg feature presents no problems, Atlas's new PS-S floor stand should be checked out. If you've ever watched an on-stage musician grapple with a stubborn microphone stand, you'll appreciate its new "touch control" clutch, which allows the height to be adjusted without resorting to a pipe wrench.

Remember the famous "Starbird" boom? It was designed by George A. Starbird in 1936, and has been popular ever since. Accurate Sound Corp. has recently acquired the rights to the Starbird Deluxe Studio Boom, which may be set at any height up to 16 feet.

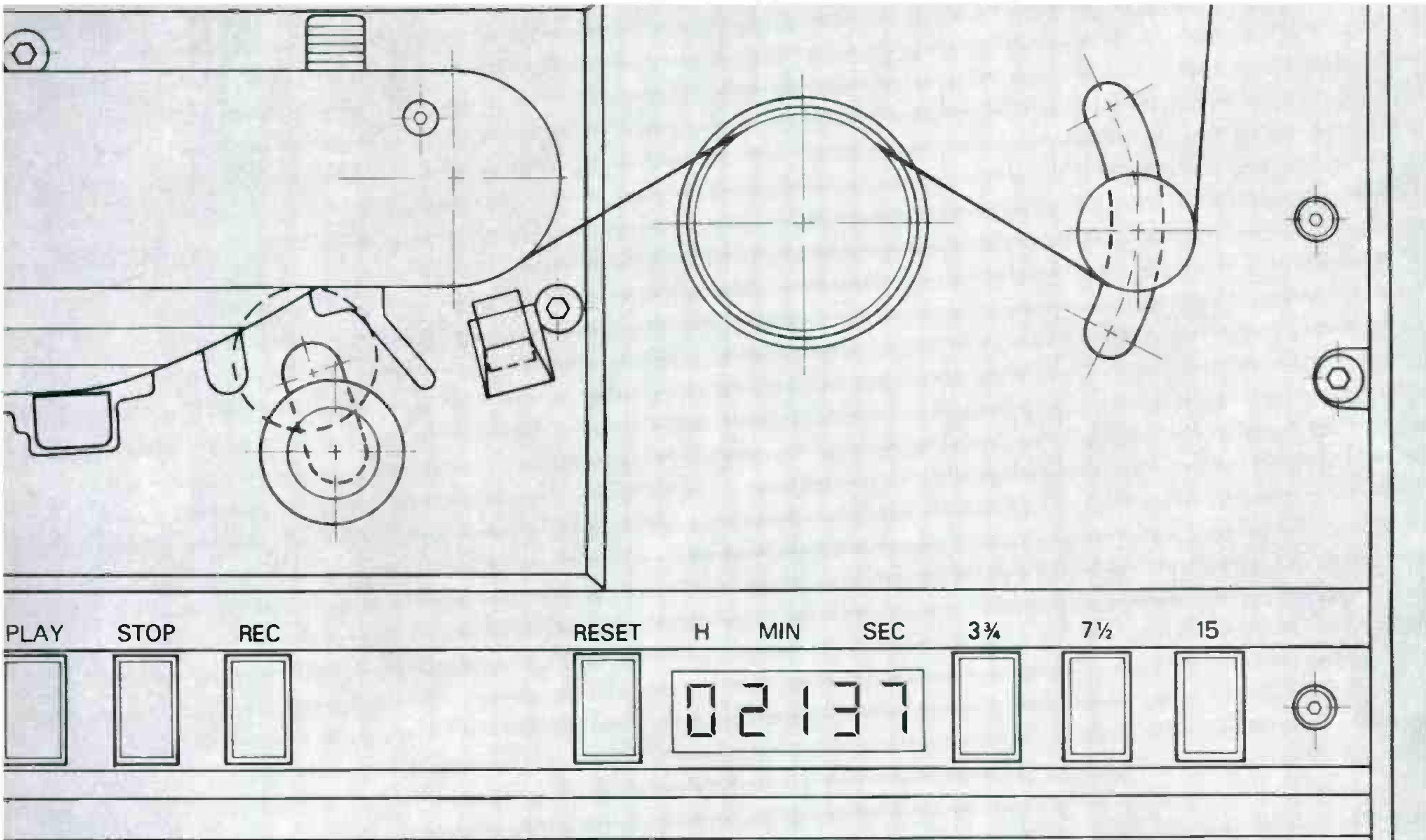
The American Photographic Instruments Co. manufactures a series of very-lightweight tripods, popularly known as "pic stands." One of these—the model 532—extends to 15 feet. It is definitely not recommended for heavy microphones, but it is ideal for a small condenser microphone, especially on remote recording sessions, where heavy traffic is not a problem. The 532 weighs less than five pounds, and its upper diameter is ½-inch, unthreaded.

MICROPHONE PLUGS

If you're like most of us, you probably think your microphone cables have "XLR" or "Cannon" plugs on both ends. Maybe they do, but on the other hand, maybe they don't, for "XLR" is merely a catalog number prefix for plugs manufactured by the (ITT) Cannon Electric Co. Equivalent plugs are made by Switchcraft, Neutrik, ADC Products and others. Once you've got one of them in hand, you can call it anything you like, but if you try to order one from any of the latter companies, you'll probably make a lot more friends if you refer to the proper catalog number. Therefore, the following brief cross-reference may be of some use.

Cannon	*Switchcraft*	*Neutrik*	*ADC*
XLR-3-11C	A3F	NC-3FC	PF3
3-conductor cable plug, female.			
XLR-3-12C	A3M	NC-3MC	PM3
3-conductor cable plug, male.			
XLR-3-31	D3F	NC-3FP	RF3
3-conductor panel receptacle, female.			
XLR-3-32	D3M	NC-3MP	RM3
3-conductor panel receptacle, male.			

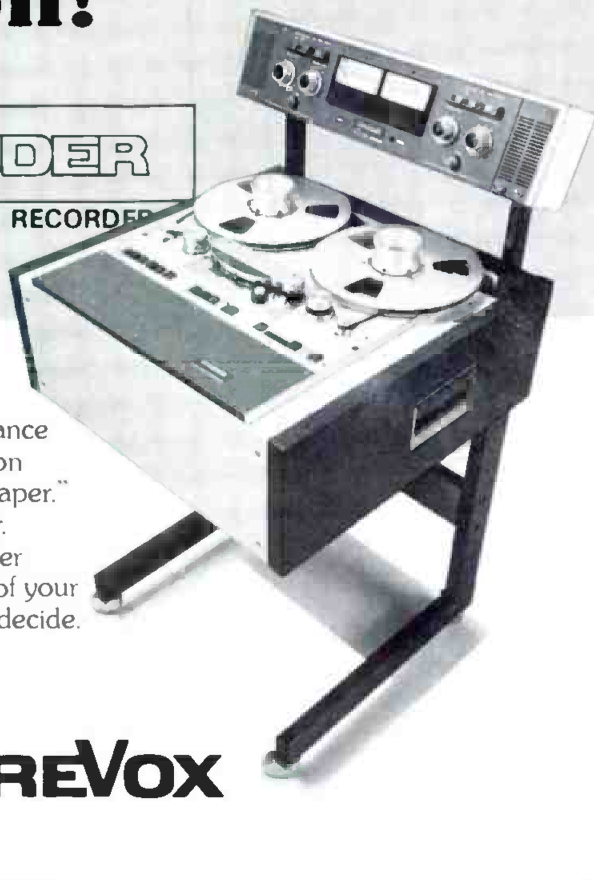
Neutrik audio connectors, available through Philips Audio Video Systems Corp.



Studer B67

It's got competitors . . . but no competition!

STUDER
B 67 TAPE RECORDER



When you buy the Studer B67 tape recorder/reproducer, you get more than just one of the world's finest tape recorders.

You are buying an engineering philosophy where performance is first and there isn't any second or third. You are buying a dedication to quality seldom seen in today's world of "make 'em faster and cheaper." At Studer, one person in every seven is a quality assurance inspector.

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To learn more, contact:

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Circle 41 on Reader Service Card

As you may guess, the "3" in the catalog number refers to the number of conductors. For other applications, up to six-conductor plugs are available in the same general format.

In the Switchcraft series, an extra solder terminal allows the cable shield to be connected to the metal plug itself. Thus, four lines (3 conductors plus shield) may be run, if desired.

The cable plugs provide various forms of cable strain relief. The Neutrik series uses a self-adjusting collet, which will grip cables with diameters between 0.157 and 0.275 inches. Switchcraft and ADC plugs have two captive screws at the rear, which bear down on the cable. In the Cannon series, two screws remove a half-round cable clamp that has a slight lip on its inner diameter. With the lip mounted away from the plug, it will clamp small cables, to 7/32-inch. Remove the clamp and reverse it (lip towards the plug) and it will grip cables up to 10/32 (5/16)-inch. (All right, how many Cannon plug users *really* knew that?)

The famous Starbird boom is now available from Accurate Sound.

BLACK BOXES ON THE STUDIO WALL

Once you've chosen your microphone stands, cables and plugs, you'll need a "black box" on the studio wall—into which, all of the above gets routed. Wireworks Corp. makes a wide variety of such "multi-boxes," with anywhere from three to twenty-seven Switchcraft D3F plugs on the front panel. The multi-boxes are designed to interface with any of the standard multi-pair cable configurations. Wireworks' own cables are made by Alpha Wire Corp., and by the way, those are Wireworks/Alpha cables seen on this month's cover photo by Robert Wolsch. (And, for more about cables, see Paul Miller's audio cable story in this issue—Ed.)

If you don't need a black box on the wall, perhaps you'd like a "blue box" at the end of your microphone snake. Sescom manufactures a series of "mic-splitters" (see Franklin Miller's story for details), and some of these use multi-pin plugs for connection to the multi-pair microphone cable. Sescom solves the problem of terminating the multi-pair cable by supplying separate termination cables, consisting of 5-ft. lengths of the appropriate number of Belden 8413 cables. As before, a mil-spec. plug connects the termination cable set to the length of multi-pair cable running out to the mic-splitter.

And, for a microphone-splitter designed for rack-mount applications, Uni-Sync's MS-1003 has ten inputs, each of which has one direct and two transformer 'bridged outputs. Each transformer output has its own ground-lift switch. The inputs and direct outputs are on the rear of the unit, while the transformer outputs are on the front panel.

CABLE TESTERS

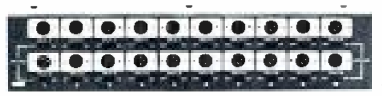
If you've done all the wiring yourself, you'll appreciate the value of a good cable tester, into which both ends of the cable are plugged. A series of LED's warns you of opens, shorts, and phase reversals. Such cable testers are made by Xedit, Wireworks, Sescom and others, with perhaps the ultimate tester coming from Switchcraft. Their model QC 1001 Cable Tester permits checking cables with up to 180 different termination (plug) combinations. If that's not enough to keep you out of trouble (or maybe get you into it), the QC 1002 adds seven adapters, which allow 350 combinations to be checked.

PATCH CORDS

If you'd rather let someone else do at least some of the wiring for you, Pomona Electronics manufactures a seemingly-endless variety of patch cords, with every conceiv-

The ultimate cable tester: Switchcraft's QC 1001.

The Uni-Sync MS-1003 microphone splitter.



H. Wilson Corp's "Tuffy Tables," for moving test gear or coffee and doughnuts.

This may be an improbable array of equipment, but it does give you an idea of the versatility of Click Systems' roll-around trolleys.

able type of plug available. Their assortment of adapters, probes cables and such will seem outrageously expensive, unless you've tried to test some equipment with a collection of home-brew cables. Many are the set-ups that have been demolished by slightly-flaky inter-connections, and a collection of Pomona's almost-indestructible cables will wind up saving you hours of grief in the long run.

FOR EQUIPMENT ON THE MOVE

If you've been in the studio business for awhile, chances are you've accumulated your share of assorted gadgets that are needed now and then, but not always. Click Audio File Systems has some light-weight roll-around equipment trolleys that may come in handy. Some are little more than an open frame on wheels, but this may be just what's needed for easy access to both the front *and* the back of the gear. For non-rack mount equipment, shelves are easily installed.

And for other roll-around chores (coffee and doughnuts, test gear, client's tapes, or whatever), the H. Wilson Corp. makes a line of "Tuffy Tables," constructed from rugged engineering plastic.

Finally, for keeping track of all your valuables, the Seton Name Plate Corp. will prepare a set of numbered custom name plates. These are anodized, dyed and etched permanently onto aluminum foil, with a 3M adhesive backing. They go on easily, but are difficult to remove, so if your equipment "travels" a lot (road crews, take note), the name plates may lessen the chance of something going astray.

For more information on any of the "nuts-'n-bolts" listed here, contact the following manufacturers. Tell 'em **db** sent you. ■

ADC Products
4900 West 78th Street
Minneapolis, Minnesota 55435
(612) 835-6800

(note: ADC stands for Audio Development Co.—not to be confused with ADC (Audio Dynamics Corp.) or Audio Developments. Audio Dynamics makes the Accutrac 4000 turntable, and England's Audio Developments manufactures compressors and limiters.)

Accurate Sound Corp.
114 Fifth Avenue
Redwood City, California 94063
(415) 365-2843

AKG Acoustics
91 McKee Drive
Mahwah, New Jersey 07430
(201) 529-3800

American Photographic Instrument Co.
10 East Clarke Place
Bronx, New York 10452
(212) 992-6000

Atlas Sound
10 Pomeroy Road
Parsippany, New Jersey 07054
(201) 887-7800

Beyer Dynamics Co.
Revox Corp.
155 Michael Drive
Syosset, New York 11791
(516) 364-1900

Cannon Electric Co.
666 East Dyer Road
Santa Ana, California 92702
(714) 557-4700

Click Systems
155 Michael Drive
Syosset, New York 11791
(516) 364-1900

Jensen Tools & Alloys
1230 South Priest Drive
Tempe, Arizona 85281
(602) 968-6231

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US distributor—Audiotechniques, Inc.
652 Glenbrook Road
Stamford, Connecticut 06906
(203) 359-2312

Neutrik Products
Philips Audio Video Systems Corp.
91 McKee Drive
Mahwah, New Jersey 07430
(201) 529-3800

Pomona Electronics
1500 East Ninth Street
Pomona, California 91766
(714) 623-6751

Sescom, Inc.
P.O. Box 4155
Inglewood, California 90309
(213) 678-4841

Seton Name Plate Corp.
592 Boulevard
New Haven, Connecticut 06505
(203) 772-2520

Starbird Microphone Booms
(see Accurate Sound listing)

Switchcraft, Inc.
5555 North Elston Avenue
Chicago, Illinois 60630
(312) 792-2700

Uni-Sync, Inc.
742 Hampshire Road
Westlake Village, California 91361
(805) 497-0766

H. Wilson Corp.
555 West Taft Drive
South Holland, Illinois 60473
(312) 339-5111

Wireworks Corp.
Box 3600
Hillside, New Jersey 07205
(201) 352-7800

XEDIT Corp.
182-25 Tudor Road
Jamaica Estates, New York 11432
(212) 380-1592

db Test Report

The Technics SL-1000MkII Turntable System

It isn't too often that a new turntable comes along that is so perfect for the broadcaster that it immediately becomes the one to have. Such a model almost was the original SP-10 Technics, but this was less than perfect particularly for fast flying starts. This problem has now been addressed by the SP-10MkII model. What this report is on is that new model installed in a massive base of obsidian material, with a Technics EPA-100 arm installed. In short, a ready-to-go system.

The turntable is three speed 33, 45, and 78. There is no vernier speed adjustment. Control of the platter is achieved by there touch buttons for the speeds, each has its own l.e.d. to indicate the selection made. A large start/stop press switch is the other control. This control is duplicated on a remote control unit which permits the actual start/stop to be operated from the console.

The power supply for the turntable is separate and can be remotely located. A connecting cable is supplied that attaches to the rear of the turntable base, as well to a.c. The power supply has a master power on/off switch.

The base comes with two insert panels for tonearms. One is uncut and can be used with any arm. The other is for the EPA-100 and has been cut to accommodate it. Mounting of the panels is via isolating rubber gaskets.

The massive base itself weighs about 30 lbs. It comes with a plastic dust cover which can be hinged back. Its legs are individually adjustable and contain damping plugs. No doubt it is the prime contributor to the solid feel the entire unit has.

The tonearm features adjustment from 0-3 grams with anti-skating compensation built in. The rear counterbalance weight has an adjustable damping plug that is to be set based on the approximate compliance of the cartridge. It does not seem to be over critical. Our tests of the arm were limited to a check for resonance. It was well damped and seemed to be centered at about 8-9 Hz. so should present no problems with warped records. The arm tracked well at low forces.

The turntable was tested for flutter, speed accuracy, rumble, and start/stop ability (its usefullness for quick cues). Flutter, using the DIN 44507 method, was 0.035 per cent. Rumble, using a CCIR weighting curve was better than −60 dB. It is difficult to measure accurately below that figure, but I believe that it is lower than the quoted number.

Speed checks of the three fixed speed proved them to be exact even under the normal load of a disc being played.

And finally: the ability to fine cueing. The platter is equipped with a braking system that stops it almost instantly when the *off* switch is pressed. Thus, the start of a selection can be found by playing the disc until the start point, hitting the stop, and then back cuing. How far to back cue? For the 33 speed, two inches was all that was needed for the platter to come to full speed with no detectable overshoot. At 45, back cue should be about three inches. Under both conditions of speed, these are virtually punch-in cues. I experimented considerably with such starts, and found them consistent and foolproof.

No question, this is a turntable for the broadcaster. The list price of the complete package described above is $1400.00. L.Z. ■

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UREI Sonipulse with AKG calibrated microphone, used. $900.00. **(213) 761-9336.**

(continued)

FOR SALE

PROFESSIONAL AUDIO COMPONENTS: AKG mics; Badap 1; Crown; dbx; Delta Lab; Eventide; Frazier; Gauss; GLI; Ivie; Malatchi; MasterRoom; Nagra; Neumann mics; Orban; Otari; Pentagon; RTR; Sennheiser mics; Switchcraft; Tascam; Uni-Sync; and UREI. These products are on demo in our showroom and in stock for immediate delivery. Our shipping is insured and prepaid. **Barclay Recording & Electronics, 233 E. Lancaster Ave., Wynnewood, Pa. 19096. (215) 667-3048 or 649-2965.**

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REELS AND BOXES 5" and 7" large and small hubs; heavy duty white boxes. **W-M Sales, 1118 Dula Circle, Duncanville, Texas 75116. (214) 296-2773.**

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Birth of the two-track masterpiece. Otari MX5050-B

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Model 1

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