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THE SOUND ENGINEERING MAGAZINE

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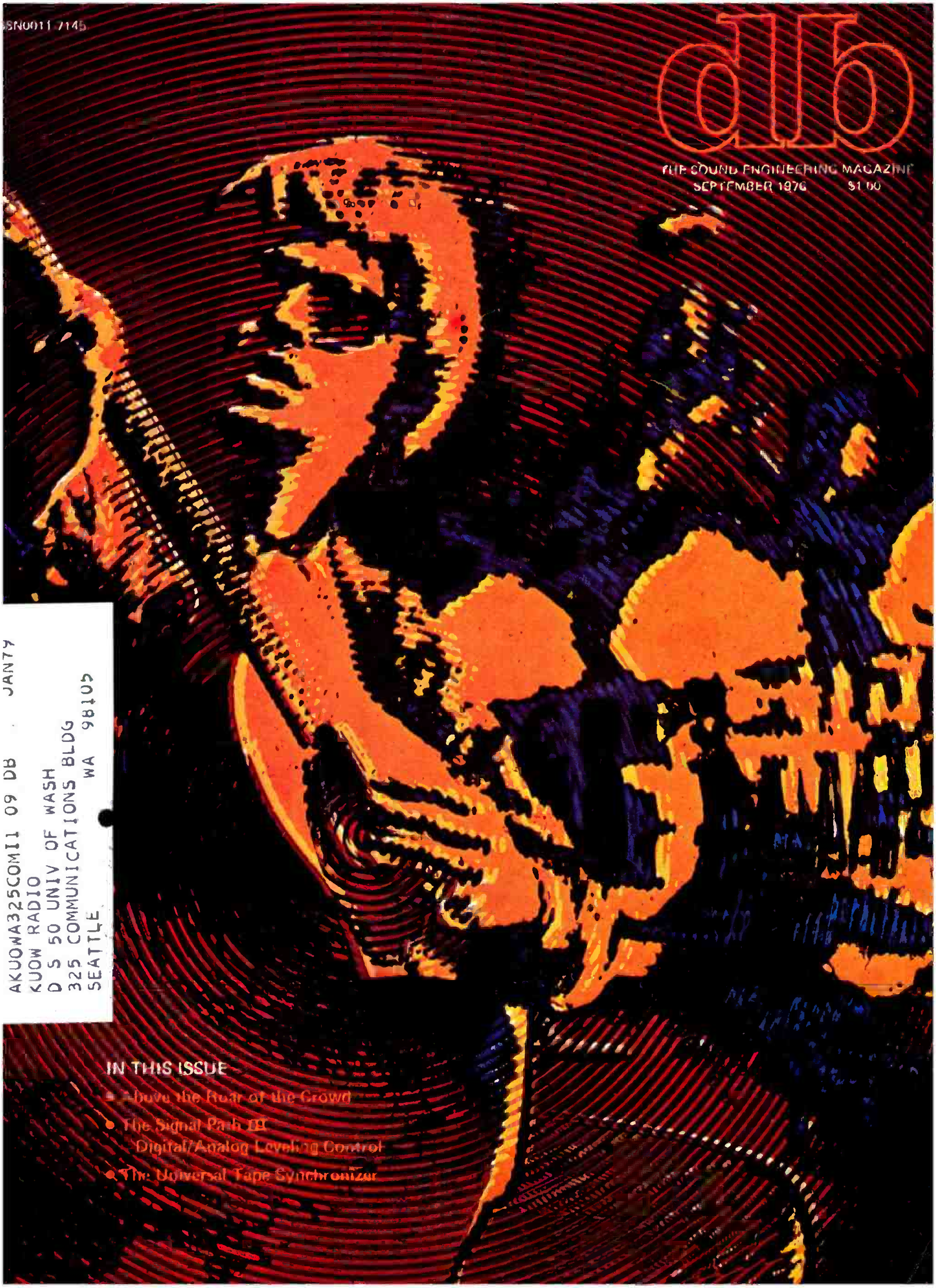

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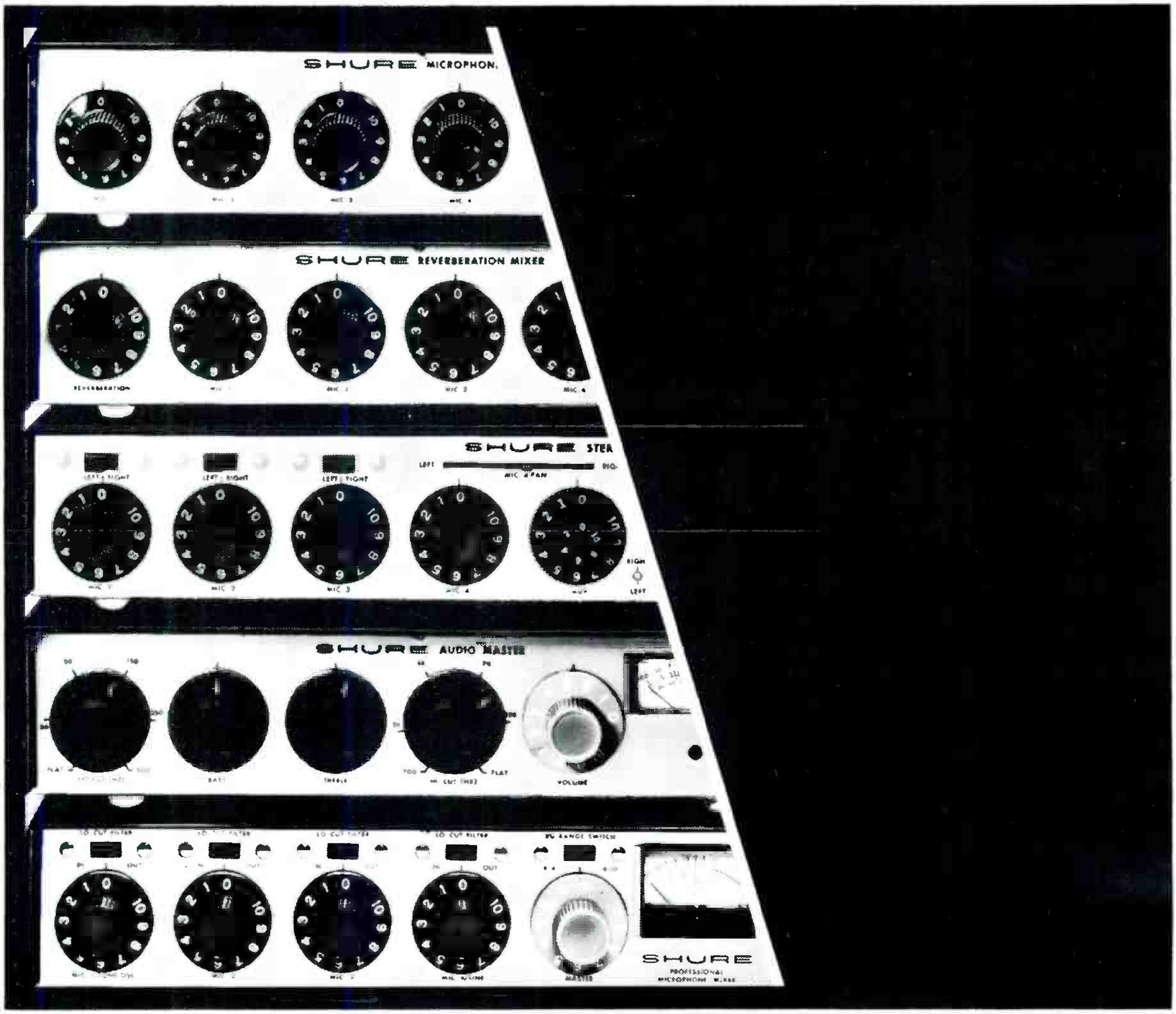
SHURE
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coming next month

Control Room paraphernalia concerns us in the October issue.

• Arthur Johnson and Robert Hess untangle misconceptions and supply clarification of the mysteries of equalization in THE WHY AND HOW OF EQUALIZATION.

• The ubiquitous vu meter takes on a new slant in Ronald Ajemian's report on THE NEW BREED OF VU METERS.

• There's a full test report on THE U.R.E.I. MODEL 200 RESPONSE PLOTTER. In less than a minute, this magic trick can plot the frequency response of a component or system from 20 to 20,000 Hz.

• A circular screen plus darkroom-manipulated color evolves the sound of contemporary music.

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

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FREE LITERATURE

IT'S NOT A GOOD IDEA TO BE LOADED BEFORE BEING TESTED

This intriguing title introduces a brochure which briefly describes the Metro-Check minicomputer, a system that checks printed circuit boards for opens and shorts. Mfr: Metropolitan Circuits. *Circle No. 92 on R.S. Card.*

CAPACITORS

An 86-page catalog provides cross references, specifications, and configurations for twist prong, electrolytic film dielectric, a.c., MICA dielectric, ceramic dielectric and d.c. Kraft, as well as information on relays, t.v./f.m. antenna rotor systems and citizen band noise filters. Mfr: Cornell-Dubilier Electronics.

Circle No. 93 on R.S. Card.

PHASE CONTROL

A new catalog describes a series of 80 amp rms silicon controlled rectifiers for general purpose phase control applications; includes nine sets of performance curves. Mfr: International Rectifier

Circle No. 95 on R.S. Card.

CONDENSER MICROPHONE CARTRIDGES

16 pages describe the various types of testing microphone cartridges, accompanied by cutaway diagrams, a survey of adaptors and accessories and a specification chart. Mfr: B & K Instruments.

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URETHANE FOAMS

Samples of various density acoustical foams are attached to this brochure, along with suggested application and description notes. Mfr: Tenneco-Chemicals

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VOICE COIL COOLERS

Voice Coil Application Bulletin, which describes the composition and operation of speakers with particular attention to voice coils (including a nice diagram of a speaker) not only leads to the product but could be used for educational purposes. A second leaflet, Adhesive Coated Aramid Paper for Low Power Speaker Voice Coils describes an adhesive coated paper used to fabricate split bobbin speaker voice coils. Mfr: Keene Corp.

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COAXIAL CONNECTORS

Plugs, jacks, and receptacles, particularly featurin gminiature and ultraminiature coaxial connectors, are covered in this catalog. Mfr: Malco.

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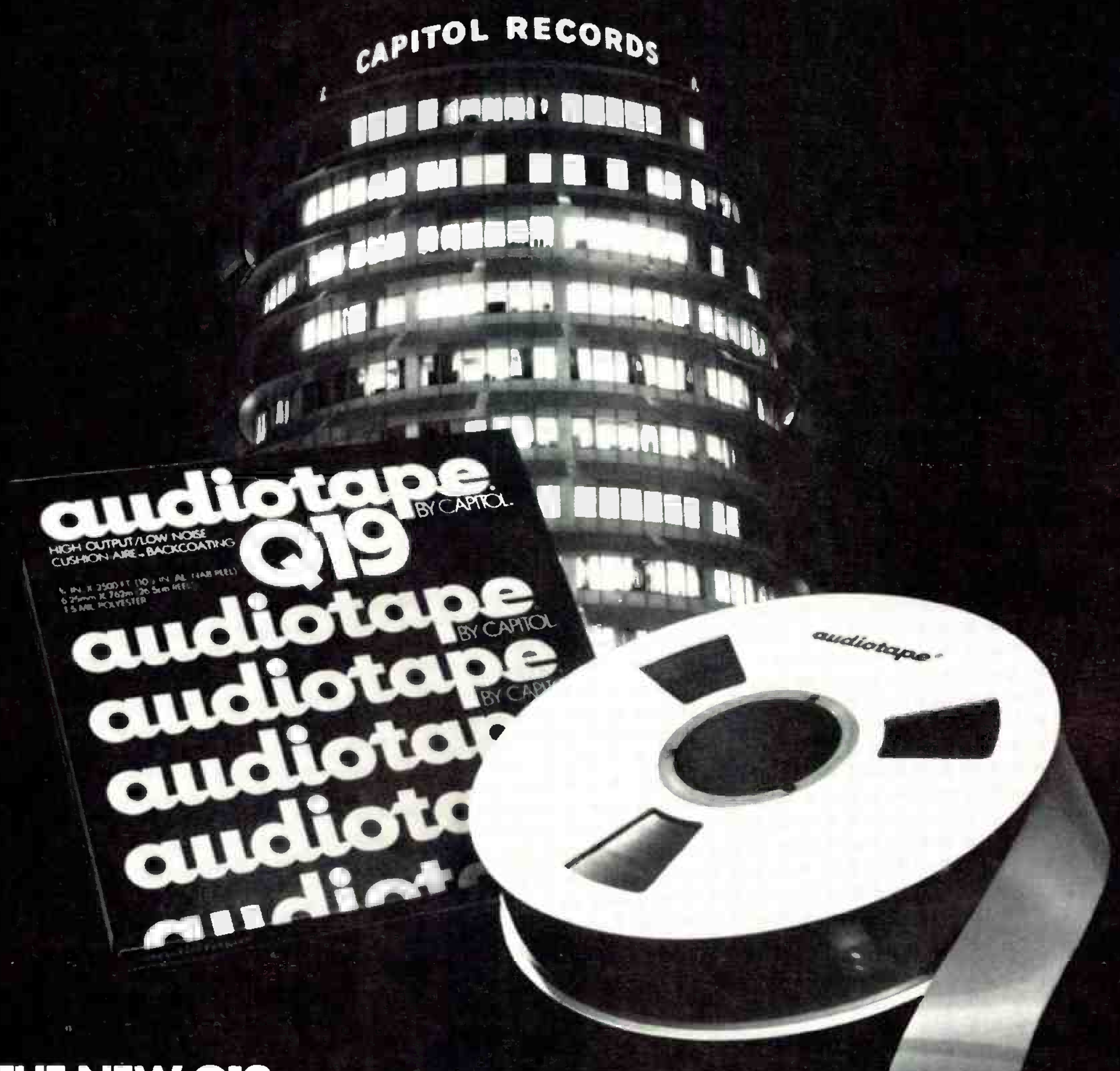

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CALENDAR

13-19 **International Audio Festival & Fair.** London, England. Contact: British Information Services, 845 Third Ave., New York, N.Y. 10022. (212) 752-8400.

20-24 **International Broadcasting Conference.** London, England. Contact: British Information Services.

14-16 **Synergetic Audio Concepts Professional Audio Seminar.** St. Louis, Mo. Contact: Don Davis, Synergetic Audio Concepts, P.O. Box 1134, Tustin, Ca. 92680. (714) 838-2288.

28-30 **Synergetic training seminar.** New York City. (See above.)

13-16 **B&K Seminar. Human Acoustics.** Contact: B&K Instruments, Inc. 5111 W. 164th St., Cleveland, Ohio, 44142. (216) 267-4800.

6-8 **Synergetic Training Seminar.** Boston, Mass. Contact: Don Davis, Synergetic Audio Concepts, P.O. Box 1134, Tustin. Ca. 92680. (714) 838-2288.

19-21 **Synergetic Training Seminar.** Washington, D.C. Contact: (See above.)

OCTOBER

1 **Society of Broadcast Engineers,** Chapter 22, Regional Convention and equipment show—open to all. Syracuse Hilton Inn, Syracuse, N.Y. Contact: Paul Barron, WCNY-TV/FM.

10-15 **Audio Visual Institute,** Indiana U. Contact: Dr. E. L. Richardson, Audio-Visual Center, Indiana Usiversity, Bloomington, Indiana 47401. (812) 337-2853.

11-14 **B&K Seminar; Designing Quiet Products.** Contact: B&K Instruments, Inc., 5111 W. 164th St., Cleveland, Ohio. 44142. (215) 267-4800.

26-27 **B&K Seminar: Microphones & Accelerometers: Their Calibration and Use.** (See above.)

29-Nov. 1 **Audio Engineering Society Show.** New York City. Waldorf-Astoria. Contact: AES, Room 929. 60 E. 42nd St., New York, N.Y. 10017. (212) 661-8528.

26-29 **Microforum '76.** London. England. Contact: British Information Services. 845 Third Ave., New York. N.Y. 10022. (212) 752-8400.

NOVEMBER

2-6 **Dixie Electronics Reps. Conference,** Boca Raton, Fla. Contact: Dixie Electronics Reps., 1720 Peachtree Rd., Suite 322, Atlanta, Ga. 30309. (404) 872-5981.

7-8 **Convention, Society of Broadcast Engineers.** Holiday Inn. Hempstead. N.Y. Contact: Mark Schubin. Society of Broadcast Engineers. P.O. Box 607. Radio City Station, New York, N.Y. 10019. (212) 765-5100, ext. 317.

8-11 **B&K Seminar: Acoustical Materials & Structures: Design, Testing, and Applications.** Contact: B&K Instruments. 5111 W. 164th St., Cleveland, Ohio 44142. (216) 267-4800.

8-12 **National Automated Production Exhibition.** Manchester. England. Contact: British Information Services. 845 Third Ave., New York, N.Y. 10022. (212) 752-8400.

9-11 **Synergetic Training Seminar, Nashville, Tenn.** Contact: Don Davis, Synergetic Audio Concepts, P.O. Box 1134, Tustin. Ca. 92680. (714) 838-2288.

17-19 **Synergetic Training Seminar.** Orlando. Fla. Contact: (See above.)

db sound with images

MARTIN DICKSTEIN

• Last month, this corner stressed the preliminary survey of the room in which a meeting was to take place, and for which you were asked to set up the audio visual equipment. Now that all the notes have been made and studied and discussed with the client and the agenda has been given to you with some information on the individual presentations, the time has arrived to get the a/v devices together and set them up.

HAVE A LITTLE LIST

The first, and one of the most important, steps you should take to avoid trouble, is to make a *complete* list of *all* the equipment you will need. According to the information you got in this theoretical situation, each of the speakers will need some audio-visual aid, and, in some cases, two or three different types. On the agenda, or in a separate listing, detail each presenter's requirements, including the items, accessories, cables, and even location of the devices in the room The list will enable you to determine cable lengths, interconnections, connectors, adapters, and so on. It will also be of assistance when the setup is made, to save time and prevent shifting of equipment after delivery to the meeting room and will also help you to plan the number and heights of tables as well as locations for all the materials you will be using.

You can be sure that if anything is going to happen during the meeting, it probably will, so the best you can do is to try to be prepared—for anything. This means a *complete* list, and spares, and the good a/v person's survival kit which is brought "just in case. . . ." Leaving anything off the list is not like missing an item or two on the old shopping list where you can go back and get it later or just get along without it if you forget anything. In the a/v business, what you forgot is exactly what you'll need, and remembering that it is lying on the bench back at the shop—no good!

The list should be made far enough in advance to give you time to think about it instead of scribbling it down in a hurry. The same goes for the equipment. Be sure it is all ready to go in plently of time to look it over so nothing is forgotten in packing, and pack it according to the list. Incidentially, your timetable also includes getting the gear to the location in plenty of time to allow for a professional setup, and a thorough checkout on the site.

CHECKING EQUIPMENT

Take a quick look at each of the pieces of equipment that will go on the job. Each piece should be checked out at this time to be sure it works well, that all its accessories are included on the list, proper plugs and adapters are available and ready to go. All associated equipment should be tested while fully connected in a test setup in the shop. This may seem like over-caution, but it is the best way for bad connectors to be found, broken connections determined, missing grounds located, and distortion from one cause or another determined in time to check it out. This also goes for improperly functioning projectors, dissolve units, or whatever else, other than sound equipment, you may need.

FILM PROJECTOR

Probably the workhorse of presentations is the film projector. Although most people are aware of how it works, and can thread a manual or load an automatic unit correctly, few except professionals know of possible problems and what to quickly do about them to keep the down-time to a minimum. The projector should be loaded with a test film, and run for awhile. It should also be stopped and started a few times.

The auxiliary speaker, if one is to be used, should be connected and the sound checked. The volume control should sound clean, not raspy or noisy at changes of level. Lamps must be checked and replaced before packing the equipment if there is the slightest doubt. They should be clean, not dark or smoky. The exciter lamp, although it is difficult to judge for age, should at least be clean and not dark looking. If you have any idea of the lifetime of the lamps, and you think they have reached nearly the limit, change them before getting out on the show. It's easier this way than when the program is under way and the lamp is very hot.

The lens should be clean and tightly in place for safety during shipping. The film path should be cleaned, cleared of old fragments of film or sticky tape which could ruin the operation, and the gate and aperture should be clean. (Incidentally, be sure the lamp is of the proper power for the throw distance and image size.) This is also the right time to check spare lamps. They should be put into the projector and turned on for testing, not just looked at. The first time you run into changing a lamp on a job and find the spare is defective you'll wish you had checked the spare.

When checking the projector completely, including the sound system, all accessories and spares and cables with connectors can be tested at the same time, and adapters checked out. Once you have satisfied yourself completely that the unit and all accessory items are ready to go, then you can pack them. This is also the right time to figure out lengths of cable for all extensions, such as to the speaker, or to the a.c. outlet, and here the original sketch and notes of the room will help.

By the way, be sure to carry take-up reels for the projector. They should be of the right size for the films you will run, and right quantity to allow you to change reels without having to rewind each film after it's over. Rewinding films during the meeting while your projector is in the same room as the audience can really be a problem. You can figure on using a supply reel for take-up on occasion, but if it's not the right size, don't use it for any film where the film will come over the edges.

SLIDE PROJECTOR

The second most used audio-visual aid (although there is seldom any audio associated with it) is the slide projector. A complete running test should be made with lamp, extension cords and remote control device, a drum or slides, and forward, reverse, and focus controls. Here, again, is a good time to work out cable lengths for the a.c., the remote control, and for support tables for the slide/film projector combination. Spare lamps should be checked in the projector, an extra remote control unit checked out and packed, and blank slides included as a professional service to the speakers who forget to bring their own. (An extra drum or two might also be of help in the event it is needed in a hurry.) It might also be wise to put a blank slide in the slide projector before you pack up. This will then be in the aperture when you put the first drum on and will allow you to change drums during the presentation without having to turn off the lamp. (This is especially helpful when the light source is Xenon, where the lamp cannot be turned out and turned on again immediately.)

Along with the slide unit, you might also check out the dissolve unit and the second projector, or the slide sync cassette player. With the cassette, it is essential that you try the unit with the projector you intend to use at the meeting. Use a tape and a drum to be sure everything works. This will also permit checking the sound with the auxiliary speaker unit you used for the film projector and all connector cables.

The sound quality should be tested with the projector and cassette player plugged into the adapters and cables just as they will be at the meeting. This way, the entire system can be judged for quality, loading, reversed wiring, grounds, shorts, etc. Also, if it is necessary during the show to have the dissolve unit and a remote control unit plugged in simultaneously for any part of the meeting, this is the best time to plug in the proper "Y" cable adapter, or to try to pull out one plug and put in the other without disturbing the setup or the alignment of the projectors. Before the meeting is the time to practice, not at the presentation. This is also an excellent time to rehearse what you will do in the event any thing happens. If the cassette goes out of sync, or if a lamp blows, or if someone steps on a cable and disconnects it, or if the sound goes bad, what would be the best way to solve the problem quickly with the least amount of fuss, and what will you need in the way of parts or tools to do it?

AUDIO TAPE PLAYER

The audio tape player, of course, is also an essential part of most presentations. This should also be checked, cleaned, and connected into the speaker system along with the other sound equipment to check connectors, loading, quality, etc., and should be played at its different speeds to be sure all is well. Again, use this chance to check the lengths of a.c. cables, extensions, and speaker cables so they can all be run hidden or covered, and out of the way of the people in the room, as you have planned according to the original sketch of the room during the survey.

The overhead projector can be a pain. This unit has to be at the front of the room near the person speaking. This means it can be set up on a small table at the front of the room, or put at the side of the room or lectern until the time comes to use it. When it is set up early, it might interfere with the projection of slides or film. It it can be folded down, perhaps it can stand ready in this position until it is used. If not, think of where it should be so that the person putting it on the table will not have to fuss with the a.c. cord. Maybe it can be plugged in and just moved to its proper position without having to fool with the cord. Perhaps using a rolling table is the best way. In any event, the ease with which the final operation takes place will depend on pre-thinking of the a.c. extension cords, the table, if necessary, and your original sketch of the room. Check the unit for cleanliness, good lamp (and spare, of course), and proper location with respect to the screen so you can save time at the setup site. The overhead can also prove to be a problem when it is positioned because of the seating arrangement and sight lines. This is something that should be discussed with your client, and the people setting up the tables and chairs in the room, and will possibly take a bit of diplomacy to set up quickly and easily.

VIDEO

If video cassettes or other tape formats are to be used in the presentation, the number and location of the receivers or monitors with respect to the audience is of utmost importance. The smaller size of the screens (compared with the projection screen) means more critical arrangement for all to see properly. They should also be high enough for good viewing from the back seats. This might mean using special tables, or extending-leg projection stands to hold the receivers.

The video player can be located at the rear of the room near the projectors, where you can run the machine, or near the front where the presenter can operate it. This must be decided far enough in advance so the person speaking can be coached on the operation, and so you can judge the length of cable you will need to set up to be able to feed both (or all) of the sets with proper splitters, distribution amp., etc.

Will the presentation require a start/stop operation on the same cassette? If so, the machine should have a pause control to permit stopping the tape without having it return to the cassette each time. This will also make it easy to start it up again and will also have the tape cued up properly if the cassette has been prepared correctly. This is the time, in the shop, to get the system hooked up, checked out, and cleaned up for proper operation during the show. All cables (video, audio, rf) to be used should be tested with splitters, and so on. To be sure there is no breakup, check the sound level. A color bar tape should be used to preset the receiver/monitor controls for the best color. The switches on the rear of the cassette player must be in their correct positions. Cable lengths and feed-through adapters, and terminations or antenna transformers should all be packed and checked off the list.

PROJECTION SCREEN

I left the projection screen out of the discussion because it seems pretty obvious that one must be taken along if there isn't one in the room. Sometimes, however, because it is so obvious, it is forgotten. The screen should be located for proper sightlines by the entire audience. The size of the screen is important, and depends on the size of the room and placement of the audience, and the space available for setting up. It will also depend on the throw distance from the projectors, the slide formats (horizontal or vertical), and brightness. Here is the time to figure the lenses of the projectors to be sure that all contingencies are covered. If one arrangement won't work, another may. This will depend on the ability of the projectionist to adapt the image size to the screen width; the wider the range of lenses available, the greater the number of possible setups he can make to fit the room. Just remember that the biggest screen you have might not always be the proper answer. Sam Goldwyn once said that a wide screen just makes a bad film twice as bad.

The thing that makes a good a/v person better than the next one is the professionalism demonstrated in arranging for the setup, the way meetings are held before the show, and the way the equipment is finally set up and how it all works out. If the presentation goes well, don't expect applause, but you might have made a good customer with a continuing business. If the show flops because of you—forget it! ■

db theory&practice

NORMAN H. CROWHURST

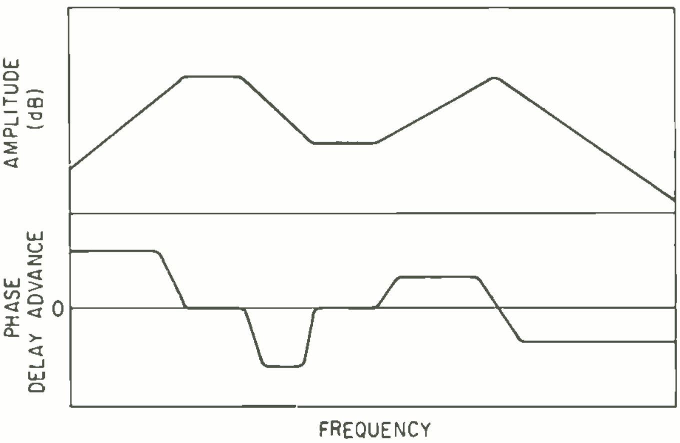


Figure 1. An idealized interpretation of the meaning of minimum phase response. Note that when amplitude response (top) is level, the phase response is zero. response is proportional to the slope of the amplitude response, shows advance for an upward slope, delay for a downward slope.

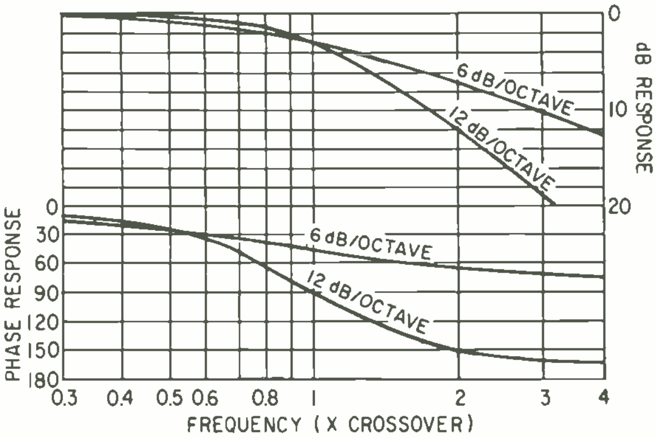


Figure 2. Amplitude and phase response of 6 dB/ octave and 12 dB/octave low pass sections, showing that they do comply with minimum phase definition.

• In considering putting together a multi-way loudspeaker system, one of the first questions about crossovers is, "How steep do we make the crossover?" Among audio people, we find two schools of thought, one of which says, the steeper the better, while the other takes the reverse view, that one should use the simplest crossover that one can get away with.

The advocates of steep crossovers do so on the basis of avoiding interaction between units in the vicinity of crossover. For example, if you use a 24 dB/octave design, there is only one frequency at which both units deliver equal energy, and by quarter of an octave away, one of the units is down to one-fourth the energy of the other. In contrast, with a 6 dB/octave design, it takes an octave to reach this ratio.

If you do not go any further, that sounds like a pretty logical argument for the steepest crossover you can afford. But anyone who has been around for some time should get suspicious of any statement that says "the more the better" in any direction, without telling what is the other side of the picture.

Why do we want crossovers, or multi-way systems, in the first place? Because it is easier to design a unit that is reasonably flat in its frequency response, for only part of the audio spectrum, than it is for the whole range. This means that if we could design a loudspeaker that is flat throughout the whole audio spectrum, we could forget about crossovers altogether.

Next, having settled for doing a really good job over only part of the audio spectrum, such a unit usually becomes erratic in its response beyond the range of frequencies for which a good degree of flatness has been achieved. You want to roll that unit off fast so its irregularities do not show, and let the other one take over, where it does the job for which it was designed.

Sounds good doesn't it? That is why so many advocates go that route, probably. Some of the better informed then start talking about minimum-phase response and non-minimum-phase response, but since this is over many audio people's heads, they'd rather not get into that. But let's see what it really means.

MINIMUM-PHASE RESPONSE

Minimum-phase response, put simply, refers to a system in which when amplitude and phase response are compared, an upward slope always corresponds with a phase advance, and a downward slope always corresponds with a phase delay (FIGURE 1). This means that where the amplitude curve is flat, phase response should be zero—no phase shift.

Referring to the constant-resistance type of filter design, which if not universal in crossovers is at least representative of the general trend, the response of each filter will be minimum phase if the ultimate roll-off rate is either 6 dB/octave, or 12 dB/octave.

The maximum phase shift, reaching its full slope of 6 dB/octave, is 90 degrees. With the 12 dB/octave type, at crossover, the slope is 6 dB/octave, and the phase shift is also 90 degrees, preserving that minimum-phase relationship (FIGURE 2).

But now let us look at how a steep type could be synthesized (FIGURE

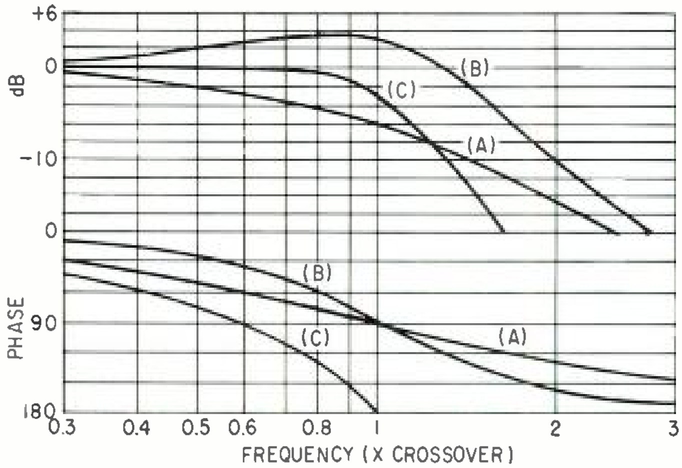


Figure 3. Synthesis of a 24 dB/octave low pass, from a 12 dB/octave, with 6 dB loss at 6 dB slope, (A); and a 12 dB/octave, with +3 dB slope, (B); combined response is at (C). Note that although amplitude response (B) has upward slope, its phase response is still delay, which violates minimum phase definition.

3). If we use two sections, one with 6 dB loss at its 6 dB/octave-slope, 90 degree point, and one with 3 dB boost at the same frequency, also at a 6 dB/octave-slope, 90 degree point, and a peak of 3.6 dB just below that, the combination will make a 24 dB/octave roll-off, with 3 dB loss, 12 dB/octave, 180 degrees phase shift at the design point.

That is one way of putting together a 24 dB/octave crossover filter. What many people do not realize is that whether you actually put it together that way or not, the relative amplitude and phase responses will be the same. So isn't that minimum phase?

To understand why not, look at the peaking network's response. Like any other 2-element filter, it has a 6 dB/octave downward slope with a 90 degree phase delay at its design point. But what about that peak? Like any other 2-element filter, its phase shift goes through a total of 180 degrees phase change, all delay for the low pass section.

At the peak, instead of having zero phase shift, it will have close to 90 degrees. Below the peak, it has a rising amplitude response, with a phase delay, not an advance. So it does not obey the minimum phase rules in this region. This means that the combined network cannot do so either.

ALL-PASS NETWORKS

Lattice networks, are called all-pass networks, which means that all frequencies are passed with uniform amplitude response, but with a relative shift in phase (FIGURE 4). Such a network is definitely not minimum phase, because a flat response should have zero phase shift.

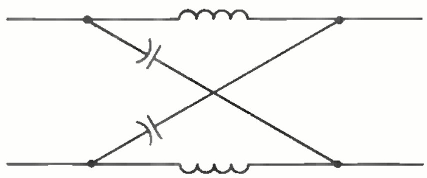


Figure 4. A simple lattice network. At low frequencies the through connection is from upper to upper and lower to lower. At high frequencies, the through connection is from upper to lower and vice versa, a 180 degree phase switch. At intermediate frequencies, the transfer swings around from one to the other, if the loading is correct, the amplitude does not change.

Now consider what happens when you put together two sections, a low pass and a high pass, to make up a crossover, using 24 dB/octave design (FIGURE 5). Each filter goes through a full phase change of 360 degrees. And if it uses constant resistance design, the phase responses will look identical, so that the output of each filter is always at 360 degrees to the output of the other.

As this really means that they are in phase, it looks really convenient; the outputs will always be completely additive, with no phase difference. What this omits is the fact that within much less than an octave, this combined output whirls through a phase change of 360 degrees. The composite

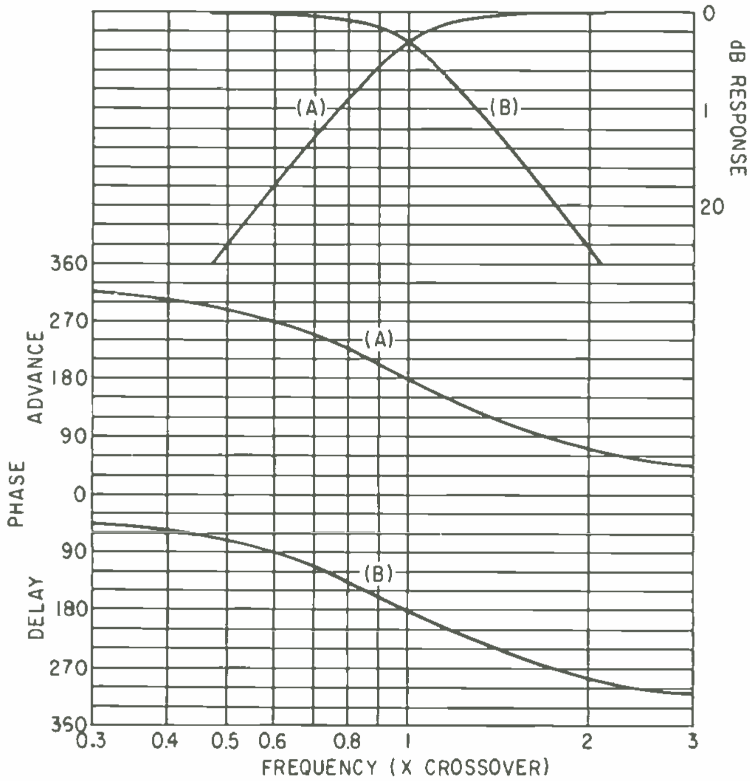


Figure 5. Response curves for a 24 dB/octave crossover. Note that the phase difference is a constant 360 degrees, which is the same as zero, on a continuous sine wave signal. The signal delivered to both outputs is always in phase.

output is certainly *not* minimum phase.

Let us look at the 12 dB/octave case a little closer. Each section shifts by an ultimate of 180 degrees and if correctly designed and used, the outputs of the two filters are always precisely 180 degrees out of phase. That is kind of convenient because all you have to do now is to reverse the connections of one voice coil and the output from both units is always in phase.

But still the phase shift passes through 180 degrees over the same frequency range that the 24 dB/octave combination passes through 360 degrees. We have just reduced the effect. The combined acoustic output is not minimum phase. But we will admit, it is better than the 24 dB/octave system.

Have I now convinced you that the simpler crossover filters are better? As I said last month, if loudspeakers behaved like resistors, this would be true. Now we can take another look at the reason we need crossovers in the first place.

WHY CROSSOVERS?

Audio people were first concerned with the irregularities in a loudspeaker's amplitude response. Because it seemed that phase did not matter, that view held for a long time. But why is amplitude response important? If a system has a gradual change in response that slopes, say 6 dB, from one end to the other of the audio spectrum, you would have to a-b it with a flat response to tell any difference. And then the only difference you would notice would be a slight change in emphasis of highs over lows, or vice versa.

But if you have a unit with a 6 dB hole or, worse yet, a 6 dB peak, in its response, the difference is much more immediately noticeable as coloration, or a characteristic tone. If the only difference is because certain frequencies get reproduced at a 6 dB higher or lower level than other frequencies, it would not be nearly so noticeable. Unfortunately, this is difficult, if not impossible, to demonstrate. The reason the shape of the amplitude response makes a difference greater than it looks on the response curve, is because it is inevitably accompanied by non-minimum phase response.

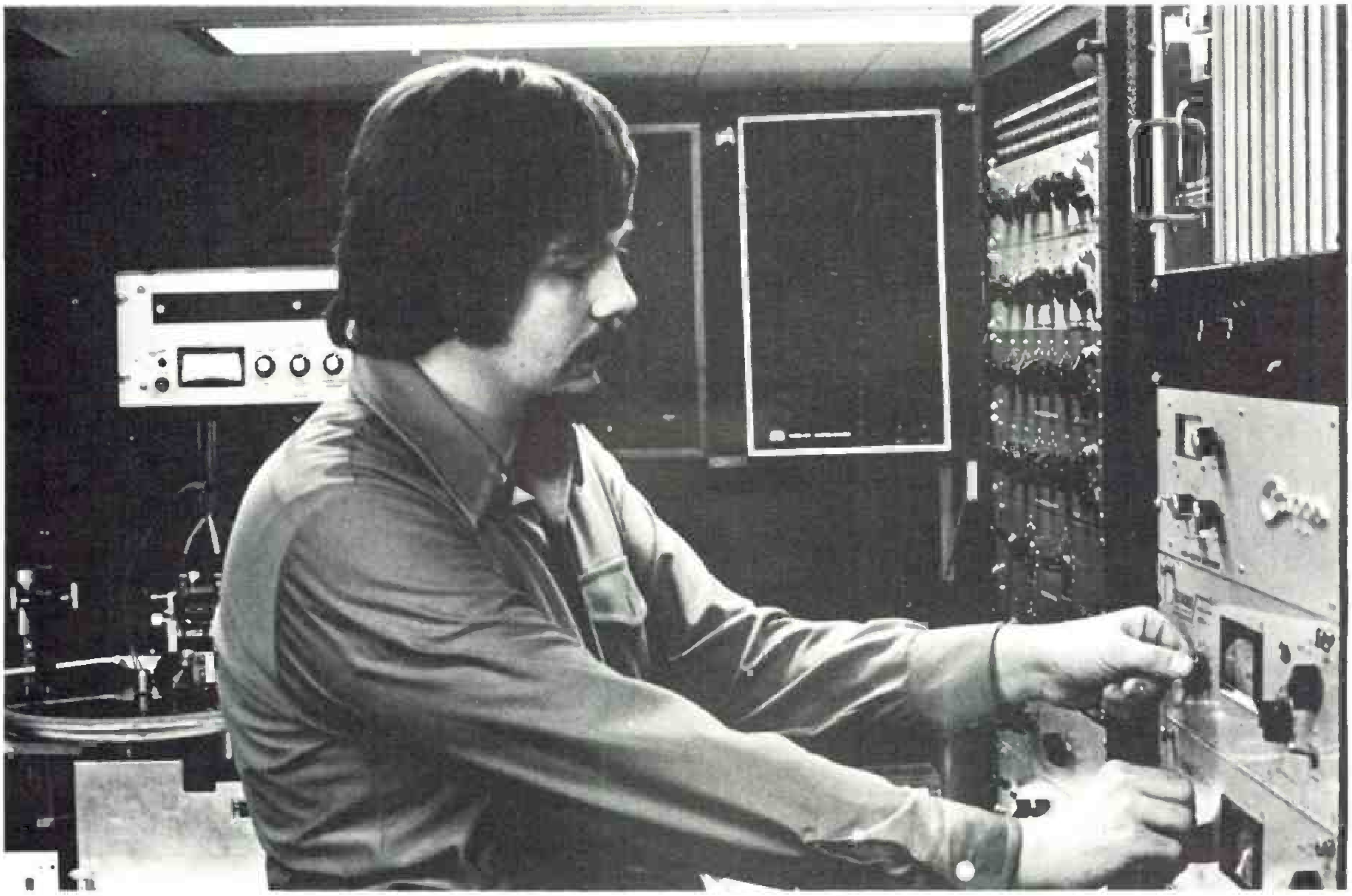
This means that the loudspeaker behaves as if certain frequencies are radiated from "further back." Other frequencies may be radiated more or less normally, but frequencies in the vicinity of the peak or hole sound as if they were radiated by a unit at the back end of a short tube creating the apparent coloration.

Now, the reason you want to use a crossover is because your carefully designed woofer, for example, gives this kind of impression if you allow it to radiate mid-range or highs. Perhaps you have a different reason for wanting to roll off the mid-range or tweeter, at its low end—to protect it from distortion. But if you use a steep slope crossover, the effect of the crossover, reproduced through two "perfect" loudspeakers (which, if you had, you would not need the crossover) has the same effect due to its non-minimum-phase properties.

To sign off this month's column, I hope I have made clear that what you do is going to be a compromise. You want to do the best possibly with what you have. Some years ago, loudspeaker designers worth their salt realized that crossovers were not the panacea. You must start with good loudspeakers, as good as you can get them. Then you can put together the best of several, by designing an appropriate crossover for the purpose.

But what slope is best, for a given combination, is something that requires careful consideration for that combination. ■

According to TRUTONE RECORDS..."The Stanton calibrated 681 series is our total point of reference in our Disc Mastering Operation"



"Carl Rowatti, Chief Engineer, adjusting the Program limiters prior to cutting a master lacquer".

Trutone can be described as a family enterprise . . . but what a family! Father Lou Rowatti is the President; Son Carl is Vice President and Chief Engineer; and daughter-in-law Adrianne handles the business end of the operation. They have great pride in their family, in their family's enterprise and in their products. That's why they insist on using the best — always.

Trutone Records in Northvale, New Jersey always uses the Calibrated Stanton Triple-E for A-B comparisons between tape and disc. They also use the Triple-E to check the frequency response of the cutter head (they'll record a 1,000 Hz tone and a 10 kHz tone twice a day to check the condition of the cutting stylus and the high end frequency response of the cutter head).

They make test cuts and play them back, using the Triple-E for reference, as high as 15 kHz all the way down to 30 Hz. Carl Rowatti says "We use the Stanton Calibrated 681 series as our total point of reference in our disc mastering operation. Everything in the studio is judged — and we think perfectly judged for quality—with this great cartridge".

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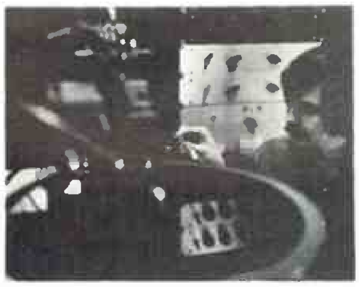
Lou Rowatti inspects a master lacquer. Adrianne checks the lathe.



Carl Rowatti adjusts the pitch computer on the mastering lathe.

Carl installs the Stanton Calibrated 681 Triple-E on the playback table.

Lou Rowatti (The Prez) adjusting the high frequency limiter in his cutting room.

Circle 27 on Reader Service Card

db broadcast sound

PATRICK S. FINNEGAN

● Telephone lines are used for a great many purposes besides normal phone conversations. Program audio is fed over the lines by broadcasters, background music services, as well as special information and marketing information networks for feed to f.m., SCA subcarriers, etc. Many other data signals, such as computer, teleprinter, and telemetry data use telephone lines. Our concern in this article is the use of local lines for sending audio around town rather than some inter-city network arrangements. If you have not had the experience of using telephone lines for audio in the past and are now faced with the prospect, the information given may help you to understand better the problems involved.

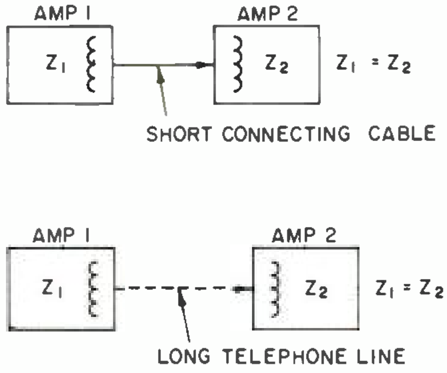


Figure 1. The two audio units must be matched, regardless of the length of line between them, or they won't work properly.

BASIC PARAMETERS

When connecting *any* two audio units together by wire circuits, short or long, such as a speaker to an amplifier or a microphone to a pre-amplifier, the two interfacing units must have their electrical characteristics matched properly or the units will either not perform properly or not at all. These characteristics must match even if there is no interconnecting wiring and the terminals are directly tied together. The important characteristics are the impedances of the sending and the receiving units, balanced and unbalanced circuit isolation, and the signal levels.

IMPEDANCES

All audio units are designed around their input/output impedances so that units can be matched up. The impedance matching (or mismatching) affects the units' operations and signal levels, as well as the bandpass achieved, so every effort should be made to match these properly.

Impedance is a value that contains both a resistive element and a reactive element. The resistive element will cause equal loss to all signals in the bandpass, while the reactive component is frequency-dependent and will cause loss according to frequency. This reactive component may be either inductive or capacitive or it may be cancelled out (zero reactance) with only the resistive element remaining.

Inductive reactance is directly proportional to frequency, that is, its value increases as the frequency increases. In audio work, the inductive reactance is usually not a problem unless large chokes, transformers, tape hands or similar inductive devices are concerned. But, except for transformers, these are usually not involved in the input/output impedances.

Capacitive reactance, on the other hand, is inversely proportional to frequency; as the frequency increases, the reactance value decreases. This reactive component will cause the most problems in audio work, whether it is designed in series or parallel with the circuit. When in shunt across the circuit, it effectively bypasses the upper audio frequencies before they can reach the load. If it is in series with the circuit, it will discriminate against the *low* audio frequencies.

INTERCONNECTION

When the two units are connected together by short-length circuits in the studio or the building, the interconnecting wiring should not affect the two units. When this is done carefully, except for a few critical circuits, such wiring is mostly not significant to the operation of the two units.

But when we wish to place our audio units across town from each other, the characteristics of the very long interconnecting lines do become very significant and call for more serious treatment. Our basic requirements of the two audio units remain the same, but now the distributed resistance and reactance of the line add up and affect our match considerably.

LINE CHARACTERISTICS

The circuit leased from the Telephone Company will be a balanced, unloaded cable pair. Each wire in this

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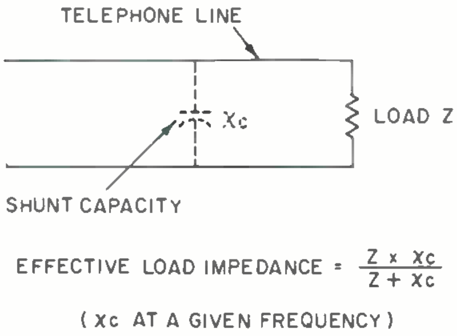


Figure 2. The shunt capacitive reactance is in parallel with the load impedance.

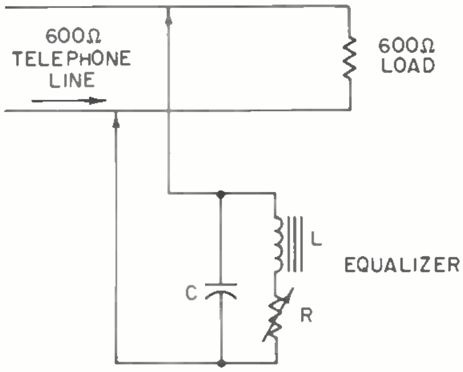


Figure 3. A simple passive equalizer that is a parallel resonant circuit. More sophisticated equalizers are available.

pair will be of very small diameter, usually of 26 gauge. Because of the small diameter of the wire, resistance can be high and will become more significant as the length increases. How much total resistance of the circuit there is depends upon the length of the circuit and the particular type of wire used, but this can typically range from 200 ohms to 500 ohms for one mile of cable length. That resistance will cause a considerable loss to the signal as it passes through the cable.

The same pair of wires will also have an approximately 0.083 μF shunt capacity across them in each mile of cable, adding directly as the cable length is increased. For example, three miles of cable will have a total of 0.249 μF shunt capacity. It doesn't take much calculation to realize what this will do to the higher audio frequencies.

Actual losses due to the resistive and reactive components in the line will be approximately 2.9 dB at 1 kHz, and the losses increase rapidly at the upper frequencies.

EQUALIZATION

Remember that we must still match our two audio units, but now we have a significant amount of resistance and capacitance between them. Unless we can reduce the effect of these components, our audio units won't work properly and signal transfer will suffer in both signal level and response. The counteraction of these line characteristics is called *equalization*. Equalization can be done by the one who orders the circuit, or an equalized circuit can be ordered from the Telephone Company. Naturally, an equalized circuit will cost more than a non-equalized circuit, the price depending on the grade (according to bandpass) ordered.

LOADING

A simple form of loading which the Telephone Company does on their regular voice circuits produces enough equalization to make a reasonably good voice channel. Loading coils are added in series with the circuit at approximately one mile intervals. The actual distance between the coils and the coil value depends upon the particular circuit, but a typical value is 6 mHy. The coils' inductances neutralize part of the line capacity but also increase the normal 600 ohm line impedance to approx. 1,000 ohms. The net result is a bandpass from approximately 300 Hz to about 3200 Hz. Above this figure, the response drops off very steeply. Although this form of equalization makes a reasonably good voice circuit, the line cannot be further equalized to improve the bandpass. The loading coils must be removed when a wider bandpass is required, and a different form of equalization performed.

LOWER THE IMPEDANCE

When the circuit is very long, the first step is to reduce the impedance of the two audio units. That is, if they are normally set for 600 ohm impedance, change the transformer taps to 150 ohms. This process in itself will cause some signal loss, but it will make equalization easier. Since the line capacity is across the load impedance, it will have less effect across 150 ohms than across 600 ohms. For example, a cable three miles in length will have approximately 0.25 μF shunt capacitance. This will have a reactive value of about 650 ohms at 1 kHz, 125 ohms at 5 kHz, and only 70 ohms at 10 kHz. At each of these frequencies, the load impedance will be a parallel combination of the capacitive reactance and the load impedance. If these values are in parallel with 150 ohms, there will be less effect than if they are across 600 ohms.

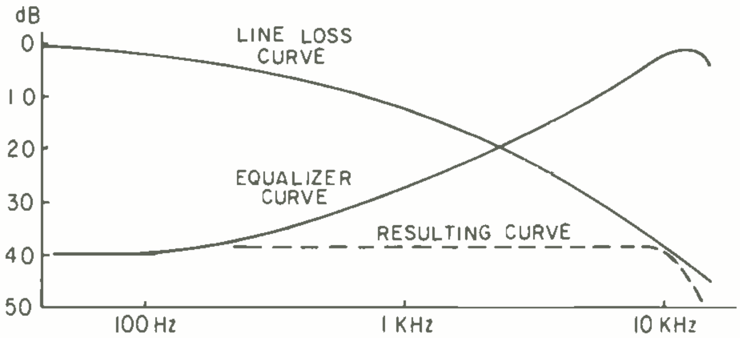


Figure 4. The equalizer curve is adjusted so that it is complementary to the line-loss curve.

BASIC EQUALIZER

The basic equalizer introduces a complementary curve to the loss curve of the line. In effect, it lowers the low audio frequency response down to the value of the high frequency response at the limit of the bandpass desired. The simple passive equalizer is made up of an inductance, capacitance and switchable resistors. This unit, when combined with the line capacity, will create a resonant peak to occur somewhere in the upper audio range. By resonating with the capacity, that partially cancels some of the capacity of the line. The lower skirt of the resonance curve is broadbanded by the use of the switchable resistors which lower the resonant circuit's Q.

LOSSES

Equalization will cause considerable signal loss in the circuit; these losses vary with the circuit itself and the bandpass desired. The wider the bandpass, the greater the losses, so keep this in mind and only equalize the bandpass necessary for the program requirements. For example, a circuit intended only for voice signals does not need a circuit with a 15 kHz bandpass. Although the losses must be made up, there is a limit that can be fed into the head end of the line. This is +8 dB at maximum. More than that will cause cross-talk into other circuits and the Telephone Company frowns on that. The higher the circuit losses are, the worse the signal/noise ratio becomes.

NOISE

Telco measures noise differently than we do. They use a noise reference value and figure above this reference, which is set at 90 dB below 1 mW (600 ohm circuit). If the circuit (to them) has a noise figure of 50 dB, this means 50 dB *above* the reference. To us, this is 40 dB *down* from program reference of zero dB, or, if we are sending at +8, the noise level is down —48 dB.

STEREO

When stereo signals will be sent over a pair of lines, the circuits call for more critical consideration. For example, a musical program from an auditorium may be sent back to a recording studio and later receive air play on an f.m. station with its matrix arrangement, or anywhere that the program may be summed for mono use.

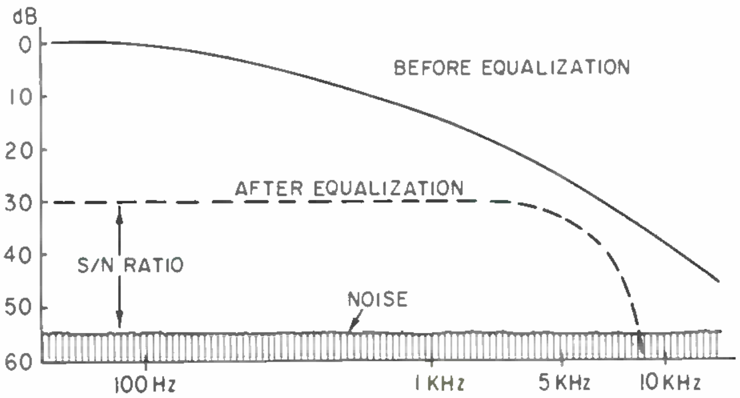
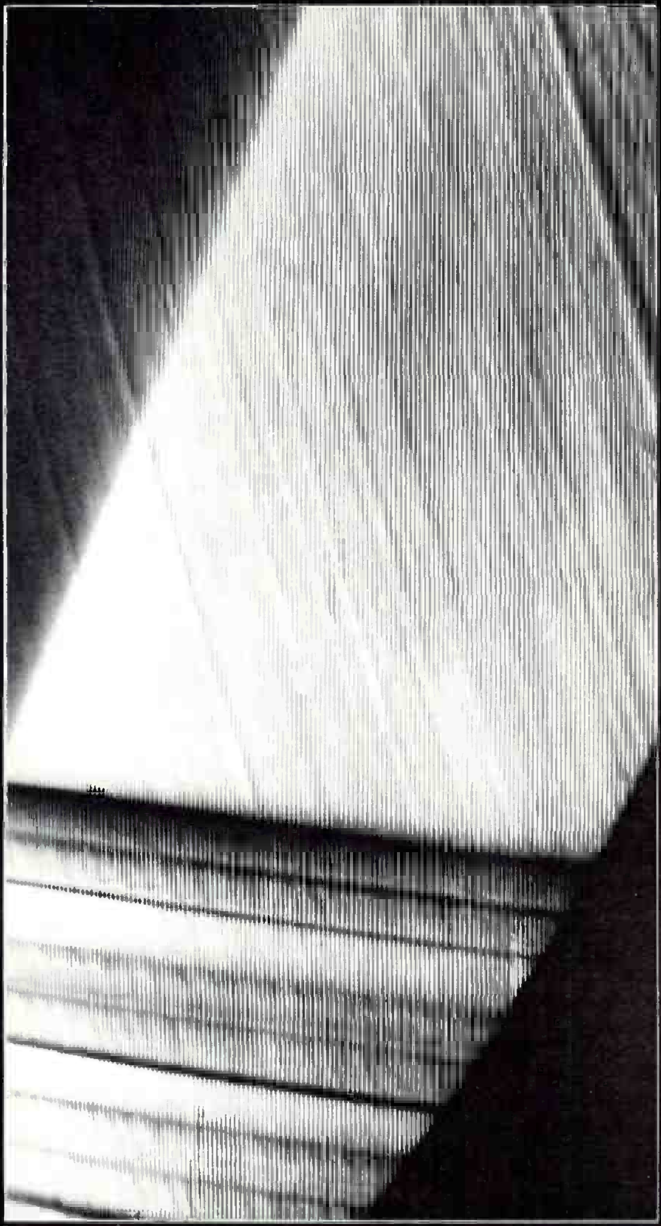


Figure 5. When there is too much equalization, the s/n ratio will suffer.

The two circuits must be identical in amplitude response and phase, or separation will be lost between channels. So, equalize very carefully and make each curve identical. But this won't be easy. Phase is basically line length and there isn't much you can do about that, so consider using one of the electronic phase adjusters, such as the Garron phase enhancer.

SUMMARY

Audio units connected together must be matched or they will not perform properly, and this is so whether they are next to each other or across town from each other. The long telephone line does have distributed capacity and resistance that will make the matching more difficult, or impossible, so its effect must be reduced by equalization. But equalization introduces losses and should not be overdone, since these losses will cause the signal/noise ratio to suffer. If stereo is to be fed over a pair of circuits, match them up very carefully and identically. ■



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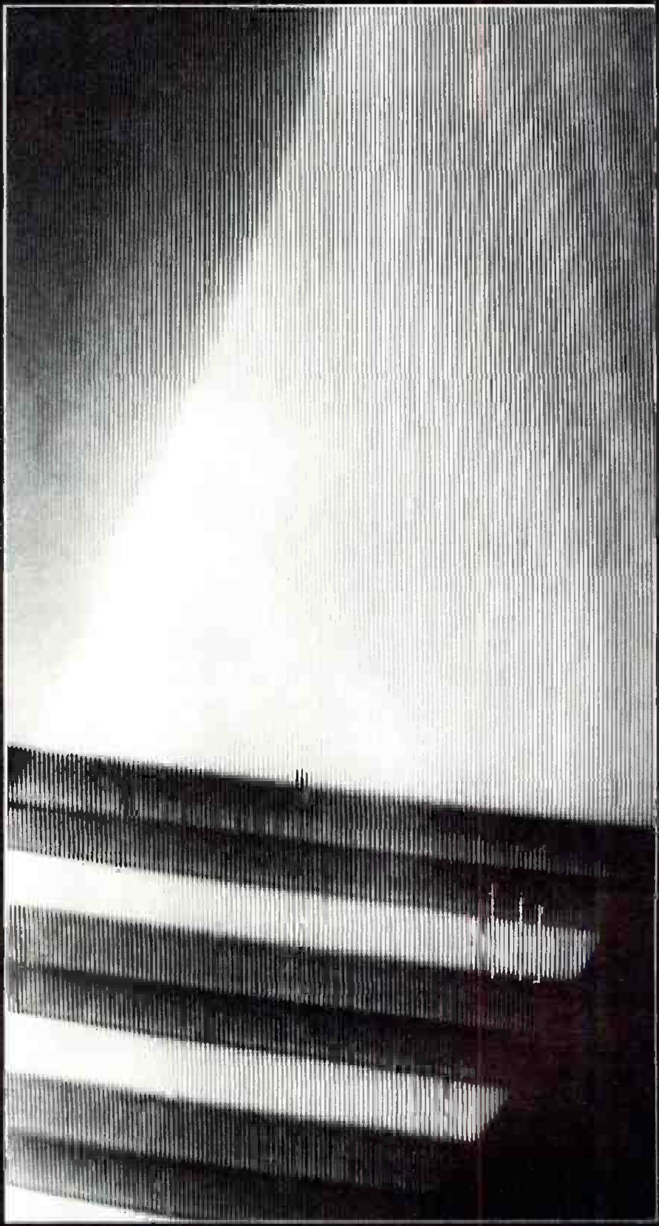

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• An eight-channel version of this manufacturer's MX-5050, MX-5050-8 features half-inch tape, 15 and 7½ in./sec. tape speeds, and a variable speed (± 10 per cent) d.c. capstan servo as standard. The unit offers front panel edit and cue, motion sensing, synchronous reproduce, XLR connectors, variable or fixed outputs (at +4 or —10 dBm, switch selectable), front adjustable bias and equalization, and a built-in test and cue oscillator.

Mfr: Otari Corp.
Price: $3.995.
Circle 50 on Reader Service Card

PORTABLE GRAPHIC EQUALIZER

• This battery-operated Six Band Graphic Equalizer is designed for portable use with instruments. The six frequency bands fit the total response of bass, electric guitar, or keyboards. Each band can be selectively boosted or cut by as much as 18 dB. The dynamic range exceeds 100 dB.

Mfr: MXR Innovations, Inc.
Price: $79.95.
Circle 51 on Reader Service Card

WHITE NOISE GENERATOR

• A digitally generated noise signal is radiated by MZL 400 acoustical white noise sound source. It is claimed that the sound spectrum is totally independent of ambient temperature, operating voltage, and the aging of components. The unit, employing cmos integrated circuits, is battery-operated.

Mfr: Sennheiser Electronic Corp.
Price: $78.
Circle 52 on Reader Service Card

KEYBOARD MIXER

• Designed especially for mixing keyboard instruments, Keyboard 100K provides completely independent volume, bass, and treble controls for up to six different keyboard instruments. Each channel has two input jacks, one for high level instruments and the other for low level instruments. Distortion is claimed at 0.08 per cent and high output level capability up to 10 volts rms.

Mfr: Tapco
Price: $189.
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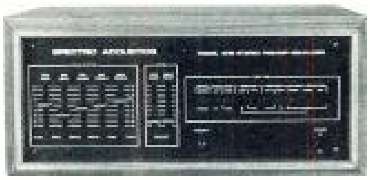
GRAPHIC EQUALIZER



• Five band model 101B preamplifier/equalizer features oil-damped slide controls and hybrid gyrator/coil design which can be pushbutton patched into line or tape outputs or bypassed for e.q. in/out comparisons. The device has dual taping facilities with twin monitor circuitry, low impedance buffered outputs, bi-directional bypass copying and simultaneous recording from any selected course. High voltage/high current driver i.c.s on all outputs are direct-coupled, capable of driving a dozen recorders or power amps simultaneously.

Mfr: Spectro Acoustics Inc.
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- **The Tape Recorder**
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● Jumbo l.e.d. readout figures on DSC-1 digital stop clock can be read 25 feet away. Elapsed-time displays include hours, minutes, seconds, and tenths of seconds, with a maximum count of 9 hours, 59 minutes, 59.9 seconds. The clock may be controlled through pushbuttons or remotely via switches. It is possible to start and stop the count any number of times without resetting the digits to zero, in order to accumulate the total time of separate events.
Mfr: Caringella Electronics Inc.
Price: (Kit) $79.95; (Assembled) $99.95.
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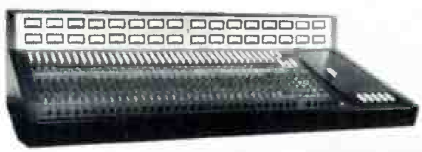
MULTIPLE SIGNAL CONSOLE



● Inputs for 32 microphone, 32 line, 32 tape, seven line level direct to control room monitor, and four echo returns are supplied on model 1032-32 console. There are 32 direct outputs, four echo send, four quad, two stereo, one mono, four control room monitors, two studio and cues A, B, and C outputs. A 15 frequency graphic equalizer is included on all 32 inputs and also on echo return. 32 vu meters monitor control room, quad, echo, stereo, mono and cue C. There are straight line attenuators for all inputs and mono, plus two four-gang for sub-mix and one for quad, and a two-gang attenuator for stereo. A generator is provided for level setting and calibration. Talkback is possible on most functions. Tape remote controls are provided for five tape recorders/reproducers and a 480 point bay field for 96 tapes, four echo return inputs, 16 spares, 66 blanks for potential added applications, and four signal generator outputs.
Mfr: Spectra Sonics
Price: $56,576.
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MODEL 8038
24, 28, or 32 Input Channels
24 Track Metering and Monitoring
4 Reverberation Sends
8 Reverberation Returns
4 Cue Sends

MODEL 8048
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These two consoles have evolved from years of custom engineering of large desks for leading studios around the world. Now you can take advantage of Neve craftsmanship, quality construction and reputation for unequalled reliability and service, at a price you can afford. Should the 8038/8048 not suit your requirement, we have other mixing consoles starting at $8,000 covering virtually any application in professional sound. Move into the Neve world of excellence. It's in your future.

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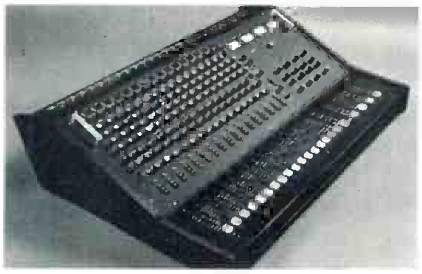
16-TRACK REMIX CONSOLE



• Complete stereo and remix monitoring with no patching necessary for up to 16-track recording are available on Series II consoles. The units have up to 24 inputs and 4 or 8 mixing buses. Two stereo effects returns are provided for. Other features include four auxiliary sends with pre-post fade selectivity, input overload detector, two fixed frequency shelving equalizers, two overlapping sweepable frequency peaking equalizers, variable hi-pass filtering, patch points, post fade direct outputs, and Waters 105mm. conductive plastic faders. Options include a sound reinforcement output section, carbon-element faders, and an ATA flight case.

Mfr: System & Technology in Music, Inc.

Price: From $4,830.

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MODULAR RECORD-PLAYBACK/ PLAYBACK EQUIPMENT

• Century II tape cartridge equipment offers a modular system of mono and stereo same-size record-playback and playback units in single, dual, and triple table top mounts and triple rack mount. Features include both secondary and tertiary cue tones, primary cue tone defeat, audio mute on playback, logging signal in- and out-put, and built-in splice finder. A high speed option is available. The unit uses peak reading record level meters with led overload indicators and digital tone detect. The dual unit, either twin playbacks or a record/ playback and playback unit, measures 11⅜ x 5¼ x 15 ins.

Mfr: Sparta Div. (Cetec)

Circle 58 on Reader Service Card

The clubhouse/grandstand of Chicago's Hawthorne Race Course.

HERBERT JAFFE

Above the Roar Of the Crowd

Sound-level sensing controls automatically raise the level of racetrack public address above finish-line hysteria.

POST-TIME: the last-minute ripple of urgency at the betting window is cut short by the bell and the announcer's call "and . . . they're off!" In the suddenly hushed stands, a rumble of tension builds to a roar as the horses thunder around the track. Quarter-pole—half-mile pole—three-quarters. As they come into the stretch, excitement peaks in a 110 dB explosion of sound—loud enough to drown out the crucial 30-second finish and announcement of winners.

Unfortunately, when the sound is lost, money is lost. Without audible sound, people can't hear the names of horses being scratched, which alternates are running, or when the race is starting. Without sound, bets aren't placed in time. Without intelligible sound for an accurate race call, to quote one veteran racetracker, "It's just a bunch of horses running."

That no longer happens at Hawthorne Race Course, the 84-year-old track in the suburbs of Chicago. A thoroughbred and harness racing establishment that pioneered with the first public address system in both grandstand and clubhouse, Hawthorne has made history again with the newest state-of-the-art system that combines sound-level sensing controls and professionally selected loudspeakers to overcome the variations of radically different acoustical and environmental extremes throughout its eleven public areas.

Initiated to refurbish the original, frequently-patched 20-year old system, the new installation was designed by Peter C. Mitchell, production engineer of Audiotech (then known as A/V Technical Services) the Chicago-based sound specialist firm. The sound reinforcement system for racing is a 2200 watt binaural, ten group/

Herbert Jaffe is director of marketing at Atlas Sound, Parsippany, N.J.

Paired with color c.c.t.v. monitors in a complementary playback/daytime paging system, environmental speakers keep announcements audible over the noise of the crowd.

20 zone complex—two loudspeaker zones to a group. In addition to new electronics for existing loudspeakers and sound columns, it teams a unique audio-level sensor, 250 per cent more amplification power, and over 100 new environment-resistant loudspeakers to provide total coverage with automatically modulated volume (up and down in proportion to ambient crowd noise or levels of excitement) and keeps the race call as clean, clear, and intelligible during the two minutes of the contests as when bets are being placed.

OPERATIONAL FLEXIBILITY

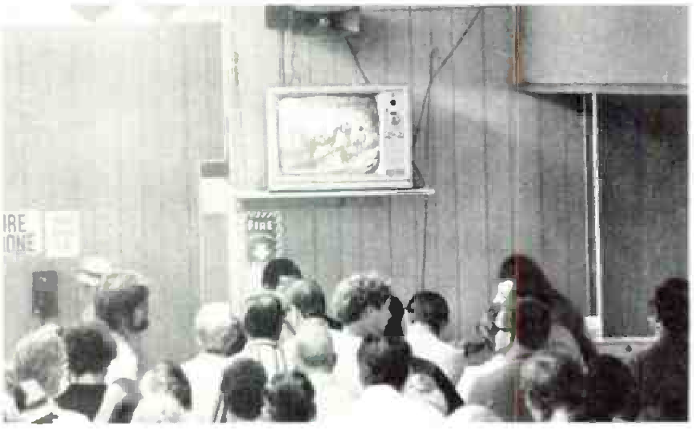
The refurbishment program, started in 1974, was designed primarily to provide audibility for crowds ranging from the typical 25,000 Saturday attendance, to a high of 37,000 patrons. Emphasizing operational flexibility and quality of reproduction to achieve its total-audibility goal, the new sound reinforcement system was developed after extensive research and evaluation of equipment, as well as restoration of all usable components of the old system.

To meet the flexibility/quality criteria, Mitchell blanketed the grandstand's outside seating and skirt areas with twenty-four Atlas Sound RC-6 re-entrant projector horns, equipped with PD-30T 30-watt compression-drivers, mounting them on 60-ft. centers parallel to the downward seating slope on the beams of the super-structure area some 50 feet high. Since the grandstand area is outdoors, yet 50 per cent confined within a roof and back wall, the 360 degree dispersion of this radial reflex-type horn assured heavier direct penetration of distance and noise without being "overdone," says Mitchell.

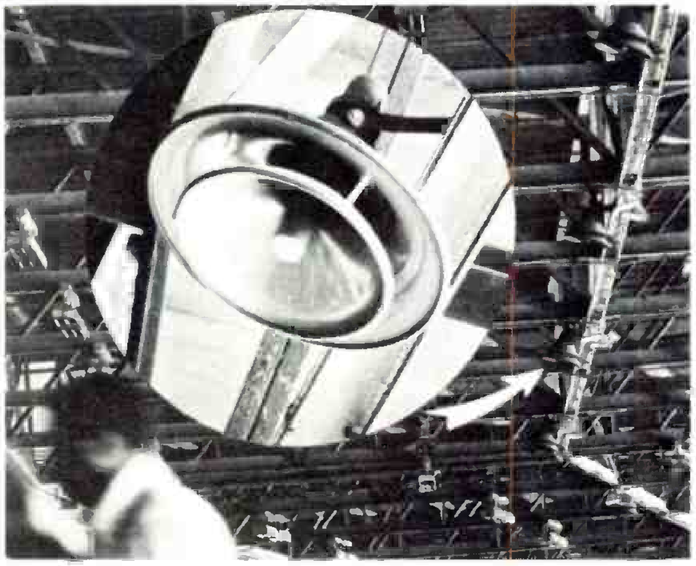
Each weather-sealed all-metal loudspeaker has a 25-inch diameter bell, is 13¾ inches long, and provides a low-end cut-off of 140 Hz. Shure Brothers M-63 Audiomaster sound shapers were provided with each RC-6 loudspeaker zone to eliminate the "boomy" frequencies which can result from resonance-coupling of symmetrically-installed speakers.

In addition to the new RC-6 speakers, twenty existing Executone re-entrant speakers were equipped with new transformers and relocated to provide additional coverage of the binaural group in the grandstand and clubhouse skirt area.

Another component of the major binaural system was



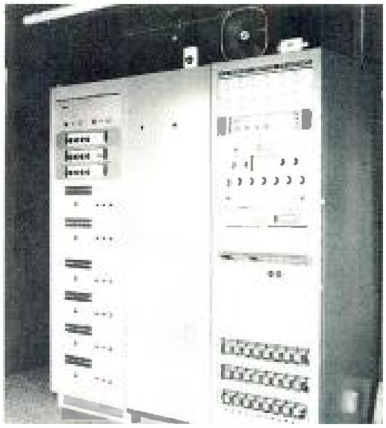
One of 35 closed circuit t.v. units, with an environmental speaker above it.



Mounted (arrow) on structural beams of the grandstand superstructure, speakers provide dispersion and audio penetration in the outdoor grandstand area.

Atlas Sound's APC-30T wide angle re-entrant speakers used to provide the medium penetration and horizontal dispersion required in heavily peopled carpeted areas throughout the building interior. Installed on ceilings and support beams, the wide-angle 30-watt speakers had an additional advantage for Mitchell: a 3-way omni-pur-

Double re-entrant projector speakers are mounted on support beams and ceilings with a three-way adjustable bracket.



Located in the main control room, the fan-cooled control console incorporates an automatic lighted relay speaker network crossover (bottom right) for emergency transfer of speakers without dispersion loss in case of amplifier failure.

pose mounting bracket. "You can mount one on a 22 degree angle and still aim the loudspeaker to project specifically where you want," he says.

All binaural-system speakers are linked to a Robins-Fairchild Ambicon 653-SP noise compensating control unit that instantly adjusts gain to accommodate differentials in needed sound. Sound levels are automatically set to correspond to the degree of crowd enthusiasm picked up by a strategically located noise-sensor microphone, resulting in what Mitchell calls a "Lincoln" system—"for the people, and governed by them!"

FAIL-SAFE

The sound reinforcement system components are located within a fan-cooled console in the main control room, where a racetrack operator-technician has complete audible/visual monitoring and overview at all times. In the event of amplifier failure, the console provides an automatic relay network with illuminated display for emergency transfer of speaker output without transmission loss. A television playback/daytime paging facility—150 watts, binaural, two zone sound system operating in conjunction with Hawthorne's thirty-five color closed-circuit t.v. units complements the main system.

The playback system performs two functions—audio accompaniment for the instant video replays of the races and paging during non-race periods. In addition to thirty-five speaker/c.c.t.v. combinations in the Gold Cup, Terrace Room and Turf Room areas, miscellaneous Atlas Sound APC-30T, area-speakers of this auxiliary system also supply daytime paging in fourteen behind-the-scenes locations.

System inputs include the announcer's finish and stand-by microphones, f.m. tuner, eight-track tape deck, cassette deck and turntable. A Shure Brothers SE-30 gated compressor microphone line mixer is used for paging and c.c.t.v. audio of the auxiliary system and a six-channel Precision Electronics G7 microphone/line preamp provides mixdown and control of all input functions of the sound reinforcement system.

The announcer has a multi-function selector switch to combine input of both systems or to choose main public address or television playback independently, enabling him to conclude the announced race with a trailer message transmitted only through the television network and select area-speakers of the auxiliary system.

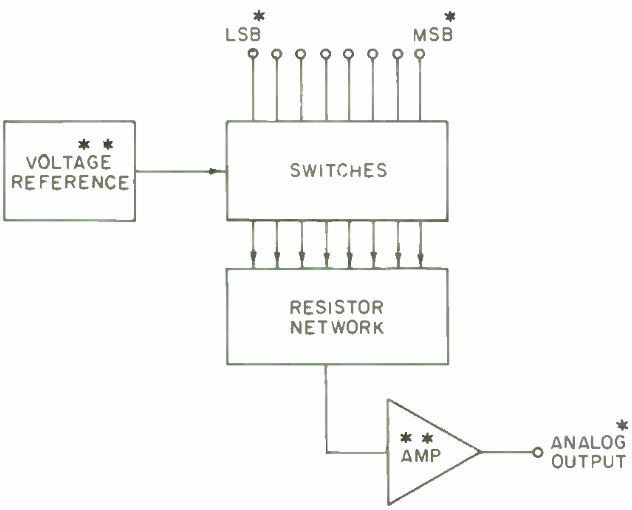
Much as he appreciates the flexibility of the audio facilities, Hawthorne announcer Phil Georgeffe, who provides the oral commentary at six tracks in the midwest, is even more impressed with the excellent quality of sound reproduction. "It takes a sharp, incisive sound for the clarity you need in a race call," he comments.

Mr. Georgeffe looks forward to similarly optimum sound during future announcing assignments at other nearby tracks. Triggered by the satisfaction of customers and Hawthorne management alike, Mitchell and Audiotech are installing two more automatically adjusting environment-overcoming sound systems, one at Arlington Park and the other at Sportsman's Park. ■

WALTER G. JUNG

The Signal Path: Part III Digital/Analog Leveling Control

Control precision can be adapted to audio level control, the basic function which feeds most other controls.



CODE TABLE (B BITS)

Digital Input (MSB LSB)	Analog Output
1 1 1 1 1 1 1 1	255/256 V ref.
0 1 1 1 1 1 1 1	127/256 V ref.
0 0 1 1 1 1 1 1	63/256 V ref.
0 0 0 0 0 0 0 1	1/256 V ref.
0 0 0 0 0 0 0 0	0

Figure 1. Block diagram, basic d/a converter.

IN THE PREVIOUS two installments of this series we have discussed audio signal sources of two distinctly different types—the function generator and the sinusoidal oscillator. Now we will examine modern means to control audio signals in amplitude, by both digital and analog means, and using a combination of both.

When you get right down to it, the control of audio signal levels is one of the most basic control functions. Many other control functions depend directly or indirectly on the control of an audio signal level. Fader position is an obvious one; panning is merely the case of a specially controlled complementary fader. Equalization, once you get beyond the filters that define the spectrum shape which is under control, is also a level control function. Even the filters themselves can be moved about in frequency via gain control, as can the filter shape. Compression, expansion, and limiting are also specialized cases of level control. And, the various noise reduction systems in current use depend heavily on gain control elements.

Audio instrumentation also uses gain control techniques to establish generator levels, filter and oscillator frequencies, and so on. Home audio equipment already uses quad and Dolby i.c. chips which incorporate gain controllers, and more general uses will develop in the future. It is obvious the systematic and predictable control of level in the signal path is a most important requirement, and certainly one which bears close study.

Currently, in studio equipment, automation is a well known term; the computer is now and will further be making its presence felt, on both small and large scales. With this installment we'll begin to look into digital techniques of contolling gain; in later ones we'll look at analog means also. Then, some system applications which exploit the advantages of both will be examined, and we'll show how these techniques can perform other control functions as well.

(continued)

This is the third in a series of Signal Path articles written for **db** *by Walter Jung, a freelance technical writer.*

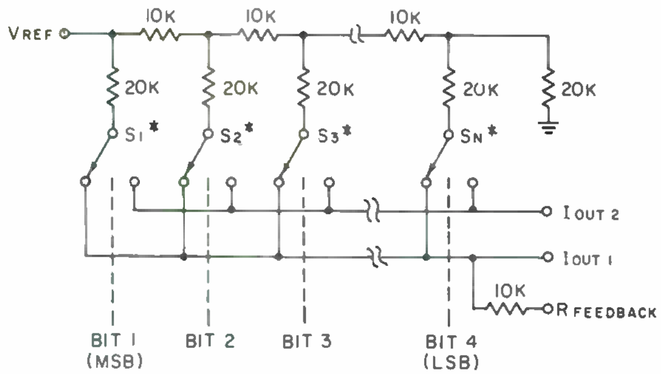


Figure 2. Functional diagram, 7520 family of multiplying d/a converter. All switches are shown in the "l" state.

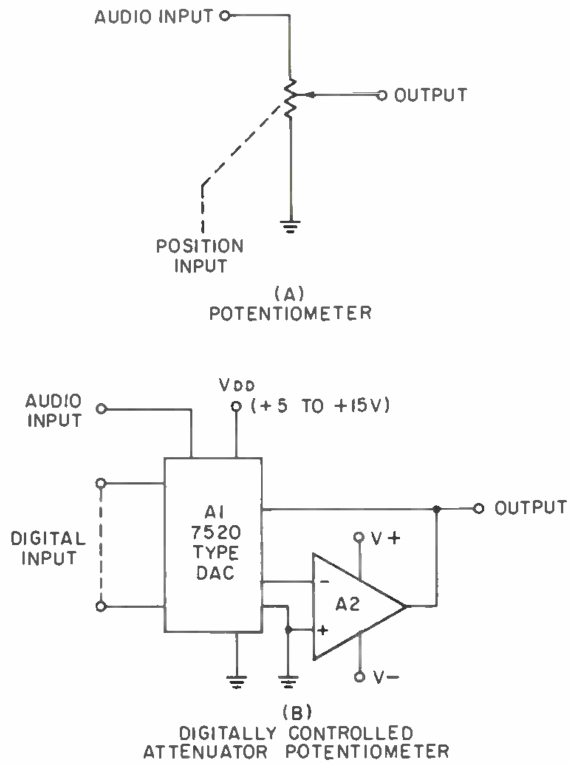


Figure 3. Operation of a multiplying DAC is analogous to a potentiometer. (A) Potentiometer. (B) Digitally controlled attenuator.

THE D/A CONVERTER

Let's lead off with the digital/analog converter—perhaps the most basic interface element between the digital world of the computer and the analog world where our audio signals naturally occur.

FIGURE 1 is an illustration in block form of a d/a converter. There are four basic elements of this machine: a voltage reference (typically 6 to 10V), a set of switches which connect the legs of a resistor network to either the reference source or to ground, the resistor network which scales the currents in binary fashion (½, ¼, ⅛ etc.) and a buffer amplifier. In the switching section, there is one switch for each bit of the converter. A digital *1* applied to a single bit will turn on that switch and cause a current to flow in the appropriate leg of the resistor network—a digital *0* turns it off. For an *N* bit converter, there will exist $2^N - 1$ discrete output voltage steps. Full scale output occurs when all bits are *1* (on), and is equal to $\frac{2^N - 1}{2^N}$.

The LSB, which is equal to the resolution of the converter, is equal to $\frac{1}{2^N}$.

Digital/analog devices come in various grades, or number of bits, for differing accuracy and resolution requirements. For example 6-, 8-, 10- and 12-bit converters are currently available in i.c. form; an 8-bit converter would, for instance, provide $2^8 - 1 = 255$ output voltage steps, such as the example in the block diagram.

To put things into a more practical perspective, suppose the converter example shown were to use a 10 volt reference voltage. Then its full scale voltage output would be 255/256 times 10, or 9.961 (rounded off to nearest mV). The least significant bit in terms of voltage output, would be 39mV (1/256 × 10).

A d/a converter such as we have just described operates from a fixed reference voltage, i.e., the 10V just described. However this is not always true; d/a converters can also operate from variable reference voltages. Some types will operate from some small positive voltage up to 10V, as one example. This type is termed a *multiplying* converter. Examples of it in i.c. form would be the 1406 and 1408 by Motorola, and the 559 by Analog Devices.

An even more valuable type to us as audio engineers is a *two-quadrant* multiplying converter, which has the capability to accept reference inputs of either polarity, or a.c. With good linearity via the reference input, this device may be used directly to attenuate or scale audio signals by making the audio signal the reference. The output is then simply the audio signal, multiplied by the digital input. Some units which are capable of this type of operation are the 7520, 7521, 7522 and 7530 by Analog Devices, and the 331-8 and 331-10 by Hybrid Systems. Of these, the 7520, the 7530, the 331-8, and the 331-10 are all pin-compatible and functionally similar. It is this type of converter which will be discussed primarily in the applications which follow, since it is inherently applicable to audio use.

AN R-2R LADDER D/A CONVERTER

A functional diagram of the 7520 type of converter is shown in FIGURE 2. This is an R-2R ladder resistor network structure, with the bit switches connecting the 20k ladder termination resistors to either of two output busses. The 1-out-2 bus is normally grounded, so a switch connection to this bus is an *off* condition. The 1-out-1 bus is the active output, which connects to the summing point of an external op amp. Note that for simplicity only four switches and ladder sections are shown; in actuality, the 7520 and 7530 have 10 bits of resolution (10 switches) while the 7521 has 12. The switches used in this family of devices are of cmos construction; this feature provides low distortion operation with a.c. ladder inputs (more on this in a moment).

Even further insight into the operation of this type of d/a converter may be had from FIGURE 3. Here 3(A) shows a simple pot controlling an audio signal, while 3(B) shows a 7520 type d/a controlling a similar signal. Rotation of the pot shaft causes its output to increase (or decrease). Similarly, increasing the digital word input to A1 will increase the output signal, just as was true in the case of the pot. Conversely, decreasing the digital word input will lower the output, analogous to moving the arm of the pot towards ground. A point of difference which should not be overlooked, however, is the fact that the output from the digitally controlled attenuator is from an op amp, and is therefore available at a low impedance and not subject to loading ills, as is the pot. The 7520, as is, does not contain an op amp; it is added externally.

A DIGITALLY CONTROLLED ATTENUATOR

FIGURE 4 is a "for real" circuit showing all the details of how a 7530 (or another of the 7520 family) converter can be used to build a high performance digitally controlled attenuator. This circuit is the practical realization of the block diagram sketch of 3(B).

Here the audio signal to be attenuated is fed into the 7530's ladder at the V_{ref} terminal, pin 15. As discussed above, the state of the 10 internal cmos switches determines how much of the signal appears at the 1-out-1 line, pin 1. The ladder input impedance is nominally 10k, so this circuit does not load a 600 ohm source excessively. Further, the impedance on the line is constant, regardless of the switching state of the d/a, so it cannot reflect back any level changes to the source.

A2 is the op amp used to buffer the d/a output and converts its scaled audio output current into a usable audio voltage. This is accomplished in conjunction with the internal feedback resistor between pins 16 and 1, with pin 16 being the d/a output. In this circuit, A2 is a 301A, an op amp which achieves a 30 mHz bandwidth at audio frequencies, and a 10 V/μsec. slew rate. These factors enable the distortion performance to be superior.

Performance-wise, the circuit is somewhat difficult to assess. Thd could only be measured at an input level of 7V rms (full scale), where it was 0.004 per cent or less below 10 kHz, rising to 0.008 per cent at 20 kHz. At lower levels, the distortion drops into the noise level, which measures about 90 dB below 3V rms out. Intermodulation measurements are similar; at 7V rms equivalent output, i.m. measures 0.008 per cent, and quickly drops down to the residual (0.003 per cent) at lower levels. Neither form of distortion is directly affected by the state of the attenuation set into the d/a. With all bits off, the feedthrough (leakage) in the audio spectrum is quite good; —90 dB at frequencies below 1 kHz, rising to about —70 dB at 10 kHz. The circuit has a max. nominal gain of unity, with 10 bits of control. It can drive loads of 2k or more, and handle signals up to 7V rms (+17 dBV).

SOME GENERAL CONSIDERATIONS

Scrutiny of the behaviour of this digital attenuator reveals some general characteristics which must be considered to apply digital techniques to audio most effectively.

The code table given with the circuit of FIGURE 4 is, of course, not complete; it just represents the major codes and is intended to give a general picture of operation. You can note that for all *1*s, we have a nominal unity gain condition, or zero dB gain. Each time a high order bit (most significant bit and succeeding lower bits) is turned off, the output drops a nominal 6 dB, or one-half that previously. This follows down through the scale and is true for any number of bits (8, 10, 12 or whatever). The attenuation or gain change span is defined by the resolution

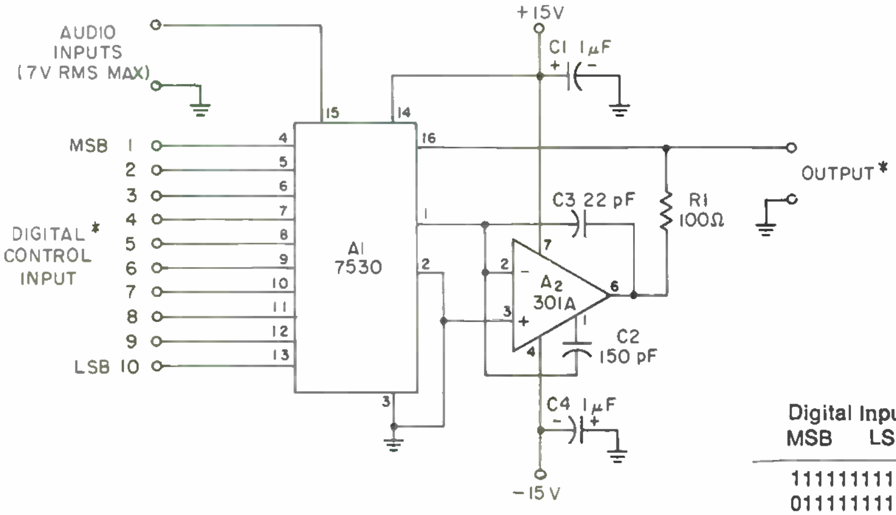


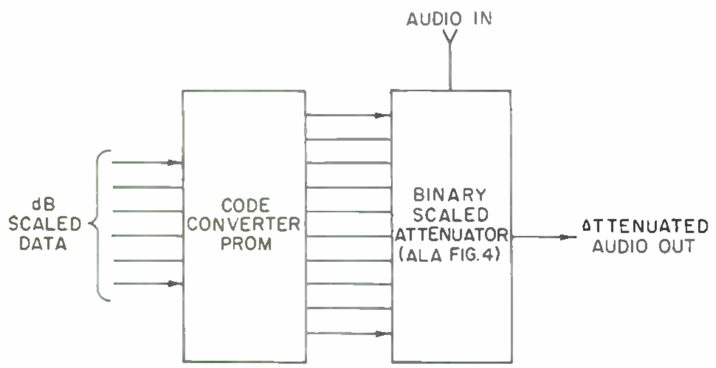
Figure 4. High performance digitally controlled attenuator.

CODE TABLE (10 Bits)

Digital Input MSB LSB	Output Ratio	dB
1111111111	1023/1024	0
0111111111	511/1024	—6.04
0011111111	255/1024	—12.08
0001111111	127/1024	—18.13
0000111111	63/1024	—24.22
0000011111	31/1024	—30.38
0000001111	15/1024	—36.68
0000000111	7/1024	—43.30
0000000011	3/1024	—50.66
0000000001	1/1024	—60.21

of the converter, and is equivalent to $\frac{1}{2^N - 1}$. Here for instance $2^N - 1 = 1023$, and this yields attenuation of 60.2 dB. 8 bits would yield 48.13 dB, 12 bits would yield 72.25 dB, and so on. This basic ratio defines the attenuation capability of a given converter, and, as it is easy to see, has definite limitation for audio use. How this can be circumvented we'll cover in a moment.

A further limitation (insofar as audio use is concerned) is the *linear* nature of the converter's scaling. Binary scaling as shown here is not really what we'd like for audio use, since we'd really like to change in *dB* increments per bit. All converters currently available, regardless of the number of bits, operate with either a straight natural binary or binary coded decimal scaling (to my knowledge). To change the operation so that it may be scaled as x dB per bit requires some additional hardware, and introduces further application considerations.



CODE CONVERTER TRUTH TABLE (6 STEPS)

Prom Input		Attenuator Input	
dB	Digital Word	Number	Digital Word
0	000000	1023	1111111111
1	000001	913	1110010001
2	000010	813	1100101101
3	000011	725	1011010101
4	000100	646	1010000110
5	000101	576	1001000000

Figure 5. A code conversion method of dB attenuation sealing.

CODE CONVERSION

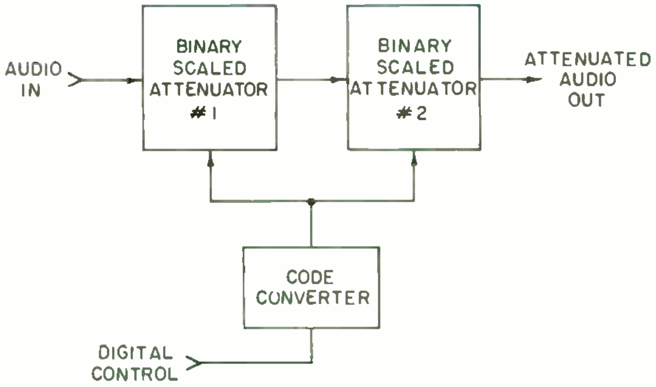
One aproach to this problem is to precede the binary scaled attenuator with a code converter which translates dB weighted data into the appropriate binary data, which then addresses a binary attenuator (such as shown in FIGURE 4). This scheme is shown in FIGURE 5.

The code converter is simply a PROM which has been programmed to deliver the prescribed output binary codes for specific dB increments. Suppose, for instance, we wanted 1 dB increments of change from the FIGURE 4 circuit. 1 dB is a ratio of 0.8913. This ratio would then be used to calculate a truth table, progressing through the entire dynamic range of the binary d/a actually used. For a 10 bit unit this would theoretically yield 60 dB of attenuation (true), but unfortunately the resolution will not allow nice even 1 dB increments as the bottom of the scale is approached. So, fairly large errors develop. Over the upper 30 dB of range, however, resolution is within about 0.25 dB or better. The partial truth table of FIGURE 5 shows the first six steps of the PROM input/output pattern arranged for 1 dB increments of attenuation for each lsb increase in the PROM address.

One thing which may seem apparent from this scheme of things is that it is seemingly wasteful of the d/a capability because it only uses 60 of the 1023 possible states. This is true, if you only look at in that sense. But what the higher resolution converter does buy is greater range and resolution. The range gives a larger active span of dB to work with, the resolution greater dB accuracy over more of that range.

INCREASED DYNAMIC RANGE AND RESOLUTION

A rather simple expedient to extend both the total dynamic range of attenuation and the resolution is just



CODE TABLE—CASCADED STAGES

Digital Input	Output	
MSB LSB	Ratio	dB
1111111111	(1023/1024)²	0
0111111111	(511/1024)²	—12.08
0011111111	(255/1024)²	—24.15
0001111111	(127/1024)²	—36.26
0000111111	(63/1024)²	—48.44
0000011111	(31/1024)²	—60.76
0000001111	(15/1024)²	—73.37
0000000111	(7/1024)²	—86.64
0000000011	(3/1024)²	—101.33
0000000001	(1/1024)²	120.41

Figure 6. Extension of the dynamic range by cascading attenuation stages.

to cascade two attenuators (such as the FIGURE 4 circuit). This scheme is illustrated in block form in FIGURE 6.

The idea is just the same as any other two attenuators in series (such as pads for instance); the total attenuation is simply the sum of the individual dB ratings. Here two 60 dB circuits, cascaded, yield a total attenuation range of 120 dB. The digital control inputs to the two stages are just wired in parallel, bit for bit. This yields a new code table, the major codes of which are shown.

The code table shown is applicable to the two-attenuator inputs, *not* to the code converter. The code converter can now be made to generate even more accurate dB increments, and at the same time smaller ones if desired. With 1 dB step coding, for instance, you can remain within less than ½ dB to about 60 dB down, and within 0.1 dB or less to 30 dB down or more. With the basic idea set forth in FIGURE 5, you can make up new coding to take advantage of this increased capability.

What we have been discussing above has been the *theoretically* achievable accuracy, assuming a perfectly linear converter in both cases. Such is not the case in real life, of course; device imperfections must be lived with. In this case, reality does not appreciably dull the argument. Using the loosest grade part from the 7520/7530 types with only 8 bit linearity (± 0.2 per cent) will degrade the accuracy around 30 dB by only 0.2 dB in the worst case. At 60 dB the worst case error is around 1 dB (this assumes trimming for full scale calibration).

FURTHER CONSIDERATIONS

At this point, one thing which may have occurred to you is that there is one significant factor which the d/a converter brings to audio control—its precision. Indeed, this is perhaps the best that the digital world can provide for us, the precise means of achieving a desired end, and at reasonable cost too.

But it is not all a bed of roses, either. This new means of controlling levels can create fresh problems as it solves longstanding ones. Consider for instance what happens when you suddenly change a digital attenuator's state at a signal peak. You get a fat switching transient, because the digital switch goes in nanoseconds, much faster than even the highest audio period. One means to circumvent this problem is to change gain at a relatively slow rate, over a 50 ms or so period—in essence, to make the switch transition *soft*. This can be accomplished by *ramping* the digital input with small dB increments per ramp step. The sound can then be made to approximately a soft fade.

This installment has begun to explore the world of digitally controlled audio, beginning with the most basic element—the attenuation element. Future installments will delve further into the implications of digital control of audio signals. ■

REFERENCES

1. Jung, Walter G., "Audio Application Ideas for Cmos Dacs," *Analogue Dialogue*, Vol. 9, No. 4.
2. Sheingold, Daniel (ed.), Analog Digital Conversion Handbook, Analog Devices, 1974.

PRODUCT SOURCES

7520, 7521, 7522, 7530
Analog Devices
Box 280
Norwood, Mass. 92062

331-8, 331-10
Hybrid Systems
87 Second Ave.
Burlington, Mass. 01803

FREDERICK E. MAJEWSKI

The Universal Tape Synchronizer

An inexpensive tape synchronizer records a 60 Hz synchronizing signal on one track of the recording tape as a reference for compatible playback.

THE UNIVERSAL TAPE SYNCHRONIZER is an inexpensive, easy-to-duplicate tape synchronizer, designed to work with inexpensive synchronous motor stereo and 4-track tape recorders. It can also be used with more expensive capstan-controlled multi-track machines. It was originally designed to synchronize 4-track stereo playbacks with video tapes for t.v.-f.m. stereo simulcasts, using a Sony TC-654-4.

Due to capstan slippage or tape stretch,* a recorded tape will not necessarily play back at the exact rate at which it was recorded. By recording a 60 Hz synchronizing signal on one track of the tape and using it as a reference to control the capstan speed, you can essentially place electronic sprockets on the tape, and compensate for this problem.

The Universal Tape Synchronizer can use the 60 Hz a.c. line, an external 60 Hz crystal standard, or a color t.v. 59.94 Hz vertical drive signal as its reference. It can also process these signals for recording on the tape as a synchronizing signal. A manual speed control aids in initial synchronization to another device.

The outputs are a 60 Hz square wave or a 60 Hz sine wave to drive a capstan power amplifier and a d.c. control voltage that can be set to any offset between +5 and −5 volts. A companion 50W power amplifier is recommended. It has sufficient output to drive most inexpensive synchronous capstan motors. If high power is needed, there are many commercial capstan drivers available.

Unlike some synchronizers, in the absence of a sync signal, the output switches to the 60 Hz center frequency or 0 V d.c. control voltage to maintain nominal tape speed. The Universal Tape Synchronizer can accept any sync signal from 200 mV to 10 Vp-p, square wave, sine wave, or anything in between. Its input filter will filter out transients and even separate sync signal from audio on the same track.

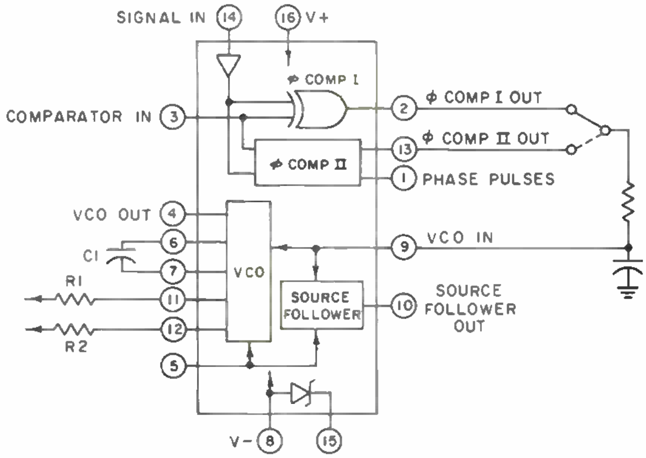


Figure 1. Cos/mos phase-locked loop.

THEORY OF OPERATION

The heart of the synchronizer is the 4046 cos/mos phase locked loop i.c.-3. (See FIGURE 1) This 16-pin i.c. contains two phase comparators, a voltage controlled oscillator (vco) and a source follower. The center frequency of the vco is set by a single resistor and capacitor pair. (R1-D1 FIGURE 1) However, unlike most other phase-locked loops, the range of the vco about the center frequency can also be set to desired limits by R2. Phase comparator 1 is an exclusive *or* gate that requires symmetrical signal and reference inputs. Phase comparator 2 is edged-clocked and

Frederick E. Majewski is the assistant chief engineer at WHYY-TV, WUHY-FM in Lansdowne, Pa.

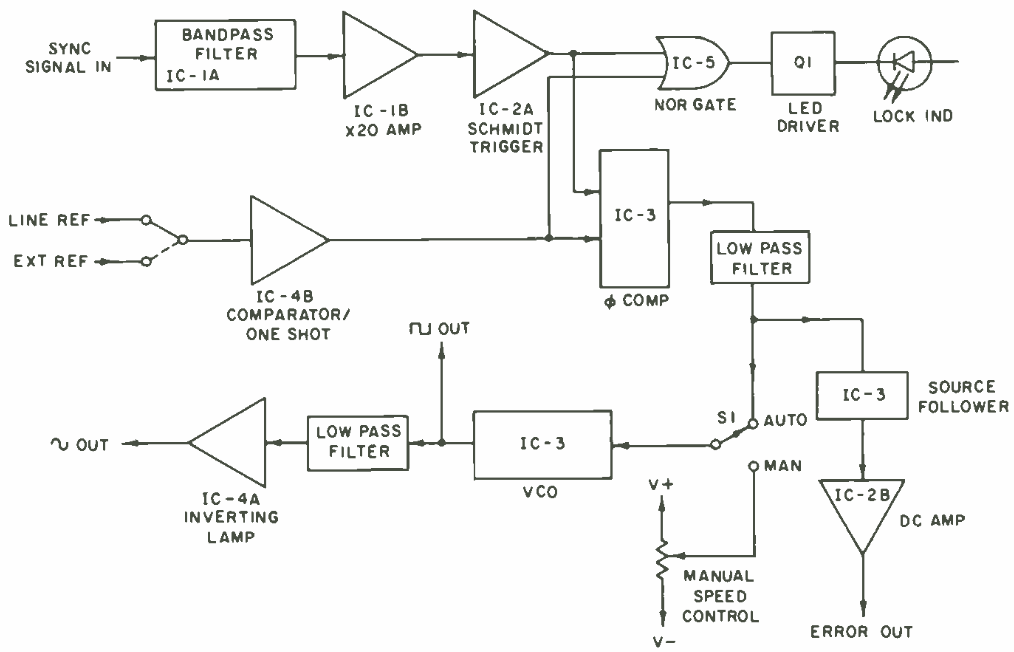
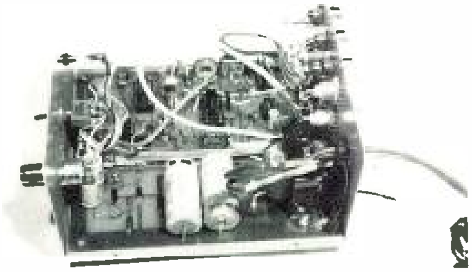


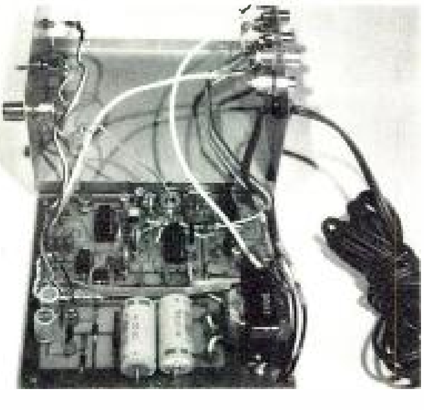
Figure 2. Block diagram.

Tape Sync. and capstan power amp (front view)

Tape sync and amp (rear view).



Tape sync (internal).



Internal with circuit board removed.

does not require symmetrical inputs. It has the added advantage of not locking to harmonics of the input signal. However, phase comparator 2 has two advantages that are important to the Universal Tape Synchronizer. It has high noise immunity and returns the vco to the center frequency in the absence of a lock signal.

The 60 Hz sync signal from the tape first passes through a bandpass multiple feedback filter (i.c.-1A Figures 2 & 3). Its gain, determined by R1 and R4, is 0.9; and its 3 dB response, determined by R1, R3 is from 50 Hz to 70 Hz.

The sync signal is a.c. coupled by C3, to eliminate the effects of d.c. offset, to inverting amplifier i.c.-1B. It has a gain of 20 determined by R6 and R8. I.c.-2A is a comparator with part of its output fed back to the non-inverting input (through R11, R10). This produces a hysteresis, or "snap action," to create a fast rise square wave from the slow rise 60 Hz sine wave. The gains of i.c.-1A and i.c.-1B are such that a good square wave is produced from a weak sync signal but not from noise or stray a.c. pickup when no sync signal is present.

I.c.-4B is used as a differential comparator. When the jumpers (L1, L2) are in the line position (L), it produces a 60 Hz square wave from the secondary of T1

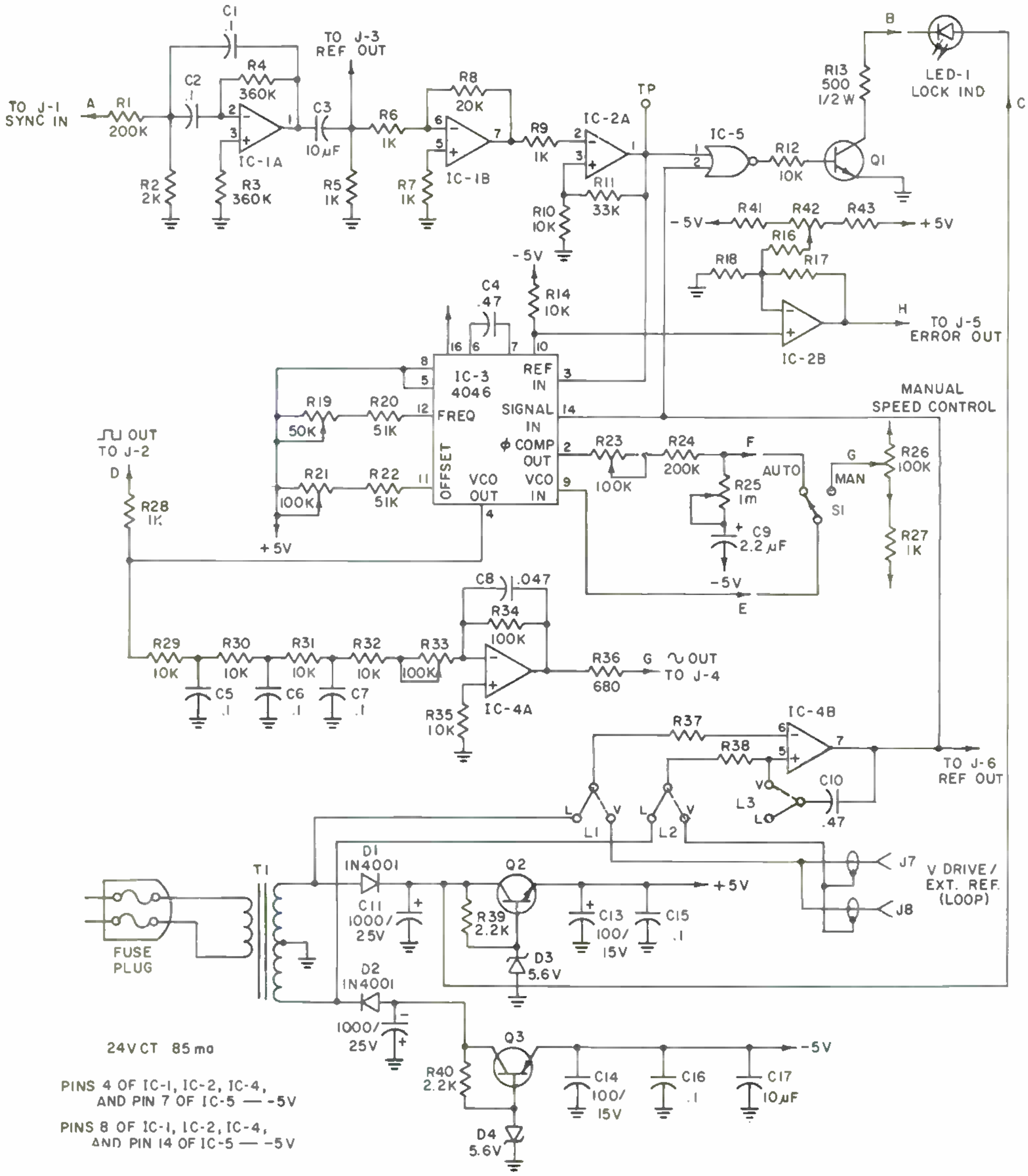


Figure 3. Universal Tape Synchronizer schematic.

(a.c. line reference). If an external 60 Hz timebase is desired as a reference, it can be fed to J7 or J8 and the jumpers moved to the V position. When vertical drive is used as a reference, L3 is also moved to the V position. Positive feedback through C-10 turns i.c.-4B into a one-shot to produce a symmetrical square wave from the short vertical drive pulse. The pulse width is determined by the combination of C10, R38.

The reference square wave is applied to the signal input of the pll (i.c.-3) and the processed sync square wave to the comparator input. They are compared in the phase comparator, which produces a 120 Hz 50 per cent duty cycle output when the two signals are 90 degrees out of phase. This is the lock condition for the pll. The phase comparator output (pin 2) passes through a low pass filter, made up of R23, R24, R25 and C9. Standard pll low pass filter designs are useless in this application because of a mechanical device (the tape recorder) in the feedback path. The values were obtained by trial and error and required no adjustment between the two tape machines tested. R23 controls the gain of the filter (lock in time) and R25 controls the damping (overshoot).

The frequency of the vco section of i.c.-3 is controlled by R19, R20, C4 and the voltage applied to pin 9. R19 sets the center frequency of the vco, which should be 60 Hz with 0 V on pin 9. R21 sets the offset, or vco range, which should be from 50 Hz to 70 Hz.

The vco square wave output is current limited by R28 and appears on J2 for external use. It also passes through a low pass filter (R29, R30, R31, C5, C6, C7) and is amplified by inverting amplifier i.c.-4A. C8 helps integrate the waveform to make it resemble a sine wave. R33 controls i.c.-4A's gain and should be set so that the positive and negative peaks are just clipped. The output at J6 goes to the power amp (FIGURE 4) where it is converted to 110V a.c. 50-70 Hz to drive the capstan motor.

The processed synchronizing signal from i.c.-2A and the reference signal from i.c.-4B are compared in *nor* gate

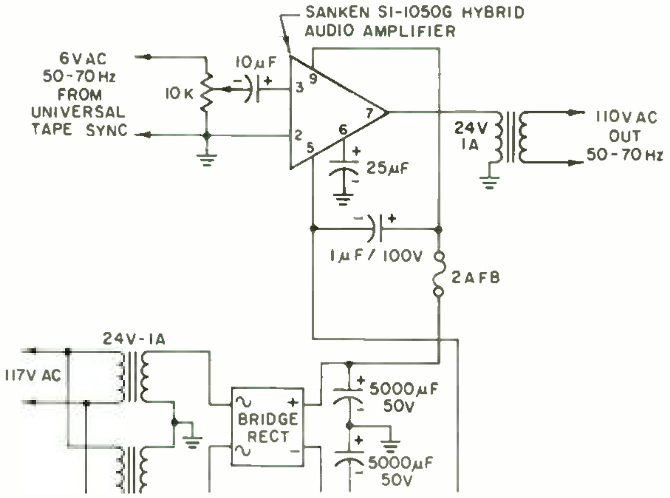


Figure 4. Capstan power amplifier.

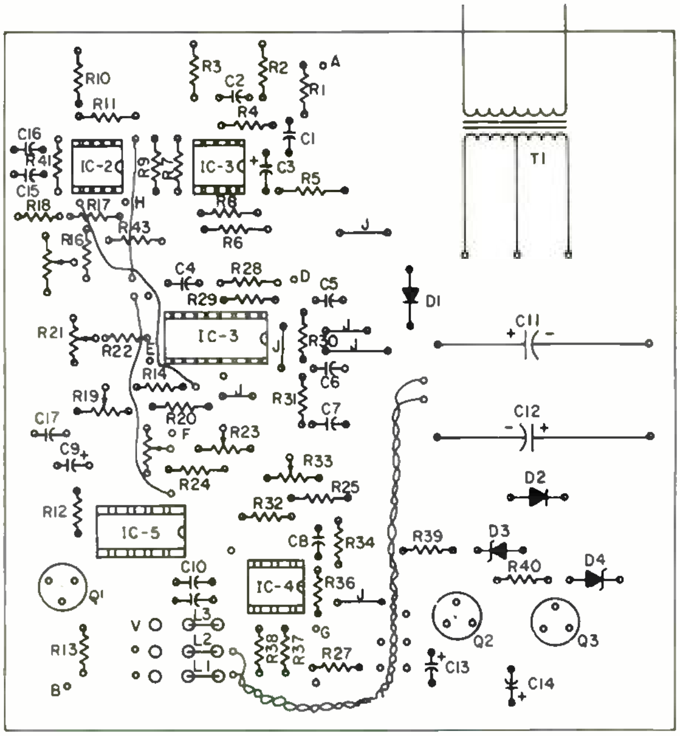


Figure 5. Component layout.

i.c.-5. When the sync signal and reference are locked (90 degrees out of phase) the output at pin 3 of i.c.-5 is a 60 Hz 25 per cent positive duty cycle waveform. This drives Q1 and the led *lock* indicator. If the synchronizer loses lock, the lock indicator will flash once for each cycle (16.6 ms) of slippage.

The d.c. vco input is internally coupled to a source follower contained in i.c.-3. R-14 is its load resistor. This control voltage is amplified by i.c.-2B and appears at J5. The gain of i.c.-2B is set by R18 and its d.c. offset is set by R42.

OPERATION AND ADJUSTMENT

With S-1 in the manual position and the manual speed control (R26) centered, the vco should be adjusted to 60 Hz with R19. R21 should be set so the manual control has a range from 50 to 70 Hz at its extremes. The two adjustments interact so they will have to be done several times, until they are correct.

A 60 Hz square wave reference signal for recording is available at J6. The reference will be the a.c. line or vertical drive or other external input depending on the positions of L1, L2, and L3. A square wave was found adequate for a synchronizing reference, but if a sine wave reference is desired, simply patch the output of J6 to J1 and use the output of the filter (J3) as the reference.

If vertical drive is used as a reference, R38 may have to be trimmed, due to tolerances in C10 and i.c.-4. It should be adjusted for a symmetrical square wave at pin 7 of i.c.-4.

For playback, the synchronizing signal is fed to J1. J6 feeds the power amplifier which drives the capstan motor. If the synchronizer is used with a d.c. servo capstan, the control voltage is taken from J5. Since different machines require different control voltage ranges, the d.c. offset of the error signal is made adjustable by R42. With the manual control at the center of its range, R42 should be set for the nominal voltage required by your particular tape recorder.

With the synchronizer operating in the manual mode, minimum speed, set S1 to the auto mode. The recorder should lock up within one second with no oscillating or overshoot. If not, adjust R23 (gain) and R25 (damping). R23 will control the lock up rate, and R25 will control the overshoot. With some d.c. capstan-controlled recorders, depending on the servo response, it may be necessary to increase the value of C9. ■

PARTS LIST

IC-1, IC-2, IC-4—LM 1458 dual op-amp
IC-3—CD 4046 cos/mos phase locked loop
IC-5—CO 4001 cmos nand gate
C1, C2, C5, C6, C7, C15, C16—0.1 µF 50V disc ceramic
C3, C17—10 µF 15V electrolytic
C4—0.47 µF 50V disc ceramic
C8—0.047 µF 50V disc ceramic
C9—2.2 µF 15V electrolytic
C10—0.47 µF 50V disc ceramic
C11, C12—1,000 µF 25V electrolytic
C13, C14—100 µF 15V electrolytic
D1, D2—IN 4001
D3, D4—5.64 Zener Hep Z0407
Q1, Q2—2N2219
Q3—2N2905
R1, R24—200k, ¼W 5%
R2—2k ¼W 5%
R5, R6, R7, R9, R27—1k ¼W 5%
R8—20k ¼W 5%
R10, R12, R14, R16, R17, R18, R28, R29, R30, R31, R32, R35, R37, R38—10k ¼W 5%
R11—39k ¼W 5%
R13—510 ohm ¼W 5%
R19—50k trimpot
R20, R22—51k ¼W 5%
R21, R23, R33—100k trimpot
R25—1 meg. trimpot
R26, R34—100k ¼W 5%
R36—680 ohm ¼W 5%
R39, R40—2.2k ¼W 5%
TI—117V Primar 24 VCT secondary 85mA. Stancor P-8394.

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CORRECTIONS

The following errors have been noted in SIGNAL PATH II, SINE WAVE OSCILLATORS, by Walter G. Jung, in the July issue. Kindly note these corrections.

1. Figure 3: Reverse either D1 or D2; correct connection is inverse parallel.
2. Figure 4: C3 value is 0.1 μF.
3. Figure 5: Reverse either D1 or D2; correct connection is back-to-back (as in Figure 6).
4. Figure 7: There should be a connection shown between A1's output and the junction of R3-R1.
5. Figure 8: There should be a connection shown between A1's output and the junction of R3-R1. Also, Q1 is an N channel device.
6. P 39, eighth line from bottom, right column: "Thermal conductivity" should read 'temperature coefficient."

Copies of all issues of **db—The Sound Engineering Magazine** starting with the November 1967 issue are now available on 35 mm. microfilm. For further information or to place your order please write directly to: **University Microfilm, Inc.
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• A new position, director of corporate planning, at GenRad, Inc. of Concord, Mass. has been filled through the promotion of **J. Robert Held,** formerly vice president for finance at the company's Environmedics Division.

• **Bob Patterson** has been appointed eastern regional sales manager for **Hy-Gain Electronics Corporation,** of Lincoln, Nebraska. Mr. Patterson's territory will include eastern Pennsylvania, the New England states, New York City, and Washington, D.C. His headquarters will be at Andover, Mass.

• **RCA** broadcast equipment played an important part in the summer Olympic Games in Montreal. Sixteen units of their TR-600 video tape recorder were used by member nations to aid in beaming coverage of 21 different sports to 70 countries throughout the world. 3.500 "Color-Trak" RCA t.v. sets formed a closed-circuit t.v. system that permitted viewers to monitor up to ten different events as they happened. Representing a value of $1.5 million. the receivers were the largest number of t.v. sets ever assembled for one purpose.

• **Audio Plus,** of Great Neck, N.Y. has been appointed to represent the **Nortronics Recorder Care Division** in Metropolitan New York, including New York City, Nassau, Suffolk, Westchester, and Rockland Counties, as well as northern New Jersey. **William P. Kist** and **Steve Weill** of Audio Plus will provide technical and sales assistance to local dealers.

• The appointment of **Harry N. Larkin** to the post of marketing director has been announced by **LPB, Inc.** of Ambler, Pa. Before joining LPB, Mr. Larkin was with the **Ampro Corp.**

• New product design and technical evaluation for **Radio Shack,** of Fort Worth, Texas, will be under the supervision of **Alfred M. Zuckerman** in his new capacity as chief engineer. Before joining Radio Shack, Mr. Zuckerman was with the **Benjamin Electronic Sound Company.**

• A new slate of officers was elected at the June 5 meeting of the **Intercollegiate Broadcasting System's** Board of Directors. **Jeff Tellis,** general manager of **WPKN-FM** at the university of Bridgeport, was reelected president. Chairman of the Board is **Donald A. Grant,** with **Dr. George Abraham and David W Borst** vice-chairmen for east and west. **Herbert B. Barlow, Jr.** was elected secretary, **Fritz Kass,** treasurer, and **Rod Collins,** vice president for programming. Station relations directors are **Dick Gelgauda** and **Norm Prusslin,** and **Rick Askoff** was re-elected as executive director for the system. Other board members include **Patricia Montieth, Paul Brown, Ludwell Sibley,** and **Jim Cameron.**

• An association for the development of and manufacturing audio and related equipment has been formed between quad patent holder **Peter Scheiber** and **Al Chesrow,** president of **Deltek, Inc.,** of Bloomington, Ill. under the auspices of Deltek. The first product will be a decoder for recovery of 360 degree positional information from SQ-coded and standard stereo sources.

• **Marketing Plus.** of South Edina, Minn, a newly formed sales rep firm, will represent **Koss Corporation** products in Minnesota and North and South Dakota. Marketing Plus is headed by **Joseph Purtell** and **Boyd Lester.**

• Products from **Rupert Neve and Company, Ltd.** of London, England have radiated out to many parts of the world recently. Orders have been received for consoles from the **Australian Broadcasting Commission, Queensland Television** in Brisbane, Australia, **Radio Television Singapore,** the **European Parliament** in Luxembourg, Swedish recording company, **Platina,** and **Industrias Electricas y Musicales Peuranas S.A.** in Lima, Peru.

• **TAPCO Corporation** of Edmonds, Washington, has announced the appointment of **Charlie Kester** as communications manager. Mr. Kester will be in charge of corporate communications, advertising, and sales training. He comes to TAPCO from the **Acoustic Control Corporation.**

• **Earl F. Keaton** has been elected president and general manager of **Gulf Telephone and Electronics, Inc.** of Houston, Texas. Mr. Keaton has been a long-time associate in the **Dynalectron Corporation,** Gulf's parent company. He has also been connected with the **NASA Goddard Space Flight Center,** the **Polaroid Company** and has been a principal in his own company, **Pathfinder Services, Inc.**

• A midwest facility employing about 100 people is under construction by the **Sony Corporation** at the Air World Center in Kansas City, Mo. The facility, which will have 200,000 square feet on a site of 13 acres, will house warehousing and distributing functions and a test center. Completion is scheduled for 1977.

• **Sela,** manufacturers of portable sound mixers. has appointed **Audio Services Co.,** 565 Fifth Ave., New York, N.Y. as their sole U.S. distributor. Audio Services will also handle warranty service.

• An extensive complex. known as **La Voix du Zaire** is being constructed by **Compteurs Schlumberger Sodeteg,** and **Thomson-CSF** at Kinshasa, Zaire. The complex will house the functions of national radio, television, film, and the official Zaire Press Agency.

CORRECTION: On the **People, Places & Happenings** page of our June issue, **Mr. George Gaal** was incorrectly listed as a principal of **Professional Audio Systems Engineering, Inc.** Mr. Gaal is not associated with that firm.

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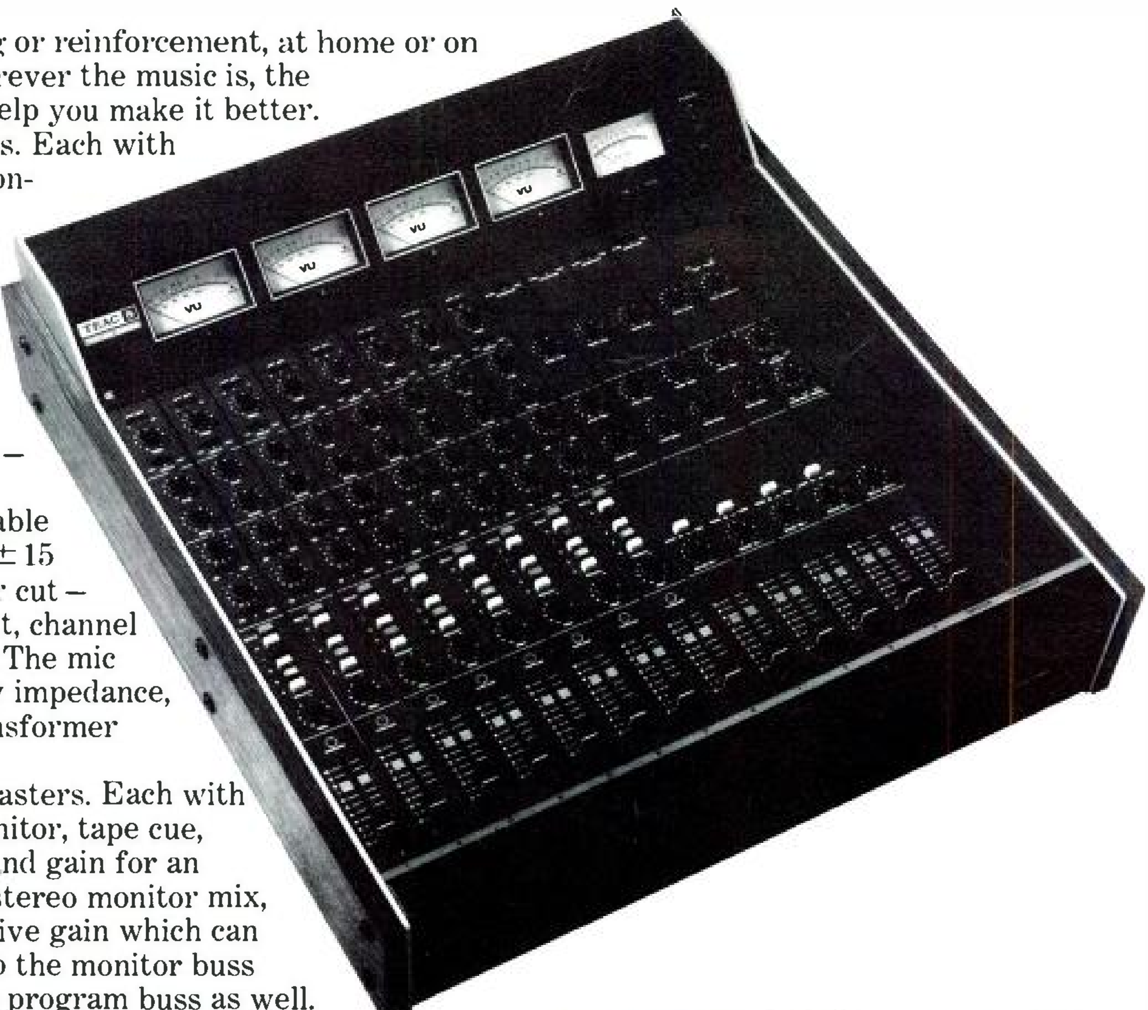
less than $1500.00*



For recording or reinforcement, at home or on the road, wherever the music is, the Model 5 can help you make it better.

Eight inputs. Each with attenuation controls, LED overload indication, two foldback sends – one before and one after the EQ and fader – two bands of EQ – 4 selectable frequencies, ±15 dB of boost or cut – solo, direct out, channel assign & pan. The mic inputs are low impedance, balanced, transformer isolated.

Four Submasters. Each with buss/tape monitor, tape cue, monitor pan and gain for an independent stereo monitor mix, and echo receive gain which can be assigned to the monitor buss only, or to the program buss as well.

Multiple patch points. 3 provisions for headphone feeds; switchable output levels; accessory send/receive on each input; buss in on each output; plus cascade facilities for interconnecting two Model 5's.

Fully modular, portable, and quiet (the equivalent input noise is 125 dB). So if you have more talent than money, check out the Model 5 at your nearest TEAC Tascam Series dealer. Just call toll free (800) 447-4700** for the name and location of the one nearest you.
**In Illinois, call (800) 322-4400.

*Actual resale prices are determined individually and at the sole discretion of authorized TEAC Tascam Series dealers.

TEAC®
TASCAM SERIES

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