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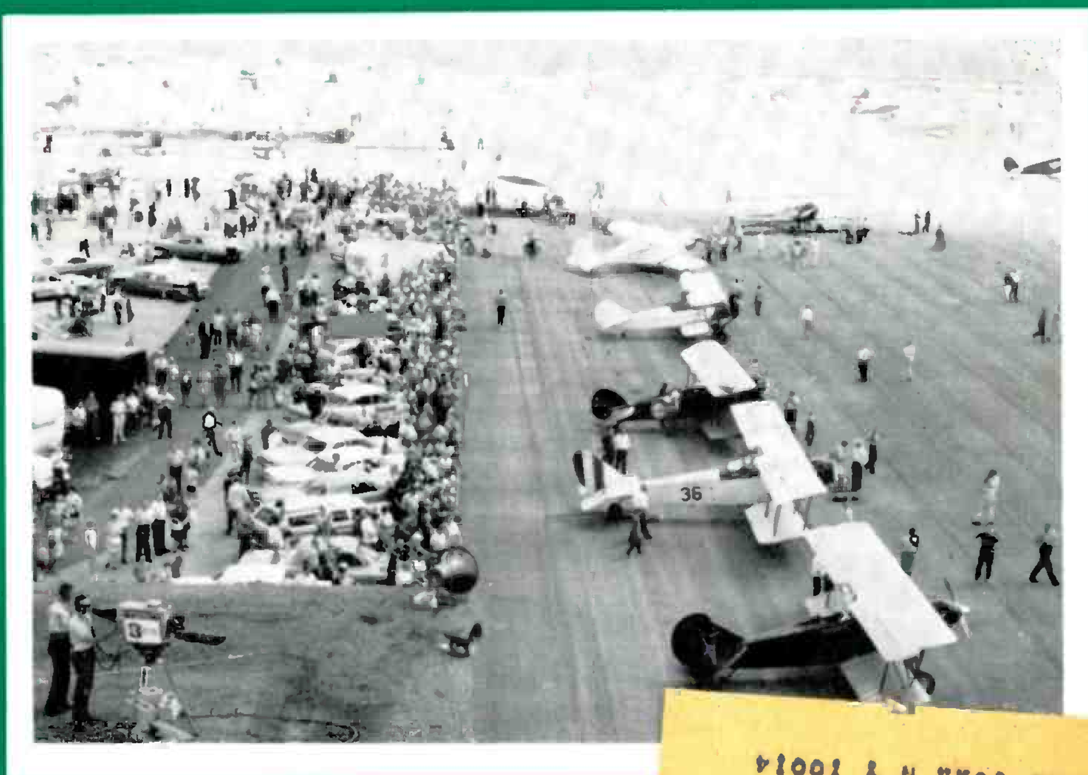
THE SOUND ENGINEERING MAGAZINE

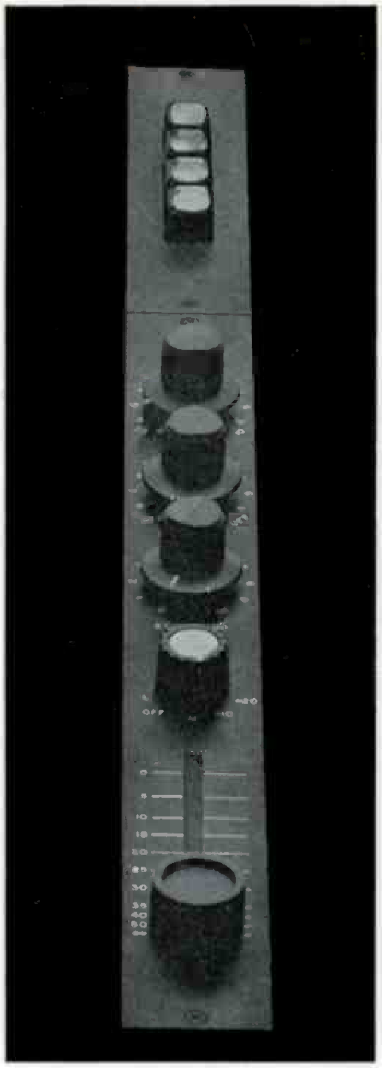
FEBRUARY 1969 75c

Stereo Microphone Techniques

A Super-Power P.A. System

Improved Studio Monitoring





Coming Next Month

Charles E. Wiegand, Jr. of Wiegand Audio Labs has prepared a story on his console design that was pictured on the cover of our December 1968 issue.

The author will detail the system and its functions as well as explaining the practical need for this complexity.

Improved Monitoring with Headphones, part 2 by Howard Souther reveals the decision to seek a solution to headphone quality through the use of electrostatic elements. The search that resulted in a finished product using such elements without the need for an independent power supply is a fascinating story.

In The Design of a Sequencer, Robert C. Ehle explains the circuitry employed in part of the music synthesizer he described in his November 1968 article, The Design of a Synthesizer.

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

About the Cover

You can see, in the lower left, next to the t.v. camera, two horns, one large and one smaller, that form part of the coverage of an air show. Elliott D. Full describes the super-power amplification system and the coverage in his article on page 30.

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THE SOUND ENGINEERING MAGAZINE

FEBRUARY 1969 • Volume 3, Number 2

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MONTHLY DEPARTMENTS

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Letters

The Editor:
John McCulloch's article regarding disc cutting was way overdue. It has been my experience that far too many "recording engineers" know little or nothing about the mechanics of disc mastering. This is particularly true of these engineers who have migrated from the broadcast industry, and the younger generation of audio engineers in general.

This is a sad state of affairs. Let's face it, the greater part of the product of the recording industry winds up on disc. Yet I must turn down numerous job applicants, who otherwise might have had a good chance with us, simply because of a total lack of disc experience. Somehow engineers seem to be under the impression that once they get the sound they want on tape, the rest will take care of itself.

I can not stress too strongly how much a background in disc cutting will change a mixer's outlook or approach to recording, for he begins to record on tape with an understanding and recognition of the problems and limitations of the disc medium. Many engineers today feel that if they get it on tape, someone should be able to get it on disc. Unfortunately, this is not necessarily true, many times the over-all sound must be sacrificed to get it on disc when a little attention during combining or live recording could have circumvented the problem.

Jerrold J. Ferree
Chief Engineer
United Recording Corp.
Hollywood, Calif.

The Editor:
The editorial of December 1968 strikes a familiar note. We have been working with officials of the forthcoming Providence Model Cities Program to institute a program.

Our plan will hopefully (pending, of course, federal approval) include speech training for a limited number of youngsters at classrooms located in the target area, plus transportation to our suburban studios for several hours training per week in tape handling, recording, and for more promising students, facets of broadcasting that would produce fledgling announcers and newsmen.

We plan to make use of a fully-equipped production room and perhaps the

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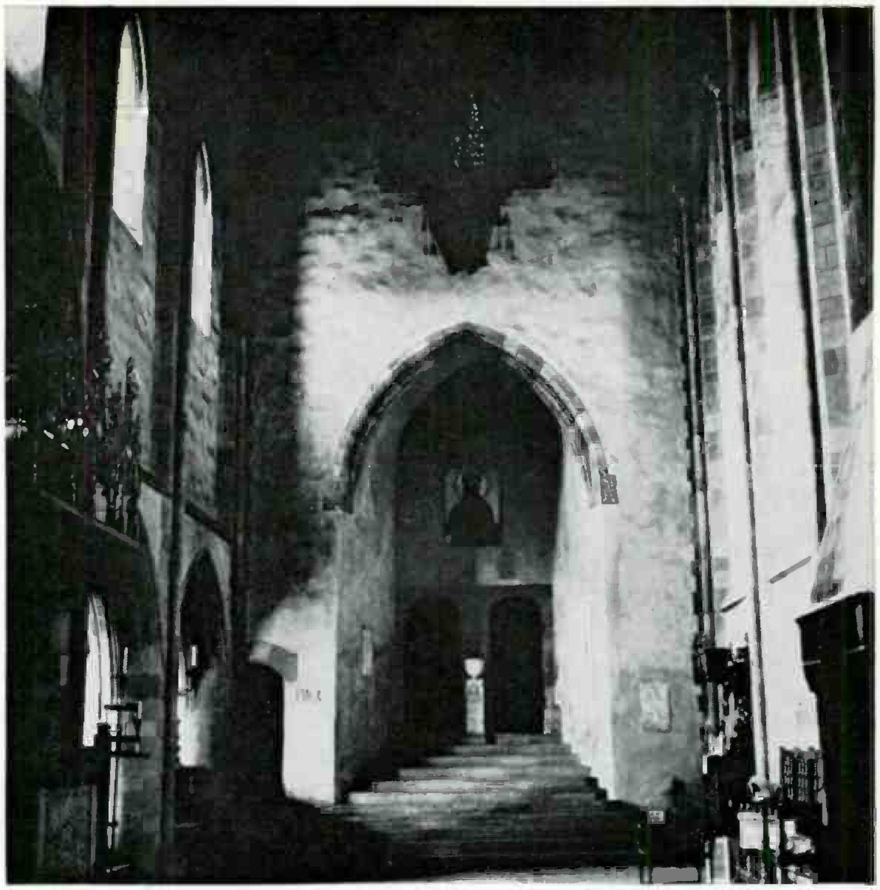

The great hall of the Hammond Museum. This room is the location of the organ played by Richard Elsasser on Nonesuch H-71200 ("Yankee Organ Music") and H-71210 (Organ Symphony No. 5 by Charles-Marie Widor)

AR-3a speaker systems were designed for home music reproduction. Nonesuch Records uses them as monitors at recording sessions.

Nonesuch Records recently recorded several volumes of organ music played by Richard Elsasser at the historic Hammond Museum near Gloucester, Massachusetts. To make the recording, Marc Aubort of Elite Recordings, engineering and musical supervisor, used Schoeps microphones, and Ampex 351 recorder, Dolby A301 Audio Noise Reduction apparatus, and several pieces of equipment which were custom made. To monitor the input signal and to play back the master tape, Aubort used an AR amplifier and 2 AR-3a speaker systems.

The AR-3a speaker system is priced from $225 to $250, depending on finish.

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entire newsroom during later evening hours. Our scheme was born at a meeting between Frank Pierce of the Providence Poverty Program and myself at a news-media meeting in late November.

Les Brown
Program Director
WXTR
Providence, R.I.

The Editor:

I've been following the LETTERS with great interest. . .particularly the recent series on television audio. Mr. Haskett's note leaves me speechless!

Both television and radio network audio is sent by single-side-band over 6 kHz-equalized lines, and the transmitted and received quality far surpasses flat to 15 kHz response! The problem with t.v. audio is the same problem that most "tea-pot" stations suffer from; too little attention to quality response from all sections in the audio chain at the station.

The F.C.C. requires that audio adhere to the rules through the main mic channel. . . .and that's about where most quality-control of audio ceases. In my eleven years in audio and radio I have met only three engineers who pursued their audio quality past the main mic channel. Most have indicated that since the board meets the proof specs, the audio is fine. I worked at a station in 1964 where the mic channels were superb, but where the turntables went up to 5 kHz and the tapes only up to 4500 Hz! Most of the problem was in equalization or mis-matched components.

I strongly suspect that very few t.v. stations, much less radio stations, have checked to see if over-all response through each component in their audio system is as good as their main mic channel. If engineers paid more attention to over-all quality instead of required quality I'm sure the t.v. audio question would be settled very quickly.

John E. Thayer
Chief Engineer
KIKX-AM
Tucson, Arizona

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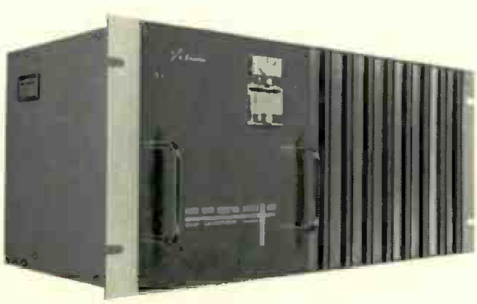
Have you sent us a change-of-address notice? It takes time for us to change your plate so let us know well in advance of your move. Be sure to send us the complete new address *as well as your old address*. Include both zip numbers. Keep db coming without interruption!



The Audio Engineer's Handbook

GEORGE ALEXANDROVICH

● When building a system, every step of the way has to be planned and carefully thought out. You must know what you want the system to do, how well, and under what circumstances. You cannot plan on feeding a microphone pre-amplifier a signal of −80 dBm and get 65dB s/n fort he entire system. The most you can hope for would be 45 or 50 dB, and this with the best equipment. Obviously, dB's do not come that easy when measuring noise.

Aside from planning a level distribution in your system, it is important that the coupling between the individual system components, stages, or sections does not deteriorate the performance of this system. Methods of interconnecting components and system wiring is the subject of this month's discussion.

The equipment selected for the system or a console may consist of a variety of makes and brands. There may be tube equipment along with transistorized gear. There may be both high- and low-impedance devices which have to be interconnected and used in the same circuit. There may be equipment using integrated circuits and modern operational amplifiers. Still, with all the variety of equipment one has to work with, there is the same set of basic ground rules to follow for composing a workable and operating system.

There are basically two approaches to console wiring using shielded wire. One is by using wire with exposed shield (braided), tied into one common harness or cable. In this way all shields are tied together at a multitude of points making numerous contacts to chassis and frame. There is no distinction between the beginning or the end of the shield. This system is practical only with braided shields and is being used in a number of consoles manufactured for broadcast and audio applications. One disadvantage of this system is that all the ground contacts made to the chassis are not always solid enough; sometimes they produce noises, and in strong r.f. fields may cause signal pickup by individual conductors. However an advantage of this system is that it does not require careful planning of grounding techniques.

The second method depends on the use of wire using insulated shields similar to Belden wire 8451. Double-conducted shielded wire is recommended for use throughout where shielded wire is required. An advantage of this method of using insulated shields is that the possibility of picking up hum or stray r.f. fields is minimized. In audio work in particular, where most modern equipment can give us s/n ratios of 65 to 80 dB, it makes sense to employ the best known methods of eliminating the possibility of the system being affected by external interference.

A set of rules on the connection of shields has evolved from the experience of many audio engineers, system designers, and installers. By following these rules, one can eliminate practically all problems of shielding and ground loops resulting from improper shielding. And I mean problems caused *only* by shielding. Improper layout of the signal wire can lead to other troubles which will be discussed later.

1. *All shields should be grounded at one end only.*
2. *Shields should never be used as a conductor for audio signals or power feed (except where there is a separate power supply for condenser microphones). Any time there is a current flow through the shield, there is nothing to prevent the field generated by this current flow to be induced into the signal wires.*
3. *Shields covering input wires to the amplifier should be grounded at the amplifier input.*
4. *Shields should never be left floating (ungrounded).*
5. *In balanced lines, shields should be connected to chassis ground or the center tab of a transformer if it is grounded.*
6. *The following wires should be shielded:*
   - *A. All mic cables.*
   - *B. All interconnecting wires in the system carrying line level signals or lower. (All wires 1 inch or longer for mic levels, 4-5 inches or longer for line levels).*
   - *C. Wires from test oscillator.*

7. *Wires that should not be shielded:*
   A. *Speaker wires.*
   B. *Power wires.*
   C. *Some control wires if current that passes through them does not have high-level transients.*

It should be remembered that excessive shielding can sometimes cause trouble. Shields have a certain capacity to the wires they protect. If shields from two separate lines are tied together, they may produce a crosstalk between the two lines, especially if grounding of the shields is done improperly. In general, however, excessive shielding cannot harm the performance of the system if it is done properly but it will add unnecessarily to the expense of construction. Since all shields are tied to a frame or chassis ground, connection of the signal wires to the ground should be done with discretion.

Let us set up a few basic rules for connecting signal and power wires. In most modern transistorized equipment, signal ground and power ground may turn out to be the same wire. If an active component such as an amplifier is not transformer terminated, the signal ground may be positive or negative. You cannot intermix two kinds of equipment in one system unless separate power supplies are used. Using isolation transformers on the input and the output of each amplifier or device would also solve the problem.

In interconnecting system components, power wires are normally installed first. It is strongly suggested that, in order to eliminate any possibility of motorboating or oscillations of separate active system components, separate wires be used from the power supply to these devices. It is even more convenient to use power supplies with the remote sensing. It means extending the lowest impedance point of the power sources right next to the load.

If there are any amplifiers involved in the system, and individual power lines are not practical for every amplifier, then separate wires should be used to power separate groups of amplifiers of two to three units per group. Using heavier wires in this case would improve the stability and lower the impedance of the power supply. Although many amplifiers do have decoupling filters in their power inputs, motorboating may occur at the cut-off frequency of the filter. That is the main reason for following precautions in wiring. Not many good amplifiers today are Class A; most are Class AB. Thus, when there is no signal fed through the amplifier, current consumption is very low. With an increase of signal power, drain increases — creating variable voltage drop across the power wires. Since more than one amplifier or device is connected across the power lines, this effectively modulates the current to each one of the connected devices. This increases the crosstalk between channels. Decoupling at each amplifier and low impedance of the power supply in conjunction with proper wiring techniques are good assurances that leakage between channels will be low.

When all power lines are installed the signal cables are wired in. Use double-conductor shielded wires. In choosing the wire you should be guided by the performance, specification, and conditions pertinent to this installation. For instance, different shields have different shielding capabilities. Braided shields may be 70 to 85 per cent effective with no appreciable shield inductance. There are wire-wraparound shields that (aside from offering imperfect shielding) have inductance which may act as an antenna for pickup of r.f. signals. 100 per cent shielding is offered by foil-shielded cables but these cables can be used only in immovable interconnections, never as a microphone lead-in wire. These cables, when disturbed, are microphonic and produce crackling noises that are non-compatible with the strict requirements for any serious audio work. For harnesses, foil-shielded cables are ideal. Low-capacity braided shields are recommended for mic cables.

If the impedances of the components to be interconnected differ, especially, if one source is high or even medium, long shielded wires are to be avoided. Even the capacity of a three-foot long wire (with approximately 15 to 20 PF per ft.) will tend to roll off the response of the 50 k ohm circuit at 20 kHz. The longer the cable the higher the impedance and the more pronounced the frequency roll off will be.

1. *When interconnecting parts of the system always carry signal ground along with the hot wire.*
2. *Always use shortest route between components (within reason).*
3. *Do not forget to leave enough slack for access to the equipment for servicing.*
4. *Do not run signal wires too close to power transformer — even shielded wires can pick up hum.*
5. *Do not lace signal wires along with supply lines.*
6. *Do not connect to the chassis ground or any other ground signal wires. First, finish wiring the whole system, then by experimentation find the point which will give the lowest noise. Most likely, this point would be in the connections between the power-supply ground wire and the point where the signal ground of the amplifier is connected to the shield. This is because grounding the power supply wire will isolate any currents induced into the ground wire of the power supply from the signal part of the system.*

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The Feedback Loop

ARNOLD SCHWARTZ

• The modern radio network center has evolved from an originator of studio programs into a complex communication center. A central network facility, with rare exceptions, no longer originates music or entertainment programs from its studios; local stations have taken over this function. The network does provide to the local stations news, sports, and special features. This is a function that only a large and well-staffed organization with diversified talents can competently handle. Because of the extensive coverage of events, information is constantly pouring into this central facility. Special events such as an election or space shot result in an even greater flow of information than is normally present. This information must be recorded, processed, edited, and utilized in some fashion in the preparation of the outgoing feeds to the local stations. Internally generated feeds must also be distributed throughout the network facility. The communication system that handles this information flow consists of a complex of electronic equipment and, more importantly, of a large and hand working staff. To fulfill his function, each staff member must have convenient and rapid access to all or part of the incoming, internal, and outgoing communication lines.

The American Broadcasting Company in planning their new radio network facility, which was put into operation in New York City last April, was faced with the problem of designing such a communication system. The design had to solve current operating problems and also permit room for the inevitable expansion to come in the years ahead. Probably the most important aspect of the system design concerned the engineers who operate the equipment in the various control rooms. A major problem facing the control room engineer is that he may have as many as one hundred different feeds available at eight or ten different locations within his technical area. In a typical control room there are console remote input channels, cartridge record channels, and reel-to-reel tape recorders — each having fifty or more feeds available. The dimensions of this problem can be appreciated by realizing that there are twelve major technical areas in the ABC Radio Network.

The requirements of the ABC system for selecting feeds are; rapid selection, convenience, and freedom from ambiguity. The hardware must be relatively compact and allows for maximum independence of the control room. Each control room should be able to operate with direct access to the available feeds without constant recourse to the master control room. This allows faster and simpler operation, and also reduces the communication burden within the network facility. The use of jack fields was rejected as being too slow, they cannot be located so that they are equally convenient from all operating points in the control room, they are not compact, and they require that the master control operator be constantly involved in the selection of feeds to the studios. The responsibility for the design and construction of the new facility was in the hands of the ABC *facilities engineering group* and the *network operations group.* McCurdy Radio was awarded the contract to supply the equipment and supervise the installation. Based upon the above requirements, ABC engineering and operations people, and McCurdy engineers concurred in the decision to use a centralized switching system with remote control at each of the input points in the control rooms.

The heart of the ABC radio network feed selector switching system is a large bank of fifty-position step swithces located in the central equipment area. One step switch is assigned to each of the input points (console remote inputs, etc.). The control module is compact, approximately 9 x 5¼ inches, and can be conveniently located either adjacent to the appropriate console fader, or mounted on a tape recorder. A photograph of the control module is shown in Figure 1. A photographic negative reads out the forty-nine available feeds (the fiftieth position is off). The negative can be changed quickly and with a minimum of expense when the nomenclature changes. Panel-engraving obsolescence, with its accompaniment of unsightly pieces of tape and grease-pencil marks, is thus eliminated. A selection is made by punching the lettered button (marked *A* through *E*) and then the numbered button (marked *1* through *9*) opposite the desired feed. The remotely located step switch will then move almost instantly to the correct positions and a lamp behind the selected feed is illuminated (see Figure 1), confirming the operator's selection. When only a letter button is actuated the feed immediately under that button is selected.

In order to accommodate an even larger number of feeds, each step switch and associated control module, has sixteen selections that are in turn selected at the master control console. Each of these sixteen signals is selected by a step switch and control module, the latter being mounted on the master-control console. As the ABC system is now constituted it is possible to have eighty-two feeds available at each console remote input, at each cartridge recorder, and at each reel-to-reel tape recorder. The system could be expanded to 131 feeds if necessary. Figure 2 shows a reel-to-reel tape recorder with its

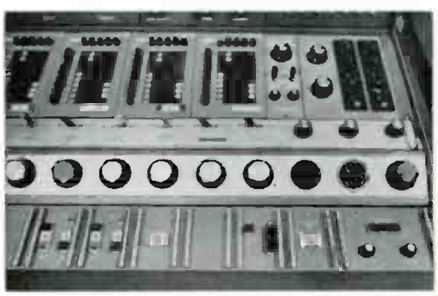

Figure 1. The control module at ABC radio's new control installation.

Figure 2. A reel-to-reel tape recorder with its selector module used at ABC radio.

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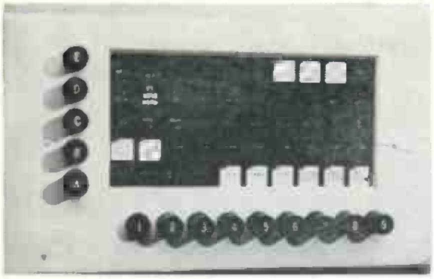

Figure 3. One of the ABC radio control consoles along with the selector modules mounted behind the faders.

selector module, and FIGURE 3 shows a typical control console with the selector modules mounted behind the faders. The tape recorder can be operated as a completely independent device with access to all feeds, even though it is located some distance from the control console. Operation of the selector modules at the console is straightforward, with the operator having control of the four channels within a 21-inch span of panel space. An important feature worth noting is that the operator can at any time, and from almost any position in his work area, quickly check which feeds are up on his equipment.

Since the same control module and panel configuration is used throughout the installation, no confusion need arise when an engineer is moved to a new area. Each module is terminated in a plug so that all movable equipment, such as the tape recorder (shown in FIGURE 2) can be interchanged for servicing or to augment facilities in a busy area. Additional receptacles for the control module can be wired in places where equipment is used only on occasions and permanent installation is not economical.

In a recent conversation with Harry Curtis, New York manager of technical operations, Mr. Curtis expressed his satisfaction with the system concept, and with its ability to effectively handle the complex of incoming and outgoing feeds. The one year that this system has been in operation has proven that by careful planning for *both* men and equipment ABC has come up with an efficient and flexible method of handling the difficult requirements of a modern radio network.

Theory and Practice

NORMAN H. CROWHURST

• The thought of making an electronic sequencer that will play any predetermined tune any time the button that starts it is pressed was exciting, or intriguing to say the least. So I rigged up a sequence of 2N395's (the type I'd been using in multivibrator circuits) open-ended, instead of as a back-and-forth multivibrator (FIGURE 1). It worked.

All transistors are normally conducting, saturated. The first capacitor is charged to full supply voltage, because the 22k at the left is connected to its negative terminal, while the base of the first transistor holds its right terminal at supply positive.

When the push button is pressed, the left side of the first capacitor gets connected to supply positive, so the right side is positive of supply positive by a voltage equal to supply voltage, until the resistor at the right discharges it. This renders the first transistor cut-off, or non-conducting.

When all transistors are conducting, all capacitors have virtually zero potential across them, because both sides are practically at supply positive potential (less the slight drop from emitter to base and collector respectively). But when the first transistor is cut off, the second capacitor, coupling it to the next stage, has its left side charged up to supply negative, quite quickly, through the non-conducting transistor's collector resistor.

When the first transistor starts conducting again, which will be when the charge on the first capacitor has leaked away through the 22k resistor circuit in its base (or when the button is released, if this happens first) the collector of that stage quickly moves to supply positive, cutting off the second transistor. This charges the third capacitor while the second is discharging toward the point that will allow the second transistor to start conducting again.

In short, the sequence follows on to the end of the line, each transistor staying cut off for a time determined by the time constant of the coupling capacitor in its base circuit, and the base resistor. With 50 mFd capacitors, which I used, and the 22k resistors shown, the time constant is 1.1 seconds. The cut-off time is 0.7 of this, as calculated in an earlier issue, which makes the time interval for each transistor 0.77 second.

This checked out according to theory, as close as the value tolerances allowed. Now to apply the idea to controlling frequency of a tone generator.

First, I knew that the output from each stage must be level during its interval. Taking an output from each collector would rise rapidly at cut-off, but not instantaneously, which wasn't what I wanted. Also, taking it directly from the collector would not give me any frequency control, because they'd all have the same voltage. And tapping the collector resistor would mean that when the transistors are conducting, the tapping point would not be at zero (supply plus) voltage.

Adding an emitter follower to each stage would solve both these problems. Current would flow in the emitter follower when the transistor to which it is connected is cut off, and *vice versa*, which will square off the waveform. And using the emitter to supply plus resistor as a potentiometer (FIGURE 2) enables the *on* voltage to be adjusted, while the *off*

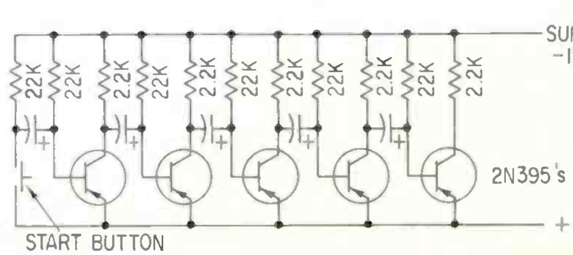

Figure 1. This sequence was rigged as an open-ended multivibrator and it worked.

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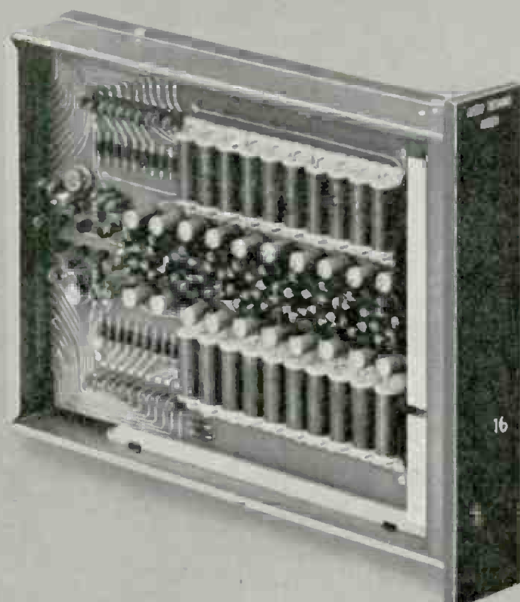

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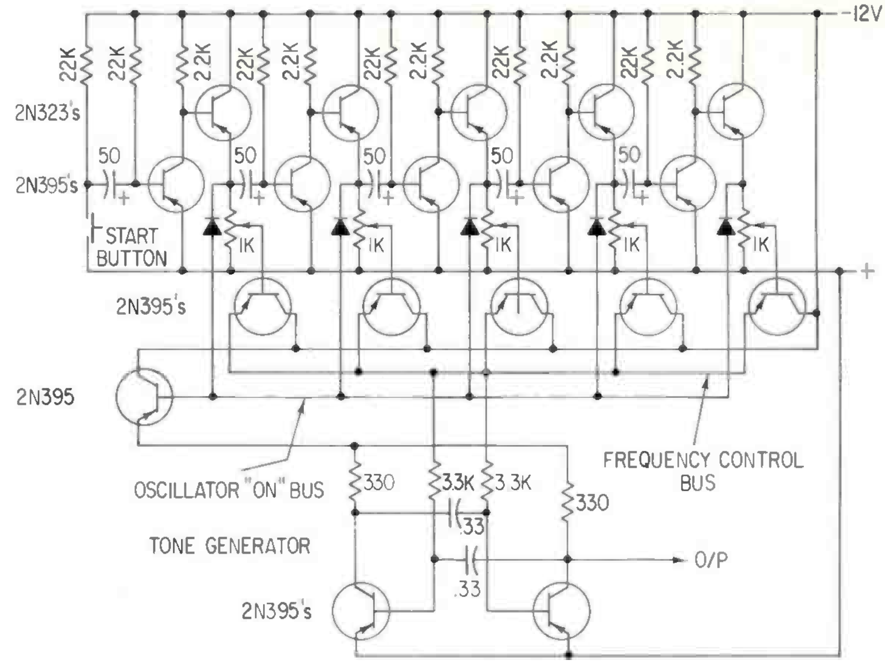


Figure 2. Emitter followers have been added to each stage of Figure 1 along with operating controls.

voltage always returns to supply plus (when the emitter follower is not conducting).

Now to select the frequency of a single tone generator, a series of emitter followers connected in parallel, with the base of each connected to one slider, enables the voltage at the common emitter line always to be that from the stage that is non-conducting, whose emitter follower is on, and providing the predetermined voltage at the potentiometer slider.

One more little trick, to switch the oscillator on at the beginning of the sequence and off at the end, is a series of diodes from each emitter follower emitter that is connected to another follower that supplies the voltage to the collectors of the tone generator. Whenever any one of the main emitter followers is conducting, its emitter is negative, so that diode conducts, making the single emitter follower that switches the oscillator *on* conduct. When all stages have returned to saturation, and all emitter followers are non-conducting, the oscillator is switched off.

That worked fine, with just one hang-up (or perhaps you could call it two). If I set the frequency controls so the notes run progressively down the scale, it works fine, I hear all the notes played in sequence. But if I set the frequency controls any other way, it doesn't work out. And sometimes it makes a difference how I touch the button. If I give it a good solid push and hold it until the first tone of the sequence is finished and then release, it works consistently. But if I leave go before that, all kinds of things can happen. And if I hold it too long, until the sequence is finished, when I leave go another odd kind of sequence starts. It's too "touchy."

I eventually figured out that, although every stage is conducting and every emitter follower supposedly cut off, these facts are only relative. There is some gain in this condition, and so the button actually triggers every alternate stage into non-conduction, when pressed. So what happens is that a whole series of sequences is started, one at every alternate stage. Each progresses to the right, until the last one; the one that starts at the first stage, reaches the end.

Then, if I held the button so the first capacitor remains discharged until I release, and that's after the sequence is finished, leaving go can trigger the second stage and all subsequent alternate stages, by a similar process.

To overcome this, I merely need to kill the gain that causes it. First I tried changing values in collector and base circuit of the main sequence. That didn't help enough. So then I tried putting a resistor in the coupling-capacitor lead. I found that 330 ohms was enough to kill these spurious starts, and let me have a single sequence that starts at the beginning and works through to the end.

But the button could still trigger some unusual effects, if I wasn't careful how to press it (with a good solid pressure) until the first two tones of the sequence were played.

I've run out of space so it will be next month before I can show you the final circuit that produced a workable system.

Editorial

PRESSURE IS RISING for federal regulation of performance standards in the audio field.

Professional audio is dominated by published specifications that are, with few exceptions, both meaningful and honest (despite minor skirmishes over the validity of some specific standards).

Confusion and outright misrepresentation are rampant in *consumer* audio, however, even in such obvious and simple specifications as the power capability of an amplifier.

Voluntary standards do exist within the consumer industries. The EIA (Electronics Industry Association) publishes one which most manufacturers of packaged goods (consoles, compacts, etc.) use and abuse. The IHF (Institute of High Fidelity), representing the hi-fi components industry, uses a more stringent standard, consistent with its image of selling superior quality. But neither organization stresses the continuous-power/distortion rating used in professional audio, so both standards result in power measurements that are higher than continuous-power specifications. Add to this the misleading quotation of *peak* power as though it were regular power, and the advertised figures have doubled again.

And now there is a new twist. A growing number of manufacturers are specifying the power of their amplifiers at a number plus or minus 1 dB. For example, an amplifier is quoted as delivering 50 watts ± 1 dB.

Seem clear enough? But that manufacturer is *not* saying his product has a production tolerance of plus or minus 1 dB around the 50-watt average. He has taken the plus 1 dB figure that his amplifier may on occasion reach, and called *that* a plus or minus 1 dB figure. Clearly, a credibility gap!

How does this relate to professional audio?

If this deceptive selling trend continues, we may face governmental interference in this area. More important, since our stringent standard is probably unacceptable to the consumer industry a federal power-response standard for both fields may compromise professional usage.

It seems likely that there will be a federal power standard. We are for it . . . but it must not work against us. It must not dilute the continuous-power measurement technique.

It must not prohibit usage. It should require every power quotation to state *in full* the derivation of the specification. This would fully protect the consumer and the honest manufacturer. L.Z.

Circle 66 on Reader Service Card →

MODEL 100

W.A.L. MODEL 100 INPUT/REMIX MODULE

SPECIAL FEATURES

- Extended range input attenuator
- Choice of two line or one line and one mic. input
- Special effects insertion switch and indicator lamp
- Pre/post echo send switch and echo send control
- Cue switch and level control
- Choice of 9 front panel colors
- Highest quality Grayhill subminiature switches
- Mu-metal shielded toroidal input transformer

DESCRIPTION

This solid state amplifier module contains a host of functions which are outlined as follows:

1. A low noise microphone preamplifier with low impedance balanced input.
2. A straight line vertical slide attenuator for microphone or line mixing.
3. A line or microphone input selector switch with a line position and four microphone positions for 0, 15, 30, and 45 db of attenuation to prevent overload on signals from very loud sound sources or to act as an extra line input to allow input switching between the output of two line level sources such as console and recorder outputs.
4. A built in no-loss program equalizer with a high and low frequency boost and cut control. Low frequencies are selectable at 40 or 100 Hz., while high and midrange frequencies are 1.5, 3, 5, and 10 Hz. Both high and low boost controls have steps at plus and minus 0, 2, 4, 6, 9, and 12 db of attenuation.
5. Echo send level control and echo pre/post selector switch allows setting echo or reverb level plus a choice of taking the echo signal before or after the vertical slide mixer and the built-in program equalizer.
6. Built in echo and direct solid state line amplifiers to feed outputs to selector switch panels.
7. Cue or audition switch and control which switches the control room speakers from normal program material to the output of the cue buss which is fed from the cue level control and switch on each Input/Remix Module in a console system. This permits sample mixes on the cue buss without disturbing the recording signal controls.
8. External special effects insertion switch allows switching in an external special effects unit such as a graphic equalizer, a sharp cutoff filter, a large program equalizer, or a tape delay echo system into any Input/Remix Module without the use of a patch panel. A red lamp indicates when the special effects equipment is already in use with another module in the console.
9. An instrument panel mounted V.U. meter in a console system can be switched between the two line inputs of a Input/Remix Module being used in a separate remix section of a console system to allow A-B switching.

WIEGAND *Audio* LABORATORIES

3402 WINDSOR ROAD · WALL, NEW JERSEY 07719

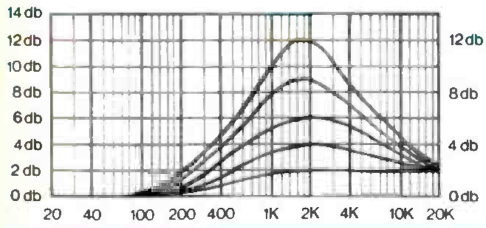
PHONE: (201) 681-6443

W. A. L. MODEL 100
INPUT/REMIX MODULE

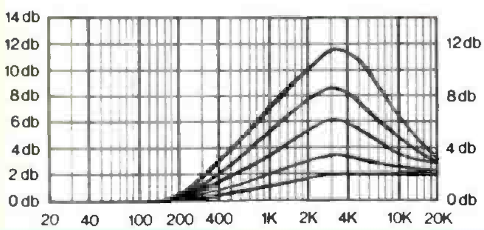
SPECIFICATIONS

Gain:	70 db
Input level:	Switch selection for line, mic., −15 db mic., −30 db mic., or −45 db mic. or line 2.
Frequency response:	15 Hz—25 kHz, ±.5 db at any level up to an output of +18 dbm.
Distortion:	less than .35% T.H.D. at +18 dbm.
Program output:	+18 dbm maximum into a 600 ohm load. Equivalent swing of +20 dbm into our active combining network input of Model 300.
Echo output:	Same as program output specifications.
Cue output:	Same as program output specifications.
Source impedance:	50 or 200 ohm mic. bal., 600 ohm line.
Output impedance:	Designed to work into a 600 ohm load. Internal output impedance is approximately 5% of rated source impedance.
Equivalent input noise:	−125 dbm.
Output noise, atten. zero:	84 db below +4 dbm.
Maximum input level with mic. 0 db position:	−26 dbm.
Equalization, low frequency:	Selectable at 40 or 100 Hz. Switchable in steps of plus 2, 4, 6, 9, 12, and 0 db.
Equalization, high frequency:	Selectable at 1.5, 3, 5, or 10 kHz. Switchable in steps of plus 2, 4, 6, 9, 12, and 0 db.
Attenuation, high and low:	100 and 10,000 Hz in steps of −2, 4, 6, 9, 12, and 0 db.
Output isolation:	Over 70 db between echo and program outputs.
Power requirements:	24 volts D.C. regulated at 188 ma. (Class A)
Physical dimensions:	1½" wide, 17½" high, 5¼" deep not including connector.
Special effects impedances:	Input 600 ohm, output 600 ohm. Extra gain in module allows up to 15 db insursion loss in Special effects unit.
Special effects Indicator lamp:	26 to 30 volts D.C.

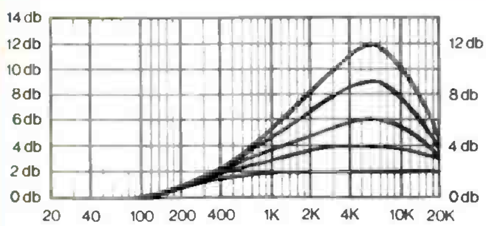
FREQUENCY CURVES



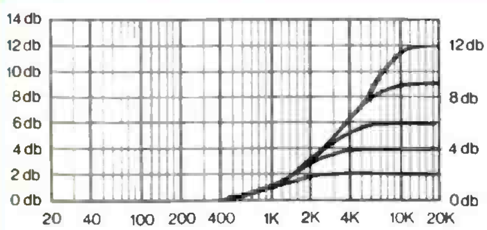
1,500 Hz HIGH FREQUENCY EQUALIZATION



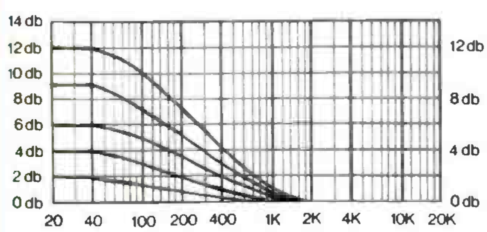
3,000 Hz HIGH FREQUENCY EQUALIZATION



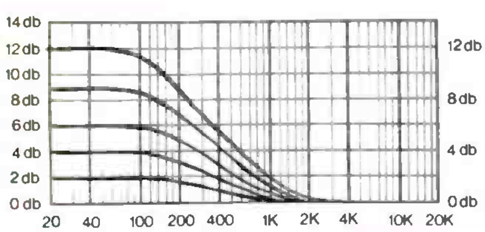
5,000 Hz HIGH FREQUENCY EQUALIZATION



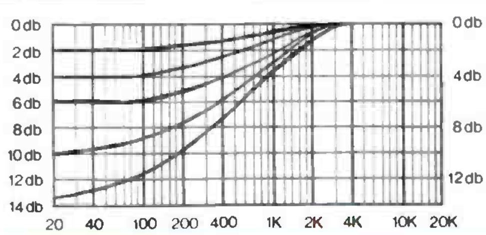
10,000 Hz HIGH FREQUENCY EQUALIZATION



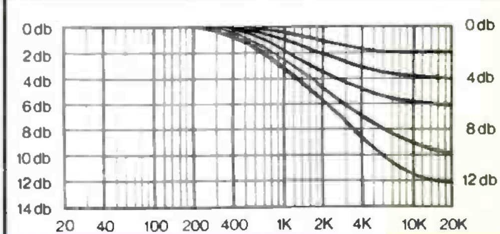
40 Hz LOW FREQUENCY EQUALIZATION



100 Hz LOW FREQUENCY EQUALIZATION



100 Hz LOW FREQUENCY ATTENUATION



10,000 Hz HIGH FREQUENCY ATTENUATION

MODEL 300

W.A.L. MODEL 300

MASTER RECORD/REVERB MODULE

SPECIAL FEATURES

- Two 22 input active combining networks
- Self contained reverb system electronics
- External echo send and return controls
- Master echo send V.U. meter
- Master record level slide mixer
- Two line amplifiers
- Built-in slating relay
- Choice of 9 front panel colors

DESCRIPTION

This Module fills the gap left in other console systems by providing in a single front panel mounted package, those functions that are required for each track of each recorder in a studio system. It replaces several separate modules plus provides some functions not found at all in other console systems. The use of this and our other modules greatly simplifies console wiring since all electronic and control functions are contained in front panel mounted plug-in modules instead of many separate isolated units mounted under the control surface. The W.A.L. model 300 contains a master record vertical slide mixer, a master echo send level control, a master echo send edgewise mounted vertical V.U. meter, two echo return controls to allow mixing the internal reverb and the external echo signals, two 22 input active computer type combining networks (one for direct inputs and one for echo inputs), a solid state current feedback amplifier and a preamplifier for use with an externally mounted Hammond Organ reverb spring delay pan, a balanced line level output amplifier, an echo send line amplifier, and a reed relay to allow placing slating information on a record track by the push of a button on the W.A.L. model 400 Monitor/Playback Control Module which contains the slating/talkbalk microphone preamplifier.

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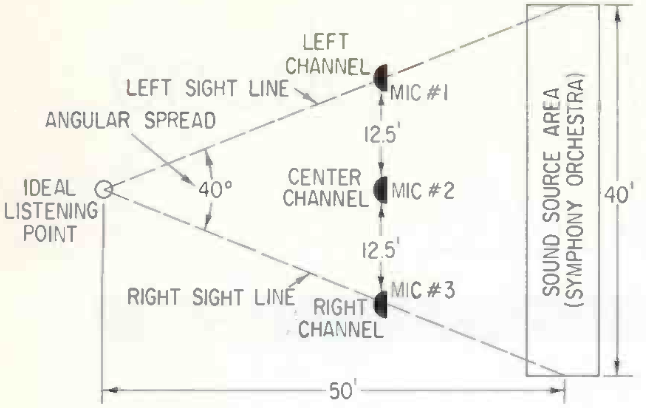

Figure 1. The listening-angle principle is used in this stereo mic placement for a large orchestra.

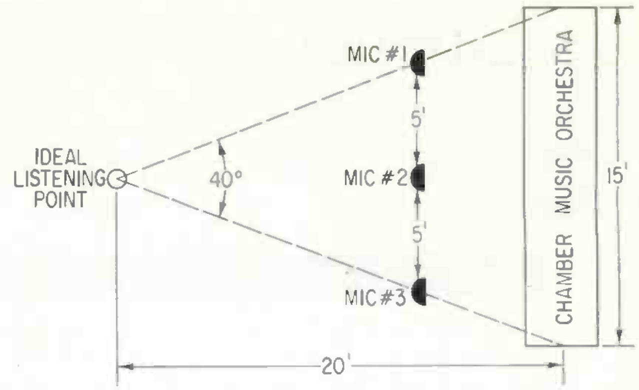

Figure 2. The angular separation of stereo mics for a small orchestra.

angular degrees in order to derive the proper linear separation of the transducers. For example, let us assume that the sound source width, or spatial distribution, of musical instruments in a symphony orchestra is about 40 feet across the stage. Normally, in order to obtain an esthetically pleasing balance between the direct sound and the multi-reflections of the concert hall, an ideal listening point is assumed to be located about 50 feet away from and directly in front of the orchestra. This point corresponds to the most favorable seat at the original performance, in which there is a smooth blending of sound without annoying bounces or other unnatural effects. Under these conditions, the angular separation formed between the hypothetical listener and the extremes of the sound source would be approximately 40 degrees.

From this listening point, let us visualize a pair of sight lines (FIGURE 1) forming both sides of the listening angle, one running to the extreme left and the other to the extreme right of the sound source. The left and right microphones are then positioned somewhere along the sight lines with the center microphone placed equidistantly between them. If the microphones are located too *close* to the sound-source area, there will be an exceedingly high ratio of direct to indirect sound with a consequent loss of the illusion of spaciousness. Furthermore, the reproduction will be harsh and lacking in musical quality, resulting in a poor orchestral blend. Placing the microphones too *far* from the sound source will produce a very small ratio of direct to reverberant sound, so that the excessive reflections would reduce the sense of directionality, resulting in a loss of clarity and definition. Between these extremes of microphone placement, an area can be found where the natural tonal balance is satisfactory, and the relative loudness of the instruments clearly defined.

Initially, a good procedure is to mount the two outside microphones so that the distance between them is about one half the distance between the ideal listening position and the sound source, which in the set-up shown in FIGURE 1, comes to about 25 feet. Normally, the principal axes of the microphones are directed straight ahead in the lateral plane toward the sound source. In some cases, however, depending upon the acoustic characteristics of the hall, better results may be obtained by inclining the outside microphones either slightly inward or outward in the horizontal plane. All three microphones should be elevated about 15 feet and aimed downward toward the sound source. The final adjustment in microphone positioning is accomplished when a compromise is reached between an acceptable balance and the desired perspective, which are evaluated in terms of subjective listening tests.

Now let us assume that we are arranging the microphones for a small chamber music orchestra in which the spatial

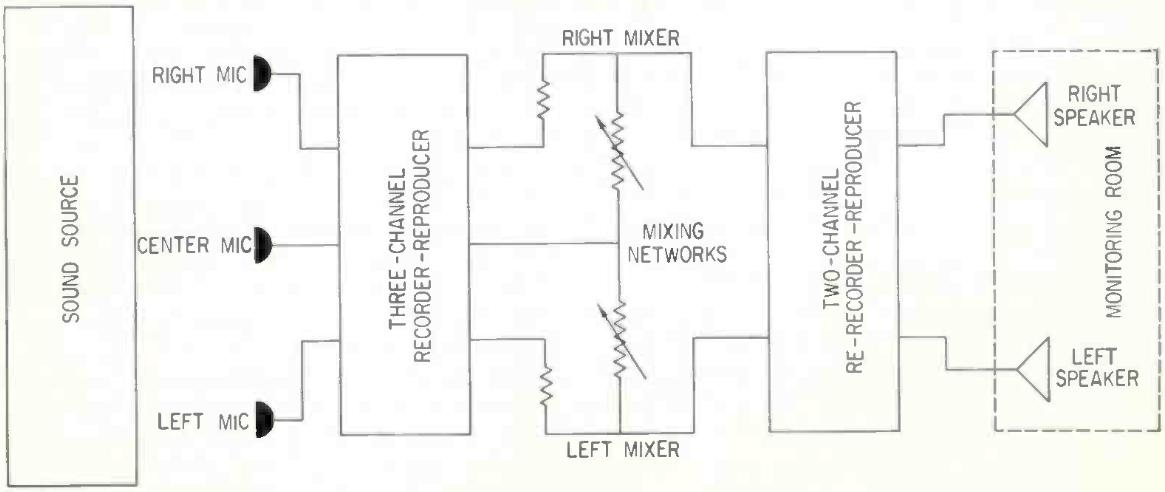

Figure 3. The mixing and dubbing process translates a three-channel recording system into a two-channel playback one.

separation between the musical instruments is only about 15 feet. Here there is an inferred shift of the ideal listening point toward the sound source, so that the effective angular spread is substantially independent of the sound source width. As shown in FIGURE 2, the outside microphones are positioned about 10 feet apart for the same 40 degree angular separation. In reproduction, the same angle should be formed between the listener and the two monitor speakers, for optimum stereo reception.

An important consideration is the choice of a suitable directivity pattern. As a rule, the omnidirectional microphone, which is equally sensitive in all directions, is most useful when widely distributed sound sources have to be picked up in halls of short reverberation times. Cardioid types, on the other hand, are more suitable when it is desired to receive most of the reflected sound from the front of the hall, ignoring reflections from the rear. Since they pick up only about one third of the reverberant sound energy, they can be employed in very lively halls of long reverberation times. When it is necessary to exclude unwanted reflections from the sides of the hall, bidirectional cosine microphones provide a very useful pattern. Here the principal axis of the figure-eight curve responds mainly to reverberant sounds from the front and back of the hall, usually giving a satisfactory over-all response.

In a spaced-microphone configuration, there may be more than three actual microphones employed in the recording session, since several microphones may be operating to serve one of the three channels. In the block diagram shown in FIGURE 3, each microphone channel is recorded independently on a separate track of a three-track tape recorder. The program material is then combined through mixing networks into left and right channels, and subsequently re-recorded to produce a two-track stereo master tape and, if required, a mono composite of the separate sections. During the mixdown, the engineer can modify or supplement the program material by rebalancing, equalizing, or adding artificial reverberation, as necessary.

COINCIDENT MICROPHONE TECHNIQUES

A common method employed in two-channel stereo transmission consists of a pair of microphones mounted coincidently, *i.e.*, in such close proximity that the differences in the arrival time of the sound are negligible. A specific form of this method is the *mid-side*, or m-s intensity technique, in which the stereo effects depend primarily upon interchannel amplitude differences. In a typical set-up, the system consists of two condenser microphones mounted axially one above the other, and contained in the same housing with their respective preamplifiers. One of the units, designated as the mid microphone, is a cardioid type which is oriented to receive the sound equally from all sections of the orchestra. The other unit, referred to as the side microphone, has a cosine, or figure-eight characteristic, and is arranged so that the principal axis of reception is parallel to the sound source. Hence, in the total pickup pattern in FIGURE 4A, the null plane of the figure-eight curve coincides with the principle axis of the cardioid. An alternate form of m-s intensity stereo (FIGURE 4B) combines the directivity pattern of the cosine microphone with that of an omnidirectional type.

In operation, the coincident microphone pair is mounted at a central location in front of the orchestra. Any sound reaching the cosine microphone from the extreme right will produce a signal 180-degrees out-of-phase with that of the extreme left. Those signals which do not originate from left or right of the sound source produce no signals at the side

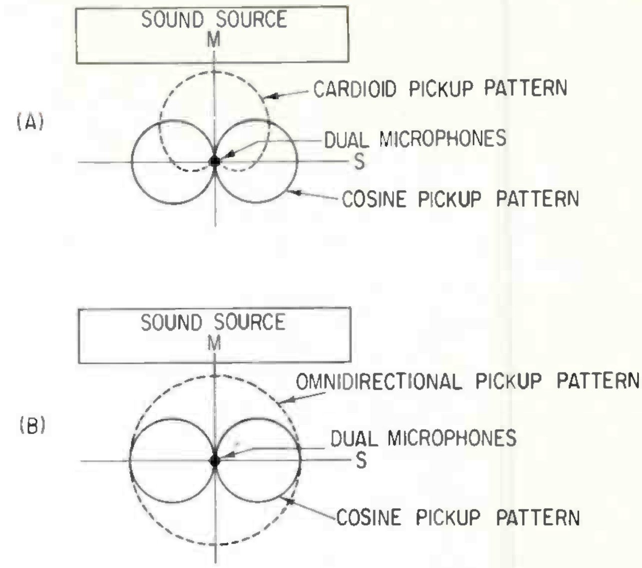

Figure 4. Polar diagrams of two variations of a coincident microphone system for m-s stereophony.

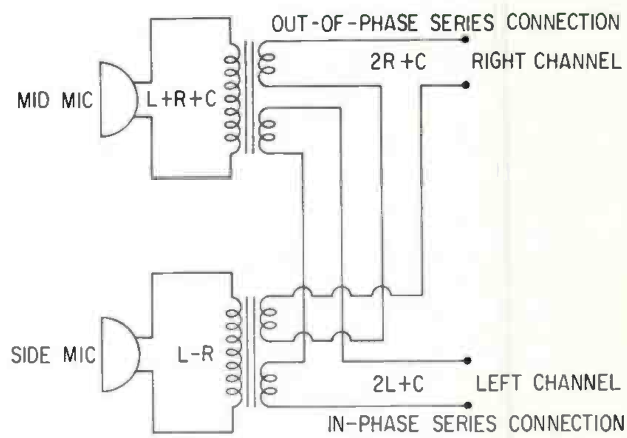

Figure 5. A matrixing circuit for signals produced by a m-s mic system.

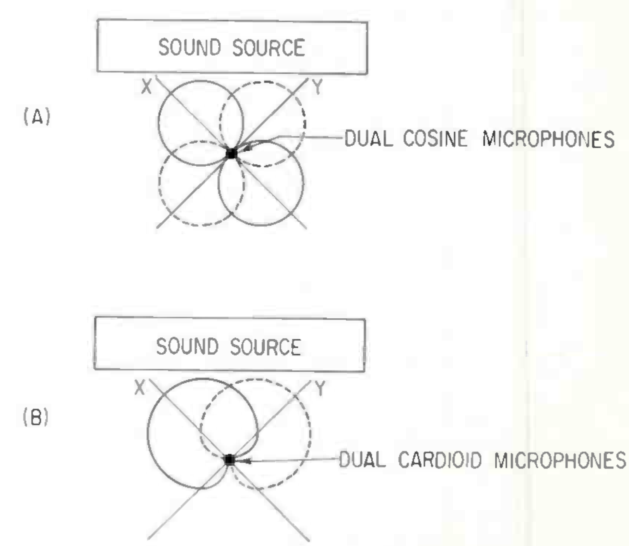

Figure 6. Polar characteristics of two versions of a dual mic system for X-Y stereophony.

Figure 7. A typical condenser stereo microphone, the AKG C-24 (courtesy of North American Philips Co.).

microphone, so that we designate its output as *left minus right*, or L−R. Since the mid microphone picks up the entire information, namely the left, right, and center, we may represent its output as L+R+C. The left and right stereo signals are subsequently derived by respectively adding and subtracting the two microphone outputs by means of a matrixing process. In the matrix circuit shown in FIGURE 5, two identical microphone transformers with exactly balanced secondary windings, are wired in series adding to obtain the sum component; while the other windings are connected in series opposing to yield the difference. Both microphone outputs are then combined in phase to produce a signal containing left information plus a reduced center-signal component, namely (L+R+C) +(L−R) = 2L+C. The out-of-phase signal for the right channel is obtained by subtracting the two outputs to yield (L+R+C) − (L−R) = 2R+C. Hence, the required left and right stereo signals are effectively separated with some center information remaining in both channels.

In actual practice, the matrixing levels are carefully controlled by the sound engineer so as to make the recording system more flexible. The use of differential controls permits the relative gain of the sum and difference signals to be boosted or attenuated over a limited range. For example, by raising the level of the difference signal at the expense of the sum component, increased separation between both channels can be achieved. Conversely, attenuating the difference component causes the center signal to become emphasized so that the stereo effect is less pronounced. Since the sum channel carries an acceptable single-channel transmission, a mono characteristic can easily be produced.

In another intensity-difference system, the polar characteristics of a dual bidirectional microphone system form two figure-eight patterns mutually perpendicular to each other, with the principle axis of each microphone inclined 45 degrees to the sound source. This is referred to as the X-Y system (FIGURE 6A) in which one cosine microphone picks up sound essentially from the left, while the other receives information mainly from the right. Any program material arriving at the forward quadrant of the X-Y axis will be equally received by both microphone channels to make up the in-phase component. Out-of-phase components, however, will give rise to amplitude differences in both channels. Using a similar matrixing circuit to the m-s system, the different components can be accentuated or reduced in relation to the sum components, so that the final stereo presentation can be made broader or narrower as desired. To increase the working angle over which the system is capable of functioning, a double cardioid stereo microphone (FIGURE 6B) can be used to advantage.

FIGURE 7 shows a pictorial view of a professional condenser stereo microphone designed to accommodate the various forms of intensity-difference stereo. Using a separate control unit, the directional characteristics of the dual microphone system can be changed during operation to omnidirectional, cardioid, cosine, as well as a number of intermediate pickup patterns.

MULTI-MICROPHONE TECHNIQUES

In multiple microphone arrangements, a main stereo microphone pair provides the general sound image with the appropriate perspective, while supplementary microphones function singly or in pairs to emphasize particular sections of the orchestra. The over-all stereo effect is to produce a well-

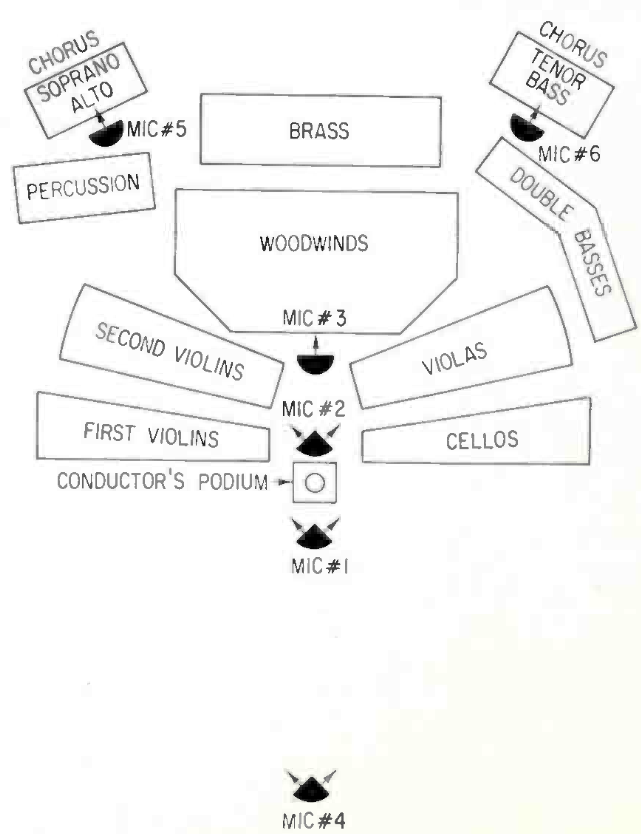

Figure 8. A multi-mic arrangement for symphony orchestra employs three coincident pairs, one spaced pair, and one accent mic.

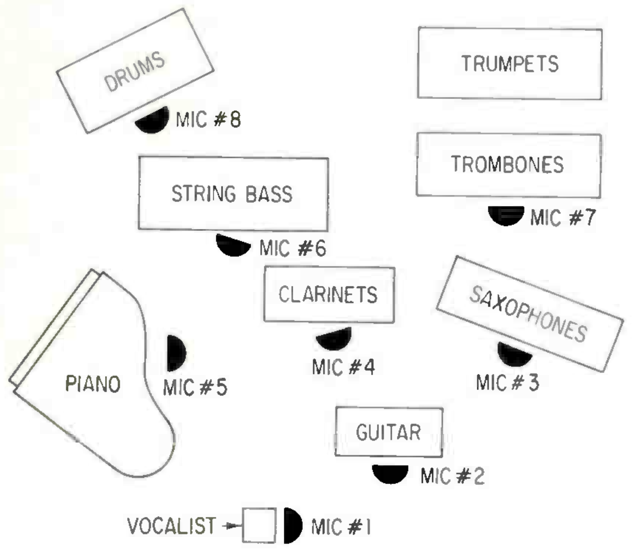

Figure 9. This close-microphone system uses an independent eight-channel arrangement.

defined image, so that the point from which each sound source appears to originate in the recording hall yields a corresponding image in the monitoring room.

To demonstrate some of the actual problems involved in microphone placement, let us assume that we are experimenting with a multi-mic set-up during a rehearsal of a full symphony orchestra and chorus in a large recording hall. In the particular instrumental layout shown in Figure 8, the first violins are placed on the conductor's left while the cellos and violas are grouped toward his right. As a starting point, a suitable choice for the main pickup microphone could be a dual cardioid type operating in the X-Y mode, and mounted about 15 feet over and slightly behind the conductor's podium. In monitoring the reproduction, the optimum position of this microphone pair (1) is usually determined by the balance of the string section in relation to the brass and woodwinds. Owing to the relative weakness of the string instruments (which require a higher ratio of direct-to-reverberant sound), they tend to be obscured by other types with more substantial tonal quality. Hence, if microphone 1 does not yield a proper balance, a subsidiary stereo cardioid pair (2) could be used to emphasize the strings. This microphone pair should be elevated about 8 feet with the left unit aimed at the first violins, and the right unit directed toward the violas. Since the violas are characterized by an inherent veiled tonal quality, the possibility of introducing artificial reverberation to the right channel should be considered. The use of this additional microphone pair to reinforce the strings is acceptable, provided that care is taken that the directional information does not conflict with that of the main stereo pair. If necessary, a single cardioid accent microphone (3) could be used to emphasize the woodwinds, in order to further improve the tonal balance. The output of this microphone can subsequently be divided between left and right stereo channels by means of a pan-pot control.

At this stage, it might be decided that the amount of reflected sound from the boundary surfaces of the hall is not great enough to convey an impression of breadth and liveliness. This spacious effect could be attained by placing a distant cosine microphone pair), operating in the X-Y mode, at a high position to the rear of the hall. In this region the direct sound is negligible compared with the reverberant sound, thus contributing to an over-all softening of the reproduced tone. Of course, if the general background noise (due to ventilators, air-conditioning units, etc.) is objectionable, this microphone can be dispensed with and artificial reverberation added instead.

Now let us consider the problem of achieving a satisfactory balance between the instrumental sections and the chorus, which in this case, is arranged in two sections behind the orchestra. To adequately reinforce the choral groups without unbalancing the orchestral pickup, it would be feasible to employ two spaced cardioid microphones (5 anc 6), each microphone aimed at one section of the chorus.

A difficulty may arise when two stereo microphone pairs pick up a soloist who happens to be located between the pairs, so that the time difference involved is great enough to prevent a single resultant image from being formed. When this occurs, any movement of the soloist can cause the reproduced sound image to leap from one side of the stage to the other. Fortunately, this effect is negligible when the grouping consists of similar voices (or instruments), so that a widely spaced microphone arrangement is satisfactory for large choirs. Another problem may be encountered when sound from a particular instrument is picked up at comparative levels between an accent microphone and one unit of the main stereo pair. Here the two microphones concerned act as a spaced pair and produce an undesirable directional characteristic. This directional information, which depends upon interchannel time differences, conflicts with that normally derived from the main stereo microphone pair which, in turn, depends upon amplitude differences.

CLOSE MICROPHONE TECHNIQUES

In the case of popular music, more latitude in microphone placement is possible since the acoustic environment does not place such exacting demands on the recording perspective as they do in classical music. Here individual cardioid microphones are arranged in a close-mike pickup (which is actually a special form of the multi-mic technique), utilizing as many units as required to obtain the presence and intimacy desired in today's pop music. As shown in Figure 9, each microphone picks up predominantly one program source, or group of sources, so that the various sections of the orchestra are effectively isolated during the recording process. For example, one microphone may be placed to pick up the woodwinds, another oriented toward the brass, a third positioned close to a vocalist, etc. — each output being fed to a separate channel of a multi-track tape recorder. Although an eight-channel system is given here, it is not uncommon to employ 12, 16, or even 24 channels in today's recording sessions.

It is obvious that there is no spatial effect in this arrangement because the isolation on each channel is more or less complete, and there is no chance for the precedence effect to operate. Since the reflected sound is negligible, the system must operate on the amplitude difference theory where the position of a sound source is determined by the relative loudness produced during the reproduction. Close-miking enables the engineer to control the balance of each sound source independently, so that by repositioning the sound images during the electrical mixing process, it becomes possible to create perspectives that did not exist in the original performance. Moreover, by adding the proper amount of artificial reverberation to any particular track, weak instruments can be raised to dominant solo level and those with large carrying power can be subdued. In the mix-down, the multi-track program material may be re-mixed any number of times until the engineer obtains a satisfactory stereo and mono master tape.

Consistency of sound track quality on an endless variety of locations and sets can be dramatically improved with the remarkable Shure SM5 Boom Microphone. It "hears" the dialogue rather than the ever-changing character of the surroundings.

Because its cardioid directional pattern is uniquely uniform with frequency and symmetrical about its axis, the SM5 is singularly independent of the effects of environment. Even in extreme shooting situations (such as with tight sets, low ceilings, hard walls, low microphone angles, traffic or air conditioner noise and rumble, and changing distance) the SM5 minimizes sound coloration and ambient noise pickup. Equalization changes—on the set or in transfer—are seldom, if ever, necessary.

The highly effective attached windscreen completely encloses the two-stage mechanical filter, so that there are no external "rubber bands" for the wind to "strum." The absence of response-correcting inductors or impedance transformers assures freedom from hum.

Call on the Shure SM5 to solve your most annoying boom problems!

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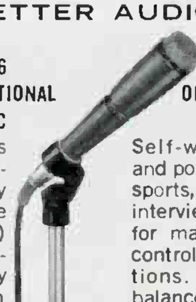

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Circle 68 on Reader Service Card

Improved Monitoring With Headphones

HOWARD T. SOUTHER

The inadequacies of present monitoring facilities occupy this month's part. In part 2, the author will present a possible solution to this problem through an improved headphone design.

MODERN RECORDING DISPLAYS the employment of diverse and remarkably refined techniques. Signal-to-noise ratios in the last few years have been increased by an entire order of magnitude. This has been brought about by improvements in disc cutters, like the Westrex 3-D, the D signifying its fourth round of improvement. Amplifiers for driving the cutter exist which permit 6dB of added level without danger of cutter burnout. Dolby is distributing an electronic system for recording the master tape which approaches yet an additional 10dB of signal over noise.

Coincident to better signal-to-noise has been the increased transient response allowed by the much greater dynamic range, while distortion products of all kinds have plummeted downwards. A degree of spectral control over the signal has been achieved which approaches redundancy; we are now confronted with consoles which provide a program equalizer over every pot, above which is an individual reverb control and an array of push-buttons which allow the microphone channel to be monitored separately, and then inserted into any one of several different output channels with the pressure of the finger-tip.

In contrast to this superabundance of control over the electrical signal is the relative lack of control over the monitoring of the recorded material. Not only do monitoring facilities vary in type and quality from one record producer to another, but within the organization from studio to studio. Some of the finest recordings have reached the field as the result of lucky accident. Other fine possibilities have died a-borning, simply because they were made to sound good in poor monitor rooms, the inference being that they could sound good in the field only on poor playback equipment which distorted it exactly the same way as did the monitor.

In what follows we may not expose a rigorous solution to this problem, but we may have some claim on initiative by disclosing a way to control and standardize the monitoring environment. Here are some of the principal considerations invoked:

Signal-to-Noise. The average isolation offered by a monitor-room door is 40 dB. From this we must subtract air-conditioning rumble and the ever increasing noise caused by a growing group of listeners, such as the producer, artists, visitors, etc. The ambient noise is now far above the capabilities of the system, unless monitoring takes place at very high levels, approaching on many cases non-linear operation of the ear.

Variations in Monitor Room Acoustics. Each monitor room has its own characteristics. To generalize, there are the standing waves which produce peaks and valleys in the lower range, the degree of liveness remaining from its absorbtive qualities in the high range, and its reverberation time. These combine in varying proportions to make reproduction in one monitor room like that heard in no other. However, an indirect frame of reference can be established: You can say that if a recording made in a certain monitor room has a characteristic sound that makes it sell well, another performance that has the same characteristic sound while being monitored in that same room will sell well also. The frame of reference is lost if the performance is made through a different monitor room.

Variations in Monitor Loudspeakers and Their Placement. An attempt is made usually to standardize on a particular speaker type within one recording organization, but types and makes vary between one organization and another. Little attempt is made to standardize the placement of the speakers at distances and directions in front of the monitor position, and this varies frequently from six to twelve feet and 40 to 120 degrees. Some monitor speakers are available with an unusually flat response characteristic, but only on the axis.

HEADPHONES FOR CONTROLLED MONITORING ENVIRONMENT

For over thirty years headphones have been used for recording dialogue in motion pictures, but scoring the picture with wide-range music has demanded monitor rooms which in their ultimate have simulated an 800-seat theater with normal speaker complement; a very large and unwieldy monitor environment indeed.

The question is asked "why have headphones been limited in the motion-picture business to dialogue recording?" The answer is because dialogue is reproduced at a much higher level than that at which it was recorded, a 12 dB-per-octave roll-off starting at 400 Hz is employed for "dialogue equalization", and consequently little demand is placed on the headphone for bass response. The headphone, in addition, is not called upon to deliver response above about 8000 Hz for motion picture work. This is excellent accommodation to what has been a low state-of-the-art in headphones.

In the field of high-quality sound reinforcement — a typical example is the new Los Angeles Music Center — where

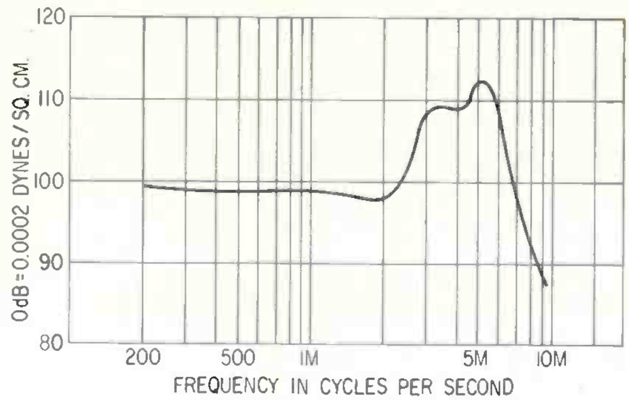


Figure 1. The response of the THD-39 earphone in an ASA Type 9A mechanical coupler.

a controlled monitoring environment is offered through even "poor" headphone listening. The sound engineer is placed in the center of the audience with his controls in a portable mixer, and a direct correlation between what he hears through the headphones and the actual performance is achieved by taking the headphones on and off as he balances the reinforcing system. Herein lies an important problem with headphones — the inability so far to equal or exceed loudspeaker range response.

SOME LIMITATIONS IN PAST HEADPHONE DESIGN

Measuring Methods. What little has been available to contribute to headphone development has stemmed from audiometry equipment. An ASA standard coupler exists as of 1954 which gives a calibrated response from an arbitrary not-so-good headphone used for a standard, the THD-39. An average ear responds to the coupler/headphone result as in Figure 1.

Some work has been engaged in with flat-plate adapters to allow later types of headphones to be measured. However, correlation with what is actually heard has been very poor, the reference range is by far too restricted with many chances for error because of the extrapolations required.

Bass Response. The chances for excellent bass response in headphones have been good, but not realized. The headphone against the ear plays into a theoretically closed cavity; this should allow linear response below the audible range. A leak between the headphone and the outside air, however, can radically change an impedance match of 400 acoustical ohms to that of mismatch with the infinitely low impedance in the outside air, and with deleterious effect. This can be offset to a degree by large diaphragms, but their relatively high mass and compliance promotes a resonance in the higher bass range followed by rapid fall-off in the first three and one-half octaves.

High Response. One of the most difficult hurdles to overcome is that of achieving extended high end response, especially above 8000 Hz. The compliance of the trapped air volume between the diaphragm and the ear acts as a shunt capacitance across the acoustical circuit, and if this is large, as in most headphones, the ability to transmit high frequencies to the ear mechanism is grossly impeded, no matter how well the diaphragm vibrates.

THE DESIGN

It would appear from the foregoing that control of the monitoring environment is necessary, and that headphones

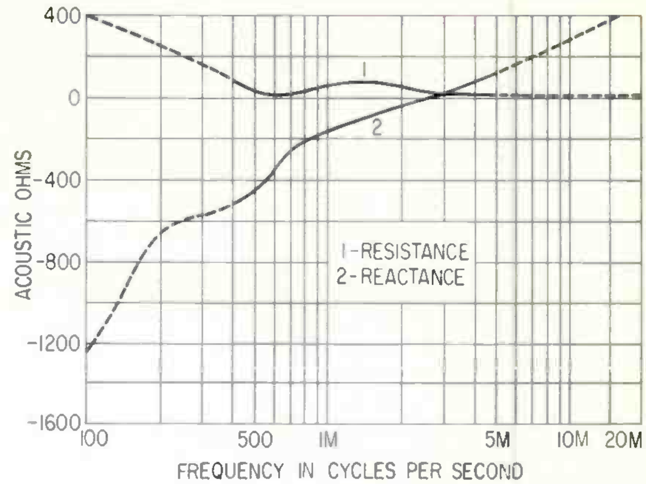


Figure 2. The acoustic impedance of the ear. Dashed lines in the upper range represent extrapolations by the author, J. C. Steinberg, in his book The Sense of Hearing.

may offer a solution. Headphones must have equal or better response range to achieve this purpose, while adding no further distortion products to the perceived signal. With this premise before us, we now proceed with design.

To reproduce the low range with facility, the diaphragm mass and compliance should be large. This implies a high, flexibly suspended mass. To reproduce the high range well, the mass and compliance of the diaphragm must be small. This requirement eventuates in a moving member attaining high stiffness, a thin section, and small diameter.

Within the range of 200 to 8000 Hz, we find cone-type miniature speakers employed in present headphones to favor the lower part of this range. Those headsets displaying good response from 1000 to 5000 Hz embody smaller, microphone-like generating elements about an inch and one-half in diameter.

The best sounding headphones available are those subscribing to the accepted but rather emperical formula:

$$F_u \times F_l = 500{,}000$$

where: F_u = upper cutoff frequency in Hz
F_l = lower cutoff frequency in Hz

This formula has worked well through the years, especially in the motion-picture industry, where, for example, the size of the bass speaker was limited to 50 Hz and about 48 inches in depth behind the screen (a whole, costly row of seats!) and the high end to 10 kHz because of release printer slippage. In present headphones, the "better sounding", or more musically-balanced assemblies, lean towards a dull high end to match the deficient bass range, so avoiding classification as "screamers".

The fact that an extended high end is not desirable with present headphone designs is a boon to the designer of conventional units, for the ear cavity adjacent to the diaphragm acts acoustically as a shunt capacitance, thus making uniform response at the ear difficult to attain. The THD-39 virtually cuts off at 8 kHz (Figure 1).

Granting control of the listening environment as a prerequisite, and headphones as a subscription to this requirement of good listening, we find ourselves now confronted with what appears to be a deficiency in the medium of our choice: poor range response, both at the high end and the low end, seemingly inherent in the dynamic generators common to the present headphone art. An examination of an alternate sound generator appeared in order.

(To be concluded next month.)

A Super-Power P. A. System

ELLIOTT D. FULL

It takes a lot of public address power to fill an airfield. High-power p.a. can be extremely expensive. Here is one way in which costs were brought down without a sacrifice in performance.

STATION KXIC AM/FM HAS for many years been providing a 100-watt audio system for this city's 4th of July celebration. The audience has grown from five or six thousand to crowds approaching twenty thousand. Area swim meets, for which we provide audio, also have been growing in size.

The annual *Antique Aircraft Association Fly In* at Ottumwa, Iowa, a national affair, offered us a challenge to get into an unusually large p.a. operation.

The need to build a spare power supply and modulator for our a.m. transmitter fairly well solved our increased p.a. problems. All we needed to do was to construct a complete 300-watt booster with a Vari-match output transformer. This class B amplifier designed with a pair of 811A tubes could then be used to "double in brass" by merely changing output plugs. Our old Conelrad transmitter was a gold mine of parts so that the total expenditure for additional parts did not exceed $65.00. KXIC already owned a fine 25-watt p.a. with output taps up to 500 ohms. This unit would work very well as a driver. The new booster was built on one 13-by-17-in. chassis. The chassis also holds a 1500-volt, 500-mil power supply and a watt meter. Silicon rectifiers were used and shunted with 250k equalizing resistors to produce a typical class B 300-watt unit. About 10 per cent feedback has kept the maximum distortion below 7 per cent which is perfectly acceptable for such a powerful unit. A switch on an input r.c. circuit allows a 16 dB roll off below 200 Hz — a speaker saving device. When this system is used at maximum power for p.a., the driving amplifier's bass is rolled off as well, so that occasional overloads will not cause speaker damage.

Once the booster was built and operating, the speakers became the next concern. Over the years we had acquired a number of projection horn speakers. A Jensen horn with a 100-watt DD100 driver, along with several 25-watt units gave us a start to our high-power system. These would not, of course, take the power of the new booster, so it was decided to buy a University 4A4 four-driver horn and to equip it with four 40-watt drivers.

The next problem was to match the 4-ohm 4A4, the 8-ohm DD100, and a 16-ohm speaker to the 500-ohm line output of the large booster. The solution to this fairly knotty question arrived in the form of one of two 10-volt, 20-amp filament transformers we had in stock as spares.

As can be expected, the transformer's 110- to 125-volt tapped primary is on the inside by the core. All that had to be done was to remove the shells, the 10-volt secondary winding, and add the new secondary. As the transformer carries little audio power below 300 to 400 Hz, the inductance of the primary makes it possible for this winding to carry the 400-volt line output without excessive losses. The core losses would be greater when the transformer is used at audio frequencies, but the amplifier had plenty of power. More important, such transformers are not available from the nearest jobber.

As an example of how we worked all this equipment into one operating unit to cover an air show, lets continue with the design of the matching transformer. Assume 400 volts, approximately 300 watts, coming out of the amplifier. A large commercial matching transformer on the 1500-ohm tap gives the 100-watt speaker 106 watts. To achieve a match then, the transformer being rewound has to have an 800-ohm primary. It should then deliver 25 volts to the 4-ohm speaker and 20 volts to a 16-ohm, 25-watt speaker. These three speakers, considering their power and angle of radiation would be able to cover the whole show consisting of about five-thousand people and up to five-hundred aircraft.

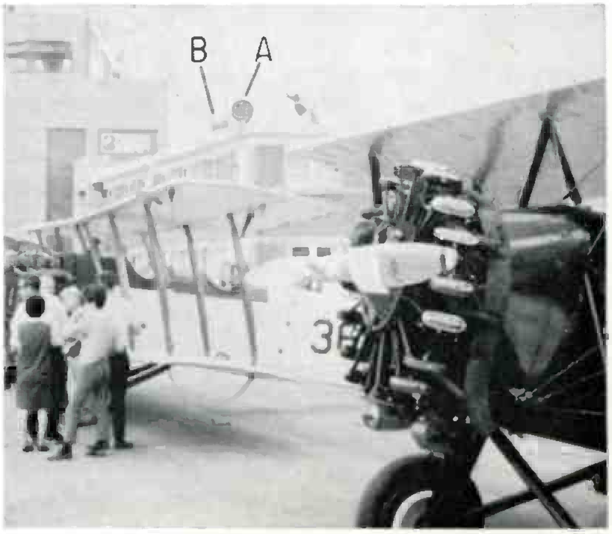


Figure 1. Two of the speakers were placed on the roof of the airfield's administrative building. At (A) the large 160-watt speaker, and at (B) the smaller 25-watt horn may be seen. A separate wide-angle horn (not seen) covered an airplane display area.

Elliott D. Full is vice-president, engineering at KXIC AM/FM, Iowa City, Iowa.

25 volts out of the transformer operating on a 400-volt line is the same ratio as 7.4 volts off a 115-volt line. With the primary connected to the 115 Va.c. line enough No. 16 wire was wound over the primary to give 7.4 volts. This was then tapped at 5.7 volts for the 25-watt speaker. Several other windings were also added so that other speaker configurations can be used for other jobs. The transformer was then reassembled and the taps brought out to a terminal strip. Many people dealing with normal audio equipment do not consider shock hazards once the back of a tube-type amplifier is closed up. A 400-volt line should be treated with extreme caution. Normal power-line type grounding techniques should certainly be used.

Figuring impedances across the square of the turns ratio has everything comes out quite closely. No allowances have been made for transformer losses, as specialized equipment is needed and the impedance to be matched is only a nominal one. After a day of nearly continuous operation the transformer is only slightly warm.

The equipment was tested at the station's transmitter site. We are fortunately located well out in the country. With these speakers operated "full bore" the low hills surrounding us echoed like a rifle would in a mountain canyon.

With the tests completed and everything working fine, all the equipment was loaded into the company's Piper Comanche airplane after the right seat was removed. Then it was flown to the site of Ottumwa, Iowa. The big 160-watt speaker was placed on the roof of the main terminal and aimed at the flight line about one quarter of a mile away (Figure 1, arrow A). We needed the power here as aerobatics and taxiing aircraft would test any speaker. The 25-watt speaker was located next to the big one (Figure 1, arrow B) and covered a much smaller, closer area. The 100-watt wide-angle speaker covered the display area from the roof of another building. About ninety valuable antique aircraft were parked here. Again, the many operating aircraft engines necessitated this much power.

The Ottumwa Municipal Airport, where the AAA annual *Fly In* is held, is an old military field with very good facilities and an antique aircraft museum run by the AAA. Power necessary to achieve 100 per cent coverage of the whole field would be impractical and in many locations, painful. We did not plan to cover those visitors who were in the immediate vicinity of the larger taxiing aircraft.

An expert on old aircraft gave a running commentary most of the time during the two day event. All usual p.a. duties were performed in addition to the commentary. One *unusual* duty was to find a crankshaft for a 1920 engine so it's owner could get home.

A system of this size needs an operator who watches the gain and the power meter. About a year before we had blown our four 40-watt drivers in a very short period of time by not watching gain. Speakers carrying powers of this order should be placed above or beyond the crowd to be covered. This allows *the square law* to afford a measure of eardrum protection. (One year we had to install a 100-watt speaker on a moment's notice, so we located it near a crowd on a car roof. The "hole" carved in the crowd after a short period of operation closely resembled the radiation pattern published by the manufacturer.) For maximum efficiency a horn speaker located high should be aimed short of its most distant objective, or most of the top half of its radiation pattern will be wasted.

This job proved to be a real challenge. But our measure of success must be indicated by the fact that we received an invitation back to the next air show a full 364 days before it is to be held.

Sound with Images

MARTIN DICKSTEIN

HOW MUCH RESOLUTION IS USEFUL?

• *In September, 1966, a t.v. product specialist in the Burbank, Calif., office of RCA's Communications Electronic Systems Division wrote a most interesting paper with the above title. We are grateful to RCA in Burbank and Camden for their permission to reprint this discussion, a most timely comment even today:*

The resolution race of today's television market is much like the horsepower race in the auto market a few years ago, as each television manufacturer tries hard to quote a few more lines than his competitor. The horsepower race led people to ask "how much can be properly used?"; the same question is very timely as concerns television resolution. The limit on horsepower is the road and the condition of the road. The useful limit of resolution is the acuity of the eye.

The eye can, under the very best conditions, see an object with a viewing angle of one minute of arc (FIGURE 1). Under average conditions, the eye will reliably see an object with a viewing angle of two to three minutes of arc.

Many factors will influence the ability of the eye to see an object, one of the biggest being the contrast between the object and its surroundings.

Television resolution is measured in terms of object size with respect to picture height. A resolution of 300 lines means the object size resolved is 1/300 of the picture height, or 300 lines of that size would just fill the picture height.

For the purposes of this discussion, all measurements of size and distance will be expressed in terms of picture height, in order that we may easily compare all information.

The first consideration of useful limits of resolution must be based on the viewer's distance from the monitor. The recommended viewing distance is between four and ten times the picture diagonal (picture size).

This picture size (diagonal) is 1.527 times picture height, or 1.527h. (1)

For this discussion viewing distance will be stated as the ratio (R) between picture size and viewer distance, so that results developed apply to any picture size.

Stated as a working formula, then:

Distance in terms of picture size

$$(R) = \frac{\text{Distance}}{\text{Picture Size}}$$

Both stated in the same unit of measurement:

Substituting (1) above for picture size we have:

$$R = \frac{\text{Distance}}{1.527\ h}$$

Rearranging, we have:

$$\text{Distance} = R\,(1.527\ h) \qquad (2)$$

We now have viewing distance expressed in terms of picture height (h) and picture size vs. viewing distance (R). Using simple trigonometry, we can now establish a formula which will tell us what object size the eye can see at a given distance [FIGURE 2], measured in terms of picture height in order that we may relate this size directly to resolution, [FIGURE 3].

$$\tan \tfrac{1}{2} A = \frac{\tfrac{1}{2} X}{R\,(1.527\ h)}$$

therefore:

$$X = 2\,(\tan \tfrac{1}{2}A)\,R\,(1.527\ h) \qquad (3)$$

For viewing angles of 1 min. and 3 min. insert tan of ½A or tan ½ min. and tan 1½ min., rearrange and simplify to:

$$1 \text{ minute},\ X = R\,(.00041\ h) \qquad (4)$$

$$3 \text{ minutes},\ X = R\,(.0014\ h) \qquad (5)$$

At a viewing distance of four times the picture size (R = 4) and a minimum viewing angle of one minute:

$$X = 4\,(.00041\ h) \text{ from (4) above}$$

$$= .00164\ h = \frac{164}{100{,}000}\,h = \frac{1}{610}\,h$$

= 610 lines of resolution useful. (6)

At the same distance, with a minimum viewing angle of 3 minutes:

$$X = 4\,(.0014\ h) \text{ from (5) above}$$

$$= .0056\ h = \frac{56}{10.000}\,h = \frac{1}{180}\,h$$

= 180 lines (7)

Similarly, the useful resolution at the maximum viewing distance of 10 times picture size would be:

at 1 minute, 245 lines
at 3 minutes, 72 lines

These figures show the maximum useful resolution to be 610 lines, and this only close to the picture, under ideal conditions. From these figures, it is easy to see that system resolution beyond about 400 lines is useless to most viewers, and that even 400 lines of resolution will be hard to see unless contrast and other viewing conditions are carefully tailored.

That concludes the paper on resolution written two and one half years ago. We welcome any comments, thoughts or experiences you may have had in the resolution race.

Figure 1. The limit of resolution of the eye is limited, under the best of conditions to a viewing angle of one minute of arc.

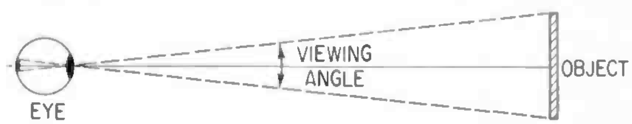


Figure 2. Using the formula (2) in the text we can establish what object size the eye can see at a given distance.

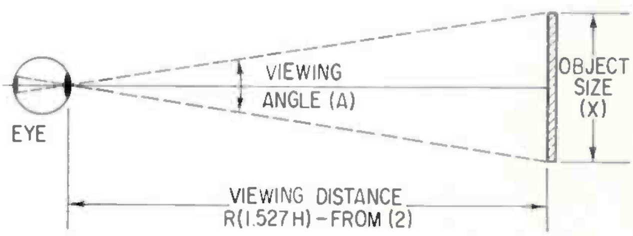
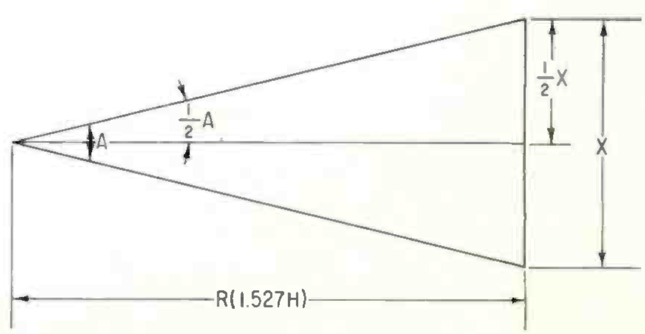


Figure 3. The size established in Figure 2 can be related directly to resolution.



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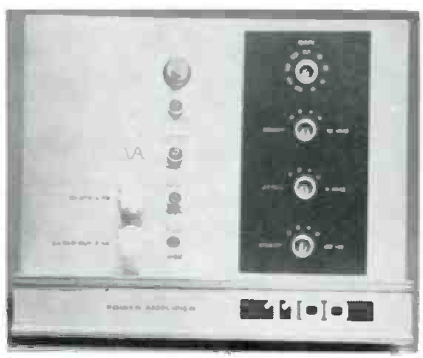
Mfr: Electro-Sound Inc.
Price: $35.00 (tire only: $8.50)
Circle 31 on Reader Service Card

MONITOR AMPLIFIER



• Seventy-five watts in a plug-in configuration make this new monitor amplifier ideal for control-room and sound-reinforcement applications. Special circuitry is designed to assure dependability and includes protection against accidental overload or output shorting. The model 1100LU is equipped with an input transformer, preamplifier, and equalizer. The equalizer provides up to 6 dB boost at 40 Hz, up to 6 dB attenuation at 4 kHz, and up to 6 dB of boost at 12 kHz. There are 2 dB per step switches for all three frequencies. Other specifications include the rated output of 75 watts (+42 dBm) continuous, clipping at over 100 watts mid-band frequencies, ±1 dB response 20-20,000 Hz, and noise is 88 dB below rated output. Tonal harmonic distortion is less than 0.5 per cent at rated output over 30-15,000 Hz, midrange distortion is under 0.2 per cent.
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People, Places, Happenings

• April 17th is the date for the **1969 Midwest Acoustics Conference,** the third annual one. It will be held at Northwestern University and the Orrington Hotel, both in Evanston, Illinois (Chicago suburb). According to **Dr. J. Norton Brennan,** this year's chairman, the participating institution will be the University of Wisconsin, which will provide the technical program for the one-day conference. The program is being organized by **Professor Richard A. Greiner** of the Department of Electrical Engineering, University of Wisconsin at Madison. The Midwest Acoustics Conference is held annually to provide a medium whereby midwest universities can report to industry on their work in various fields relating to audio and acoustics.

• **Anthony N. English** is a recent appointment to the position of vice-president and general manager of **United Recording Electronics Industries.** He will direct all manufacturing, sales, and administrative activities of the company. U.R.E.I. is a manufacturer of professional audio equipment, under the labels **Universal Audio, Waveforms,** and **Teletronix.**

• **Roland Wittenberg** has been appointed vice-president engineering at **Melcor Electronics Corp. Julius Brick,** president of the Farmingdale, N.Y. firm announced that Mr. Wittenberg will be responsible for all aspects of Melcor's product lines. In the past, he has been president of **Dynatran Corp.,** president of **Ameridex Corp.,** and a senior engineer with **Reeves Instrument Corp.** Mr. Wittenberg holds a BSEE degree from Rennselaer Polytechnic Institute and an MSEE degree for Polytechnic Institute of Brooklyn.

• The largest music-recording control console yet built by **Rupert Neve and Company Limited,** has been installed in the New York studios of the **Vanguard Recording Society, Inc.** The Cambridge, England-based firm's console embodies twenty-four input channels, sixteen output groups, four echo groups, and two fold-back groups. The console also has comprehensive four-speaker monitoring and re-mix for sixteen tracks. This is their second installation for Vanguard, the first being a sixteen-channel mastering console. Neve has recently completed two consoles installed in Spain and is currently working on units for re-recording desks for **Pye Records, Ltd.,** and **Associated British Pathe Ltd., both of England.**

• **Shure Brothers, Inc.,** has announced the promotion of four executives.

V. F. Machin has been promoted from his former position of vice-president in charge of marketing to the newly-created post of senior vice-president in charge of marketing and manufacturing. In his new position, he will be directing all Shure manufacturing operations, as well as continuing to supervise all marketing activities for the company.

Raymond Ward, formerly distirutor sales manager, has been named vice-president in charge of sales, also a newly-created post. He will now have responsibility for the direction of all domestic and international sales activities for the company.

Replacing Raymond Ward as distributor sales manager is **Roger Ponto** who will direct all sales activities conducted through domestic distributors of Shure products.

Tom Burks, formerly sales office manager has been named marketing services manager. In this position he will be responsible for sales activities to catalog accounts, public-address equipment manufacturers, and acounts in the educational market.

In announcing these promotions, **S. N. Shure,** company president described them as "important changes designed to advance our program for continued growth and business expansion."

Robert W. Byloff, a leading innovator in color television broadcasting and president of Reeves Video Division of Reeves Broadcasting Corporation, died January 18th, of a heart attack. He was 48.

Mr. Byloff was a member of the joint RCA/NBC systems engineering team that began the first successful broadcasting of color television for NBC.

He joined Reeves in 1959 to build a video facility that has since grown to become the largest independent video services company in the world. Techniques developed by Mr. Byloff while at Reeves have become standard operating practices in the entire television industry.

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The sound of Koss ESP-6 headphones is as new and unique an experience as the very first time you heard the sound of Koss Stereophones. Remember the dramatic and expansive sound? Like nothing you ever heard before. And now Koss has done it again.

Through the development of the world's first electrostatic driver element for headphones, Koss engineers have created a sound lively in its own reality; smooth and relaxing in character. It is a sound beyond the limits of ordinary voice coil-and-cone-type driver elements. It is a sound reachable only through electrostatics.

Through the development of practically weightless, self-energized diaphragms never before possible for headphone uses, Koss engineers have eliminated the moving cone driver element to achieve a frequency response obtainable in the past only through the use of complex electrostatic speakers like the Koss-Acoustech X system. And only now have Koss engineers been able to employ these techniques for headphone listening. And to prove it, an individual machine-run response curve comes packed with each headset.

Dramatic Koss ESP-6 Electrostatic Stereophones feature a dynamic level indicator in each earcup to protect against overload when signal reaches very high sound pressure levels. Constructed with earpieces of sturdy, high impact plastic, coiled cord extending to 10 feet, comfortable foam rubber headband, and fluid-filled ear cushions for effective seal, the ESP-6 will withstand rugged use required by professional recording. Yet uniformly reliable and superb sound reproduction need not be sacrificed.

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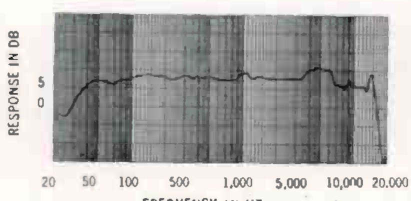

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The biggest changes in the RE55 are mechanical. For this microphone is even more rugged than the 655... long known as one of the toughest in the business. There's a solid steel case and new, improved internal shock mounting for the RE55. Plus a satin nickel finish that looks great on TV long after most microphones have been scarred and scratched almost beyond recognition.

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