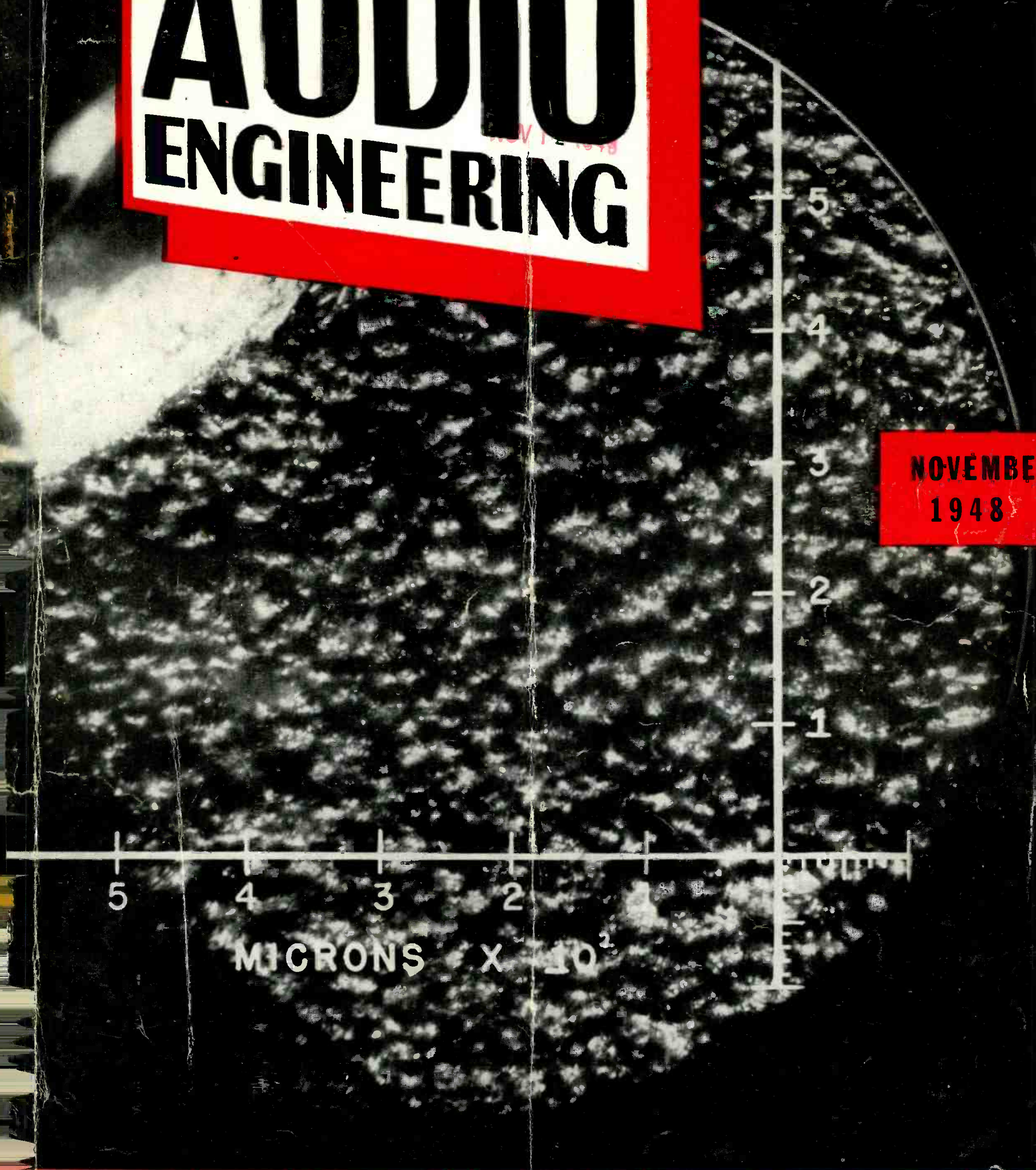
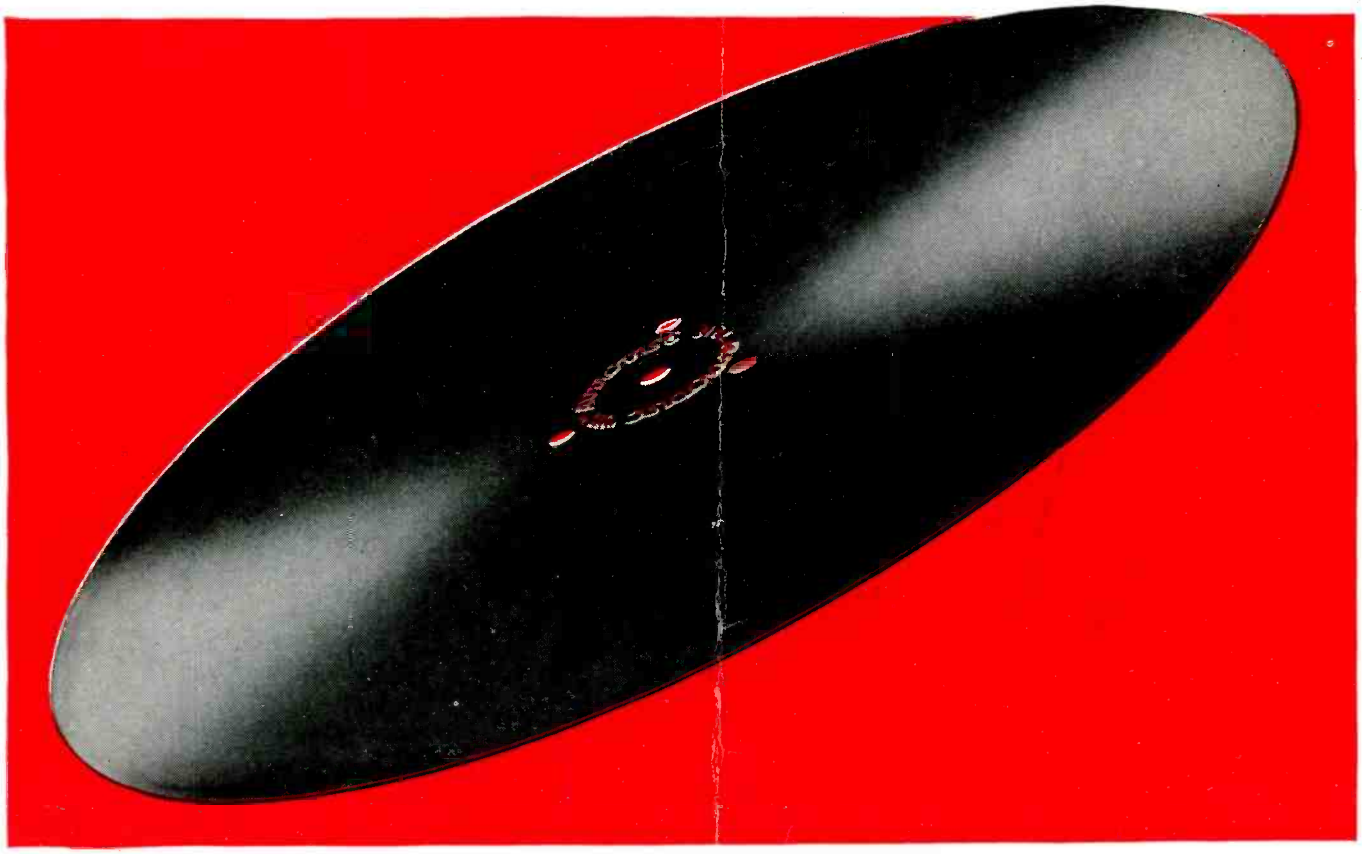
AUDIO ENGINEERING

NOVEMBER 1948



THE JOURNAL FOR SOUND ENGINEERS

FOR FINER ALL-WEATHER RECORDING



Now **audiodisc*** *lacquer provides permanent resistance to humidity*

Excessive humidity has long been one of the industry's major problems—both to the manufacturer and to the recordist. Humid conditions in factories have frequently held up production and caused excessive spoilage. Also, discs which have absorbed too much moisture make poor recordings. The noise level increases progressively while recording and the cut gets greyer and greyer.

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1. The Improved Audiodisc Formulation has eliminated all production difficulties due to excessive humidity. During the past summer no trouble was encountered, even with humidity as high as 90%.

2. Countless Tests in our "weather room" have proved the new Audiodiscs to be remarkably resistant to moisture absorption. Discs subjected to a temperature of 90° at 80% to 90% humidity for many weeks show no increase in noise level while recording. Ordinary discs, under the same conditions, show a noise level increase of from 15 to 25 db. The most conclusive proof of all, however, has come from the field—for during the past summer, one of the most humid on record, our customers have reported no difficulties in recording or reproduction due to humid conditions.

3. This "Weather-Proof" Feature has been achieved without any basic change in our lacquer formulation. Recordists will therefore continue to note the outstanding qualities in recording, playback and processing which have made for Audiodisc leadership.

•

This improved humidity-resistant lacquer is now used on all Audiodiscs. It is your assurance of finer, all-weather recording—with the same consistent, uniform quality which has characterized Audiodiscs for a decade.

*Reg. U. S. Pat. Off.

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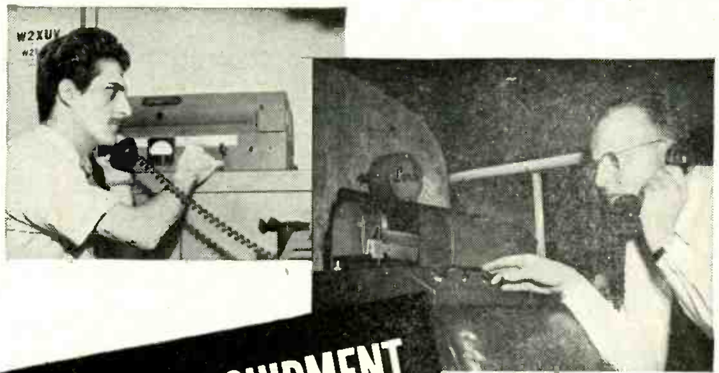
they speak for themselves **audiodiscs**

Sitting before the sound-photo machine in the New York Daily Mirror radio car, John Reidy, the Mirror's chief photographer, advises the Mirror office (extreme left) that a photo is coming through.

The heart of the sound-photo receiving equipment at the office is Sylvania's Glow Modulator Tube, Type R1130B. A pin-point of light emitted by this tube is focused on a sheet of photographic paper attached to a revolving cylinder. As the cylinder revolves, the photograph is faithfully reproduced as it is being broadcast!



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LINK FM SETS—SYLVANIA-tubed, too—transmit on-the-spot photos to paper's home office!

Pioneer in the radio car field, the New York Daily Mirror has made excellent use of its sound-photo apparatus. Equipped with Link Radio Communication units, the Mirror car has been in a position to scoop other newspapers by radioing pictures taken on the spot as soon as they are developed in the mobile photo-lab.

In these units, 36 highest quality Sylvania tubes, ranging from Lock-Ins to standard glass and GT tubes, help insure trouble-free operation of this ultra-modern method of photo-news reporting! For information on Sylvania tubes, see Sylvania Distributors, or write Radio Tube Division, Emporium, Pa.

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Established 1917

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NOVEMBER, 1948

Vol. 32, No. 11

COVER

Photomicrograph of the surface of a treated germanium crystal amplifier unit. Photographed by Winston Wells especially for AUDIO ENGINEERING.

AUDIO ENGINEERING (title registered U. S. Pat. Off.) is published monthly at New York, N.Y., by Radio Magazines, Inc., J. H. Potts, President; S. R. Cowan, Sec'y-Treas. Executive and Editorial Offices at 342 Madison Avenue, New York 17, N. Y. Subscription rates—United States, U. S. Possessions and Canada, $3.00 for 1 year, $5.00 for 2 years; elsewhere $4.00 per year. Single copies 35c. Printed in U. S. A. Entered as Second Class Matter July 29, 1948 at the Post Office, New York, N.Y., under the Act of March 3, 1879.

The 1304 is *TOPS!*

TOPS in
Reproduction quality—operating convenience

REPRODUCTION QUALITY? The Western Electric 1304 Set combines the 109 Type Reproducer Group with its extremely low intermodulation distortion and a unique new driving mechanism (shown in Fig. 1) that cuts flutter to a value lower than many standard recording equipments.

Even the small amount of flutter originating in the mechanism's simple gearing is damped in the novel filter of Fig. 2. Result: a flutter level, *including wow,* of less than 1/10 of 1% *at both 78 and 33-1/3 rpm.*

The platter has been isolated from the sources of rumble by means of the drive isolation coupling (Fig. 4), the fabric belt, and by mounting the entire drive mechanism on rubber vibration mounts (Fig. 3). The large drive pulleys, the use of large belt wrap around, and an adjustable spring loaded idler pulley prevent belt slippage problems.

OPERATING CONVENIENCE? Speed change-over at the throw of a switch. Acceleration to 33-1/3 rpm in *1/9 revolution*—to 78 rpm in *less than 1/2 revolution.* Rapid slowdown — no overdrive — convenient flange on platter for quick stopping.

And playing time variation is *less than ± 2 seconds in 15 minutes!*

Scientific placement of elements facilitates operation. An annular groove in the platter makes it easy to grasp edge of 10- or 12-inch records. 706A Guard provides automatic arm rest, keeps stylus from dropping on panel, catching in turntable felt, or striking edge of revolving platter.

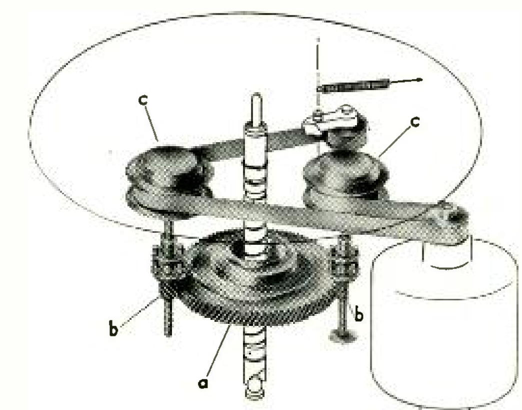


FIG. 1

A single helical ring gear (a), is permanently meshed with two pinion gears (b), each driven by an overriding clutch (c). Reversing direction of motor rotation disengages one overriding clutch, engages the other to change platter speed. Permanently meshed gears eliminate possibility of flutter caused by wear of engaging and disengaging.

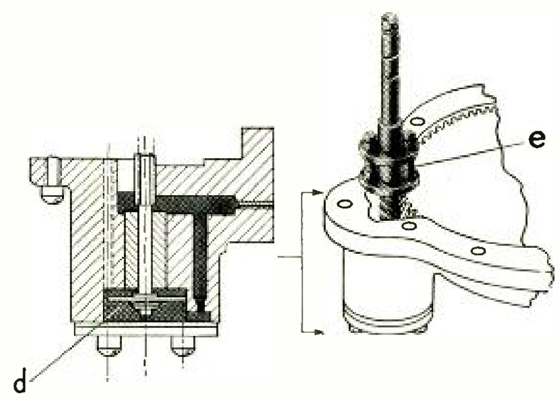


FIG. 2

As shown in cut-away view, a coupling (e) allows each pinion and associated shaft to move a short distance along its axis. The bottom of each pinion shaft projects into an oil-filled chamber (d) for damping axial motion. Because of the helical gearing and the high inertia of the turntable platter, irregularities in the drive tending to cause flutter are taken up and damped in axial motion of the driving pinion.

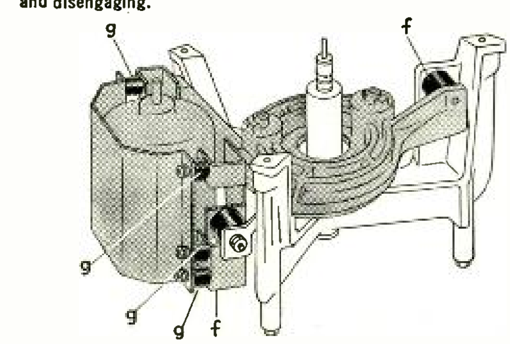


FIG. 3

The entire mechanism, including motor, floats separately from frame and platter shaft on three large rubber mountings (f). Motor, in turn, is isolated from the gear system by smaller rubber mountings (g) and the use of belt drive.

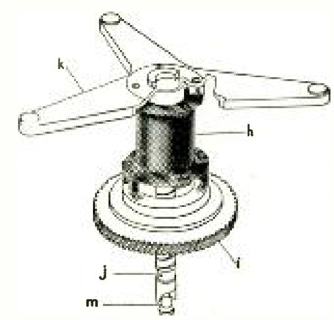


FIG. 4

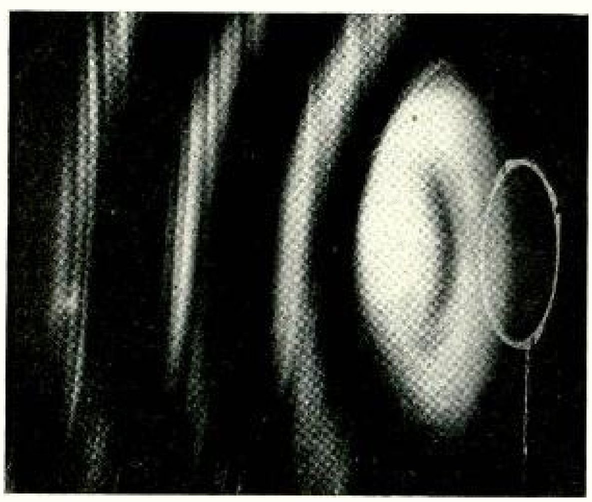
Drive isolation coupling (h), provides the only connection between driving gear (i), platter shaft (j) and platter support (k), completing the separation of drive mechanism from platter. This coupling—very rigid in rotational plane, highly flexible in all others—transmits the driving motion, but isolates the rumble-causing motion. Platter and support ride on a hardened single ball thrust bearing (m).

EDITOR'S REPORT

COVER PIX

PERHAPS you may have wondered why the remarkable cover photographs by Winston Wells which we have used in our last three issues have been presented with so little editorial comment. Here's the story: We had planned on running a cover picture showing a sound wave traveling out from a loudspeaker and, because this is strictly a job for a top-flight audio engineer rather than a commercial photographer, Winston Wells volunteered to undertake it. Two methods were available to produce sound waves in a form which could be photographed, the *schlieren* (spark) method, and one described by Bergmann, by which a smoke pattern is formed in air at high relative humidity.

After considerable thought, Wells decided to use the latter method because, while extremely difficult to set up with the facilities available, it promised far more spectacular results. Two days were spent in preparation,



but when everything was ready it was found that the sound power required to produce a pattern was so great that other occupants of the building complained and it had to be abandoned. By this time we were perched on our deadline, minus the desired cover photo. But the resourceful Mr. Wells cooked up another method, one never before used so far as we have been able to discover, yet so simple that a mere explanation of the method was sufficient to convince your editor that it was unquestionably feasible. But, because the photograph had not yet been made at the time the contents page describing it had to go to press, our description was pretty vague. The photograph, which is again reproduced on this page, represents a 3300-cycle note traveling out from a small speaker. The separation between the waves is thus of the order of four inches. The signal level used was only a few milliwatts and the same technique can be applied in determining the slip stream pattern of an airplane propeller, in plotting standing wave patterns in a room, leakage patterns around shielding, either at r-f, microwave, or audio frequencies, etc. So many things can be done by this method that Mr. Wells has applied for a basic patent, and will present a complete story about it in an early issue.

The cover picture for this month, also by Winston Wells, likewise required a great deal of work. It shows a germanium crystal which has been chemically treated to improve its characteristics as an amplifier, and should be of extraordinary interest to crystallographers, as well as to audio engineers, for whom this new development holds great potentialities. Recent investigations show that it presents great advantages in transconductance and power efficiency over vacuum tubes now used. From the scale shown, the degree of magnification can readily be calculated. In photographs of this type, crystallographers prefer flat lighting for maximum detail, but for spectacular effect, sharp contrast and brilliant highlights are desired. Note that Wells has produced both.

Our next scheduled cover is another presentation of a sound pattern, which undoubtedly will hereafter be called a "Wells" pattern, showing a unique application of this technique.

PERSONAL MENTION

Just a few days ago Dr. W. W. Wetzel of the Minnesota Mining and Manufacturing Company joined me for lunch and discussed, among other things, some of the new developments in tape recording. His company, as you know, is producing excellent magnetic tape as well as the sticky kind. Dr. Wetzel has promised us more articles on magnetic recording in the near future. One which deals with an extremely important point is now awaiting company clearance.

Mr. W. S. Barrell, managing director of the recording subsidiary of E.M.I. in England arrived a few days ago. His organization produces HMV records, famous throughout the world. They record flat, at the outside turn, up to 20,000 cycles as a matter of course and their pressings lose very little. Mr. Barrell addressed a meeting of the Audio Engineering Society in New York and told of the work of the British Sound Recording Society, of which he is president. An arrangement for an interchange of papers between the two Societies has been made.

—*J.H.P.*

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— Letters —

Sir:

Mr. Redman's recent letter proposes a solution to your desire to include in AUDIO ENGINEERING material on television without conflicting with present policies. Of course, any journal exists because of the readers it attracts; and if broadcasting engineers predominate among yours, then the journal should certainly be on an everything-except-r.f. sound broadcasting basis, thus including, audio engineering, television, facsimile, wired music and wired television.

For my part, I believe that you can attract enough readers among the "informed laity," as musicians, public address engineers and recordists, to keep AUDIO ENGINEERING expanding with present editorial policies.

Vincent Salmon,
6220 S. Moody Ave.
Chicago 38, Ill.

We agree. *Ed.*

Sir:

I just wanted to let you know that I agree with the first two paragraphs of Mr. James F. Redman's letter (September issue, page 4) that AUDIO ENGINEERING is doing a most outstanding job.

One never gets the impression that certain articles are used just to fill out the book. Each presentation is informative, helpful, and authoritative. As far as publication is concerned, the audio field—surely the electronic field of greatest interest among the largest number of people—has finally come into its own.

Your September editorial especially appealed to me. I can probably push a slide rule about as fast as average and I keep a good supply of pencils and paper on hand, but when I read a radio magazine I want to keep a cigarette in my hand, not a log table. When the time comes to apply the design information the formulæ will be used, but before that, an article must be readable. Of course, when the time comes to apply the data, RADIOFILE will also be useful in finding it (advt.).

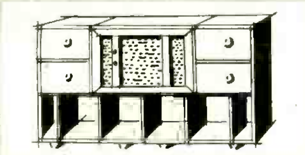
I want to compliment you, in addition, on the makeup of the magazine. The choice of type faces is excellent and the use of color and bleeds extremely tasteful. The Audio Design Notes are very useful (though sometimes, as in September, the audio application seems to get lost) and Canby's column is always entertaining, though not necessarily particularly informative.

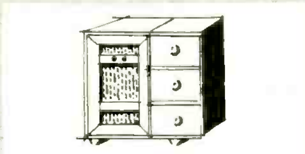
I might as well get in on the TV controversy, too. It might be true, theoretically, that adding a TV section to the magazine would give the reader more information for his money. But I believe it would detract from the flavor of the magazine. We now feel that AUDIO ENGINEERING is the audio oracle. AUDIO ENGINEERING would, in other words, lose its distinctiveness as a specialist's publication.

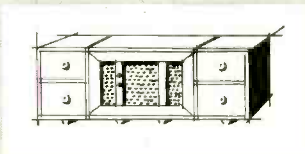
In short, you have an excellent publication which is doing a fine job. Please don't dilute it with other material.

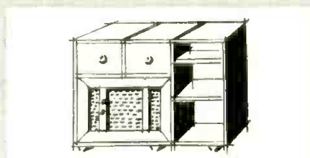
Best wishes for continued success.

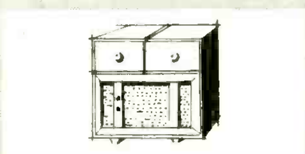
Richard H. Dorf,
255 W. 84th St.
New York City.

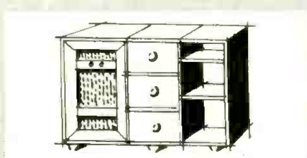


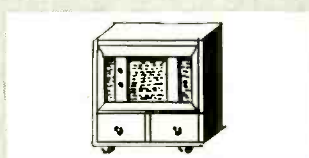


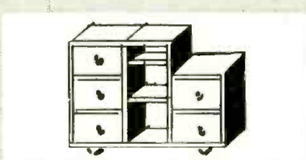


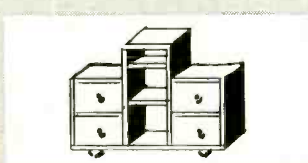












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Audio System Design Fundamentals

HOWARD A. CHINN*

First of a series for audio engineers.

THE DESIGN of audio facilities for broadcasting, recording, or for sound reinforcing (public-address) applications divides itself naturally into two parts; (*a*), the design of the individual components and (*b*), the design of the complete system utilizing these components. Many of the essential components, such as microphones, amplifiers, volume controls, loudspeakers, volume indicators, etc., are readily available in several grades and in various forms and sizes. Thus, except for certain specialized applications, such as are found in the more elaborate and meticulous audio installations, it is usually possible either to use commercially available components that fulfill the requirements or to design them readily by employing conventional methods. Under these circumstances, the design of audio *systems* themselves can be approached on the assumption that most of the components required are readily available.

The most complex audio installation can generally be broken down into smaller units. For example, in a broadcasting plant the audio facilities may be segregated into studio, portable, program distribution, building and office monitoring, sound reinforcement, recording and allied systems. The basic principles governing the design of audio systems apply equally well to broadcasting, to sound recording (phonograph records, transcriptions, motion picture), to public-address, wired-music and other applications.

The most important considerations in any system are the operating requirements that must be fulfilled. Establishment of these determines the elaborateness of the installation, the grade of equipment and workmanship required, the physical layout, and, of course, the over-all cost. The most extensive installation can be successfully formulated, however, by faithfully following certain fundamentals. From an electrical viewpoint these include establishment of adequate signal levels, avoidance of overloading, observance of proper impedance matches and a systematic method of portraying the basic circuit design. From a physical layout viewpoint, consideration must be given to convenience and ease of operation, visibility and legibility of controls, lines of vision into adjacent areas and similar considerations. This paper concerns itself only with the first of these two broad categories, namely, the circuit fundamentals.

Program Levels

The prime consideration in the design of any high-quality audio system is the establishment of proper signal levels at all points in the system. This

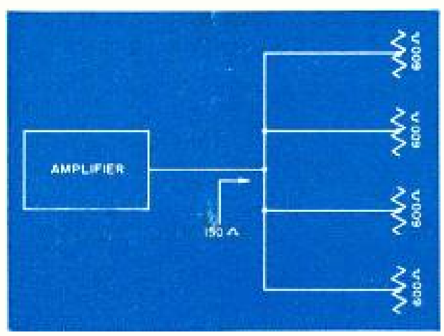

Fig. 1. In this diagram the amplifier output load is 150 ohms, but the load facing each section is only 86 ohms.

stems from the need (*a*), for obtaining the required audio levels for the purpose in hand, (*b*), for avoiding overloading in so doing, and (*c*), for maintaining the desired signal-to-noise ratio. Knowing the signal intensities required, overloading can be averted simply by the proper choice of components. The successful achievement of the required signal-to-noise ratios, on the other hand, hinges upon the basic system design and the skill with which its actual realization is undertaken.

The problem becomes more and more acute as higher and higher quality systems are demanded. High-quality systems imply, among other things, wider frequency bands and higher signal-to-noise ratios. On the other hand, wide-band electronic devices usually entail lower output level (microphones), lower sensitivity (microphones, loudspeakers and phono pickups) or lower gain (amplifiers). As a matter of fact, the output levels of present-day wide-band microphones and pickups are often so low that, where involved, they may be the limiting factor in achieving a high signal-to-noise ratio. Under circumstances such as these, it is obviously necessary to avoid any degradation of the signal-to-noise ratio because of poor system design.

System Noise Levels

Audio system components, when properly designed and carefully installed, need not contribute any appreciable noise to a system other than that caused by thermal agitation. In other words, using the components that are available today, no problems need be faced because of noise caused by hum, microphonics, or poor contacts.

On the other hand, thermal noise is a limitation that is always present. It establishes the limit below which the circuit noise cannot be reduced. By the same token, it automatically establishes the signal level that must be maintained in order to achieve a specified signal-to-noise ratio. Similarly, it determines the lower limit of useful output from microphones and pickups.

As is well known, the thermal agitation voltage, *e*, of a circuit element is given by the following equation:[1,2]

$$e = \sqrt{1.605 \times 10^{-20} \times \Delta f \times R} \quad \text{(volts)}$$

where Δf = bandwidth in cps

R = resistance of element in ohms

Audio system levels are usually expressed in terms of either "dbm" or "vu", depending upon which term is applicable.[3] It is therefore convenient also to express noise on a similar basis. Assuming a band-width of 15 kc and carrying through the calculation, the thermal noise level in a circuit having matched, pure resistance source and load impedances is found to be equiva-

*Chief Audio-Video Engineer, Columbia Broadcasting System, Inc., N.Y.C.

1. J. B. Johnson, "*Thermal Agitation of Electricity in Conductors*" *Physical Review, Vol.* 32, *No.* 1. *pg.* 97 (1928)
2. *H. Nyquist, "Thermal Agitation of Electric Charge in Conductors" Physical Review, Vol.* 32, *No.* 1, *pg.* 110 (1928)
3. *H. A. Chinn, "VU vs. DBM"*, AUDIO ENGINEERING, *Vol.* 32, *No.* 3, *p.* 28 (*March* 1948)

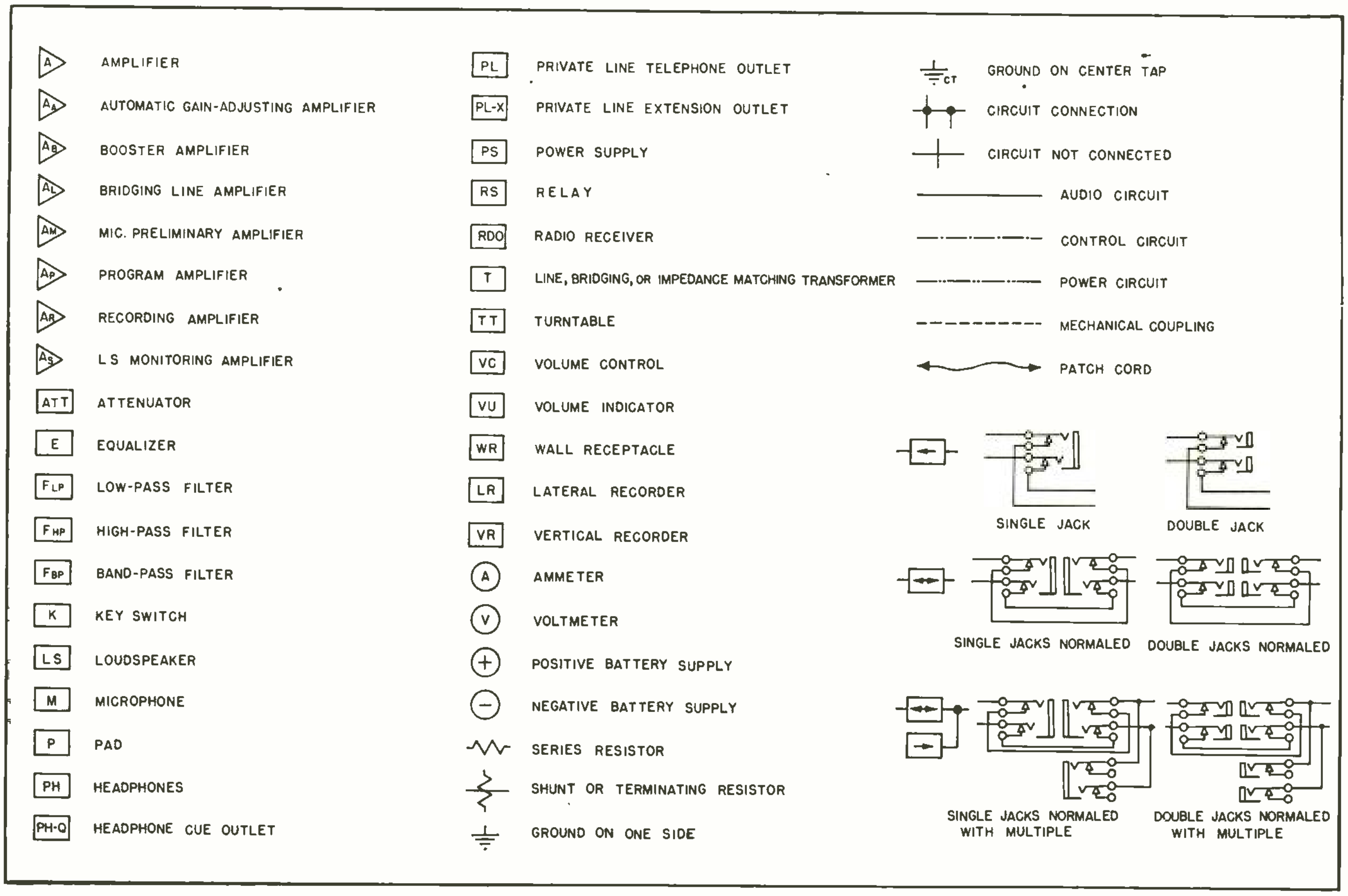

Fig. 2. Standard symbols used in audio engineering drawings.

lent to —129 vu. This is the theoretically lowest noise level that can be achieved in any circuit having a 15 kc band-width. In practice, with careful design, this figure can often be approached within a few decibels. It can never be equalled, however, since there is always some residual noise present in an actual system. In any event, knowing the lowest possible noise level, the minimum signal level required to achieve any desired signal-to-noise ratio is immediately known.

For example, assume a 60-db signal-to-noise ratio is required. Under these circumstances the signal level can never be permitted to fall below —69 vu. Since, in practice, the noise level will always be somewhat higher than the theoretical minimum, some safety factor should be applied. A typical good engineering practice would be never to allow the signal level to fall below —60 vu. In some services the signal-to-noise requirements may not be so stringent as the example cited. In any case, the basic principle outlined will apply.

A simple rule is therefore established. If, at any point in a system, the loss entailed by the use of some device (e.g., a mixer control, a matching network, an equalizer, a filter, or a bridging coil or pad) will cause the level of the desired signal to fall below the permissible minimum, amplification must first be provided to avoid degradation of the signal-to-noise ratio. Once the signal-to-noise ratio of an audio system suffers, it cannot usually be reclaimed since both the signal and the noise are always amplified equally. That point in the system where the signal-to-noise ratio is the lowest is the controlling one—it is the weakest link in the chain and a stronger link further on can do nothing to overcome the basic weakness.

Overloading

The avoidance of overloading in audio system design simply requires the proper choice of components. As has been detailed elsewhere[3], a 10-db safety factor is usually allowed between the sine-wave level that corresponds to the maximum permissible distortion and the safe level for program material. That is, if a given component is capable of handling a maximum sine-wave level of +20 dbm, without exceeding specified distortion limits, it may be safely used with program levels of +10 vu. To be sure, greater factors of safety may be used but they are not generally economically justifiable.

In any discussion of overloading in an audio system, the capabilities of the amplifiers are usually given prime consideration. It is equally important, however, to recognize that other units, particularly equalizers, filters, matching and bridging transformers, loudspeakers, and recording heads also have their limitations. The response, impedance and/or distortion as a function of frequency may change markedly with level in some of these units. Their characteristics at both high and low operating levels should always be investigated.

Impedance Matching

Rigorously speaking, except for simple elements such as a fixed resistance attenuation pad or a terminating resistor, few audio components exhibit abosolutely constant input or output impedances at all operating levels and throughout the audio spectrum. Furthermore, the impedance of some devices may be a function of the adjustment of controls (e.g., ladder attenuators, variable equalizers, and adjustable wave filters).

Fortunately, except for filters and some types of equalizers, the impedances of many components are reasonable, constant and uniform over at least a fair portion of operating range. It is the impedance values that obtain in these regions (where, incidentally, the phase angle is usually small) that are

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[3] *Loc. cit.*

generally specified as the nominal impedance of the device in question.

Most audio components are designed to operate from a source impedance and into a load impedance that is a pure resistance of finite value. As a matter of fact, the performance of audio equipment is usually specified and measured on this basis. It is therefore good engineering practice to provide the proper resistive source and load impedances to insure that the completely assembled system will perform as predicted from the characteristics of the individual components. The average audio system lends itself readily to this treatment since volume controls (other than ladder networks and simple potentiometers), differential networks, attenuation pads, terminations and other constant resistance devices are almost universally interposed between the other elements of an audio system.

Under these circumstances the problem of providing the proper source and load impedance for each component simply requires the observance of two rules:

a. Interpose constant resistance elements between components whose input and output impedances do not provide the proper load and source impedances.
b. At a junction point where more than two components meet, scrutinize the impedance relations from the viewpoint of each branch of the circuit.

It is this latter requirement that often trips the novice. For example, consider an amplifier that is to feed four identical loads, each of which is intended to operate from a source impedance equal to its input impedance (*Fig.* 1). Since a numerical example will quickly illustrate the point, assume each load has an input impedance of 600 ohms. The four in parallel will result in a net load impedance for the source of 150 ohms. If, now, the generator has an output impedance of this value and is designed to accommodate a like load impedance, all seems well.

This is not the case, however. Consider the impedance from the viewpoint of any given load. Looking back from any given load, one sees the 150-ohm output impedance of the amplifier in parallel with which there are the other three 600-ohm loads. In other words, the net impedance that each load faces is just under 86 ohms—far from the 600-ohm value that the conditions of the example specified. The solution to the problem, of course, is the use of a differential matching network—but that is a subject in itself and will not be treated here.

The principle cited must also be observed in the reverse situation—that is, when several sources are to be combined into a single channel. A very common example of this kind is in mixer circuits where a number of program sources are brought together into a common channel. Differential networks are again indicated if impedance mismatches are to be avoided.

Block Diagrams

A very effective way to practice the four principles that have been expounded is to make use of a systematic method for portraying the circuit that is being designed. Many years ago[4] a method was evolved which has withstood the test of time and is worth the repeating. Basically, it is nothing more than the well-known block diagram. It is the embellishments on this diagram, however, that make it so useful.

First, a series of standard symbols are used (*Fig.* 2.) For simplicity and speed in drafting practically all the symbols are rectangles of uniform size. The letters within the boxes are chosen so as to suggest, insofar as possible, the item being identified. The connotation of each symbol is almost so obvious that once reviewed they require almost no further identification.

Next, the conventional block diagram is adorned with those features shown in *Fig.* 3 that are applicable to the case in hand. It is this supplementary information that really establishes the value of method of presentation being described. Taking up the items in the

[4] *H. A. Chinn, "Broadcast Studio Audio Frequency System Design," Proc. I.R.E., Vol.* 27, *No.* 2 (*February* 1939)

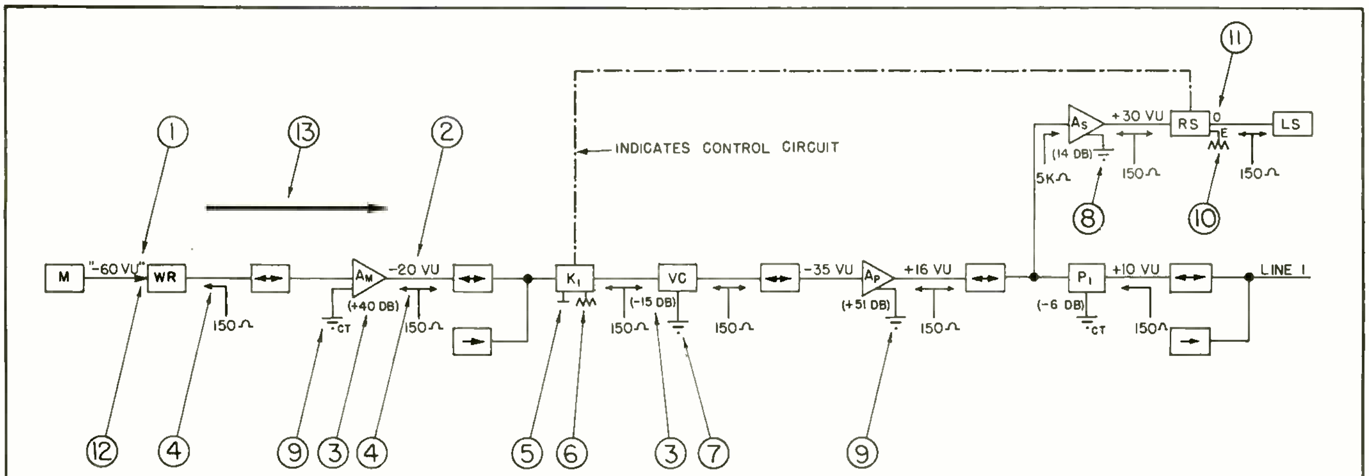


Fig. 3. Application of symbols in a typical block diagram showing signal levels.

1 QUOTATION MARKS INDICATE LEVEL SHOWN IS "EFFECTIVE" VALUE.
2 INDICATES PROGRAM LEVEL AT DESIGNATED POINT IN CIRCUIT.
3 INDICATES GAIN OR LOSS IN ASSOCIATED UNIT OF EQUIPMENT.
4 INDICATES CIRCUIT IMPEDANCE IN DIRECTION INDICATED.
5 INDICATES TERMINATING SHORT CIRCUIT ON INPUT WHEN KEY IS IN OFF POSITION.
6 INDICATES TERMINATING RESISTOR ON OUTPUT WHEN KEY IS IN OFF POSITION.
7 INDICATES COMMON SIDE GROUNDED.
8 INDICATES ONE SIDE OF OUTPUT WINDING GROUNDED.
9 INDICATES CENTER TAP GROUNDED.
10 INDICATES RELAY COIL ENERGIZED IN THIS POSITION.
11 INDICATES RELAY COIL OPEN IN THIS POSITION.
12 INDICATES CABLE CONNECTOR.
13 INDICATES DIRECTION OF FLOW.

order in which they are illustrated:

Item 1: Microphones are intended to work into what is essentially an open circuit (although their output impedance is of finite value), consequently their output levels cannot be rigorously stated in terms of vu (or dbm). For system design purposes, however, a microphone may be considered to have an "effective" output level that is stated in the conventional terms for designating audio levels.

Item 2: Along the circuit, each time there is a change in program level, the new level is shown. This practice keeps the designer constantly informed as to conditions that might lead to overloading on one hand and degradation of the signal-to-noise ratio on the other. The levels shown are generally the peak program levels that may be expected under normal transmission conditions.

Item 3: The loss or the gain, for normal setting of the controls, of each circuit element is indicated. These data are useful in calculating program levels along the circuit and in selecting the particular component that will fulfill the requirements.

Item 4: The internal impedance of the component in the direction of the associated arrow head is indicated. In those instances where applicable, arrows face in both directions showing that the impedances are matched. In others, an arrow faces in one direction only. For example, the microphone has a finite output impedance, but the input of the associated preliminary amplifier is very high.

Items 5 *and* 6: Circuits that are controlled by key switches are often terminated or shorted when the key is in the "off" position. Appropriate symbols for both conditions are shown. Unless otherwise noted, the value of the terminating resistor corresponds to the iterative impedance of the circuit.

Item 7: Often one side of a circuit element is grounded. In fact, it is usually mandatory in the case of T and pi-section attenuators, pads, equalizers and filters. These often have one circuit that is common to both the input and output circuit. When this is the case and this common leg is grounded, the connection is shown attached to the lower side of the block symbol.

Items 8 *and* 9: In some cases one side or the center-tap of an input or an output circuit is grounded. This is indicated by showing a ground connected to the proper side of the block symbol. When the center-tap is grounded, the letters "CT" are added. When unaccompanied by any notation the ground is assumed connected to the appropriate side of the circuit. By showing all grounds on a block diagram, the need for an isolation coil automatically becomes evident as, for example, when a one-side grounded component is to be connected to a center-tap grounded element.

Items 10 and 11: When relays are used to transfer a circuit from one point to another, it is important to know which path prevails when the coil is energized and which when it is open. A good engineering practice is to have all relays at rest (coil circuit open) when "on the air" or when in normal service. When it is possible to follow this practice, a failure of the relay power supply will not interrupt the program transmission. The letter "O" indicates the circuit when the coil is open; the letter "E" indicates the circuit when the relay coil is energized.

Item 12: When a plug is used to connect one component to another, it is shown by an arrow head.

Item 13: Tracing the progress of a signal through a complex system is greatly facilitated by the use of direction-of-flow arrow. Another helpful convention is to arrange the block diagram so that the flow is always from left to right unless otherwise indicated. When, for one reason or another, this direction of flow cannot be adhered to, a flow-arrow should be shown.

Special attention is called to the convention where key switches or relays having multiple inputs or outputs are involved. The connection to the switch arm is shown on one side of the block symbol while all connections to the switch points are shown on the opposite side of the block. Audio circuit connections are never shown to the top or bottom edge of the block, these being reserved for terminating resistors or for battery control circuits, when shown.

LEGEND		
CODE	DESCRIPTION	MFR.
A_M	MIC. PRELIM. AMP.	LANGEVIN 116-A
A_P	PROGRAM AMP.	RCA BA 3A
A_S	L.S. MONITORING AMP.	RCA BA 4A
K_1	KEY SWITCH	WE 2ARN
LS	LOUD SPEAKER	RCA LC-1A
M	MICROPHONE	WE 639-B
P_1	FIXED "H" PAD, 6DB, 150:150 Ω	DAVEN 154
RS	RELAY	WE U-1017
VC	VOLUME CONTROL	DAVEN T-330
WR	WALL RECEPTACLE	CANNON UA3-13
→	JACK	WE 239-E

Fig. 4. Legend used with block diagram.

Legends

Even though standard symbols are used, each block diagram should be accompanied by a legend (*Fig.* 4). The legend serves the purpose of identifying each symbol for those who have only casual contact with them. Equally important, it also provides a complete description, including the manufacturer and type number, of every item.

Conclusion

The basic design of even the most complex and elaborate audio system can be greatly simplified by adhering to certain fundamentals, the more important of which have been described. These include the establishment and maintenance of adequate signal levels as determined by the required signal-to-noise ratios, the avoidance of overloading and the observance of proper impedance matches.

Perhaps most important of all, from the viewpoint of simplifying the work, is the employment of a block diagram containing *all* the essential facts that influence the design. In some estab-

[*Continued on page* 41]

Audio Engineering Society, San Francisco Section

Newly elected officers of the recently formed San Francisco Section of the Audio Engineering Society. Left to right: Myron Stolaroff, Ampex Electric Corp.; Jack Mullen, Bing Crosby Enterprises; Harold Lindsay, Ampex; Frank Lennart, Ampex; Don E. Lincoln, Audiophone; I. R. Ganic, Audiophone, (Chairman); Walter T; Selsted, Pacific Broadcasting; Dick Beck, C. C. Brown, and Ross H. Snyder, KJBS-FM.

Design of a Continuously Variable
AUDIO SIGNAL GENERATOR

BRUNTON BAUER*

This instrument meets the need for a stable, accurate unit for testing high-quality audio apparatus.

OVER THE LAST FEW YEARS audio engineers have tightened up on fidelity requirements so strenuously that one day's laboratory equipment is the next day's commercial amplifier. This state of affairs calls for higher-caliber measuring equipment than, for example, the time-honored audio oscillator. A unit has been needed which could emit a known value of signal at a known impedance level with extremely low distortion. Continuous variability and self-contained design are additional desirable features.

Use of the term *signal generator* for the piece of apparatus illustrated herein implies that a high degree of control and measurement can be exercised over the output signals, and major effort in the over-all design has been applied in that direction. In this generator it has been possible to keep distortion below 0.1 per cent from 50 to 20,000 cps and below 0.25 per cent between 20 and 50 cps. This has been accomplished with a circuit combining a resistance-tuned oscillator with a tuned amplifier that is tracked with the oscillator in such a way as to cause rejection of harmonic voltages while passing the fundamental along to the output amplifier. Arrangement of these sections is shown in *Fig.* 1, the block diagram.

The output amplifier has a negative feedback loop for distortion reduction. The output of this amplifier goes to the output system consisting of a vacuum-tube voltmeter, a three-section attenuator, and a line-matching transformer—a combination permitting the transmission of a signal at known level from a known impedance source. An electronic voltage regulator is included in the power supply to provide steady plate voltages throughout.

All equipment necessary to making complete gain measurements is thus consolidated in one unit—with the exception of the receiving-section indicator. Because of the attenuator arrangement, this needs only to be the simplest indicating instrument or vtvm capable of revealing a change in level. Both this instrument and the meter in the signal generator are eliminated from the accuracy of measurement because they are kept at a fixed reading by adjustment of the attenuator. Typical equipment-test setup is shown in *Fig.* 2.

Output frequency is selected by the combined use of the frequency control and the range switch. The total range of 20 cps to 20 kc is covered in three ranges: *Times* 1—20 to 200 cps; *Times* 10—200 to 2000 cps; and *Times* 100—2000 to 20,000 cps. This control gives complete coverage of the audio spectrum and makes it possible to study any part of it in extreme detail for detection of peaks which might pass unnoticed if the analysis were limited to discrete steps of frequency.

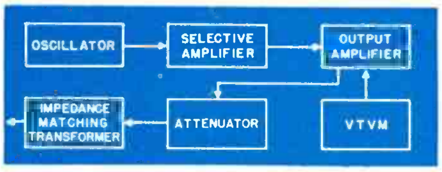


Fig. 1. Block diagram of generator sections.

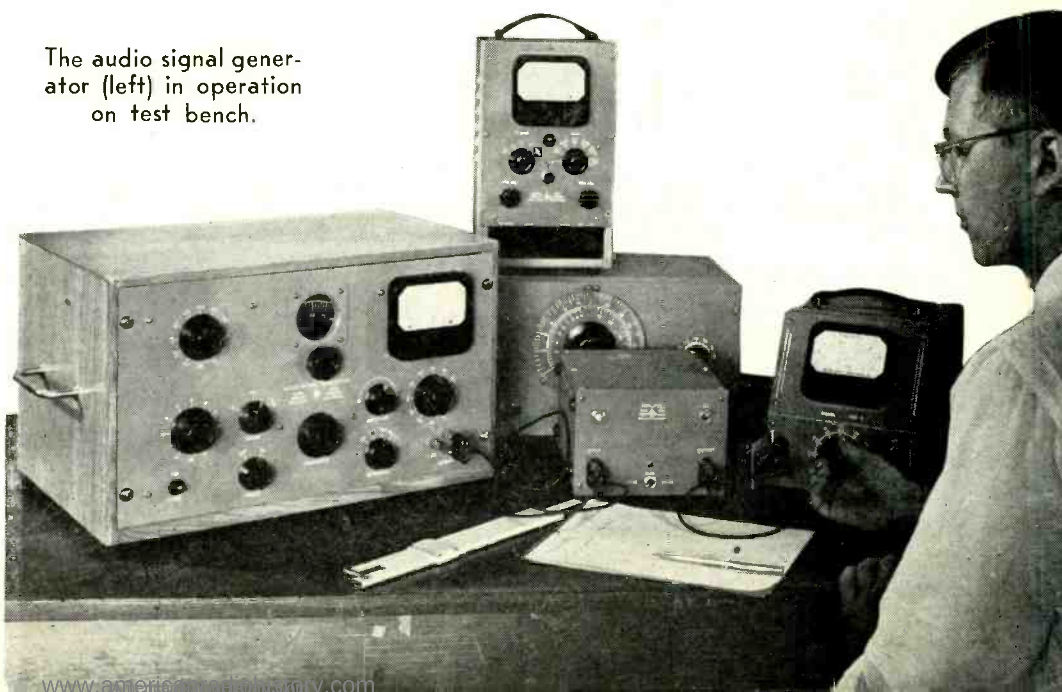
Application of the carrier null method of measuring percentage modulation of an f-m transmitter, for instance, requires this continuous variability. In this measuring technique an unmodulated 200-kc i-f signal containing the full f-m swing of the transmitter is monitored on a communications-type receiver with a several-hundred-cycle beat note while sinewave modulation is applied to the transmitter in increasing amplitude. The beat note will pass through successive nulls at a known series of combinations of modulating frequency and percentage modulation as the carrier disappears in accordance with Bessel-function relationships. Smooth frequency variation is essential in the detection of these nulls.

Fig. 2. Typical test set-up.

Oscillator

The complete signal of the audio signal generator is shown in *Fig.* 3. The oscillator section consists of tubes V_1 and V_2. This section is in the form of a conventional resistance-tuned oscillator. The two stages are resistance coupled and positive feedback through

The audio signal generator (left) in operation on test bench.



Chief Engineer, Hewlett-Packard Co., Palo Alto, Calif.

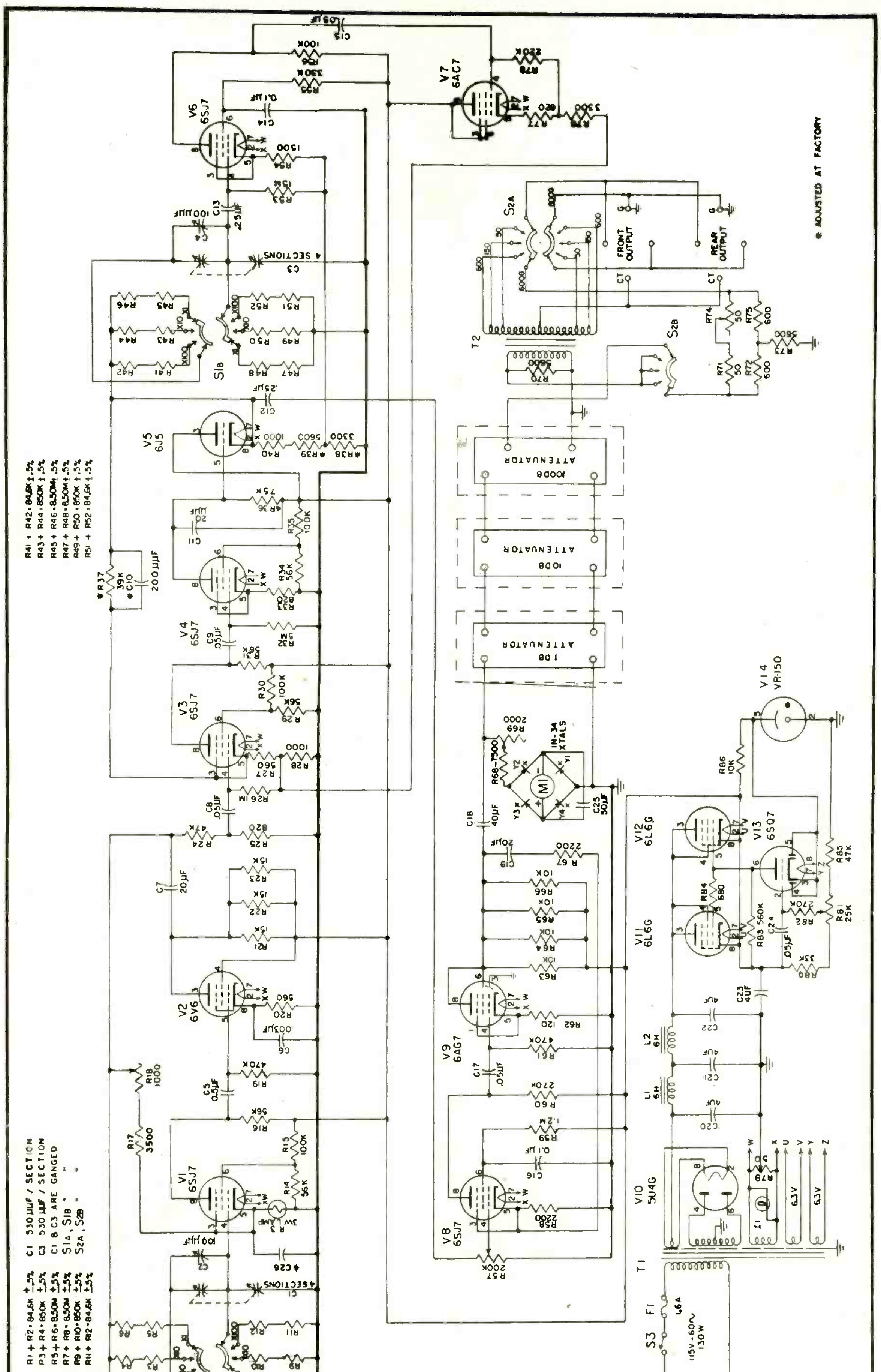


Fig. 3. Complete circuit diagram of audio signal generator. Tuned amplifier includes V_3 to V_7.

the resistance-capacitance network associated with the A section of switch S_1 and the four-section capacitor gang C_1 sustains oscillation and determines oscillation frequency. These circuit elements have low temperature coefficients for over-all stability. Frequency calibration is accurate within 1 per cent.

The oscillator has a negative-feedback loop R_{18}, R_{17}, and R_{13}. Resistor R_{13} is a non-linear element consisting of a 3-watt, 120-v lamp which acts as an automatic amplitude limiter and maintains the system at its optimum operating point. This lamp is standard except that it is subject to the following selection: Voltage from the high side of R_{24}, measured to ground with a high-impedance voltmeter, must read 28 volts $\mp$ 1 volt. It can be adjusted by potentiometer R_{18} with a lamp of the proper characteristics in the socket. An appreciable percentage of production lamps fall outside these limits and cannot be used.

Tuned Amplifier

Electron tubes V_3, V_4, V_5, V_6, and V_7—together with their related circuit elements—comprise the tuned-amplifier section of the signal generator. Tubes V_3 and V_4 are conventional voltage amplifiers. Cathode follower V_5 has, in its cathode circuit, a Wien bridge tuned to a null at the fundamental frequency of the output. It is tuned by another resistance-capacitance network identical with that in the oscillator section described previously. Capacitance is varied by another four-section unit of the tuning capacitor ganged to the first. The second section, B, of switch S_1, ganged to section A, switches the three sets of resistors for range determination.

Tuning this bridge to a null eliminates the fundamental from the following stages without affecting the harmonics present. These are amplified in V_6—a conventional voltage amplifier connected across the center of the bridge—and fed to V_7, a cathode follower, whose cathode circuit is returned to that of V_3. Because the cathode of V_7 is in phase with that of V_3, negative feedback is established at all frequencies except the one to which the bridge is balanced—the output frequency. Output from this section of the generator is taken from the top of the bridge circuit—directly from the cathode of V_5—and has the selective characteristic necessary for tracking with the oscillator and rejecting harmonics to the extent of 10 db or more of distortion reduction.

The tuning capacitor can be seen in *Fig.* 4, a view directly down on top of the open unit. *Figure* 5 shows the tuning section in more detail, the rear deck having been removed for better visibility. The two sections of the tuning capacitor are tracked at the factory and driven by a stainless-steel flexible cable sheathed with beryllium copper for abrasion resistance. The oscillator section of the gang is to the right in both views. The amplifier is on the opposite side.

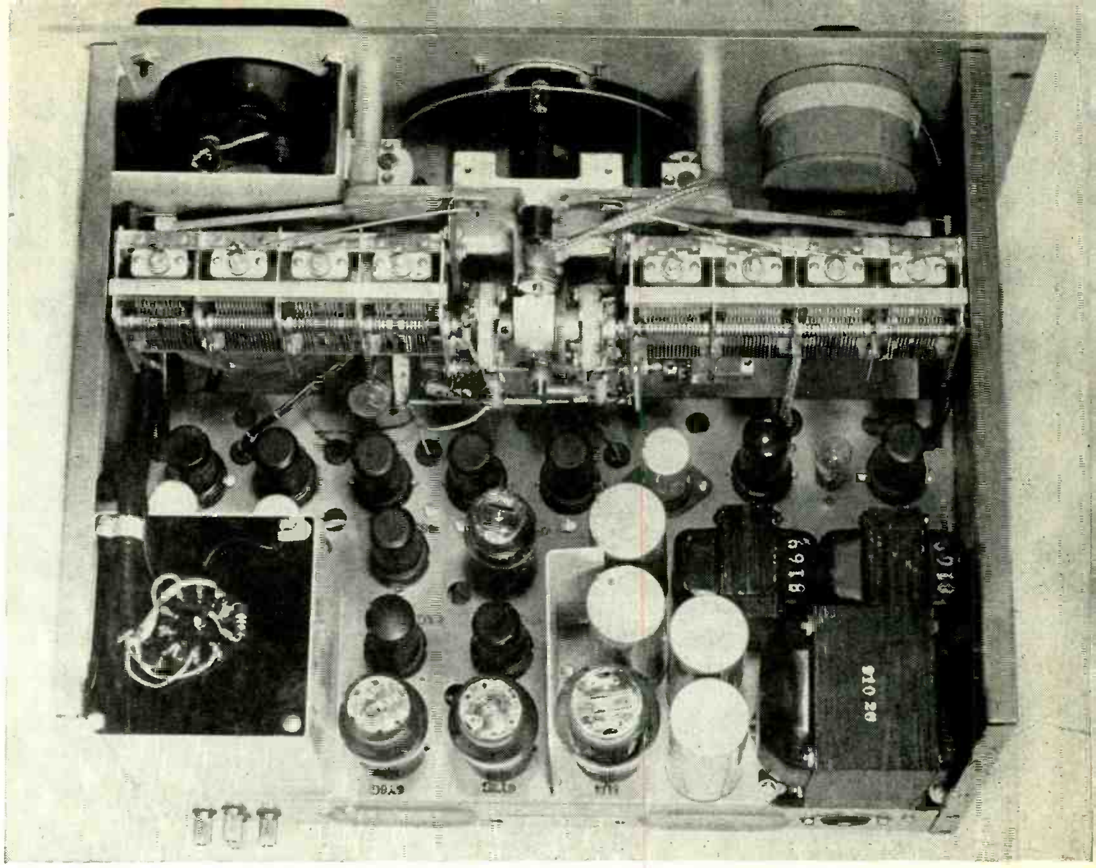


Fig. 4. Top view of signal generator from behind panel. Oscillator section is to the right, tuned amplifier to left.

Output Amplifier

Tubes V_8 and V_9 are the major parts of the output amplifiier, V_8 operating as a driver for output tube V_9. Variable resistor R_{57} in the grid circuit of V_8 serves as an amplitude control. Pentode V_9 is connected here as a triode for working into very low-impedance loads with a minimum of distortion. Negative-feedback loop C_{19} R_{67} from the plate of V_9 to the cathode of V_8 further minimizes distortion.

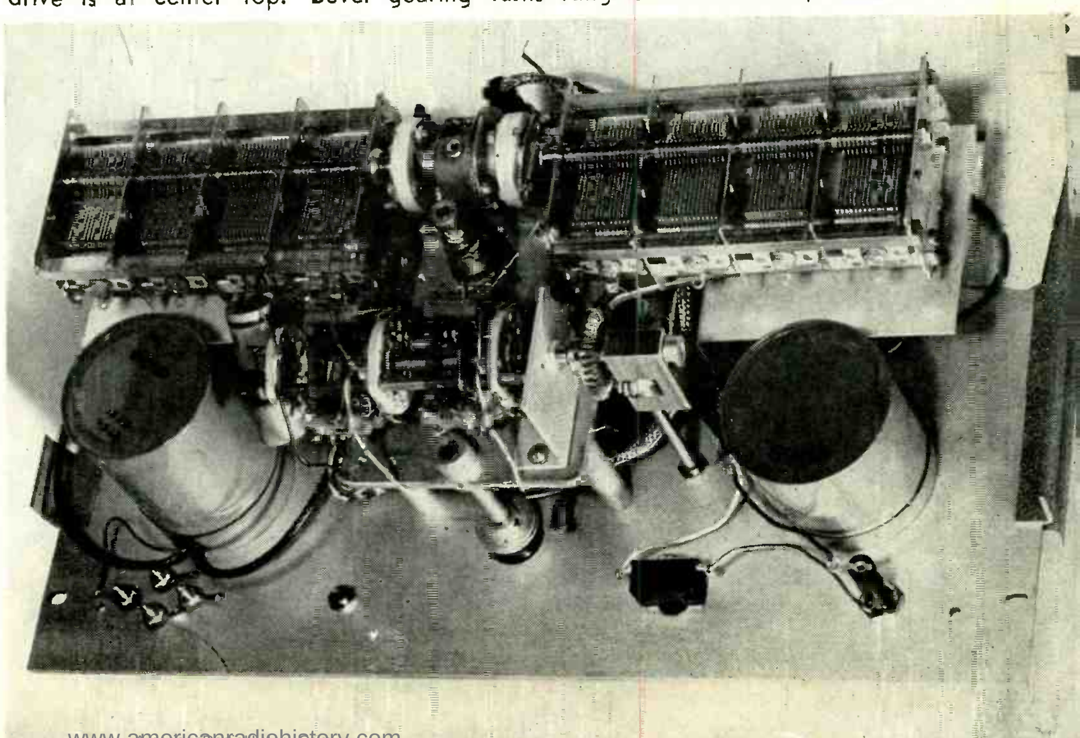
Vacuum Tube Voltmeter

The voltmeter, M_1 is an 0-to-1-ma, 4-in.-square d-c instrument connected into a full-wave bridge rectifier with four 1N34 germanium crystals as rectifier elements. This meter is calibrated both in volts and dbm (zero dbm: 1 milliwatt into a 600-ohm load). Series resistance is provided by R_{68} and R_{69}—the latter providing adjustment for calibration.

There are three sections in the attenuator which gives a range of 111 db in

[*Continued on page* 41]

Fig. 5. Rear view of unit with power-supply deck removed. Capacitor-gang drive is at center top. Bevel gearing turns range-selector switch, center.



Characteristics of Amplifying Crystals

S. YOUNG WHITE*

Performance characteristics of experimental germanium crystal amplifier and oscillator units.

IN ORDER to make efficiently a large number of characteristic curves of amplifying crystals, it is necessary first to design a mounting so that the catwhiskers can be adjusted to an accuracy of about .0001 inch in the *X* and *Y* planes. The *Z* plane, which is the compression of the spring catwhisker, is not so critical. If a compression of .0025 inch is desired, a tolerance of half a thousandth is adequate.

The design shown in *Fig.* 1 meets the requirements rather nicely. As in the previous few articles, we use a mounting that can plug into a miniature tube socket. The crystal, mounted on its .073 diameter rod, which is a quarter-inch long, is held in a brass cylinder, and can be locked with a set-screw. The catwhiskers are procured from IN21A silicon crystal units, as these are longer and straighter than those that come with the IN34, as a rule. The crystal face is forced firmly against two Bakelite blocks, A-A, and the height of these blocks is held to .0001 inch, and thus establishes the *X* axis. Between the blocks is a mica sheet about a thousandth thick, which forms an insulating barrier between the catwhiskers, thus establishing the *Y* axis.

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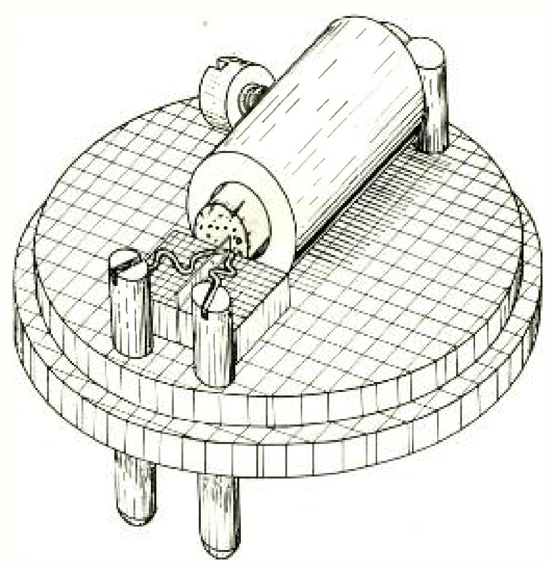


Fig. 1. Experimental mounting for germanium crystal amplifier unit.

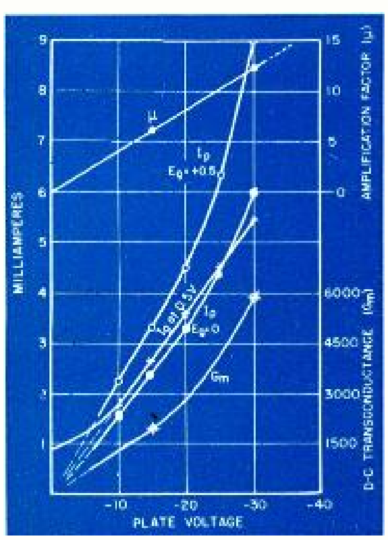


Fig. 2. Static characteristics of "soft" crystal.

In assembly the crystal is retracted .0025 inch by a spacer. A catwhisker is loosely laid in a slot in the pin of the miniature tube base, and forms a compound angle with the crystal face, which causes it to fall in the pocket formed by the barriers and the crystal face. You might be able to visualize it as jamming a broomstick in a corner of a room—it is firmly guided into position by the two baseboards and the floor. By watching an ohmmeter we establish light contact with the crystal face. The assembly is then soldered.

The spacer is then removed, and the crystal forced firmly into the brass cylinder. The Bakelite barrier stops it, and the contacts are forced back 2½ thousandths, and held apart by the mica spacer. With a one-thousandth spacer, separation of points is about 1½ thousandths.

By using thinner mica, closer location of the two points is obtained but the tendency toward cross-leakage from the plate to the grid is enhanced. After all, we may operate with over 60 or 80 volts between these points, and it is a miracle that the leakage is so small.

The whole assembly is made eccentric, so by rotating the crystal after pulling it out by any distance in excess of 3 thousandths, and then firmly pressing it back in place against the Bakelite barrier, we have a new spot. Incidentally, there are so many spots on such a crystal it is impossible to burn them all out.

We have one trouble with this assembly—it works very well with crystals whose ends are truly squared off, but not so well with those crystals lapped at an angle. The early crystals we bought were about 90% true on the ends, but later batches show as much as 50% off true.

This assembly will take ordinary untreated crystals from an IN34, of course. Some of them make good oscillators, but they are very non-uniform, as even partial treatment of the crystal makes it much more uniform in useful performance.

Test Gear

A test set up was developed as shown in the photograph. The two meters show grid and plate currents, and the two voltage controls are set to the desired point by an adjustable series resistor. Grid voltage is furnished by a built-in battery, or can be obtained from extenbal source. Plate voltage is secured from a 250-volt supply. A built-in plate loading transformer with tapped primary and six secondary loads provides all plates impedances from 100 ohms to 100,000 ohms at 1,000 cycles. High-frequency oscillator assemblies can also be plugged in, and all leads are accessible for inserting meters or loads.

Static Characteristics

There are three types of crystals to discuss. We start with the untreated crystal of high plate impedance, about 50,000 ohms, and of otherwise highly variable characteristics. The flashover point is medium, say 50 to 60 volts, but it will not withstand repeated flashovers, which rapidly lower the flashover points to the 5-10 volt region.

A partly treated, or "soft," crystal is interesting. The flashover is slightly higher, but again it will have its flashover point lowered by repeated use. Its usable stable transconductance reaches high values, and the number of active spots is about 50%. The input impedance is unfortunately very low and plate impedances may be from 5,000 to 20,000 ohms.

The fully treated, or "hard," crystal, has a quite high flashover, often above 100 volts, which is seldom lowered below about 80 volts by repeated flashovers. Its transconductance is often somewhat lower than that of a soft crystal. It has the very fortunate characteristic that these flashovers weld the point of the catwhiskers to the crystal face, so that unchanging current is drawn. There is no pressure that will really give equally steady conditions on an untreated or "soft" crystal. Plate impedance is about 10,000 ohms, and the input resistance is higher than that of a soft crystal.

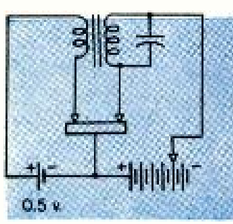
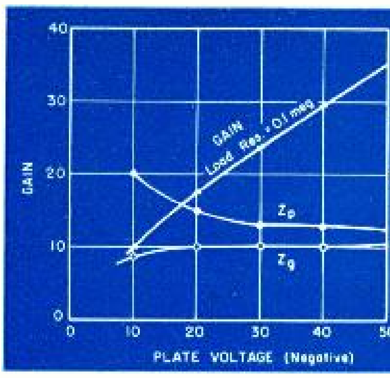


Fig. 5. Experimental oscillator circuit.

The static curves of *Fig.* 2 are from a soft crystal, which we will show in dynamic and also oscillating condition (*Figs.* 3 and 4). There is one odd fact about these static curves that makes them doubtful for calculation purposes. If we slowly run the curves by rotating a grid bias potentiometer, for instance, the curves are correct. But if we switch the voltage on suddenly, there is a sluggish response—that is the meters will jump to a new value, and then drift for about two seconds, the drift accounting for about ten per cent of the total change.

The curves show high values of transconductance, 6,000 at 30 volts. This is a stable condition. Note the change in grid volts is only 0.5. The mu is around ten, and varies with plate voltage.



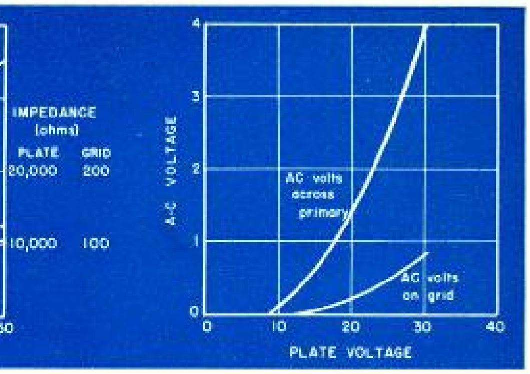


Fig. 3 (left). Dynamic characteristics of "soft" crystal.
Fig. 4 (right). Oscillator performance curve.

Note that the grid current starts at 1 ma and steadily increases to about 5 ma. Some of this is due to leakage from the plate.

Dynamic Curves

Using the same soft crystal, the plate impedance was taken by inserting a variable resistive load in the plate and

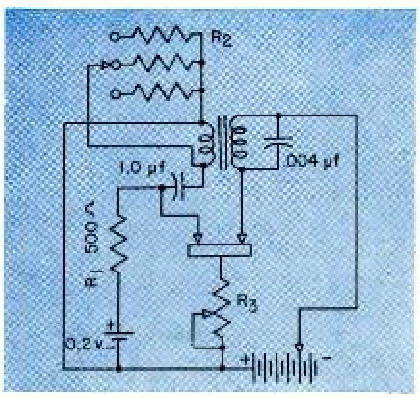


Fig. 6. Improved oscillator circuit.

adjusting it until the voltage drop across the 100,000 ohm plate resistor is halved. It will be noted that the plate resistance is constant from 25 volts up. The gain values are normal, the highest gain we ever obtained into 100,000 ohms being about 50.

An interesting point is that the change in input impedance due to plate voltage variation is small, and if the plate be optimum loaded the reflection of this dynamic load causes not more than 2 percent change in input resistance.

Oscillator Use

This same crystal oscillated quite nicely. The first circuit (*Fig.* 5) used direct feedback through the output transformer. This gave interrupted oscillations, so a grid condenser and leak (*Fig.* 6) was inserted. This circuit gave much smoother oscillations. Oscillation started at 8 volts on the plate, and reached ten volts rms at 40 volts, plate. The wave form was peculiar, as is shown in *Fig.* 7, but remember a vacuum tube does not give a sine wave either when an output transformer is used for the feedback element.

*R*2 is the bank of resistors across the secondary of the Stancor 10 watt universal output transformer built in the test equipment. These resistors tuned the oscillator frequency through one octave—800 cycles to 1600 cycles—with a change of resistance from 200 ohms to 2000 ohms, without affecting the voltage output. When fed to an amplifier and speaker, it sounded like a rather pleasing electronic organ, and various "tones" could be had by varying the plate voltage to limits shown in the wave form figure. Optimum turns ratio for smooth oscillation was 4 to 1 (8000 ohms to 500 ohms). At 40 volts, plate, the oscillator maintained an output signal of ten volts for four hours without changing in intensity more than 2%.

A cathode resistor *R*3 was inserted and stopped oscillation with a plate voltage of 30 volts when set at a value of 400 ohms.

The 400-ohm cathode resistor was also inserted when we were investigating stage gain. It raised the input impedance from 100 to 1000 ohms. Unfortunately, the gain fell to about zero. The input signal was 0.1 volt, across cathode resistor 0.1 volt also, and across plate load 0.5 volts.

When amplifying a sine wave signal, the output signal distortion is unnoticeable on an oscilloscope.

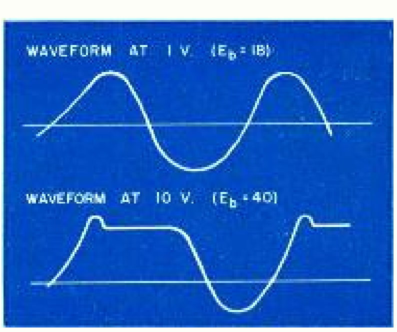


Fig. 7. Oscillator output waveforms.

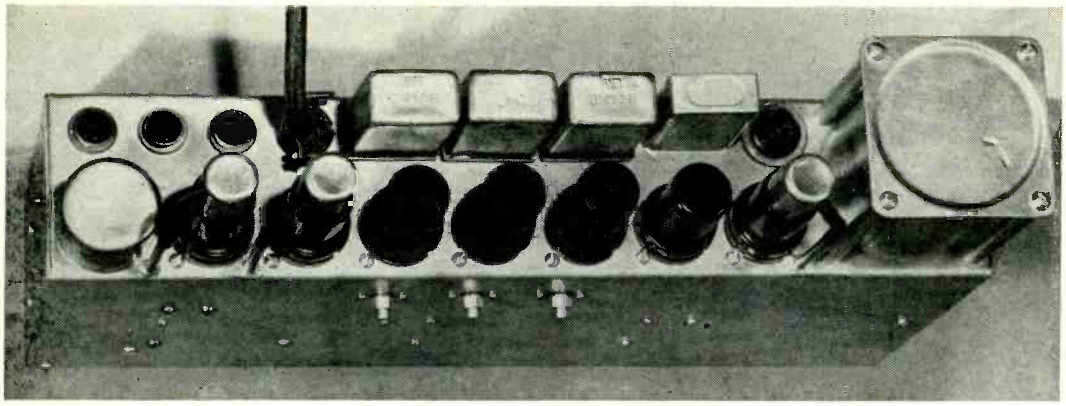


Fig. 1. External view of the control amplifier, showing the unique arrangement of the components. Operating controls are on the opposite side of the chassis.

Elements of RESIDENCE RADIO SYSTEMS

C. G. McPROUD*

PART III

A description of the control unit, with design data on the phonograph equalizer and noise suppressor circuits

IN THE FIRST TWO ARTICLES of this series, both AM and FM tuners were discussed as suitable input sources for a specific type of Residence Radio system. These units were designed with two primary considerations—quality of reproduction and convenience in use. Tuners alone do not make a radio receiver, and this article covers the next stage in the system.

**Managing Editor*, Audio Engineering.

The phonograph preamplifier and the noise suppressor are grouped into one unit known as the control amplifier, so called because it is the only point in the system where any control over volume or frequency response is exercised. It includes a number of circuit sections, but is constructed as a single unit. Throughout this entire system, professional practices are followed to a large extent, and the output of the control amplifier appears as a 600-ohm line feeding the power amplifier, which has a transformer input. This practice has the advantage of separating the various sections into units which may be changed individually if a desire for modification should arise or if new developments make it advisable, thus avoiding scrapping a complete system as would be necessary if it were built in one unit. It is not economical from the standpoint of tubes or chassis or plugs and cables, but it does provide a high degree of flexibility.

The control amplifier consists of a phonograph equalizer with two turnover points, a preamplifier, switching facilities for selecting various inputs, a Scott-type noise suppressor, and a compensated volume control. Since the suppressor can serve as a low-pass filter, and since the volume control furnishes almost complete compensation for the Fletcher-Munson hearing curve, no tone controls are employed. If they are ever considered necessary, they may be installed in the 600-ohm line to the power amplifier without disturbing either chassis, and constant-

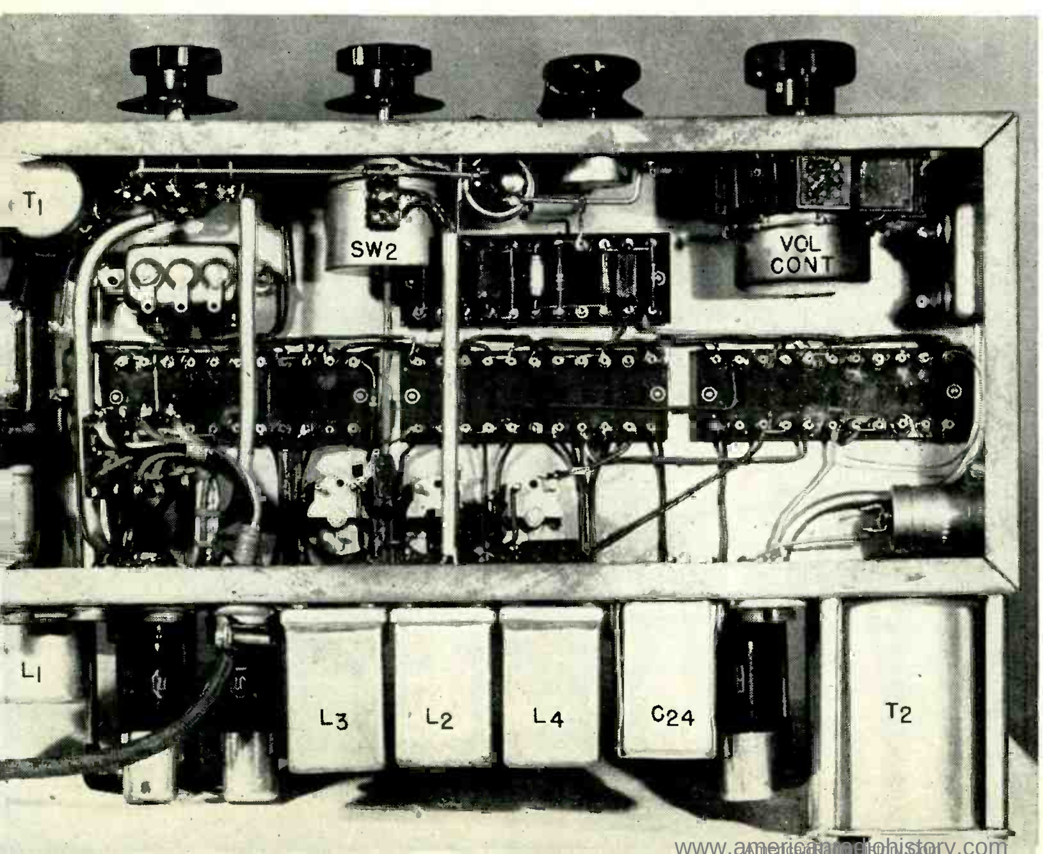


Fig. 2. Internal view of the chassis prior to completion. The input transformer is at the upper left corner, and resistors and capacitors will be mounted on the Bakelite strips. Note use of copper tubing for shielding long leads.

impedance equalizers can be used in accordance with professional practice. So far, no such controls are deemed necessary.

The physical arrangement is shown clearly in *Fig.* 1. The chassis is a standard 7x15x3 radio chassis, but all the tubes and some other components are mounted on one of the 3x15 sides, with the controls on the opposite one. This makes parts accessible and provides a desirable chassis arrangement. The plate and filament power are obtained from the main power supply which furnishes 225 volts, regulated, for the plates, 12 volts d.c. for six of the heaters, and 6.3 volts a c. for the seventh. *Figure* 2 shows the inside of the unit before completion, with some of the resistors and capacitors mounted on the strips. Note the use of solid copper tubing for shielding leads that must be run any great distance. These tubes are shaped and soldered in place, and the wire pulled through afterward. Tubing provides better shielding and is self-supporting.

Electrically, the control and amplifier consists of a phono-preamplifier and equalizer, an input stage, two high-frequency gate tubes, one low-frequency gate tube, and the output stage, with the necessary side amplifier and rectifiers to actuate the suppressor gates.

Phonograph Equalizer

A more accurate low-frequency equalization curve, with lower losses, can be obtained from a magnetic pickup equalizer by the use of low-impedance elements. The unit used here consists of the elements to the left of T_1 in *Fig.* 3, the over-all schematic.

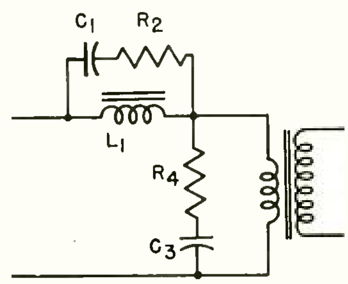


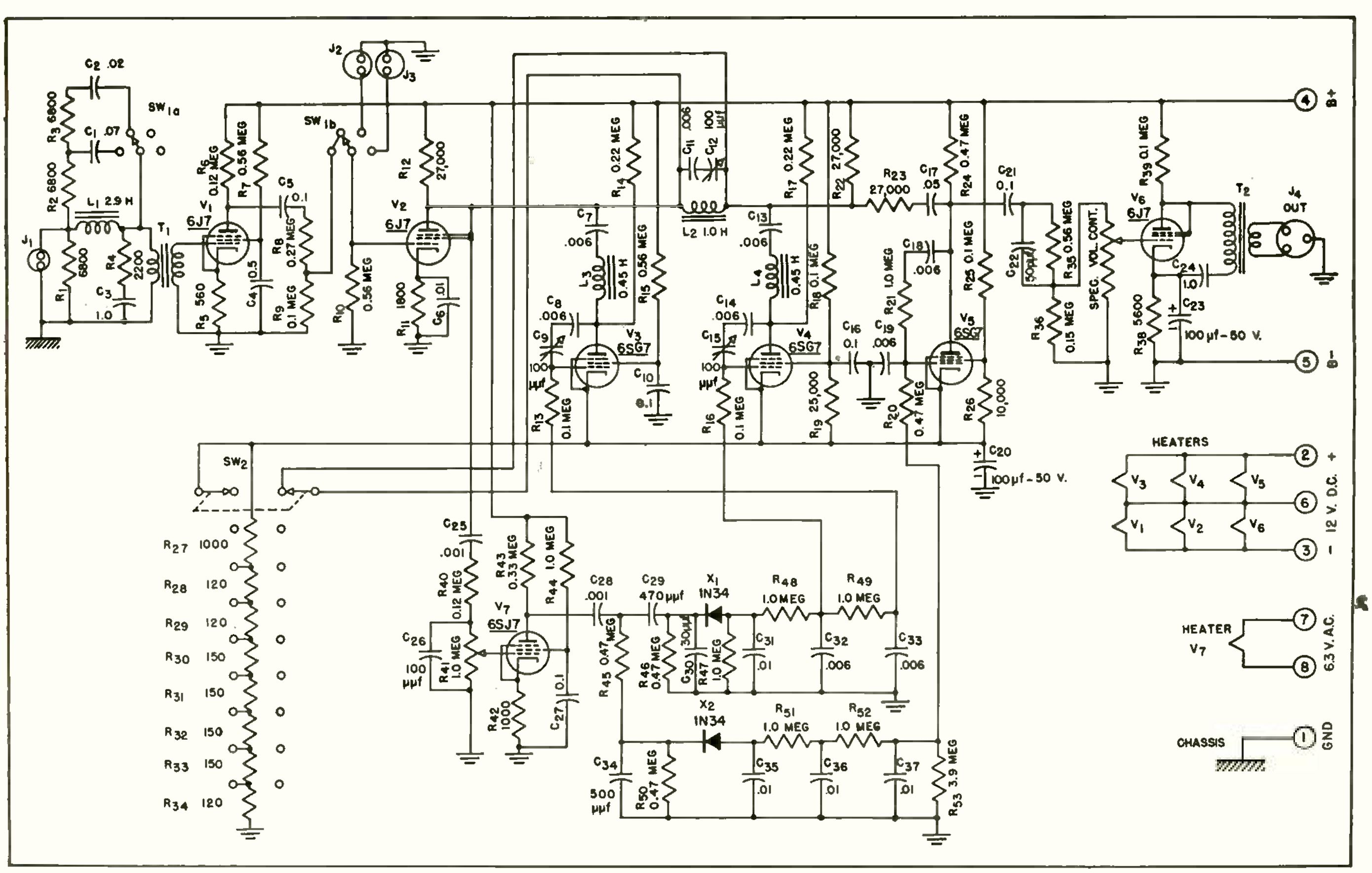
Fig. 4. Simplified circuit of the phonograph equalizer.

To adjust such an equalizer requires the use of an audio oscillator, but once the principles are understood, an ideal equalization curve can be obtained.

The basic equalizer circuit is shown in *Fig.* 4. The series inductance L_1 is tuned by the capacitor C_1 to the turnover frequency and with R_2, serves to adjust the response in that region. The shunt elements, R_4 and C_3, control the low-frequency rise. The value of R_2 is somewhat critical, as shown at (*A*) in *Fig.* 5. The easiest method of adjusting C_1 and R_2 is to reduce the latter to about 2,000 ohms, and then to vary C_1 to obtain a dip at the turnover frequency. Then increase R_2 gradually until the curve is flat above the turnover. This will result in a smooth but fairly sharp turnover. The over-all schematic shows two capacitors, C_1 and C_2, together with two resistors, R_2 and R_3. These provide two turnover frequencies, one at 800 cps and one at 400 cps, a compromise between the common 300-and and 500-cps points.

The resistor R_4 adjusts the slope of the curve below the turnover, as shown at (*B*) of *Fig.* 5. In order to make measurements on this type of circuit, the oscillator should be connected with a series inductance equivalent to that of the pickup to be used. A simpler procedure is to connect the actual pickup between the oscillator and the input terminals, as at (*C*). The input transformer provides a voltage step-up to the grid of the first tube. Any good quality, well shielded input transformer should suffice, with the primary intended to work from a 250 to 600-ohm source. Resistor R_1 across the input terminals is used to flatten the response over the higher end of the spectrum, and a switch on the phonograph motor board shunts an additional 1800-ohm

Fig. 3. Complete schematic of the control amplifier and noise suppressor.



resistor across the pickup to provide a roll-off almost equivalent to that required for NAB transcriptions, and also necessary with the new Columbia LP records.

This type of equalizer is more efficient than RC types, and has a total loss of only 20 db, not counting the gain in the transformer. Actual measurements on this unit show a voltage of .023 on the grid of V_1 for an input of .016 volts at 1,000 cps. The transformer has an impedance ratio of 1:120, or a voltage step-up of approximately 11, yet the equalization realizes the total of 20 db. Typical RC equalizers lose up to 40 db for complete correction, and in addition, the curve is considerably more rounded at the turnover point. The response curve of this equalizer is shown in *Fig.* 6. The preamplifier tube, V_1, provides a voltage gain of approximately 31 as used in this circuit.

The Noise Suppressor

The input to the second stage, V_2, is preceded by a selector switch. Two positions are used for records, the ganged section, SW_1, adjusting the turnover frequency. One other position is used for radio input, and the fourth is a spare. R_8 and R_9 serve as a voltage divider to balance the output of the preamplifier to the radio input.

The noise suppressor section is similar to that previously described by the writer[1]. However, some modifications have been made, and the reasons therefor will be discussed. C_6 across R_{11} is used to equalize the frequency re-writer.[1] However, some modifications in the high frequencies to compensate for all the circuits "hanging" on the

[1] "General Purpose 6AS7G Amplifier," C. G. McProud, AUDIO ENGINEERING, June, 1948.

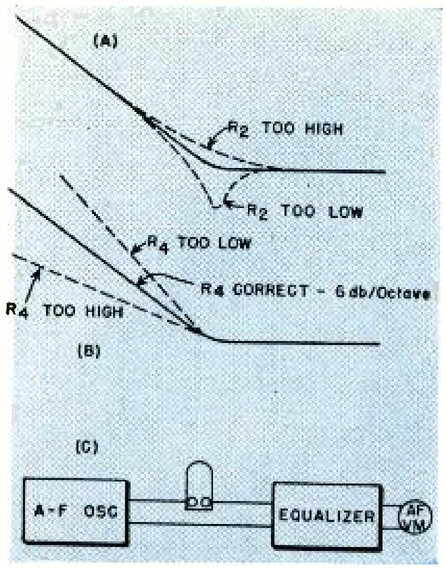


Fig. 5. Phonograph equalization curves at A and B show the effect of resistor values. C shows method of connecting oscillator to obtain correct results.

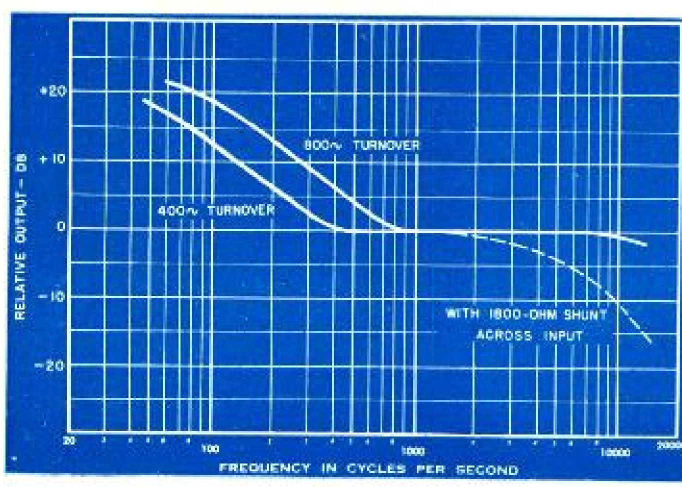
high-impedance line from the plate of V_2 to the volume control. C_{11} isolates the plate voltage from the stator of the trimmer C_{12} so the latter will not be "hot" to chassis. C_8 and C_{14} perform similar functions.

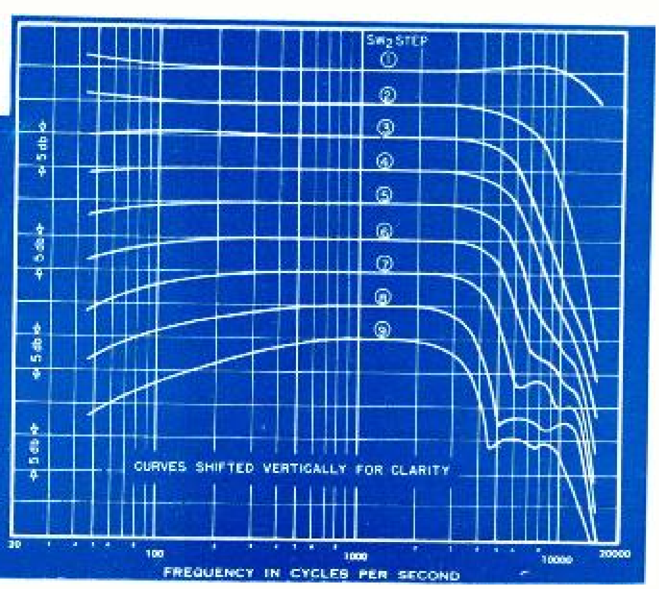
The coils used in this particular suppressor will probably not be generally available. They have a Q of around 60 at 1000 cps—too high a value—and L_2 is 1.0 H; L_3 and L_4 are both 0.45 H units. The exact values of these coils are not important, provided the tuning capacitors are chosen to tune the circuits to the required frequencies. Any coils having an inductance ratio of approximately 2:1, and with the larger being from 0.8 to 2.0 H should be satisfactory. The Q should range from 10 to 20. The final adjusting operations can accommodate these variations easily.

Filter circuits which employ high-Q coils are capable of extremely sharp cut-off, and as a result, it is not uncommon for the response to rise 2 to 3 db above normal just before the steep droop. This peak usually results in a poor-sounding output, and its presence is readily recognized by a trained ear. By way of digression, the filter circuits used in this type of noise suppressor are actually of the band-elimination type, and only the lower portion is normally measured. If the measurements were carried up to 40 or 50 kc, the signal voltage would rise again, assuming the other circuits would pass the higher frequencies. Such a filter is normally designed to work between specific impedances. As the suppressor operates, the various tuned circuits must be altered in order to remain correct for a given source impedance. However, in the suppressor, the capacitance of the shunt circuits—that of the reactance tubes—is the only element that is varied. Also, the coil L_2 in the series leg and its tuning capacitors are not changed.

According to filter design theory, practically any combination of coils and capacitors can be made to serve in a band-elimination circuit, but the impedance changes. However, the impedance of the source and load in the suppressor does not change appreciably, and the filter is not perfectly matched to its terminations. With high-Q coils, the peak becomes objectionable. This effect can be eliminated by loading the circuit by a resistor, which is the function of R_{22}. Its value should be adjusted to eliminate the peak just prior to the cutoff.

Fig. 6 (below). Final response curves of phonograph equalizer with two positions of the selector switch. Fig. 8 (right). Over-all response curves of noise suppressor for nine positions of control switch.





Another characteristic which is also objectionable is occasionally observed. That is the return of the response curve after the resonant point of the filter. For example, a filter of this type will have a response similar to that of the solid curve of *Fig.* 7, where f_r is the frequency of the dip due to the shunt circuits, L_3-V_3 and L_4-V_4, and $f\infty$ is the frequency of the dip due to L_2-C_{11}, C_{12}. This curve also shows, at *A*, the rise before the dropoff. The return, at *B*, often rises as much as 10 db above the dip at f_r, and this too does not sound well. Therefore, in this suppressor, L_4-V_4 is tuned to one frequency and L_3-V_3 is tuned to a frequency midway between f_r and $f\infty$, giving a curve like that of the dotted line in *Fig.* 7. Now, while the dip at f_r is not as great as before by about 6 db, the response does not rise appreciably above

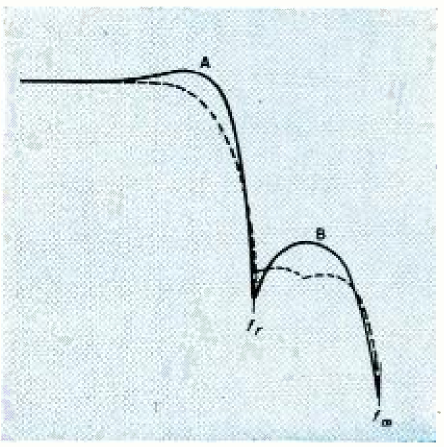


Fig. 7. Curves resulting from noise suppressor. Solid line represents initial measurement before correction, showing peak at A and return at B. Dotted line shows curve after applying corrective measures described.

f_r, and results are more pleasing to the ear. The problem now is to keep the dip at higher frequency midway between f_r and $f\infty$ during the shift of f_r by the action of the gate control circuits.

This can be accomplished readily by proper choice of the operating characteristics of the two reactance tubes. V_4 is tuned to 4,500 cps at maximum suppression, and its screen voltage is supplied from a voltage divider, R_{18}-R_{19}. This tube is connected to the side amplifier-rectifier circuit to operate more quickly than does V_3, so the lowest-frequency dip moves upward quickly in the presence of high frequencies in the signal. V_3, however, has its screen supplied by a series dropping resistor, thus changing the μ bias characteristics of the tube. With this connection, the effective capacitance of V_3 varies less slowly with a given change in bias than does that of V_4, and the second dip in the curve remains approximately midway between f_r and $f\infty$, as f_r is increased by a shift of grid bias.

The side amplifier and the rectifiers are conventional, electrically, but 1N34 germanium crystal diodes are used instead of tubes. Turning to the side-amplifier input, C_{25} couples to the signal circuit at the plate of V_2. This capacitor is small to reduce low-frequency response. C_{26} across R_{41} serves to reduce high-frequency response. Thus, the gates are controlled principally by the mid-range signal, rather than by either highs or lows. The circuits between V_7 and the two rectifiers are essentially a dividing network, and each rectifier is followed by an RC filter.

The switch controlling the suppression has nine position. At 1, the suppressor is inoperative because of a high grid bias, and another pair of contacts shorts out L_2, giving a curve which is essentially flat. At 2, the reactance tubes are still cut off, but the short across L_2 is removed, providing a roll-off down 1 db at 5,000 cps and down 13 db at 10 kc. This circuit is resonated at approximately 16 kc. In the remaining seven positions, the bias on the reactance tubes is reduced gradually, selecting a number of about equally spaced frequencies for f_r. The measured curves of the completed suppressor are shown in *Fig.* 8.

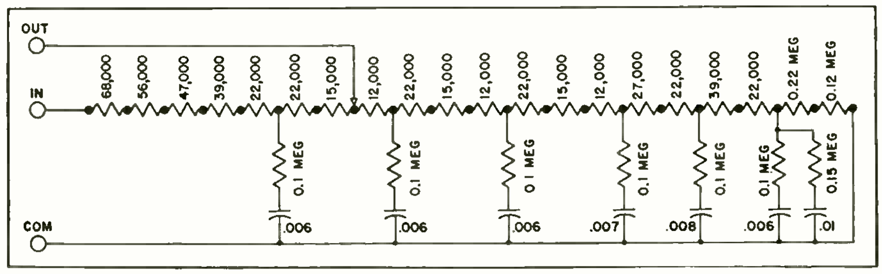


Fig. 9. Schematic of the volume control compensated for loudness contours.

The Compensated Volume Control

Following the suppressor section is the volume control. This unit is completely compensated for the Fletcher-Munson curve, and was originally described in these pages[2]. Since it is the only volume control in this system, it has more steps than the original model, but the operation is similar. This unit was assembled on an IRC commutator-type attenuator, with the capacitors mounted externally with Kovar bead seals leading through the shield case. The complete schematic of the volume control is shown in *Fig.* 9.

Until a listener becomes familiar with this type of volume control, he is certain to be somewhat amazed by its characteristics. It is necessary to adjust the over-all gain of the system so that the control is operated about five steps below maximum for normal room volume. This gives a slight boost to the low frequencies, but it is nearly correct for *normal* room volume when compared to the original sound source. As the control is turned, however, the apparent balance between low and high frequencies remains so nearly constant that it is sometimes difficult to determine whether or not the level has been changed. It is this control that relieves

[*Continued on page* 46]

[2]"Loudness Control for Reproducing Systems," David C. Bomberger, AUDIO ENGINEERING, May, 1948.

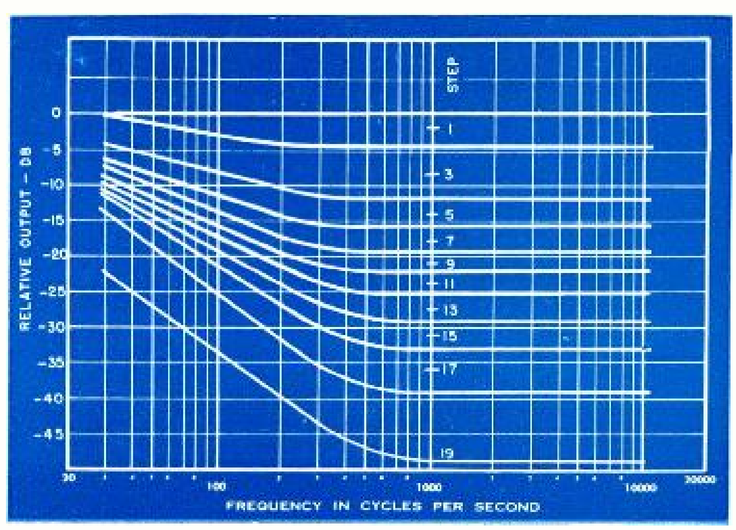


Fig. 10. Curves obtained with various settings of compensated volume control.

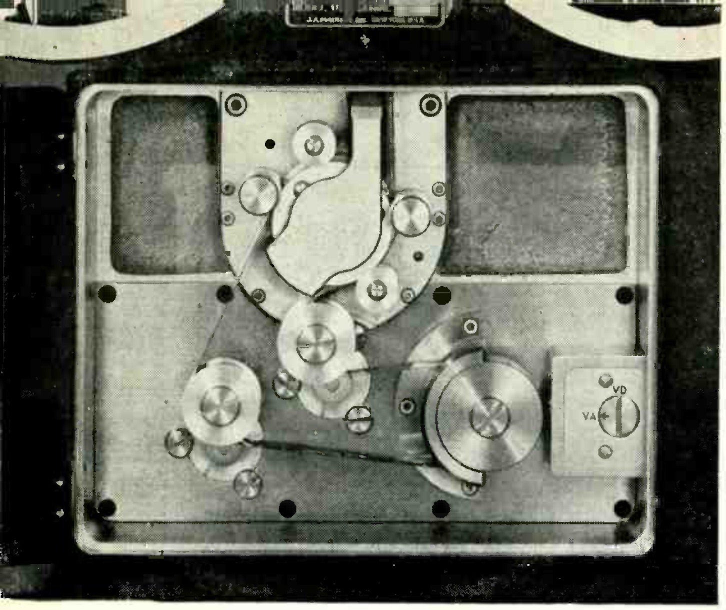


Fig. 1. The operating side of the film-moving mechanism of a typical sound-on-film recorder. The recording optical system is at the right-hand side.

Sound on Film

JOHN A. MAURER*

The author discusses in detail the methods and technique of modern sound-on-film recording.

OPTICAL methods of recording sound on photographic film were developed for the motion picture industry, which needed a type of sound record that could be carried on the same support with the pictorial record and thus maintained in synchronism with it. Because photographic film is more costly than other readily available record media, sound-on-film records have found very few applications apart from motion pictures, and most audio engineers have had no occasion to concern themselves with them. Today, however, these recording processes are of general interest because they will be needed for the recording of television programs and in preparing programs on film to be televised.

The engineer who seeks to educate himself on the subject of sound-on-film recording will find that with the exception of two books,[1],[2], one of which is in the German language and both of which are somewhat out of date, almost all of the useful literature is in the files of the *Journal of the Society of Motion Picture Engineers.* This literature covers a period of more than twenty years. As might be expected, many of the articles are not applicable to present practice. Two valuable summarizing articles exist,[3],[4], one of which is available in reprint form.

The present article and those which are to follow will attempt to set forth those facts which are of practical importance to engineers who must learn to use sound-on-film recording and reproducing equipment or who will be making decisions regarding its use. A critical bibliography will be attached to each article. Since the emphasis will be on what is of practical value today, no mention will be made of devices and methods that are no longer used, even though some of them are extremely interesting.

Photographic Processing

The most important practical difference between photo-optical sound recording and the other methods in common use is that the record on film is not complete when the recording is finished. It does not even exist in a usable form until it has been developed and printed. This means, of course, that immediate playback for checking is not possible. This is an inconvenience, but is not in itself a matter of major importance. What is important is that the photographic processes may profoundly affect the quality of the record, and, if improperly handled, may ruin it altogether. Furthermore, all of the steps in the photographic procedure are interrelated, and are strongly influenced by one another. For example, correct original exposure of the sound track is essential, but this must be established by co-operative tests between the recording studio and the film laboratory that will develop and print the records. The recording lamp current that is correct for the development given in one laboratory may be seriously wrong if the film is sent to a different laboratory. Since an error by either the recording personnel or the film laboratory may lead to bad quality, it is difficult for the beginner to determine what is at fault when results are unsatisfactory. If, however, proper checking procedures are maintained, the results of sound-on-film recording can be made as consistent as those of any other high-quality recording method.

Basic Outline

The fundamental steps in photo-optical recording and reproduction are as follows:

1. The electrical signal corresponding to the sound wave is caused to modulate

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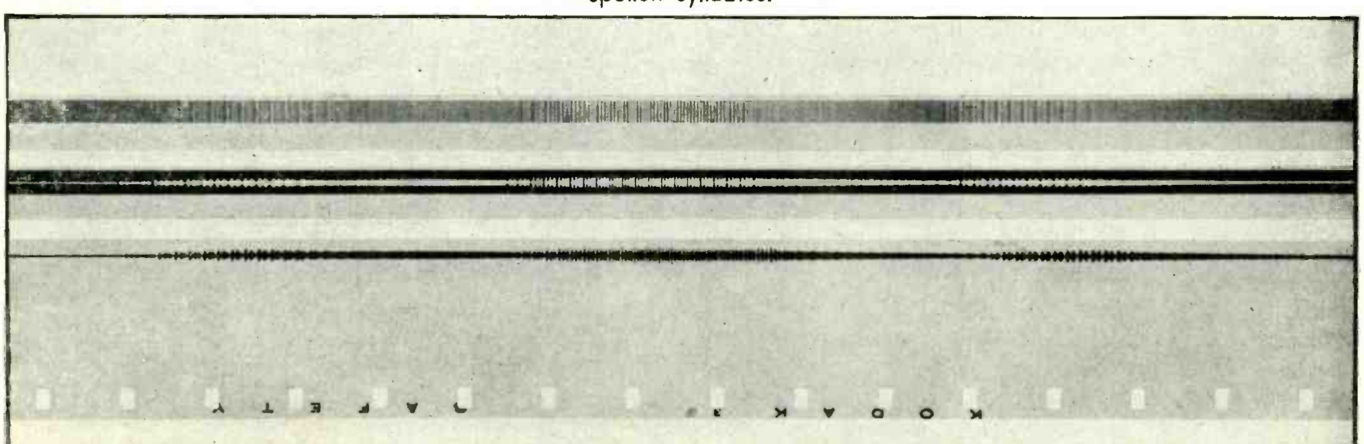
Fig. 2. A variable area negative (bottom), a variable area print (center), and a variable density print (top) of the same spoken syllables.



a beam of light, which is focused on the moving film in a mechanism such as is shown in *Fig. 1*.

2. The exposed record is developed to produce (in most cases) a negative sound track.
3. The sound track negative is printed on another strip of film, on which, in most cases, the picture image has been or will be printed.
4. The print is developed to produce the positive sound track.
5. The sound track on the print is used to modulate a second beam of light which is transmitted to a photo-electric cell.
6. The electrical signal generated in the photo cell is amplified.

Step 6 is not a photo-optical process, but is included in the outline because it presents special problems which are unlike those encountered in reproducing sound from discs or magnetic tape.

Methods of Recording on Film

Photographic processes are able to reproduce the shapes of objects with considerable accuracy; they are also capable, within limits, of reproducing differences of light intensity. Corresponding to these two aspects of photography we have two distinct systems of sound recording. In the *variable area* system the light modulator produces on the film what is essentially a graphical picture of the sound wave. Films and processes are chosen so as to reproduce the *shape* of the recorded wave as accurately as possible. Those parts of the track which are exposed to light all receive the same exposure, but the width of the exposed area varies. In the *variable density* system, on the other hand, the entire available width of the sound track is always exposed, but the exposure varies from instant to instant. Films and photographic processes are chosen so as to obtain as nearly as possible a linear relation between the variations of *transparency* in the positive sound track and the variations of *exposure* in the recording of the negative. Typical variable area and variable density sound tracks are shown in *Fig. 2*.

Light Modulating System

A typical light modulating system for variable area recording is shown diagrammatically in *Fig. 3*. The galvanometer in this system is usually a moving-iron-armature device carrying a mirror about 1/8 inch square. When an electrical signal is applied, this mirror swings about a central axis in such a way that the triangular image which falls on the slit is moved at right angles to the slit. Obviously, this varies the length of the slit that is illuminated, but the light intensity remains at a constant value. The lens L_4,

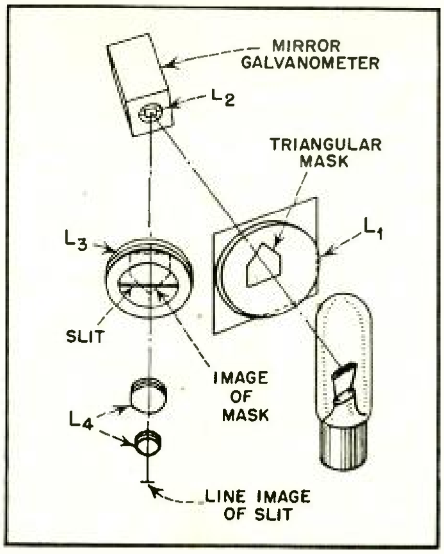


Fig. 3. Diagram of galvanometer and optical system for variable area recording.

which is essentially a low-power microscope objective, forms a reduced image of the slit on the moving film. This line image is usually about .00025 inch wide in 35-mm film recording, and not much more than .0001 inch wide in 16-mm film recording. The maximum illuminated length of the line is .076 inch in standard 35-mm recording, and .060 inch in 16-mm recording.

The corresponding light modulator and optical system for variable density recording are shown in *Fig. 4*. The "light valve" generally used for this type of recording consists of a pair of tiny stretched duralumin ribbons in a strong magnetic field. These ribbons are arranged in the same plane, with their edges about .001 inch apart, forming a slit about .200 inch long. This slit is illuminated by the lamp and lens L_1, while the lenses L_2 and L_3 form a system that images the slit on the film at a reduction of about four to one in the direction of its width and two to one in the direction of its length. Thus the final image, in 35-mm recording, measures about .00025 inch by .100 inch. The audio signal currents are passed through the ribbons, which move in such a way as to widen and narrow the slit, thus varying the amount of light which reaches the film.

Film Speed and Frequency Range

The standard linear speed of the film in 35-mm recording is 18 inches per second; in 16-mm recording, 7.2 inches per second. Thus at 7200 cycles, which is near the upper limit of the frequency range normally reproduced by theatre equipment, the wave length on 35-mm film is .0025 inch, and on 16-mm film .001 inch. The latter value is commonly believed to be near the upper limit of the resolving power of the film. That this is incorrect is clearly shown in *Fig. 5*. The peaks of the wave on this record are .001 inch apart, and it is obvious that much finer detail could be reproduced.

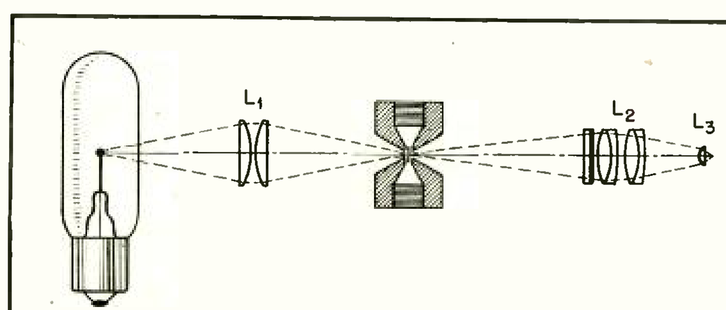
The upper limit of the frequency range in sound film recording is usually fixed in practice by the tuning point of the galvanometer or light valve. This is usually about 9,000 cycles in 35-mm practice. Substantially the same modulating devices are used on 16-mm recorders, though in the equipment built by the writer the tuning point of the galvanometer has been moved out to 12,000 cycles in order to remove the mechanical resonance peak from the working frequency range. Low pass filters cutting off at frequencies slightly lower than the tuning points of the light modulating devices are included in the audio inputs of most film recording systems.

Effect of Recording Image Width

It is obvious that in order to obtain clean records of high frequencies it is necessary to make the width of the recording light beam considerably less than the wave length of the highest frequency to be recorded. In variable area recording too wide a recording image introduces both wave form distortion and attenuation of high frequencies. A ratio of one to five between image width and wave length is about the minimum for good results. In most of the equipment now used in practice this ratio is about one to eight at a frequency of 9000 cycles.

In variable density recording, a wide recording image causes high-frequency attenuation but does not introduce

Fig. 4. Light valve and optical system for variable density recording.



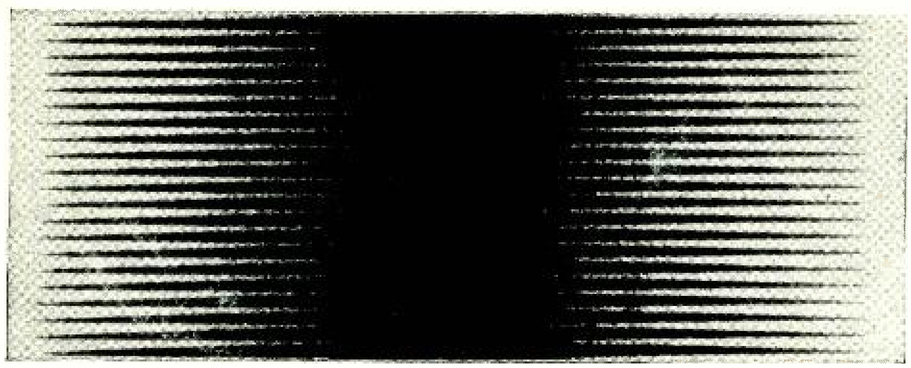


Fig. 5. Enlargement of a portion of a 16-mm variable area negative sound track of 7000-cycle sine wave.

wave-form distortion. Since the high-frequency attenuation is not serious and can easily be compensated in the electrical input to the system, recording images about twice as wide as those mentioned above are used in some variable density recording systems.

Choice of Film Types

In motion picture sound recording it must always be remembered that the film used for the positive print also carries the pictorial image, and its characteristics and the treatment it is given in development are determined primarily by the requirements of the picture. Therefore in both variable area and variable density recording it is necessary to produce a sound negative that has the proper characteristics for printing on this predetermined final type of film. The factors available for control of the final result are the choice of the negative film, the exposure in recording, the development of the negative, and the exposure used in printing the positive.

Fine grain motion picture positive film is a high contrast material, though its contrast is not as high as that of the so called "Contrast Process" materials used in photo engraving. As ordinarily developed it gives very little fog; that is, any part of the film which receives no exposure comes out almost perfectly transparent after processing. This is extremely important in variable area records, since any clouding or graying of what should be the clear areas of the positive sound track reduces the level of the reproduced sound and at the same time increases background noise.

The properties that are desirable in film for the variable area sound negative are high resolving power, high contrast, fine grain structure, and sufficient speed to yield dense images with the relatively small amount of light that can be transmitted through an optical system of the type shown in Fig. 3. Special types of film have been developed for the purpose. The most widely used of these are DuPont type 602 and Eastman types 1372 and 5372, which are the 35-mm and 16-mm sizes of the same basic type of film.

For variable density recording, a film stock of low contrast is desirable because the requirements of overall linear brightness reproduction are best satisfied when the contrast that results from the combined action of the negative and positive films is about the same as the contrast that was present in the original. (In photographic terminology, the overall *gamma* should be not much greater than unity.) Until recently no films of high resolving power, fine grain, and *low* contrast were available, and it was necessary to obtain the low contrast by carefully controlled development of inherently high contrast films. The contrast increases with the degree of development. This situation has been much improved by the introduction of Eastman film types 1373 and 5373, which have inherently low contrast.

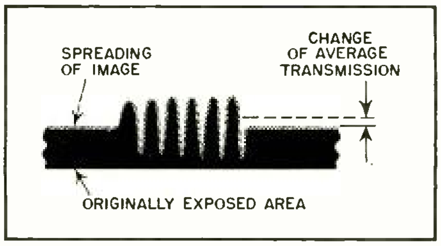


Fig. 6. Diagram of a portion of one side of a variable area negative sound track, showing how spreading of the image causes cross-modulation.

It is important to note that the higher speed films designed for picture taking, although most of them have inherently low contrast, are not suitable for variable density sound recording. The grain patterns of these films are a great deal coarser than those of the sound recording films mentioned above, and in variable density recording this results in an intolerably high level of background noise. Somewhat better results are obtained by variable area recording, but really high quality is not obtained in either case when recording on films designed for picture photography.

Variable Area Processing

In developing and printing variable area sound tracks we are principally interested in getting a high signal to noise ratio and in minimizing a form of distortion known as *cross modulation*.

Both high signal level and low noise are assured if the black areas in the sound track print are sufficiently opaque and if at the same time the transparent areas are free from fog and are clean. In good commercial practice the blacks transmit only three to five per cent of the incident light, and the clear areas absorb about the same amount. The reproduction level of such a track, if fully modulated, is within one decibel of the maximum possible. Cleanliness is the most important factor in obtaining a low level of background noise.

Cross-modulation distortion arises because the emulsion of an undeveloped photographic film is opalescent, and scatters the incident light. This scattering causes any image impressed on the film to spread somewhat beyond the limits of the area exposed. The result, as it affects the recording of a sine wave, is shown in *Fig. 6*. The negative image has both harmonic distortion, principally second harmonic, and a rectification component which causes groups of high frequency waves in the record to generate lower frequency noises corresponding to their envelopes. Speech is affected more noticeably than music. In particular, uncompensated cross modulation produces an extremely unpleasant distortion of "ess" sounds which is as though two pieces of sandpaper were brushed together at the instant each sibilant is reproduced.

Cross modulation can be eliminated in properly controlled printing because the exposure of the print is through the areas that are clear in the negative, and thus the spreading of the image is in the opposite direction. In order to obtain accurate cancellation it is necessary to determine the proper printing exposure in relation to the exposure and development used in producing the negative.

Sound Track Printing

Good cross-modulation cancellation can only be obtained with accurate printing equipment. Most sound track printing is still done on printers of type that has been in use for many years in the motion picture industry. The process is shown diagrammatically in *Fig.* 7. The negative film and the print "raw stock" are passed, with their emulsion surfaces in contact, around part of the periphery of a large sprocket. At about the middle of the arc of engagement of the two films with the sprocket there is an exposure aperture at which a light shines through the negative sound track and thus exposes the positive. The excellence of the result depends on uniform motion of the two films and on close contact between them. Unfortunately, both motion and contact are periodically disturbed by the action of the sprocket teeth in entering and leaving the perforations of the film.

Another type of sound track printer uses two film driving mechanisms similar to the one shown in *Fig.* 1 to drive the negative and positive films separately at constant speeds. An illuminating system is placed back of the negative sound track and an optical system between the two films produces an image of the negative track on the positive film. This image travels in the same direction as the positive film and at the same speed. Such printers have been used principally to produce 16-mm sound track prints from 35-mm negatives, but the writer has recently shown[5] that optical printing leads to superior results even when the negative and print are of the same size.

Processing Variable Density Records

In determining the photographic treatment of variable density records we have the problem of combining two non-linear characteristics so as to produce an overall characteristic that is linear over as great a range as possible. The light-transmission vs. exposure characteristics of the negative and positive films individually have about the same amount of curvature as the plate current vs. grid voltage characteristic of an output triode when it is driven from near plate current cut-off to the grid current point. The exposure in recording or printing determines the "operating point", and the time of development and the constitution of the developer influence the shape and steepness of the curve. When properly combined, the two curves can be made to yield an overall result that is linear up to about 90 per cent modulation of the recording light valve. Since three independent variables are available (negative exposure, negative development, and printing exposure) it is not surprising to find that even with one selection of negative film the problem has many fairly satisfactory solutions. The now familiar intermodulation type of test, which was invented for this specific purpose, is generally used to determine optimum processing conditions for variable density records. Values of intermodulation as low as four per cent are obtainable in good commercial practice. The output level of the prints, however, is at best about six decibels lower than the output of fully modulated variable area prints. When the modulation level of

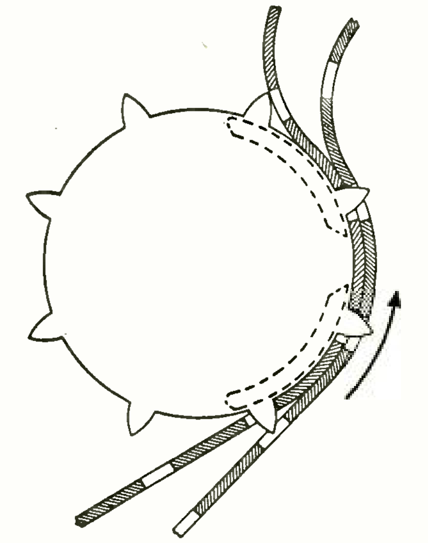


Fig. 7. Diagram of sprocket-type mechanism used for printing sound track. In practice the sprocket has 40 to 64 teeth instead of 8 as shown here.

the negative is not too high it is possible to obtain volume control over a range of about 12 db by varying the exposure in printing without increasing the intermodulation distortion enough to make it noticeable.

A major advantage of variable density records is that they exhibit no cross-modulation distortion; therefore the reproduction of the sibilants in speech is pleasing even when the records have not received optimum processing. On the other hand, improperly processed variable density records, especially of music, exhibit objectionable distortion of the familiar "overload" type.

Overload Characteristics

An important practical difference between variable area and variable density records is that in the latter, as in disc recording and magnetic recording, the overload point is somewhat arbitrary and depends upon what is considered a permissible maximum of distortion, whereas in variable area recording the overload point is perfectly definite. As long as the amplitude of the recorded wave is less than the width allotted to the track no significant distortion occurs. As soon as this width is exceeded the peaks of the waves are clipped and the distortion may be immediately noticeable. For this reason unusual care must be exercised in monitoring the signal for variable area recording.

Noise

The background noise in sound-film reproduction is due to dirt and dust on the film, abrasions caused by handling of the film, the grain structure of the photographic image itself, and hiss generated in the photo cell and its coupling circuit when the cell is exposed to light. All these sources of noise are reduced in importance if the light transmission of the sound track is reduced. This may be done in either variable area or variable density recording by applying to the light modulator a biasing current, superimposed on the audio signal, in such a way as to reduce the amount of light transmitted to the sound negative film during periods of silence or low signal level. This biasing current is removed in proportion to the peak amplitude of the incoming signal, so that when the signal reaches approximately ninety per cent of full modulation the biasing current is removed completely. This technique is known as "noise reduction." The sound tracks shown in *Fig.* 2 were made with noise reduction, and the operation of the system can readily be observed in them.

It is obvious that in "noiseless" recording the actual background noise level fluctuates continually with the signal level, but the masking effect of the signal is such that the noise is heard only during periods of low signal level. The time constants of the biasing amplifier must be adjusted correctly. Too rapid a change of bias current produces thumping sounds; too slow an "opening" action causes excessive distortion by clipping of the peaks of the waves each time the

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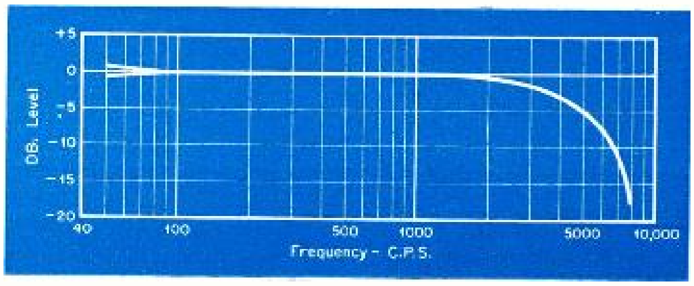
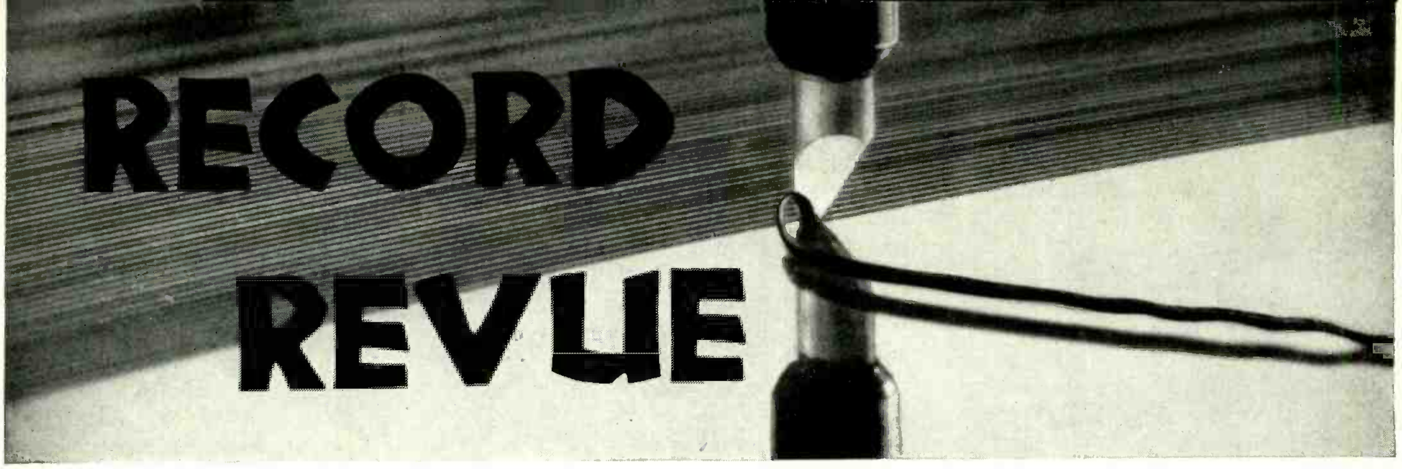


Fig. 8. Standard theater reproducing characteristic curve of the Academy of Motion Picture Arts and Sciences.



EDWARD TATNALL CANBY*

THE writing of the present articles[1] on control systems for the dynamic noise suppressor has been interesting and perhaps a bit delicate in more ways than one. Although my entire discussion concerns the outward control functions in relation to actual *musical* results to be had and therefore is well within my official province—nevertheless it is difficult not to become more involved in the technical aspects of design than is good for one of my training.

Fortunately, though I do now have certain well-backed opinions as to how I should like most to see the controls of a suppressor arranged on my own amplifier, I can say instantly that there is no question of black and white—if one grants that the dynamic suppressor is a good idea. There are not only numerous ways of using the circuit with satisfaction, but good results can certainly be achieved with any of the three leading commercial exemplars of the design, those of Messrs. H. H. Scott, Avery Fisher and John Goodell, as well as with several variants of the one-control, simplified type now appearing. Therefore I will continue by giving my own preferences were I actually to be consulted on the panel control requirements of some hypothetical new noise suppressor amplifier.

Let us begin with the premise that the supposed machine is a wide range type for use with high quality equipment, and will have "everything." (Some suppressors are designed specifically for average home conditions, with a low-cut-off system.) And that, in line with this, it is desirable to have the more complex, more flexible laboratory type controls, involving, as illustrated last September, at least two principal control centers, a "main" control, and a secondary fixed-position range control, the general form that all the better suppressor models have taken. These controls cover the functions of three basic operations (again ignoring the low-frequency gate): A) *Gate Sensitivity*, B) *H. F. Ceiling*, and C) *H. F. Floor*. Their relative importance, as reflected in the type and positioning of the controls, has been a matter of disagreement, as already outlined.

I would suggest first that the real fanatic—and most of us are that—would appreciate an amplifier with at least *three* controls, one specifically for each of these functions. The biggest question to be decided somehow is the relative importance of the three aspects. The table in the September issue will give an idea of differing opinion in actual production models.

A. *Gate Sensitivity.* As I see it, there is no question at all about the necessity for some variation in gate sensitivity on any type dynamic suppressor, so that increasing noise being passed is covered by increasingly loud music. We may dispense quickly with one obvious need, and to adjust the input level, which is bound to vary from pickup to pickup and from radio to phono. But, given an adjusted input level, there are other variables too, not only in the proportion of noise present in relation to the music but more important, the type of music and the noise-masking ability it has. A vital point is the tremendous variation in this respect found in many types of recorded music, and some further gate sensitivity adjustment at least is clearly required in order to meet such greatly varying conditions.

But the question is, how much? And more, shall gate sensitivity be the most important adjustment to be made?

Here I have found in actual experience that, once input levels are set, the needed variation in gate sensitivity, even to cover the widest variations in noise level and type of music, is *surprisingly small.* This is a fundamental point. As above said, I do not think sensitivity can be fixed (not counting input level adjustment) for all conditions. When it has been so fixed, I have resorted to making quite frequent slight adjustments of the input control to achieve what is in effect a change in gate sensitivity. But I feel that to provide more than this slight amount is a bit risky, in that it may be asking for trouble for the inexperienced user—which is almost everyone. Too great sensitivity leads to the unpleasant swish that means noise is coming through above the music in the louder passages; too little, means a generally dull tone and a near-fixed-filter action.

I would be inclined to reason, next, that since gate sensitivity is to be varied in a rather limited way, it might be wiser not to use it as the principal control function assuming that you accept my argument. The "Main" control is on most suppressors rather forcibly brought to the user's attention by some such designation as "Suppressor Action" or the like—implying that here is where listening attention should be centered. It would seem advisable somehow to subordinate the gate sensitivity control a bit—either by making another control more obvious or by merely labelling it in a more neutral way. On all the suppressor amplifiers I have tried, I have been able to accomplish this to my own satisfaction in one way or another, if only by restricting the variable sensitivity knob to a small arc of its total movement. My recommendation would be to make this function continously variable, *but through a narrow range*, and give it a neutral or subordinate positioning in the control panel, or a neutral lebal.

B. *Floor Control.* Conceivably, if a "main" control is wanted (and human nature seems to want one) either of the remaining functions might be used—*h.f. ceiling* (*B*), or *h.f. floor* (*C*), technical problems of course entirely aside. Of the two there is no question, in practice, that the *floor control* is the one that works, and, ideally speaking, I find that this control is the most useful of all. (Remember, it appears in some form in most suppressor models.)

Let us look at the possible usefulness of the floor control vs. the ceiling control with this logic, my own:

The variation we most need in noise suppression is the *degree of noise reduction.* Let us say, then, the noisier the record, the more filtering (noise reduction) we need. This, note well, is exactly as in any fixed-filter situation.

Next, note that the basic principle of the dynamic suppressor is that noise is most objectionable with a *low signal level*, therefore, filter most with the least signal. It follows nicely that our main adjustable filtering action, to meet differing conditions, must be at the low-level end of the dynamic range—the gates-closed position, or *floor*. In other words, we can treat the dynamic filter as we use the fixed filter and vary its maximum action (i.e., at low signal, gates closed) as we would vary a corresponding fixed filter. This argument would seem to me to justify trying a type of design that would make the variable floor control more prominent than the others.

It is quite possible, I admit, to get similar results by varying the gate sensitivity—in a sense this varies the effective amount of suppression, because, with less sensitivity, the gates are closed a greater proportion of the time. In fact, it is quite difficult to visualize the exact relation between these two ways of achieving roughly the same thing.

Look at it this way: When the gate sensitivity alone is varied, the floor remains fixed, no matter how far you turn the knob; what is varied is the proportion of floor to ceiling, so to speak—how much of the music is filtered toward the floor and how much comes through at the ceiling limit. The gate-sensitivity control, then, does not adjust the filtering *depth* at all; it does not provide deeper filtering action for noisier situations. Thus, whatever system my theoretical amplifier would use, floor control would have to be provided. It may be in a series of fixed steps, as some manufacturers have arranged it, or it may be continuously variable over the desired range—which clearly must be a wide one, just as a fixed filter set-up covers a wide range.

Because the variable floor seems to me to be the most useful of all the functions, I would tend to place it in the most prominent position, or otherwise (by labels, for instance) draw the user's attention to it. (Similarly, in any "automatic" compound-function, one-knob type of control, I would prefer to see a variable floor incorporated willy-nilly into whatever adjustments the user is to make with his single knob.)

C. *Ceiling Control.* There remains one more big consideration, the third possible control function, the *h.f. ceiling* (again, see AUDIO ENGINEERING for September). There are two ways in which a ceiling control here may be had: 1) via the regular attenuation treble

[*Continued on page* 35]

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*279 *W. 4th St., New York* 14, *N. Y.*

[1]*See* AUDIO ENGINEERING, *Sept.*, 1948.

TWO-WAY SPEAKERS
From Commercial Components

JOHN WINSLOW

Assembly directions for adding tweeters to cone speakers to give better reproduction.

WHILE the two-way speaker system may well be considered the ideal for sound reproduction, it is a costly device if purchased as a factory-built item, or it requires considerable work if entirely home built. The principal problem centers around the construction of the high-frequency horn. Once this is solved, the assembly of a good quality speaker system can be accomplished easily. Fortunately, several manufacturers have taken steps to solve this very problem, and a two-way speaker can now be a working reality without any sheet metal work, and with a minimum of measurement and adjustment.

Atlas High-Frequency Speaker

One of the simplest methods of assembling a two-way speaker system is to employ an

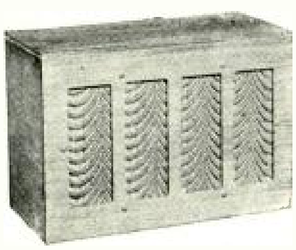

Fig. 1. Atlas HF-1 high-frequency speaker assembly, which includes its own dividing network in the cabinet which is designed to be placed on top of radio or speaker cabinet.

Atlas HF-1 speaker with an existing cone speaker in its normal baffle. This model of "tweeter" is made in the form of an attractive cabinet, shown in *Fig.* 1, which is intended to rest on top of a radio or a speaker enclosure. The dividing network is included in the HF-1 cabinet, and all that is necessary is to remove the present leads from the loudspeaker, connect them to the dividing network chassis, and run another pair of leads from the chassis to the loudspeaker. That is all there is to it. Phasing adjustments may be made by moving the entire cabinet backward or forward. and the balance between lows and highs is made by a control mounted on the dividing network chassis.

This unit is designed for an impedance of 8 ohms, which is a common value in better grade cone speakers, and works at a cross-over frequency of 1,400 cps. The h-f horn is made of cast aluminum to avoid resonances, and a moulded phenolic diaphragm is used; the magnet is Alnico V. The constant-resistance dividing network is of the parallel configuration.

This is an ideal unit when used with any good low-frequency cone as a "woofer." Users should be cautioned about the correct balance between the two speakers, since the

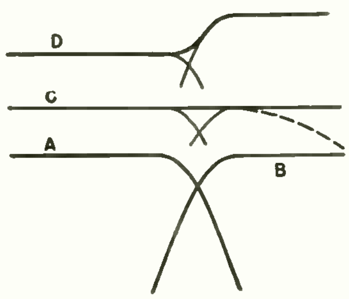

Fig. 2. Curves resulting from various components of speaker system. (A) is output of woofer; (B) is output of tweeter; (C) combined output of system when properly adjusted; (D) output when tweeter gets too much signal. Dotted line indicates action of normal tone control.

action of the control is not similar to that of a tone control. *Figure* 2 shows the output resulting from adjustment of the control. The entire band above the cross-over frequency is raised or lowered, putting a step in the response curve. The dotted line shows the action of a normal tone control. With the tone control of the amplifier or receiver in the "flat" position, the balance adjustment should be set for normal reproduction.

One advantage of the Atlas unit is that it is supplied with the dividing network already built in. While these networks are not difficult to construct if suitable measuring equipment is available, it is at least more convenient to have it already made. Some means for preventing the low frequencies from reaching the diaphragm of the h-f unit is necessary, and while this may be done by the use of a series capacitor, the network also removes the high frequencies from the woofer, preventing cone breakup and avoiding the possibility of phase cancellation at the higher frequencies.

Fig. 3. University Model 4407 tweeter assembly, with adapter ring.

If the user desires, the h-f horn may be removed from the cabinet and mounted inside a speaker baffle, since the horn is flat along the front surface, and is equipped with flanges by which it may be mounted on a flat surface. This begins to require some constructional work, however, which is completely avoided if the unit is used in its own cabinet.

University Models

Another equally simple method is to add a Model 4404 University Dual Tweeter to an existing low-frequency speaker and baffle. This unit, somewhat smaller than the Atlas, also comes with its own cabinet, and with

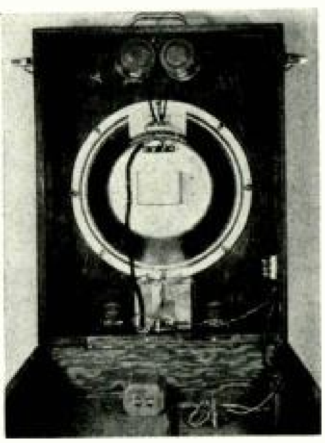

Fig. 4. Experimental assembly of 12" cone and University Model 4402, with dividing network and impedance-matching input transformer.

a high-pass filter and balancing potentiometer built in. Two small horns are used, each with its own driver, and they reinforce the woofer above about 2,000 cps. Since the filter consists solely of a series capacitor, low frequencies are kept off the h-f unit, and it is adequately protected.

Another University unit, Model 4407, is shown in *Fig.* 3. It consists of a 12" adapter ring which is used as a spacer between a 12" cone speaker and the baffle, setting the cone back far enough to accommodate a single small horn unit which mounts on a metal strap across the back surface of the adapter ring. To install this unit, it is only necessary to remove the cone from its baffle, assemble the h-f unit and horn to the strap, and attach the cone to the adapter ring. The entire assembly is then remounted on the baffle, and a two-way system results. The same type of high-pass filter may be

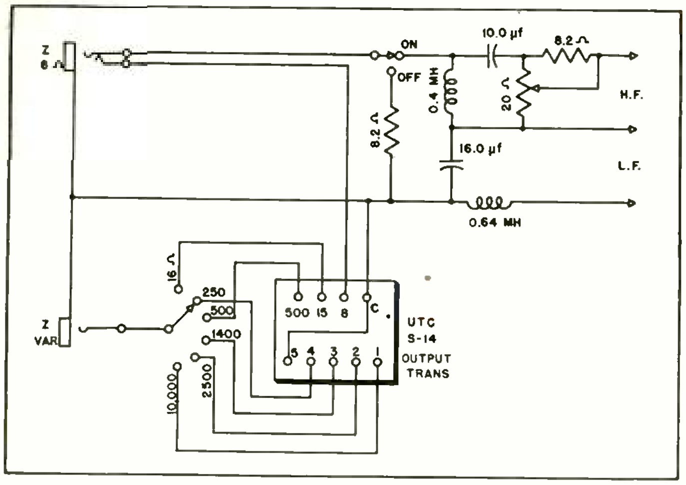

Fig. 5 (left). Schematic of speaker shown in Fig. 4.

TABLE I

DIVIDING NETWORK COMPONENTS

INPUT Z	TWEETERS TYPE	TWEETERS Z	f_c cps	L_1 mh	L_2 mh	C_1 uf	C_2 uf	R_1 ohms
8	1-UNIV	12	2,000	.636	.398	9.95	15.9	10
	2-UNIV	6*	2,000	.636	398	9.95	15.9	15
	RACON	15	1,200	1.06	.662	16.6	26.6	10
10	1-UNIV	12	2,000	.795	.497	7.95	12.7	15
	2-UNIV	24	2,000	.795	.497	7.95	12.7	10
	RACON	15	1,200	1.32	.825	13.2	21.1	15
12	1-UNIV	12	2,000	.955	.596	6.64	10.6	20
	2-UNIV	24	2,000	.955	.596	6.64	10.6	15
	RACON	15	1,200	1.59	.995	11.0	17.6	15
15-16	1-UNIV	12	2,000	1.25	.782	5.07	8.11	30
	2-UNIV	24	2,000	1.25	.782	5.07	8.11	20
	RACON	15	1,200	2.08	1.30	8.45	13.5	25

* UNITS IN PARALLEL

used with this arrangement, or a separate dividing network may be built.

The single unit and horn, Model 4401, may be installed directly on a speaker baffle; or the dual unit, Model 4402, consisting of two similar horns made as a single casting and set at an angle of approximately 30° may be used, the latter giving somewhat better spatial distribution of the high frequencies. This type of mounting entails more work, but the final results are generally more satisfactory. The impedance of each driver unit is 12 ohms, and they may be connected either in parallel or in series, depending upon the impedance of the woofer with which they are to be used. *Figure* 4 shows an experimental two-way system using a 12" Stromberg-Carlson cone and the dual tweeter, together with a dividing network operating at a 2,000-cps crossover. This box, although too small for good bass response, also incorporates a tranformer and switch, and provides input impedances ranging from 8 to 10,000 ohms, making it a general purpose test speaker. The schematic is shown in *Fig.* 5.

The Racon Two-Cell Horn

Still another model is available commercially to provide a simple assembly of a good system. This is the Racon two-cell horn and the unit designed to work with it. This combination operates at a cross-over frequency of 1,200 cps, and requires a dividing network for best results, although a series capacitor will protect the driver unit sufficiently. If this model is employed, it is necessary to provide a suitable means for mounting, such as cutting an opening in the baffle or woofer cabinet.

Any one of these three methods will give added brilliance and definition to reproduction from a high quality amplifier or radio receiver. It must be remembered that additional h-f output will tend to show up any distortion in the amplifier, so it is important to reduce distortion to a minimum when adding a separate high-frequency speaker.

In this connection, it is well to mention one fault common to many two-way speaker system users. Having a means for reproducing highs well, it is not unusual for systems to be operated with an unbalance between the two units, on the apparent theory that "now we have a tweeter so you're going to hear highs!" Good reproduction should be based on the proper balance between treble and bass, with the principal benefit being gained in a truer response in the upper middle range—1,500 to 4,000 cps.

Dividing Networks

The most satisfactory operation of any two-way speaker system will be obtained if a dividing network is employed because of the fact that the sound then comes from either the woofer *or* the tweeter (except right in the crossover range) rather than from the woofer only in the lower ranges and from both together in the upper range. It is definitely recommended that the extra effort be expended to construct the network because of the improved performance.

Table I shows the components to be used with single or dual University

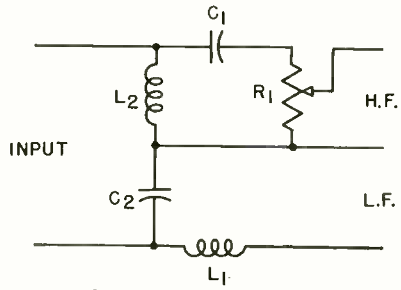

Fig. 6. Crossover network configuration for which values are given in Table I.

tweeters, or with the Racon two-cell horn and unit, with 8-, 10-, 12-, and 15- or 16-ohm woofers. Since the cross-over network is an integral part of the Atlas unit, it is not necessary to build one if this model is used.

The configuration for the network is shown in *Fig.* 6. There are a number of usable configurations, but the one shown is best suited for this purpose since the phasing for the two speakers is most nearly correct at crossover when the horns are mounted on the same baffle as the woofer. The component values were calculated from the formulas:

$$L_1 = \frac{R_o}{2\pi f_c} \text{ henries}$$

$$L_2 = \frac{L_1}{1.6} \text{ henries}$$

$$C_1 = \frac{1}{2\pi f_c R_o} \text{ farads}$$

$$C_2 = 1.6 C_1 \text{ farads}$$

where R_o is the impedance of the network, and f_c is the cross-over frequency.

The values for the balancing potentiometer, R_1, were determined for the condition where the h-f speaker is 6 db more efficient than the cone, which is about normal for average cone speakers and the nearest standard value is listed in the table. A 10-watt resistor with an adjustable tap is ideal for this application, since it should be set when the system is assembled and then left at the optimum position. Impedance changes resulting from this method of adjustment will not affect the performance noticeably.

As to making the inductances, close approximations can be obtained if a certain size of wooden spool is used and specific winding directions are followed for a given wire size. For these values of inductance, a suitable spool size has a core 1-¼" in diameter and offers a ¾" winding space. With No. 16 enameled wire, laid thirteen turns per layer, the total number of turns required for the listed inductances are shown in the curve of *Fig.* 7. No iron

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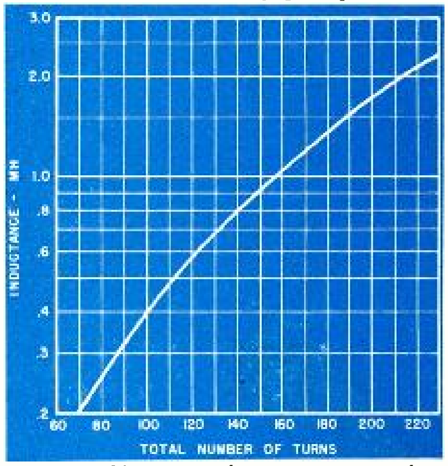

Fig. 7. Chart for determining number of turns to give inductance values necessary for various networks. This chart applies to coils wound on 1¼" core of wooden spool to a width of ¾", using #16 enameled wire with 13 turns per layer.


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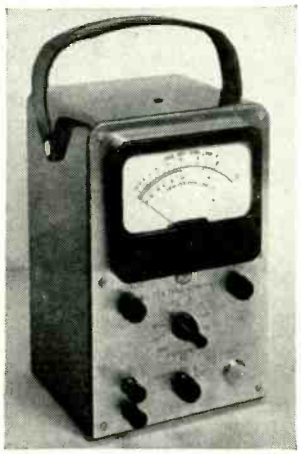
NEW PRODUCTS

NEW TONE GENERATOR

● A small, portable tone generator, designed primarily for use in broadcasting studios, is now in production, the RCA Engineering Products Department tells us.

The new instrument (Type WA-26A), combining a high-quality audio oscillator and sensitive meter, supplies a suitable tone for use in equalizing remote telephone lines.



The tone-generator circuit is an R-C type, allowing selection of ten frequencies from 50 to 15,000 cycles per second. The output is metered and calibrated in dbm. An output of either 150 ohms or 600 ohms may be selected by means of a switch mounted on the front panel. The equipment is supplied with a black leatherette carrying case. The over-all weight with batteries is 9 pounds, 4 ounces.

LOW-COST DYNAMIC NOISE SUPPRESSOR

● Low cost, remote control, and easy installation are features of the newly announced model of the famous H. H. Scott Dynamic Noise Suppressor, Type 110-A. This unit may be added to most existing a-c phonograph or radio-phonograph systems. Designed primarily to minimize the background noise on records that is so annoying to the discriminating music lover, the unit can be used for both AM and FM reception.

Further simplifying the dynamic bandpass principle of suppression, the Type 110-A covers a frequency range equal to or greater than most records on the market. A specially matched pickup is included for wide range performance and low needletalk; and a single control connected with a two-wire cable enables great flexibility in mounting and operation.

Only one connection is necessary to install the Dynamic Noise Suppressor 110-A in its position between the pickup and the amplifier. Being small and compact, it may be located in any convenient spot within the phonograph cabinet.

For further data, write Hermon Hosmer Scott, Inc., Dept. AE, 385 Putnam Ave., Cambridge, Mass.

HIGH QUALITY AMPLIFIER

● A new 10-watt model of the Brook High Quality Amplifier which makes available the advantages of all-triode audio performance to the moderate price field, has recently been announced by Brook Electronics, Inc., 34 DeHart Place, Elizabeth 2, N.J. Within the range of its power rating the new Model 12A series equals the performance of Brook 30-watt models in every respect.

A new Brook circuit development, known as the "transient peak" circuit, permits the amplifier to handle power peaks considerably higher than its 10-watt rating, at the same time holding distortion within the figures published by the manufacturer as applying to constant power output. Intermodulation and harmonic distortion are reduced to negligibility and frequency response is virtually flat from 20 to 20,000 cycles.

Distortioin analysis and complete technical specifications will be supplied by the manufacturer without charge or obligation.

NEW RECORDING HEAD

● The Indiana Steel Products Company announces completion of the development program for their new magnetic tape recording head, Model TD-704. Thoroughly tested and approved, this new head, used for both recording and playback, features high output, maximum frequency response, extremely low hum level, compact size, and efficient performance with all available quarter-inch tapes.

This new Indiana Model TD-704 is used in high impedance circuits; and gives op- Using tape with a coercive force of 300 oersteds at 7½ inches per second, the operating bias level at 40 kc is 1.7 ma and the audio signal current for standard recording

level (as defined by RMA Standards) is 0.15 ma. Recorded in such manner, the signal output from the high impedance playback winding is 5 millivolts. Impedance at 1000 cps is 1000 ohms.

For descriptive literature, write Dept. A-3, Indiana Steel Products Company, 6 N. Michigan, Chicago 2, Ill.

EQUALIZER PRE-AMP

● In response to the need for a flexible, compact pre-amplifier of moderate cost, Brociner Electronics Laboratory, New York City, announces the Model A65 Amplifier and companion Model P6-300 D.C. Power Supply. Permitting compensation for the widely varying recording characteristics in use in the past as well as today, the Model

A65 Amplifier provides 18 combinations of bass and treble curves. Turnover frequencies of 300, 500, and 800 cycles are accommodated, and the high frequency control permits adjustment to any of six response curves, ranging from flat response down to slightly more than NAB slope-off. The amplifier measures only 8¼ long x 2⅞ high x 6 inches over-all; can be mounted on a 3-inch relay rack panel, and is small enough to fit into the phonogrph compartments of most cabinets.

PROFESSIONAL DIRECTORY

— *News* —

FAIRCHILD FORMS RECORDING CORPORATION

● Formation of Fairchild Recording Equipment Corporation to combine the manufacture and sale of a new professional studio quality magnetic tape recorder with the extensive line of professional disk recording and sound equipment of Fairchild Camera and Instrument Corporation, has been approved by the latter's board, it was announced by James S. Ogsbury, president.

The new corporation, with offices at 30 Rockefeller Plaza, New York City 20. will serve the specialized requirements of the radio broadcast industry, as well as those of record and trancription manufacture, and will also apply magnetic tape recording to industrial, motion picture and aviation uses. Furthermore, recent Fairchild developments have made it possible to meet all the exacting requirements of micro-groove recording for the production of the new long-playing records.

Fairchild Recording brings together many well-known figures in the recording equipment field, with Sherman M. Fairchild internationally known for his photographic and aviation developments, heading the group as president. Wentworth Fling, who has been engineering head of Fairchild Camera's recording equipment division, is operating vice president. Jay H. Quinn has resigned as sales manager of Gray Research and Development Corporation to become director of sales and advertising. C. V. Kettering, in charge of recording equipment sales at Fairchild for many years and an authority on the use of this equipment in education, heads a sales division specializing in educational and industrial applications; Theodore Lindenberg, the inventor of the Fairchild dynamic pickup, heads the instrument laboratory; and Gordon Mercer, for many years in charge of manufacturing for Recordisc Corporation, then engineer of Musicraft Records, Inc., later with the Gray company, and recently with Fairchild, is in charge of the electronic laboratory. John B. Wolf leaves the camera company to become secretary-treasurer.

Dr. D. G. C. Hare, recently president of Deering-Milliken Research Trust and an outstanding authority on magnetic recording, has been retained by the new corporation as technical consultant and advisor. During World War II, Dr. Hare was director of the Airborne Instrument Laboratory at Mineola, N.Y., where magnetic recording techniques were applied to anti-submarine detection and guided missile and proximity fuse counter measures..

Mr. Ogsbury said the steady growth of the recording equipment division and the desire for further expansion, especially in the magnetic tape field convinced the management of Fairchild Camera that organization of a separate company would make for a more efficient operation and closer cooperation with users of this type of equipment.

Personnel of the new company has been carefully selected for knowledge and ability in recording and reproduction of sound on both disk and tape, and the staff will devote its entire effort to provide the industry with the finest and most up-to-date precision equipment, Mr. Fairchild said, adding that the firm will soon announce a number of new and advanced pieces of recently-developed items.

AES MOVES

● Audio Equipment Sales, a division of F. Sumner Hall, Inc., has announced the removal of its executive offices from 923 Eighth Avenue, New York, to larger quarters at 153 West 33rd Street, New York 1.

Audio Equipment Sales manufactures Audio-Line Jack Strips, Patch Cords, Jacks, Plugs, and the Audio-Rule — a new device for quickly determining the exact values of resistance for attenuators, without calculation.

RECORDING STANDARDS

● Reports on recording and reproducing standards from four of nine project groups working on phases of the problem are being studied by the National Association of Broadcasters' Recording and Reproducing Standards Committee.

The executive group of the NAB committee, meeting in New York, has also adopted audio frequency response limits for primary and secondary magnetic tape recording standards, to be associated with tape speeds already adopted.

The tape speeds, approved at a previous meeting, were 15 inches and 7½ inches per second, the former for high fidelity work, the latter for portable and mobile equipment.

Continuance of the NAB recording characteristics for both lateral and vertical transcriptions as now used throughout the industry was recommended to the committee. Slight changes in the working of this standard were proposed, which would more realistically define the curves so widely accepted.

The standard as to signal-to-noise ratio on disk recording was somewhat revised and adopted as a proposal. The "Glossary of Terms and Definitions for Disk Recording" was studied in connection with standards on acoustical terminology recently proposed by the American Standards Association (ASA).

The section of the ASA proposal concerning record terminology was found to depart in some respects from NAB proposals. It was agreed that it was advisable to re-study the NAB and ASA proposals with a view toward the early adoption of a coordinated terminology.

The various groups were given specific suggestions to aid them in preparing final proposals for the next meeting of the executive committee, scheduled to be held in New York, October 15.

Proposals prepared at the meeting will be sent to the NAB Engineering Executive Committee for presentation to the Board of Directors at their November meeting.

Preesnt at the meeting were Robert M. Morris, ABC, Chairman; W. S. Bachman, Columbia Records, Inc.; H. A. Chinn, CBS; C. J. LeBel, Audio Devices, Inc.; R. A. Lynn, NBC; G. M Nixon, NBC; W. E. Stewart, RCA-Victor; and Neal McNaughten, NAB.

TWO-WAY SPEAKERS

[*from page* 30]

should be used in constructing the spools, and they should be mounted with brass screws through the center. One pound of wire will make both of the coils for any combination shown except the last one in the table, which will require almost 1½ pounds.

Paper capacitors must be used for these networks, and low-voltage types are quite acceptable, since there will never be more than 25 volts across them, even at 30-watt levels. Many of the surplus houses list suitable values at low prices. If it is not possible to obtain a single capacitor of the desired value, the total may be built up by connecting smaller values in parallel.

Performance Tests

In order to make comparisons of the performance of these systems, two types were assembled. *Figure* 8 shows a single baffle using the Racon unit with a 15-inch Jensen cone while the dual University tweeter and a 12-inch cone were shown in *Fig.* 4. For testing, the Racon-Jensen combination was mounted on a 6 cu. ft. enclosure and the response measured using a warble-tone frequency record through a flat amplifier as the source, the sound output measurement being made with an Electro-Voice

Model 630 dynamic microphone working directly into a high impedance a-f voltmeter. This microphone has an excellent response up to around 11,000 cps and is quite adequate for measurements of fair accuracy.

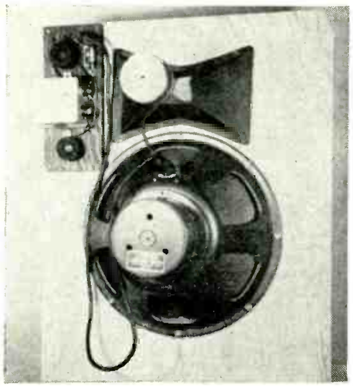


Fig. 8. Baffle with two-cell Racon horn and 15" Jensen cone and dividing network for 1,200-cps crossover frequency.

The smaller unit was measured in its cabinet and the Atlas unit was connected with the Jensen speaker, using the Atlas dividing network and with the Racon unit disconnected. After adjusting levels, the runs were made, and the results were consistently flat with all models. All three units gave appreciable output up to 13,000 cps. so the h-f performance was quite acceptable throughout. Naturally, the small box was deficient in the lower register, due to its inadequate size. The measuring set-up is certainly open to question, since all measurements were made in a normal living room, but the principal reason for making them was to check performance through the cross-over region. Further checks were made with an oscillator, but standing waves made these measurements less reliable.

Subjectively, the two systems with the lower cross-over had more presence on speech than the University model. but on music there was little noticeable difference—except for the poorer bass from the small box, which is only 14"x18"x7" inside. However, any of the three systems was noticeably more realistic than a good single speaker, and the time spent in assembling any of the types will be well repaid in listening pleasure.

RECORD REVUE

[*from page* 28]

control on the amplifier, and/or 2) via a separate filter system with step positions.

There is little doubt that some form of control of this gate-open (ceiling) situation is necessary in any system that will reproduce over 6 or 7 kc. Just where it is to appear. technically, would depend on the response curves attainable in the design. Given similar slopes, the fixed-position and continuously-variable controls are not different in any important way; but if the fixed-filter curves cut off more steeply, there can be a considerable difference in effect between the two. (Note that many amplifiers include both types of action, the boost-attenuate knob for treble, and another for bass, plus the fixed cut-off. usually in the semi-standard positions of 5. 8. 10 and 20 kc.) In my own amplifier I would definitely want to have both types available, for use separately or in combination.

Why a ceiling control? Let us examine the ceiling studiously. This is, again, the situation where we have full volume, gates wide open, no suppression. There are three factors in the input signal, important in reverse order: 1. Tonal range of the useful signal (the music). 2. Noise input. 3. Distortion in the useful signal.

1. Tonal range is a fairly simple business. Since a large number of records have little on them in the higher tonal ranges—virtually all pre-war records—it may be desirable to cut down maximum gates-open (ceiling) tonal response and thus eliminate a lot of noise that comes through where there is virtually no useful sound. Here the ceiling control acts as an alternative automatic filter setting (the floor control being the other) for high-level conditions, when we may want the full range of available musical signal—but not more. This, again, is simply common fixed-filter usage, applied in this case to full-volume conditions only.

2. But suppose we have a really wide range signal from a record—beautifully encased in noise. Not so simple. Here it is, alas, a matter of having your cake and eating it; but what we most need is *flexibility*, to deal with the cake as our taste demands. With ceiling unlimited, every full-volume peak opens up all the way and allows the full noise content to come through—in this case plenty of it. And here a vital psychological principle operates: *continuous, smooth* noise is far less consciously noticeable than noise that abruptly or rhythmically changes. Thus to suppress some noise and let through the rest full-strength actually makes the noise more objectionable than with no suppressor action. This is a delicate situation, and one in which the suppressor is at its weakest. By taking advantage of smoothness in noise content we can usually overlook more of it (and get more highs to our ears) than by suppressing some, *but not all.* And the farther we suppress some of it (the lower we make the floor) the more noticeable, by cantrast, is the unsuppressed noise that comes through at the ceiling.

In such embarassing cases perhaps it is best to cut out the suppressor, entirely and work on the fixed-fiilter principle, for smooth, steady noise. But, I find, it is often possible to gain an advantage over this—*if* you have adequate flexibility in both ceiling and floor, as well as gate sensitivity. You may decrease the gate sensitivity; that allows fewer peaks to get through. *But,* when the gates do open up, especially in sudden, percussive peaks, the noise is still suddenly too great. Far better, I find, to cut down the ceiling, and thus shave off any and all sudden peaks to a mean top level that is guaranteed (according to taste) not to throw sudden blasts and swishes of noise at you, and yet to give an optimum quantity of highs under the circumstances. The over-all technique here is to reduce the stretch between floor and ceiling until the quick changes in noise level are not too objectionable. But with no ceiling limit this is obviously impossible.

3. A similar effect that is considerably more annoying is the distorted signal so often combined with loud noise as above. Without distortion we might tolerate a lot of smooth hiss and even quite a bit of swish (changeable hiss) in order to get through the clean and pleasant highs of a good recording. But there are still a fabulous number of recordings, old and new, that are distorted, often rather badly—and this distortion is frequently at its worst in the

[*Continued on page* 38]

RECORD LIBRARY

In this spot a continuing list of records of interest will be presented. This list specifically does not suggest "the" best recordings or versions. It will draw predominantly but not entirely from postwar releases. All records are theoretically available, directly or on order: if trouble is experienced in finding them AUDIO ENGINEERING will be glad to cooperate. Records are recommended on a composite of musical values, performance, engineering; sometimes one, sometimes another predominates but records unusually lacking in any of the three will not be considered. Number of records in album is in parenthesis.

Pre-War Classics — exceptional recording (restricted range); highest musical quality.

Bach, Organ Works, vol. I.
Albert Schweitzer, organ of All Hallows, Barking.
Columbia M 270 (7)

Bach, Cantata No. 4, "Christ lag in Todesbanden".
Bach, Cantata No. 140, "Wachet auf".
Orfeo Catala, Barcelona
RCA Victor M 120 (5)

Beethoven, Symphony No. 3 ("Eroica").
Columbia M 285 (6)

Beethoven, Symphony No. 7.
Columbia M 269 (5)

Beethoven, Symphony No 9 ("Choral")
Vienna Philharmonic Orch. Weingartner.
Columbia M 227 (8)

Beethoven, Symphony No. 2.
Boston Symphony, Koussevitsky.
RCA Victor M 625 (4)

Beethoven, Septet, opus 20.
(solo players)
RCA Victor M 571 (5)

Handel, Water Music Suite (arr. Harty).
London Philharmonic, Harty.
Columbia X 13 (2)

Haydn, Symphony No. 45 ("Farewell").
London Symphony, Sir Henry Wood.
Columbia M 205 (3)

Haydn, Symphony No. 92 ("Oxford").
Paris Conservatory Orch. Bruno Walter.
RCA Victor M 682 (3)

Mozart, Piano Concerto in A, K. 488.
Marguerite Long, orch. cond. Gaubert.
Columbia M 261 (3)

Mozart, Sinfonia Concertante in Eb, K. 364.
Albert Sammons, violin, Lionele Tertis, viola, London Phil. Harty.
Columbia M 188 (4)

Mozart, Symphony No. 29, No. 34.
Boston Symphony Orch. Koussevitsky.
RCA Victor M 795 (5)

Mozart, Symphony No. 35 ("Haffner").
London Philharmonic Orch. Beecham.
Columbia M 399 (3)

Mozart, Quartet No. 17 in Bb, ("Hunt"), K. 458.
Budapest Quartet.
RCA Victor M 763 (3)

Mozart, Serenade for Thirteen Wind Instruments, K. 361.
Chamb. Orch., Edwin Fischer.
RCA Victor M 743 (3)

peak volume passages, the very ones that are passed through untouched by the suppressor with unlimited ceiling. Moreover, distortion being what it is, even those records that have a highly restricted legitimate tonal range still send distortion through at the higher levels, *well above the nominal undistorted range of the music!*

The trouble that distortion in the signal makes in any suppressor without ceiling control is only too obvious when you consider that since most distortion is worst in the peak passages, the suppressor actually works in reverse—it suppresses the less noticeably distorted highs at low-level passages, lets through to a maximum of 10 kc or so every good burst of loud distortion that comes along. Is it any wonder that many demonstrations of the dynamic suppressor have gone sour? When high distortion is combined with high noise level, the effect of the unlimited-ceiling action is nothing less than dreadful. The number of musically good records that boast this unlucky combination of noise and distortion at peaks is appalling.

The most vital function of the ceiling control, then, is to limit distortion. Without any question, the most important single difficulty I've run into is the highly unpleasant effect of those unchecked bursts, let suddenly through the gates when there is no ceiling to control them—and sounding the worse in contrast with the low-level passages where distortion is sharply controlled by the closing of the gates.

For most purposes a single ceiling filter position at 8000 kc cut-off would take care of a very large part of the distortion problem and a good deal of the sudden swish from too sudden letting through of high-frequency noise. But for records with limited tonal range one is likely to want a lower cut-off, probably at the usual 5000 position; and so a ceiling control on my theoretical suppressor-amplifier would probably have graduated steps at the standard positions of 20 (unlimited), 10, 8, 5 kc, in addition to the attenuator control.

* * * * * *

Needless to say, this discussion of the three basic outward controls desirable on a really flexible suppressor-amplifier ignores many internal problems of design which, though they obviously affect performance of any of these functions, are beyond the scope of any but the laboratory designer himself. I mention briefly two, for reference, on which there is significant disagreement. First, the delay in release time in the gate action. (Attack time isn't particularly controversial; fast enough to open for all music, but slow enough to skip over ticks and pops). Release time, which is easily adjusted via condenser values, is bound to be a compromise to suit many different requirements. The McProud simplified suppressor (*August issue*), for instance, has to my ear an unusually long delay—excellent for percussive music (piano, *pizzicato* strings, etc.) but possibly not as well suited to other musical situations. Delay time clearly has much to do with the quality of "swish" achieved when noise bursts through. Were it possible, I would like to see on my super-duper complete amplifier-suppressor an *adjustable delay control*, probably with, say, three settings. A simple and inexpensive addition for those who delight in extra controls. (*Now it's four!*)

A final area of doubt and disagreement is between those who count on as slow, wide spread to the gate opening action (so that most of the time the gates are working within the partly-opened area) and other designers who deliberately arrange for a very rapid opening action. Either is quite simple to design—the difference is one-hundred per cent a matter of *musical* effect, in the actual operation of the sppppressor. McProud, it will be noted, goes so far as to take his control signal from the output, thereby making it "boost itself" so that the gates open very quickly once action is initiated. With the material of this article in mind it can be seen that this involves some pretty fundamental questions of operational theory. I would like to hear Mr. Goodell and Mr. McProud have this one out, for the benefit of us all! I will venture no opinion at this point, being occasionally a cautious soul.

Recent Recordings

Albeniz, Iberia, Books I and II.
Claudio Arrau, pianist.
Columbia 757 (5)

Two of the four "books" of this series of brilliant Spanish piano works, the top output of their sometimes-trite composer and piano music that rates high in content and energy, as well as mood and entertainment value. This album contains some of the less familiar of the set—*El Puerto, Rondèna, Almeria*, as well as the often heard *Fête-Dieu a Seville Evocation*, and *Triana*, the best known of all. If you like the Spanish idiom—at least the Spanish as we outsiders know it,—this music will appeal enormously. The playing by Arrau is just a bit chilly.

De Falla, Seven Popular Songs
Carmen Torres, soprano. John Newmark, piano.
RCA Victor DM 1223 (2)

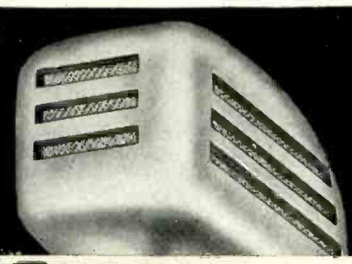
Here is a wilder, more genuine Spanish style, full of the strong oriental feeling that is so remarkably potent in Spanish folk song and dance. These are actual folk songs, set with elaborate accompaniment for piano, beautiful works in themselves, by Manuel De Falla. The strange melodies here will remind one of other music of De Falla, notably the *Amor Brujo* music, of which two recent recordings have been released. Carmen Torres is Spanish, has a "white" kind of voice without as much exotic color as we somehow expect from that country; but she knows the fancy and difficult ornamentation of these songs obviously well. Her mike technique is a bit bumptious—extreme low levels and very high ones—but nothing to spoil the music. Very fine piano recording in the accompaniment.

Schumann, Liederkreis, opus 39.
Helen Traubel; Coenraad Bos, piano.
Columbia MM 752 (5 10")

One of Helen Traubel's earliest recorded albums was of Schumann songs; here she is back at the same stand, after countless recordings of everything from Isolde's Love Death to Home, Sweet Home. A most complete contrast to the Spanish music above; this is fresh German Romantic, youthful, a bit naive, highly European. Traubel does a better job here (with her huge voice) in accuracy and control and economical expression than she did in the earlier work, but it is still a bit monolithic, without the wonderful bounce and flexibility of meaning that Lotte Lehmann or Elizabeth Schumann can put into this sort of music. Not bad at all, though. The recording is vocally beautiful; but—power of a name—the piano is almost entirely inaudible, a faint, tinny clanging somewhere off in the far background. How long will it be before recording engineers learn to keep a stiff upper lip when in the Presense of a Name! (This album recalls the famous Flagstad-Melchior Wagner duet album, with a faint and tinny Symphony Orchestra somewhere far, far

(if racks are employed), and terminal block data are all that is needed to undertake the assembly and wiring of the audio facilities involved.

Finally, practical design consideration dictates the avoidance of apparatus components that are operated to



away. Same story and there are plenty of others, notably among recordings by famous violinists.)

Vaughan Williams, Concerto for Oboe and Strings
Mitchell Miller, Saidenberg Little Symphony. Mercury DM 7 (3)

Vaughan Williams, Mass in G Minor (1923). Fleet Street Choir (London), T. B. Lawrence Decca London EDA 57 (3)

ation at 15000 cycles. An image .0004 inch wide gives similar results in 16-mm reproduction. At one-half the frequency of infinite attenuation the attenuation due to scanning beam width is 3.9 db.

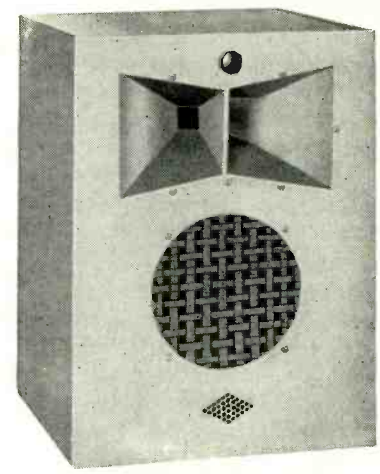
The scanning light beam in a film reproducer must be quite accurately adjusted if good results are to be obtained. The important adjustments are for focus, lateral position, and "azimuth". The last term refers to the adjustment by which the line image is placed accurately at right angles to the direction of motion of the film. Departure from the right angle by more than about five minutes of arc introduces noticeable wave form distortion in the reproduction of variable area tracks. In the reproduction of variable density tracks no wave form distortion is introduced, but high frequency attenuation is increased.

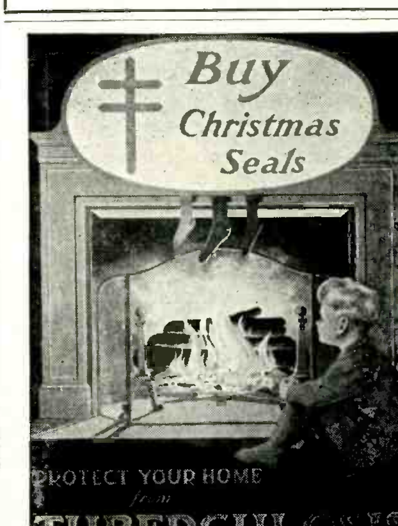
In reproducing variable area records it is necessary to have constant illumination over the length of the scanning light beams. This requirement is not easy to meet in practice. Fortunately a falling off of about 15 per cent at the ends of the beam does not produce objectionable distortion, and most systems perform within this limit when the light source is properly centered.

The photo cell and the circuit by which it is coupled to the amplifier deserve more attention than is sometimes given. If the modulated light beam is brought to the cathode surface of the cell in such a manner that the illuminated area on the cathode changes with the modulation, any unevenness of cathode sensitivity will introduce wave form distortion. If the cell is coupled by a resistance of several megohms, as has been done in many cases, there is considerable attenuation of frequencies as low as 5000 cycles, and harmonic distortion up to several per cent may be introduced at this point. If the operating bias on the cell even slightly exceeds the rated maximum (usually 90 volts) the cell hiss will be high, and the cell may "flashover" due to gas ionization when no film is in the light beam. Under this condition cells deteriorate rapidly. All these defects can be avoided in a properly designed system.

Performance

Photographic recording processes are capable of extremely high fidelity. For example, as reported by Hilliard[6], push-pull 35-mm sound tracks totaling .200 inch in width, which are employed for original motion picture studio recording, are capable of a signal to noise ratio of 74 db. Single 35-mm sound tracks recorded by similar methods





give signal-to-noise ratios of 60 to 65 db. Intermodulation values as low as 4 per cent are not unusual. The recorded frequency range in 35-mm practice is usually 8000 cycles, although on several occasions records which included the range to 15000 cycles have been demonstrated. Extension of the frequency range is a commercial rather than a technical problem, since losses in the 35-mm film amount to no more than 6 db at 10,000 cycles or 11 db at 15,000 cycles when the usual methods of printing are used.

In theatrical motion picture practice the quality standards are considerably lower than those indicated above. The standard reproducing frequency characteristic is shown in *Fig. 8*. The volume range of sound in release prints is usually limited to about 40 db, since experience has shown that low level passages in recording are likely to be masked by audience noise. The sound negatives for release prints are produced by re-recording from the push-pull originals to single tracks, with, naturally, some increase in distortion. Often as many as twenty original tracks are combined into one in this re-recording process.

Original single track records on 16-mm film, as the writer has demonstrated,[7] can carry a frequency range of 10,000 cycles provided the print is made by the optical method. At 10,000 cycles the film losses amount to 15.5 db. As is well known, it is practical to use this amount of "tip up" of the high frequencies. The characteristic required is not much different from the NAB characteristic for disc recording. The signal-to-noise ratio with this frequency range is 45 to 50 db, and harmonic distortion does not exceed 2 per cent with practical tolerances in film processing. Most 16-mm reproducing systems, however, are equipped with inexpensive loud speakers and for this and other primarily economic reasons have been designed to respond well only up to about 5500 cycles. Because of this limitation in reproduction it has been general practice to limit the frequency range in recording to approximately 6000 cycles. Since the 16-mm projector is usually operated in the same room with the audience, it is usual to limit the volume range in 16-mm recording to not more than 25 db (in some cases to no more than 15 db!) in order that the sound level will always override the machine noise. It is to be expected that the now rapidly increasing use of 16-mm sound film in television broadcasting where these reproducer limitations need not apply will lead to a greater utilization of the quality possibilities of this medium.

[The second article in this series will discuss in detail the design of light modulating devices and optical systems for both variable area and variable density sound-on-film recording. Ed.]

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RADIO SYSTEMS

[*from page* 23]

the necessity for tone controls, and the writer is convinced that this circuit is one of the outstanding contributions to the audio art in some time. The combination R_{35}, R_{36}, and C_{22} serve to adjust the total gain to a suitable amount to take advantage of this type of control, and the response curves of *Fig.* 10 represent the output at various settings. C_{22} again adjusts frequency response occasioned by the residual capacitance of the volume control structure.

The two level-adjusting voltage dividers used in this amplifier could have been eliminated and the levels adjusted by selection of plate-load resistors, or by various other means. However, for the experimenter it is thought better to have some variable elements in the circuit. Once adjusted, these networks cause no trouble, yet they provide a measure of flexibility to allow for future modifications.

In this amplifier, 6J7's were used

because of the grid connection separation. The only tube with a-c filament supply is V_7, which is not actually in the signal circuit. The other heaters are wired in a series-parallel arrangement, with the center tap brought out to a terminal. Normally operating from a 12.6-volt d-c supply, terminals 2, 3, and 7 of the terminal strip may be connected together, and 6 and 8 strapped together, and all tubes fed from a 6.3-volt a-c or d-c supply. The regulated 225-volt plate supply for this unit comes from the main amplifier-power unit. Since 6SG7's have a semi-remote cutoff, they are more desirable for the reactance tubes than either the 6SJ7 or 6SK7 types.

No difficulties should be encountered in adjusting this unit, provided the constructor has available an audio oscillator and an output meter, preferably a sensitive a-f voltmeter. All tuning circuits in the suppressor are made adjustable, and the resistor values for SW_2 can be determined easily. Since it is not expected that experimenters will follow the circuit exactly, this description has attempted to trace the steps taken in the design and adjustment of the noise suppressor circuits and of the phonograph equalizer. However, it is believed that if this entire circuit is duplicated, equivalent results should be obtained.

Components

Resistor and capacitor value are all shown on the schematic, *Fig.* 3, but the characteristics of the transformers and inductances used are not so obvious. In the unit shown L_1 is 2.9 H; L_2 is 1.0 H; L_3 and L_4 are 0.45 H. For these coils, UTC VIC adjustable inductances are suitable and readily obtainable. The types chosen should be used as near the minimum inductance setting as possible, to insure the highest Q of the coils, which will then be around 15. Therefore, L_1 should be a *VI-C*15; L_2 should be a *VI-C*13; and L_3 and L_4 should be *VI-C*11 units.

The input transformer used is a Langevin 401B, but suitable substitutions would be UTC A10, ADC 215A, Thordarson 20A05, Chicago BI-1, or Stancor A-4351. The output transformer used is a Western Electric 132C, but UTC LS-27 or HA-113, ADC 315A, Thordarson T-22S92, Chicago BO-1, or Stancor A-3315 should give equally satisfactory results.

The power amplifier and power supply units for this particular system will be described in the next article in this series, which will conclude with a final article on a two-way corner speaker.

ADVERTISING INDEX

Broadcasting's Best...

—now available from stock

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