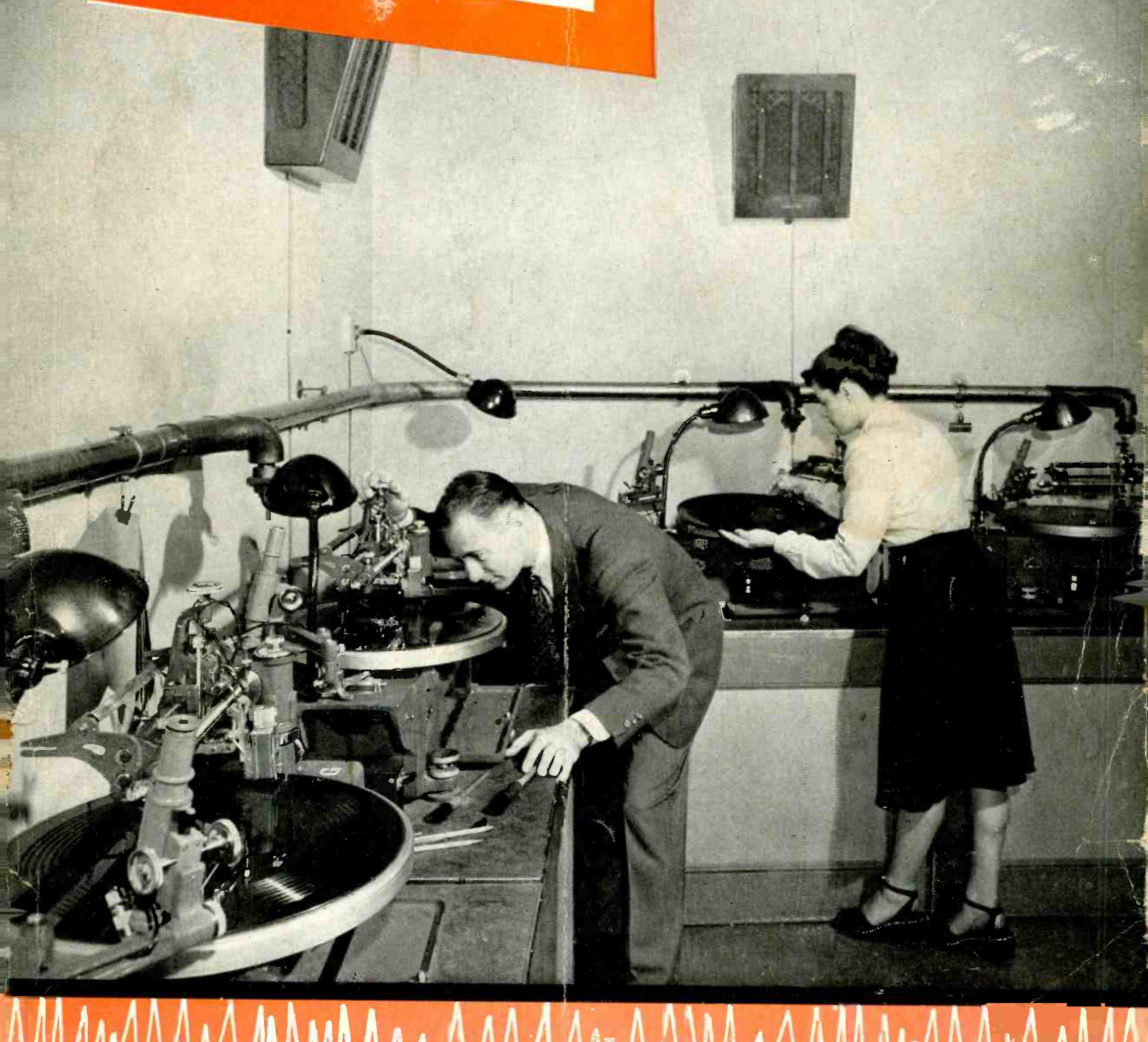
AUDIO ENGINEERING

NOVEMBER 1947



THE JOURNAL FOR SOUND ENGINEERS

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AUDIO ENGINEERING

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CONTENTS

NOVEMBER, 1947

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COVER ILLUSTRATION

Presto 8-D Recorders in operation at the Empire Broadcasting Corporation in New York City.

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EDITOR'S REPORT

BIOGRAPHY OF A SCOOP

• ON WEDNESDAY morning, November fifth, we put this issue to bed. It hadn't gone too smoothly; Winston Wells was still too ill to complete the scheduled installment of his series on electronic organs, and illustrations for another article had become stymied in Washington. But out of the Golden West had come the first of an excellent series on magnetic tape recording, by Dr. Wetzel, as well as Professor Morrical's fine story on mixer networks.

It was a good issue, but we weren't altogether happy. We wanted to run a story about a new noise suppressor, but the engineer was taken off the job to complete some other work. We had tried to get the technical data on the H. H. Scott suppressor, but Mr. Scott had made other commitments. We could, of course, report the lecture to be given by him at the National Electronics Conference, later in the month, but getting complete diagrams would be out of the question. So we closed the issue without a noise suppressor story.

Two hours later, a friend phoned, "I was talking to John Goodell in Saint Paul just a few minutes ago, and he says he's planning to do a story on the H. H. Scott Dynamic Noise Suppressor, which he is manufacturing." Ten minutes later the long-distance operator got my call through to John in Saint Paul. "But," he said, "if I do write this story for you, how long will it be before it is published?" If he didn't mind writing all night, and could get the complete story to me in three days, I told him, I would break open the current issue and run it as a lead.

Two days later, he phoned me. The article was ready, he said, but Saint Paul was being blanketed by a blinding snowstorm, so no planes were flying. Trains would be too slow to meet our deadline, so what to do? But mail planes sometimes fly when passenger planes don't, so we took a chance.

Saturday, November eighth, the article arrived, right on schedule. Just how it got through the thirteen-hundred-mile journey in less than twenty-four hours, in the midst of a blizzard at the sending end and a sixty-mile-an-hour gale and rainstorm in New York, I'll never know, but all honor to our mail system.

But we weren't, even then, altogether out of the woods. The two schematics which accompanied the article had to be redrawn, and one (see page 8) was rather complicated. To get them done in time, we'd have to persuade our draftsman to work over the week-end. Early Sunday morning we drove over to his home—he has not been able to get a phone installed—catching him just as he was leaving to take over an unexpected assignment for the day as usher in a church. How he did it, I do not yet know, but the next morning one of the drawings was on my desk, completed, and the other had been scaled so we could dummy up the story. Then, from our printer in Pittsfield, Massachusetts, came word that the manuscript had been received and was being type-set.

So that is how we were able to bring you the first technical data on the Scott Dynamic Noise Suppressor. Being an editor is fun, sometimes.

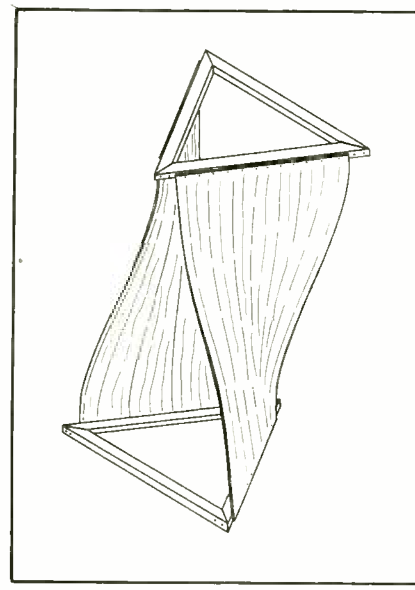
MORE RECORD DATA, PLEASE

• NOW that better phonograph pickups and amplifiers are available, the need for more techinal data on records is emphasized. To use this equipment intelligently, the cross-over frequency and amount of pre-emphasis used in recording should be known to the purchaser. If this is done, then both the manufacturer and user can be assured that the record is being reproduced as well as possible with the equipment used. It is not enough simply to provide the operator with controls sufficiently flexible to accommodate all recording characteristics and to rely on his judgment to select the proper reproducing characteristics. After all, there may be a group listening, each member of which has different hearing characteristics, so that the operator's idea of what constitutes good reproduction may not coincide with that of others with better hearing. Children, especially, usually have keen hearing and it is important that they get good reproduction if they are to cultivate a taste for the best.—*J. H. P.*

TECHNICANA

Sound Diffuser

● Here's a cheap and simple sound diffuser for studios and monitor booths. Make two five-foot equilateral triangles out of 2 x 4s. Then take two 4' x 8' sheets of two-ply and nail the bottom of 4' ends to two of the 5' sides of a triangle, so the sheets make an angle of 60 deg. Then nail the top 4' ends to the other triangle, but twist the sheets of plywood so they make a 60 deg. angle the other way. Thus each sheet is continuously bent in two planes. It can't drum, it stands up, can be moved around, diffuses beautifully,



and if you hide a mike in it you improve the directivity. (Note: the 4' end of the plywood should be centered in its 5' side of the supporting triangles.) These can be slung from cables in any position.

Three at right angles make an astounding improvement in a small studio or monitor booth. The test is to play 1,000 cps through a horn and walk around with a mike and note the VI excursions before and after. I have seen 23 db before, using a ribbon mike, and 3.5 db after, showing the reduction in standing-wave nodes and peaks by adding multipath diffusion. Due to diffraction, the back sides of the flats do a lot of good so they should not be stood against a wall. Three-ply is better mechanically, but it takes a couple of good men to get the thing together.

J. N. A. Hawkins,
844 N. LaJolla Ave.
Hollywood 46, Calif.

EFFECTS OF PHASE SHIFT

● Reporting on a serious group of experiments to determine the effects of phase in monaural perception, R. C. Mathes and R. L. Miller, of Bell Telephone Laboratories, bring forth several interesting phenomena in the September 1947 issue of *J. Acous. Soc. Am.*

These experiments demonstrated that the envelope waveshape of a complex tone—which is determined to some extent by the relative phase of the fundamental and its harmonics—has an important bearing on the tone quality. While it has been generally believed that the quality of a tone was independent of the relation between the fundamental and the harmonics, these experiments were made to prove or disprove this theory as well as a number of others stated by Helmholtz in his "Sensations of Tone" 1895. By means of balanced modulator circuits using techniques similar to those employed in frequency-modulated transmitters, it is possible to provide a variety of signals with entirely controllable phase relations, without the necessity of employing harmonics of a single oscillator to ensure a constancy of phase relation. When separate oscillators are used, it is almost impossible to maintain a definite phase relationship during the experiment.

Observers wore headphones for these tests to avoid the possibility of standing waves interfering with the results, as would most surely be the case if a loudspeaker were used. Among the conclusions arrived at are the following:

1. The tone quality depends upon the components present, and, over certain frequency and volume ranges, upon the envelope shape, which is determined by the relative amplitudes and phases.
2. With no change in relative amplitude of the harmonics present, the ear is able to detect changes in phase only, provided they are sufficiently great.
3. Phase shift determines the degree of harshness of the tone.

A companion paper, appearing in the same issue of the *Journal*, indicates the results of a group of experiments of one of the same authors in connection with the "Masking Effect of Periodically Pulsed Tones." Both papers are extremely complete, and indicate a true scientific approach to well-known problems. The latter experiments have proved, among other results, that repeated pulses of two different frequencies exert a masking effect which is dependent upon the time interval between the pulsing, with the greater masking occurring when the interval decreases.

SOUND FOR 8-MM FILMS

• Attempts to employ standard sound recording practices to 8-mm film have been made to some extent, but from a consideration of the conditions, it is obvious why the results are not comparable to the 35-mm theatre films, or even to the 16-mm home movies, according to Marvin Camras, of the Armour Research Foundation, in an article in *J. Soc. Mot. Pict. Eng.* for October 1947. This article, "Magnetic Sound for 8-mm Projection," was first presented in April at the SMPE convention in Chicago, and has been referred to occasionally in other literature.

[*Continued on page 41*]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915-S*	Crystal	—50	60-7500	7 ½ ft.	11.50
615	Dynamic	—57	55-7500	7 ½ ft.	16.50
615-S*	Dynamic	—57	55-7500	7 ½ ft.	18.00
615-SR**	Dynamic	—57	55-7500	7 ½ ft.	20.00
215	Carbon	—26†	200-4000	48 in.	8.25
215-S*	Carbon	—26†	200-4000	48 in.	9.75
215-SR**	Carbon	—26†	200-4000	48 in.	10.00

*with slide-to-talk shorting switch
**with slide-to-talk relay control switch and hang-up hook
†using input transformer

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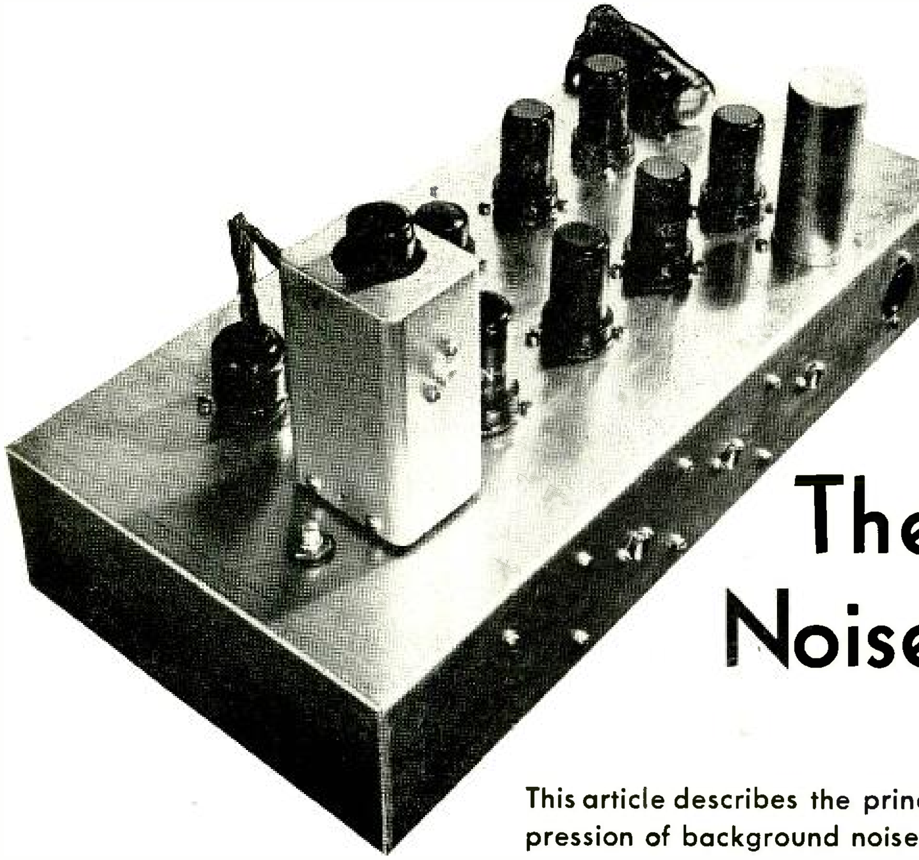


Six-tube Dynamic Noise Suppressor used with Goodell Radio-Phonographs. Shield can with internal shock mounting at left is plug-in preamplifier compensated for Pickering cartridge. 6AL7 indicator tube is used to observe gate action. One section is connected to high-frequency and one to low-frequency reactance tube circuits.

The Dynamic Noise Suppressor

JOHN D. GOODELL*

This article describes the principles and circuits for the dynamic suppression of background noise developed by Hermon Hosmer Scott.

IN ALL METHODS of music reproduction, background noise is a limiting factor. In the reproduction of disc recordings, the irritation of needle scratch was accepted for many years as an evil unavoidably associated with wide range reproduction. Countless methods of eliminating the noise without affecting the brilliance of the reproduction were investigated and rejected.

Many engineers and listeners with a high degree of tonal appreciation found it necessary to develop a psychological rejection of the needle scratch in order to enjoy full range reproduction. This was so prevalent a concept that upon first hearing the Dynamic Noise Suppressor in action at the National Electronics Conference in 1946, many observers were incredulous to the point of refusing to believe that there was no trick involved in the demonstration. Others were so thoroughly "acclimatized" to the presence of scratch that they believed they were observing an appreciable change in the quality of reproduction when the scratch was removed, until they had listened repeatedly to the same passages with the dynamic suppression circuits in and out of the system.

The Dynamic Noise Suppressor will not remove all the background noise. It will not preserve all of the frequencies present in every record under conditions of maximum suppression. It *will* improve the signal-to-noise ratio incredibly, and with proper adjustment will preserve the band pass necessary for full range reproduction to a degree that makes it impossible for most listeners to observe a change in tonal quality even when the suppressor is rapidly switched in and out of the system during the playing of records. The reduction in noise obtained will vary with the hearing characteristics of the listener and with other factors of a psychological nature. Obviously, with a listener incapable of observing anything over 4000 cycles per second, the effective noise reduction will not be as appreciable as with one whose hearing has wide range response characteristics. As a practical matter, the reduction in high- and low-frequency noise is approximately 20 db. The losses involved in quality of repro-

The Minnesota Electronics Corp., 6th & Minnesota Sts., St. Paul 2, Minn.

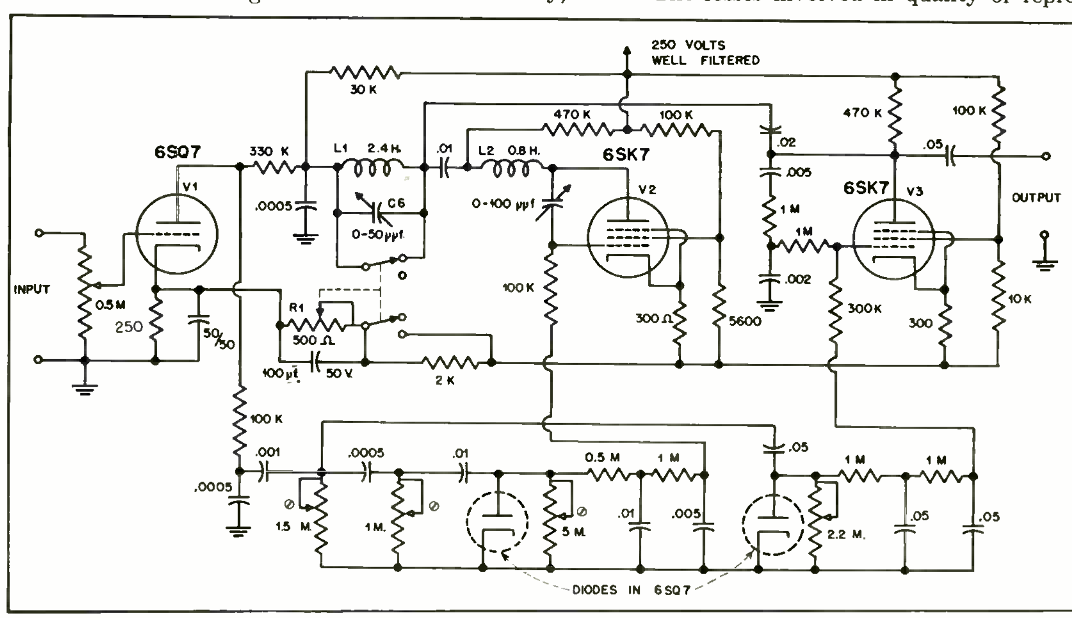


Fig. 1. Dynamic Noise Suppressor. Many variations of switching circuits are possible. R1 is often 5-position switch. Circuit parameters and component values are typical of designs for conventional home radio phonographs but are not necessarily identical to those used by any particular manufacturer. 6SJ7s are sometimes used as reactance tubes. Various voltage amplifier tube types may be substituted. 6H6 may be used for diodes.

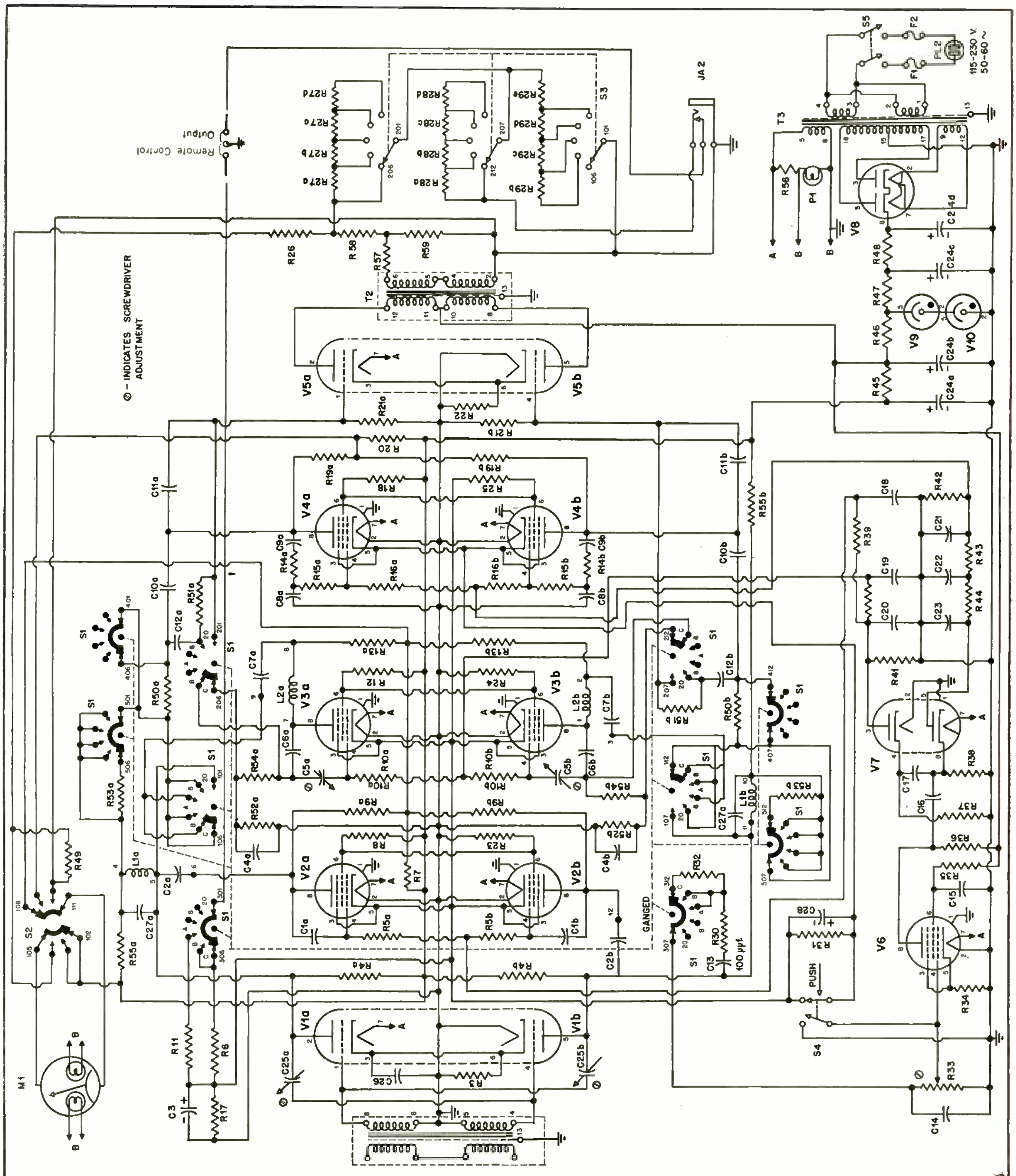


Fig. 2. The 910-A Dynamic Noise Suppressor for broadcast stations. Push-pull circuits conform to standard broadcast engineering practice with usual advantages of push-pull arrangements. Elaborate switching and remote control facilities aid in obtaining optimum conditions on wide variety of records with convenience to operation.

duction are in an order of magnitude that places them in the category of laboratory observation. The design and construction of the noise suppressor for application in any particular amplifying system requires careful engineering, and the adjustment of the controls requires some experience in order to obtain optimum results.

There are countless possible variations of the basic circuits to fit special problems. Each manufacturer producing a Dynamic Noise Suppressor has developed switching and control facilities, as well as more or less elaborate circuits in accordance with the viewpoints of various engineers and the requirements of the equipment in which it is incorporated. There is also a certain amount of individual opinion as to the most desirable conditions of operation, which is typical of all equipment designed in the highly controversial field of audio engineering.

Fundamental Circuits

The fundamental circuits consist of controlled reactance tubes. In *Fig. 1* is shown a relatively simple Dynamic Noise Suppressor using one high-frequency and one low-frequency section. V_2 is designed to appear as a variable capacitance. In series with L_2 it forms a sharply tuned filter shunted across the system. The capacitor between grid and plate of the reactance tube in series with the grid return forms a phase shifting network. A portion of the signal passing through the system is thus applied to the grid so as to produce an alternating current in the tube that leads the signal voltage applied to the plate by 90 degrees. The magnitude of the voltage applied to the grid at any frequency is a function of the impedance of the capacitor between grid and plate. Consequently the capacitive current through the tube and the effective value of the tube viewed as a capacitance will vary with the tuning of this capacitor and will change the resonant frequency of the series LC circuit.

The effective mutual conductance

$$\left(g_m = \frac{\Delta\, ip}{\Delta\, eg}\right)$$

of the tube under static conditions is a function of the screen voltage produced by the dividing network in the screen circuit and the cathode bias resistance, since these values determine respectively the plate current versus grid voltage curves and the static operating point. Increasing the effective mutual conductance creates the effect of making the tube appear as a smaller capacitance and tunes the circuit towards a higher frequency. The opposite effect is produced by lowering the effective mutual conductance. The cathode bias, of course, tends toward degenerative self adjustment in terms of the signal and thereby affects the apparent Q of the entire circuit.

A portion of the input voltage to the system is obtained from V_1, which is simply a voltage amplifier stage, and is filtered and rectified through the d-c control circuits. The rectified voltage is applied to the grid of the reactance tube so that the mutual conductance is under dynamic control of the input signal. These filter circuits must be so designed that the control voltage is obtained from components of the signal in the upper fundamental range of the musical scale. Fortunately the region it is desirable to attenuate to reduce high-frequency noise is in the extreme upper fundamental and harmonic portion of the musical spectrum. Thus, whenever appreciable signal energy is present in the high-frequency region, there must be fundamentals present in the upper mid-range. If the control voltage is derived from the range it is desirable to attenuate, it becomes possible to drive the reactance tube "open" with the noise. If the control voltage is obtained from a low-frequency range where no appreciable high harmonic content exists, the "gate" is driven open during passages when there is only noise present in the high-frequency spectrum.

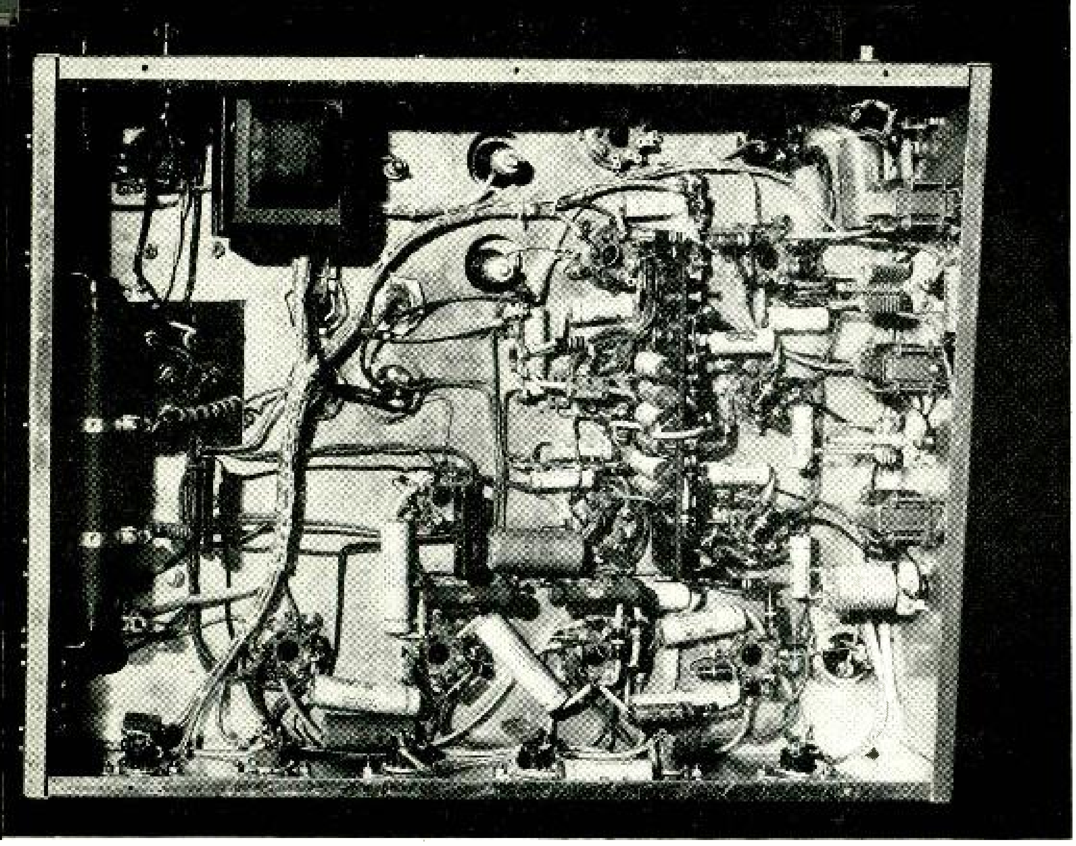
Time Constants

For proper operation of the suppressor it is essential that the control voltages be derived from carefully selected bands of frequencies. The time constants of the con-

Unit	Description
R1	RWV-600
R2	RC21-47
R3	RC21-1000
R4 a & b	RC21-22k
R5 a & b	RC21-100k
R6	RC21-22 ohms
R7	RC21-440 (2-220 series)
R8	RC31-100k
R9 a & b	RC21-470k
R10 a & b	RC21-100k
R11	RC21-110 (2-220 parallel)
R12	RC31-100k
R13 a & b	RC21-470k
R14 a & b	RC21-1 Meg.
R15 a & b	RC21-1 Meg
R16 a & b	RC21-220k
R17	RC21-2200
R18	RC31-100k
R19 a & b	RC21-470k
R20	RC21-800 (470 and 330 series)
R21 a & b	RC21-1 Meg.
R22	RC21-1000
R23	RC21-6800
R24	RC21-5600 (6800 & 22000 parallel)
R25	RC21-5600 (6800 & 22000 parallel)
R26	RC21-5500 (2200 and 3300 series)
R27	RWC-910-1
R28	RWC-910-1
R29	RWC-910-2
R30	RC21-470k
R31	RC21-6800
R32	RC21-220k
R33	RCV-500k-2S
R34	RC20-550 (220 & 330 series)
R35	RC20-470k
R36	RC20-100k
R37	RC20-470k
R38	RC20-470k
R39	RC20-1.0 Meg.

Unit	Description
R40	RC20-1.0 Meg.
R41	RC20-1.0 Meg
R42	RC20-1.0 Meg.
R43	RC20-1.0 Meg.
R44	RC20-1.0 Meg.
R45	RC40-1000-2 RC 30-470 series
R46	RC30-470
R47	RW-500-10 watts
R48	RW-1000-10 watts
R49	RC30-3300
R50 a & b	RC20-10k
R51 a & b	RC20-4.7 Meg.
R52 a & b	RC20-220k
R53 a & b	RC20-15k
R54 a & b	RC20-4.7 Meg.
R55 a & b	RC20-330k
R56 a & b	RW20-5 ohms
R57	RC20-235 ohms (2-470 parallel)
R58	RC20-235 ohms (2-470 parallel)
R59	RC20-687 ohms (1000 & 2200 parallel)
C1 a & b	CM20-200 mmfd.
C2 a & b	CM30-.01 mfd.
C3	CEC-500 mfd.-25v
C4 a & b	CM20-200 mmfd.
C5 a & b	CCV-100 mmfd.
C6 a & b	CM30-2500 mmfd.
C7 a & b	CM30-.01 mfd.
C8 a & b	CM30-5000 mmfd.
C9 a & b	CM30-5000 mmfd.
C10 a & b	CM30-.02 mfd.
C11 a & b	CM30-.01 mfd.
C12 a & b	CM30-.01 mfd.
C13	CM20-250 mfd.
C14	CM20-100 mmfd.
C15	CM30-.01 mfd.
C16	CM30-2500 mmfd.
C17	CM20-500 mmfd.
C18	CM30-5000 mmfd.
C19	CM30-5000 mmfd.
C20	CM30-.01 mfd.
C21	CPM-.01 mfd.

Unit	Description
C22	CPM-.01 mfd.
C23	CPM-.01 mfd.
C24	CEB-20 mfd.-4 sec.
C25 a & b	CCVD-7 mmfd. (dual unit)
C26	CEC-100 mfd.-25v.
C27 a & b	CM20-15 mmfd.
C28	CEC-25 mfd. 25v
S1	SRW-105-1
S2	SRW-25-1
S3	SRW-45-1
S4	SRP-21 x A
S5	SWT-21A
M1	MRD-1 VU Meter
T1	TRAE4-1
T2	TRA4-1
T3	TRIP-1
910A-20	Terminal Strips
910A-21	Terminal Strips
910A-22	Terminal Strips
	HAU-910 Handles for Chassis
	CHL-A-910
(L1	2 CHI-A) 2 coils
(L2	2 CHI-B) 2 coils
PL2	PL-2 Twist-lock Power Plug
	2CAP-2 Connecting Power Cable
	MDU-2 Mounting Plate
	PLJ-4 Jones Plug
	SOJ-4 Jones Socket
V1	6SL7
V2 a & b	6SK7
V3 a & b	6SK7
V4 a & b	6SK7
V5	6SN7
V6	6SJ7
V7	6H6
V8	6X5GT
V9	OD3
V10	OD3



Under chassis view of Dynamic Noise Suppressor incorporated in Goodell Amplifier shown with arrangements for plug-in of control cables in front panel. Trimmers in tuned circuits shown at right have screw driver adjustments. Screw driver adjustment of potentiometer in upper right corner permits compensation of input level from various sources to provide correct drive to gate circuits.

trol circuits must be designed to provide rapid opening of the "gates" so that the brilliance of staccato passages and cymbal crashes will be maintained. The time constants controlling the "gate" closing are equally important. These values must permit sufficiently rapid closing to eliminate the hangover of needle scratch described as "swish," but must not close so abruptly as to have an appreciable effect upon the reverberation following abrupt crescendos. For optimum results on all types of records, elaborate systems may provide switching facilities for these components. The choice of these values is one of the important factors in eliminating any recognition of the gate action by the listener.

Tube Adjustment

The potentiometer in the cathode circuits is used to adjust the static mutual conductance of the reactance tubes, thus setting the minimum band pass. It will be clear that many different tube types may be used for the reactance tubes with somewhat different results and requiring, of course, an adjustment of component values for optimum conditions.

The low-frequency reactance tube V_3 is designed with a phase shifting voltage divider in its plate to grid circuit such that the signal developed on the grid produces an alternation of the current through the tube that lags the voltage signal applied to the plate by 90 degrees. This circuit appears as an inductive reactance shunted across the system. The effective mutual conductance of the tube is varied by a control voltage applied to its grid, thus tuning the inductive current to produce varying degrees of low-frequency attenuation.

To produce the most desirable results, the high-frequency circuits are designed for sharp cut-off characteristics. The low-frequency circuits do not require such sharp attenuation but should be somewhat steeper than that obtained with the usual *RC* arrangements. Clearly, the low-frequency control voltage must be obtained from a portion of the spectrum that does not include high frequency fundamentals, but must not be developed from low frequencies in the range of motor rumbles and other noises it is desirable to attenuate.

In these, as in all analogous filter circuits, the shape of the curves, particularly with regard to steepness of cut-off, will be considerably affected by the input and output terminating impedances.

The tuned circuit consisting of L_1 and C_6 is adjusted to produce sharp cut-off in a region between 10 kc and 16 kc, depending on associated designing. The purpose of this circuit is to attenuate all frequencies beyond the maximum pass band and to eliminate noise that might appear in a region where the resonant curve of the reactance tube circuit rises above its point of sharpest tuning. Obviously it is possible to tune a single reactance tube circuit only over a limited range with the control voltage available. In the more elaborate systems, an additional high-frequency reactance tube is used to permit a wider maximum pass band and tuning of the fixed filter to 16 kc. Theoretically, this principle could be carried on indefinitely if there were any application where frequencies reaching into the ultrasonic range were to be included in the maximum pass band. In the simpler systems, or through switching arrangements, it is often desirable to tune the fixed filter to 10 kc in order to eliminate heterodyne whistles from adjacent radio channels when used for radio reception.

In table model sets, where the low-frequency cut-off is sufficiently high to eliminate motor rumble, a one-tube Dynamic Noise Suppressor consisting of a single high-frequency reactance tube circuit is often adequate. Many engineers have been so plagued with high-frequency needle scratch for years that they have paid little attention to low-frequency motor rumble as a serious factor with reasonably good motor mountings. In experiments with various versions of the Dynamic Noise Suppressor it has been found that elimination of the high-frequency noise immediately makes the low-frequency noises more apparent and more irritating. For this reason alone the low frequency circuit is well worth including. Actually, it is essential to include the low-frequency section for reasons associated with maintaining a correct aural balance. It is well known that the characteristics of human hearing are such that a band pass of correct balance is more acceptable than an extension of either end of the spectrum when one end is attenuated. This is usually expressed in terms of the product of the lowest and highest frequencies in the pass band. By proper coordination of the high and low frequency reactance sections and their associated control circuits, it is possible to maintain a very satisfactory balance in the action of the Dynamic Noise Suppressor. In high-quality systems this characteristic is imperative if satisfactory results are to be obtained.

Other Advantages

There are a number of secondary advantages obtained with the Dynamic Noise Suppressor. Among these is the fact that tendencies toward oscillation through mechanical or acoustic coupling in systems with extended low frequency response are greatly minimized. In the reception of recorded programs from radio stations, particularly FM, where the noise suppressor is not used in the transmitting system, the background noise may be limited just as effectively as in phonograph operation. Other types of interference, such as tube noises in pre-amplifiers, transmission line noises, hum and ordinary static, are greatly reduced in nuisance value.

The remarkable results obtainable with properly designed and operated dynamic noise suppressors are aided considerably by inherent characteristics of human hearing and by the structure of musical sounds. Most music contains passages where the required pass band is very limited. It is also true that the frequency

[*Continued on page 43*]

Design and Use of Mixing Networks

K. C. MORRICAL *

An authoritative article on a subject about which little has been written.

IT IS NECESSARY in many audio and acoustic applications to combine the signals from several sources in a known and controllable manner. A common example is the mixing of the outputs of one or more turntables with that of studio microphones, magnetic recorders, etc. In these applications the connections between the various units of the system are made at standard impedance values, i.e. 600, 250, 50 ohms, etc., and mixing accomplished by means of circuits termed mixer networks.

In general, the complete mixer network contains provisions for level control of the individual channels and of the outgoing signal, as well as performing its fundamental requirement of combining the signals in such a manner that all necessary circuit requirements are met. Since such circuits are seldom mentioned in the literature, and yet are valuable in audio frequency engineering, a brief description of their proper use seems to be in order. In this respect, it is noteworthy that even in manufacturer's catalogs incorrect mixer networks are shown, the use of which would result in certain circuit irregularities as described below.

General Considerations

A mixer network must:

1. Combine the signals of the various sources in such a manner that all circuit requirements are met,

and in addition may:

2. Provide fixed or variable attenuation for each channel in order that the relative signal levels can be adjusted.
3. Provide fixed or variable attenuation for the outgoing signal.

In order to simplify the discussion, and present the results without undue mathematical effort, the following assumptions will be made:

1. All source impedances and the load impedance are purely resistive. This condition is satisfied in almost all applications, particularly at the point in the system where mixing is done. The transformer inputs and outputs of amplifiers usually meet this condition, as do most microphones, pick-ups, etc. Mixing is practically never done at microphone level, but after some amplification, and therefore crystal microphones and pick-ups and condenser microphones are never used to feed the network.
2. All sources have equal impedances. This condition is also satisfied in almost all applications. Instances of departure from this assumption will be discussed later.
3. Source and load impedances are equal. The remarks under (2) apply here.
4. No frequency discrimination is desired in the mixer network or attenuators.

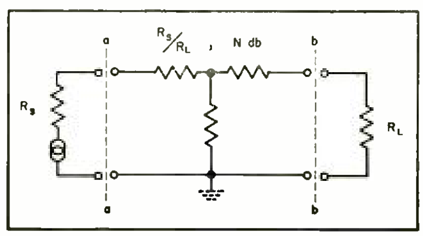


Fig. 1. Connections for a representative attenuator.

Consider now some facts about attenuators. *Fig. 1* makes use of a "T" section although the remarks will equally apply to any other type of section. The description of and the computation of various attenuation networks have already been described.[1,2] The three defining characteristics of the attenuator are:

1. The value of the impedance out of which it operates.
2. The value of the impedance into which it operates, and
3. The amount of loss it introduces. Thus three equations are provided to solve for the values of the three resistances.

If the load, R_L, is connected at plane *bb*, then the generator at plane *aa* will look into a load R_S. It will not see a value R_S if the load is not connected. Likewise, if the source, R_S, is connected at plane *aa*, the load at plane *bb* will look into a source R_L. It will not see this value R_L if the source is not connected, or, of course, if a source of some value other than R_S is connected. Therefore, if either the source from which the pad is fed, or the load into which it operates is incorrect there will be a reflection loss arising from the mismatch. However, if the incorrect load is applied at plane *bb*, the resulting situation is much more serious than if only some reflection loss was being experienced, for the following reason. The resistances in the pad for the desired attenuation were computed on the basis of the specified load resistance, and, therefore, if the load connected is of some other value, the pad will not introduce the loss which is expected of it. That is important enough to state again: *if the attenuator is incorrectly loaded, it*

Operator at controls of Western Electric 25B standard type speech input console.

[1] P. K. McElroy—Designing Resistive Attenuating Networks, *Proc. IRE*, Vol. 23, No. 3, p. 213, March 1935.

[2] A.P. Hill, Resistance Attenuator Networks, Motion Picture Sound Engineering, p. 459, D. Van Nostrand Co., Inc.

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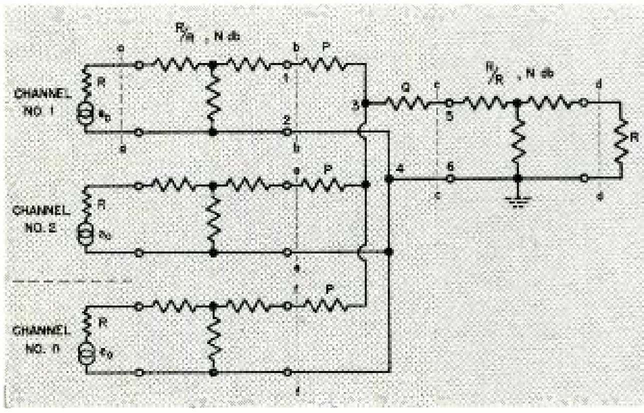


Fig. 2. Connections for first type mixer network.

will not introduce the loss for which it was designed. It is common practice in measurement work to obtain values of unknown quantities by means of adjustable or variable attenuators. This use, where highest accuracy is wanted, is most open to serious errors from improper termination.

Incorrect loading of attenuators, for one cause or another, is the most frequent origin of error in attenuators and mixing circuits. For instance, there was once a circuit shown which connected four 200-ohm controls in series and loaded the combination on 800 ohms. Actually, of course, each control was loaded on 1400 ohms instead of the 200 ohms necessary to yield the correct value of attenuation, and the numbers on the dial were without meaning. After several such experiences, a wise operator will always check this point before using strange equipment.

First Type Mixer Networks

Consider the general case of n sources all of impedance R to be combined and operated into a load whose impedance is R. Such a system is shown in *Fig. 2.* For channel 1 to be properly terminated, it should see a load R at plane *aa*, which it will if the following attenuator looks into a load R at plane *bb*. Likewise, the load should see a source R at plane *dd*, which it will if the master attenuator sees a source R at plane *cc*. The simplest method of accomplishing this is by the use of auxiliary resistors in series with each line. Since all sources have an impedance R, it is obvious that the order is of no importance and therefore the value of the series resistance is the same for all incoming lines. These have been designated as P, while the series resistance in the outgoing line is Q. Therefore there are two unknown resistors requiring two equations.

The impedance to the left at plane *cc* is comprised of Q in series with n parallel circuits of P and R in series. Hence,

$$R = Q + \frac{R + P}{n} \qquad (1)$$

Similarly, the impedance to the right at plane *bb* is comprised of P in series with the parallel combination of $n - 1$ parallel circuits of P and R in series and one circuit of Q and R in series. Hence,

$$R = P + \frac{\frac{R + P}{n - 1}(R + Q)}{\frac{R + P}{n - 1} + R + Q} \qquad (2)$$

(Note that the same result would be obtained by choosing plane *ee* instead of plane *aa*)

Solving these two equations for P and Q results in

$$P = Q = \frac{n - 1}{n + 1} R \qquad (3)$$

The resistances P and Q naturally introduce losses in addition to that of the individual attenuators. This loss, known as the mixing loss, can be determined by considering the generated voltage e_o, and supposing the attenuators to be removed. This can be done, of course, because the connections are on an image-iterative basis,[3] and any number of such networks can be inserted or removed without introducing mismatch. The power absorbed to the right of plane *bb* is

$$P_b = \frac{e_o^2}{4R} \qquad (4)$$

Progressing along the circuit junction by junction, the voltages at each point are determined to be

$$e_{12} = \frac{1}{2} e_o$$

$$e_{34} = \frac{R - P}{R} e_{12} = \frac{1}{n - 1} e_o$$

$$e_{56} = \frac{R}{R + Q} e_{34} = \frac{1}{2n} e_o$$

The power absorbed to the right of plane *cc* is

$$P_c = \frac{e^2_{56}}{R} = \frac{e_o^2}{4n^2R} \qquad (5)$$

and the mixing loss is

$$M.L. = 10 \log \frac{Pb}{Pc} = 10 \log n^2 \qquad (6)$$
$$= 20 \log n$$

The values of P and Q and of $M.L.$ for various numbers of input channels are shown in Table I.

[3]T. E. Shea, Transmission Networks and Wave Filters, p. 87, D. Van Nostrand Company, Inc.

Second Type Mixer Networks

There is another type of mixing circuit which omits resistance Q, and therefore the loss introduced by it, and instead loads the network on an impedance R_L which must be calculated to establish matched conditions. This circuit is shown in *Fig. 3.* Following the procedure used in solving the other circuit, the impedance to the left at plane *cc* is comprised of n parallel circuits of P and R in series. Hence

$$R_L = \frac{R + P}{n} \qquad (7)$$

Similarly, the impedance to the right at plane *bb* is comprised of P in series with the parallel combination of $n - 1$ parallel circuits of P and R in series and one circuit of R_L. Hence

$$R = P + \frac{\frac{R + P}{n - 1} R_L}{\frac{R + P}{n - 1} + R_L} \qquad (8)$$

Solving these two equations for P and R_L results in

$$P = \frac{n - 1}{n} R \qquad (9)$$

$$R_L = \frac{2n - 1}{n^2} \qquad (10)$$

The mixing loss can also be found in a similar manner by determining the voltages. Thus

$$e_{56} = \frac{R - P}{R} e_{12} = \frac{1}{n + 1} e_o$$

and the power absorbed to the right of plane *cc* is

$$P_c = \frac{e^2_{56}}{RL} = \frac{e_o^2}{R} \frac{n^2}{(N + 1)^2 (2n - 1)} \qquad (11)$$

The mixing loss is

$$M.L. = 10 \log \frac{P_b}{P_c}$$
$$= 10 \log \frac{(n + 1)^2 (2n - 1)}{4n^2} \qquad (12)$$

The values of P, R_L and $M.L.$ are shown in Table II for various numbers of input channels.

Remarks

The choice between types of circuits depends for the most part on other design requirements. The first type is by far the simplest to use because of the fact that all associated line impedances are the same value. Thus, the 600-ohm outputs of four different amplifiers can be combined and the network is able to be

TABLE I

Values of P and Q and of $M.L.$ for various numbers of input channels in First Type Mixer Networks.

n	P/R or Q/R $\frac{n-1}{n+1}$	Mixing Loss 20 log n
1	0	0
2	1/3	6.02 db
3	½	9.54 db
4	3/5	12.04 db

loaded on the standard 600-ohm input of the following amplifier if the first type mixing circuit is employed. All that is required is five 360-ohm series resistors connected in star fashion. The mixing loss resulting is 12.04 db.

If a 12 db mixing loss is too great, use may be made of the second type mixing circuit with a loss of only 4.37 db, a savings of 7.67 db as shown in Table III. Four 450-ohm resistors will be necessary. However, the output impedance will now be 262.5 ohms, a non-standard value which it will be found difficult to match. The inset of *Fig. 3* shows the use of a transformer which transforms the impedance R_L to the R of the incoming lines, or, of course, to any other standard value so that standard impedance attenuators and loads can be used. A transformer is not an ideal device, usually producing frequency and phase discrimination, and moreover it costs money. Such a "mixing transformer," as it is called, is supplied by at least one manufacturer for the case of 500-ohm lines; all other values are usually special. In general, the resulting savings in mixing loss is not worth its cost in transformer, weight, and space and can more economically be balanced by higher amplification.

It was mentioned above that the series resistors in the first type mixing are equal and are connected in a star pattern. The star pattern is also true of the lines, since they connect to these resistors.

TABLE II

Values of P, R_L, and $M.L.$ for various numbers of input channels in Second Type Mixer Networks.

n	P/R $\frac{n-1}{n}$	RL/R $\frac{2n-1}{n^2}$	Mixing Loss $10 \log \frac{(n-1)^2(2n-1)}{n^2}$
1	0	1	0 db
2	½	¾	2.27 db
3	2/3	5/9	3.47 db
4	¾	7/16	4.37 db

Therefore the distinction between incoming channels and the outgoing channel disappears, as can be seen by rearranging *Fig. 2* into a star pattern. Signals can be put in on any line and taken out on the remaining lines. An interesting development in this respect is the "inverse mixer" in which a single signal is divided and sent to several different loads. The inverse mixer finds use in large scale, multi-channel sound reinforcement work where a cross-over microphone is employed between those feeding the separate channels. The output of this microphone is divided and sent to both channels, so the transition from one channel to the other as the artist moves across the stage is not as abrupt as it might otherwise be. Another application of an inverse mixer is where two coherent sources are required for balance or comparison circuits.

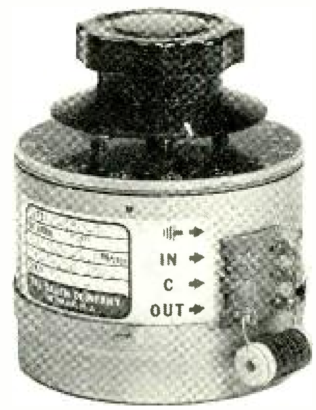
TABLE III

Difference in Mixing Loss due to two types of Mixer Networks.

n	△M.L.
1	0 db
2	3.75 db
3	6.07 db
4	7.67 db

Extension to Unequal Impedances

To prevent unnecessary complication of the foregoing discussion, it was assumed that all impedances were equal, etc. It does happen that in most instances in which mixing of the signals is desired, the input and output impedances are all equal, usually 600 ohms. However, there



Attenuator with attached built-out resistor. (Daven)

is only one resistor added per line, and therefore if unequal impedances are used in the circuit of *Fig. 2*, a number of equations equal to the number of unknowns is available by considering the impedances presented at each of the planes *bb*, *cc*, *ee*—*ff*. There will be $n + 1$ equations instead of the two equations used here, and they will not be expressed in the relatively simple forms of Eqs. (1) and (2), but solution can be made if such a situation arises. The mixing loss in this case would be quite high, high enough to consider the use of transformers to produce a common impedance value.

The impedance transforming action of attenuation networks may also be made use of in mixer circuits. The circuit in *Fig. 1* illustrates this situation wherein the pad can be considered to be matching the load, R_L, to the source, R_S. The minimum insertion loss, however, is not zero but is a function of R_S/R_L. Pads can be used to remove small impedance mismatches, but for large impedance mismatches the remarks of the preceding paragraph apply and resort to transformers should be made. Non-resistive networks are not usually present in the applications under discussion and are beyond its scope.

Precautions

The precautions to be exercised in the use of a single attenuator have been mentioned earlier. The precautions to be used in connection with mixers all derive from this fundamental one:

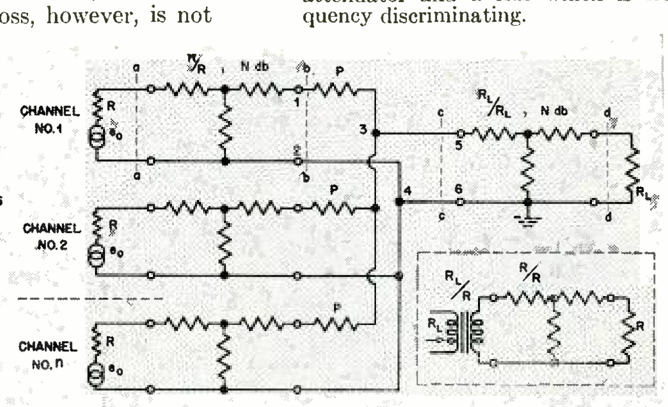
1. Always be sure that the attenuator is operated into the output impedance specified on it. Otherwise the loss will not be as stated and errors will result.

Typical T-pad. (Tech. Labs.)

Some other precautions are:

2. If the load is an amplifier which has a "600-ohm input," make certain that this is meant, instead of an input suitable for operation out of a 600-ohm source. There is a difference, and if the input does not measure 600 ohms, add parallel resistance (or series resistance if too low) until it does.
3. Make certain that no sources are disconnected, or if they are, that upwards of 30-db loss remains in the attenuator of each circuit disconnected. Otherwise the load impedance presented to the attenuators in the remaining channels will be wrong and the indicated losses in error.
4. Level adjustment of loudspeakers, telephone receivers, etc., is best made at the input of the amplifiers supplying them. These devices have strongly reactive impedances which are therefore a function of frequency. The result is an improper load on the attenuator and a loss which is frequency discriminating.

Fig. 3. Connections for second type mixer network.



Review of the Present Status of Magnetic Recording Theory

W. W. WETZEL*

PART I

In this series of three articles, Dr. Wetzel presents the first complete discussion of magnetic tape recording theory for engineers.

IN VIEW OF THE growing postwar interest shown by engineers in magnetic recording and because of the scattered and sometimes inaccessible literature on the subject, a review of the basic theory and the present status of the art is of value. This article is designed to familiarize engineers who will ultimately make use of this recording medium with current views on some of the factors which govern the quality of sound reproduced from tapes and wires.

It seems advisable to divide the article into several short sections. Part I will explain the measurement of some basic properties of magnetic media and the effect of these properties on recordings. Part II will cover phases of recording, playback and erase. In Part III consideration will be given to noise background, distortion, equalization and the effects of media velocity.

This presentation is made on the basis of tape as the recording medium, and the text is illustrated with experimental results obtained on magnetic recording tape. This is a consequence of our laboratory having collected the predominant amount of its information on tape rather than wire.

Minnesota Mining & Mfg. Co., 900 Fauquier Ave., St. Paul 6, Minn.

All of the discussion on fundamental magnetic properties applies to wire as well as tape. This is also true of the record, playback and wipe theory. Some forms of distortion exist in wire which are not inherent in tape, and these will be evaluated in the concluding paragraphs.

Although magnetic recording has a relatively venerable history, there remain a number of phenomena upon which scientists working in the field do not agree, or for which basic explanations have not been found. An attempt will be made to present several points on view on controversial subjects.

Hysteresis Loop Tracer

Preceding a discussion of the pertinent magnetic properties of recording media it may be of general interest to describe briefly the instrument with which these properties are determined. We are interested in measuring constants such as coercive force, remanence and saturating field which are derivable from the saturated hysteresis loop. Useful information is also obtained from the unsaturated and minor loops as we shall show. Some instrument on which hysteresis loops may be studied is an essential to a laboratory studying these magnetic materials.

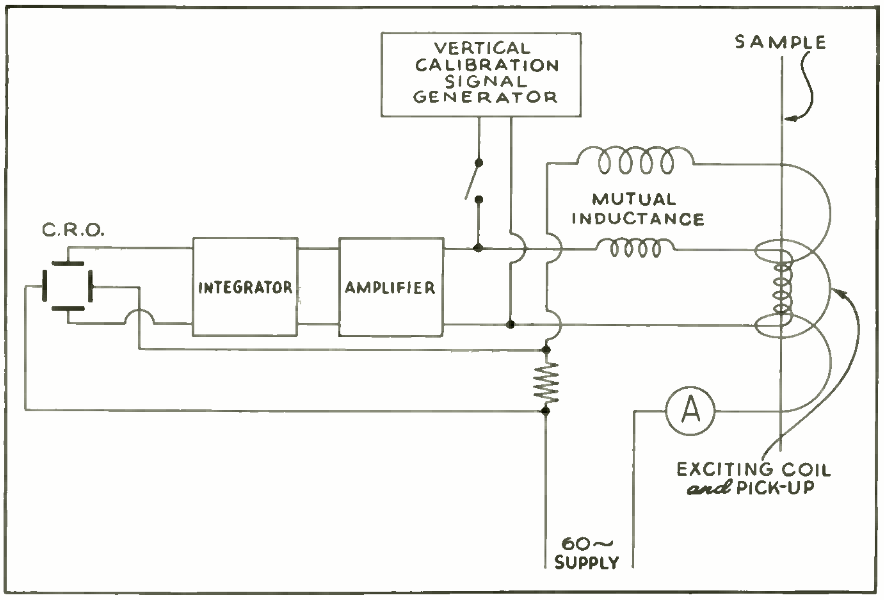


Fig. 1. Schematic diagram of a hysteresis loop tracer.

The requirement of a commercial laboratory where many samples are evaluated in a day is primarily that of speed. A high degree of precision is not demanded by most exploratory work, reproducibility of five per cent satisfying many of the requirements. High sensitivity is essential since the entire magnetic layer on a tape has a cross section of about 2.5 x 10^{-4} square inches and permeabilities are extremely low. The cross section of active material may be increased by stacking tapes in bundles for the purposes of measurement. This has the advantage of providing average values over many lengths of the tape but a marked disadvantage in the time required to prepare the sample for testing. It is considered desirable, because of the ease of sample preparation, to operate on a single tape.

These requirements for speed, reasonable accuracy and high sensitivity in a determination have been met in a number of laboratories by one form or another of a sensitive hysteresis loop tracer. This instrument produces the hysteresis loop on the screen of a cathode-ray oscilloscope. Here the values of interest may be read directly, or, if they are to be studied in detail, the loop may be photographed for leisurely analysis.

The components of a hysteresis loop tester are shown schematically in *Fig. 1.* The essential parts are the exciting and pickup coils, the mutual inductance for balancing the output to zero when no sample is present, amplifier, integrator and calibration equipment. The exciting coil, which supplies the a.c. field in which the sample is examined, is connected in series with the primary of the mutual inductance. The pickup coil is connected in series opposing with the secondary of the mutual inductor which is so designed that its output cancels that of the pickup coil in the absence of a sample. When a sample tape is inserted in the pickup coil a resultant signal is supplied to the amplifier which is proportional to the time derivative of B-H. B, the magnetic induction, is the number of lines of force per square centimeter existing in the magnetic medium. H, the applied field, is a measure of the flux lines per square centimeter of the exciting field in the absence of magnetizable material. The change in flux in the pickup caused by the insertion of a sample is the addition of a B contribution and the subtraction of the H previously existing in the space occupied by the magnetic sample. Hence this instrument responds to the B-H characteristic of the sample.

The output signal for the small samples examined requires amplification before application to the oscilloscope plates. Integration is needed to give deflections proportional to B-H rather than the derivative. H may be calculated from the current and the turns on the exciting solenoid. The horizontal deflecting voltage may be obtained from a series resistor and the gain of the horizontal amplifier adjusted to give a convenient deflection from a given value of H. Similarly the vertical scale may be calibrated through a knowledge of the number of turns on the pickup coil and the application of a standard calibration signal.

It is seen that with calibrated horizontal and vertical deflections proportional to H and B-H respectively a hysteresis loop will be traced on the C.R.O. screen and absolute values in c.g.s. units may be read from the trace.

Instead of reading B-H directly a value $\emptyset$ equal to the increment in flux in maxwells due to the insertion of a sample is usually read. If the cross section A of the sample is known, B-H and finally B may be obtained from the known value H of the applied field and the relation

$$B - H = \emptyset / A.$$

A discussion of the design details of one form of the loop tester is scheduled for publication shortly.[1]

A similar piece of equipment has been described by Long and McMullen.[2]

It should be recalled that in any properly designed permeammeter provision must be made to remove magnetic poles from the region of measurement. This is necessary to eliminate the demagnetizing forces of the poles at the point where B is determined. In loop tracers the effect of poles is made negligible by magnetizing a relatively long piece of tape in the essentially uniform field of a solenoid and confining the pickup coil to a short region in the center of the solenoid. The B-H curve traces are therefore characteristic of the basic magnetic properties of the sample. Having determined the basic magnetic properties the effects of demagnetizing forces, which depend upon the geometry of pole distribution, can be calculated in certain cases. The effects of demagnetizing forces are important in magnetic recording and will be discussed later.

Fig. 2 shows a view of a variation of the Wiegand-Hanson loop tracer. *Fig. 3* shows a photograph of a hysteresis loop for one type of tape. The possibly unfamiliar shape of this curve is due to the fact that B-H vs H is being traced rather than B vs. H. This results in the saturated condition approaching a horizontal rather than a 45° line as the asymptote.

Magnetic Constants

The magnetic constants of interest to us may be read directly on or derived from readings taken from the loop tracer.

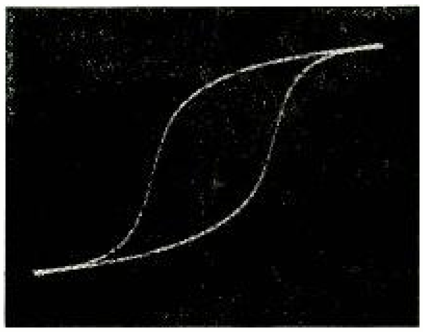


Fig. 3. Photograph of a hysteresis loop characteristic of one type of high coercive force recording tape.

Fig. 4 illustrates a possible hysteresis loop where the applied field H is plotted against $\emptyset$. Three values which concern us, the saturating field, remanent flux and coercive force, may be read directly from the chart. If the value of the cross section A of the active magnetic layer on the tape is known, we obtain B_r, the remanence, from

$$B_r = \emptyset_r / A.$$

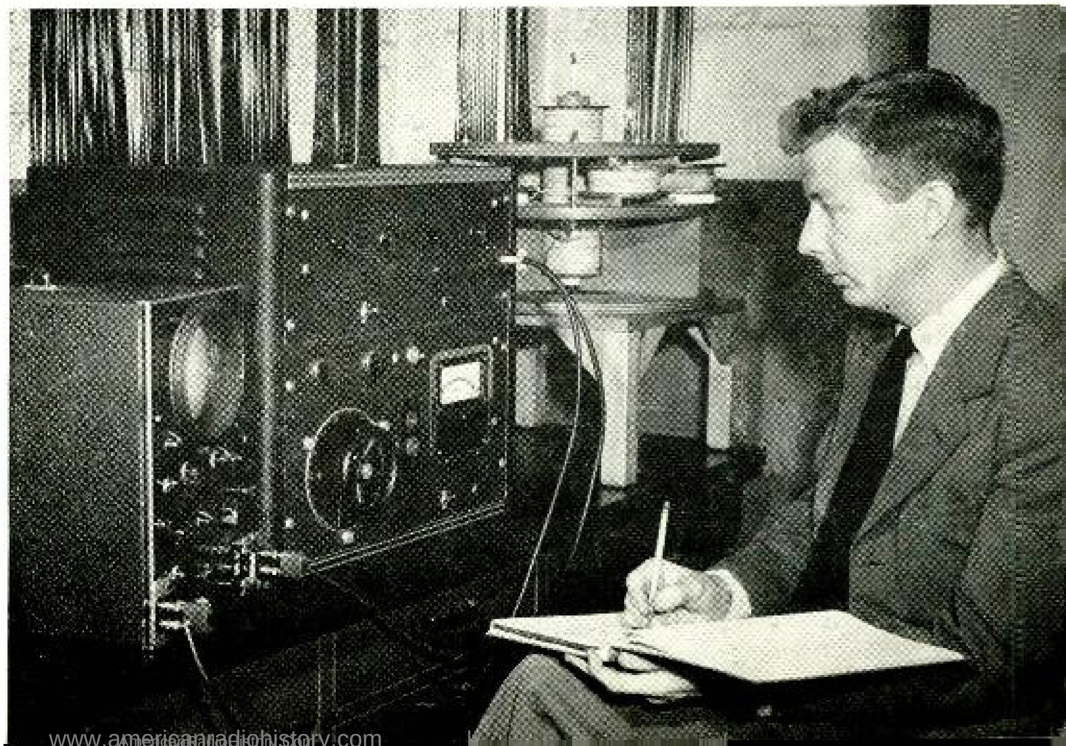
If A is expressed in square centimeters B_r is in gauss. The remanence to a great extent governs the signal output of a tape at long wavelengths.

As will be shown later, the effective output of a tape is controlled at low frequencies by remanent magnetization. The dominant response is due to the material in the active layer immediately adjacent to the playing heads. The contribution of deeper layers becomes increasingly unimportant as depth is increased. For reasonable thickness of active material on a tape it is not true that a 6 db increase in output may be obtained by doubling the thickness of the coating. Doubling the coating thickness will of course double $\emptyset_r$ without affecting B_r. However, doubling B_r will be shown to double the low frequency output. It seems probable that B_r is more nearly characteristic of the behavior of a tape than $\emptyset_r$.

Because of the difficulty of measuring A within an error of $\pm 20\%$ for either wire or tape, it has been proposed that $\emptyset_r$ rather than B_r be used to specify the recording medium. As B_r appears at present to more nearly characterize tape response, this proposal has not as yet been generally accepted and the materials continue to be specified in terms of remanence.

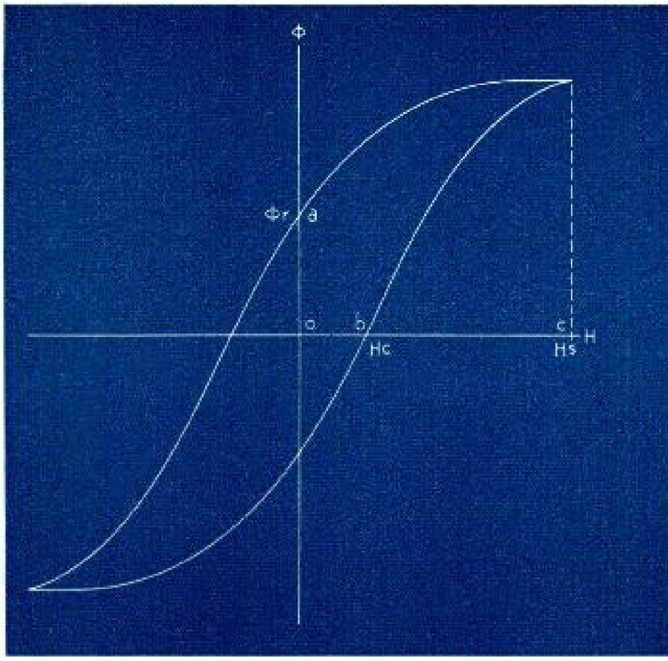
Saturation field H_s, the distance *oc* *Fig. 4*, may be obtained from the curve tracer by direct reading. For practical purposes the lowest value of H which brings $\emptyset_r$ to its maximum value may be defined as H_s. For many materials which saturate slowly this definition does not lead to easily reproducible values. As will be discussed later, H_s is of use in determining the value of the erase field required to obliterate a signal on the tape. Erase fields are now established experi-

Fig. 2. A modified Wiegand-Hanson loop tracer in operation. The long solenoid in the center background is the exciting coil in which a sample tape has been inserted. The smaller symetrically spaced coils are mutual inductors used for calibration and null point balance. The amplifier, integrator and power supply are housed in the relay rack. Traces of the hysteresis loops are observed on the C.R.O. screen. One useful modification in the instrument is an automatic cutoff of the primary current when the temperature of the solenoid reaches 100° C. The temperature sensitive element is a thermistor embedded in the windings of the exciting coil.



1. D. J. Wiegand and M. W. Hanson, "A 60-Cycle Hysteresis Loop Tracer for Small Samples of Low-Permeability Materials," *AIEE Transactions.*

2. T. E. Long and G. D. McMullen, "A B-H Curve Tracer for Magnetic Recording Wire," *AIEE Transactions, 65*, 146, March 1946.



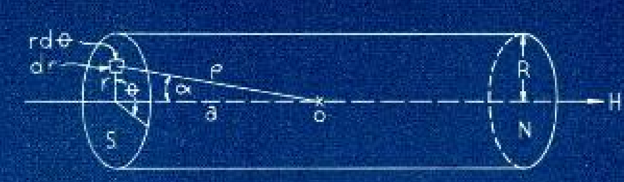


Fig. 4 (left). The form of hysteresis curve obtained on the loop tracer where the flux increment $\emptyset$ is plotted against the exciting field H.

Fig. 5 (above). Axially magnetized cylinder used to illustrate demagnetization principles in recording tape.

mentally for any given tape and H_s is only used qualitatively to predict the relative ease of wipe of two tapes. Some indefiniteness in determinations of H_s may therefore be tolerated.

The coercive force H_c is measured as the intercept *ob* on the H axis. H_c is the field in oersteds necessary to reduce the induction $\emptyset$ to zero after the material has been saturated. It is a measure of the ability of a medium to retain its state of magnetization under the influence of demagnetizing fields and, as will be explained later, is a measure of the short wavelength output of a tape.

It should be noted that, through usage in magnetic recording laboratories, it has become customary to express the coercive force of recording materials as the value of the field at which $\emptyset = 0$ rather than the value at which $B = 0$. From the equation relating the quantities

$$B - H = \emptyset / A$$

we see that the definitions approach equivalence only when B is large compared with H, i.e., for high permeability materials. In the low permeability materials, such as we are considering, the two definitions lead to values which differ by large percentages.

Demagnetization Effect

We have so far considered plots of $\emptyset$ where measurements were obtained under idealized conditions in a hysteresis loop tester. The conditions are idealized to the extent that terminating poles in the tape were kept remote from the region of measurement. This was done to evaluate the basic properties of the material independent of geometry of pole distribution. In recording, this ideal situation is not generally realized. While in a loop tracer the dominant poles are several inches from the region where induction is determined, the separation of the poles from the point of maximum induction in a recording is measured in tenths to thousandths of an inch. In recording, effects of demagnetization forces are usually negligible at low frequencies but become considerable at high frequencies. An explanation of the phenomena will help in understanding the behavior of magnetic records at high frequencies.

Instead of considering directly the demagnetizing forces occurring in a recording tape where geometry complicates the mathematics it will be helpful if we consider the simplified problem of demagnetizing forces in an axially magnetized cylinder. The principles are the same and general conclusions for tape may be drawn from the analogy.

Suppose a cylinder of magnetizable material is placed in a magnetic field H so that the axis coincides with the direction of the field as shown in *Fig. 5*. The cylinder will become magnetized with poles appearing over the entire surface. Let us make two simplifications, which will not affect our general qualitative conclusions, by assuming that poles are formed only on the ends of the cylinder and that the density distribution over each end is uniform and equal to m unit poles cm.[2] We are interested in evaluating the induction B at the center O of the cylinder since the higher the value of the induction the g r e a t e r the pole strength. (In a recording tape the output signal is proportional to the pole strength and therefore to B.)

In the complete absence of poles the induction is given by $B = \mu H$ where μ is the permeability of the medium. When poles are present they act to decrease the effective value of H. The presence of an elemental south pole of strength $m\, r\, dr\, d\theta$ on a point at the origin is to decrease the effective value of H by an amount

$$mr\, dr\, d\theta \,/\, \rho^2 \cos \alpha$$

or

$$amr\, dr\, d\theta \,/\, \rho^3$$

For the decrease in effective field due to the poles covering one face of the cylinder we get:

$$am \int_0^{2\pi}\!\!\int_0^R \frac{r\, dr\, d\theta}{(a^2 + r^2)^{3/2}} = 2\pi m \left(1 - \frac{a}{(R^2 + a^2)^{1/2}}\right)$$

or

$$2\pi m\,(1 - \cos A)$$

Where A is the angle subtended at the center between the axis and the circumference of the cylinder end. Since there are two faces which have identical effects at the center, we have for the demagnetizing force $4\pi m\,(1 - \cos A)$, and since it acts to oppose H, the equation for B is:

$$B = (H - 4\pi m\,[1 - \cos A])$$

It can be seen if the cylinder length is increased, keeping the radius constant, $\cos A \rightarrow 1$ and the term due to the demagnitization force becomes small. This is analogous to the situation in the loop tracer where the angle subtended by the ends of the tape at the center of the pickup coil is small enough to permit disregarding the demagnetizing force in our measurement of $\emptyset$.

As the cylinder is made shorter, A increases toward 90° and the demagnetization force increases although never actually reaching the value of H which gives rise to it.

With different geometry the same condition exists in magnetic records. With very long wavelengths where the poles are far removed from the point of symmetry, the value of the induction B is not affected. As the wavelength on the record is decreased, the demagnetizing forces increase, lowering the induction without, however, being able to reduce the value completely to zero.

Through considerations of the effects of demagnetizing forces it is possible to draw conclusions concerning the effects of remanence and coercive force on the playback characteristic of tapes. Theoretically at least, it is possible to obtain complete quantitative data on tape records from B vs. H curves and demagnetization constants. The measurements are somewhat tedious and the results as yet not as satisfactory as actual record and playback tests. The approach through calculation remains attractive since it offers a means of evaluating tapes without involving the recording and playback head characteristics.

It is evident in the example of the cylinder that the greater the induction B in a magnetic material the greater the value of m as each line must terminate at a pole. One can show that the demagnetizing field H_d for a given distribution of poles is directly proportional to B, or

$$H_d = -CB$$

This equation, where C the demagnetizing constant contains all the geometry, can be combined with a B vs. H curve, which defines the magnetic behavior of the medium freed from geometry, to obtain the effect on B of demagnetizing forces. This may be done graphically as shown in *Fig. 6*.

Lines 1 and 2 represent plots of H_d against B for two values of geometry or wavelengths on a tape. Line 1 where the value of C is small represents the effect at intermediate wavelengths. Line 2 where C is large gives the demagnetizing field as a function of B for a relatively short wavelength. The points of intersection of each line with the hysteresis loop defines the point of equilibrium for the material characterized by the loop under conditions of geometry and demagnetizing force defined by the line.

For very long wavelengths where $C = 0$ an *ac* saturation field recorded on the moving medium will develop a maximum induction of B_r gauss. Intermediate and short wavelengths will result in decreasing remanent inductions of B_{r1} and B_{r2} respectively. If a material were to double its coercive force, B_r remaining constant, the intersection of the hysteresis loop and Line 2 would occur at about twice the value of B_{r2} illustrated. The low wavelength or high frequency output would in this case double without much effect on the low frequency response. Conversely, doubling the remanence while holding the H_c value constant would double the low frequency output with little effect on the high frequencies.

In a later section these predictions on the effect of coercivity and remanence will be shown to hold in terms of frequency response and output level on record-playback tests. It will be shown that, within limits, something may be done to tailor-make a tape to response specifications through the selection of remanence and coercive force of the active material.

Increase of Induction During Replay

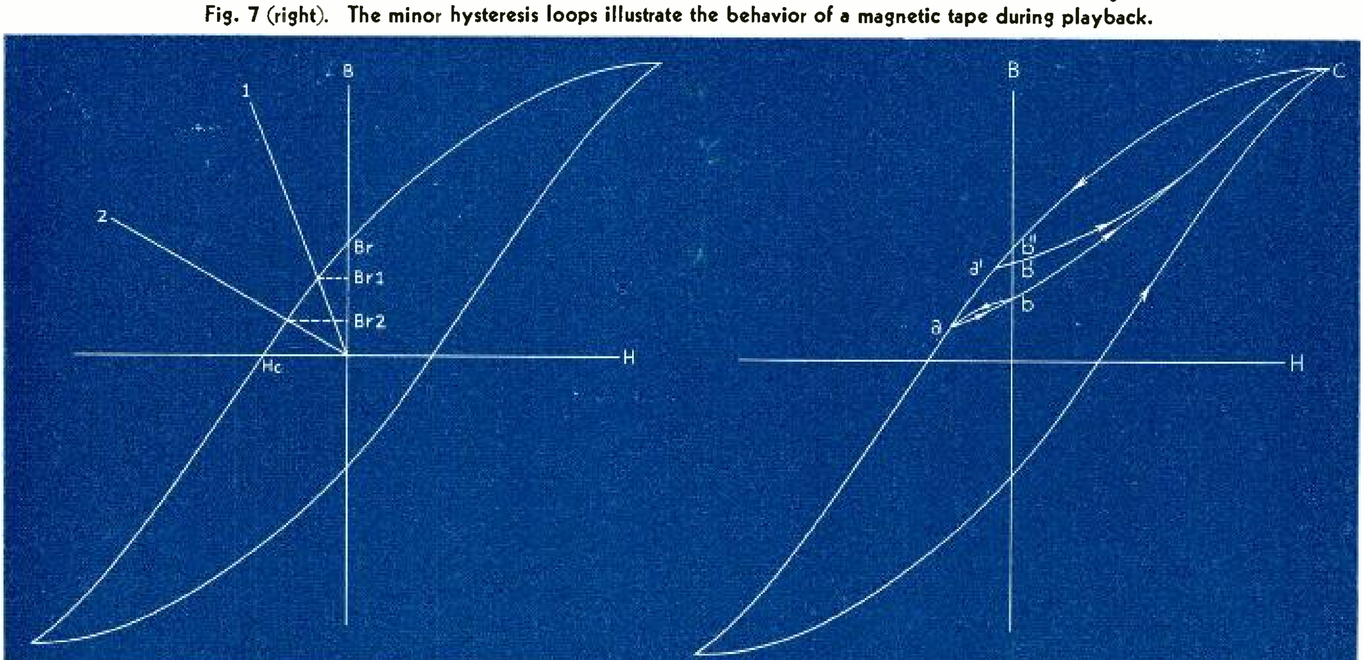
We have examined some of the effects of the demagnetization forces in a tape after it has passed the recording head. Even though the same magnetizing forces are employed (we have been considering saturation fields), the remanent flux in any magnetic recording medium drops as the wavelength is decreased. There is an additional effect on playback where the magnetized material contacts the playback head which tends to restore the output to some degree. The per cent recovery in induction will be shown to become greater the shorter the wavelength recorded.

The recovery effect during replay can be understood by reference to the minor hysteresis loops *ac* and *a'c* shown in *Fig. 7*. If a hysteresis loop has been established and through some means we reverse the field at point *a* after the material has passed from *c* to *a*, instead of retracing the path *aa'b''c*, the state will traverse a minor loop *abc*. If a material, such as the rod of *Fig. 5*, has been brought to saturation (point *c*, *Fig. 7*), and the field H removed, the magnetic state traces the curve *cb''a*, coming to equilibrium with the demagnetization force at point *a*. If we then remove the demagnetizing force, neutralizing the poles by placing a "U" shaped keeper in contact with the faces of the cylinder, the condition of magnetization changes. The path described is *ab* on the minor loop and the new equilibrium point is *b*. A portion but not all of the induction lost through the demagnetizing force is recovered through the application of the keeper.

During playback of a wire or tape, the poles on the medium are to a large extent neutralized by the core of the playback head and the effective induction increased, although never to the point which would have been achieved had demagnetization forces not been present originally. This increase of induction represents useful flux increase as the output at the playback head depends on the value of B in the tape at the time of contact with the head, not the value of B either before or after contact.

Neutralization is effective only as long as the poles on the tape are in contact with the playing head. After passing the head the original state of magnetization is restored without loss to the magnetic record. The analogous case in the magnetized cylinder is the removal of the keeper. The magnetic state of the cylinder then passes from *b* to *a* along the upper branch of the minor loop as indicated by the arrow. [*Con. on page 39*]

Fig. 6 (left). This normal hysteresis loop is used to show the effects on Br of two conditions of demagnetization.
Fig. 7 (right). The minor hysteresis loops illustrate the behavior of a magnetic tape during playback.



Two-Way Speaker System

C. G. McPROUD*

PART I

The first of three articles describing the design and construction of an excellent two-way speaker system.

THE THEATRE-TYPE loudspeaker system, commonly called "two-way," has long been recognized for a number of reasons as the optimum arrangement for the reproduction of speech and music. By providing separate speakers for the low- and high-frequency ranges, each designed for its own particular duties, no compromises are necessary in either to cover an extremely wide band of frequencies. The clarity of speech reproduced by the two-way system is undoubtedly the result of a small, relatively stiff diaphragm which handles the bulk of the speech frequencies, or at least their harmonics. The distribution of sound energy over a wider angle than is possible with a single-cone direct radiator gives a more uniform characteristic over an entire room, and the low-frequency cone can be sufficiently flexible to permit the wide excursions necessary for efficient bass reproduction.

*Managing Editor, *Audio Engineering*

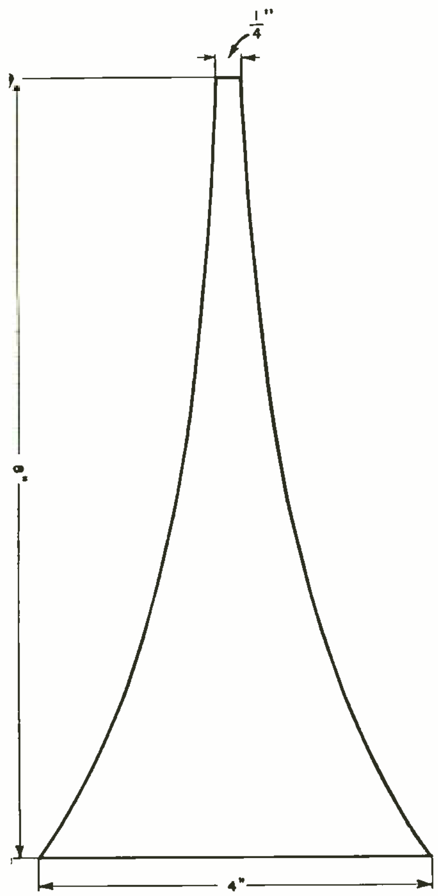


Fig. 1. Developed section of exponential horn, doubling in cross-sectional area for each inch of length along the axis.

Many experimenters, engineers, and ordinary listeners would use the two-way system if it were not for its relatively high cost. The simple baffle or the reflexed enclosure can be constructed easily by almost anyone who is reasonably handy with tools, but the problems of multicellular horn construction appear to be difficult. However, once the method of building the horn is learned, it is seen to be quite simple. After getting together the necessary tools, a complete horn should be built in about eight hours' time. If the constructor plans to make a number of sets of horns, he will do well to make his own patterns for the throat coupling. For the builder who wishes only one such unit, it is simpler to buy the throat ready-made.

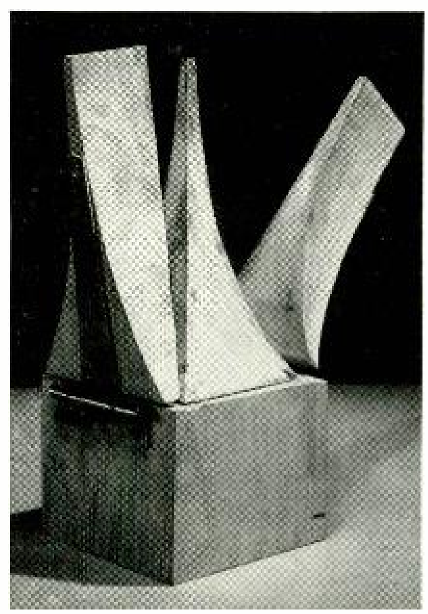


Fig. 2. Wood forming block used for shaping and soldering the individual horn sections to uniform dimensions.

Description

For the benefit of the newcomer, a two-way system consists of a low-frequency cone speaker in a suitable baffle, a high-frequency diaphragm-type unit with a suitable horn, and a dividing network to channel the low frequencies to the cone and the high frequencies to the horn unit. The design and construction of dividing networks has been covered previously in these pages,[1] and will not be repeated in this series. The bass speaker may be housed in a conventional reflexed cabinet, or in an infinite baffle, or in one of the more elaborate horns such as the Klipsch. The last is ideal, but it is bulky, requires a cross-over at least as low as 500 cps, and since the walls of the room become a part of the horn, its use in apartments is frowned upon—usually quite vigorously —by one's neighbors.

The high-frequency section of the speaker comprises the unit itself, coupled to a horn which serves to load the diaphragm and to distribute the sound over a wide angle. The dividing network is a relatively simple circuit arrangement which can be assembled quite easily by the average constructor.

Of all the components required for a two-way system, the only ones offering any apparent difficulty are the high-frequency unit and the horn. The information contained in this article covers the construction of a multi-cellular horn, and satisfactory speaker units are available at a cost of about $15. Allowing about $5 for the shaping block, and $3 for the throat casting and the machine work on it, the total cost of a high-frequency horn and unit should be less than $30. The quality of reproduction which can be obtained from a system of this type should easily justify the cost and labor involved.

Horn Requirements

Two basic types of horn construction have been used to provide exponential loading for the high-frequency unit. One comprises a single exponential horn with a number of partitions which aid in the distribution, while the other consists of a number of individually-exponential horns mounted in a group with their throats and mouths joined to provide, essentially, a single opening at each end. Thus, the unit is coupled to the joined throats, and the joined mouths serve as the distributing area. This latter form of horn appears to be the simpler to build, since each separate horn cell is identical with every other one.

One manufacturer of these horns has built dies in which the entire assembly can be molded of Bakelite in one piece, but

[1] "*Design and Construction of Practical Dividing Networks*," *C. G. McProud, Audio Engineering, June, 1947.*

this is out of the question for the builder of one set of horns. A die for this purpose costs upward of $5,000. Once the die is made, however, horns can be turned out quite cheaply.

The grouped horn sections must be coupled to the opening in the high-frequency unit. This opening is usually circular, and the outside of the throat fitting is threaded. Joining the throats of the separate horns to the speaker units requires a throat casting. This casting may be made by any foundry from a pattern which is not hard to make. The first pattern the writer ever made was for a horn throat, and the casting obtained was perfectly satisfactory, much to the amazement of the writer (and probably the foundry).

Horn Design

Without going into the primary development of the number of individual cells required to obtain a suitable angular distribution (since it was assumed that if one commercial two-way system had eight cells, the same arrangement should be suitable for the homemade set), let it be said that the desired grouping was the 2x4 horn, consisting of eight cells. This gives a distribution of approximately 100° in a horizontal plane, and about 50° in a vertical plane. It also furnishes a reasonably-sized unit which is not too bulky for the average living room, and which provides satisfactory coverage of most of the listening area.

The formula for an exponential horn is based upon the requirement that the area of the cross-section must double in a given length along the axis of the horn. This length controls the cut-off frequency. A second requirement for the individual horn sections is that the perimeter of the mouth shall be not less than one wavelength of the lowest frequency to be reproduced. The cross-over frequency was selected at 900 cps on the premise that this value approximated the more conventional 800-cps cross-over, yet provided a slightly smaller horn with a wider angle of distribution.

With the 900-cps cross-over frequency, a mouth perimeter of 14.9 in. is indicated. Thus if the mouth were 4 in. square, the horn size will be sufficient. At the throat, it should have a dimension of not less than ¼ in. square, since any smaller opening would be difficult to work with. This provides a total throat area of 8x¼x¼in., or 0.5 square inches.

It has been determined that when an exponential horn doubles its area every 12 inches of length, it will reproduce satisfactorily a frequency of 64 cps. If it doubles its area every 6 inches, its cut-off frequency is 128 cps, and so on. For a horn capable of reproducing a minimum frequency of 900 cps, the area should double at intervals of 0.854 in. Resorting to round figures, a suitable interval is selected as 1 in., which corresponds to a frequency of 768 cps.

Using these figures, and contemplating the use of a horn of square cross section, the over-all length of each section is determined to be 8 in., and the longitudinal section of the individual horn may be plotted as in *Fig. 1*, which is half scale.

Now comes a gimmick which simplifies the construction of a set of horns. When perfectly square horns are joined, with the center lines of the various horns each being radii of a sphere, there is a narrow diamond-shaped opening between adjacent vertical pairs of horns at the mouth. This opening must be covered, since some deadening material must be employed around the horn sections to prevent resonances. However, if the shape of the mouth is changed slightly to a trapezoid, the diamond-shaped opening between the pairs reduces to a straight line, and the

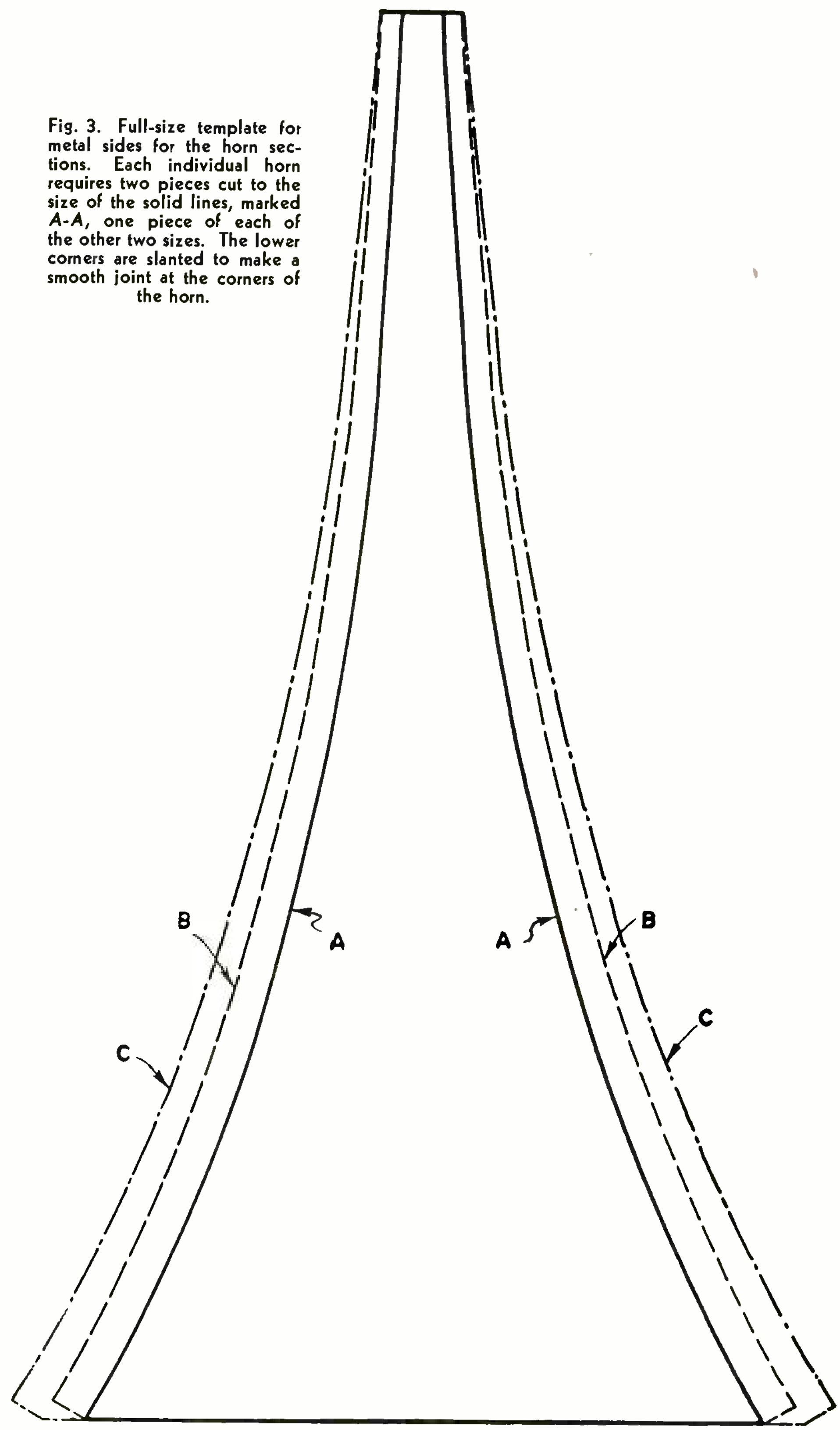


Fig. 3. Full-size template for metal sides for the horn sections. Each individual horn requires two pieces cut to the size of the solid lines, marked *A-A*, one piece of each of the other two sizes. The lower corners are slanted to make a smooth joint at the corners of the horn.

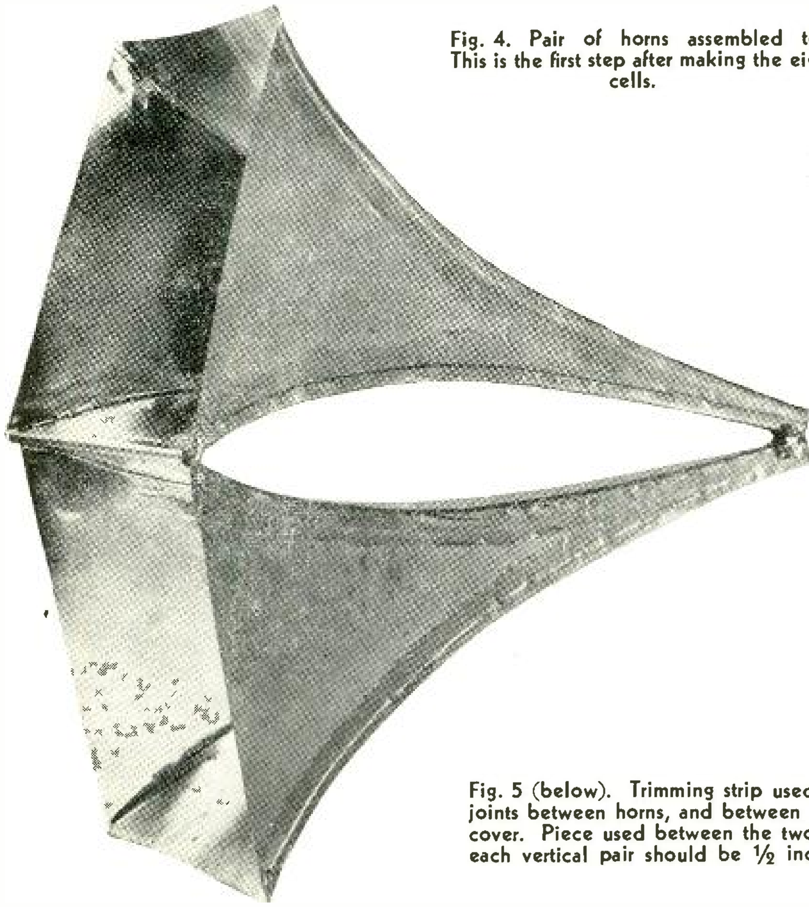


Fig. 4. Pair of horns assembled together. This is the first step after making the eight horn cells.

assembly is simplified considerably since a simple folded strip of metal may be used to fasten the units together. By a series of calculations, it has been determined that the mouth of the individual horn should be 4 in. on each of the three sides, and 4½ in. on the fourth. The long dimension becomes the dividing line between the two cells of each vertical pair.

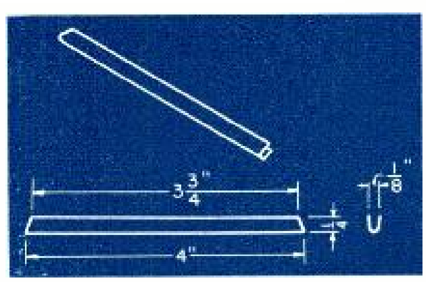
This shaping of the horn section takes place gradually throughout the entire length. Thus at both the mouth and throat ends, the horns join smoothly without any intervening spaces.

Forming the Horns

The method of forming the individual sections ensures that each will be of the proper shape, and that the joints are well soldered. Early attempts at horn making without the forming block resulted in poorly shaped sections, and they were difficult to solder together without any leaks.

The forming block, shown in *Fig. 2*, consists of a solid block of wood, preferably maple or birch, shaped to the predetermined curves, and equipped with two clamping blocks. To make such a block, use a piece of wood 5 in. square and 12 in. long. On one side, lay out the horn shape commencing with a ¼-in. throat at the top and a 3 15/16-in. width at the lower end. The shape should be centered vertically on the block, with the throat at one end. On the opposite side, lay out the horn shape again, using a ¼-in. width at the top and a 4 7/16-in. width at the bottom. This allows for the 4- and 4½ in.

Fig. 5 (below). Trimming strip used to cover joints between horns, and between horns and cover. Piece used between the two horns of each vertical pair should be ½ inch longer.



widths when the sides of the trapezoid are extended to the 5-in. square of the block. The curves marked *A-A* on *Fig. 3* may be used, since they have the correct shape and are of full size. The additional length of the sides for the one wider section may be compensated for by a slight elongation of the curve at the throat end.

With the two opposite sides marked, the block is then cut in a band saw, being tilted slightly as the cut progresses so as to join the lines with the saw cut. This work had best been done in a cabinet maker's shop. The sides are not cut off completely, but are left joined to the block by the sections approximately ½ in. wide at the lower end.

Now slip a piece of cardboard into the two saw slots to make the block solid, and lay out the horn shape on one of the remaining sides, using a ¼-in. throat and a 4-in. mouth. Mark these two sides for identification both above and below the base line so that the correct parts will be retained for later use, after the cutting is completed. Then, still using the band saw, cut along the lines just drawn. The block is now left flat on the saw table, since these two cuts must be parallel. Then cut along the base lines just far enough in to free the outside sections, leaving only the form corresponding to the horn shape, resembling a concave-sided pyramid.

The final operation is to take the two marked sections and mount them back in place, using heavy T-hinges, as shown in *Fig. 2*. The remaining pieces may be discarded.

The method of using is simple. Pre-cut pieces of sheet metal are placed between the clamp blocks and the center form, leaving the metal extending equally on each side, and a large C-clamp is applied at the top of the form and tightened securely. Two additional pieces of sheet metal are then placed on the open sides between the extending lips, and the latter are peened down tightly over the form, thus providing a perfectly shaped section. Solder the four sides carefully, using 50-50 solder. Release the clamp blocks, and remove the completed horn section. The time required to make a single section should be less than five minutes.

Preparing the Sheet Metal

To expedite the assembly of the horn sections, it is advisable to cut all the sheet metal first. Since the final horn assembly will be filled with a deadening material, the only duty of the sheet metal horns is to hold this material in place while it cools, and there is no need to use heavy sheet metal. The recommended material is known as "Coke Tin" in the lightest grade obtainable, being approximately 0.010 in. thick. This material is easy to cut, easy to form, and thin enough to solder rapidly, yet is sufficiently heavy for this application. It comes in sheets 20 x 28 in., and for the entire set of horns, three sheets are required.

Three sizes of metal are necessary for each individual horn section, since two of the sides lap over the others, and because of the trapezoidal shape. *Fig. 3* is a

Fig. 6. Special model of "Baby" speaker unit designed for use with high-frequency horns.

full-size pattern for the three required pieces. For each section, two of the narrowest shape are required, and one each of the other two. Thus, for an eight-cell horn, sixteen pieces are necessary of the smallest size, and eight of each of the others. Once the pieces of tin are cut out, the assembly of the horn sections should proceed rapidly.

Assembly of Sections

After the eight horn sections are completed, they must then be assembled into pairs. It will be noted that each horn mouth is trapezoidal in shape. Holding a pair of horns together with the long sides of the openings adjacent to each other, and the throat ends aligned, solder the corners of the mouth openings. Then solder the throats together, and the pair appears as shown in *Fig. 4*. This completes the preliminary joining of the vertical pair, and all four pairs should be so soldered. For this operation the use of rosin-core solder is recommended, since it will reduce the tendency to loosen the other joints. Flow solder on the sides of the horns at the throat and file flat so that a straight edge along the sides touches only the edges of the mouth and throat openings. Using the same method as for joining the individual sections together, assemble the four pairs into a single unit, keeping the edges of the throats as close together as possible. Use plenty of solder at the throat so as to make a solid structure. The use of a fine-pointed flame from an alcohol torch simplifies this operation, although it is not essential. After the eight sections are assembled together, use a square file to clean up the throat openings and make the edges come together with a sharp dividing line.

The front or mouth edges of the horn are now trimmed by the use of folded strips of tin such as those shown in *Fig. 5*. These are slipped over the adjacent edges of the horns, clamped down in a vise, and soldered in place, again using 50-50 solder. The reason for using hard solder is that when the horns are completed and covered, the space between them is to be filled with melted roofing tar. When sufficiently hot to flow easily, this tar will melt rosin-core solder. If the joints are loosened, the tar will leak out, and additional work is required to remove it.

After the assembling operation is finished, place the narrow side of the horns on a piece of tin and cut the two side covers, leaving a ¼-in. overlap to bend over the top and bottom covers, and bringing the ends flush with the mouth openings. The throat end should be ¼ in. short of reaching the end of the horns. Before soldering on, bend the upper and lower lips over at an angle of 90°, toward the inside of the cover. Folded trim pieces, similar to those used to cover the joints between the mouth openings, are used to finish the edges between the side cover and the end pair of horns.

Now cut similar covers for the top and bottom of the assembly, allowing enough metal to make a smooth concave surface, although not so deep a curve as to touch the horns. Cut a 1-in. diameter hole in the piece that is to be the bottom, centered from side to side, and about 3 in. from the throat end of the horns. This hole is used for filling with the tar. The top and bottom are then slipped under the lips on the side, soldered in place, and trimmed along the front edges with the bent strips. This completes the assembly of the horns into a single unit, and the remaining work consists of preparing the horn throat and soldering it in place, and filling the spaces between the cells with the deadening tar. Make sure that all openings at the corners of the horns are soldered closed. Any surplus can be filed off after the horns are filled.

The High-Frequency Unit

The selection of a suitable high-frequency unit can become involved. Some speaker unit manufacturers advertise models designed to cover the range from 1,000 to 15,000 cps, 3,000 to 16,000, and so on. While good performance to over 10,000 cps is important, it is also important that operation be satisfactory down to at least an octave below the cross-over frequency. Thus a unit which is capable of handling 300 cps will perform better in the octave between 1,000 and 2,000 cps than a unit designed to extend down to 1,000 cps as a minimum. In the writer's opinion, the principal advantage of a two-way system is its ability to handle the upper-middle frequencies adequately, even at the expense of the range above 10,000 cps. Although many f-m programs exceed this frequency, and wide-range a-m receivers are capable of it, few phonograph records have any appreciable signals of higher than 10,000 cps.

The throat coupling on the high-frequency speaker unit has a 5/8-18 thread, and the hole has a diameter of 0.5 in. The area of the opening is therefore $\pi\ (0.5)^2\ /4 = 0.19635$ sq. in. This area must double with every inch of length, and it must join with the throat area, which is 0.5 sq. in. Thus the length between the opening in the unit to the throat of the horn can be determined by calculation to be 1.35 in. and the hole must therefore gradually change its shape from a 0.5-in. circle to a 0.5 x 1.0-in. rectangle, distributed in a circular arc.

This is done in the throat, which is a brass casting also serving as a mechanical

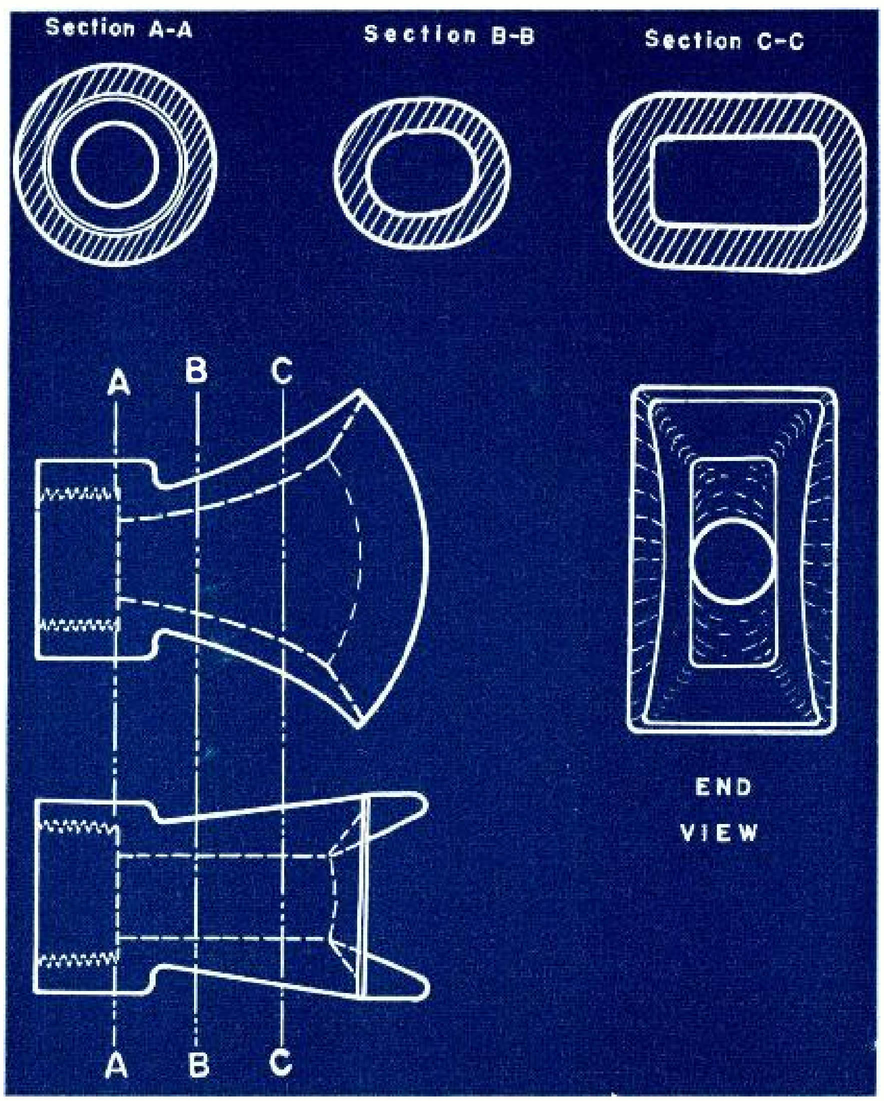


Fig. 7. Plan, elevations, and sections of horn throat coupling, approximately 9/10 full size. Center hole is ½-inch diameter.

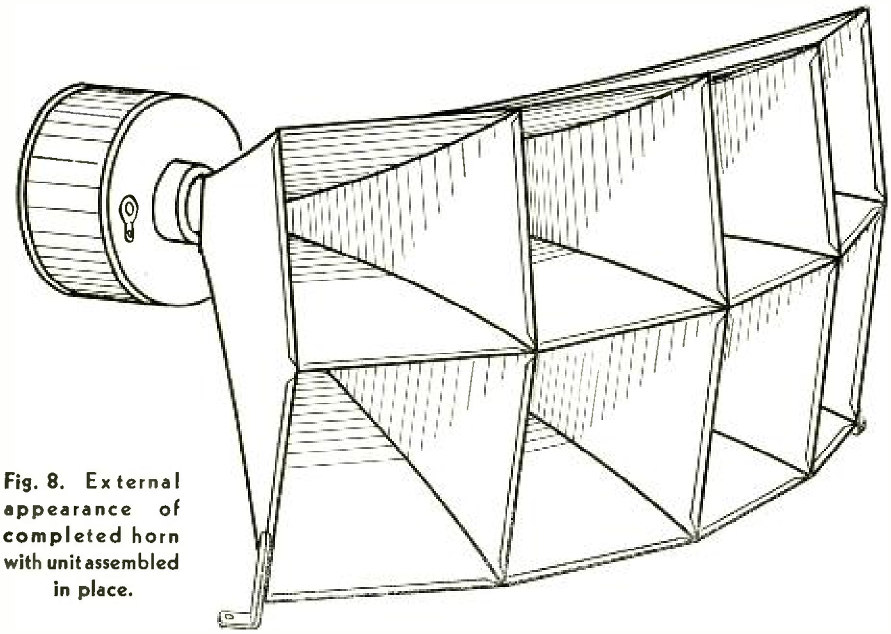


Fig. 8. External appearance of completed horn with unit assembled in place.

coupling between the unit and the horns. *Fig. 7* shows side and plan views, together with sections at various points. From this detail, the pattern can be made readily by anyone familiar with pattern making. However, since it is believed simpler to purchase the throat casting ready made, no instructions are given for making the pattern. The constructor who wishes to make his own is referred to books on the subject. Suffice to say that the pattern consists of a wood replica of the finished coupling, so arranged to part in the center, using short lengths of dowels to keep the two halves in alignment, and having two extensions for the core. The core itself is moulded, also in two parts, and baked before the mould is made, and is placed in the sand mould to provide the opening through the casting. It is moulded in a "core box" which is also made of wood to provide a form of proper shape. After the casting is made, it is machined to make a tight joint at the shoulder, and threaded to fit the speaker unit. The castings made available with the unit have this mechanical work completed.

The throat casting should be thoroughly tinned on the inside, then soldered to the throats of the assembled-and-covered horns. This job requires a lot of heat, and again the small alcohol torch is helpful. Solder should be flowed smoothly to make a neat joint. After cooling, clean up any lumps of solder in the inside of the throat coupling using a round or square rattail file, as the location indicates. Solder two 1 x 1½ in. angle brackets to the sides of the assembly at the front corners for mounting.

Deadening the Horns

At this point, the deadening material is poured into the opening in the bottom cover. Ordinary roofing tar is suitable; once it cools, the entire horn assembly is a solid structure. Melt the tar in an open-topped container, taking care not to get it too hot. It is advisable to melt only a small quantity at a time, adding more chunks as the liquid tar is poured into the horn. A two-pound coffee can makes an ideal container. *Melted tar can cause painful burns, and considerable caution should be exercised in handling it.*

As the tar melts, pour a quantity into the opening and tilt the horns to fill all the corners first, then the throat end. It would suffice to have a coating on the outside of each horn section, but there is no way to make sure of a thorough coating except to fill the entire space. After it is filled and cooled, solder a round tin cover over the opening. Any tar that may have leaked out at the joints can be removed by using gasoline on a rag.

An air brush is helpful in painting; however, a small brush can be used to reach the small ends of the horns if no air brush is available. The object of painting is to forestall rust and to give a suitable outside appearance. When dry and the speaker unit is screwed on tightly, the high-frequency speaker is complete, resembling *Fig. 8*.

The second article of this series will describe the remaining details of the two-way speaker system. Once the high-frequency horn is finished, the unfamiliar part of the work is complete.

IRE-WCEMA Convention

NUMEROUS AUDIO TOPICS were taken up during the combined Institute of Radio Engineers and West Coast Electronic Manufacturers' Association convention in San Francisco, Sept. 24-28. Attendance and interest at both functions were high. Over 750 attendees paid registration fees at the IRE meetings while 3200 registered at the WCEMA exhibit hall.

During the IRE meetings, these speakers made the following points in the course of presenting papers related to the audio field:

John Hessel of the Signal Corps Engineering Labs, Fort Monmouth, said that when war is speeded up, communications problems grow at an even more rapid rate. Considering only the advances made since the war, military communication engineers are presently faced with the necessity of producing communication systems with better performance than existing commercial systems. But at the same time, this equipment must be serviceable under battle conditions and operable by almost totally unskilled soldiers and sailors. This is a severe challenge. He pointed out that a large part of this problem is audio in nature by citing the fact that during the late war, 95 per cent of all communication traffic was by wire line.

Ralph D. Bennett, of the Naval Ordnance Lab., Washington, D. C., in discussing applications of electronics to underwater ordnance, spoke of the hydrophone and bridge circuits used in the study of underwater sound. The frequency spectrum from zero to 100 cps is split into two ranges: 0-1 and 1-100. The high section is recorded directly on disks and the low is pen-recorded through combination in a bridge with a 1000-cps tone.

J. W. McRae, Bell Telephone Labs., New York, described early tests now being made on the AT&T New York to Boston frequency-division multiplex telephone system and said that a month's operational experience has now been had on the first, longest, and (presumably) most difficult link of the chain—New York to Jackie Jones, Conn. During the month he reported a total of 3 minutes during which the signal faded below the 20-db level provided for in the repeater. He also mentioned the fact that in production tests the flat characteristic of the system is held within 1/10 db over 12 mc. This is necessary where any number of repeaters operate in cascade.

John K. Hilliard, from Altec-Lansing Corp., Hollywood, provided the details on several of the high-quality speaker systems produced by his company. In discussing cabinets and enclosures, he stressed the fact that no substitute has

[*Continued on page 41*]

Audiometry of Normal Hearing

C. J. LeBEL*

Applications of special interest to audio engineers are discussed.

FEW ENGINEERS seem to realize the availability of a tool for measuring the hearing characteristics of persons with "normal" ears. The limited number who have used this tool have in too many cases misunderstood the scope and precision of the results which can be so obtained. Finally, they have overlooked an application which seems to supply a remedy for a perennial source of trouble in one part of the audio field.

The instrument which we refer to is, of course, the audiometer. Both frequency and intensity range are wide enough, in any medically approved audiometer, to handle cases of normal as well as of subnormal hearing. A typical audiometer consists basically of an audio oscillator and amplifier of high stability, an accurate attenuator of 110 db range, and a dynamic receiver. By means of compensating networks, mechanical linkages, or other means, the reference level is made to conform to the normal threshold-of-hearing curve of sound-pressure and frequency. By manipulation of the attenuator and oscillator, any sound pressure between 10 db less than normal threshold level and 80 to 100 db above normal may be presented to the listener, at any of the usual hearing test frequencies from 64 or 128 up to 8192 or 11584 cycles. The presentation is controlled by a fading circuit so that the attack and decay of the tone are gradual and free from click.

In use, the operator presents a given tone at successively reduced levels, until the observer has only a fifty per cent chance of telling whether or not the tone is on. This is threshold level. Intensity is adjusted in 5 db steps. A graph of hearing acuity, with respect to frequency, is called an *audiogram*.

Applications

The usual applications are:

1. Medical diagnosis
2. Hearing aid fitting
3. Measuring hearing of workers in noisy shops at start and finish of employment.
4. Measuring normal hearing applicants for employment in positions where judgment of sound quality is essential.

To this the writer would add a new use:

5. Measuring the hearing of prospective purchasers of custom-built high-fidelity home reproducing equipment.

In the following discussion the audiogram examples are discussed on the basis that both ears are alike. Evaluation of more complex conditions would require more space than is presently available, but in general we may consider the better ear (if they are widely different) or an average (if closely alike) as characteristic.

Usual Applications

Medical diagnosis by audiometric methods is too well developed and too specialized to require comment here. It is safe to say that every modern otologist either owns or has the use of an audiometer.

Hearing aid fitting by audiometer has been found a highly desirable practice by those manufacturers who were willing to put forth the effort to develop a fitting procedure with a sound, quantitative basis, and were willing to spend the money required to equip all their dealers with the necessary instruments and train them in the correct technique. One alternative is a speech-hearing test. This has two defects:

a. It has insufficient resolving power; i.e., at best it can indicate only very gross differences.

b. It is not sufficiently dependable. While the results of a speech-hearing test correlate somewhat with more refined methods of measurement, the correlation is not sufficiently good from the customer's point of view. He is buying not a probability of fit, but a definitely acceptable performance. It has been estimated that for every 5% of the customers who are not satisfactorily fitted, a minimum annual advertising expenditure of about $100,000 will be required.

Another alternative is to provide a hearing aid with substantially no adjustment, and make no attempt at all at acoustical fitting. This has been advocated in a recent publication. As in so many other audio instances, the data are impeccable but the interpretation is faulty. Investigation showed that the results were based on tests of 18 persons. This immediately raised a question of accuracy of sampling, for it may be questioned whether it is possible to get a fair sample of the hard-of-hearing in only 18 persons. Inspection of such of their audiograms as were available showed that by the standards used in hearing-aid research the hearing loss was very close to uniform with respect to frequency in the region which is critical. In short, the tests showed that for the group which has substantially uniform loss, an instrument having substantially uniform frequency response is most satisfactory!

We might reinforce this with an amusing example. Through inadvertence, a hearing-aid company for one year marketed an instrument with negligible provision for frequency response variation. The management and dealers, not being well informed, went through a vigorous "fitting schedule" which impressed the customer. The next model had genuine provision for response variation over a wide range, and fitting charts to use this to advantage. Apart from improved fitting provision, the fatigue factors were unchanged. It was quickly noticed that a larger percentage of the prospective sales were being closed, so that more sales resulted from the same number of prospects. In other words, a change from an essentially uniform response instrument to one with genuine fitting provision had proven worth while.

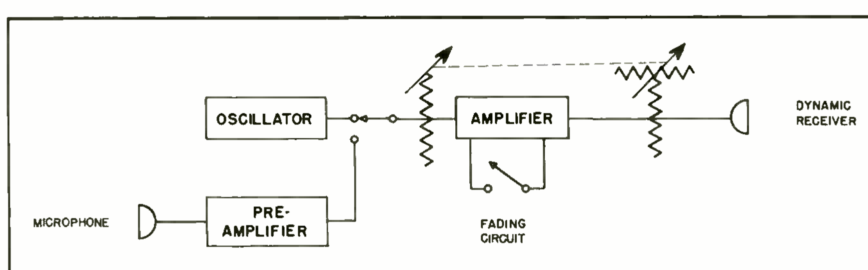


Fig. 1. A typical audiometer circuit.

Perhaps the trouble with the 18-person study lay also in the fact that a species of articulation test was employed. It has been found that a person may do very well on such a test, yet refuse to wear the hearing aid so tested because it is too fatiguing. Unfortunately, some refuse to recognize the fatigue factors because they are subjective and hence qualitative.

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Nevertheless they are a very real influence in the user's mind.

Aural trauma produced by work in an excessively noisy shop is often indistinguishable from the effects of age and of previous employment. Therefore, a *before* as well as an *after* audiogram is necessary. If this practice had been generally followed during the war, shipyards (for example) would have been spared many expensive lawsuits.

Aural Test of Key Personnel

In broadcast studio operating and in high quality radio receiver tests it is essential that the personnel have "normal" hearing. Therefore a number of organizations have made it a practice to secure audiograms of all applicants for such positions. Discussion of the handling of such results with a number of executives seems to indicate a lack of knowledge of the basic characteristics and limitations of the audiometric process, and it is felt that a few facts might be of interest to engineers in the field.

A deviation of 10 db from the 0 db "normal" line is still considered normal by the otologist. Literally applied, this would permit a range of variation of 20 db. Since we are interested more in frequency response than in absolute acuity, the audiogram of *Fig. 3* would be unacceptable to the broadcaster, even though technically "normal." On the other hand, the audiogram of *Fig. 4* represents a person with a normal frequency response, but subnormal general acuity. He probably would do perfectly well in a broadcast job, for any monitor amplifier is likely to have 10 or 15 db reserve power. Actually, such a person would probably not realize that his hearing was technically below normal, for a 15 db loss is not a social handicap. In listening to radio receivers, he would simply have to keep his ear closer to the loudspeaker, but again would probably be entirely satisfactory.

We come now to a more difficult question, posed by the audiogram of *Fig. 5*. Superficially, this appears to show a reduced extreme high-frequency response. Actually, what it shows is somewhat in doubt. The following factors may be at work, separately or together, to increase the observational error:

1. According to medical standards, a 5 db error in calibration does not render an audiometer unacceptable.

2. While we have a U. S. Public Health Survey—National Bureau of Standards standard of intensity for 8192 cycles—the 11584 cycle value is based only on a manufacturer's limited tests. There is no national standard at that frequency (or at the other half octave frequencies, either).

3. Standing waves in the ear canal represent a serious source of possible error at 8192 cycles, and are even worse at 11584 cycles.

4. The transfer of calibration from the BuStan's standard receivers to a manufacturer's own type of receiver has to be done by loudness balance procedure. Since this is a subjective method, errors due to individual differences in listener ear cavity and differences in receiver acoustic output impedance can exist unless a large number of listeners is used. Unfortunately, loudness balance work demands considerable skill, and there are probably not over a dozen or so who can handle that type of work well. It will be found that a manufacturer's own transfer will usually check the Bureau at all frequencies but one or two, and at the latter a divergence of 2 or 3 db may exist. We are not justified in accepting the Bureau data as final, for the manufacturer may have as many and as skilled technicians for the transfer. At best, we can split the difference and say that at some one of the higher frequencies an additional error of 1 or 2 db can occur. The National Bureau of Standards has been fully cognizant of this possible source of error for a number of years, and would like to retransfer using the probe tube method. Then a hundred unskilled subjects could be used, for they only lend the use of their ear cavities, and make no observations themselves. Unfortunately, world events have forced the precedence of other projects, and have robbed the section of several good men; hence this project has necessarily had to wait.

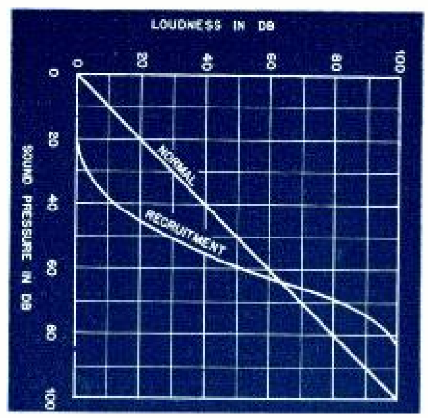


Fig. 10. Recruitment.

5. We would therefore repeat the test, having the subject shift the receiver for best results at 8192 cycles and above, and might still disregard small variations.

In *Fig. 6* we find an extremely common defect, a notch at 2896 cycles or 4096 cycles. Prolonged exposure to very loud sounds will create this—it is often called "aviator's notch." It is invariably quite narrow—not over an octave at the base. The notch may also be an artefact, caused by the audiometer receiver. One manufacturer for years used a non-moisture-proof receiver. In damp summer weather the receiver response might drop 10 to 20 db, at two or three frequencies, with almost complete recovery when the weather became less humid. Since several thousand such units were sold, the possibility of encountering an off-calibration receiver is great in summer.

We have no evidence that such a narrow notch will have any significant effect on judgment of quality, if the notch does not represent too great a loss. A 20 db notch is 20 db over so narrow a frequency range that we are disposed to consider it negligible.

In short, read an audiogram with judgment, and evaluate the variation with some consideration for possibility of artefacts. Be sure that the audiometer used was calibrated correctly, and that it has not changed since calibration.

High Fidelity Addicts

Virtually everyone has hoped to use a home reproducing system with better quality than that generally available. Some people have the means to gratify that desire. They listen to the best commercially available material, decide they want something better, and approach the custom builder. Now that poses a nice problem. If the custom builder is to remain in business, he must have a record of highly satisfied customers. His customers have a right to dislike the quality of the commercial set, for it has too often been made to sell rather than to listen to. Nevertheless, some customers can never be satisfied, for their hearing is such that normal sound is forever beyond their grasp, no matter what the frequency response of the system is. A number of audiograms will best illustrate this:

Graph *A* of *Fig. 7* shows the ideal customer, with average hearing. Graph *B* shows another good case, with a uniform 20 db loss over the entire spectrum. In the latter case rather more amplifier reserve power might be desirable. Incidentally, a 20 db loss is so small that it often goes unnoticed socially.

A highly undesirable hearing characteristic, from the custom builder's point of view, is shown in *Fig. 8*. With accentuated bass, and no usable hearing above 3000 cycles, we have a person who can never appreciate improved bandwidth. Unless he realized that the higher overtones were forever out of reach, and would value especially clean low frequency reproduction, purchase of a custom-built unit might lead to serious dissatisfaction.

An example of a hearing defect which could be compensated for by suitable networks in the reproducing amplifier is shown in *Fig. 9*. On the other hand, a reproducing system which was so compensated would sound very unpleasant to the listener with normal hearing; it would be an exceedingly poor advertisement for the manufacturer.

A hearing defect which might lead to very curious reactions would be that of recruitment. In engineering terms this

[*Continued on page 37*]

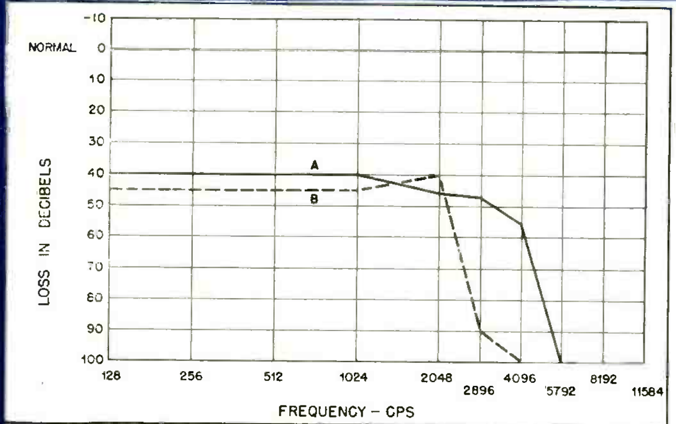


Fig. 2. Hard of hearing conditions that can best be distinguished by audiometric means.

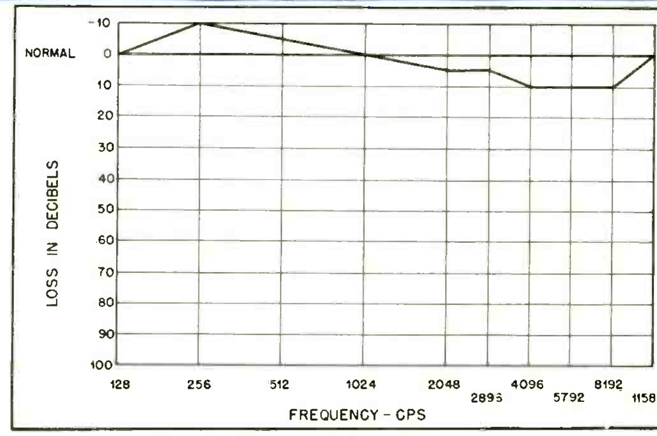


Fig. 3. Unacceptable audiogram.

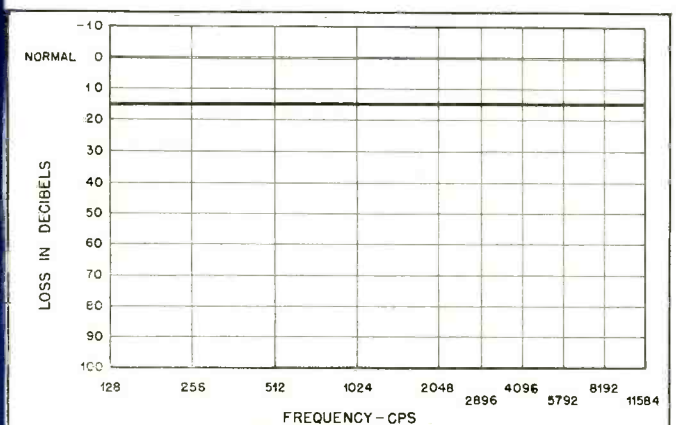


Fig. 4. Acceptable audiogram.

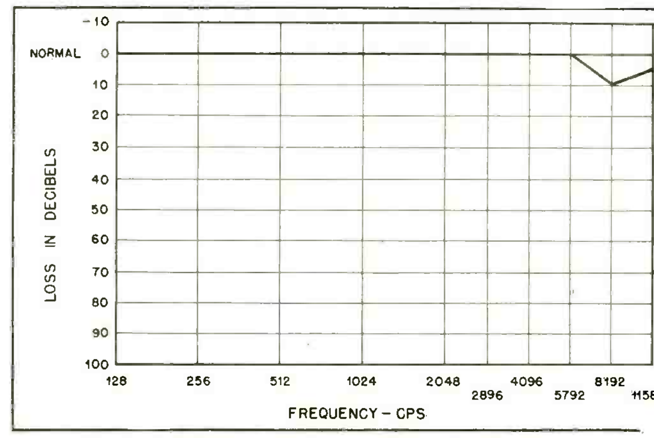


Fig. 5. High-frequency loss or observational error.

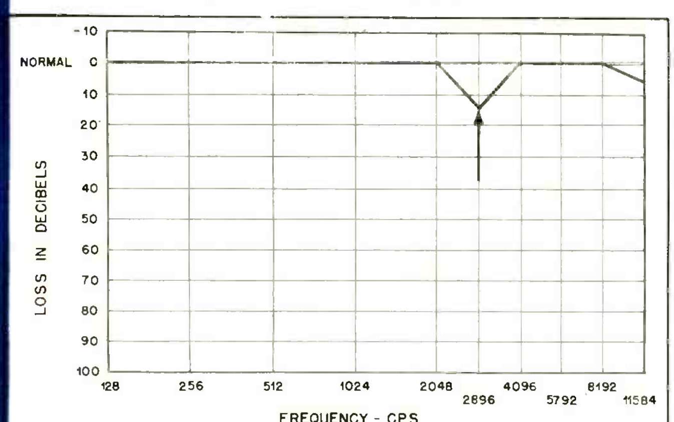


Fig. 6. Aviator's notch.

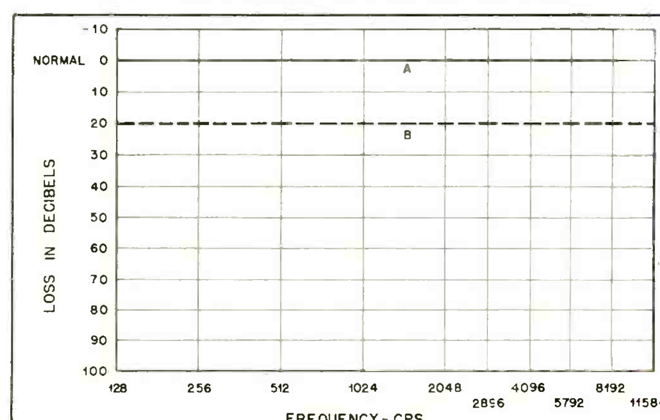


Fig. 7. Audiograms of ideal customers.

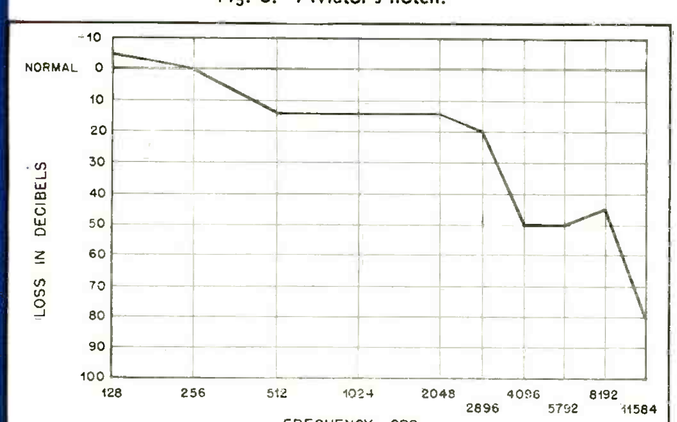


Fig. 8. Audiogram of a low-fidelity customer.

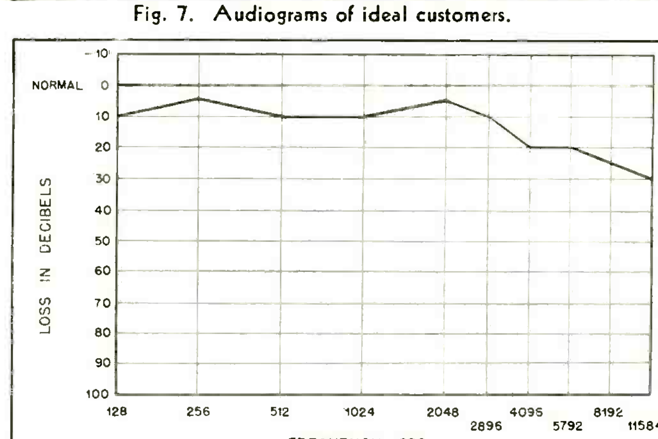
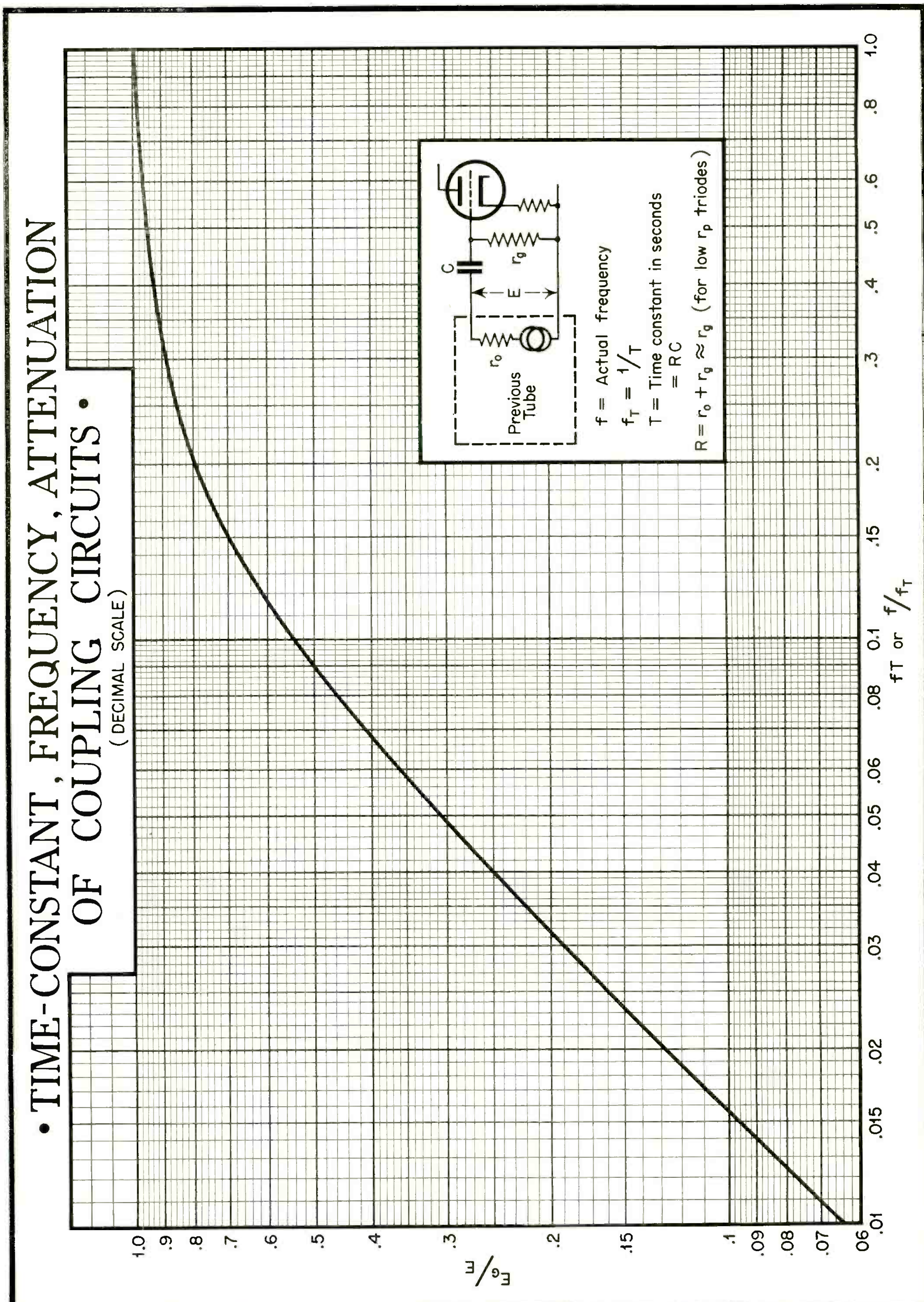
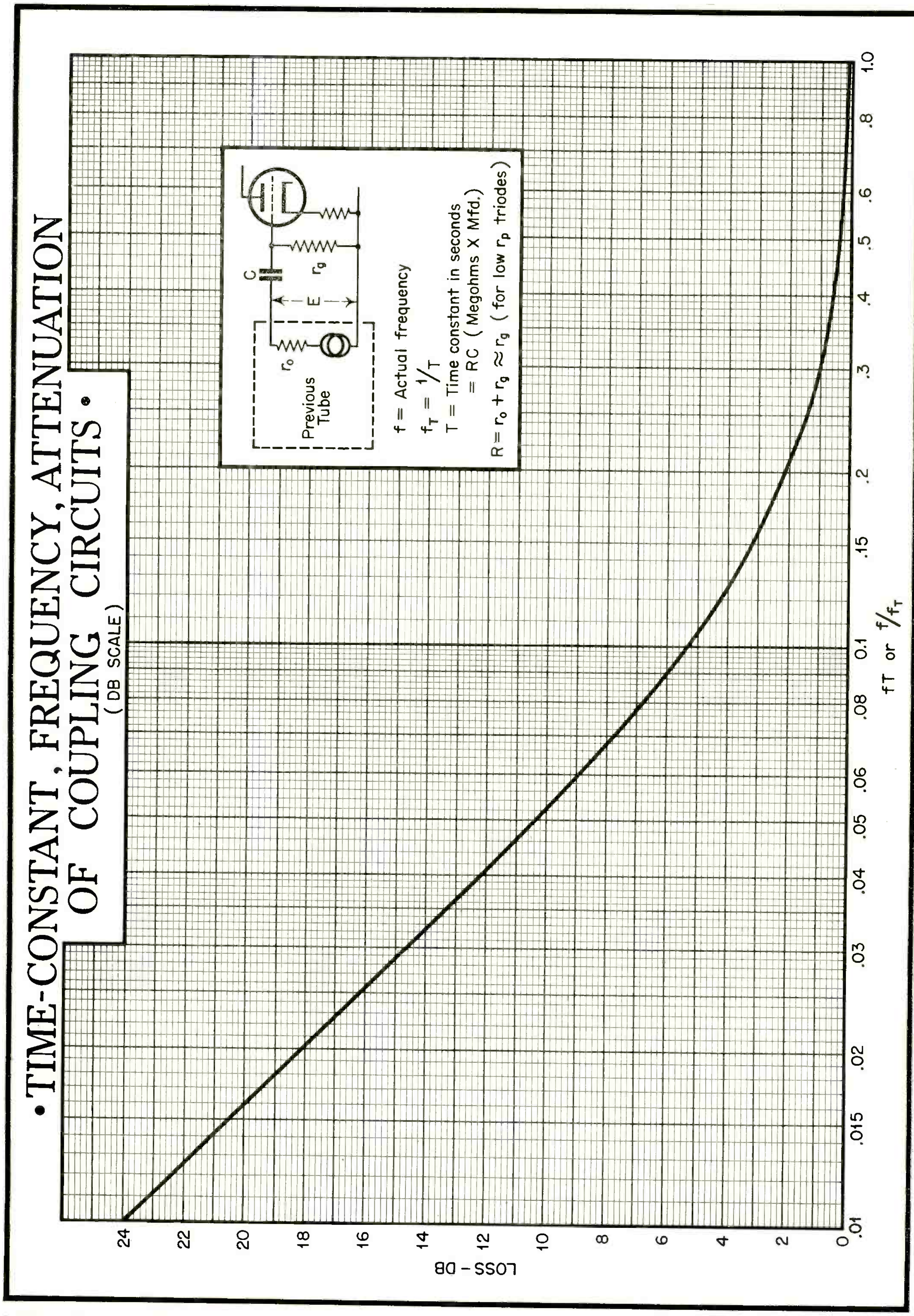


Fig. 9. Audiogram of a customer who can be satisfied only by alienating his normal hearing friends.



• TIME-CONSTANT, FREQUENCY, ATTENUATION OF COUPLING CIRCUITS •
(DECIMAL SCALE)
Previous Tube
r_o
C
E
r_g
f = Actual frequency
$f_T = 1/T$
T = Time constant in seconds = RC
$R = r_o + r_g \approx r_g$ (for low r_p triodes)
E_g/E
fT or f/f_T



AUDIO ENGINEERING NOVEMBER, 1947 31

TABLE II (all dimensions in mils)

Grooves per Inch	Groove to Land Ratio 70 to 30: Value	Depth T for R equal to 2.3	2.0	1.5	0.5	0.25	Groove to Land Ratio 60 to 40: Value	Depth T for R equal to 2.3	2.0	1.5	0.5	.25
87° Angle of Cutting Stylus	.5269W						.5269W					
96	3.84	2.80	2.94	3.16	3.61	3.73	3.29	2.25	2.39	2.61	3.06	3.18
100	3.69	2.65	2.79	3.01	3.46	3.58	3.16	2.12	2.26	2.48	2.93	3.05
110	3.35	2.31	2.45	2.67	3.12	3.24	2.87	1.83	1.97	2.19	2.64	2.76
112	3.29	2.25	2.39	2.61	3.06	3.18	2.82	1.78	1.92	2.14	2.59	2.71
120	3.07	2.03	2.17	2.39	2.84	2.96	2.63	1.59	1.73	1.95	2.40	2.52
125	2.95	1.91	2.05	2.27	2.72	2.84	2.53	1.49	1.63	1.85	2.30	2.42
136	2.71	1.67	1.81	2.03	2.48	2.60	2.32	1.28	1.42	1.64	2.09	2.21
150	2.46	1.42	1.56	1.78	2.23	2.35	2.11	1.07	1.21	1.43	1.88	2.00
160	2.31	1.27	1.41	1.63	2.08	2.20	1.98	0.94	1.08	1.30	1.75	1.87
416	0.89			0.21	0.66	0.78	0.76			0.08	0.53	0.65
515	0.72			0.04	0.49	0.61	0.61				0.38	0.50
80° Angle of Cutting Stylus	.5959W						.5959W					
96	4.34	3.06	3.23	3.51	4.06	4.20	3.72	2.44	2.61	2.89	3.44	3.58
100	4.17	2.89	3.06	3.34	3.89	4.03	3.58	2.30	2.47	2.75	3.30	3.44
110	3.79	2.51	2.68	2.96	3.51	3.65	3.25	1.97	2.14	2.42	2.97	3.11
112	3.72	2.44	2.61	2.89	3.44	3.58	3.19	1.91	2.08	2.36	2.91	3.05
120	3.47	2.19	2.36	2.64	3.19	3.33	2.98	1.70	1.87	2.15	2.70	2.84
125	3.34	2.06	2.23	2.51	3.06	3.20	2.86	1.58	1.75	2.03	2.58	2.72
136	3.07	1.79	1.96	2.24	2.79	2.93	2.63	1.35	1.52	1.80	2.35	2.49
150	2.78	1.50	1.67	1.95	2.50	2.64	2.38	1.10	1.27	1.55	2.10	2.24
160	2.61	1.33	1.50	1.78	2.33	2.47	2.23	0.95	1.12	1.40	1.95	2.09
416	1.00			0.17	0.72	0.86	0.86			0.03	0.58	0.72
515	0.81				0.53	0.67	0.69				0.41	0.55
70° Angle of Cutting Stylus	.7141W						.7141W					
96	5.21	3.50	3.72	4.09	4.84	5.02	4.46	2.75	2.97	3.34	4.09	4.27
100	5.00	3.29	3.51	3.88	4.63	4.81	4.28	2.57	2.79	3.16	3.91	4.09
110	4.54	2.83	3.05	3.42	4.17	4.35	3.89	2.18	2.40	2.77	3.52	3.70
112	4.46	2.75	2.97	3.34	4.09	4.27	3.83	2.12	2.34	2.71	3.46	3.64
120	4.16	2.45	2.67	3.04	3.79	3.97	3.57	1.86	2.08	2.45	3.20	3.38
125	4.00	2.29	2.51	2.88	3.63	3.81	3.43	1.72	1.94	2.31	3.06	3.24
136	3.68	1.97	2.19	2.56	3.31	3.49	3.15	1.44	1.66	2.03	2.78	2.96
150	3.33	1.62	1.84	2.21	2.96	3.14	2.86	1.15	1.37	1.74	2.49	2.67
160	3.13	1.42	1.64	2.01	2.76	2.94	2.68	0.97	1.19	1.56	2.31	2.49
416	1.20			0.08	0.83	1.01	1.03				0.66	0.84
515	0.97				0.60	0.78	0.85				0.48	0.66

but they should not prevail where the sound-track grooves are embossed and the same playback stylus is used, as in home-recording, for then *Fig. 1* shows the conditions.

However, the unit pressures on contact areas are of interest where embossed sound tracks are concerned. Depending upon the material embossed a certain amount is displaced by plastic flow, as is indicated by the ridges which appear at the sides of the grooves; but it appears that compression of the material of the record disc may account for the major volume of the groove.

Even with cut grooves such ridges are formed, to a possibly lesser degree, but increasingly with the reduction of the included angle of the cutting stylus. That is to be expected because the more nearly horizontal any thrust is against the groove walls of a horizontal disc, the more effective is the pressure vector for producing plastic flow to the upper free surface of the disc.

When ethyl cellulose record discs were marketed just before World War II, they bore pronounced ridges on only one side of the sound track. That made an additional support to guide the playback stylus inwardly. Such records were cut by a conical stylus whose inside face was truncated by a plane approximately parallel to the axis of the cone. The Dow Chemical Company owns a patent covering such a stylus and method of cutting sound tracks.

The present method of cutting sound tracks still leaves ridges above the sides of grooves and an effort is made to avoid them, although it is questionable whether they are objectionable, *provided* they do not move inward by delayed cold flow, and rub noisily against the side of the stylus, or cause processing trouble. Rising above the surface, they may be more susceptible to handling scratches or distortion, but they provide a greater guiding surface for the playback stylus. There may be a belief that they encourage starting a stylus in the lands between grooves, but the writer was unsuccessful in attempting to do that with embossed records which had higher ridges than the cut records.

Further studies concerning the plastic flow of cut and embossed records might encourage some important changes in that art. It is certain that playback styli rarely fit the sound grooves of lateral-cut and pressed records. This results in initial point contacts until the sound groove is sufficiently indented to provide a supporting area determined by the physical properties of the disc material and the pressure upon the playback stylus normal to the surface of the disc. Thereby the fidelity of the original cut sound track suffers injury if proportional elasticity is exceeded.

With embossed sound tracks, using the same stylus to record and reproduce, this objection does not hold true, for; the stylus has a greater bearing for the same pressure displacement (fits the sound track); and a lower pressure is placed upon the stylus for playback than was used in recording.

A certain resiliency or lateral flexure in the stylus seems common in phonograph pickups as a necessary evil since high resonant frequencies may thereby be introduced and are often blamed as surface noises from the record. N. W. McLachlan states in his book*, "One is apt to conclude that it (needle scratch) is entirely due to the needle passing over tiny irregularities in the surface of the material of the record. This is not so. The greater part of the scratch is due to resonance of the steel needle, which may occur between 3,000 and 7,000 cycles according to the type of pickup. Although the resonance of the needle itself is well above 5,000 cycles, the frequency is reduced by the armature mass. During operation the needle is subjected to a continuous shower of blows, and although highly damped by rubber buffers, it is in a state of perpetual oscillation at its resonance frequency. Even on non-playing passages the microsopic

*"The New Acoustics," 1936, Oxford Univ. Press. p. 88.

particles of which the record is composed keep up an incessant bombardment."

Although there is lateral flexibility for spotting the stylus in the groove, where the sound track guides the pickup arm as in conventional phonographs, yet the mass of the arm offers sufficient inertia to permit the resonant flexure of the stylus.

But where the pickup arm is not guided by the spiral sound groove of the record, as in the Wagner-Nichols embossing recorder described in the May, 1947 issue of AUDIO ENGINEERING, some other method of flexibility had to be devised. Although not mentioned in that article, it may prove to be an important development which will reduce the resonant frequencies of the stylus.

The method is to suspend the crystal pickup case as a simple pendulum which swings laterally in a vertical plane radially to the record disc. The period (P) of oscillation in seconds of such a pendulum equals two pi times the square root of its suspended length (L) in feet divided by the acceleration of gravity (32.16 feet per sec. per sec.). This simplifies to the quantity 1.108 times the square root of the pendulum length in feet. Since the pendulum length in this recorder is estimated at 3/8 inch its frequency is 5.1 cycles per second. To have a natural frequency as high as 30 cycles per second its pendulum length would have to be as short as one one-hundredth of an inch.

The deformation laterally in the sound grooves of lateral-cut plastic discs is also encouraged by the reliance of the sound groove to guide the pickup arm; and it is further promoted by the eccentric swing common to such arms. The imperfect fit of playback styli to cut sound grooves intensifies such deformations.

The use of abrasive fillers in commercial phonograph discs in order to grind down a steel playback stylus to fit a sound groove is no solution to the problem in a modulated sound groove for it grinds the stylus to fit the convexities of the walls of the sound groove (and abrades them to a loss of fidelity) and thus provokes a poorer contact with the rest of the walls.

Undulation of Sound Grooves

With the usual lateral-cut record the spherical tip of the playback is of a larger diameter than the bottom of the groove so never rests on the bottom. The bearing of the stylus in the sound groove thus falls on the conical walls of the stylus with a corresponding wedging action if it fits the groove.

To facilitate and encourage sound groove pressure studies the writer offers some of the trigonometry involved in computation of stylus bearing areas.

The area a of the spherical segment surface of the tip of the stylus equals $2\pi R\,[R-(R^2-r^2)^{1/2}]$ where R is the radius of the sphere and r is the radius of the segment section at the point of

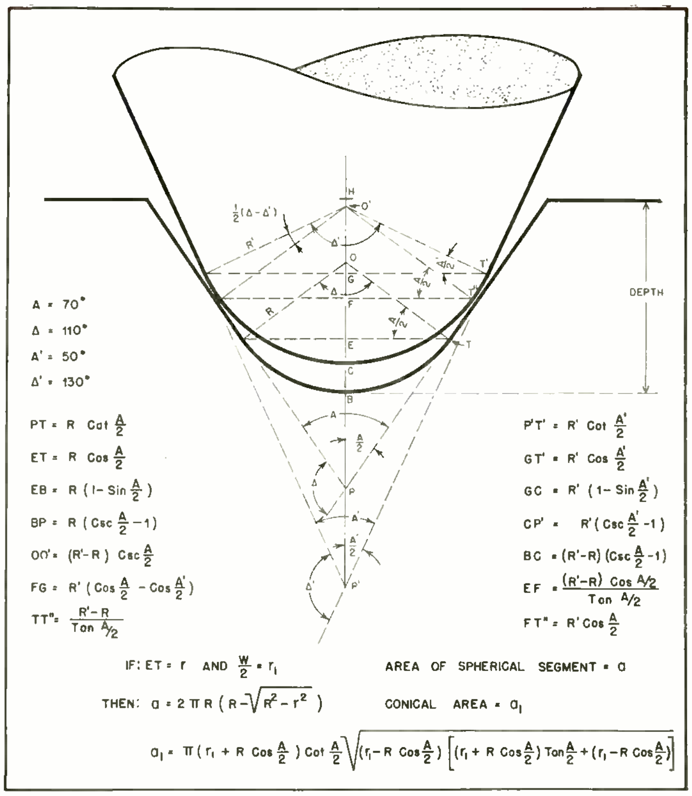


Fig. 2. Play-back stylus in sound track.

tangency with the stylus cone, namely, OT in *Fig. 2*. When penetrating a plane surface r would represent the radius of the indentation, as in a Brinell hardness test. Therefore, $r = R\cos\frac{A}{2}$ and $a = 2\pi R\,[R-(R^2-R^2\cos^2\frac{A}{2})^{1/2}] = 2\pi R^2\,[1-\sin\frac{A}{2}] = 2\pi R^2$ covers $\frac{A}{2}$.

Angle A	M	N
90°	$3.1416\,r_1 + 2.2214\,R$	$(2.0000\,r_1^2 - 1.4142\,r_1 R)^{1/2}$
87°	$3.3106\,r_1 + 2.4014\,R$	$(1.9490\,r_1^2 - 1.4508\,r_1 R + 0.0268\,R^2)^{1/2}$
80°	$3.7440\,r_1 + 2.8691\,R$	$(1.8391\,r_1^2 - 1.5305\,r_1 R + 0.0932\,R^2)^{1/2}$
70°	$4.4867\,r_1 + 3.6753\,R$	$(1.7002\,r_1^2 - 1.6383\,r_1 R + 0.2012\,R^2)^{1/2}$
60°	$5.4414\,r_1 + 4.7124\,R$	$(1.5774\,r_1^2 - 1.7321\,r_1 R + 0.3170\,R^2)^{1/2}$
50°	$6.7372\,r_1 + 6.1059\,R$	$(1.4663\,r_1^2 - 1.8126\,r_1 R + 0.4384\,R^2)^{1/2}$

For various values of the including angle A the area a equals

For 90° $a = 1.8403\,R^2$
87° $a = 1.9581\,R^2$
80° $a = 2.2445\,R^2$
70° $a = 2.6793\,R^2$
60° $a = 3.1416\,R^2$
50° $a = 3.6278\,R^2$

For the area (a_1) of the curved surface of the truncated cone which corresponds to its penetration in an embossed record the formula is far more complicated. Only one-half of these areas of spherical segment and truncated cone exert pressure in the process of embossing a sound groove, after the initial penetration of the stylus and when it is inscribing a sound track.

Where r_1 is the radius of the transverse section of the cone at the surface of the record or equal to one-half the width of the unmodulated sound groove, and r is the radius of the cone where the spherical segment joins it, $a_1 = \pi\,(r_1 + R\cos\frac{A}{2})$ times $\cot\frac{A}{2}\,\{(r_1 - R\cos\frac{A}{2})\,[(r_1 + R\cos\frac{A}{2})\tan\frac{A}{2} + (r_1 - R\cos\frac{A}{2})]\}^{1/2}$

For tabulation with various values of the included angle A this may be expressed as $a_1 = M \times N$ where $M = \pi(r_1 + R\cos\frac{A}{2})\cot\frac{A}{2}$ and $N = \{(r_1 - R\cos\frac{A}{2})\,[(r_1 + R\cos\frac{A}{2})\tan\frac{A}{2} + (r_1 - R\cos\frac{A}{2})]\}^{1/2}$

But there are no such areas of contact for the playback styli of both lateral and vertical-cut sound tracks where they make only line or point contact with the groove until they have penetrated it or some deformation has appeared at the contact. In the case of the stylus having the same included angle and larger tip radius than those of the groove one-half

[*Continued on page 37*]

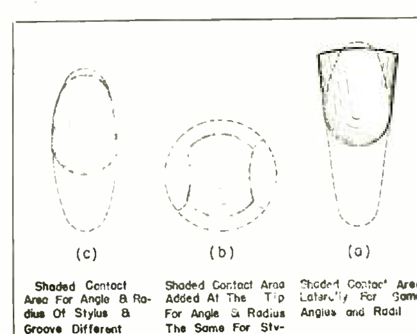


Fig. 3. Areas of contact differing styli tips.

RECORD REVUE

In this department the author, who is a very well-known record critic, will review monthly record releases of outstanding technical, as well as musical, quality.

EDWARD TATNALL CANBY

AS A MUSICIAN bungling into an engineer's private magazine, I occupy a somewhat anomalous and very stimulating position on this page. I speak in terms that belong to the engineer, not the musician, and yet I am no trained engineer by a long shot. Speaking of shots, I take a good many good ones in the general direction of what both musicians and engineers are after, but I certainly don't guarantee all hits, especially when I stick my neck way out and make positive statements as to frequency range, distortion, microphone placing and the like! Yet it seems worth making them just the same.

These statements are estimates, analyses, forecasts—guesses if you will, made as shrewdly and as cannily as possible, but still not laboratory-guaranteed. Certain objective facts are of course ascertainable (and one I hope soon to reduce to objectivity is the exact range at which a record actually does cut off or cease to be effective). But even such a simple thing as surface noise, for instance, is by no means a matter of plain mechanical grading. It's clear enough to any record collector, and to FM broadcasters who try to play records minus filtering, that record scratch is only partly measurable as so much noise; just as important is the type of noise, its constancy, its rhythm.

Even more important is the type of music with which it is associated—in fact this last factor is so vital that, when you come down to it, it takes a musician to judge the degree of "unpleasantness" in the surface scratch of any given record, and a musician (with engineering knowledge of course) should man the filter controls. Or put it a better way: it takes close listening, not to the scratch, but to the *music* (and an engineer can do this perfectly well!). A "hiss meter" or other objective measurement, however accurate, will never prove much, even if it allows for variation in reproducing equipment and for a dozen or so other technical factors. It is what the ear hears—what the mind hears, that matters.

In a recent article I have defined "high fidelity"—another un-pin-downable factor of a lot greater complexity—as "a relatively high degree of naturalness or faithfulness to the *imagined* original sound." That may seem a strange definition, but remember that in the first place all "fidelity" is basically an illusion, never a literal (binaural) reproduction; and secondly, the realism of that illusion may be very convincing, yet the listener actually cannot compare it to the original of which it's supposed to be a reproduction, because he never has heard the original and never will. Hence "fidelity" is always a faithfulness to an *imagined* original. A very philosophical point, admittedly. But, if true, then a large part of the utter confusion as to what is "pleasurable" in reproduction (as one listening test on fidelity has put it), can be chalked up to it, and to the unfortunate fuzziness of that original in a lot of people's minds! It's a nice idea in any case and I hope it will be the occasion for a few hot engineering arguments.

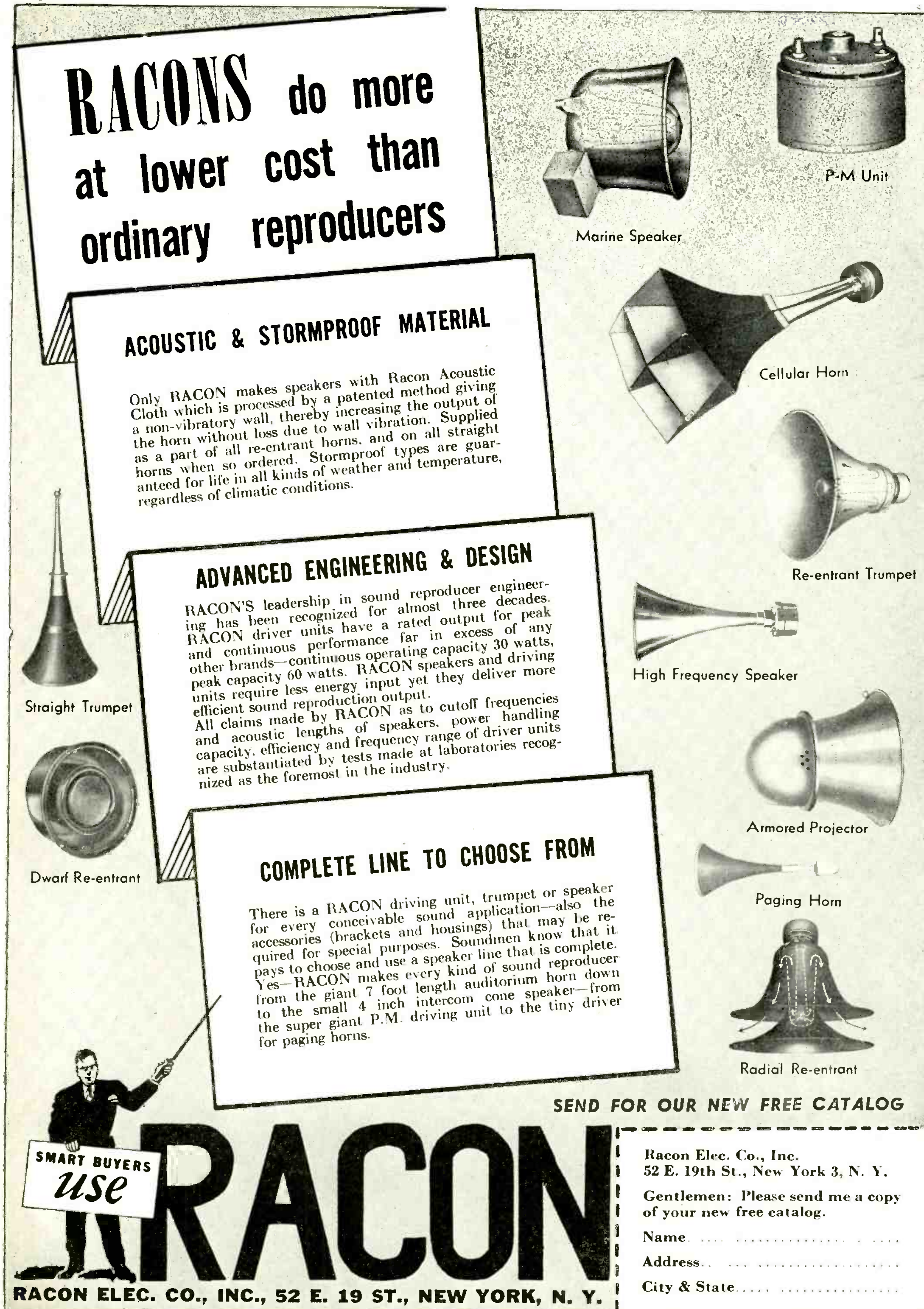
Surface Noise

I take surface noise and fidelity as samples, because the same human element applies to the judgment of more engineering factors in recording than a circumspect and methodical engineer might like to admit. I have, for instance, gone off the deep end more than once in the matter of judging the comparative tonal range of various recordings. I have flatly labelled one recording wide range, another definitely restricted—only to have the engineer in charge later assure me that the two were made under identical circumstances. Yet I'm not too abashed. Though perhaps I ascribed the wrong reason, there was and is a clear musical difference—an apparent difference—between those records. Without question one gives the *impression* of wider range than the other. One sounds filtered, the other sounds bright and clear. Why? (I'm sure that if the engineers who made the records knew there would have been no difference. Both would have been "bright and clear"!)

One may of course argue most fruitfully as to what the possible causes are for such paradoxes. One lives and learns. It seems there is a new factor to consider each day, and a large number of them are musical. There is the weather, too, and the mood of the performers—and don't think these don't matter. I am about persuaded, to bring in another line, that the accentuation technique (Maxfield) now used in much recording and broadcast gives an *impression* of decidely greater tonal range than, say the "overall" single mike type of placement, even though there is actually no greater range. Partly, I suspect, because the accentuation tends to give a sharp, close-to edge instrumental color simultaneously with a feeling of distance, an effect that of course is wholly unnatural, but very realistic.

Yet equally significant when it comes to apparent range in a recording, is the composer's use of his instruments. A transparent, thin orchestra, with many solo passages especially for brasses, woodwinds and solo violin, gives a wider range effect than a solid, full, heavy orchestration, recorded identically. The perfect example for both these points is the Strauss "Bourgeois Gentilhomme" album

[*Continued on page 45*]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NEW PRODUCTS

DUO-CONE LOUDSPEAKER

The new high-fidelity Duo-Cone Monitor Loudspeaker, RCA Type LC-1A, recently developed by Dr. H. F. Olson of the RCA Laboratories, is capable of reproduction over the range from 30 cycles to 15,000 cycles.

The LC-1A loudspeaker consists of a 15-inch permanent magnet type mechanism. The cones are of the direct radiator type and

consist of high and low-frequency units mounted together coaxially and sharing the same axis and cone periphery angle.

The high-frequency unit is a two-inch cone with an aluminum wire wound voice coil, which effects a very desirable high flux density and a low mass, necessary to obtain an extended high-frequency range.

The low-frequency cone employs a 15-inch diaphragm with a high-mass voice coil to produce a desirable low-distortion factor, a wide angle (120° at 15,000 cycles), extended low-frequency range, and a low fundamental resonant frequency, which is approximately 35 cycles. Above this frequency the reluctance due to the mechanical compliance of the cone suspension system does not appreciably retard the movement of the cone, and therefore minimizes distortion.

The LC-1A cabinet was especially designed with an open bass reflex port to provide the most desirable bass accentuation for general acceptance. However, since some individuals prefer a flat low-frequency response, the port may be closed easily by a manual control.

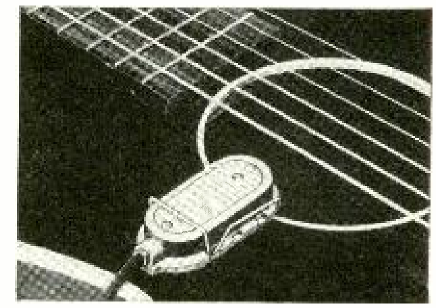
The cabinet of the RCA Duo-Cone speaker is attractively styled in two finishes: two-tone umber gray with satin chrome trim (MI-11411) to blend with all associated RCA equipment, and a bleached walnut veneer (MI-11411-A) for use in finely furnished listening booths and executive offices.

Both cabinets are provided with a brushed chrome control panel at the top of the right side. A channel selector switch may be employed to connect the speaker to any one of 10 high-level audio busses. When this method of bridging is desired, the panel will also accommodate a 15-ohm power attenuator.

CONTACT MICROPHONE

● A new contact pick-up microphone, E-V Model 805, that can be used on all vibrating musical instruments, is announced by Electro-Voice, Inc., Buchanan, Michigan. Designed to enhance the reproduction of instrumental music . . . suitable for guitar, banjo, mandolin, violin, viola, cello, harp, drum and piano. Increases natural sound volume and enriches tonal effects.

Frequency response is 40 to 8000 c.p.s. Can be used with any amplifier having a high



impedance input. Output level: .1 to 1 volt, depending on type of instrument. The generating element is an inertia-type crystal, sealed against moisture and acoustic feedback.

For further information, write to Electro-Voice, Inc., Buchanan, Michigan. Export Division: 13 East 40th St., New York 16, N. Y., U.S.A. Cables: Arlab.

VELOCITY MIKE

● The new Amperite velocity microphone represents a concrete advance in this type of microphone. The new design incorporates all the advantages of a "ribbon" microphone and eliminates the few previous causes of dissatisfaction.

For example, it will give top-quality reproduction on *either close talking* or *distant* pickup. Performers may shout directly into the new Amperite velocity without "blast-

[*Continued on page 47*]

PROFESSIONAL DIRECTORY

Normal Hearing

[*from page 24*]

would be non-logarithmic hearing, a characteristic which would not follow the Weber-Fechner law. In practice (*Fig. 10*) this usually means presence of hearing loss when listening to weak sounds, but less or no loss when hearing a loud sound. It might be considered as an ear with a built-in volume expander. Since in critical cases the usable listening volume range may be reduced to a zone only 5 db wide, it can become a problem.

Of course, discretion must be used in interpretation. As in connection with employment matters, small irregularities are not significant.

For a final note, some precautions should be observed if the audiogram is to be significant.

1. Be sure that the customer has not taken analgesics within the few hours just preceding the test.

2. Be sure that the customer has not imbibed too freely within the few hours just preceding the test. It is quite surprising to see the (temporary) effect of a few cocktails, on hearing.

3. Severe fatigue can produce a definite effect too.

Conclusion

Audiometric data on the normal hearing forms an invaluable adjunct to many audio activities, but the results must be realistically interpreted.

Musical Acoustics

[*from page 33*]

of the bearing surface is that bounded on a cone with spherical tip by a plane which intersects the same and is parallel to a side of the cone. The sectional area would be a portion of a parabola bounded at the ends by a circle and an ellipse when the stylus does not penetrate sufficiently to bear against the bottom of the groove. See *Fig. 3a*.

Where the tip radius and angle of the playback stylus are the same as those of the groove (a condition common to embossed home recording), there is added to the above areas the balance of the spherical segment whose height (or penetration in the bottom of the groove) equals the (normal to the walls of the stylus cone) indentation of the groove side walls multiplied by the cosecant of one-half the included angle of the stylus.

Fig. 3b shows the vertical projection of the additional spherical tip area and its considerable relative addition to the side bearing areas (projected normal to the slope of the stylus cone). It indicates the desirability of the same radius of playback stylus tip and sound-groove bottom in facilitating reduced unit bearing pressures upon the material of which the record is composed. This seems to be possible only with home-recorders.

In commercial recordings the playback stylus has a different angle and radius than the groove. Exactitude in grinding the tip radius is more difficult than grinding a conical slope. The use of an edged cutting stylus to make the groove precludes the possibility of a conical stylus from fitting such groove as the groove must vary in width and angle of cross-section as the cutting stylus follows a modulated sound track. An exaggerated sketch of this condition is shown in *Fig. 4a*. This results in a bearing on only one side of the playback stylus so only one half that quoted for the conditions of *Fig. 3c*. And if the angularity of the cutting edge to the track is too great for the slopes of the undulations of the sound grooves it may cut only a portion of a path and drag backwards between points (1) and (2), as indicated in *Fig. 4b*.

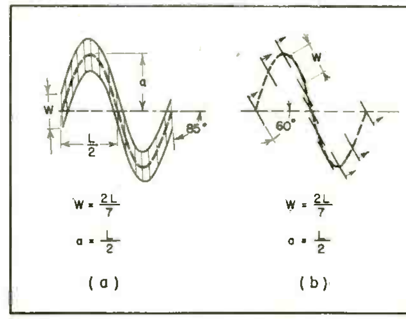


Fig. 4a. Variation of width in lateral-cut groove.

Fig. 4b. Effect of wrong cutting angle in sound groove.

So the bearing surface condition, as shown in *Fig. 3c* gives, if bearing against both sides of the groove, an area as projected normal to the slope of the playback stylus cone twice that which is an elliptical segment of a cone terminated by a smaller ellipse and a segment of a sphere. For the same penetration normal to the stylus conical slope the projected area is seen to be materially smaller than that area of *Fig. 3a*, upon which the internal dotted line shows its superposition. Where the area of *Fig. 3b* is added to twice that of *Fig. 3a*, as in home-recorders with embossed records using the same stylus for recording and playback, the bearing area is seen to greatly exceed that with the usual commercial phonographs.

Some limitations affect embossed sound tracks. Embossed sound tracks can also give a path which lacks fidelity unless the radius of the section of the stylus cone at the face of the disc (that is half of the unmodulated sound-track width) has a maximum radius not greater than the minimum radius of curvature of the center line of the modulated sound track curves. The larger such diameter the greater is the required unmodulated sound track velocity and, or, the lesser is the amplitude of stylus swing.

Recording Theory

[*from page 17*]

Since after playback the state of magnetization has returned to its initial condition, playback energy is not derived at expense of the magnetic energy stored in the media but from the energy supplied by the drive mechanism. This is directly analogous to a generator with a permanent magnet field where the output energy is drawn from the power source rotating the generator not from potential energy in the field magnets.

If demagnetizing forces for a certain wavelength bring the condition of magnetization to point *a*, *Fig. 7*, some longer wavelength will result in a state represented by a'. Upon playback it passes to b'. It is readily seen that the per cent change due to playback is greater the shorter the wavelength on the record.

The output of a magnetic tape is modified by two opposing tendencies. The output decreases with decreasing wavelength as the result of demagnetizing forces. The per cent recovery on playback increases with decreasing wavelength. The sum of the opposing tendencies is a net decrease in output with decreased wavelength.

Conclusion

We have seen from a qualitative discussion that variations in two of the magnetic constants affect the output of recording tapes. Actually the effects are not trivial and in the construction of recording materials careful consideration must be given to the proper balance of remanence and coercive force.

It is not proper at this point to speculate on the choice of optimum values of coercive force and remanence since all the evidence is not before us. Noise levels, erase conditions and velocity of tape drive must be taken into consideration as modifying and limiting our selection. The discussion of optimum magnetic constants will be deferred until these new factors have been considered in Parts II and III of this article.

Testing by Ultrasonics

[*from page 30*]

a beryllium ceramic shell, and be cooled by cold gas. If we operated outside the firebrick the temperature problem is easy, but we might wish to operate in the melt directly. Then the cooled ceramic tube picks up an inch thick layer of lower temperature steel which lowers the surface temperature on the ceramic nearly a thousand degrees for 2800-degree steel, so we would be well protected. This layer is quite constant and quite thin in relation to the total path through the mix, so any error introduced would be small. We would have a terrific change in transmission when the material melted, so we could shoot one beam along the bottom of a pot and get absolute indication when the charge was melted in a batch process.

Testing By Attenuation of Signal

This subject has been mentioned at some length several times in this series. It is most useful in flow processing to measure concentration of aerosols or dispersoids (small particles in gas or liquids, respectively). It suffers from being an amplitude system, and thus needs a transmitted signal of constant amplitude in the mix, and a stable and well calibrated receiver.

We have no clear concept as to just how the signal is weakened by a very small concentration of these particles. It does not seem to be a Tyndall or scattering action, but a lowering of Q of the mix by simple absorption. In many cases the dispersoid is of considerably different sonic impedance than the fluid, so that we can assume a certain reflection loss at the boundary layer of each particle. But in other cases, such as Kraft pulp, the loss in a six-inch path is half signal with only one-tenth per cent of pulp. Yet the sonic impedance of the pulp is almost exactly that of water, which is the fluid.

Reflection From Boundaries

The sonic impedance of any medium is the product of the density times the sonic velocity. In British units, it is pounds per cubic foot times the velocity in feet, and in c.g.s. units it is density in comparison with water times velocity in centimeters per second.

The simple boundary layer case is where you go from one medium to another and both are infinite in thickness. The wave is prime normal to the interface. If the sonic impedance of the first medium is Z_{s1}, and the second medium Z_{s2}, then the per cent reflection R is simply

$$\left(\frac{Z_{s1} - Z_{s2}}{Z_{s1} + Z_{s2}}\right)^2$$

If either or both mediums are only a fraction of a wavelength thick, the formula is fairly complicated, but if the wave arrives at an angle it is quite difficult to calculate the resultant wave pattern.

The sonic impedances of liquids and solids are not very far apart, but in most cases gas, while it has a respectable value of velocity, has a density only a ten-thousandth or so of either one, so the reflection is practically 100%. Thus it is almost fatal to have a very thin gas layer form on a diaphram in liquid, as the energy must go into the gas with a very large loss, and then again suffer a similar loss in going from gas to liquid.

A practical point in this case is to note that, under some conditions at least, a rough diaphram will "gas up" more readily than a well buffed one, as gas seems to form in tiny irregularities in the diaphram face, and then spread over the whole face. Some work has been done in attempting to discourage the formation of gas by a biasing voltage between the diaphram and the liquid, with little success. We have observed cases where gas spread all over a transmitter assembly and gave a resistance of over 100 megohms to tap-water.

For general use a monel diaphram is not too bad, but in boiling acids some considerable searching must be done to find suitable diaphram material.

Testing Uniformity

The Germans during the last war made considerable use of ultrasonic testing for uniformity of sheet stock, shell cases and the like. In this country some quite promising work was done in passing rolled stock between a transmitter and receiver under water, and by watching the signal attenuation quite small discontinuities were detected.

A very simple and bright idea was developed for determining thickness of metal by a one-sided test. A quartz crystal is put in an oscillating circuit and the crystal brought in intimate contact with the sheet metal. This formed half of a Langevin Sandwich and the frequency of the metal path added to the frequency of the crystal itself. The new frequency gives the thickness of the metal.

The future of supersonic testing looks quite bright, but a series of new techniques are called for in many cases.

IRE-WCEMA

[*from page 22*]

yet been found for physical size in proper reproduction of the lower reaches of the audio spectrum.

Donald Erdman, Triplett & Barton Co., Burbank, Calif., gave a paper on ultrasonic flaw-detection and demonstrated a portable echo-type unit.

Exhibitors at the WCEMA portion of the show included:

Altec-Lansing Corporation
American Microphone Company
The Brush Development Company
Burlington Instrument Company
Dalmo Victor Company
Eitel-McCullough, Inc.
Electro Engineering Works
Essex Wire Corporation
Gilfillan Bros., Inc.
Girard-Hopkins
Heintz & Kaufman, Ltd.
Hewlett-Packard Company
Hoffman Radio Corporation
The Institute of Radio Engineers
Kaar Engineering Company
Lake Manufacturing Company
The Langevin Company, Inc.
Lenkurt Electric Company
Magna Electronics Company
Remler Company, Ltd.
Sangamo Electric Company
Snyder Manufacturing Company
Sound Equipment Company
Stephens Manufacturing Company
Stromberg Carlson Company
Sylvania Electric Products, Inc.
Tartak Speakers, Inc.
Triad Transformer Mfg. Co.
Universal Microphone Company
Western Electric Company
Westinghouse Electric Corp.
Westline Electronics Company
Weston Electrical Instrument Corp.

Among large quantities of equipment of special interest to high-fidelity enthusiasts was a demonstration by Norman B. Neeley Enterprises of a Magnecord wire recorder playing into a James Lansing speaker system. Many critical listeners were impressed with the clean reproduction obtained from this combination. The recorder is reported to be flat within 2 db from 50 to 12,000 cps; to have a signal-to-noise ratio of over 45 db; and to have a maximum of 1.5 per cent harmonic distortion. The unit has, in addition, numerous features making for simple and straightforward operation.

Technicana

[*from page 5*]

Using a sound-storage index composed of the width of the sound track in mils multiplied by the speed of the film past the slit, in inches per second, the relative values come out to be 1800 for 24-frame 35-mm film, 576 for sound-speed 16-mm film, and only 108 for sound-speed (24-frame) 8-mm film. This is only 6 per cent of the sound storage index for standard film, and is a fair indication of the inadequacy of the 8-mm film for conventional recording methods. Add to this the difficulty of obtaining consistent development along the sprocket holes or along the edge of the film, and the problems are somewhat increased.

A relatively simple sound-pickup arrangement has been suggested which could be added as an accessory item to most 8-mm projectors with a minimum of change, and when using a film speed of 24 frames per second, the resulting sound quality compares favorably with that of superheterodyne radio receivers; the response curve is essentially flat to 3,000 cps, and down 10 db at approximately 4,300 cps.

The most important feature of this equipment is that the equipment may be added to existing silent projectors, and if the movie enthusiast wishes to do his own recording, there are no complicated problems—the recording is simply done on the completed film after editing, using a separate machine for the recording. At any time, the sound can be removed and rerecorded if desired, simply by using an erasing coil in conjunction with a high-frequency tone.

Since most motors employed on 8-mm projectors are of the series type with relatively poor regulation, it may be necessary to change the motors to a more-constant speed type, such as a standard a-c induction motor. Additional stabilization of speed may be found desirable, if music is to be reproduced.

Noise Suppressor

[*from page 10*]

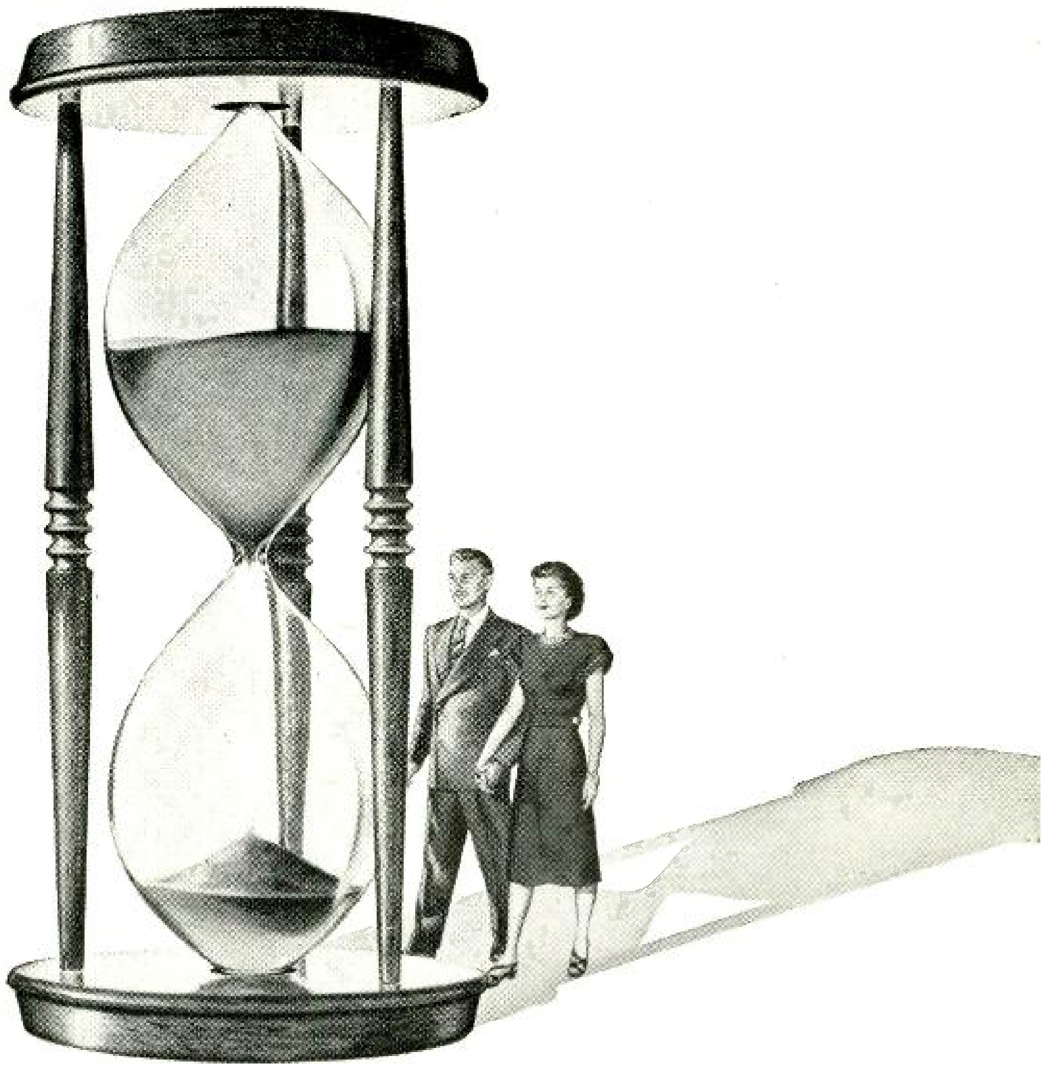
response range of the ear is greatly restricted at low intensity levels. Under many conditions of playback, listeners find it desirable to operate the reproducing equipment at high average intensity levels. Under these circumstances the observed noise level during quiet passages of the music or during intervals between notes is particularly distracting.

It is this latter circumstance that makes it possible to produce limited noise reduction with volume expanders and other "vertical" noise suppressors. These systems operate on a variation in amplitude of the entire (or block portions of) the frequency spectrum. For various reasons, including the fact that the control voltages are not derived from limited frequency bands, these systems are generally inadequate. This is particularly evident in connection with abrupt transients, such as the sharp clicks that characterize noise developed by vinylite and similar record materials. One form of vertical noise suppressor is a selective volume expander where the frequency spectrum is divided into three channels. The high and low-frequency channels may be independently driven by obtaining the control voltage after the dividing networks. A serious disadvantage is the fact that the entire high and low frequency ranges are opened "vertically" when only a limited portion may be needed for full range reproduction. An inverse form of this effect has been described.[1]

Pass Band

The Dynamic Noise Suppressor is of the "horizontal" type, providing dynamic adjustment of the pass band from approximately 2½ octaves to full range. This means that the signal-to-noise ratio is optimum at all times in terms of the instantaneous requirements of the music versus background noise. In broadcast work and in some home radio-phonograph installations, it is advantageous to be able to extend the controls electrically for remote operation. Fortunately, the controlled components are in portions of the circuits to facilitate this arrangement.

The most convincing evidence of the success of this method of noise suppression is actual demonstration. Many devices that work wonders in the laboratory do not prove successful in the field. The Dynamic Noise Suppressor is no longer questionable in this regard for it has withstood the test of widespread installation in radio stations and in home radio-phonographs and has won wide acclaim among engineers and music critics. It is of some importance to emphasize that the effectiveness of the results obtained may sometimes be limited by associated components and economic compromises



in design policies. A serious problem in the industry is the selection and training of sales personnel in dealers' stores to learn enough about the merchandise they sell to make effective demonstrations and to instruct their customers properly. The finest radio-phonograph, with or without a Dynamic Noise Suppressor, makes a poor showing if it is improperly operated.

The Dynamic Noise Suppressor does not eliminate the value of tone control circuits, but, on the contrary, makes it possible to operate high and low boosting circuits without the usual attendant increase in noise level. It will not produce music from blocks of concrete, but it will produce the most music and the least noise possible from any given record.

References

1. Goodell, John D., and Michel, B. M. H., Auditory Perception, *Electronics*, July 1946.
2. Scott, H. H., Dynamic Suppression of Phonograph Record Noise, *Electronics*, December 1946.

Record Revue

[*from page 34*]

(*Columbia M693*) reviewed last month. An opposite example is Concert Hall Society's Symphony for Strings, by William Schumann, which on casual hearing may sound definitely restricted in range (it isn't) because of the solid, homogeneous string tone without instrumental solos, the over-all type pickup, and rather dry, unresonant acoustics.

In these cases and plenty of others, isn't the impression of tonal range as important as the actual range itself?

Almost any subscriber to this magazine can muster far more extensive testing equipment than I have for the judgment of what there is to be heard on a record. If it were purely a matter of electrical testing, then this department would surely be superfluous—nor would there be much to argue about. But judging records, judging the effective, the practical over-all engineering as well as the music, is a matter as much for the ear as for the instruments, which are merely guideposts along the way. It is a matter of a trained ear and a good memory, for careful and comparative listening, for the application mentally of as many checks and balances as are available through experience. It is a matter of mental gymnastics, if you wish. Perfect technical equipment is not the answer; rather a knowledge of the defects and limitations of one's own equipment so that these may be discounted. One pins down as much as one can in the way of fact, one tries this and that setting. A GE pickup, a Pickering, a crystal, a couple of speakers, good and bad, a cheap amplifier as well as a good one—and then the work really begins, the correlating and the discounting.

As for going out on a limb, I'm quite

willing to if it is, so to speak, a constructive limb. Better to take a good guess (based on a lot of comparative experience) as to frequency range or mike placement than to ignore these factors entirely when they obviously count. Better to ascribe a very wrong reason for an actual fault than none at all! Better to make boners, learning the hard way, than to be over cautious. Boners or no, I'm confident that this department can pass on its batting average. If it can, the reason is again that the human ear is after all the greatest, simplest, most accurate testing instrument as to results that there is; because it is direct-coupled to the mind, whereas all the 'scopes and multitesters and sweep frequency records in the world can't define a simple word like "pleasurable" in sound reproduction.

Recent outstanding records of interest to the sound engineer:

Humperdinck, Hansel and Gretel.

Metropolitan Opera Company. Rise Stevens, Nadine Connors, etc. **Columbia M OP 26** (**2 vols.**)

This is the first of the new American-recorded complete operas with the Met company. Done with what appears to be a well worked out variety of the accentuation technique, it combines extreme clarity of instrumental and vocal detail (excellent and very intelligible word sounds) with a good sense of over-all liveness. This is in no way an attempt to reproduce "opera house realism," but rather a new kind of sound, artificial perhaps, but very much alive and realistic nevertheless. Appearance of very wide range (see article!).

Britten, Young Person's Guide To The Orchestra.

(Variations on a theme from Purcell's Abdelazer Suite. See Audio Engineering, October)
Liverpool Philharmonic, Malcolm Sargent. **Columbia M703**

English Columbia is rivalling English Decca in a similar type of recording. This is a part-humorous work (no narrative), a kind of "quick-brown-fox" of the instruments, with passages for every one in the orchestra. The recording is soft, resonant, with mild high end and usual low turnover of European recording. Impression of lack in highs on first hearing, but careful listening shows a very wide range. Highs can be boosted to taste.

Beethoven, Rasoumovsky Quartets, Opus 59, Nos. 1, 2, 3. (3 albums).

Paganini Quartet **Victor M1151, 1152, 1153**

A fine example of the importance of other factors than merely wide range in the matter of "high fidelity." These are finest quartet recordings in postwar period, possibly excepting some by Concert Hall Society. Big, full, natural tone, like an orchestra: shows the real power of a quartet when heard as it was intended to be heard—close to. Range is plenty wide enough—yet it is interesting to find that Victor's prewar recordings of the Budapest Quartet are almost identical with these in effect and are just as satisfactory. Range in both is probably restricted to 8500 or so. (See Mozart, Quartet No. 17, "The Hunt," *Victor M763*).

Bach, Concerto in D Minor for Two Violins.

Heifetz (both parts). RCA Victor Chamber Orchestra. **Victor M1136**

An interesting job of technical stunting. The synchronization is remarkable and it is impossible to tell which part was recorded first; quality is virtually identical. Yet technical imperfections still present. The sides differ markedly in "curve"; the first side is rather dull, lifeless, the last suddenly brilliant and harsh. The sense of liveness also seems to have suffered in the necessary dubbing; the recording is rather lifeless compared with other Victors.

Musically this merely proves how right old Bach was in his idea of a duet between two different personalities. With both played by same person the music is unbearably monotonous, completely lacks the intended give-and-take that makes it worth listening to.

Tschaikowsky, Symphony No. 1 ("Winter Reveries").
Santa Monica Symphony, Rachmilovitch. **Disc 801**

This is more satisfactory than previous recordings by this movie musicians' orchestra. It is on good surfaces, has a wide range. But again ill-advised acoustical characteristics and poor microphoning give a very false sound, which quickly disturbs most music lovers. "False," in that it sounds Hollywoodish, as though recorded on a sound stage. Engineers study this as sample of how not to record this type of music. Playing is excellent.

Wagner Program. (Siegfried Idyll, Faust Overture, Ride Of Valkyries).
NBC Symphony Orchestra, Toscanini........ **Victor M1135**

Strange that almost every Toscanini recording is, like this one, harsh, unmellow, and frequently so heavily recorded that few pickups can take the vibration without transmitting some unpleasant distortion. Softer portions are beautiful in the reproduction. Other conductors recording with this orchestra seem to get different results. An interesting question—is it merely the conductor's type of playing, or are there actual engineering differences? Ask Victor! For comparison try *Victor M1132* (Beethoven Piano Concerto)—same orchestra, different conductor. Presumably same equipment, etc.

New Products

[from page 36]

ing" effects in the reproduction. There is very little difference in pick-up as you pull away from the microphone. The abrupt change in output usually experienced on moving away from the microphone is eliminated. The harmonic distortion is less than 1%. The ribbon itself has a peak of 10 cps; above that, it is practically flat. The entire microphone has a rated frequency response of 50 to 11,000 cps plus or minus 2 db. Output is –62 db.

The discrimination with angle from 60 to 10,000 cps is less than 5%. (Angle discrimination in Studio Diaphragm Microphones is at least 400% higher).

The microphone comes complete with cable connector and a switch which is "hidden" in the back of the microphone in such a way as to prevent tampering.

The microphone is available in two models: Model RBHG (high impedance); Model RBLG, 50-200 ohms. Complete information, prices, will be promptly sent by the Amperite Co., Inc., 561 Broadway, New York 12, N. Y. Canadian Address—Atlas Radio Corp., 560 King St. W., Toronto, Ont.

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HIGH FIDELTY TUNERS

● Meissner has added to their line of precision built products new tuners for high-fidelity reception. The A.M. coverage of both units is 527 to 1620 kc and the newly

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Operated at 105-125 volts, 50-60 cycles input the power consumption for the AM-FM Tuner is 80 watts; 60 watts for the AM Tuner. The output is 8 volts at high impedance terminals and .75 volts at 500 ohm terminals, with a hum level 60db below output level. Both units are provided with phono input terminals so that they may be used with record players.

For further data, write the Meissner Manufacturing Co., Mt. Carmel, Ill.

ADVERTISING INDEX

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