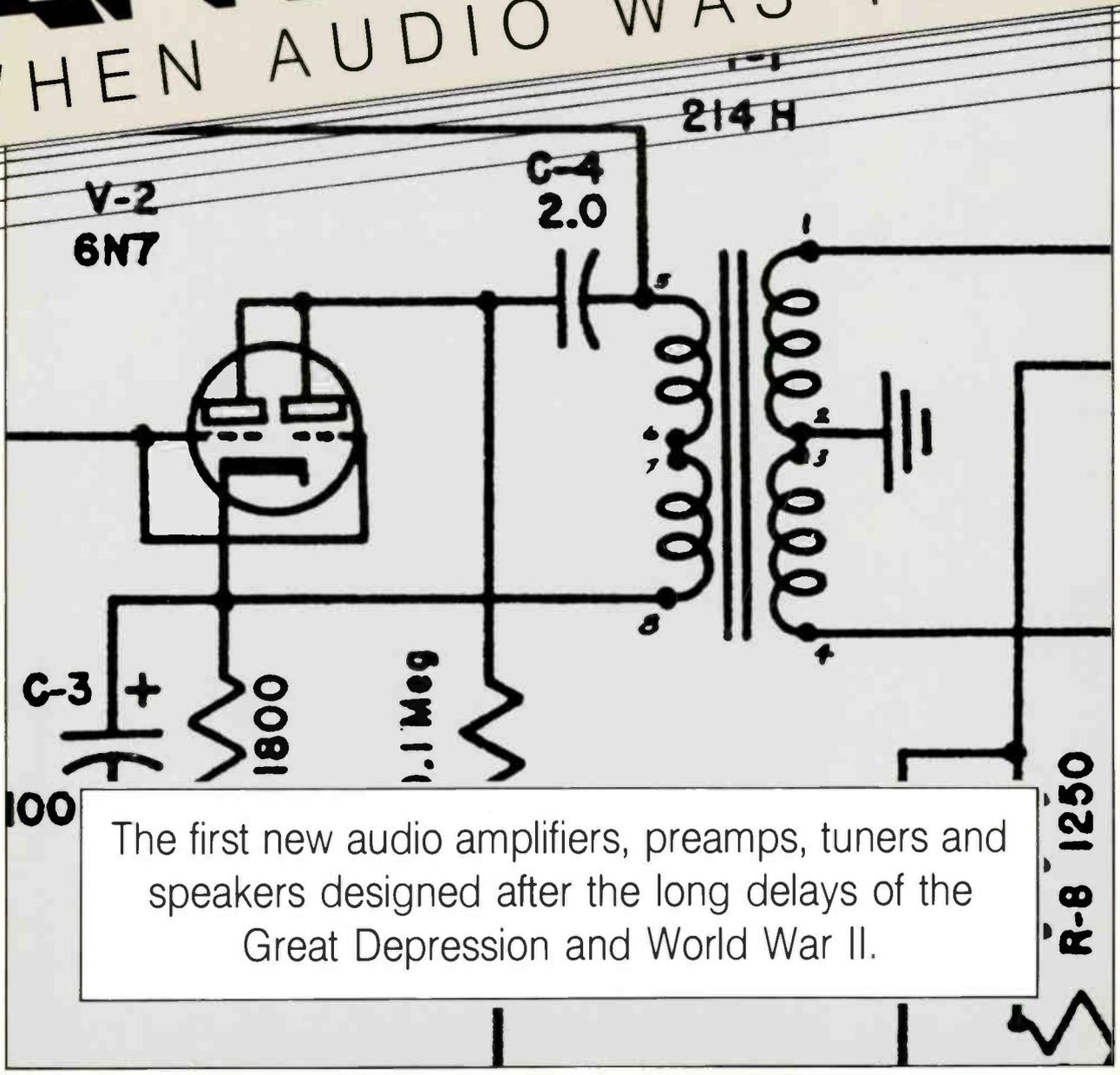


# AUDIO ANTHOLOGY

VOLUME ONE

WHEN AUDIO WAS YOUNG



The first new audio amplifiers, preamps, tuners and speakers designed after the long delays of the Great Depression and World War II.

R-8 1250

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VOLUME ONE  
1987**

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# audio anthology

VOLUME ONE

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From May 1947 to December 1949

by

C. G. McProud

Editor

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# Preface

Unless your earliest memories of sound reproduction are steeped in the smell of varnished cabinets with chrome-plated bowls filled with steel and cactus needles, as well as a handle on the side that needed cranking often to keep the platter spinning, you will not, perhaps realize how surprising and startling the new world of electronically reproduced records seemed in the late forties. Radio was a matter of large pieces of furniture in the living room which, if they were expensive enough, included short wave and a green eye that looked sleepy when tuned.

A great deal of the basic design work on sound reproduction had begun in the late twenties and early thirties. One need not look further than the papers from Bell Labs or the pages of *The Gramophone* to find discussions of sound quality, tracking distortion, and directionality. EMI's patents on our current two channel reproduction system, based on Alan Blumlein's work, were obtained in 1932.

The enforced dormancy imposed on audio technology's flowering by the Great Depression and World War II resulted in a rocket-like acceleration in its development afterward. C. G. McProud founded *Audio Engineering Magazine* in 1947 and thus provided a central clearing house for what designers and enthusiasts were doing to develop the hardware for producing high quality sound. Only three years after the magazine's inception, McProud reprinted a selection of articles, all carefully corrected, from the publication's pages.

Those of us who were just discovering the contrast between the mechanically reproduced sound from the old phonographs, and compared it to the new electronically reproduced version, had an experience that caused an almost unquenchable thirst for more. At the time *Audio Engineering* was being first published, Armstrong's FM was an untried novelty, almost simultaneously, Goldmark's longplay disk appeared. The GE variable reluctance cartridge had stylii (a new word we all learned quickly) for both microgrooves and the old 78s as well. It was a heady time when developments of far reaching significance were appearing in every other issue.

This is an appropriate place to offer a small tribute to the remarkable work of C. G. McProud. His magazine was highly professional, the content carefully checked and presented, and the range of topics remarkably broad. His work laid the groundwork directly for the founding of the Audio Engineering Society. McProud's work, both as editor, designer, engineer and author, is a monumental achievement. His work was far more than theory. He was a thorough and careful engineer, but also an accomplished technician and artisan. His handiwork in realizing the projects he designed was as elegant as the designs themselves.

We believe the record of those days, as contained in McProud's anthologies, is something many in this generation will welcome. This first one, if the reception is enthusiastic, will be followed by others.

Edward T. Dell, Jr.  
Editor/Publisher  
*Audio Amateur* and *Speaker Builder* Magazines

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# *Foreword*

Since the first issue of AUDIO ENGINEERING made its appearance in May 1947, those interested in audio—either as a profession or as a hobby—have found many articles which they considered useful or valuable in their work. The demand for these articles exhausted early issues long ago, and newer readers have been unable to locate them.

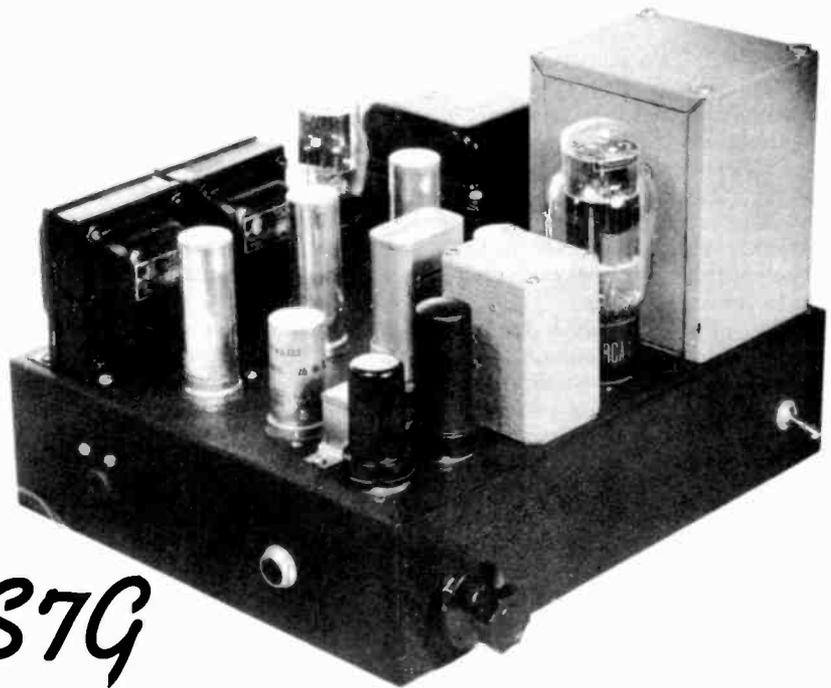
The articles attracting the greatest interest are those which were directed to the audio hobbyist, largely because professional audio men learned of AUDIO ENGINEERING long before the hobbyist did and their files are complete. It is primarily for these newer readers that this collection of articles has been compiled. Most of the articles about amplifiers, loudspeakers and enclosures, and phonograph equipment in general are here reproduced, and will serve to bring many readers' files up to date. Others may prefer to have these articles in one volume instead of in separate magazines which are not readily stored or indexed, and not as convenient for reference.

Many of the articles in this volume have been brought up to date with additional notes, substitute transformer listings, or minor changes, and errors in the original printing have been corrected. To those who have so anxiously searched for unavailable back issues and so regularly requested reprints, this volume is appreciatively dedicated.

C. G. McPROUD, *Editor*  
AUDIO ENGINEERING

New York  
June, 1950

# High Quality Amplifier with the 6AS7G



C. G. McPROUD

Outstanding performance from an ultra-modern amplifier

**A**LTHOUGH not particularly new, the 6AS7G twin triode recently publicized by RCA as an audio output tube is an interesting device, and natural curiosity prompts the experimenter to use the tube in an amplifier to determine its capabilities. The 6AS7G, briefly, was originally designed with two principal uses in view—one being a high-current substitute for the usual 2A3 in tube-controlled voltage regulator circuits, and the other as a damper for the horizontal sweep circuits in television receivers where greater power is required than was previously obtainable with existing tubes. In this particular application, the requirement is for a short duty cycle, but one in which the current is relatively high. Recently, however, it has been given considerable attention as an audio amplifier.

Physically, the 6AS7G is a fascinating tube. In cross section, the elements resemble the sketch in Fig. 1, with each plate being composed of two separate and very deep channels, with the short dimension serving as the plate proper, the longer sides being used for heat dissipation. Above and below the element structure are the two radiators for the grids, also used to dissipate the heat. Compared with the 2A3, the 6AS7G is almost the equal of two of the former tubes. Plate dissipation for the 2A3 is 15 watts, whereas that for the 6AS7G is 13 watts *per section*, making a total of 26 watts for the tube. Thus it is seen to be useful as an output tube.

### Advantages

There are a number of advantages to the 6AS7G which are not obtainable in any other output triode. In the first

place, it uses a heater-cathode construction, eliminating the problem of reducing hum from this source. The two sections of the tube are completely separate, permitting the use of separate cathode resistors to obtain grid bias, and making it a simple matter to balance plate currents in the two sections. The plate

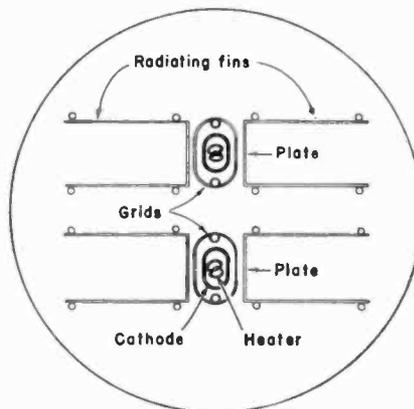


Fig. 1. Cross-section of element structure of the new 6AS7G, showing the unique plate construction with adequate area for heat radiation. Grid radiators are alternately above and below the element structure.

resistance is relatively low, being of the order of 670 ohms at the operating point selected. With this value of plate resistance and the high-current capability of the tube, the plate efficiency is seen to be high. The amplification factor of the tube is 2.1. When working into a 3000-ohm plate load, the effective output impedance of the amplifier—at a 16-ohm winding, for example—is 7.2 ohms, which provides a good damping

factor for an amplifier used to drive a loudspeaker.

### Design of the Amplifier

Proponents of high-quality reproduction are divided over the relative merits of triodes versus pentodes-with-feedback, but the current trend seems to be back (?) toward the use of triodes for the output tubes in the better apparatus. With this in mind, the 6AS7G was investigated, and since it appeared to have certain advantages over the other tubes of similar power output

TABLE 1  
AUDIO AMPLIFIER SERVICE  
Values are for each unit

Maximum Ratings, Design Center Values:		
PLATE VOLTAGE	250 max.	volts
PLATE CURRENT	125 max.	ma
PLATE DISSIPATION	13 max.	watts
PEAK HEATER-CATHODE VOLTAGE		
Heater negative with respect to cathode	300 max.	volts
Heater positive with respect to cathode	300 max.	volts
Typical operation, Class A <sub>1</sub> push-pull amplifier. Unless otherwise specified, values are for both units.		
Plate	200	250 volts
Grid	-90	-125 volts
Cathode Resistor (per unit)	1500	2500 ohms
Peak AF grid to grid voltage	180	250 volts
Zero signal plate current	120	100 ma
Max. signal plate current	126	106 ma
Effective load resistance (plate to plate)	4000	6000 ohms
Total harmonic distortion (less than)	4	4 per cent
Max. signal power output	11	13 watts
Amplification Factor (per unit)	2.0	2.0

\*It is essential that precaution be taken in equipment design to prevent subjecting the tube to full load current of 250 ma before its cathodes have reached normal operating temperature. The cathodes require approximately 15 seconds to attain normal operating temperature. Unless this precaution is observed, the cathodes will be seriously damaged, if not completely ruined. In speech amplifier service, as indicated under typical operating conditions, the plate voltage may be applied simultaneously with the filament voltage.

capabilities, an amplifier was designed around it. This amplifier was designed on the basis of the highest fidelity of reproduction, and used broadcast quality transformers. It was not intended that any short cuts would be employed, but simply that a good, stable, reliable amplifier should result. The finished amplifier reasonably well justifies the time spent, since the performance comes up to the expectations.

The values published by RCA for typical operating conditions are shown in Table 1. This table indicates that the tube draws a high plate current, requires a high grid bias, and is operated with separate cathode resistors to develop this bias. The high bias also indicates a high signal voltage applied to the grids, which poses somewhat of a problem, as will be discussed later. Under the 200-volt condition, the plate dissipation is 12 watts per unit, which appears to be too

close to the maximum to provide a factor of safety. Therefore, a slightly higher bias voltage was chosen, in order to reduce the plate dissipation to 10 watts, as well as to reduce the plate current through the primary of the transformer to a more reasonable value.

Commencing the design, the output stage is considered first. A study of the typical operating conditions indicates a plate voltage of 200 and a bias of -95 volts, making a total of 300 volts, allowing for a 5-volt drop in the output transformer, which may be a trifle high. However, a supply voltage of 300 is practicable, and the amplifier is designed on that basis.

Using high quality transformers restricts the choice considerably, but two transformers of Audio Development Company manufacture appear to be suitable for this application. The Type 314C output transformer is equipped with a

primary for 3,000 or 5,000 ohms, and with secondary impedances of 600, 150, 16, 8, 4 and 2 ohms, which provides sufficient versatility for most uses. Since the 6AS7G should be operated with a relatively low resistance in the grid circuit for optimum performance, the coupling to the preceding stage should preferably be a transformer. Another reason for the use of a transformer is that it is difficult to obtain sufficient voltage swing from any existing tube in a resistance-coupled circuit unless the grid resistor is made large. Reference to tables in the tube handbook indicate that a 6AU6 would provide adequate grid swing as a resistance-coupled driver, but it seems rather ludicrous to employ two miniature tubes to drive a combined output tube. Furthermore, the added complication of a phase inverter is avoided by using a transformer.

Since some voltage step-up is desired

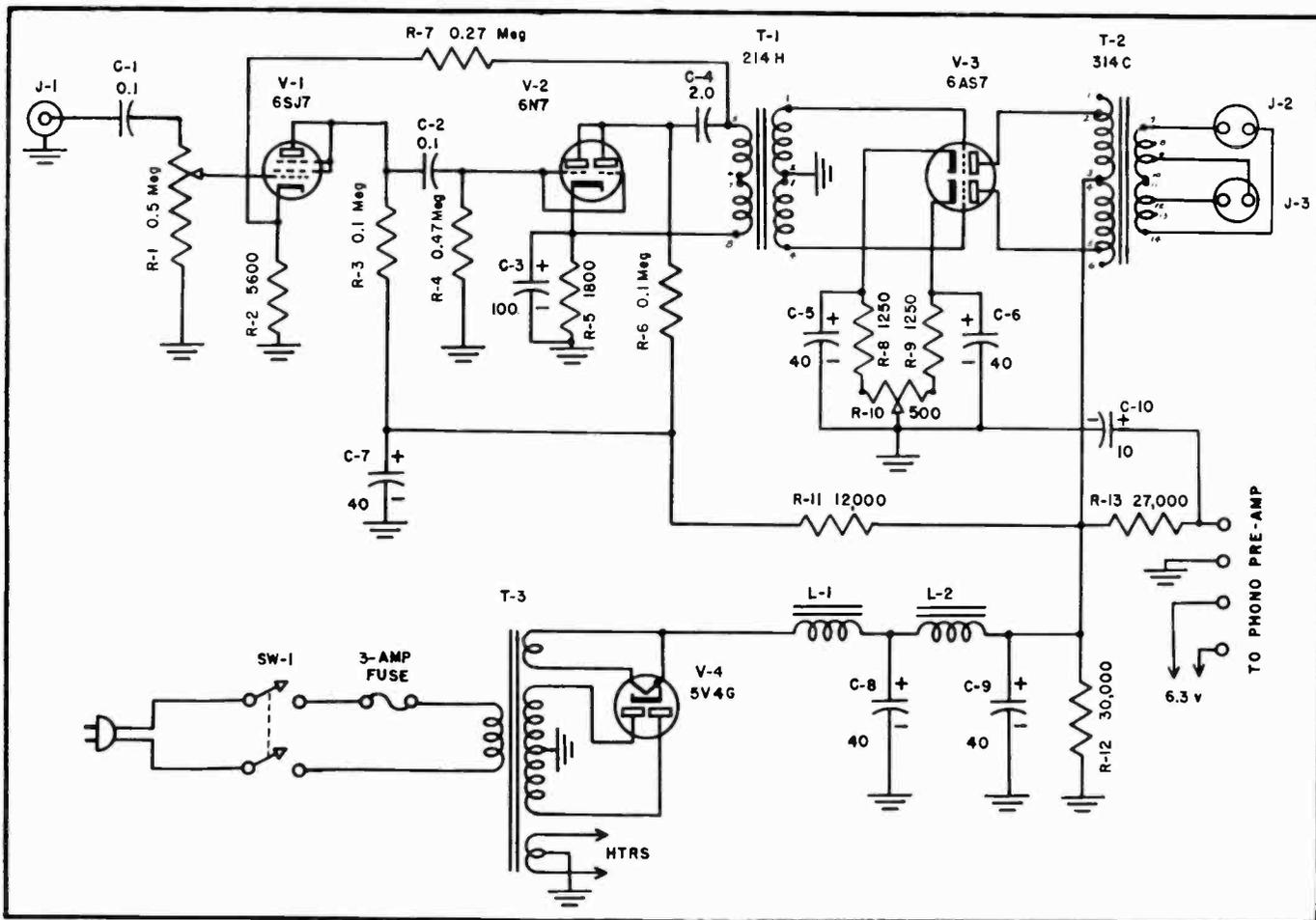


Fig. 2. Schematic of 6AS7G amplifier, showing provision for furnishing plate and filament power to external pre-amplifier.

- C<sub>1</sub>—0.1  $\mu$ f, 600-volt, tubular
- C<sub>2</sub>—0.1  $\mu$ f, 600-volt, Aerovox 618B
- C<sub>3</sub>—100  $\mu$ f, 25-volt, CD UP4AJ23
- C<sub>4</sub>—2.0  $\mu$ f, 600-volt, CD TJU-6010
- C<sub>5</sub>, C<sub>6</sub>—40-40  $\mu$ f, 150-volt, CD UP4415
- C<sub>7</sub>, C<sub>8</sub>, C<sub>9</sub>—40  $\mu$ f, 450 volt, CD UP4045
- C<sub>10</sub>—10  $\mu$ f, 450 volt, CD BR1045
- J<sub>1</sub>—input jack, Amphenol 80C
- J<sub>2</sub>—600-ohm output jack, Amphenol 80PC 2F
- J<sub>3</sub>—8-ohm output jack, Amphenol 80PC2F

- L<sub>1</sub>, L<sub>2</sub>—7.0 H, 140 ma choke, Stancor C-1421
- R<sub>1</sub>—0.5 meg. volume control, IRC D13-133
- R<sub>2</sub>—5,600 ohms, 1 watt
- R<sub>3</sub>, R<sub>6</sub>—0.1 meg, 1 watt
- R<sub>4</sub>—0.47 meg, 1 watt
- R<sub>5</sub>—1,800 ohms, 1 watt
- R<sub>7</sub>—0.27 meg 1 watt
- R<sub>8</sub>, R<sub>9</sub>—1,250 ohms, Ohmite 10 W. Brown Devil
- R<sub>10</sub>—500-ohm wirewound potentiometer

- R<sub>11</sub>—12,000 ohms, 2 watts
- R<sub>12</sub>—30,000 ohms, Ohmite 10 W, Brown Devil
- R<sub>13</sub>—27,000 ohms, 1 watt
- T<sub>1</sub>—Audio Development Co. 214H, 10,000/80,000 interstage
- T<sub>2</sub>—Audio Development Co. 314C, 3,000-5,000/600-16-4, output
- T<sub>3</sub>—400/0/400 at 200 ma; 5 v at 4 amps; 6.3 v at 5.5 amps, Stancor P-6165

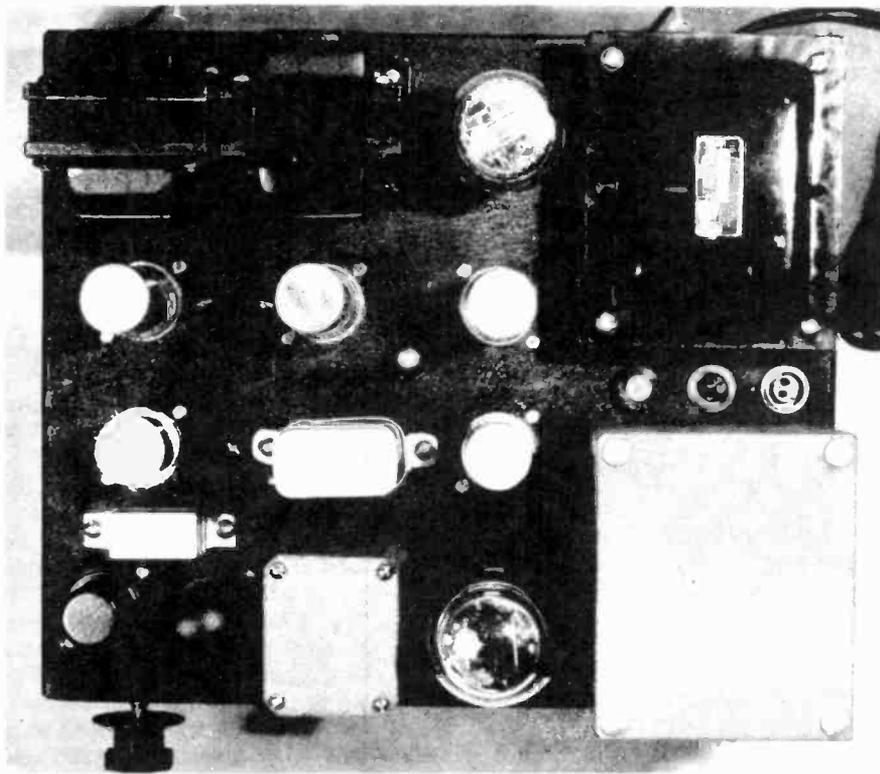


Fig. 3. Top view of amplifier, showing placement of parts. Note simplicity of arrangement. Capacitor above interstage transformer is  $C_4$ , used in shunt feed circuit. Note use of inverted can capacitor for coupling between 6SJ7 and 6N7, shown directly above the two tubes.

from the interstage transformer. the 214H was selected. This transformer has an impedance ratio of 10,000 to 80,000, resulting in a voltage gain of 2.83. The peak grid-to-grid swing is—with the bias chosen—190 volts, or an rms swing of 138 volts. This requires a signal at the plate of the driver tube of 138/2.83, or 49 volts, a value which is readily obtainable from triodes of the 6J5 type. However, an equal amount is obtainable from the 6N7 with ease, and with approximately twice the voltage gain in the tube. The 6N7, with the two sections in parallel, is often used as a driver tube since it is capable of furnishing rather a large amount of power. Although the 6AS7G is operating strictly Class A, some power is to be dissipated in the feedback circuit, used to improve the characteristics of the driver, which is the hardest-worked tube in this combination.

Relatively little total gain was required from the amplifier, so it was considered suitable to use a triode as the first stage. In order that the gain could be increased readily by the simple change to a pentode connection, the 6SJ7 was selected. With this tube lineup, the total gain of the amplifier, without any feedback, was calculated at 68 db. Planning on 20 db of feedback, a net gain of 48 would result, which is still more than sufficient for a "main" amplifier.

#### Power Supply

The power supply is conventional, using a choke input and two sections for adequate hum filtering. One of the

requirements of the 6AS7G is that the plate voltage should not be applied until the cathodes have attained normal operating temperature. This requirement is waived in speech amplifier service where the bias is obtained from cathode resistors, but since it is of some importance in certain applications, it is considered

desirable to limit the application of the plate voltage to some extent by using a 5V4G as the rectifier tube. This tube may be operated with up to 500 volts on each plate for choke-input filters, and is eminently suitable for this purpose.

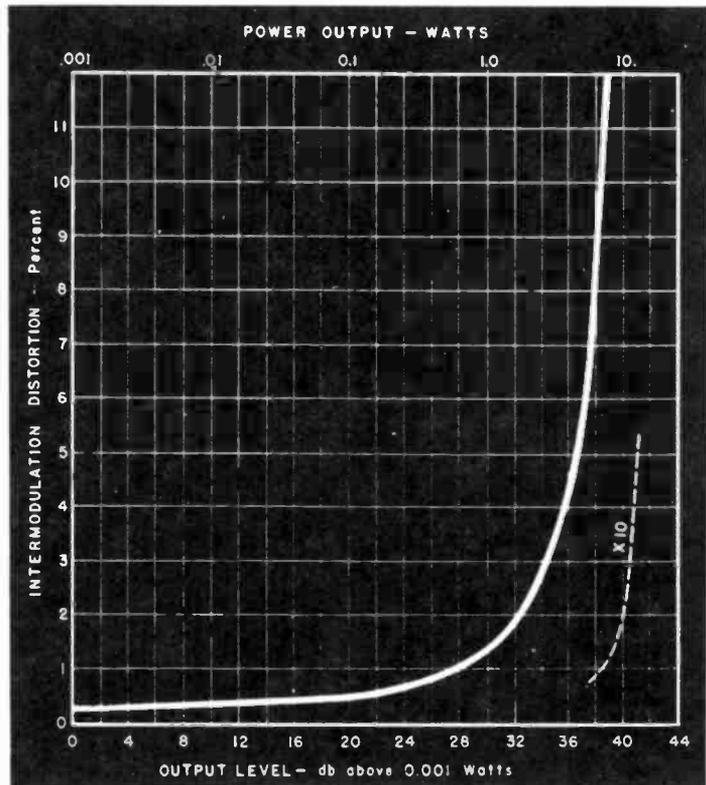
Desiring an amplifier suitable for continuous operation with adequate reserve to keep heating to a minimum, the power transformer selected is designed to furnish 200 ma at 400 volts each side of center tap, 5 volts at 4 amps for the rectifier heater, and 6.3 volts at 5.5 amps for the amplifier heaters. The 5V4G requires but 2 amps, and the tube complement of the amplifier draws 3.6 amps, which means that the entire power supply is "coasting."

#### Balancing Plate Currents

Returning to the output stage, some provision is desired for balancing the plate currents of the two sections of the 6AS7G, so instead of using separate resistors of equal value, each being 1500 ohms, it was decided to employ two 1250-ohm resistors, connecting to the ends of a 500-ohm potentiometer with the arm grounded. This arrangement provides sufficient control to balance the plate currents. Each cathode is by-passed with a 40- $\mu$ f, 150-volt electrolytic capacitor, a dual unit being used to conserve space.

The plate load resistor for the 6N7 is 0.1 meg, being coupled to the primary of the interstage transformer by a 2.0- $\mu$ f 600-volt oil-filled capacitor. At this same point the feedback resistor is connected, providing the isolation for both circuits with the same capacitor. The

Fig. 6. Intermodulation distortion curves at various output levels.



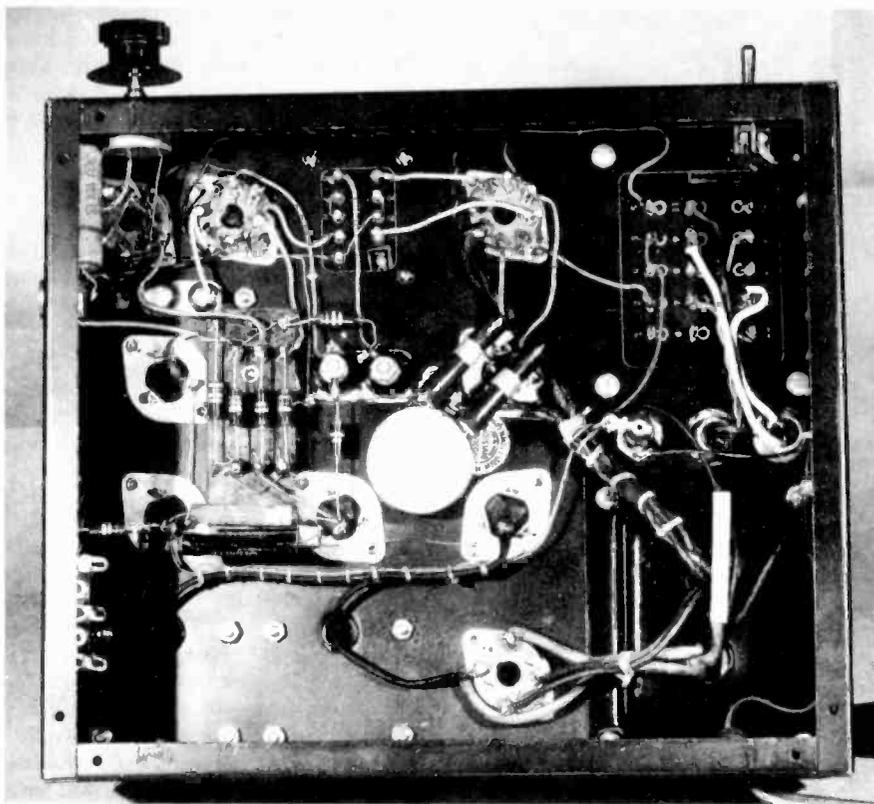


Fig. 4. Under-chassis view of amplifier.

cathode of the 6N7 is by-passed with a 100- $\mu$ f, 25-volt capacitor. The feedback resistor connects to the unby-passed cathode of the 6SJ7. The complete schematic is shown in Fig. 2, with the arrangement of parts both above and below the chassis shown in Figs. 3 and 4.

As constructed, the gain of the amplifier without feedback is 67 db, measured from an unterminated 500-ohm gainset, or an actual gain of 61 db. With 26.7 db of feedback, the gain reduces to 34.3 db, without a transformer at the input.

#### Increasing Gain

If more gain is desired, the first stage may be changed to the pentode connection, with an increase in gain of approximately 20 db, and a 600/50,000 input transformer will give nearly 20 db more. Referring to the schematic of Fig. 5, it will be noted that the changes necessary to convert the first stage to a pentode are relatively simple—only one resistor is changed, and one resistor and one capacitor are added. Changing the value of the cathode resistor from 5,600 ohms to 680 ohms automatically compensates for the difference in gain for the two tubes with respect to the feedback connection, and provides approximately the same amount of feedback—in db of gain reduction—when using the same resistor for R7.

#### Performance

With the design considerations outlined, the amplifier was constructed as

shown, and then tested to determine how well the 6AS7G tube performed. The principal information required of an output stage is the maximum power output, and the output at a stated percentage of harmonic distortion, usually 5 per cent. Modern engineering and testing standards require information as to the intermodulation distortion in addition to the above. It is generally accepted that an amplifier designed for high-quality reproduction must be capable of

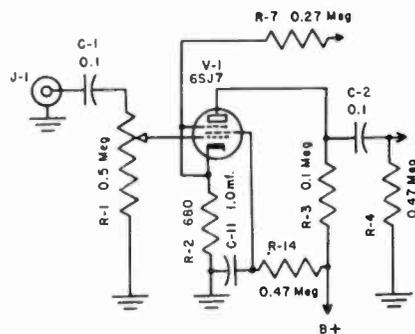


Fig. 5. Schematic of changes to circuit of  $V_1$  when used as a pentode for increase of approximately 20 db in gain. Only required changes are in cathode resistor  $R_2$  from 5,600 ohms to 680 ohms, and addition of  $R_{14}$  and  $C_{11}$ . No change is necessary in feedback resistor,  $R_7$ .

furnishing sufficient output power at a harmonic distortion of less than 1 per cent to be completely satisfactory, and that the intermodulation distortion should be under 2.5 per cent for optimum performance.

This amplifier underwent the necessary tests, and the results are gratifying. The power output at 20 per cent intermodulation distortion, equivalent to 5 per cent harmonic distortion, was measured at 9.5 watts, and at 4 per cent intermodulation distortion (1% harmonic distortion approximately) 3 watts. The intermodulation distortion at 5 watts output is 6.3 per cent, and at 1 watt, 1.3 per cent. This indicates that the performance of the amplifier is satisfactory for the highest quality requirements for home or monitoring applications, using efficient speaker systems where the power requirements are not as severe as in public address systems.

One of the principal advantages of the 6AS7G shows up in the measurement of hum, this value being  $-45$  dbm. When compared to the conventional 2A3 amplifier, this is somewhat better, and it should be noted that no special precautions were taken to minimize hum, nor were any of the "tricks" used. The measured gain is 34.3 db with feedback, and since the frequency response is flat within 0.5 db from 20 to 15,000 cps, it is not considered necessary to reproduce the curve. Intermodulation distortion curves are shown in Fig. 6. At 1 watt output, the output impedance at the 600-ohm winding is 258 ohms; at the 16-ohm winding, it is proportionately less than 8 ohms which provides excellent damping for a loudspeaker.

#### Conclusions

Tests of this amplifier indicate that the 6AS7G is an ideal substitute for a pair of 2A3s or 6B4Gs in the output stage of a quality amplifier. The low voltage and high current requirements of the tube reduce possibility of failure in filter capacitors, especially when electrolytic units are used, and the simplicity of the amplifier makes it ideal for applications where continuous service is required. Although phase inverter circuits are considerably less expensive than high-quality transformers, there is one thing to be said in favor of the transformer—once installed, it may be depended upon for years of service, and its characteristics are not dependent upon the gain of a tube or the constancy of a resistor. When shunt-fed circuits are used, there is little chance of burning out the primary (unless the coupling capacitor fails) and no trouble need ever be anticipated with a good transformer. The characteristics of a transformer-coupled amplifier may be counted on to be constant as long as the unit is in use.

The use of the 6AS7G in quality amplifier service definitely indicates a trend toward simple, reliable design, and if operated at reasonable plate dissipation, the tube should give continued satisfaction.

# AUDIO DESIGN NOTES

## MICROSECONDS-DB CONVERSION

NUMBERS IN COLUMNS ARE FREQUENCIES IN CYCLE PER SECOND

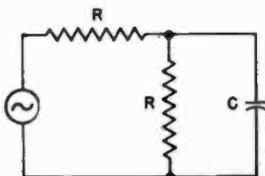
DB. GAIN OR LOSS	100 $\mu$ sec	75 $\mu$ Sec	50 $\mu$ Sec	25 $\mu$ Sec
0	500	670	1,000	2,000
1	850	1,130	1,700	3,400
2	1,200	1,600	2,400	4,800
3	1,600	2,150	3,200	6,400
4	2,000	2,700	4,000	8,000
5	2,400	3,200	4,800	9,600
6	2,800	3,700	5,600	11,200
7	3,250	4,300	6,500	13,000
8	3,800	5,100	7,600	15,200
9	4,400	5,900	8,800	17,600
10	5,000	6,700	10,000	20,000
11	5,600	7,500	11,200	
12	6,300	8,400	12,600	
13	7,100	9,500	14,200	
14	8,000	10,700	16,000	
15	9,000	12,000	18,000	
16	10,000	13,300	20,000	
17	11,200	15,000		
18	12,600	16,800		
19	14,200	18,900		
20	16,000			

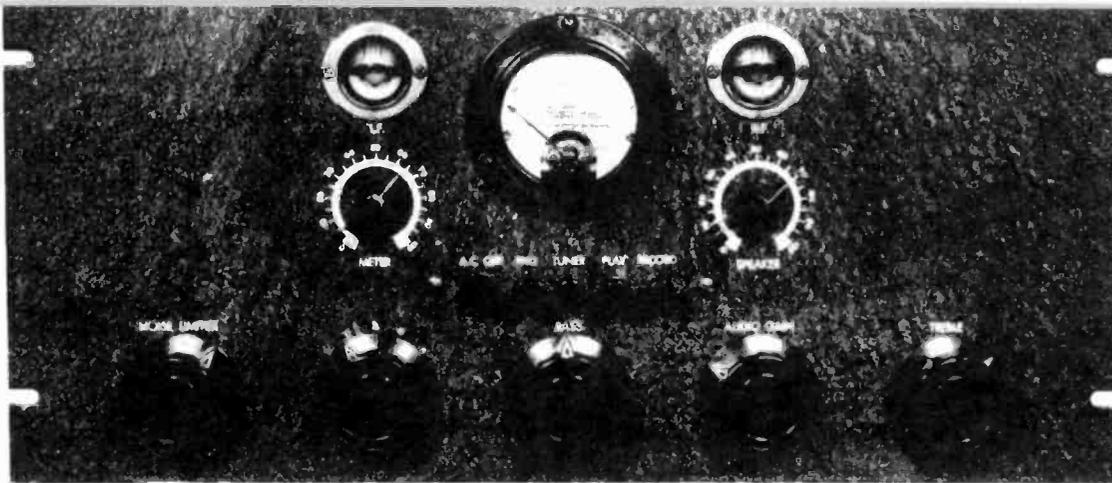
WHERE NUMBERS HAVE BEEN ROUNDED OFF, THE FREQUENCY ERRORS DO NOT EXCEED 1 PERCENT.

RC = time constant ( $\mu$ Sec)

TO FIND THE FREQUENCY AT A GIVEN LOSS OR GAIN FOR ANY OTHER TIME CONSTANT ( $t_x$ ), MULTIPLY FIGURES IN COLUMN 1 (100  $\mu$ Sec)

BY  $\frac{100}{t_x}$





Panel view of amplifier described in this article.

# General Purpose 6AS7G Amplifier

C. G. McPROUD

Modifications of the 6AS7G high-fidelity amplifier to provide bass and treble tone controls, a dynamic noise-suppressor, high-gain input stage, and recorder feeds—sectionalized for flexibility.

**A**LTHOUGH the three-stage high-quality amplifier of the preceding article leaves little to be desired in the way of performance, it has elicited many requests for information as to the best method of adding some form of tone control in the circuit. The gain of the amplifier is sufficient to accommodate certain types of tone controls, but others require still more amplification in order to perform correctly. The feedback placed around the first two stages eliminates the interstage coupling circuit as a location for tone controls, and in general, the amplifier is not suitable for this modification.

Added to this difficulty, some interest has been shown in the possibility of adapting the basic circuit to accommodate a dynamic noise-suppressor amplifier, and as a further incentive, one request was received for circuit data to

permit the connection to a crystal cutter for use as a recording amplifier. In order to make the unit still more complete, it was decided to incorporate a low-level stage for use with a magnetic pickup of the Pickering or GE Variable Reluctance type.

The amplifier described in this article was accordingly designed and built as an answer to all these requirements. It is relatively complicated in construction—but only because it has a large number of components. The layout is straightforward, and the adjustments necessary to put the noise-suppressor section into operation are not difficult, *provided* an audio-frequency oscillator is available. One feature of this design is that it is sectionalized so that the basic amplifier may be constructed either with or without the noise suppressor, or it may be constructed without the preamplifier

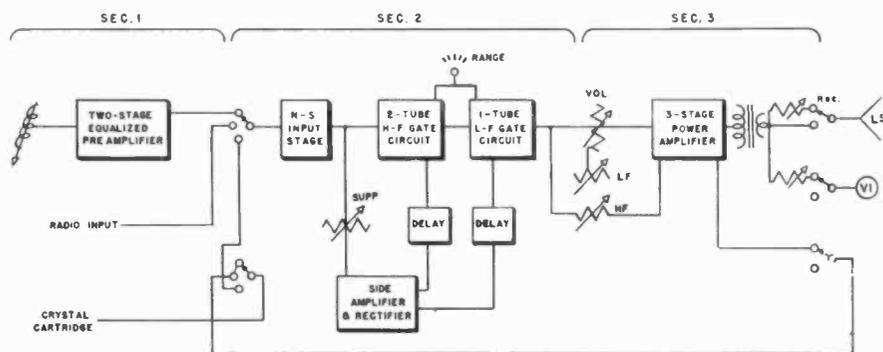
stage if it is to be used with an external preamplifier or with a crystal pickup. By so sectionalizing the design, it can be adapted readily for any specific requirement the user may wish.

## Circuit Arrangement

The basic arrangement of the amplifier, reduced to its simplest form, is shown in the block diagram, *Fig. 1*. The power supply is omitted for the sake of simplicity. The first section includes the two-stage preamplifier, equalized by feedback to compensate for the low-end recording characteristic of commercial records. A three-position switch permits the selection of the desired input source—either phonograph, radio, or a recorder.

The second section is the dynamic noise-suppressor amplifier, which follows the H. H. Scott circuit (with some modifications lifted from the Goodell version of the original Scott amplifier). This section incorporates an input stage, providing a source impedance of the proper value, and enough gain to actuate the side amplifier which furnishes the control voltages. The input stage is followed by a two-tube high-frequency gate circuit and a single low-frequency gate circuit, together with the necessary control amplifier and rectifiers. The output of this section may then be fed directly into the output amplifier. This consists of three stages, essentially identical with the original 6AS7G amplifier. It employs a tapped volume control to provide an increase in low-frequencies which may be reduced at will by the low-frequency tone control, and a treble control which in-

Fig. 1. Block diagram of sectionalized amplifier suitable for wide variety of uses.



creases or decreases the high-frequency response as desired.

The output circuit contains switching arrangements which connect the speaker directly to the secondary of the output transformer for normal use, or through a variable resistor for monitoring, and which also connect a volume indicator and the recorder to the output stage during recording. Although the switching appears to be complicated in that it requires a number of operations to change from record to playback, such is not the case because of the use of a push-button switch which performs all of the switching operations quite simply.

The type of recorder used will control the switching circuits to a great extent. For the disc recorder unit used in this particular job, it is necessary to connect the crystal cartridge to either the input or the output of the amplifier. Most recorders employ a separate unit for playback, which simplifies this switching. It is desirable to connect the cutter of the disc recorder unit to the plates of the output tube through capacitors, giving a constant-amplitude characteristic over the entire range. With other crystal cutters, some series resistance should be employed to provide a characteristic which is similar to standard phonograph records. Low-impedance magnetic cutters will naturally be connected to a source of the correct impedance. Various arrangements for these connections are shown in Fig. 7.

The power supply section is conventional, with the exception of the d-c filament supply which is simply a connection between ground and the center tap of the high-voltage secondary through the heaters of the first two tubes,  $V_1$  and  $V_2$ . Both of these tubes have 12-volt heaters, and they are connected in series. After the amplifier is completed, a value of bleeder resistor is selected which will make the total current drain equal 140 ma. It will be noted that this is slightly below the normal value, but the low-level stages operate perfectly with the lower current, and are somewhat less susceptible to the slight a-c component remaining in this supply. The amplifier is quite low in hum, but because of the large number of stages it is desirable to take all possible precautions.

### Tone Control Methods

The methods of adding high- and low-frequency tone controls to the circuit are relatively simple. For general use in home reproduction systems, it is usually considered desirable to employ a tapped volume control so that when the output level is lowered, the frequency is altered somewhat in accordance with the Fletcher-Munson curve. A control with a single tap will not give complete compensation, but it is some improvement over an untapped control. The circuit used for this compensation consists of a

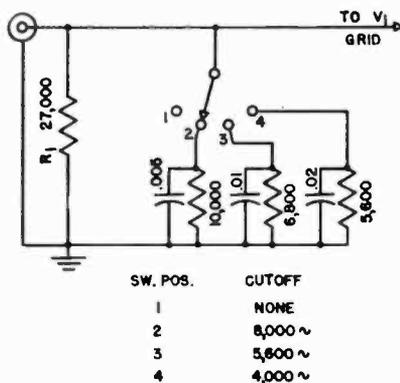


Fig. 2. Input circuit for use with low-level magnetic pickups when noise-suppressor section is not employed.

resistor and capacitor connected in series between the tap on the control and the low end of the control, usually ground. If a potentiometer is connected across the capacitor, the amount of compensation is reduced as the shunting resistance is decreased. This serves quite adequately for the low-frequency control. For smoothest operation, the shunting potentiometer should have an audio taper, and should be connected so that clockwise rotation of the control increases the resistance across the capacitor, thus increasing the bass response. With a 1.0-meg volume control, the recommended resistor and capacitor values are 27,000 ohms and 0.01  $\mu f$ . A 0.25-meg shunting potentiometer provides a smooth control of bass response.

In most amplifier designs, it is not considered desirable to utilize the feedback circuit for tone control purposes, since it reduces the amount of feedback available, and hence negates the beneficial effect of the feedback. This is particularly important in the case of a pentode amplifier, with feedback over the output and driver stages. However, with this amplifier the feedback is employed over only the driver and input stages, and its primary advantage is obtained over the low- and middle-frequency ranges because that is where the highest signal voltages are encountered. Therefore, with at least 20 db of feedback in use normally, it is felt that it will not affect the performance ad-

versely if 10 db of this feedback is used up in the high-frequency tone control circuit. The signal voltage at high frequencies is comparatively low, and the driver stage will not be required to furnish as much signal voltage as at the middle and low frequencies.

Therefore, if a capacitor is shunted across the cathode resistor to which the feedback circuit returns, it will reduce the feedback at high frequencies, and thus increase the output. A potentiometer in series will permit variation of the amount of treble boost. If another capacitor in series with a potentiometer is connected across the entire volume control, the high frequencies may be reduced at will. Since highs will not be boosted and cut simultaneously, the two potentiometers can be combined so that clockwise rotation will increase highs and counterclockwise rotation will decrease highs. To make this circuit perform smoothly, the taper on the control must be the reverse of the standard audio taper. With these two potentiometers, a resistor, and the three capacitors, quite satisfactory tone controls for both bass and treble are provided.

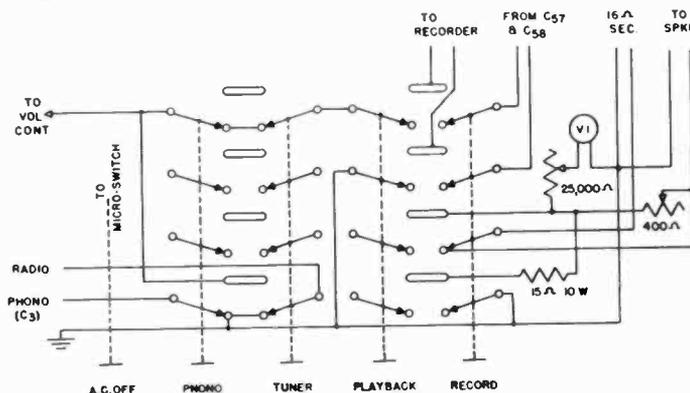
While some high-frequency cutoff is an advantage when reproducing phonograph records, no separate control is provided for this purpose since the dynamic noise suppressor section performs this function. It may be used simply as a controllable low-pass filter, with no dynamic action, or the suppressor control may be advanced so that the signals themselves control the opening of the gates. However, if the suppressor section is not included, it is suggested that a four-position switch be added, with the necessary resistors and capacitors, giving various cutoff frequencies. This is shown in Fig. 2, and was described in another article.<sup>1</sup> This control is not necessary if the noise suppressor section is included.

### Construction Features

Since individual constructors will rarely choose the same components, a complete

<sup>1</sup>High-Frequency Equalization for Magnetic Pickups, by C. G. McProud, starting on bottom of page 48.

Fig. 3. Wiring of push-button switch for selecting the use to which the amplifier is to be put.



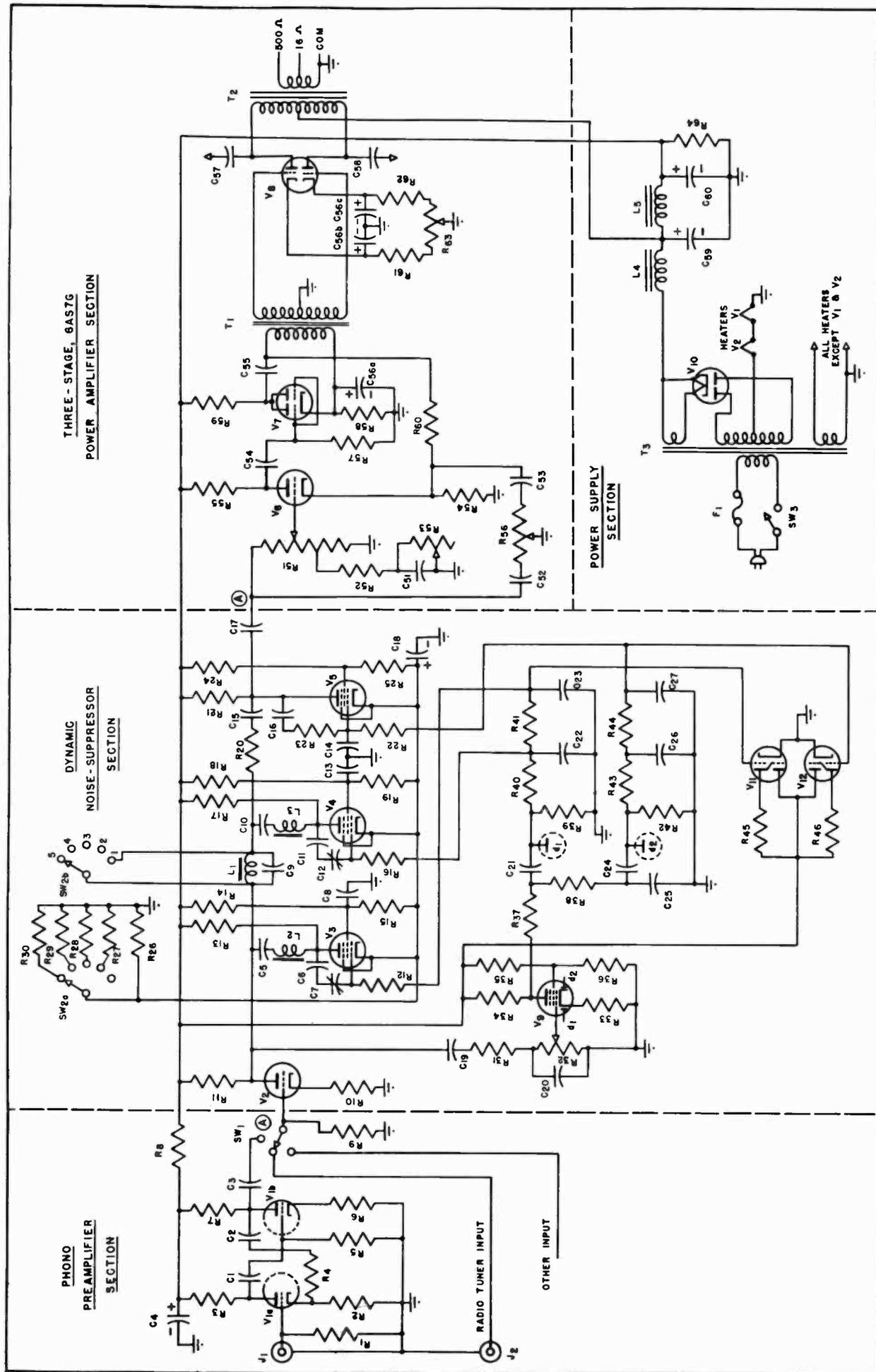
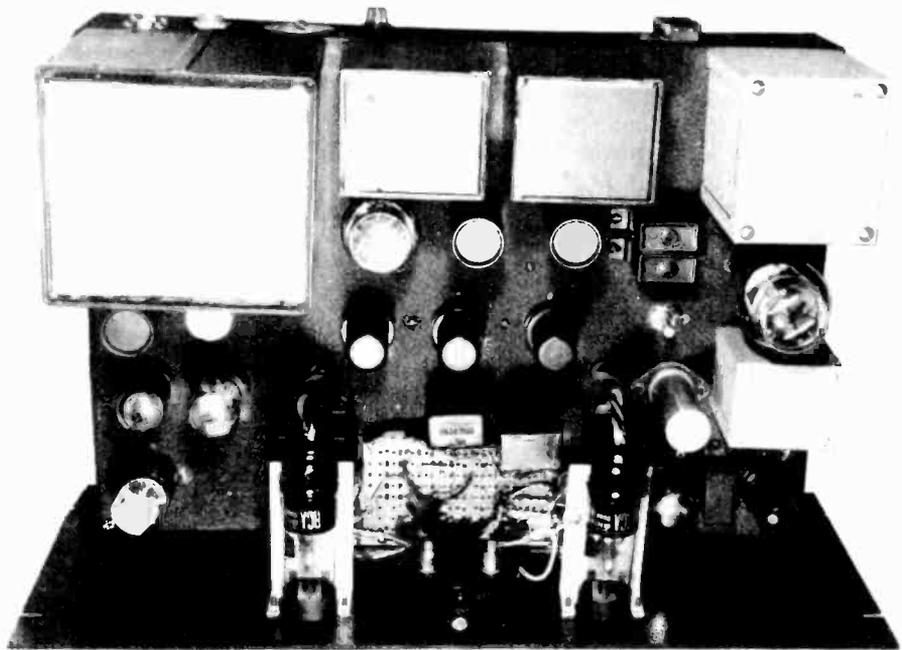


Fig. 5. Complete schematic for the sectionalized amplifier. By combining desired sections, practically any requirements may be met. V<sub>1</sub>—12SL7GT, V<sub>2</sub>—12J5GT, V<sub>3</sub>, V<sub>4</sub>, V<sub>7</sub>—6AB7, V<sub>8</sub>—7A4 (6J5), V<sub>9</sub>—6N7, V<sub>10</sub>—6AS7G, V<sub>11</sub>—5Y4G, V<sub>12</sub>, V<sub>13</sub>—6E5.

## PARTS LIST

- $C_1, C_{27}, C_{28}, C_{29}, C_{31}, C_{32}$ —.01  $\mu f$ , 600 v., paper.  
 $C_2, C_{21}$ —.003  $\mu f$ , mica.  
 $C_3$ —.04  $\mu f$ , 600 v., paper.  
 $C_4, C_{29}, C_{30}$ —40  $\mu f$ , 450 v., electrolytic.  
 $C_5, C_6, C_{10}, C_{11}, C_{14}, C_{16}$ —.006  $\mu f$ , mica.  
 $C_7, C_{17}$ —7-45  $\mu f$  adjustable, in parallel with 33  $\mu f$  fixed mica.  
 $C_8, C_{12}$ —.03  $\mu f$ , 600 v., paper.  
 $C_9$ —80  $\mu f$  (select to resonate  $L_1$  to 15,000 cps. approx).  
 $C_{15}$ —.035  $\mu f$ , 600 v., paper  
 $C_{17}, C_{18}$ —0.1  $\mu f$ , 600 v., paper.  
 $C_{19}$ —100  $\mu f$ , 50 v., electrolytic.  
 $C_{20}$ —.002  $\mu f$ , mica.  
 $C_{20}, C_{21}$ —100  $\mu f$ , mica.  
 $C_{22}$ —.001  $\mu f$ , mica.  $C_{25}$  500  $\mu f$ , mica.  
 $C_{26}$ —2.0  $\mu f$ , 600 v., oil filled.  
 $C_{26}$ —40-40-20/150-150-25 electrolytic.  
 $C_{27}, C_{28}$ —0.5  $\mu f$ , 600 v., oil filled (omit if not for recording).  
 $F_1$ —3 amp fuse in Littelfuse holder.  
 $J_1$ —Phono pickup input.  
 $J_2$ —Radio tuner input.  
 $L_1$ —1.5 h  
 $L_2, L_3$ —0.375 h  
 $L_4, L_5$ —10 h, 200-ma power supply choke.  
 $R_1, R_{21}$ —27,000 ohms,  $\frac{1}{2}$  w.  
 $R_2, R_7$ —2,200 ohms,  $\frac{1}{2}$  w.  
 $R_3, R_4, R_7$ —0.12 meg, 1 w.  
 $R_8, R_9, R_{23}, R_{20}, R_{21}, R_{22}, R_{23}, R_{24}, R_{25}$ —1.0 meg,  $\frac{1}{2}$  w.  
 $R_6$ —56,000 ohms, 1 w.  
 $R_{10}, R_{24}$ —2,700 ohms, 1 w.  
 $R_{11}$ —27,000 ohms, 1 w.  
 $R_{12}, R_{16}$ —0.1 meg,  $\frac{1}{2}$  w.  
 $R_{13}, R_{17}, R_{20}$ —0.22 meg, 1 w.  
 $R_{16}, R_{18}, R_{24}, R_{25}, R_{26}, R_{29}$ —0.1 meg, 1 w.  
 $R_{18}, R_{19}$ —22,000 ohms, 1 w.  
 $R_{27}$ —18,000 ohms, 1 w.  
 $R_{21}$ —0.47 meg, 1 w.  
 $R_{22}$ —0.33 meg,  $\frac{1}{2}$  w.  
 $R_{24}$ —10,000 ohms, 1 w.  
 $R_{24}, R_{25}$ —2,200 ohms, 1 w.  
 $R_{27}$ —470 ohms,  $\frac{1}{2}$  w.  
 $R_{28}$ —330 ohms,  $\frac{1}{2}$  w.  
 $R_{29}$ —220 ohms,  $\frac{1}{2}$  w.  
 $R_{30}$ —68 ohms,  $\frac{1}{2}$  w.  
 $R_{31}$ —0.27 meg,  $\frac{1}{2}$  w  
 $R_{32}$ —1.0 meg, IRC D13-137, suppressor control.  
 $R_{33}$ —560 ohms,  $\frac{1}{2}$  w.  
 $R_{34}$ —0.33 meg, 1 w.  
 $R_{35}$ —33,000 ohms, 1 w.  
 $R_{37}, R_{38}$ —0.22 meg,  $\frac{1}{2}$  w.  
 $R_{39}$ —3.9 meg,  $\frac{1}{2}$  w.  
 $R_{40}, R_{41}$ —1.0 meg,  $\frac{1}{2}$  w. (in Amphenol MEA-6 socket)  
 $R_{42}$ —1.0 meg, tapped for tone compensation, IRC D13-137X  
 $R_{43}$ —27,000 ohms,  $\frac{1}{2}$  w  
 $R_{44}$ —0.25 meg, bass tone control, IRC D13-130.  
 $R_{45}$ —1.0 meg, treble tone control, IRC D14-137.  
 $R_{46}$ —0.56 meg, 1 w.  
 $R_{47}, R_{48}$ —1,750 ohms, 10 watts, Ohmite Brown Devil.  
 $R_{49}$ —500 ohms, 4 watts, wire-wound control.  
 $R_{50}$ —30,000 ohms, 20 watts, Ohmite Brown Devil.  
 $SW_1$ —1 cct, 4 position, rotary switch, Centralab 1402 (or use push-button switch; see text).  
 $SW_2$ —2 cct, 5 position, rotary switch, Centralab 1404  
 $SW_3$ —SPST toggle switch (or use push-button with Micro-Switch).  
 $T_1$ —ADC 215C interstage transformer, turns ratio 1:3.  
 $T_2$ —ADC 315F output transformer, 3,000/-500, 16, 8, 4, etc.  
 $T_3$ —400-0-400, 200 ma 5 v. 3 a; 6.3 v. 6 a.



Top view of amplifier chassis.

layout of the chassis is not shown. The unit as built occupied a 12 x 17 x 3 chassis quite completely, and an 8  $\frac{3}{4}$ -in. rack panel was used for the front. The power transformer and filter chokes were surplus items, but the table at the end of this article lists a number of possible transformer and choke selections which should be obtainable from regular stocks. While the first 6AS7G amplifier was constructed with broadcast quality ADC transformers, this one uses the industrial line which is somewhat less costly. Excellent results were obtained, however, and unless the highest possible quality of all components is desired, it is suggested that these are quite satisfactory. All other transformers shown in the table were selected from specifications, but it is expected that the results obtained would be about in proportion to the cost of the components.

In the interests of simplicity of operation, a push-button switch was chosen for the selector switch. Again resorting to surplus stocks, a switch was located which consists essentially of four separate 4pdt switches, with a fifth push button bar which actuated no switches, only releasing all other buttons. This switch was mounted just above the chassis with the buttons extending through the panel. A normally-closed micro-switch was mounted under the chassis, with a spring actuator extending upward through a slot and so positioned that the push-button bar causes the switch to operate when the button is depressed. Thus one position of the switch is marked "A.C. OFF," and the amplifier is turned off whenever this button is depressed. When any other button is pressed, the "off" button is released, and the power is turned on. The other four push buttons are wired for radio tuner, phono,

playback, and recording. The wiring of the switch is shown in Fig. 3. One advantage of the push-button switch is that it permits connection to either radio or phonograph pickup as a source for recording simply by depressing two buttons simultaneously.

In order to maintain a proper recording level, it is desirable to incorporate a volume indicator in the amplifier. When used with a crystal cutter recording unit, the output transformer should be terminated with a resistive load and the cutter connected to the plates of the output tube through 0.5- $\mu f$  capacitors. However, the recording level required—of the order of 30 volts—furnishes too high a level to the monitor speaker, so the switching is arranged to connect a 16-ohm terminating resistor across the secondary of the output transformer, and insert a 400-ohm rheostat in series with the voice coil. Thus it is possible to adjust the speaker volume to a de-

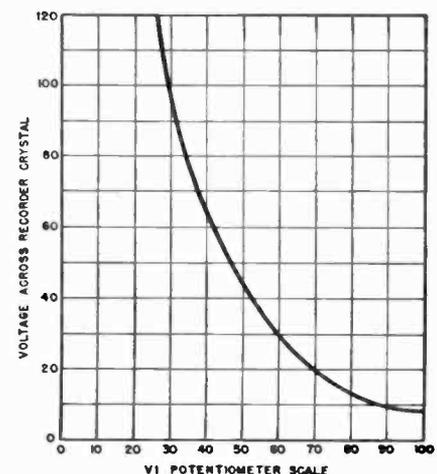


Fig. 4. Calibration of VI potentiometer for voltage across recorder head at zero indication

sirable level while maintaining the correct recording level to the cutter.

The volume indicator is a standard db meter, calibrated at 1.73 volts for zero indication. A 25,000-ohm potentiometer in series with the VI provides a control over the output level, with the calibration of the scale for this resistor being plotted on a curve, Fig. 4, so the desired output level may be obtained. The potentiometer used for this circuit is a grid-bias control, and while the curve is apparently reversed, it seems correct in use because the meter deflection is increased as the knob is turned clockwise.

From the photo of the amplifier, it will be noted that the panel is labeled quite profusely, giving a professional appearance. This is made possible by the new Tekni-Cals, which provide a wide range of identifying names. They are easy to apply, inexpensive, and of excellent appearance.

#### Adjustment of Suppressor

The circuit of the dynamic noise-suppressor amplifier is shown in the complete schematic, Fig. 5, and since the operation of this circuit has been described many times in the literature, no further mention of the principles underlying this section will be made here. It will be noted that it is practically identical to the Goodell amplifier, even to the physical layout of the schematic. In any discussion of this circuit, the coils have been described as rather critical, and of high Q. These will undoubtedly be difficult to obtain—one source of supply of 2.4 and 0.8 henry chokes used in one model of the suppressor being ADC, which supplies them under part numbers 414D and 414E respectively. One of the former and two of the latter would be required. Another source of suitable coils would be the UTC adjustable types, VI-C10 and CI-C12—again requiring two of the former and one of the latter. All of these types are very satisfactory for this unit. Another suitable coil, available in surplus at the time of construction, was found to have a measured inductance of approximately 1.5 H, and is center tapped, providing 0.375 H across either half. The half between terminals 1 and 2 has the higher Q, and should be employed for the inductances  $L_2$  and  $L_3$  while the entire coil is used for  $L_1$ . These are the coils used in the amplifier shown, and they work satisfactorily.

Once the complete amplifier is constructed, the alignment of the noise-suppressor section is not particularly difficult, but an a-f oscillator is essential. The value of the capacitor  $C_9$  is specified, and is determined so as to resonate  $L_1$  and  $C_7$  at 15,000 cps. Some noise-suppressor amplifiers employ a variable or adjustable capacitor for this circuit, but the adjustment is not critical, and a calculated value is adequate if the value

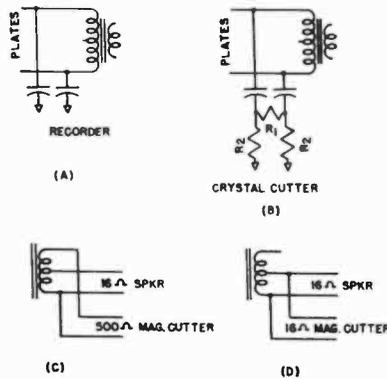


Fig. 7. Change in output wiring for feeding different types of recorders.

may be obtained with fair accuracy. With the switch  $SW_2$  on position 2, capacitors  $C_7$  and  $C_{12}$  are adjusted for minimum signal at 9,000 cps,  $R_{32}$ —the suppressor control—being at the minimum position so there is no opening of the gates by the signal itself. Then, turning  $SW_2$  to position 5, check the frequency of minimum output, which should be around 4,000 cps. Minor adjustments in the values of  $R_{15}$  and  $R_{19}$  may be employed to cause the circuits to “track” at 9,000 and 4,000 cps. Any change in the resistor values will necessitate returning of  $C_7$  and  $C_{12}$ , so the work is of the nature of a “cut and try” process, but no trouble was experienced in adjusting the first model, so it is assumed that the work may be duplicated by a careful constructor with the assurance that the final unit will work as it should.

The time constants for the rectifier circuits are quite satisfactory as shown. Longer release times may be obtained by an increase in the values of  $R_{41}$  or  $C_{23}$  for the high-frequency gates, or of  $R_{44}$  or  $C_{27}$  for the low-frequency gate. The value of  $R_{31}$  may have to be adjusted for the individual amplifier. This should be such that with about three-quarters rotation of  $R_{32}$  the gates open and close with the applied signal. The average signal at the arm of  $SW_1$  will normally be around 1.5 volts, whether from a tuner or from a phonograph pickup. This will give adequate signal level to cause the side amplifier to operate with the resistor value given in the parts list. It may be stated safely that if the circuit values are followed accurately, the amplifier should perform in the normal manner for a noise suppressor.

#### Construction Hints

As with any high-gain amplifier, it is necessary to exercise normal care in shielding grid and plate circuits, especially where there are any long runs. This does affect the frequency response if carried to extremes, and minor compensation may be effected by the addition of a small capacitor across  $R_{11}$ . It is at this point that the high-frequency losses may be corrected if found necessary. However, with the parts layout shown in the photo, such compensation will probably not be necessary.

Because of the large number of wires in a circuit of this type, it is desirable to cable the wiring. This necessitates either of two procedures—a complete full-size wiring diagram may be made first and the individual wires laid in place using a forming board with finishing nails driven in at points where the cable makes a bend or where wires branch off. After all the wires are in place, the cable may be laced up. The other method appears to be simpler in that it does not require the full-size wiring diagram, but once the wiring is in place it is often difficult to form it into cables smoothly. In the long run it is easier to cable the wires outside of the chassis. After the lacing is completed, the form is put into place and the wires cut off at suitable lengths for connection to the sockets and other components. The form is then removed, the insulation stripped back, and the wires tinned. Then the form is replaced and connections made and soldered, a very rapid process after the initial cable forming is completed.

Measurements made on the complete amplifier indicate an output of 5 watts at 7 per cent intermodulation distortion, with frequency response curves as shown in Fig. 6. The upper curves show the effect of the high- and low-frequency tone controls with the volume control at one-third rotation and the suppressor range switch,  $SW_2$ , on position 1. The lower curves show the response for positions 2, 3, 4, and 5 of the range switch and the suppressor control,  $R_{32}$ , off. Note that the amount of suppression increases as the range is narrowed, which is a desirable condition since the worst records necessitate a narrower transmission band as well as greater suppression outside the band.

#### POSSIBLE TRANSFORMER AND CHOKE SUBSTITUTIONS

Mfgr	T <sub>1</sub>	T <sub>2</sub>	T <sub>3</sub>	L <sub>1</sub> , L <sub>2</sub>
ADC	215C*	315F*	515D	415D
ADC	214H†	314C†		
Chicago	BI-8	BO-9	PCC-200	RC-12200
Freed		F-1951	F-424A	F-633
Peerless	G-212-Q	S-240-Q	R-560-A	C-390-A
Siancor	WF-28	HF-65	P-6165	C-1646
Thordarson	T-20A23	T-22S72	T-22R34	T-20C55
Triad	HS-25	HS-84	HS-215	HS-315
UTC	LS-21	LS-55	LS-70	LS-92

† original specification for 6AS7G amplifier

\* as employed

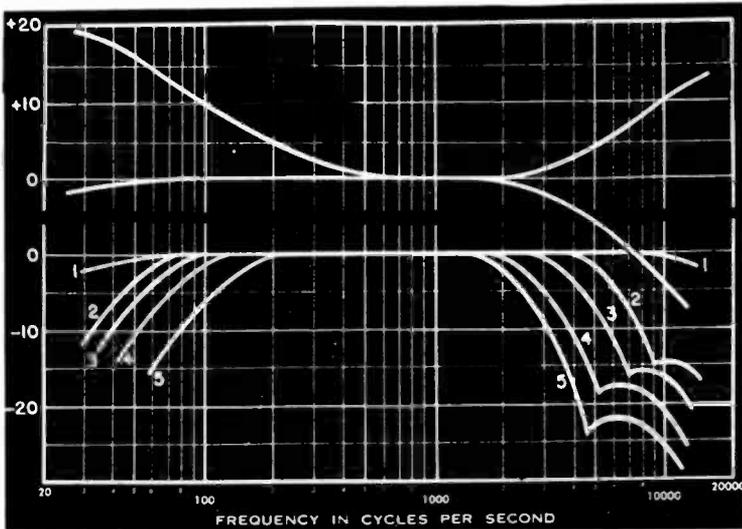


Fig. 6. Frequency response curves (upper) effect of tone controls; (lower) effect of noise suppressor in various positions of range switch.

### Adaptation of Sections

Since this amplifier is laid out so as to be flexible in construction, it may be well to indicate the various arrangements possible. The simplest arrangement consists of the preamplifier and the output section, which simply omits the center portion of the circuit, connection being made between points "A." This provides sufficient amplification for use with a low-level magnetic pickup and

furnishes the tone controls desired by most users. The high-frequency cutoff for the pickup should be added in the form of the circuit of Fig. 2 across  $R_1$ .

When desired for use with an ordinary crystal cutter, the wiring of the output circuit and the feed for the recorder should be modified as shown at (B) of Fig. 7. (C) and (D) show connections for 500-ohm and 16-ohm magnetic cutters respectively. The wiring of the complete output circuit for feeding the re-

recorder is shown in Fig. 3, complete with VI and speaker circuit wiring.

If the previous 6AS7G amplifier has already been built, the first two sections may be connected to the output section with results similar to those obtained with the complete single-unit amplifier.

### Conclusion

Although the parts list specifies the transformers and chokes employed in the amplifier as built, some substitutions may be desirable, depending upon availability of parts. Therefore, the table shows a number of components which should perform similarly.

Assuming the use of good components throughout, this amplifier should give completely satisfactory performance for the record enthusiast or for the home recorder. The output power is not sufficient for use with professional disc cutters, but reasonably satisfactory results may be obtained with the better quality of either magnetic or crystal cutters, assuming that the correct coupling circuits are used between the transformer and the magnetic types of cutters. For ordinary reproduction of records or of radio programs, this amplifier will be found to answer most of the requirements of critical listeners.

# Tubeless Hi-Fi Tuner

High-fidelity addicts will remember the Western Electric 10A Radio Receiver as, for its time, one of the best from the standpoint of quality but it was not commercially available to the home user. The circuit of this tuner was simplified and appeared on the market as a wide range unit, employing a four-gang capacitor and a total of eight coils (two of them untuned) and three or four tubes. Having a wide pass band, it was useful only in close proximity to radio stations, and the sensitivity was not very great for this reason. However, the audio quality was excellent, and there are many of these wide-range tuners still in use.

Utilizing the same circuit principles, a new tuner has recently appeared which serves practically the same purpose. It is broad, and thus suffers from no side-band cutting; it has low sensitivity, which is desirable for tuners of this type; and it has remarkable audio quality. The circuit of the tuner is shown in Fig. 1, and it will be seen to consist of two antenna coils essentially back-to-back, with a two-gang

tuning capacitor and a negative-mutual coupling coil (EL-56 on the schematic). The 1N34 Germanium diode serves as the detector, with the signal being developed across the resistor.

This tuner is designed for use in metropolitan areas where there are likely to be a number of radio stations within a radius of 20 to 25 miles, and when used with a good antenna from 75 to 100 feet in length will give an audio output ranging from .05 to 0.5 volts. Even with a shorter antenna, satisfactory results are obtained with an output of less than .001 volts, pro-

viding the signal is fed into a high-impedance microphone input channel of a high-quality amplifier.

The coils specified for this tuner are the products of J. W. Miller Co. of Los Angeles, and while a complete kit is available—consisting of the coils, tuning capacitor, slide-rule dial, and chassis—it is possible to assemble the tuner with any desired chassis and capacitor, provided it covers the tuning range. For satisfactory results, it is necessary that high-Q coils be employed, and this requirement is fulfilled by the 242-A coils specified.

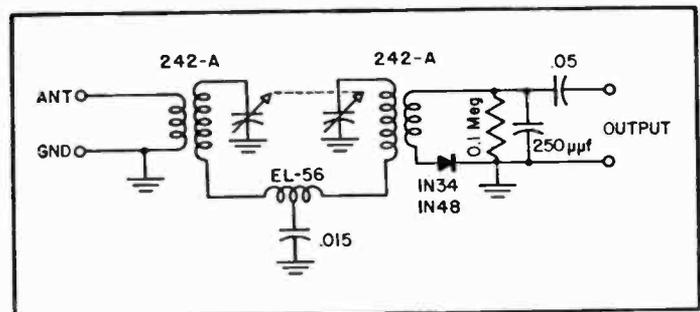


Fig. 1.

# Compact 6AS7G Amplifier for Residence Audio Systems

C. G. McPROUD

A new amplifier of exceptional performance especially designed for modernization where cabinet space is limited.

**M**ANY AN EXPERIMENTER or audio hobbyist has the desire—and often a definite need—for a high-fidelity amplifier, but is at a loss for sufficient space to install it in an existing cabinet or piece of furniture. So far, many of the writer's amplifier designs have been adequate for good quality reproduction, but none was arranged specifically for use by anyone desirous of modernizing a reproducing system because they were all laid out with a view to accommodating the components in a normal amplifier arrangement.

To solve any problem, it is first necessary to recognize its existence—the rest follows naturally. For a modernization problem, the requirements may be stated as follows:

*Electrical:* Around 5 to 6 watts of high-quality audio power.

Switching to select standard and microgroove phonograph pickups, and two additional positions for AM and FM radio inputs.

Sufficient gain and low-frequency equalization for low-level magnetic pickups.

Separate high- and low-frequency tone controls.

To these may be added as desirable features a volume control compensated for loudness levels, and means for equalizing the levels of the various inputs so the compensated volume control works at its optimum position and to avoid undesirable level changes when switching between inputs.

*Physical:* Amplifier and power supply small enough to fit into reasonable spaces.

Control facilities which may be mounted on a small panel space separate from the amplifier.

Considering these requirements separately, the first is fairly obvious. The reason for modernizing is to obtain a better quality of reproduction. This demands good components, and sufficient power to handle peaks without danger of overload. Since it is more economical and usually provides better overall quality to use a high-quality loudspeaker with a good reproducing

system, a fairly high efficiency is generally encountered. Most high-quality speakers will provide plenty of volume for home use with much less than one watt of *average* power although more is necessary, of course, to handle the peak levels. Therefore, it is felt that five watts should suffice for practically any home system. It goes without saying that frequency response should cover the range from 30 to 15,000 cps, and that distortion must be held to an absolute minimum. The hum level should be so low that no sound is audible from the speaker in the absence of signal.

#### Multiple Inputs

Practically every reproducing system is used for more than one input. Since the advent of long-playing, microgroove records, it seems logical to include an input for a second pickup, with a single selector switch connecting the chosen input source to the amplifier.

Low-level magnetic pickups are firmly established, and any good amplifier must be designed to accommodate them without the need for an external preamplifier. As is well known, these pickups require equalization of the low-frequency spectrum, in addition to considerable gain to make their output comparable to that of a radio tuner. The microgroove pickups are slightly lower in output in most instances, due largely to a lower level on the record itself.

Although not generally known, a conventional crystal pickup can be fed into an equalized preamplifier, and will often sound better than if used with a high-impedance input. As far as the frequency response is concerned, this is easily explained. A crystal pickup may be regarded as a generator of zero impedance in series with a capacitance. An average crystal pickup, for example, has a

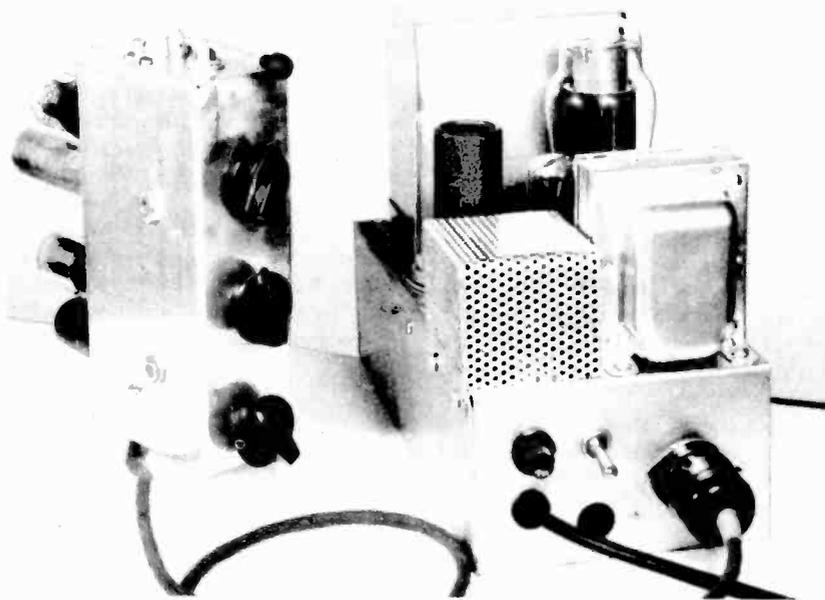


Fig. 1. Compact, two section amplifier with control section arranged for mounting in small panel area.

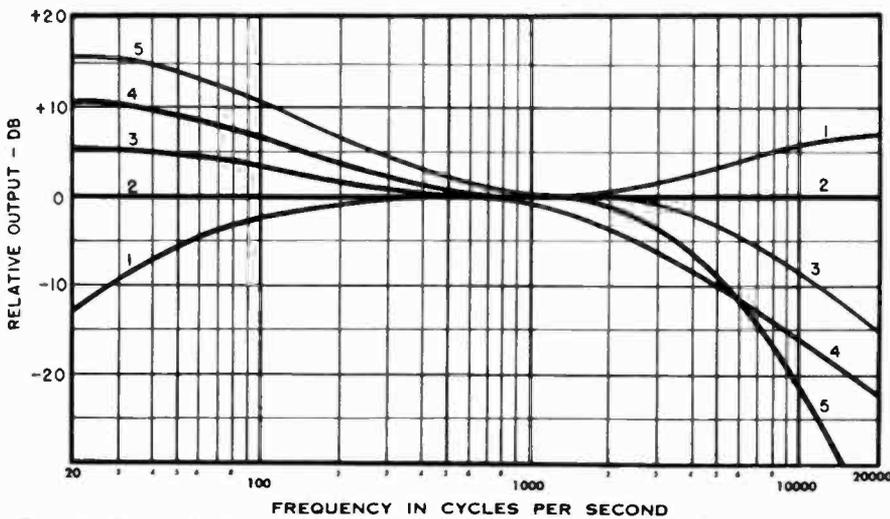


Fig. 2. Tapped high-and low-frequency tone control switches provide fixed response curves.

capacitance of around 1500  $\mu\text{mf}$ . When such a pickup is fed into a resistive load, it has a natural droop of 6 db/octave below the frequency at which its reactance equals the value of the resistance into which it feeds. Thus, it has a "turnover" frequency of 500 cps when fed into a 0.2-meg load. Now, while a crystal pickup is a constant-amplitude device and delivers a constant voltage into a high resistance load up to the turnover frequency (of the record) from a disc cut with the normal 6 db/octave droop below the turnover, the low-resistance load causes a loss in bass response equivalent to that of a magnetic pickup. But the preamplifier corrects for this loss, so the output is again "flat." The voltage output of the crystal is higher than that of the magnetic pickup, so the loss due to the low-resistance load may be accepted readily. Since high-frequency equalization is still necessary for the crystal pickup, it is still necessary to add a resistor shunted by a capacitor in series with the high side of the pickup to make it workable with a high-gain preamplifier, if wide-range reproduction is to be obtained. Therefore, this type of preamplifier is reasonably suitable for crystal pickups.

Most users want some tone controls so as to be able to obtain desired response curves. While the compensated volume control reduces the need to a large extent, satisfactory reproduction of phonograph records demands some roll-off control, and also a sharper cut-off for particularly noisy records. Varying degrees of bass boost are also desirable. Therefore, both low- and high-frequency tone controls are employed, providing five curves for each as shown in Fig. 2. These are step controls

rather than continuously variable potentiometers because more suitable curves are obtainable. The low-frequency control provides a 5-db boost at 50 cps, a flat position, and boosts of 4.5, 9, and 13 db at 50 cps. The high-frequency control provides a boost of 6 db at 10,000 cps; a flat position; a roll-off down 3 db at 4400 cps and 8 at 10,000; an NAB roll-off down 3 db at 1600 cps and 16 at 10,000; and a cut-off down 5 db at 3500 cps and 21 at 10,000. Listening tests have adjudged these steps to be desirable.

The particular type of fully compensated volume control used is that

described in the article on page 39. The level adjusting is accomplished by means of three 0.5-meg potentiometers, with the microgroove pickup having no built-in adjustment, since the overall amplifier gain is designed to fit this input. If further adjustment is required, it may be accomplished externally.

The amplifier, shown complete in Fig. 1, is built in two sections—one is the output stage and the power supply, while the other is the control unit, with all the other stages. The power section is built on a standard 5 x 10 x 3 chassis, while the control section is housed in a 3½ x 7 x 2 chassis, made from 7 x 11 x 2 standard aluminum chassis. The two sections are connected by a three-foot cable which carries all power and signal circuits except for the a-c switch line, which is separate. The a-c switch is not a part of the control section, but is to be mounted at a convenient location on the panel.

### Circuit Description

In general, amplifier design progresses backward, first involving the selection of the output stage, then adding the earlier stages to provide sufficient gain to drive the output tubes. Because the 6AS7G has so many advantages as a power output tube, it was chosen again for this application, in spite of the fact that it is relatively hard to drive.

It may also be said that the power

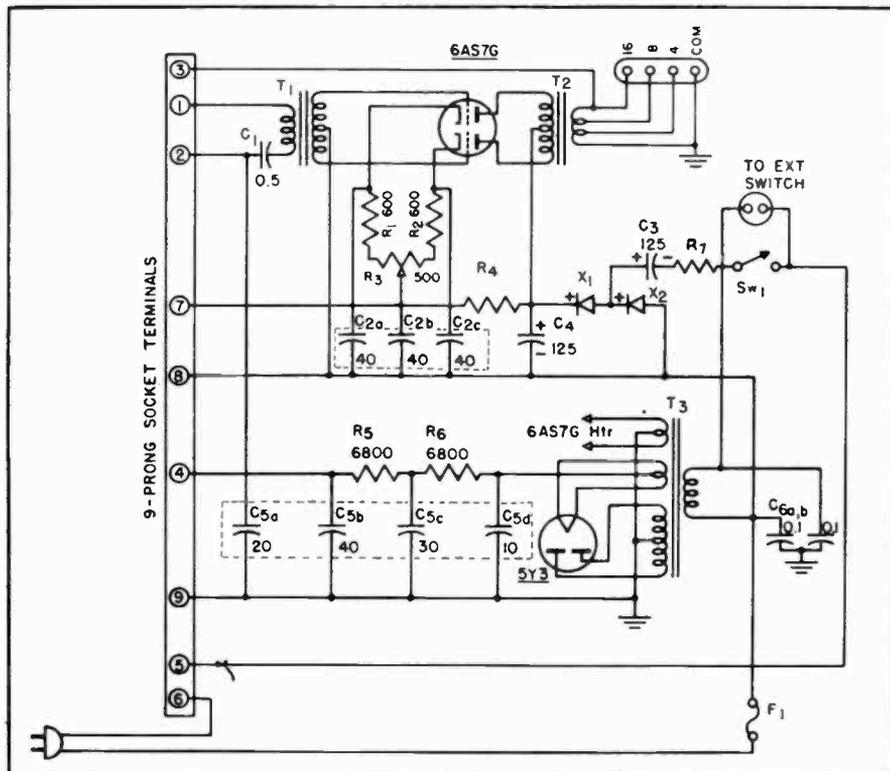


Fig. 3. Schematic of power amplifier section of the two-unit amplifier.

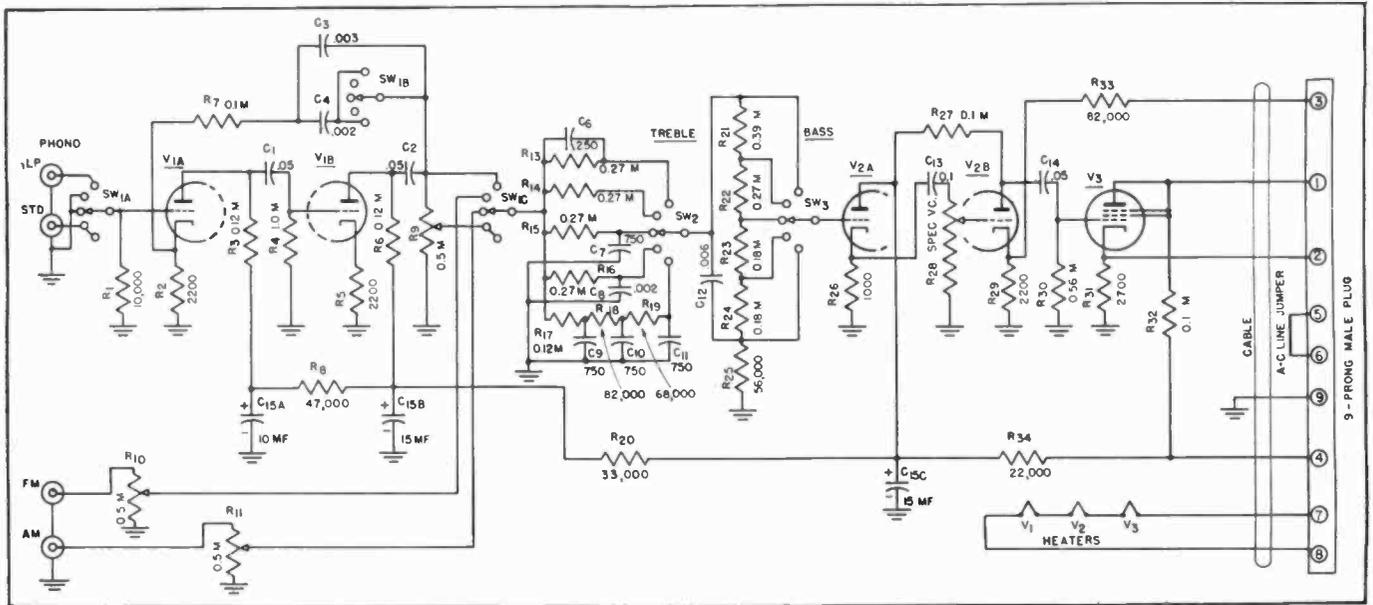


Fig. 5. Schematic of the control section.

supply requirements for this tube are fairly severe since it draws a rather heavy plate current. Normally, this necessitates a large power transformer and one or more large filter chokes. However, one of the requirements of the tube may be considered an advantage—because it needs an input transformer, the output stage can be completely isolated from chassis ground, thereby permitting the use of a voltage-doubling selenium rectifier circuit, as shown in the power section schematic, Fig. 3. This arrangement furnishes up to 150 ma at approximately 300 volts from a 117-volt a-c line. The 6AS7G draws about 120 ma, and an additional 30 ma is fed through a bleeder for heater current of the three input tubes. Thus the low-level heaters are energized by rectified alternating current. Considering the use of a push-pull output stage, the capacitor provides sufficient filtering for humless reproduction.

The input stages require a d-c supply which may be grounded to the chassis, and in addition, the 6AS7G heater must be energized. Thus a small power transformer is used with a conventional rectifier and an RC filter circuit. The 6.3-volt filament winding on the transformer is used only for the 6AS7G, since the other tubes have a d-c heater supply.

The output transformer is massive and occupies a large portion of the chassis area. The space underneath the output transformer is occupied by the two 125- $\mu$ f capacitors on a bracket; the channel-type, push-pull input transformer is also under the chassis, as are the coupling and line bypass capacitors. The selenium

rectifiers are mounted on Bakelite strips above the chassis, and are protected by a perforated screen cover. The capacitors in the voltage doubler power supply—the two 125- $\mu$ f units and the triple 40- $\mu$ f unit used for cathode bypass of the 6AS7G—are insulated by cardboard tubes. All connections except the output are made on one end of the chassis: a 9-prong socket for the control section feed; the a-c line cord, fuse, and switch; and a small two-prong socket for a remote power switch which is in parallel with the chassis switch. This permits a pair to be run up to a panel-mounted power switch, thus eliminating any a-c circuits from the interconnecting cable.

#### Control Amplifier

The control amplifier is of unique design, since three controls are mounted on the front apron of the

chassis, and one on the end, and with the tubes on the rear apron projecting away from the panel. The controls are arranged so the unit may be mounted either vertically or horizontally, thus being adaptable to almost any cabinet space available. The selector switch is on the end of the chassis, with an operating lever which extends through the panel. Also mounted on the end are the two pickup jacks and the level-adjusting potentiometer for the standard pickup. The decoupling capacitor and the two radio input jacks are mounted on the rear apron along with the tubes, while the radio level-adjusting potentiometers are on the "top" of the chassis. The power cable comes out of the end opposite the selector switch.

To simplify wiring into the circuit, both tone controls are assembled completely on their switches as shown in Fig. 4. All resistors connecting to

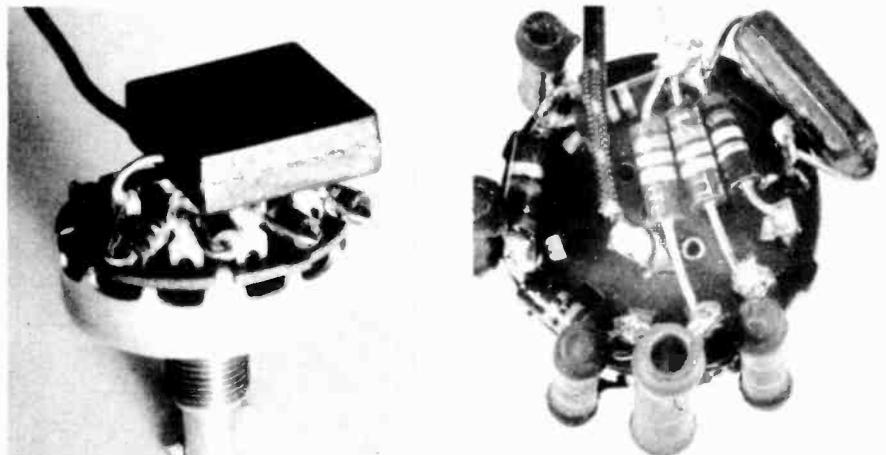


Fig. 4. Assembly of tone controls on standard switches to provide units easily wired into the control section. Left—bass control; right—treble control.



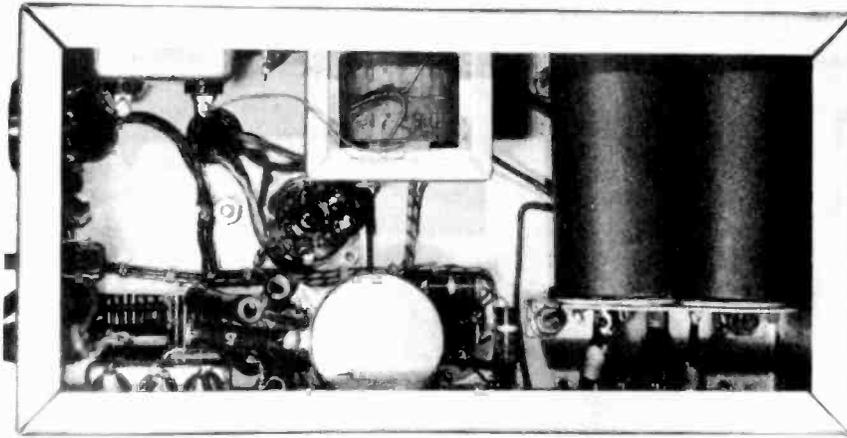


Fig. 7. Underside of power section chassis.

trols in the positions for flat response. However, this did not provide sufficient gain, and the components used for this purpose were later omitted. The elimination of this feature in no way hampers the operation of the amplifier or its flexibility of frequency-response correction, due to the presence of the high-frequency control, SW<sub>2</sub>. At position 4 of this switch, the circuits are so arranged that the correct roll-off is provided for the reproduction of LP records. Actually this makes the amplifier more flexible because any high-frequency correction can thus be used, making it possible to play any type of record.

The performance of any amplifier depends to a large degree upon the quality of the components used. At the time this amplifier was designed, it was desired to use high-quality transformers throughout, and to make the power section as compact as possible the input transformer had to be mounted underneath the chassis. This ruled against a cased type, and few manufacturers list high-quality transformers in open-frame mounting. The unit employed was designed to work between a single 12SJ7, triode connected, with no d.c. in the primary, and to obtain adequate driving voltage for the grids of the 6AS7G it was required that the step-up ratio should be fairly high. To get this performance and retain a wide frequency range, the transformer is wound in two sections, both placed on the center leg of a conventional E-I core. Measured frequency response of the transformer itself indicates a droop of 1 db at 30 cps and at 30,000 cps. The output transformer, also special, shows a droop of 1 db at 17 cps and at 120,000 cps when operated without feedback, and drooping 1 db at 15 cps and 62,000 cps when operated with feedback. Similar output transformers are available in the standard Freed line under number F-1951 with output impedances of 1.2 to 30 ohms, and under

number F-1950 for impedances from 50 to 500 ohms.

The following table indicates transformers of high quality which are generally obtainable from jobber stocks and which should perform satisfactorily, since their characteristics are similar to those used in the original amplifier.

TABLE I			
Mfgr	T <sub>1</sub>	T <sub>2</sub>	T <sub>3</sub>
Audio Development Co.	214H <sup>1</sup> 215C <sup>1</sup>	314C 315F	515C (large)
Chicago Transformer Div.		BO-6	PCC-70
Peerless Transformer Div.	G-252-Q	S-240-Q	R-196-A
Stancor	A-4750	A-3800	P-4078
Thordarson	T20A22	T22S70	T22R02
UTC	LS-21 <sup>1</sup> CG-132 <sup>1</sup>	LS-57 <sup>2</sup> LS-55 <sup>3</sup> CG-16	R-54

<sup>1</sup> Insufficient space under chassis to mount these models.  
<sup>2</sup> Voice-coil secondary only.  
<sup>3</sup> Voice-coil and line secondary

#### Construction Hints

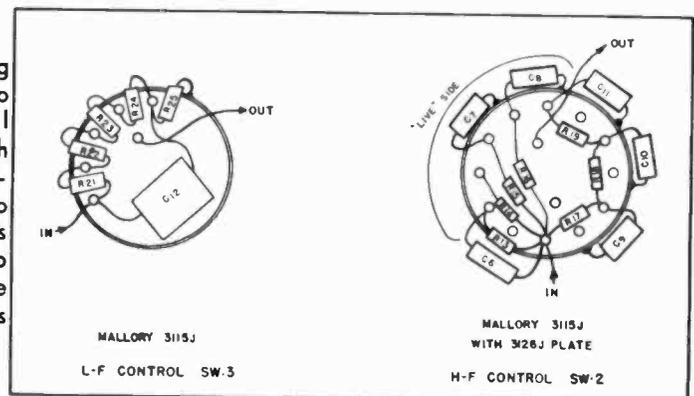
There are a number of suggestions which may be of interest in the construction of these two units. Referring first to the power section, it will be noted that the two filter capacitors, C<sub>3</sub> and C<sub>4</sub>, are mounted on a bracket under the output transformer. Therefore, they are necessarily installed after the wiring to the output transformer is completed. The balancing potentiometer, R<sub>3</sub>, is also mounted

on a bracket so that its shaft may be adjusted through a hole on the chassis between the two capacitors mounted on top. The selenium rectifiers are mounted on a 6-32 threaded rod which passes through two strips of Bakelite attached to bent-up angles on the chassis. Two saw-cuts are made ¼ in. apart and extending for two inches along the chassis. At the center of these two cuts, another cut is made between them. This frees two "flaps" which may be bent up to mount the Bakelite strips. The cover is bent up from perforated metal to prevent accidental contact with hands or tools.

The value for the resistor R<sub>4</sub> is shown as 15,000 ohms. This gives approximately correct current through the filament string, but the current should be measured, and the value of R<sub>4</sub> adjusted to give 36 volts across the heaters. Resistor R<sub>7</sub> is used to reduce the peak current through the selenium rectifiers, and 5 ohms is a suitable value. This resistor is connected between SW<sub>1</sub> and C<sub>3</sub>.

The push-pull input transformer is mounted directly below the 6AS7G socket, using leads as short as possible. This will normally require that the leads be connected before the transformer is bolted in place, and since the leads are likely to be of relatively light wire, care should be exercised in this operation. The two bathtub capacitors are mounted on opposite sides of the chassis, with tie-points installed adjacent to them. One is needed

Fig. 8. Wiring diagram of two tone-control switches. Switch plate on H-F control is changed to furnish tie points on five-step frame. Only one contact arm is used.

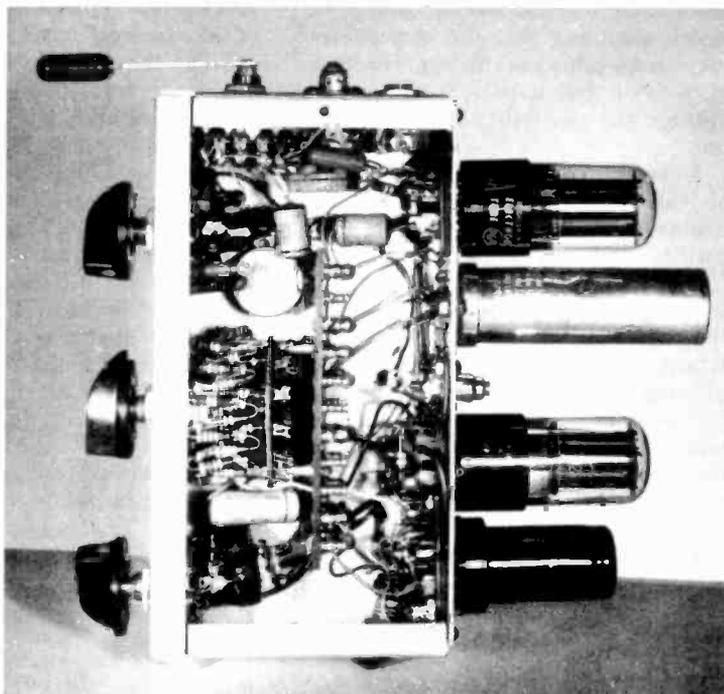


to make the connection to the plate end of the primary, and the other serves to hold  $R_7$ .

With some transformers it is probable that there will be a tendency to oscillate at some super-audible frequency. This may require some experimentation, but it is suggested that a small capacitor across each half of the primary of the output transformer, or possibly across the secondary of the input transformer, will suffice to eliminate this trouble. Probable values will be in the vicinity of .002  $\mu\text{f}$ . *Figure 6* is a partial wiring diagram of parts employed in the power section, while *Fig. 7* is a photograph of the underside of the completed amplifier chassis.

The preamplifier section shown in the photograph of Part 1 is somewhat smaller than the average constructor may wish to employ, but it was built in this manner to keep the space required to a minimum. The chassis was fabricated from a 2 x 7 x 11 aluminum chassis base, cutting it apart at the center. The sides of the chassis are then cut  $1\frac{1}{4}$  in. from the open end, and folded in to provide a  $\frac{3}{8}$ -in. angle. The top is folded down, resulting in a chassis approximately  $3\frac{1}{2}$  x 7 x 2. The tube sockets and the electrolytic capacitor are mounted on this section, with the controls on the opposite side of the chassis. The input selector switch is mounted on one end, with a lever extending through the panel for its operation. If the chassis length were extended to eight or nine inches, the selector switch could then be mounted on the front in line with the other controls, and it is quite probable that it would be easier to install in a cabinet because of the difficulty in cutting a neat slot

Fig. 10. Interior of input section, showing the mounting of resistor strip on rear of compensated volume control.



to pass the lever-type arm used to actuate the selector switch.

In the parts list several references were made to the text for further explanation. The volume control,  $R_{28}$ , is the loudness control described in detail on page 39, and consists of a Centralab 1443 switch on which are mounted the resistors and capacitors necessary to obtain the desired compensation.

The high-frequency tone control switch,  $SW_2$ , needs a little further description. The series of switches selected for the tone controls consists of small units which are desirable in such a compact amplifier. However, the exact assembly of contacts is not obtainable, so the switch used was made by using the frame and mechanism

from a 3115J switch with the contact plate from a 3126J switch. This gives a number of tie points for the resistors used for the cut-off circuit, as well as for the input connection. The capacitors in this circuit are all mounted directly on the switch, being soldered to the frame for ground connection. This is not usually considered the best practice, but it must be remembered that this section of the amplifier does not have any a-c circuits in it, and there is little chance of ground loops causing hum trouble. Suffice that the unit as constructed exhibits no troubles from this source. The exact arrangement of the tone controls is shown in *Fig. 8*. Similar methods were used for both, in that all parts are wired directly to the switch, but the low-frequency control is a standard 3115J switch without modification.

The over-all schematic of the input section is self-explanatory and represents the final form of the circuit after all the performance data were determined. The wiring diagram, *Fig. 9*, indicates the arrangement of parts and the wiring between them, while the photograph of *Fig. 10* shows the method of mounting the resistor strip on the back of the volume control switch.

Unless the builder is reasonably well experienced in construction of small amplifier equipment, it might be desirable to increase the overall size of the input section. It is definitely possible to construct the unit in the size shown, but it must be admitted that it is extremely compact, and as the size is reduced the complexity of construction is increased. Of course, once the amplifier is completed

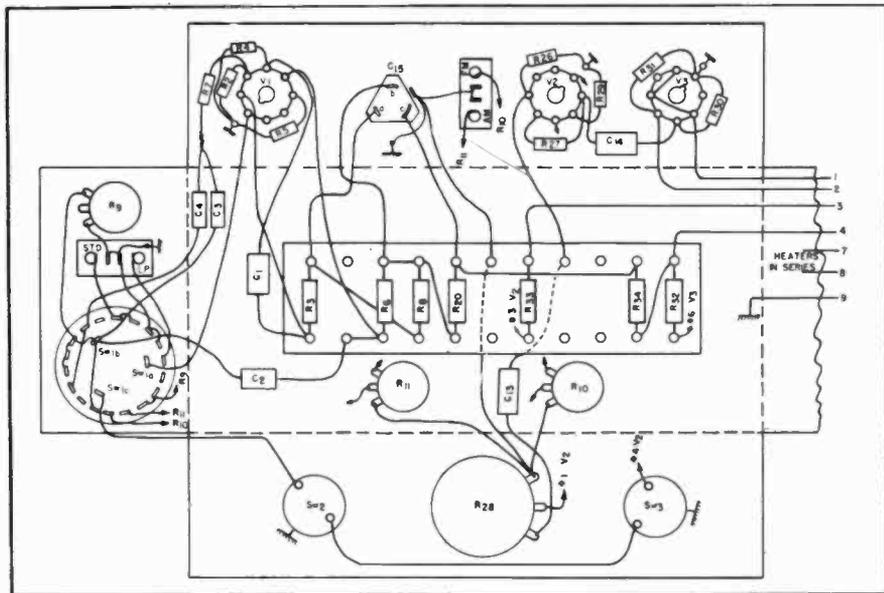


Fig. 9. Input section wiring diagram.

there should be no need to get at it again, assuming that the constructor takes reasonable care in the selection of resistor and capacitor ratings to prevent the possibility of failure in use.

A more conventional construction of this entire amplifier would undoubtedly reduce the hum level still further, and if a larger power transformer were used—one which could supply the 0.9 amps of filament current required by three 6-volt tubes in the input section—the electrostatic field existing between heater and cathode of the present first 12SL7 would be reduced. If made in two units, however, it would be preferable to use a separate cable from the power section to the preamplifier to carry the heater current, although the signal from the

preamplifier to the power section is of a relatively high level and it is possible that no trouble would be encountered from this source.

If additional gain is required, the cathode follower section of  $V_2$  can be changed to a conventional amplifier, thus giving approximately 32 db more gain than with the present arrangement. For the uses for which this amplifier was designed, however, this should not be necessary. Another possibility is that a 6SN7 could be substituted for the 6SL7 used as  $V_2$ , (if a filament transformer were being used, together with 6-volt tubes) again using the amplifier connection rather than the cathode follower, and the increased gain would be of the order of 10 db. This suggestion would only apply if the filaments were arranged

to be supplied from a transformer winding, since the 12SN7 will not operate in a series string with the 12SL7 and the 12SJ7, inasmuch as the 12SN7 draws a filament current of 0.3 amps. These are design modifications, and some ingenuity on the part of the constructor will be necessary to arrive at the exact desired result. It is felt, however, that the amplifier as described performs satisfactorily, and that no changes are necessary for the purpose for which it was designed. The amplifier was designed for one application—that of modernizing an existing installation, or for providing a control arrangement which could be adapted to cramped quarters with the greatest of ease. This requirement is fulfilled adequately by the arrangement shown.

## PARTS LIST

$C_1, C_2, C_{14}$	0.05 $\mu$ f, 400 v, paper	$R_{19}$	68,000	$C_2$	40-40-40/150, electrolytic, with insulating tube
$C_3$	.003 $\mu$ f, mica	$R_{20}$	33,000	$C_3, C_4$	125 $\mu$ f, 350 v, electrolytic, with insulating tube
$C_4, C_8$	.002 $\mu$ f, mica	$R_{21}$	0.39 meg	$C_5$	40-30-20-10/450, electrolytic
$C_6$	250 $\mu$ f, Centralab Hi-Kaps	$R_{23}, R_{24}$	0.18 meg	$C_6$	0.1-0.1, 600 v, oil filled, bathtub type
$C_7, C_9, C_{10}, C_{11}$	750 $\mu$ f, Centralab Hi-Kaps	$R_{25}$	56,000	$R_1, R_2$	600, 5-watt
$C_{12}$	.006 $\mu$ f, mica	$R_{26}$	1000	$R_3$	500-ohm wire-wound potentiometer
$C_{13}$	0.1 $\mu$ f, 400 v, paper	$R_{27}, R_{32}$	0.1 meg, 1-watt	$R_4$	15,000, 10-watt adjust to provide 36 volts across filament string.
$C_{15a}, b, c$	15-15-10/450 electrolytic	$R_{28}$	special volume control (see text)	$R_5, R_6$	6800, 2-watt
$R_1$	10,000 (all values $\frac{1}{2}$ -watt unless otherwise specified)	$R_{30}$	0.56 meg	$R_7$	5, 5-watt
$R_2, R_5, R_{29}$	2200	$R_{31}$	2700	$SW_1$	SPST toggle switch
$R_3, R_6$	0.12 meg, 1-watt	$R_{33}$	82,000, 1-watt	$T_1$	push-pull input, special channel mounting, no d.c. in primary; Freed 17290
$R_4$	1.0 meg	$R_{34}$	22,000, 1-watt	$T_2$	4000 ohms plate to plate, 4-8 $\times$ 6-ohm secondary; Freed 15929
$R_7$	0.1 meg	$SW_1$	Mallory 3136J	$T_3$	325-0-325 v at 50 ma; 5v at 2a; 6.3v at 2.5 a. Freed F-413
$R_8$	47,000, 1-watt	$SW_2$	Mallory 3115J, modified (see text)	$X_1, X_2$	200 ma selenium rectifier
$R_9, R_{10}, R_{11}$	0.5-meg potentiometer	$SW_3$	Mallory 3115J		
$R_{13}, R_{14}, R_{15}$		$V_1, V_2$	12SL7		
$R_{16}, R_{22}$	0.27 meg	$V_3$	12SJ7		
$R_{17}$	0.12 meg				
$R_{18}$	82,000				

The parts not specifically described for the power section are as follows:

# FM Tuning Indicator

L. B. KEIM

## Features of a new indicator tube for FM receivers.

AS the FM band continues to gain in popularity and widens in the offering of its program coverage, an increased use of these static-free facilities will be made not only by the listening public, but also for recording studios, transcriptions of programs and tape recordings for future use. Such services require that no distortion at all be present in the signal source, consequently an accu-

rately tuned receiver is the answer.

Appearance on the market of such precise tuners as the Browning RV-10 FM tuner, or RJ-12A FM/AM tuner, possessing tremendous gain ahead of the audio-restoring stages, precludes the employment of the familiar 6U5 tuning eye, connected so as to indicate the maximum limiter current, as a means of resonance indication. For the amount of voltage de-

veloped across the limiter grid resistor is often so great as to overlap the tuning eye. Hence, it is impossible to find the exact center of the carrier excursion during modulation. It can also be shown that a slight misalignment of the intermediate frequency channel will lead to a false tuning point, and consequent audio distortion, especially when a high percentage modulation signal is being received.



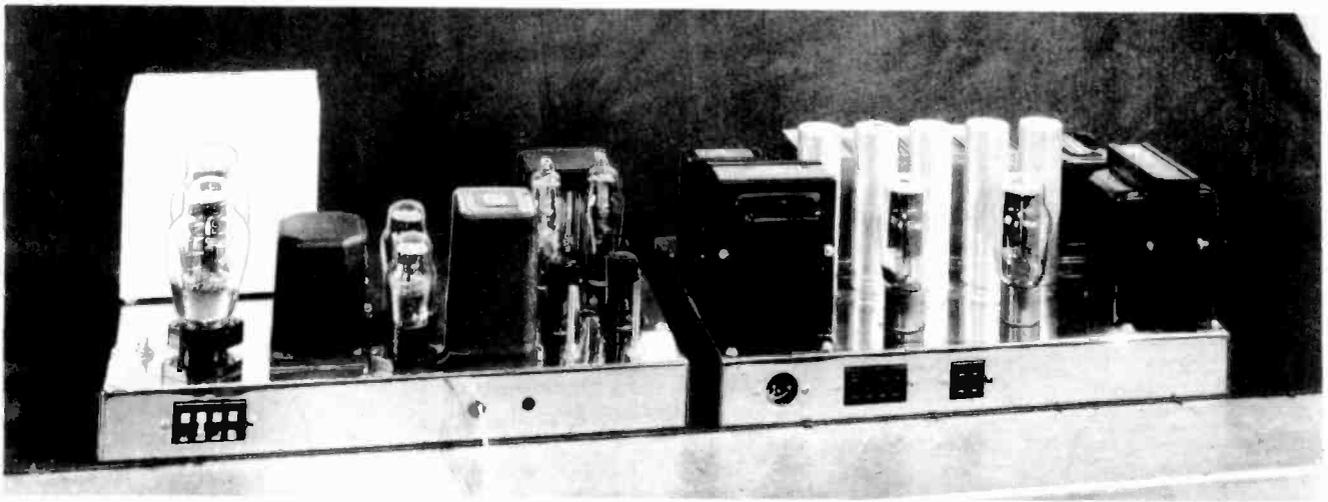


Fig. 1. The high-fidelity, high-quality amplifier and its power supply.

# 30-Watt High-Fidelity Audio Amplifier

CURTISS R. SCHAFER

A general-purpose medium-power amplifier of broadcast quality.

**T**HE LOW-MU TRIODE AMPLIFIER enthusiasts are at it again, and in view of the impracticability of some of the designs that have been offered within the past few years, the author would like to point out a few fundamental requirements that must be met by a truly high fidelity amplifier which also can furnish sufficient power to "fill" a small auditorium, for loudspeaker testing, or for driving a wax or acetate cutter. These basic requirements are:

1. *Sufficient power output and gain.*<sup>1,2</sup>

2. *Low listening fatigue*, which is primarily insured by low intermodulation distortion.<sup>3,4</sup> The distortion products should result only from second and third harmonics instead of the higher order harmonics usually generated in beam-power tubes. Single-frequency harmonic analysis is useful only in determining some of the operating parameters of the tubes and transformers involved.

3. *Good transient response*, which results in a particularly clean-cut reproduction of speech. Whistling consonants are evidence of parasitic oscillations on peaks. The specific factors responsible for good transient response are (a) good high-frequency response, (b), low phase rotation,<sup>5-8</sup> (c), low internal impedance as seen by the loudspeaker or cutting head, and (d),

low hysteresis distortion, especially at the higher frequencies, in any transformers that may be involved.

It should always be remembered that we are not dealing with sine waves in the reproduction of speech and music. This is important in the matter of phase rotation, for instance; two sine waves out of phase still add up to a sine wave, but two waveforms containing harmonics, and out of phase, add up to a new and different waveform which does not resemble either of the originals. Excessive phase shift in an amplifier sounds like high intermodulation distortion, and produces excessive listening fatigue.

4. *Good input vs. output linearity.* This demands that each voltage amplifier stage be capable of supplying several times the actual voltage required to drive the following stage.

5. *Reliability, ruggedness and ease of servicing*, both with regard to locating the faulty part and replacing it easily. This calls for the use of oil-filled or oil impregnated condensers, hermetically sealed transformers and chokes, resistors which do not get noisy with age and/or temperature variations, and tubes which are of simple design structurally, have their elements well braced, and do not have close grid-to-cathode spacing. Close element spacing offers the probabilities of high microphonism, grid emission and wide variations in electronic para-

eters with small variations in tube geometry. Glass envelope tubes usually have a lower gas content than metal envelope tubes. In addition, glass tubes are an aid to rapid servicing, as a bright spot on a cathode, a gassy rectifier, or an open filament are easily noticed.

## Design

With these elementary considerations in mind, we began the design of an amplifier. After a year of loudspeaker listening and testing with amplifiers rated at 10, 15, 20, 30, and 40 watts output, it was decided that an output of 30 watts with less than 5% total intermodulation distortion was required. We felt that this output should be obtained with triodes rather than beam power tubes: first, because the absence of a feedback loop would result in a simpler and more easily serviced amplifier; second, because the higher order harmonics generated by beam-power tubes (running up through the tenth) were considered undesirable, even though relatively low in amplitude. The value of distortion selected is so low that we felt we could ignore it when the amplifier is used in listening tests.

The output triodes were selected from a list of the following types: 50, 2A3, 6A3, 6B4G, 6A5G, 6AS7G, 3C33, RJ-563 and DRJ-564, and 300A. The first five of these belong to the same generic type, the 6A5G being a heater-



mumetal cored interstage transformer, of the author's own design, with a stepup ratio of 1:2. The UTC LS-22 could just as well be used here. The 76 tube will handle almost 50% more driving voltage than the 6J5 or 6C5.

The amplifier is push-pull all the way; second harmonic distortion is pretty well down, and less plate supply filtering is required than for single-ended stages. The first stage uses Western Electric 347As, which generate very low values of hum, microphonics and fluctuation noise.<sup>11</sup> The 1603 triode connected is an acceptable substitute.

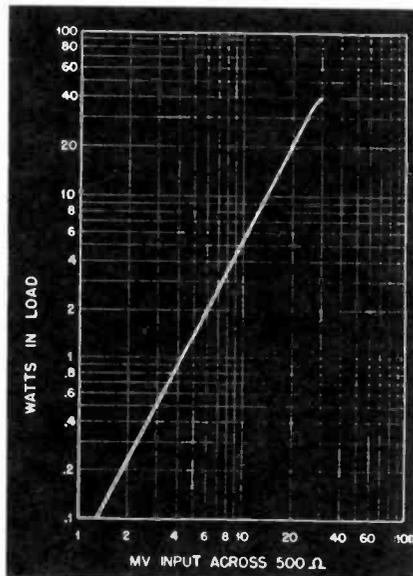


Fig. 4. Voltage sensitivity graph.

A variation of the Thordarson degenerative tone control circuit is used, because it will (a) handle a fairly high value of input signal, (b) give a 1:1 voltage "gain" even in the flat response position, and (c) introduce no measurable harmonic distortion of its own. Here again, 1603s triode connected may be used in place of the 6C5Gs.

Two 45-volt batteries, Burgess 5308 or equivalent, were selected as being the most reliable and economical source of bias voltage. No current is drawn from them, and from previous experience the author has found that they are good for about two years before their voltage begins to drop at all. The power supply circuit is conventional throughout. The 5R4GY was selected for rectifier service because its voltage ratings are high, it is economical of filament power, and electrolysis at its lead-in wires proceeds at a much slower rate than it does in the 5Z3 or 5U4-G. No electrolytics are used.

Figure 1 is a photograph of the amplifier, with its power supply chassis at the right. Eight-prong Jones plugs

and sockets are used for inter-connecting the two, with a four prong Jones plug for the speaker. A four-prong socket is also provided on the power supply chassis so that heater and plate supply power may be taken for a pre-amplifier or tuner. The 10  $\mu$ f 1000-volt input filter is made up of the five 2- $\mu$ f 1000-volt cylindrical cans on the power-supply chassis. The output transformer is at the left rear of the amplifier chassis, and the cast case holding the two bass tone control chokes is between the 76s and the 6C5s. Metal 6C5s were used at the time the picture was taken, but 6C5-Gs are used in their places now. The two dual 100,000-ohm tone control potentiometers are Mallory, Type LL. The 347As are shown at the right rear, just this side of the input transformer.

Figure 2 is a schematic of the amplifier, and Fig. 3 of the power supply. Figure 4 shows the voltage sensitivity and excellent linearity of the amplifier, and Fig. 5 shows the intermodulation distortion as read on an Altec Lansing intermodulation analyzer. The frequencies used were 40 cps and 12 kc. Total hum and noise are 75 db below maximum rated output. With the tone controls in the "flat" position, the frequency response at 30 watts output is uniform within 2 db from 30 cps to 25 kc, with almost all of this variation taking place above 15 kc.

The tone controls provide a maximum of 8 db boost at 50 cps and 10 db boost at 8 kc. No bass or treble attenuation is provided, as it has been the author's experience that such attenuation is rarely if ever used.

Listening tests, comparing this amplifier with 15, 30 and 40-watt amplifiers of well-known manufacturers, have confirmed the low distortion and excellent transient response of this design.

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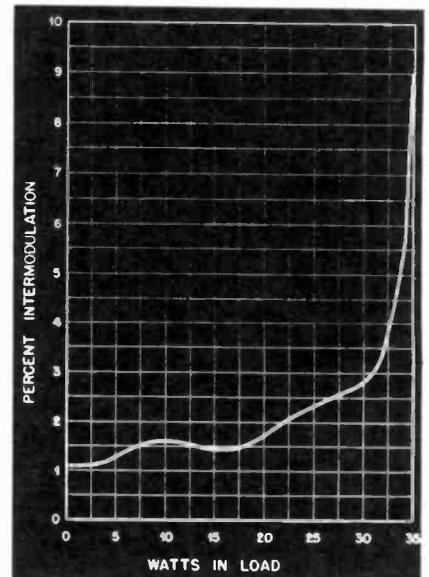


Fig. 5. Intermodulation distortion at various levels.

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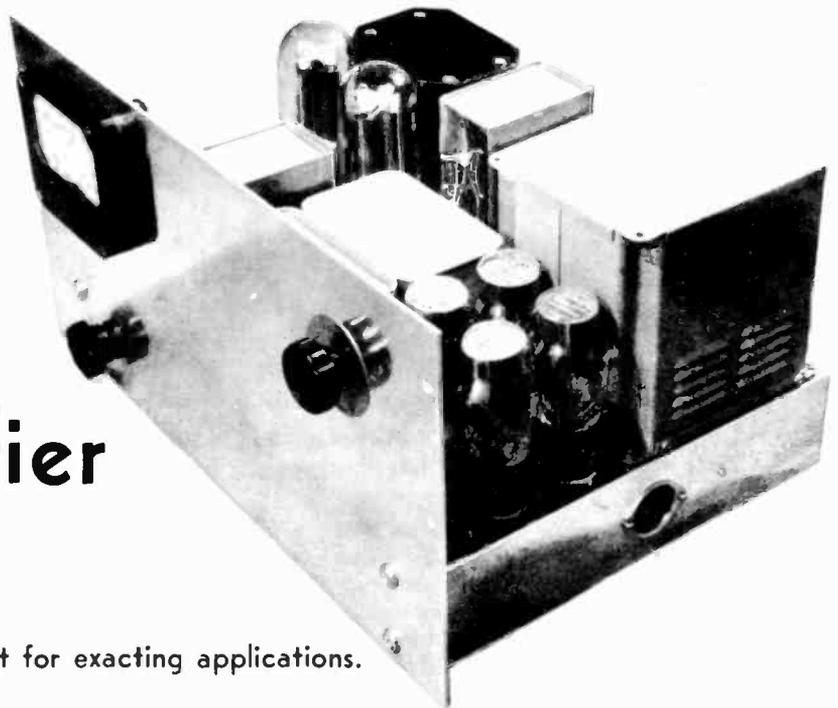
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POSSIBLE TRANSFORMER SUBSTITUTIONS						
	Input	Interstage	Output	Power	1st Choke	2nd Choke
ADC	215A	215D	315E			
Freed			F-1967			
Langevin			300A			
Peerless	K-251-Q	G-212-Q			C-455-A	
Triad	HS-4	HS-27	HS-94			
UTC	LS-12	LS-22	LS-6L4	S-43	S-31	S-27

Three-quarter view of high-fidelity amplifier chassis.

WALTER T. SELSTED  
and  
ROSS H. SNYDER

# High-Power Triode Amplifier



The authors describe a high-quality audio unit for exacting applications.

**A**N AUDIO AMPLIFIER of high gain, high power, high stability, wide frequency range, low distortion and low noise content was required at the outset of a series of measurements of human hearing, loudspeaker performance, and recording fidelity. Because this unit was to be a link common to all the measurements, its characteristics had to exceed substantially the best expectations of the electro-mechanical equipment involved.

Calculation indicated that with loudspeakers of average efficiency peak powers of as much as 25 watts might be needed to attain the desired sound levels. A frequency range of 30 to 20,000 cycles within 1/2 decibel would

exceed the requirements of measurement, but good voltage regulation of the output would have to accompany the response. Because the amplifier would be used at low as well as high output levels, internal noise would have to be kept below audibility compared to the zero vu level. It was anticipated that the measurements for which this equipment was constructed would take some weeks or months to complete, and since the data accumulated would be correlated, extraordinary stability would be a requirement in the amplifier. Finally, every reasonable precaution had to be taken against introducing distortion products in the amplifier, of whatever obscure

origin, which would offend the most sensitive listener.

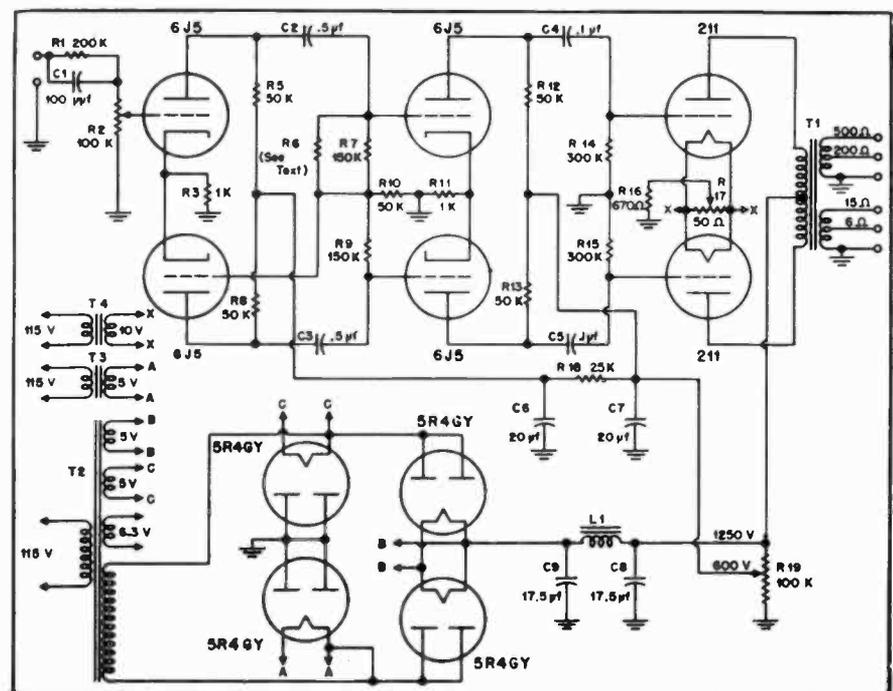
With these requirements in view, the long smooth curves of the larger triode power tubes looked attractive. An opportunity to evaluate the results to be had from the high costs usually associated with triode designs thus presented itself also.

## Low Cost Components

Funds for the experiments were no more plentiful than usual, so the possibility of using low-cost war-surplus tubes was explored. The rather formidable 211 has for a long while been in copious supply at prices comparable to those of receiver types. Power supply requirements, always severe in triode

Fig. 1. Schematic of audio amplifier.

- R1—200,000 ohms, 1/2 watt, carbon.
- R2—100,000-ohm wire-wound potentiometer.
- R3, R11—1000 ohms, 10 watts, wire-wound.
- R5, R8, R12, R13—50,000 ohms, 10 watts, wire-wound.
- R6—(see text).
- R7, R9—150,000 ohms, 1/2 watt, carbon.
- R10—50,000 ohms, 1/2 watt, carbon.
- R14, R15—300,000 ohms, 1/2 watt, carbon.
- R16—1000 ohms, 50 watts, slider adjusted to 670 ohms.
- R17—50-ohm 4-watt potentiometer, wire-wound.
- R18—25,000 ohms, 10 watts, wire-wound.
- R19—100,000 ohms, 200 watts, with slider, wire-wound.
- C1—100  $\mu$ f, mica.
- C2, C3—0.5  $\mu$ f, 600 volts, oil-filled.
- C4, C5—0.1  $\mu$ f, 600 volts, oil-filled.
- C6, C7—20.0  $\mu$ f, 600 volts, oil-filled.
- C8, C9—17.5  $\mu$ f, 1500 volts, oil-filled.
- T1 U.T.C. PVM-3
- T2 U.T.C. S-43
- T3 U.T.C. S-54
- T4 U.T.C. S-62
- L1 U.T.C. S-31



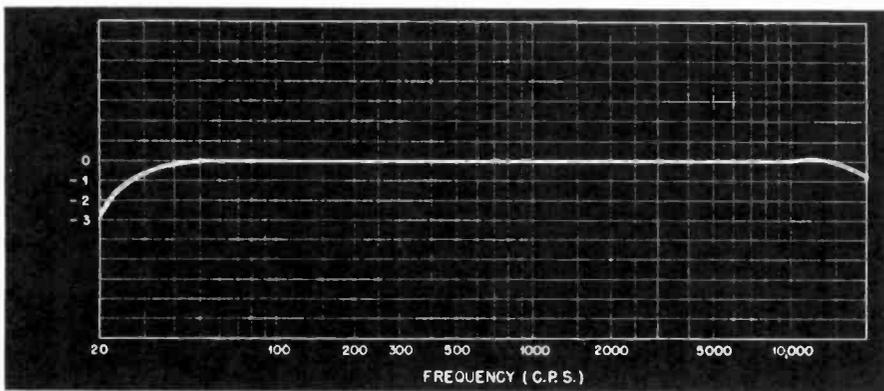


Fig. 2. Frequency response curve of audio amplifier at 30 watts output. Note that it is flat within less than one db from 30 to 20,000 cycles.

designs, at first made it appear an uneconomic choice, but a reasonable solution was found. The wide demand among radio amateurs for power equipment in this range has resulted in quantity production of the required transformers at tolerable cost. High-voltage condensers were available through war surplus. And a bridge rectifier was feasible with the low-priced 5R4GY rectifier tube, with its comfortable 250 ma current rating, and 2100 inverse peak voltage capability. The problem appeared surmountable if costly interstage transformers could be avoided.

So the 211 was tentatively selected, and in the interest of reducing drive requirements, insuring stability of operation, and eliminating the necessity of a fixed-bias supply, pure class A connection was proposed.

The availability of very high plate-

supply voltages suggested the possible use of ordinary receiver triodes in conventional resistance-coupled circuits. The calculations revealed that this would be the most critical section of the amplifier, but with 600 volts at the high-potential end of the 6J5 plate resistors these tubes, operating at nearly full rated dissipation in push-pull, would drive the 211s just beyond the requirements.

The medium- $\mu$  driver triodes dictated the use of another stage of voltage gain, which could conveniently use the same tube type. Phase inversion was removed to the input stage so that degeneration in the drivers' cathode resistor would tend to remove any unbalance in a.f. grid-to-ground voltage remaining after inversion. The 6 decibels of gain added by making the input a push-pull stage was also desirable. The need for high stability with

high gain demanded a tube inverter of unusual characteristics. The cathode-loaded inverter was rejected, despite its stability, because of the possibility of cathode-filament hum, a considerable hazard where relatively high gain follows inversion. Gain requirements ruled against the cathode inverter too. The ordinary voltage-divider type was eliminated by its necessity for constant readjustment with tube aging. The stability and gain of the conventional floating-paraphase inverter were attractive, but its never-quite-perfect balance was a disadvantage. In the end a modification of the latter type proved satisfactory: the load on the input tube was varied by shunting  $R7$  with  $R6$ , of varying values above a megohm, until a virtually perfect balance of a.f. to the driver grids was obtained. The floating-paraphase action maintained the balance over long periods of use.

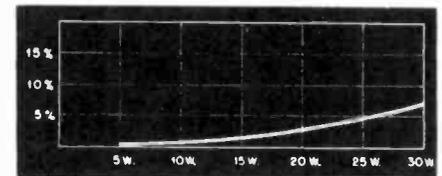


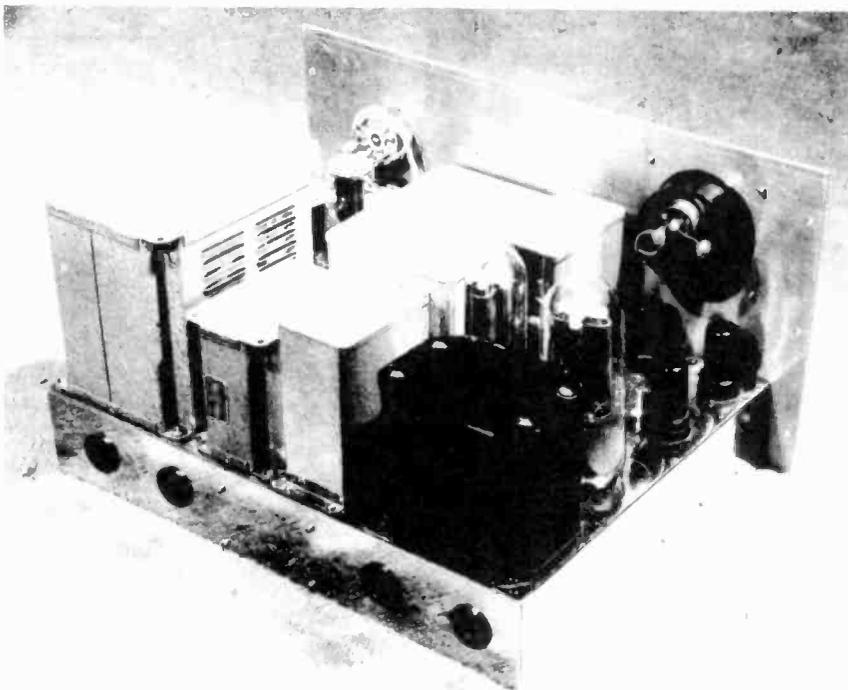
Fig. 3. Intermodulation distortion vs. power output.

The final refinement in the circuit was the addition of a short time constant filter in the input, calculated to counteract the high-frequency losses within the amplifier due to distributed shunt capacitances. High-frequency losses due to Miller effect, often experienced in high-impedance triode grid circuits, were held to a minimum by the 100,000-ohm value assigned the input potentiometer. This value likewise reduced the chances of frequency discrimination from RC effects, prominent in many higher-impedance input designs.

Again in the interest of low noise level and high stability, wire-wound resistors were used in all d.c.-carrying circuits, and oil-filled capacitors were used throughout. If a suitable quality of oil condenser is used for coupling, d.c. leakages will remain even after long use at values lower than those found in new paper condensers, keeping the grids essentially at ground potential, with attendant long-time circuit stability. In the presence of heavy a.c. fields the shielding afforded by the can construction is probably a considerable advantage too.

The photographs show how the entire unit was mounted on a single standard rack chassis. The controls illustrated

As shown, the amplifier is built on a standard rack chassis.



on the front panel are for gain and monitoring level.

It should be noted that potentially lethal voltages exist on the underside of the chassis. No high-voltage contacts are brought to the top. The equipment is mounted so that no access can be had to the underside without first disconnecting the power.

**Tests**

The intermodulation curve of Fig. 3 was taken with the Pickering analyzer, using input frequencies of 100 and 7000 cycles at 10 decibels difference in level. Total harmonic distortion analysis gave reading of 1/3 to 1/4 the numerical values of the IMD figures, at corresponding powers.

The frequency-response curve of Fig. 2 is a power line, based on 30 watts, and is therefore of significance at much more than usual operating levels. The low-frequency drop is attributed to output-transformer core-saturation.

Noise level was measured with a

shielded 100,000 ohm resistor across the input terminals, with the gain control wide open. Further amplification of the output was required to distinguish its character audibly. It was the usual random hiss superimposed upon a lower level of hum. The absolute level of noise at the output under these conditions was -52 dbm, or nearly 96 decibels below the nominal 30-watt level.

Effective internal impedance at the 15-ohm tap is 6 ohms.

Thirty watts of output was reached with about 0.1 volt drive, giving a nominal 85 db maximum gain.

Although all the tests indicated that the amplifier was suitable for its purposes, the final judgement would have to be that of prolonged listening by critical people. The nature of the distortions which offend a "golden ear" still defies analysis, but whatever their causes these offenses had to be excluded from the present amplifier if it were to be useful. The operation was ap-

parently a success, for the equipment was later used with laboratory-produced transducers capable of accurate reproduction of sounds in the 20,000 cycle area without pain to the perhaps hypercritical listening of the speaker-creators. Prolonged use has confirmed the original opinion that the amplifier is contributing no unpleasantness. Its low internal impedance is audibly an effective damp on oscillatory excursions of bass speakers employed with it.

For all of the excellent characteristics obtained here, no fuel is meant to be added to the fire of academic argument over triodes and beam tubes. The high-frequency equipment mentioned above has been used with beam amplifiers employing ample feedback with no decrease in listening satisfaction. It is believed that the requirements met by the present amplifier could have been reached equally well, and at lower cost, with beam tetrodes using generous feedback.

# Low-Power Cathode-Coupled Amplifier

RAYMOND H. BATES

Lt. Col., CAC

**T**HE amplifier described below is the result of an arduous two year search for an inexpensive, low power, good quality audio amplifier for home use with an FM and AM tuner and a phonograph reproducer.

The circuit illustrated below represents the results of experimentation with many conventional circuits that have been published in the available magazines. All were discarded one by one until the author, in desperation, tried the cathode-follower output stage. The results were completely satisfactory.

According to the limited literature available in various publications, a cathode-follower output stage, in comparison with a conventional output stage, will provide.

1. Improved low frequency response.
2. Improved high frequency response
3. Damping out of peaks in both the output transformer and speaker.
4. Less distortion at the same rated power output.
5. 100% degenerative feedback with all its benefits.

The circuit is simple and straightforward with no special tricks to reduce hum, although the amplifier constructed

has no audible hum at full gain.

The primary of the output transformer, T<sub>2</sub>, was selected so that its d-c resistance was approximately equal to the normal cathode bias resistor, or approximately 250 ohms. The primary impedance should be 5,000 ohms, with the secondary impedance selected to match the speaker to be used. The plate and screen of the 6V6 are tied together, and to the B supply. The power transformer, T<sub>1</sub>, has a secondary voltage of 250-300 v. each side of center tap, and

the choke L<sub>1</sub> is a ten-henry, 70-ma unit.

When used with the new RCA 45-rpm record attachment or a good FM tuner, together with a Jensen extended-range twelve-inch speaker in a bass reflex enclosure, the results are astonishing, and the power output more than meets the requirements for small living room use. It will be noted that the gain is not adequate for use with low-level magnetic pickups, but the simplicity of the amplifier makes it well suited for small, high-quality installations.

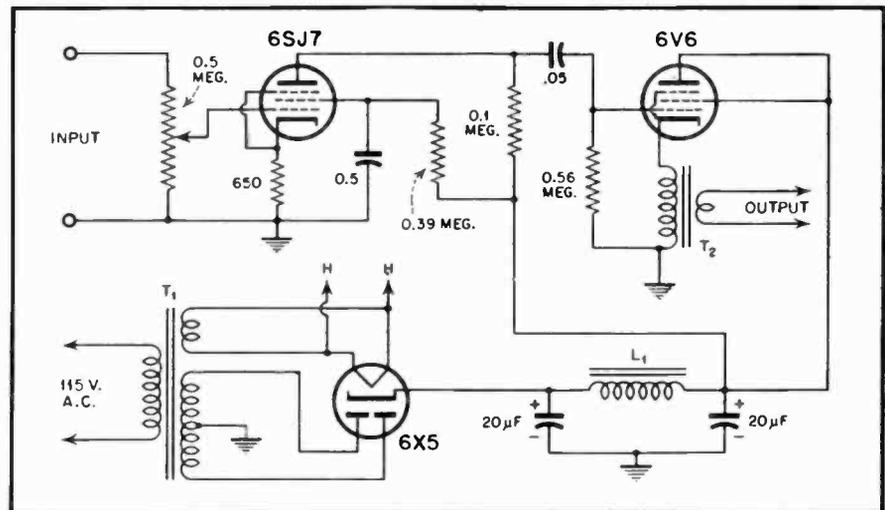


Fig. 1. Eight 6V6 tubes in push-pull-parallel cathode-follower output stage, and two paralleled power transformers give this amplifier an unusual appearance. Simple circuit arrangement provides maximum speaker damping.

W. E. GILSON, M.D.,  
and  
RUSSELL PAVLAT



# A Practical Cathode-Follower Audio Amplifier

A design which provides high damping for the loudspeaker, resulting in excellent transient response.

**S**PEAKER damping is a comparatively neglected element in the discussion of high-quality amplifiers. Specifications usually state that the frequency response curve can best be drawn with a reliable straightedge, and that the percentage of distortion is negligible at a substantial number of BTU's output per second. The absolute flatness and freedom from distortion so obtained are largely of emotional value to the designer, as a slight variation at high output cannot be detected except by measurement, because the inner ear generates up to 50% harmonic distortion

under that condition.<sup>1</sup> Such an amplifier may sound clean, or it may be quite "muddy," depending on the damping.

Such fine sounding specifications are usually accompanied by (sometimes accomplished by) the use of large amounts of inverse feedback. Thus fair damping is usually obtained, particularly if an excellent output transformer is used, or if the feedback is taken from the voice coil winding. Except

<sup>1</sup>E. G. Weaver and C. W. Bray, *J. Acous. Soc. Am.*, Vol. 9, pp. 227-233 (Jan. 1938).

in a very good transformer, there is regrettably little relation between the wave shapes of the voltages on the primary and secondary windings.

Although in comparable circumstances triodes sound somewhat cleaner than beam-power tubes to the writers, this is an uncertain subjective judgment. Many persons interested in audio reproduction will, when the proper stimulus is applied, inevitably state their dogma "Nothing can beat a pair of 2A3's." One amplifier on the market at present, with the modern equivalent of the 2A3, sounds definitely muddy, largely because of a lack of damping. No inverse feedback from the output winding is used, and the transformer is not of sufficiently high quality to provide close coupling between the tubes and the voice coil. Thus a good solid triode does not inevitably give clean reproduction. It can give very good results if used in a well designed amplifier.

The use of the 6AS7G appears to provide some improvement in damping. An experimental amplifier using this tube, constructed some time ago, sounded remarkably clean, considering that a poor output transformer was used. McProud<sup>2</sup> obtained an impedance of 7.2 ohms across the 16-ohm output winding, using the 6AS7G in a better amplifier with good components.

<sup>2</sup> High Quality Amplifier with the 6AS7G; see pages 5-8.

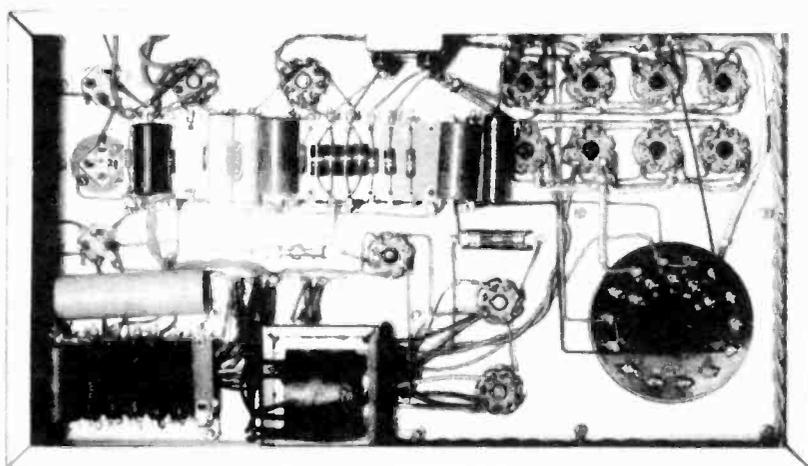


Fig. 2. Underside of amplifier chassis. Note clean, straightforward wiring layout.

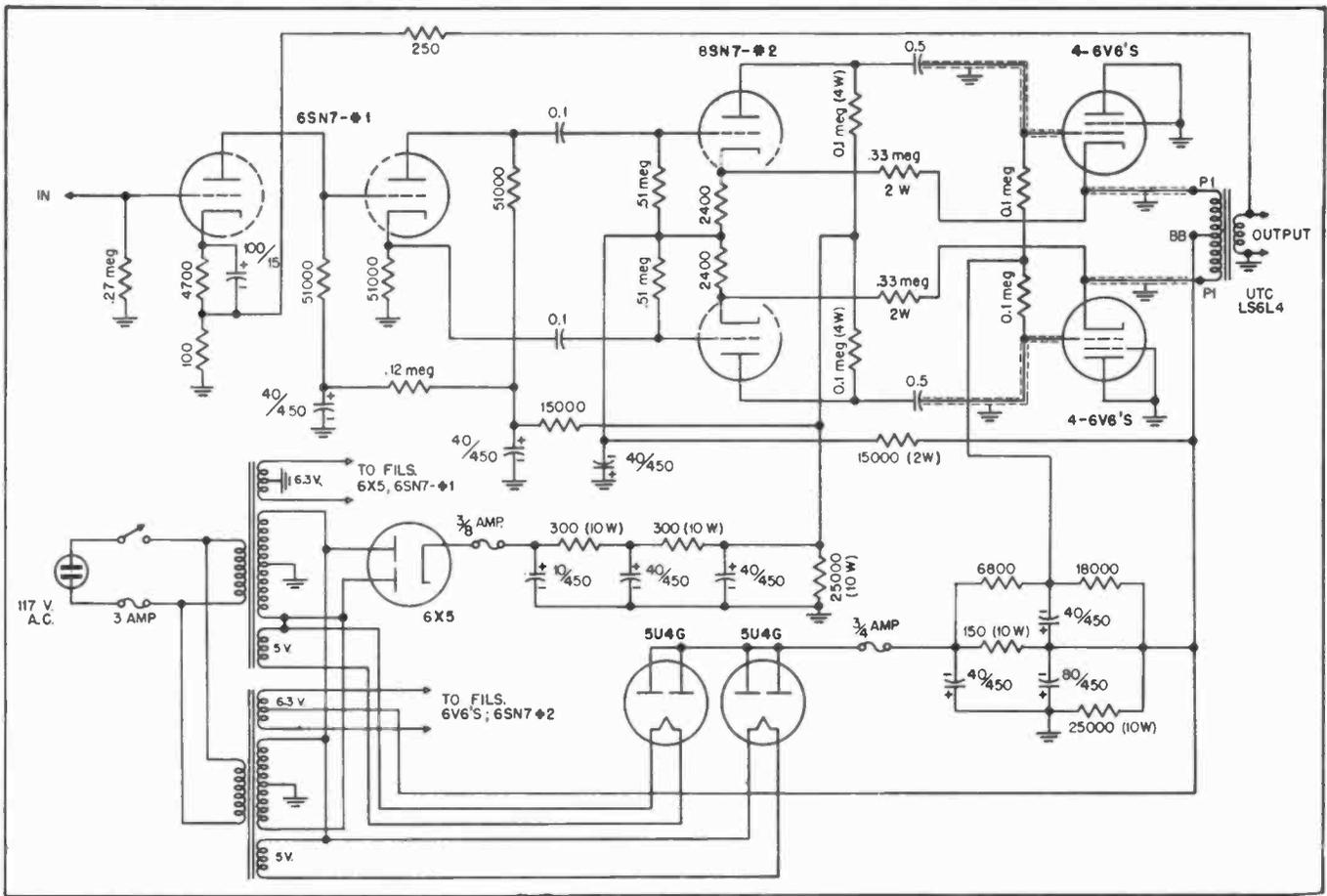


Fig. 3. Overall schematic of complete amplifier and power supply.

Considerably better damping can be obtained with a cathode follower amplifier. This type of audio output circuit has seldom been used, probably because of the high driving voltage required. The present circuit has been developed to provide adequate output with an economical source of distortion-free high driving voltage. The circuit appears slightly unconventional, but there is a definite reason for all of the departures from the usual. The amplifier is shown in Figs. 1 and 2.

#### Driving Voltage

The driving voltage is obtained by using a resistance coupled 6SN7 with a plate supply voltage of 700, and a plate voltage of 300. The cathode follower output requires high current at low voltage, and the preamplifier stages require low current at low voltage. To fulfill these requirements, two SNC transformers are used with their high-voltage secondaries in parallel. The combination is then rated at 350 volts each side of the center tap, at 300 ma. Two 5-volt and two 6.3-volt windings are also available. One of the 6.3-volt windings, with its center tap connected to ground, is used for the heaters of the early amplifier stages and for the 6X5GT rectifier, which supplies 350 volts above ground at low power for

these stages. The two 5-volt windings are used for the two 5U4G rectifiers, which are connected in the reverse of the usual manner to provide 350 volts below ground at high power, for the output stage. There is a total difference of 700 volts, across which the driver stage is operated. The remaining 6.3-volt winding, which has its center tap connected to minus 350, supplies

the heaters of the driver and output stages.

The bias for the final stage is obtained from the pulsating d-c voltage drop across the 150-ohm filter resistor, which voltage would otherwise be wasted. This is voltage-divided and filtered and gives a 25-volt negative bias which is highly stable.

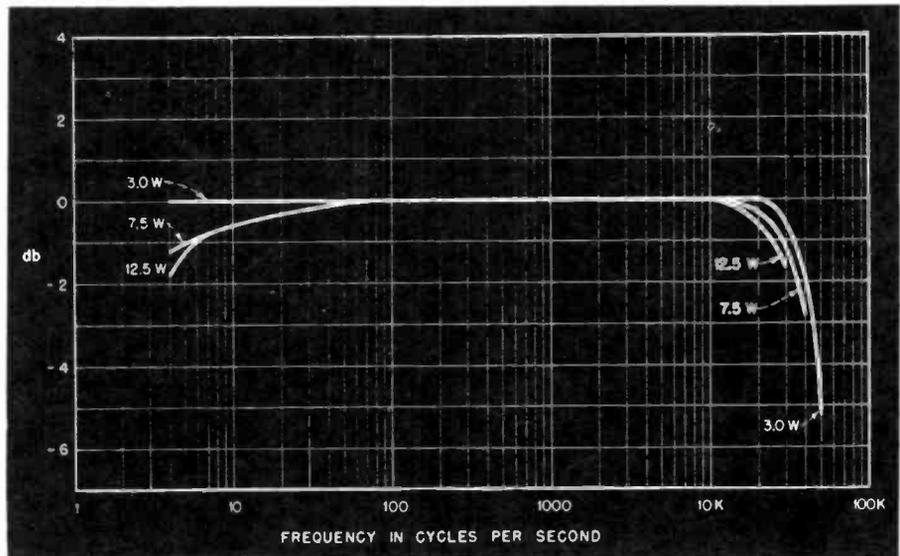


Fig. 4. Frequency response curves at different output levels. Note that range extends from 4 cps to well over 30 kc.

In order to obtain peak output, it is necessary to make the driver and cathode follower circuit slightly more sensitive, as the limitation lies in the phase-inverter output. Approximately 5 per cent regenerative feedback is added from cathode to cathode, increasing peak output and noticeably reducing peak overall distortion. This changes the output impedance imperceptibly because of the large amount of overall degenerative feedback. The regenerative circuit may be omitted if

Early models of this amplifier manifested a variety of motorboating. Because of the low impedance of the output stage, and its consequent control of the power supply voltage, in-phase motorboating is likely to occur due to the push-pull tubes acting as if they were in parallel. The first model using this circuit had the lower ends of the 2400-ohm driver bias resistors connected directly to minus 350. Motorboating occurred even with only the drivers and output tubes plugged in. This can be explained by regarding the minus 350 as the reference point, with ground moving up and down in voltage, depending on the drain by the output stage on the power supply. Ground is coupled to the grids of the drivers through various capacitors and resistors. A negative pulse on the grids of the drivers will cause the grids of the output tubes to become less negative, causing greater drain on the power supply. Ground and the grids of the driver will thus become more negative, and motorboating results. This interesting but undesired phenomenon was eliminated by inserting a 15,000-ohm resistor between the 2400-ohm bias resistors and minus 350, and connecting a 40- $\mu$ f capacitor from the junction to ground. As it is now connected, when ground moves up and down both the grids and cathodes move with it, and the output is practically zero.

#### Speaker Damping

The primary purpose in the design of the amplifier was to achieve unusually good damping. This has been carried to such an extent that the limit-

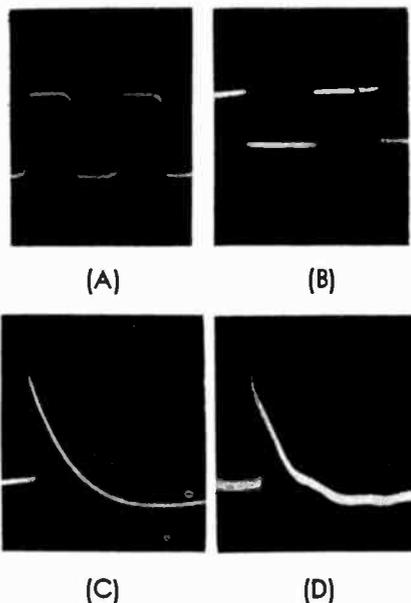


Fig. 6. Oscillograms of output for various input signals: (A) 200-cps square wave; (B) 5000-cps square wave; (C) 0.5-volt d-c pulse at input, speaker connected normally; (D) same signal as at (C), but with 10-ohm resistor in series with voice coil. Last two oscillograms show effect of damping.

ing factor is the d-c resistance of the output winding, less than one ohm. The cleanness of reproduction obtained is definitely noticeable. A good demonstration is side 4 of Columbia's "Young Person's Guide to the Orchestra", which has excellent low notes as well as the crack of a whip near the end of the record. The extreme low frequencies are more naturally reproduced because of the added control of the voice coil. There is no muddiness. Another test of the damping, somewhat analogous to definition in a camera lens, is the separation of rapidly rolled drumbeats.

It is usually suggested that the output of a tweeter be reduced by means of a resistor network, because it is higher in efficiency than a woofer. In listening tests it has been found tentatively that the introduction of a resistance in series with a good compress-

sion type tweeter reduces its sharpness and clarity, making it sound like an ordinary cone type tweeter. It is thus suggested that other methods might be tried, such as multiple tweeters spreading the sound more widely, for example.

The amplifier is used primarily in our "Gilson", a radio-phonograph combination using a Jensen 18-inch theatre speaker with two Jensen tweeters. Passably good reproduction can be obtained, however, even with a 15-inch speaker designed for home use, because of the control of the voice coil.

The frequency response is shown in Fig. 4, the total harmonic distortion in Fig. 5. The deviations shown cannot be detected by the ear.

Figure 6 shows oscillograms taken from the 16-ohm voice coil of a 15-inch speaker. The response to a square wave at 200 cps is shown at (A), at 5000 cps at (B).

The effect of damping is demonstrated at (C) and (D). A 0.5-volt d-c pulse is applied to the input, producing a transient, the shape of which depends on a number of factors. (C) shows the lack of overshoot and smooth exponential return as the coupling capacitors discharge. At (D), a 10-ohm resistor has been connected in series with the 16-ohm voice coil. The potentials produced by the marked overshoot and continued vibration of the voice coil can clearly be seen.

Reference to the photograph of the underside of the chassis may occasion some question as to the reason for the small channel-frame transformer. It supplies filament current to a tuner used with this amplifier, but is not shown on the schematic.

The output transformer specified in the original circuit is a UTC LS-6LA, but equivalent results should be expected from ADC type 315G, Freed F-1967, Langevin, 316A, Peerless S-245-Q, or Triad HS-94. With respect to the power transformers, it is obvious that the least costly arrangement is that shown, employing two transformers in parallel. It must be remembered, however, that the two transformers used must be identical, or a certain amount of circulating current will flow in the secondaries. If it should be desired to employ only one power transformer, any of those listed below should be adequate. Some of these substitute transformers may be capable of furnishing a reasonable amount of heater current for an additional tuner or a control amplifier. The suggested substitutes are: UTC CG-431, Langevin 100B, Peerless R-800-A, or Triad R-24A.

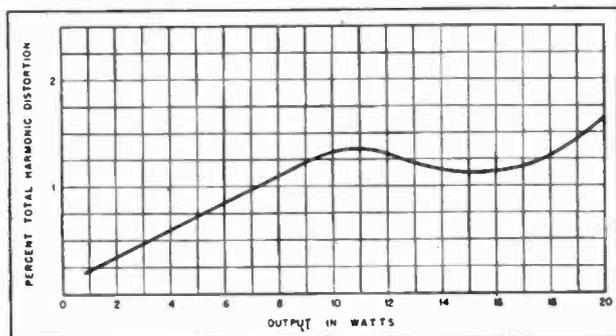


Fig. 5. Total harmonic distortion vs output at 50 cps.

Fig. 1. The two chassis comprising the amplifier patterned after the "Williamson."



DAVID SARSER

and

MELVIN C. SPRINKLE

# Musician's Amplifier

An adaptation of a famous English circuit which has already earned an enviable reputation for quality.

**A**MPLIFIERS are something like women—there are a lot of 'em. And—like women—amplifiers come in all sizes, shapes, and degrees of appeal. The readers of radio magazines have seen dozens of different amplifier designs, and audio enthusiasts are as vociferous in praise of their pet circuits as Kentucky colonels about their pet way of making a mint julep.

The writers of this article—one a professional violinist in Toscanini's NBC Symphony, owner of the famous Lamoreux Stradivarius, and a builder of amplifiers, the other a professional electronic engineer whose avocation is serious music—have both made a baker's dozen of amplifiers, but have felt that most of these amplifiers in one way or another never completely recreated a satisfactory "studio sound."

The writers, then, have pooled their resources and have come to the conclusion that there should be a simple, easy-to-make, and thoroughly foolproof circuit which gives out with that intangible something called "presence effect." In doing so, we are going to duck controversies regarding beam-power tubes versus triodes, transformers versus resistance coupling, and all the other perennial areas of enthusiastic conflicts and ideologies. All we're really after—how simple it sounds!—is to reproduce music in the home which sounds the same way it does in the concert hall or broadcast studio.

Recently, we heard about the "Williamson" circuit which has been widely

publicized in England and Australia as the absolute tops for obtaining natural reproduction. It was written up in *Wireless World* (an English publication) for April and May, 1947. This circuit has become so popular in England that it was reprinted with minor modifications in the August, 1949, issue of the *Wireless World*.

Having studied the Williamson circuit and having read the comments on its intermodulation distortion in *AUDIO ENGINEERING* for September, 1948, we became interested in the possibilities of the amplifier.

Strictly from the technical point of view, the output tubes are 807's with triode connection. The 807 has much to recommend it as an audio tube. It is a standard type, available everywhere, and though it is usually considered to be a transmitting tube, its price makes it no more expensive than other tubes used by amplifier constructors. It has a fairly high plate dissipation, and draws enough plate current to provide adequate audio power. It is a cathode type, rather than direct heated and hence the completed amplifier has no hum even when used with efficient speakers with fine bass response such as the Altec Lansing 604B.

The output transformer presented somewhat of a problem as the original circuit called for a transformer not manufactured in this country. If it were specially made to Williamson's specifications the cost would be prohibitive. A careful survey and trial on

various transformers available through jobbing channels disclosed that one transformer—the Peerless S-265Q—would best meet the rigid specifications for performance.

The resistance network in the cathode circuit of the 807 tubes contains a variable resistor  $P_1$ , which is set to give equal current in each tube. A closed circuit jack is provided in the cathode circuit of each 807 to permit insertion of a milliammeter when making this adjustment, and the plate current should be balanced by  $P_1$  to exact equality, at 50 ma per tube.

The original Williamson diagram contained a series resistor to adjust the total plate current drain of both tubes. After building several of these amplifiers it was decided to omit this control and substitute a fixed resistor, since once it is set it is never changed.

The driver stage consists of a single 6SN7GT tube connected in push pull and resistance coupled to the 807's. Here one of the points of superiority of the 807 is evident. Most low-mu triodes such as the 2A3 and its six-volt counterpart, the 6B4G, have a high bias of the order of 60 volts and, hence, require a driver stage capable of putting out signals whose peak values are of this magnitude. As a result, resistance coupling to low-mu triodes is not practicable unless elaborate precautions are taken to supply adequate driving voltage with low distortion, a difficult task. The triode-connected 807 requires about 35 volts bias and thus the

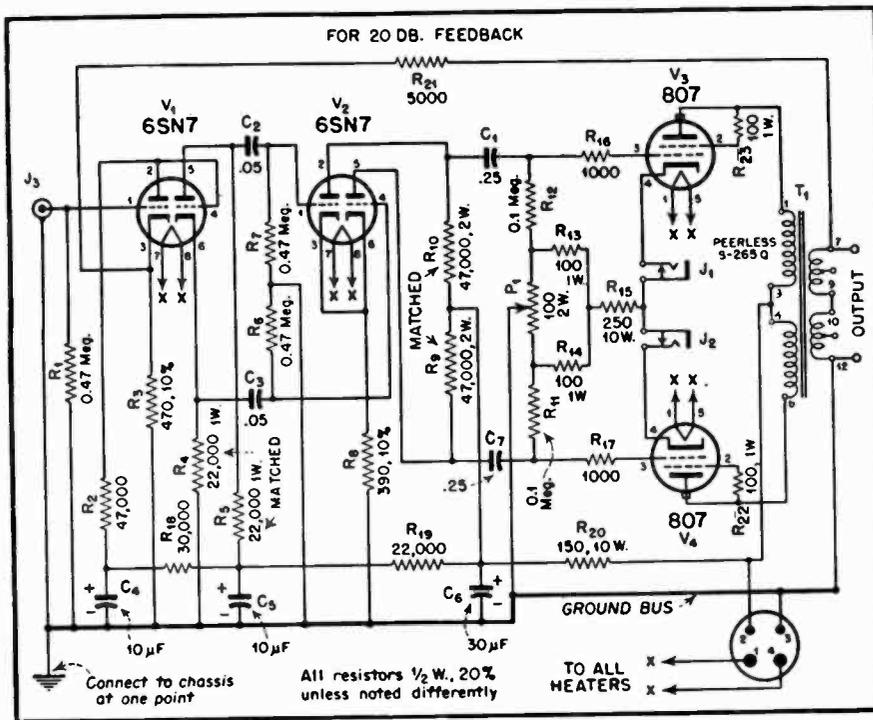


Fig. 2. Schematic of the amplifier section.

use of 6SN7 as a driver becomes practical. The first two stages are unique as they consist of a single 6SN7 tube using the first section as a voltage amplifier which is directly coupled to a "cathodyne" or split load type of phase inverter. Use of this type of phase inverter permits direct coupling, as the positive voltage on the phase inverter grid is offset by the high positive voltage on its cathode. Direct coupling to the phase inverter permits the extension of the low-frequency response in this stage to d.c. and eliminates coupling networks which can cause phase shift and low-frequency attenuation. It will be seen from the schematic, Fig. 2, that there are only four coupling capacitors in the entire amplifier, two on each side of the push pull system. Thus the response of this amplifier at low frequencies is remarkably good.

#### Inverse Feedback

It will be noted that inverse feedback is used in this amplifier. Inverse feedback has been used with beam power amplifiers, but the circuit is not confined to this type of amplifier. Our amplifier incorporates approximately 20 db of feedback which is put around four stages and the output transformer. With this much feedback there is absolutely no trace of supersonic or sub-audible oscillation, a tribute to the design of the output transformer. Feedback greatly improves the linearity and response of the amplifier and reduces the source impedance looking back into the output terminals to the

unbelievably low figure of about  $\frac{1}{3}$  of an ohm on the 16-ohm output tap. When this amplifier is used with highly damped, high-efficiency speakers, the clarity of reproduction is more than satisfactory.

In connection with source impedance, there are two schools of thought. One group holds that the source impedance should be made as low as possible by using triode tubes, by the use of inverse feedback, or possibly both. The other group maintains that the source impedance should be low, but not necessarily as small as possible—their claims being based upon listening tests. This second group feels that a representative value of source impedance should be from 0.5 to 1.0 times the load impedance; thus on a 16-ohm tap the source impedance would be between 8 and 16 ohms. This amplifier has been constructed so that the source impe-

dance is very low and those persons who prefer a slightly higher value may increase the feedback resistor (R-21) to approximately 22,000 ohms, or they may insert a resistor in series with the speaker voice coil should they prefer to retain the effects of the high feedback in this amplifier.

The power supply, Fig. 3, is built on a separate chassis to eliminate hum and also to provide a more compact construction, which simplifies installation. It is conventional with the exception that high grade 600-volt oil-filled capacitors are used in the filter. The recommended power transformer is designed for continuous operation with a drain of 200 ma in the plate supply. In practice it is found that after the amplifier has been on all day the transformer is just moderately warm.

#### Construction Details

Construction of this amplifier is simple. The amplifier and power supply are both placed on a 5 x 10 x 3 chassis; there is plenty of room for parts with complete accessibility for service should it be required. All resistors and capacitors are mounted on a terminal strip which is mounted on one side of the chassis as shown in Fig. 4. This type of construction, long used by professional amplifier builders, has been much neglected by amateur constructors and its use is encouraged to provide neatness as well as ease of construction and repair.

In order to reduce hum and undesired coupling between stages to a minimum, a grounding bus is used. This begins in the amplifier where the B- from the power supply enters the chassis, connects to the grounded side (terminal 12) of the output transformer secondary and then is carried along on the terminal strip toward the input, picking up cathode circuits and filter capacitor cans as it goes. It is grounded to the chassis at only one point near the input connector. Use of a grounding bus in amplifiers is also well known to professionals, and a large part of the

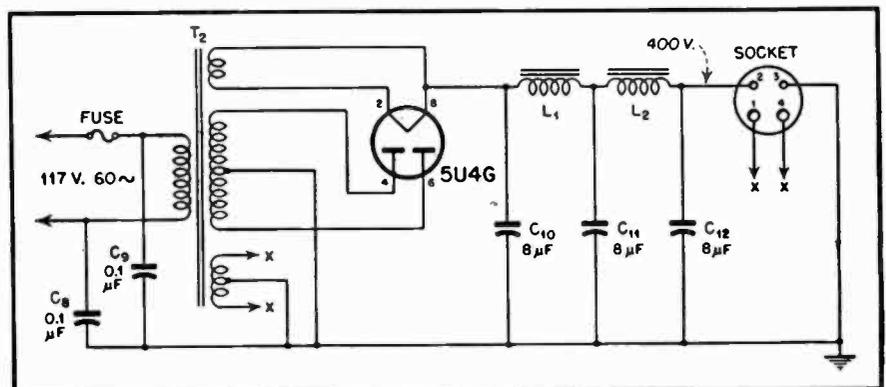


Fig. 3. Schematic of the power supply.

difficulties encountered by amateur constructors is due to haphazard ground connections. The layout of parts for the power supply is shown in Fig. 5.

At this point it might be well to review a few of the beliefs held by amateur constructors of amplifiers. Many amplifier designs which have appeared in magazines have had their virtues extolled solely on the basis of frequency measurements. The measurement of frequency response is an important property of an amplifier, but it is not the *only* property which contributes to naturalness in the reproduced sound. Frequency response is a measure of how the gain of the amplifier changes as a function of frequency. If, for example, the absolute gain is 75 db at 1,000 cps and 70 db at 10,000 cps, then the amplifier is said to be down 5 db at 10,000 cps. For this reason frequency response measurements are usually made at low output levels.

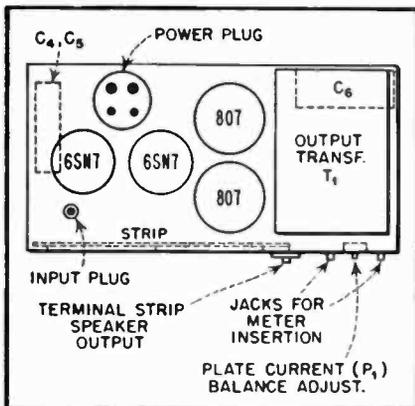


Fig. 4. Parts layout of the amplifier on a 5 x 10 chassis.

#### Power Output vs. Frequency

One important property of an amplifier which has long been neglected in the previous literature on the subject is power output at various frequencies. Magazine articles and manufacturers' literature will state that an amplifier will put out "10 watts." Sometimes they state the frequency and the degree of distortion that was present when this measurement was made. The importance of power output over the audible spectrum is so great that it cannot be over-emphasized, and it is this factor which is to a large degree responsible for naturalness of reproduction and freedom from distortion. Full power output over a wide band of frequencies is largely determined by the output transformer, the most important component in an audio amplifier.

The manufacturers of output transformers more often than not fail to make any statement as to the power output of amplifiers using their transformers at various frequencies. Regarding the transformer we used, the manu-

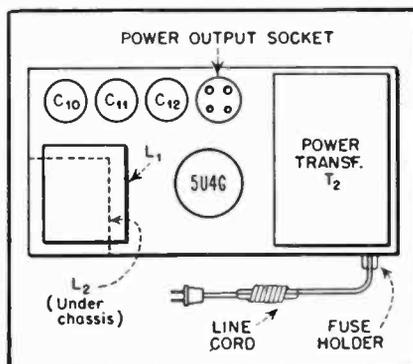


Fig. 5. Parts layout for power supply.

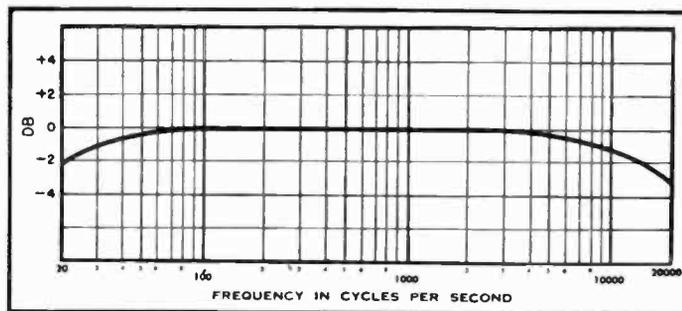
facturer states that the output will be down no more than 3 db from midrange power at 20 cps and at 20,000 cps. Audio enthusiasts who make measurements of power at various frequencies from 20 to 20,000 cps would be amazed to see the very poor power characteristic of a favorite amplifier.

#### Performance

The absolute gain of this amplifier, with about 20 db of feedback, is 70.8 db. The frequency response was measured by feeding in a signal at constant level from an audio oscillator, the signal being fed through a series resistor of 500,000 ohms which is equal to the input impedance of the amplifier. This is the customary way of measuring amplifiers professionally. The output was measured across a 16-ohm non-inductive resistor and high grade meters were used which had a minimum of frequency error. Under these conditions the amplifier is flat from 20 to 80,000 cps. There is a rise of 3.4 db at 96,000 cps and the output begins to fall at about 100,000 cps. In the revised article, Mr. Williamson states that his amplifier is flat from 20 to 100,000 cps, but he does not state the manner in which the response was measured.

The amplifier performs exceptionally well when tested with square waves. This test consists of feeding square waves into the input and observing the wave form on an oscilloscope at various points throughout the circuit. If the output waveforms have sharp corners and no tilt on top and bottom the following things can be said about the amplifier:

Fig. 6. Relative power output with respect to frequency.



1. The frequency response is flat to at least 20 times the fundamental frequency of the square waves.
2. There is no time delay; i.e., the phase shift through the amplifier is proportional to frequency.
3. There is no spurious oscillation produced by steep-wave-front signals, such as the usual music and speech programs.

Square wave testing is a sensitive method of testing amplifiers, especially with regard to phase shift, and it was found that this amplifier passed without distortion square waves having fundamental frequencies from 20 cps to 5,000 cps. These signals were passed through the entire amplifier and not just one stage. This confirms the frequency-response measurements and showed that there is uniform time delay for all frequencies between 20 and 100,000 cps. It is entirely possible that the startling realism in reproduced music is due to the excellent phase characteristics of the amplifier, as a non-

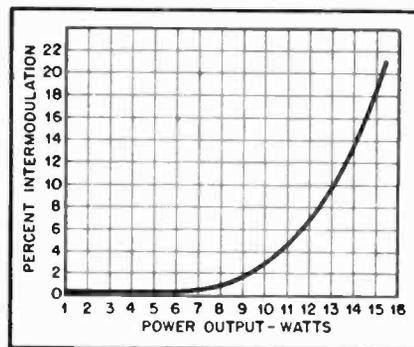


Fig. 7. Intermodulation distortion curve plotted against power output.

linear phase characteristic disturbs the arrangement of harmonics in complicated signals such as program material.

Mr. Williamson does not state the power characteristics of his amplifier, but does say that the power output is 15 watts. As our output tubes are not exactly the same as the KT66 valves originally used, we cannot make accurate comparison, but with 807 tubes and a 400-volt plate supply it will deliver 12.2 watts to the load resistor at 1,000 cps at 7 per cent intermodulation distortion. As may be seen from Fig. 6, the power begins falling off at low frequencies and is approximately 0.7 db down at 40 cps and 2.1 db down at 20

cps. It is down 2.1 db from full power at 15,000 cps and down 3 db at 20,000 cps. This power curve on the surface may appear to be excessive, but tests run by the authors on several well-known makes of audio amplifiers have shown the power to be down as much as 20 db at 20 and 20,000 cps. The overall intermodulation distortion curve is presented in Fig. 7.

On the surface it may appear that 12.2 watts is rather poor output from two 807 tubes, but it must be remembered that the 807's are triode-connected and the efficiency of triodes is much less than that of tetrodes. Moreover, the power output compares very favorably with that from type 2A3 or 6-volt equivalent triodes. The power ratings given in the tube manual for these types are the power developed in a load resistor connected plate to plate, while our power measurement was of useful secondary power. Those who have built "10-watt amplifiers" using 2A3 or equivalent tubes might find it very enlightening to measure the undistorted power in a resistor connected in place of their speaker. We have found that the power of 12 useful watts is more than enough for home volume, even with a 10-db safety factor.

The intermodulation distortion in this amplifier was checked on Altec Lansing intermodulation equipment, using frequencies of 40 and 2,000 cps. It was found to be extremely low at powers up to about 8 watts. As a matter of fact, the IM distortion for ordinary room volume powers was less than .2% which is entirely negligible. As pointed out in AUDIO ENGINEERING for September 1948, the power read on the IM equipment may be converted to equivalent single-frequency power, the usual way of rating an amplifier, by multiplying the IM meter power by the fac-

tor 1.47. This has been done in the published curve and it will be noted that the amplifier does not begin to exhibit serious distortion until sine wave power of 12.5 to 13 watts is reached. Thus the power curve for various frequencies, as reproduced in this article, is for substantially distortionless reproduction. If the reader wishes to be conservative in his amplifier performance, the power output as read from the curve should be divided by the factor 1.47 in order to get the IM meter power for a given amount of distortion.

When this amplifier is connected to a wide range speaker system and fed with good program material (e.g., local, live FM transmission) the resulting realism is so startling that it must be heard to be appreciated. The intermodulation distortion is so low that when a full symphony orchestra is playing, the various instruments are easily detected. The thud of the bass drum and the tinkle of the triangle are reproduced naturally at low levels, together with other sounds, even the squeaking of chairs and shuffling of feet. One very prominent musician, who is a composer, arranger and conductor and in whose home one of these amplifiers has been installed, said: "You not only can hear the difference between the flute and clarinet, but you can also separate the first and second clarinets when they play together".

#### Controls

It will be noted that the circuit diagram does not show any gain control. It is the authors' contention that an ideal amplifier should be one which can be tucked in some corner and forgotten while gain control and equalization should be accomplished in a separate control unit. The equalization and gain controls together with a pho-

former. It is suggested that a 0.25-meg. potentiometer be substituted for  $R_{21}$  at the time of final checking, and that the output of the amplifier be measured with the total resistance in circuit, and then again as the resistance is decreased gradually. If the output increases, it will be necessary to reverse either the primary or secondary leads to the output transformer. Then the fixed value for  $R_{21}$  may be installed. For other voice-coil impedances,  $Z$ , the value for  $R_{21}$  will be approximately  $1250\sqrt{Z}$ ,

nograph preamplifier may well be a part of an FM tuner. If it is desired, however, to incorporate a gain control, the 0.5-meg grid resistor of the input stage may be replaced with a 0.5-meg left-hand taper volume control. No equalization (tone controls) should be added to this amplifier, as the feedback loop goes from the output transformer to the first stage, and the large amount of feedback will negate any equalization introduced within the feedback loop. If equalization is desired it must be incorporated before the volume control. There is such power available at both the bass and treble end so that when the bass and/or treble boost is used, the results are positive and free from distortion at all comfortable levels. When bass boost is added the foundation music is solid and free from thuds.

To a couple of fiddlers—one who fiddles all day on a Stradivarius, the other who fiddles all day with amplifiers, this amplifier is the best that we have yet heard. We suggest that you build one and judge for yourself.

#### PARTS LIST

- C<sub>1</sub>, C<sub>7</sub>—0.25  $\mu$ f, 400 v. paper.
- C<sub>2</sub>, C<sub>3</sub>—.05  $\mu$ f, 600 v. paper.
- C<sub>4</sub>, C<sub>5</sub>—10  $\mu$ f, 450 v. electrolytic.
- C<sub>6</sub>—30  $\mu$ f, 500 v. electrolytic.
- J<sub>1</sub>, J<sub>2</sub>—Closed circuit jack, Mallory A-2 (Must be insulated from chassis).
- J<sub>3</sub>—Input connector, Amphenol 75-PC1M (R<sub>1</sub> is mounted inside this connector).
- J<sub>4</sub>—Power cable connector, Amphenol 86-RCP4 (4-pin male chassis plug).
- P<sub>1</sub>—100 ohm, 2W. wire wound potentiometer.
- R<sub>1</sub>, R<sub>6</sub>, R<sub>7</sub>—0.47 meg, 1/2 W.
- R<sub>2</sub>—47,000 ohms.
- R<sub>3</sub>—470 ohms, 1/2 W. 10%.
- R<sub>4</sub>, R<sub>5</sub>—22,000 ohms, 10%, (matched).
- R<sub>8</sub>—390 ohms, 1/2 W. 10%.
- R<sub>9</sub>, R<sub>10</sub>—47,000 ohms, 2 W. 10% (matched)
- R<sub>11</sub>, R<sub>12</sub>—0.1 meg, 1/2 W.
- R<sub>13</sub>, R<sub>14</sub>, R<sub>22</sub>, R<sub>23</sub>—100 ohms.
- R<sub>15</sub>—250 ohms, 10 W. wire wound.
- R<sub>16</sub>, R<sub>17</sub>—1000 ohms, 1/2 W.
- R<sub>18</sub>—27,000 ohms.
- R<sub>19</sub>—22,000 ohms.
- R<sub>20</sub>—150 ohms, 10 W. wire wound.
- R<sub>21</sub>—5000 ohms, 10%.
- T<sub>1</sub>—Peerless S-265-Q.

#### Power Supply

- C<sub>8</sub>, C<sub>9</sub>—0.1  $\mu$ f, 600 v. paper.
- C<sub>10</sub>, C<sub>11</sub>, C<sub>12</sub>—8  $\mu$ f, 600 v. oil filled.
- F<sub>1</sub>—Littelfuse 342001 fuse extractor post with 3AG fuse, 2 amps.
- J<sub>5</sub>—Power cord socket, Amphenol 77MIP4 (4-hole socket).
- L<sub>1</sub>, L<sub>2</sub>—Peerless C-325-A. 10 H, 120 ma, d.c. resistance 240 ohms.
- T<sub>2</sub>—Peerless R-560-A. 800 v. CT at 200 ma; 5 v at 3 a; 6.3 v at 6 a.

#### Power Cable

- 1—Female connector, Amphenol 78-PF4.
- 1—Male connector, Amphenol 86-PM4.
- 4-wire cable, of required length.

All resistors are 1-watt, 20% tolerance, unless otherwise specified.

#### Additional Notes

While no guarantee of performance can be offered when transformer substitutions are made, many constructors have assembled this amplifier with transformers other than those specified, and have reported good results. The output transformer must be an exceptionally good one, and those mentioned are thought to be of sufficiently high quality to permit this listing.

Care should be exercised to obtain the right polarity in the output trans-

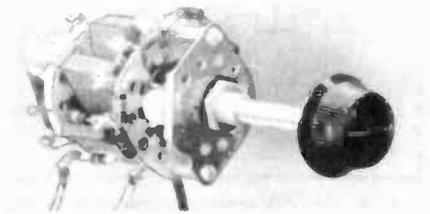
#### POSSIBLE TRANSFORMER SUBSTITUTIONS

	T <sub>1</sub>	T <sub>2</sub>	L <sub>1</sub> , L <sub>2</sub>
Acro	TO-290		
Freed	F-1959		
Partridge	"Williamson 807"		
UTC	LS-6L3	S-39	S-29

# Loudness Control for Reproducing Systems

DAVID C. BOMBERGER

By using the device described, it is possible to maintain a close tonal balance over a wide variation in output level.



Complete loudness control unit.

A NUMBER of different methods have been utilized to control the output of reproducing systems in such a manner that the tonal balance is maintained reasonably constant at all intensity levels. These controls are essentially variable equalizers which modify the gain-frequency characteristic of the system as the intensity is changed; the desired equalization being specified by the intensity vs. loudness characteristics of the human ear. An ordinary resistive potentiometer is actually a simple intensity control, while a properly equalized intensity control might be termed a loudness control. To maintain constant tonal balance as the intensity is changed, the intensity at low frequencies must be changed less than that at high frequencies. The consequence of changing the intensity equally at all frequencies has been experienced by anyone who has noted the apparent lack of bass in the average radio at low volume settings.

A common type of loudness control employs a tapped potentiometer, with a capacitor and resistor in series between the tapped point and ground. This rather elementary structure is a large step in the right direction, and is attractive because of its simplicity. This very simplicity, however, renders it capable of yielding only a moderate approximation to the ideal. The loudness control to be described is an elaboration of the tapped potentiometer, and was designed for those applications in which a considerably closer approximation to the ideal is desired, even though some extra cost is involved.

## Fletcher-Munson Curves

The basis for design of a loudness control is the set of curves shown in Fig. 1. These are the well-known data of Fletcher and Munson, and are the averaged results of measurements taken with a large number of individuals. Each curve represents a particular loudness, measured in decibels from a reference level; the ordinates of the curve

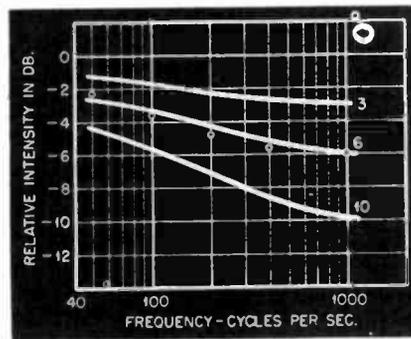


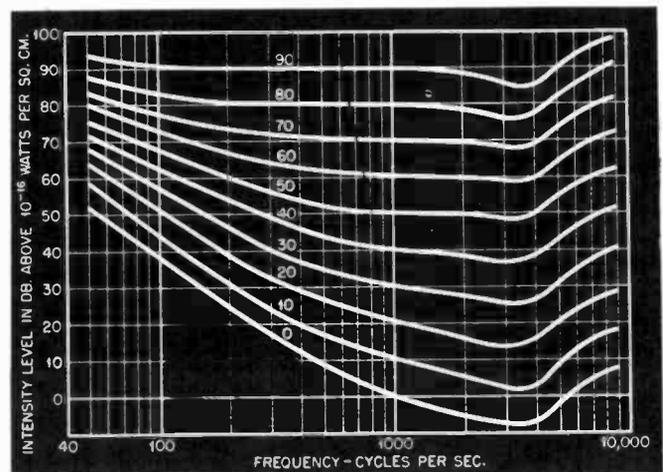
Fig. 2. Curves showing the frequency characteristic of each network section at various intensity levels.

show the intensity in decibels corresponding to that loudness. The departure of these curves from horizontal lines spaced 10 db apart on the intensity scale represents the need for a loudness control. It was concluded that the departure at frequencies above 1000 cps was relatively unimportant, and that only the low-frequency end would be considered. This leads to an appreciable simplification of the problem. Nevertheless, a single network to produce a large loudness change would require a rather complex array of elements because of

the rapid change of intensity with frequency. A large loudness change can more readily be built up by the addition of a number of smaller changes, each having an appropriate intensity vs. frequency characteristic. This procedure is facilitated by the fact that the intensity differences between adjacent loudness curves are quite similar. An excellent approximation to the ideal may be realized by a control which inserts successive units of loss, each similar to the other, and having a loss-frequency characteristic proportional to the average intensity differences between loudness curves. These averages are presented in Fig. 2 as gain-frequency characteristics for 10 db, 6 db, and 3 db loudness intervals. The 3 db interval was chosen for design; it has been found that this increment is sufficiently small for almost all applications where only the listener's reaction need be considered.

It is evident, now, that the loudness control may take the form of a switching device which inserts, successively, a suitable number of identical network sections somewhere in the reproducing system. These sections are designed, on an image impedance basis, to match the characteristic of the 3 db loudness change

Fig. 1. Fletcher-Munson curves.



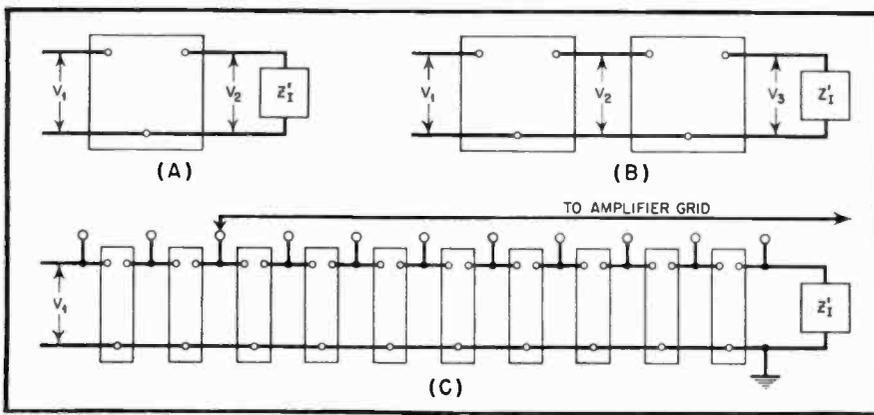


Fig. 3. In (A), representation of network. In (B), two networks in parallel, and (C), ten section in tandem.

of Fig. 2, and are inserted between proper terminating impedances. As many of these sections may be placed in tandem as are required to produce the desired loudness change. While this control would be satisfactory in its performance, the switching mechanism would be rather unwieldy. A somewhat simpler method of attaining the same end is available.

### Transfer Constant

The transfer constant of a network designed on an image impedance basis defines the complex ratio between the voltages across the input and output terminals of the network, when the output is terminated in the proper image impedance. Thus, in Fig. 3-A, the voltage  $V_1$  on the input terminals is related to the voltage  $V_2$  on the output terminals by the equation

$$\frac{V_1}{V_2} = \epsilon^\theta$$

In this equation  $\theta$  is the complex transfer constant whose real part is the loss of the network in nepers. If two identical networks are connected in tandem, as in Fig. 3-B, then three voltages exist, related as follows:

$$\frac{V_1}{V_2} = \epsilon^\theta, \quad \frac{V_2}{V_3} = \epsilon^\theta, \quad \frac{V_1}{V_3} = \epsilon^{2\theta}$$

Thus the ratio of voltages between input and output of this two-section network is defined by  $2\theta$ , and the loss in

nepers (or in decibels) at any frequency is just twice that of one section. Consider, now, ten sections in tandem with an appropriate terminating image impedance  $Z'_I$ , as in Fig. 3-C. Let the input to the network be the voltage output,  $V_1$ , of a vacuum tube amplifier. Let an eleven-position switch connect the grid of a second vacuum tube amplifier to any of the eleven connection points of the chain. If each network section has a transfer constant such that its transmission is represented by the 3 db curve of Fig. 2, and the terminating impedance is the proper image impedance for that network section, this structure will be a loudness control with 30 db total loudness change, in 3 db steps.

### Network Section

A satisfactory network section is shown in Fig. 4-A. The transmission of this section can be made to match the desired characteristic quite closely. It would, however, require ten series resistors,  $R_1$ , eleven shunt arm resistors,  $R_2$ , and eleven shunt arm capacitors,  $C_2$ , to construct the loudness control of Fig. 4-B. The number of shunt arms with their expensive capacitors may be halved by using, instead of ten 3 db sections, five 6 db sections each divided into two parts. Unfortunately, the voltage in the middle of a section is not related to that at the ends by half the transfer constant; still, the section can be divided in such a manner that exactly half the high frequency loss exists across each portion with only a minor distortion of

the low-frequency loss. The exact manner in which the section is divided depends on the image impedance of the section, which in turn requires that the element values be specified. Figure 5-A presents the design of a section which has the calculated full-section transmission shown by the circles on Fig. 2. Figure 5-B shows the complete loudness control including the terminating network. The exact image impedance can only be obtained by an infinite number of additional sections, but the elements shown are entirely adequate. The impedance level of the control is determined when  $R_I$  is chosen. This choice will be influenced by the effects of parasitic capacitance in the switching, the desirability of using RMA values for the elements, and the amplifier to be used as a drive for the control.

### Amplifier Problems

The problem of the amplifier can be analyzed as follows: consider first that the amplifier has zero internal resistance. The removal of the first shunt arm (shown connected by dotted lines in Fig. 5-B) will not affect the voltage at the input to the network, nor the transmission through it. The load presented to the amplifier is now made up of the first two series resistors, which total  $R_1$ , plus the balance of the network. This remainder has an impedance which varies from a very high value at low frequencies to  $R_I$  at frequencies above 1000 cps.

When the amplifier has a non-zero internal resistance, two alternatives are available. If  $R_I$  may reasonably be made ten times the magnitude of the amplifier resistance, the amplifier can be considered to have effectively zero resistance, and the network may be reduced by leaving off the first shunt arm. An attempt at this procedure may lead to a value of  $R_I$  which is too large for the parasitic capacitances that will be present in the switching. In this event,  $R_I$  may be adjusted so that the amplifier resistance is any convenient fraction of  $R_I$ . The network is now augmented: the first shunt arm is made the same as the others, and the difference between  $R_I$  and the amplifier resistance is inserted between the amplifier and network input. In the special case of an

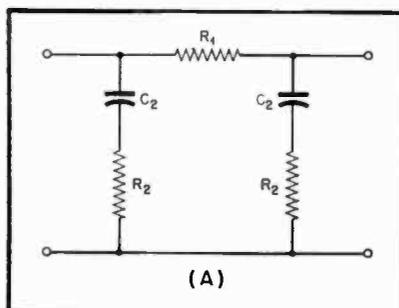


Fig. 4A. Schematic of network section.

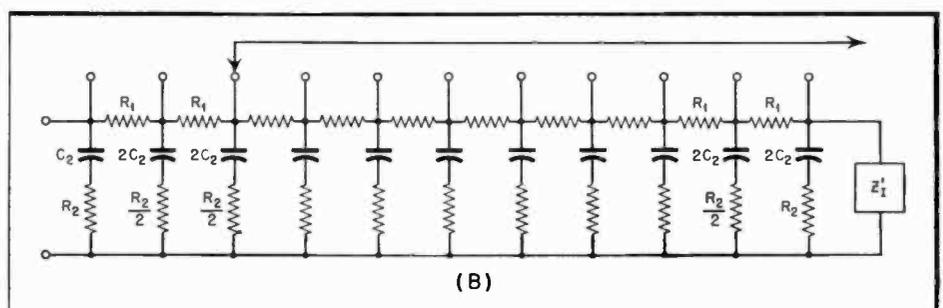


Fig. 4B. Schematic of complete loudness control using the network design of Fig. 4A.

Fig. 5A. Design of section calculated to provide the characteristic shown by circles in Fig. 2.

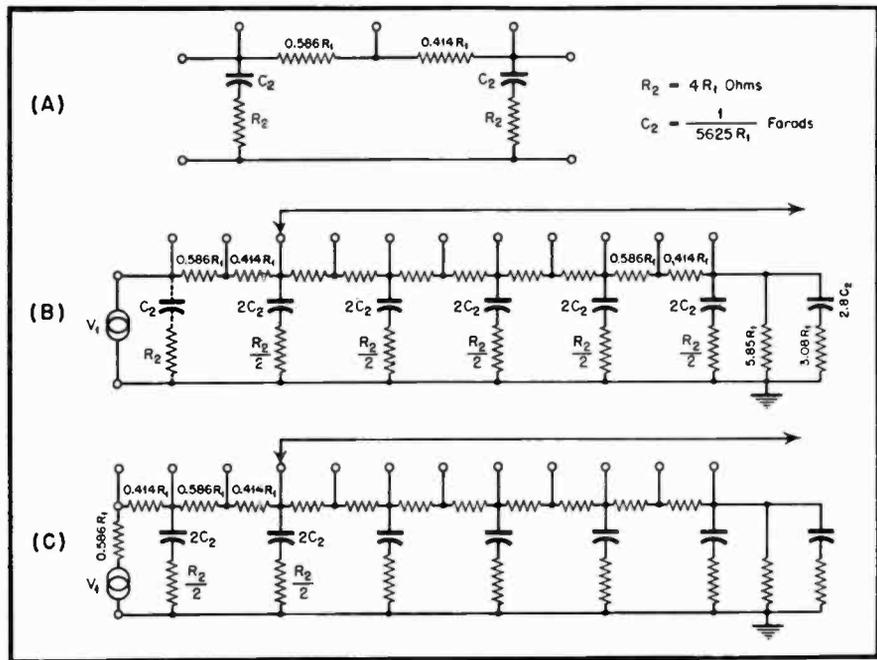
Fig. 5B. Complete loudness control, including terminating network.

Fig. 5C. Arrangement when the amplifier resistance is  $0.586R_1$ .

amplifier resistance equal to  $0.586 R_1$ , the added resistance is  $0.414 R_1$ , and an extra loudness interval is available at the amplifier terminals. This arrangement is illustrated in Fig. 5-C; it will be seen that one 3 db loudness interval is inside the amplifier, and so cannot be switched out. If  $R_1$  were adjusted to equal the amplifier resistance, the added resistance is zero; in this case, there is a 6 db loudness interval inside the amplifier.

The network elements may be mounted on a two-deck wafer type switch with shorting contacts. One deck is used for switching, and has the resistors of the series arms mounted between terminals. The capacitors of the shunt arm are mounted between corresponding terminals of the two decks. Shunt arm resistors and terminating elements are then arranged on the circular area of the second deck with a ground termination in the center. The moving arm of this deck should be removed. Constructed in this fashion, the loudness control does not require an excessive mounting space.

A representative network is one which has the basic element values:



$R_1$	50,000 ohms
$R_2$	200,000 ohms
$C_2$	$0.00355 \mu f$

Values which approximate the individual network components are:

$.586 R_1$	30,000 ohms
$.414 R_1$	20,000 ohms
$.500 R_2$	100,000 ohms
$2 C_2$	$.0068 \mu f$
$5.85 R_1$	300,000 ohms
$3.08 R_1$	150,000 ohms
$2.8 C_2$	$.010 \mu f$

A loudness control built with these elements, and driven from a low impedance source, has the measured characteristics shown in Fig. 6. The several approximations made in the design, as well as the element deviations, have but small effect on the over-all performance.

It will be recognized that this control is accurate only in producing appropriate changes in intensity. To be correct on an absolute basis, each program to be reproduced should be adjusted by a resistive control elsewhere in the system so that with the loudness control at the top position, the acoustic intensity is equal to that of the original sound.

This adjustment is rather impractical. Nevertheless, when the loudness control is simply used as a replacement for the ordinary resistive volume control, quite gratifying results are obtained. The most conclusive evidence of the superiority of this loudness control over flat intensity control is that a low level of intensity in the reproduction of music is as enjoyable as the higher level usually required for good tonal balance.

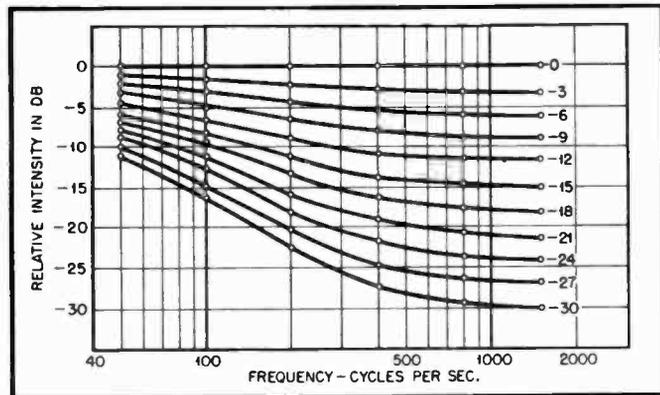


Fig. 6. Measured characteristics of loudness control constructed from data supplied in this article. It closely approximates hearing curves.

# Full-Range Loudness Control

This simplified loudness control uses commercially available components and may be assembled in an hour or so.

**S**INCE the publication of David Bomberger's "Loudness Control for Reproducing Systems"<sup>1</sup> many readers have expressed considerable interest in this device, mainly with a desire for a control having a larger number of steps so as to be more suitable for use as the only volume control in a radio and phonograph

installation. Such a unit was described in the series on "Residence Radio Systems"<sup>2</sup> but no data were given other than the schematic of the particular network employed. The photograph showed an enclosed control, using the frame of an IRC attenuator.

This control is not readily avail-

able in jobbers' stores, and the construction of a loudness control on this frame is rather a difficult procedure, so the arrangement described herein was assembled in an effort to reduce the work involved, and to use

<sup>1</sup> See pages 37-39.

<sup>2</sup> See pages 106-122.



switch deck being necessary in addition to the other parts listed.

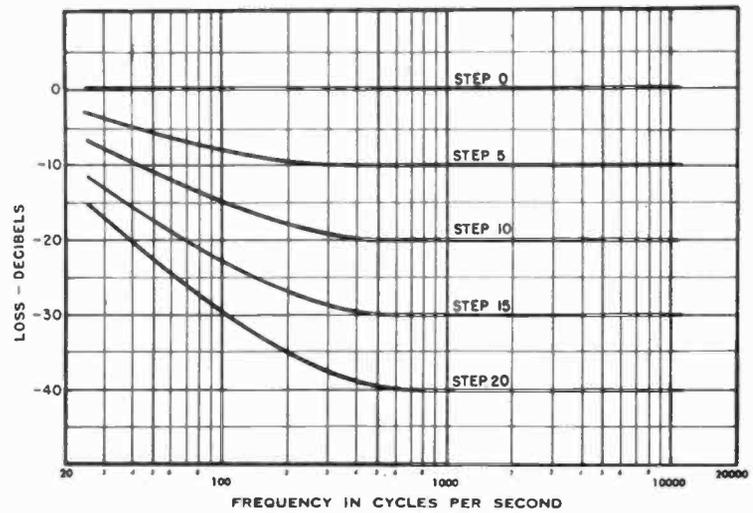
The first step in the construction of the control is to remove the rotor from the "B" switch deck. To do this, break out the thin Bakelite retaining disc on one side, and carefully work the contact arm out of the lugs so as to avoid damaging them. This deck is used only to mount the seven capacitors, with the resistors connecting to the potentiometer at the correct points.

Next, turn the 1443 control to maximum clockwise position and note the position of the contact arm. Calling this point "0" count around in a clockwise direction (viewed from the back) and make a scribe mark on points 4, 7, 10, 13, 16, and 19. Now, solder a 0.15-meg resistor to each of these marked points, with the leads through the eyelets holding the lugs, and allowing only about 1/16-in. between the resistor body and the eyelet. Attach the 0.22-meg. resistor to the lug of point 19 as closely as the others, but do not solder. At this point, the switch deck will be shown in Fig. 5.

The two screws holding the switch together are now removed—one at a time—and replaced with the longer ones, with a 1/2-in. spacer between the 1443 switch deck and the de-rotored "B" deck. The resistor leads should be fed through convenient lugs on the latter as it is put in place so as to avoid bending the short ends of the resistor leads, as would be necessary if the connections were made later. After the new deck is mounted and the nuts firmly tightened, the remaining 22 resistors may be mounted.

In starting the first of these connections, the wiring may appear to be a bit tricky, but a little experience will make it quite simple. Bend one lead of each resistor back 180 deg. around the tip of long-nosed pliers. The two leads may then be easily inserted through the lugs—one resistor from one side, the next from the other, and so on. When all the resistors are mounted and soldered, including the soldering of the 0.22-meg. resistor on point 19, the six capacitors are mounted on the "B" deck. Three unused lugs are employed for the ground connection, as shown in Fig. 6. These grounds should be strapped to the 23rd lug on the main deck, leaving sufficient wire for making the connection to the ground circuit of the amplifier. The finished control should then resemble that shown in Fig 1(A). Input connections to the control are

Fig. 4. Response curves for steps 0, 5, 10, 15 and 20.



made at points 0 and 23, and the output connections to the arm lug and point 23, the latter being the common or grounded side.

#### Installation

The loudness control may be installed between the existing volume control of any amplifier and the following grid, or it may be installed in place of the present control. It is necessary, however, that the overall gain be adjusted so that normal room volume is at about step 4 of the control. This allows some 8 or 10 db above normal for demonstration purposes, or for those occasions when such an increase is desirable, and some 40 db of range below the normal for late evening listening or for background use.

It will take some time to become accustomed to this control because the average listener fully expects the quality to change as the level is raised or lowered. Since no apparent change in *quality* is encountered, it may seem that the level is not changed, but after a week's use it is

almost certain that the listener would not go back to a simple volume control. Even at very low levels, the quality remains constant, giving real listening enjoyment with only one control.

It will be observed that the switch has discrete steps, with the detent action. If this should prove objectionable, the detent may be removed by spreading the flat retaining spring out slightly and allowing the steel ball to drop out, thus giving a continuous smooth action. In addition, this permits the shorting-type arm to shunt two contacts, reducing the level difference between points. Personally, we prefer the step-type control, but it is purely a matter of choice.

#### The Decompensator

On speech, particularly male, the loudness control may appear to be too bassy, and some means for reducing the compensation may be desired. The logical method for this is to short out the compensating capacitors. (Concluded on bottom of next page)

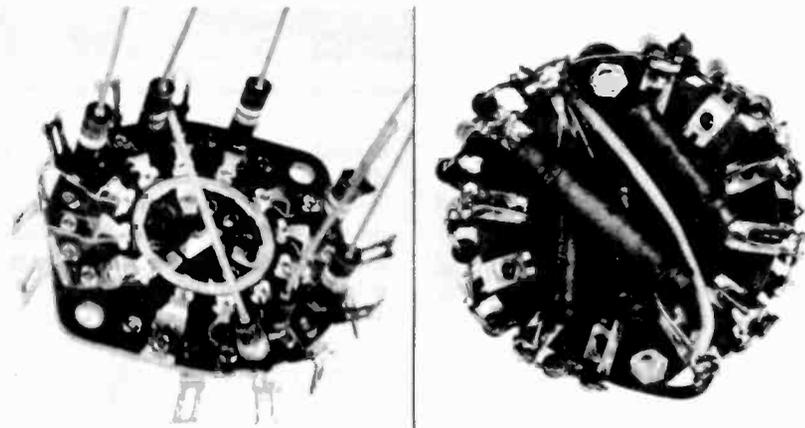


Fig. 5 (left). First step in connecting the resistors to the 1443 switch deck. The 0.15-meg shunt resistors are mounted first, as shown. Fig. 6 (right). Method of mounting the compensating capacitors on the de-rotored "B" switch deck.

# Flexible Dual Control System

HOWARD T. STERLING

Describing a continuously variable R-C system providing boost and cut for both treble and bass, without the use of inductances or feedback.

**I**N RECENT YEARS the term "tone control" has become anathema in the ears of self-respecting engineers. There has been good reason for this—usage in the field has generally been such that wherever a tone control has been used it has resulted in deterioration of quality.

The conventional treble control, as used in most inexpensive systems, acts to depress the whole high-fre-

quency range, usually above 1000 cps or so. The bass control is too often a simple variation of this which provides a rising bass characteristic by reducing the relative amplitude of the highs, and usually affects the over-all volume level. Some manufacturers, desiring somewhat more effective control, often resort to a system involving selective inverse feedback. Unfortunately, in order to

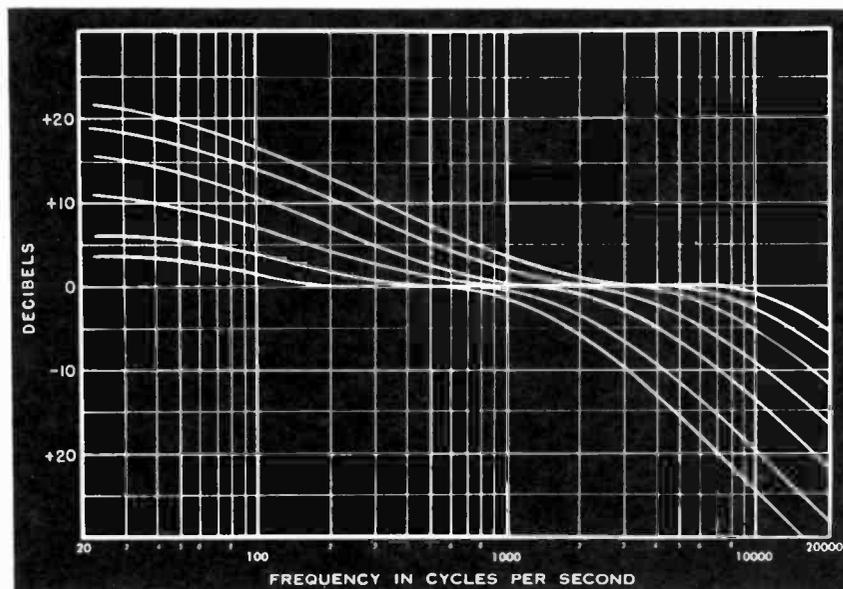
obtain significant results by this method it is usually necessary to include more than one R-C section in the feedback loop, with attendant regeneration and transient distortion. It is small wonder that with methods such as these the tone control has fallen into ill repute.

## Step-Type Controls

A more reasonable approach to the problem has been made by some of the manufacturers of so-called "professional" equipment. There the usual solution lies in the use of "step-type" treble and bass controls. It is possible by this means to get much more satisfactory performance, and typical curves (*Fig. 1*) indicate a good choice of treble cutoff and bass boost. The selling point on this method is usually that ". . . This is the type of control used in commercial and broadcast equipment." Unfortunately, most users of this system find the step-type of adjustment exceedingly annoying, and would much prefer that tone control be continuously variable.

It can be further demonstrated that it is desirable to be able to cut the lows as well as to boost them, and to boost the highs as well as to cut them. Often, particularly on AM, program material comes through with a muddy quality, and reproduction may be rendered much more pleasing and intelligible if the low-

Fig. 1. Typical characteristics of step-type bass and treble controls from "professional" amplifier.



## LOUDNESS CONTROL

(Continued from previous page)

itors progressively from the high end down. This may be accomplished with the "G" switch deck, with the shorting arm cut in half, and with the normal arm removed. To make the ground connection, it is also necessary to short the grounding ring to the arm contact on the bottom of the rotor. This may be done by adding another rivet between the two rings. The modified switch is shown in *Fig. 1(B)*. In this photo, the rotor is shown in the maximum

clockwise position, corresponding to normal compensated operation. The six lugs opposite the rotor ring are connected to the junctions of the six 0.15-meg resistors and the .005- $\mu$ f capacitors, so that as the decompensator is turned counter-clockwise the capacitors are shorted to ground. In the position shown, all six contacts are open; in the second position the capacitor associated with point 4 is shorted; in the third position, the capacitors for points 4 and 7 are shorted; in the fourth those for points 4, 7, and 10 are shorted, and so on. The stop is set to provide a

total of seven positions, the .0075- $\mu$ f capacitor remaining in the circuit at all times, since the switch will accommodate only six shorting positions.

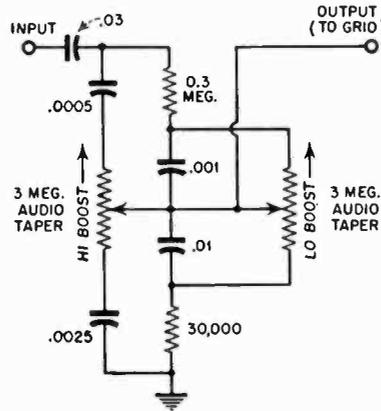
With the construction of the loudness control reduced to such a simple procedure, it is expected that many experimenters will find it worth the hour or so of work necessary to try it out because of the enjoyment certain to be obtained. The writer firmly believes that no listener will ever give up the loudness control after using it for a few weeks—it's that good.



# Low-Loss Tone Control

A simple tone control for both low- and high-frequencies which has a total loss of only 20 db.

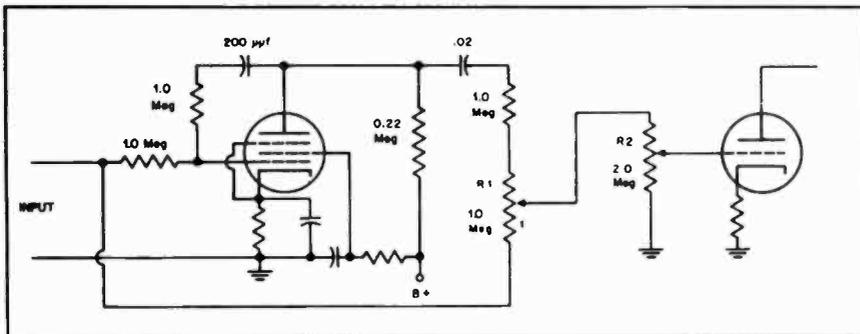
For some years the writer has been using the tone control system described in the preceding article, but with a modification which may be of interest because of a reduced loss. Both high- and low-frequency control circuits can be combined into one network with a mid-frequency loss of only 20 db, instead of the total loss of 40 db when the networks are used separately as described in the article. As a result, only one triode amplifier is needed to compensate for the loss introduced by the tone control.



The network is shown at left and the values shown give substantially the same response curves as shown in the article. The input should be connected to a source impedance not higher than about 20,000 ohms, to prevent loss at high frequencies when in the high boost position, and the output should work directly into a grid, as pointed out in the editorial note.

James J. Faran, Jr.,  
Acoustics Research Lab.,  
Harvard University.

## Feedback Equalizer



A single-tube amplifier circuit claimed to furnish sufficient equalization for magnetic phonograph pickups together with an appreciable amount of amplification is described by J. Ellis in *Wireless World* for September, 1947. The circuit, shown at left employs a single pentode with feedback to provide the desired frequency response.

Variation of the setting of  $R1$  changes the transition frequency by selecting desired proportions of the original signal and of the amplified signal with the boost on the low frequencies. If no volume control is required at  $R2$ , the arm of  $R1$  may be connected directly to the grid of the following stage, provided there is a d-c path to ground through the input device.

## DUAL CONTROL SYSTEM

(Continued from previous page)

a measured response characteristic, for various settings of the controls, as shown in Fig. 5. It will be seen, of course, that the resistor from the grid to ground of the first triode may be replaced with a volume control.

It must be understood that this system should only be used to supplement proper equalization of the various program sources, and in no way obviates the need for proper compensation in the phonograph for turnover and pre-emphasis, for normal de-emphasis on FM, or for an adequately compensated volume control.

Ed. Note: The circuits of Figs. 2 and 3 must work directly into a grid to be effective in the amount shown in the curves of Fig. 5.

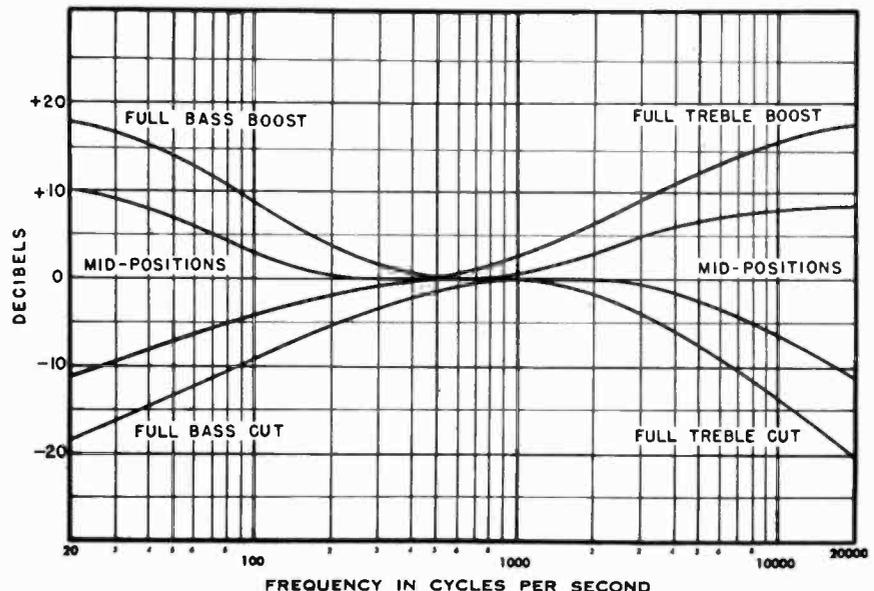


Fig. 5. Characteristics of dual tone control.

# Heater Supplies for Amplifier Hum Reduction

FREDERICK W. SMITH

A practical method for reducing hum to a minimum

**W**HEN hum is perceptible in the output of an amplifier, the listener notices a lack of "presence" in the reproduction of program material. This effect is especially objectionable where the background noise level of the signal is extremely low, as in frequency modulation reception.

The existence of hum in an amplifier usually may be attributed to three sources <sup>1,2,3</sup>:

- Insufficient plate supply filtering.
- Electrostatic or magnetic pickup in low level stages.
- The cathode heater circuits.

Of these three, the first is the easiest to eliminate, because little practical

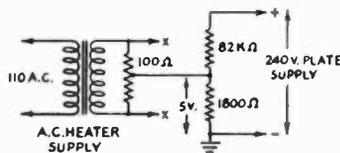


Fig. 1. Hum reduction method used in a program amplifier.

difficulty is experienced by the designer in reducing plate supply ripple below any predetermined value.

Electrostatic or magnetic pickup may be reduced by a variety of devices which include shielding, the isolation of power supplies from the amplifier proper, and the use of a non-magnetic amplifier chassis.

## Cathode Heater Hum

The remaining and most obstinate source of hum is that due to the cathode heater circuits. If the heaters of the tubes incorporated in an amplifier are supplied with alternating current of power-line frequency, a hum having both fundamental and second harmonic components will appear in the amplifier output. This hum is due to various causes. Direct sources naturally include hum pickup from heater circuit leads and hum leakage through socket capacitances. Less obvious, however, are such other sources as heater-to-cathode leakage, capacitive coupling between the heater and other elec-

trodes, or actual heater emission to other elements within the tubes.

A number of methods have been devised to eliminate some of these defects with varying success. A representative arrangement employed in a commercial program amplifier is illustrated in Fig. 1. In this circuit, a positive bias of approximately five volts is applied to the heater circuits of the amplifier stages through a hum-balancing potentiometer placed across the 6.3-volt heater supply transformer winding. The positive bias on the heaters prevents heater emission while the effects of capacitive coupling between the heaters and control grids are balanced out by proper adjustment of the potentiometer.

However, even with special precautions and care in the design of an

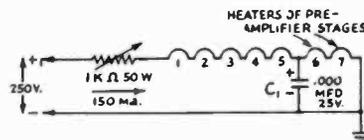


Fig. 2. Typical 150-ma d-c heater circuit.

amplifier stage having an a-c heater supply, there generally remains a residual hum amounting to from five to fifteen microvolts between grid and cathode, which is not negligible in a low-level stage.

The ultimate solution to the hum problem, therefore, is to use heater power sources that are either of the direct-current or the high frequency alternating current type. Substitution of either of these in the usual amplifier will result in a minimum reduction of five to ten db in hum level, and consequently this feature has been incorporated in many of the newer equipment designs.

The two types of heater excitation just mentioned may be classified in actual application according to whether a series or parallel heater connection is employed.

## Series Connection

With the advent of tube types having 150-ma heaters, it has been possible to employ conventional power supplies to furnish direct-current heater excitation. A typical circuit of this type described by Clark<sup>4</sup> is shown in Fig. 2. Here the output voltage of a power supply capable of furnishing sufficient current for both plate and heater circuits is applied to the series connected, 12 volt, 150-ma heaters of the amplifier tubes. Heater current is adjusted to the proper value by means of  $R_1$  and additional filtering is supplied for the heaters of the preamplifier stages by the inclusion of capacitor  $C_1$ , which is of the order of 1000  $\mu$ f.

A number of difficulties can be experienced with this type of circuit as follows: when a burnout occurs in the heater string it is difficult to determine rapidly which of the several heaters is defective. Also, in the event that either of the heaters on the ground side of capacitor  $C_1$  should open up,  $C_1$  will fail unless it is rated to accommodate the full supply voltage, thus endangering the remaining heaters in the string. It is also possible for this same filter capacitor to discharge itself through a new replacement tube, causing a second burnout, if it is not provided with a bleeder resistance.

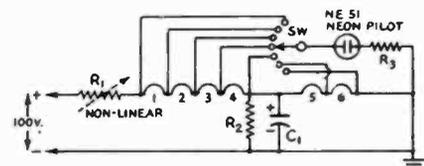


Fig. 3. Improved d-c heater supply circuit.

An improved circuit designed to avoid these difficulties is presented in Fig. 3. Here the series resistance,  $R_1$ , is a special non-linear type having a negative temperature coefficient such that initial current surges during the warmup period are eliminated. Non-linear resistors designed for this application are obtainable either from the

Keystone Carbon Co. (Type 701) or the Carborundum Corp. (Type F). A typical unit suitable for this purpose has a resistance of approximately 1400 ohms at room temperature and 200 ohms at its normal operating temperature of about 300° F<sup>5</sup>. The filter capacitor for the preamplifier stages, C<sub>1</sub>, is rated to accommodate the full supply voltage and is furnished a discharge path through R<sub>2</sub>. Finally, a simple voltmeter circuit consisting of a neon pilot lamp in series with a current-limiting resistor, R<sub>3</sub>, has been included to facilitate rapid checking of heater circuit operation. Should an "open" occur, the lamp will give no indication when switched to the ground side of the defective tube and will indicate the full supply voltage on the other.

#### Parallel Connection

The excitation of parallel-connected heater circuits naturally poses a problem in low voltage, high-current, power supply design. The heater current requirements of even a modest amplifier may be from 3 to 5 amperes, and only a rectifier of the selenium type can handle such a load efficiently. A suitable circuit employed by the auth-

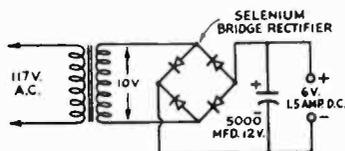


Fig. 4. Hum-free heater supply for parallel heaters.

or to eliminate hum in a recording amplifier is shown in Fig. 4. The output of a 10 volt rms filament transformer is applied to a bridge-type selenium rectifier, producing sufficient current in this case to excite all heaters except those of the power amplifier tubes. If the power supply and power amplifier are located on a chassis separate from the lower level stages, it is advisable not to employ a common ground return for both the heater and signal circuits, because any remaining hum component in the heater current will introduce a hum voltage in series with the signal which appears at the input of the power amplifier stage.

#### H-F Heater Supply

Because of space requirements, the type of supply just discussed may not be very convenient to install in existing equipment. A much neater solution to the problem of equipment modification for hum reduction may be found in the high-frequency alternating current type of heater supply. Such a supply may be compactly in-

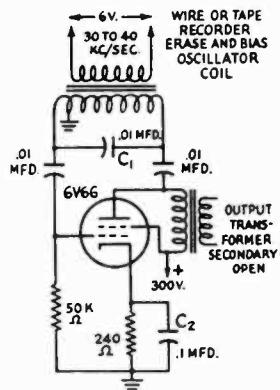


Fig. 5. High-frequency power supply for heater excitation.

stalled in a chassis space of 2" by 2" and requires but 40 ma of plate current and .45 amperes of heater current.

The circuit itself, as illustrated in Fig. 5, is based on the high-frequency oscillator circuits currently employed in magnetic tape or wire recorders to supply supersonic bias and erase current. Such a supply will furnish up to two amperes at six volts, depending on the plate voltage applied to the oscillator, and generally operates at a frequency of from 30 to 40 kc/sec.

Best results will be obtained if C<sub>1</sub>, the oscillator tank capacitor, is of the mica variety, and it should be emphasized that the cathode bypass, C<sub>2</sub>, must not be omitted, since the resulting degeneration will make the oscillator difficult to start and cause its regulation to be poor. The waveform produced by this type of circuit tends to be rather impure, and some difficulties with harmonic radiation may be experienced where such a stage is installed near or on a radio receiver or video amplifier chassis.

#### Regulated Supplies

In a number of specialized applications, not only is it necessary to employ d-c heater supplies, but in addition, all heater voltages must be closely regulated. Such instances arise in the design of the direct-coupled amplifiers associated with certain types of electroencephalographs, spectrophotometers, and apparatus for nerve potential studies.

In these cases, regulation of heater potentials serves to eliminate cathode drift resulting from the variation of electron emission velocities with cathode temperature. The necessity for regulation in these applications can be appreciated from the fact that for a typical tube type, the heater voltage must be held constant within 1% if an effective voltage fluctuation of 10 millivolts in the cathode circuit is to be avoided.<sup>6</sup>

The heater circuits employed may again be either series or parallel, depending upon the type of regulated source which is available.

For the series arrangement, the conventional voltage regulated power supply which has been fully discussed elsewhere<sup>7</sup> may be employed. In this type of supply, a single 6AS7G, used as the series regulating tube, will deliver sufficient output to supply not only 150 ma of heater current but plate currents as well. It may be noted that in certain of these units, where a highly regulated output is achieved through the use of a large loop gain, the heaters of the control amplifiers are themselves placed across the output of the supply.

If a parallel connection of the heaters is desired, a regulated supply based on the circuit illustrated in Fig. 6 may be used. Here, regulation of the rectifier unit is achieved by amplifying the voltage variations appearing across the load and applying them to the d-c control winding of a saturable reactor in the primary circuit in such a manner as to cause a compensating change in the a-c line voltage applied to the rectifier.

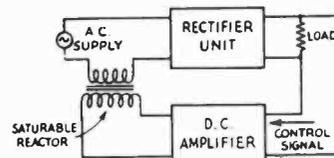


Fig. 6. Regulated d-c heater supply.

A six-volt power supply of this type designed by the author<sup>8</sup>, exhibited a change in output voltage of .05 volts with a change in load current from zero to 15 amperes, and negligible variation in output was observed for line voltages ranging from 105 to 140 volts.

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# Transition Frequency Compensation

C. G. McPROUD

The simple equalizer circuits described make it possible to correct any magnetic or crystal pickup for the various turnover frequencies used in recording.

EVERY RECORD ENTHUSIAST has found certain records that sound fine with one system, but fail dismally on another. Yet another album may sound entirely out of balance on the first system, but reproduce perfectly on the second. Assuming that both systems are reasonably free from distortion and are comparable in other characteristics, this anomaly may cause some concern.

Since the recording characteristics of the various manufacturers are not yet standardized — and it seems doubtful that they ever will be — the reproducing equipment must be made sufficiently flexible to ensure satisfactory performance with all makes of records. At present, we are faced with transition or turnover frequencies of 500 and 800 cps on records of American manufacture, while the English Decca ffr disks are reputed to be recorded with a 300-cps turnover. Obviously, no single equalization can compensate perfectly for these variations.

It is general practice for the recording characteristic to be essentially flat above the transition frequency. Below this point, a droop of 6 db per octave is normal, but this may vary somewhat in practice. The curve does not make a sharp bend at the turnover point, fortunately, on account of the simple means used to achieve the bend. Consequently, a simple means of equalization is sufficient for satisfactory correction.

Reproducing characteristics vary between magnetic and crystal pickups, so the type of equalization varies considerably. Both will be discussed here, with actual tested circuits.

## Compensation for Magnetic Pickups

The high-quality, low-level magnetic pickup, such as Audax, Pickering, and GE, require that some preamplification be supplied, and this provides a suitable place for the equalization.

A 6 db/octave boost requires at least 20 db of equalization for the entire

range. When using *RC* circuits, a total loss (in the unequalized band) of 40 db is usually sustained to secure a net boost of 20 db, such as from 500 cps to 50 cps. If the output of the pickup cartridge is of the order of 50 millivolts, let us say, the loss due to equalization gives a net output voltage of 1/100th of this amount, or 0.5 mv. To get a 1-volt output signal to feed an ordinary amplifier requires a voltage gain of 2,000, or approximately 45 per stage for a two-stage preamplifier, indicating the use of high- $\mu$  triodes. The equalizer circuit may be inserted between them.

A single-section *RC* network, such as that of Fig. 3(A), gives a rise of slightly over 3 db/octave at an insertion loss of 20 db. Putting two such circuits in series as at (B) doubles the boost, and doubles the insertion loss, of course. Thus, with proper choice of values for *R* and *C*, the required equalization can be obtained. The transition frequency is controlled by the size of the capacitors; the amount of equalization by the ratio between  $R_2$  and  $R_1+R_2$ .

Figure 1 shows a complete equalized amplifier suitable for use with a high-quality magnetic pickup.  $V_1$  and  $V_2$  may

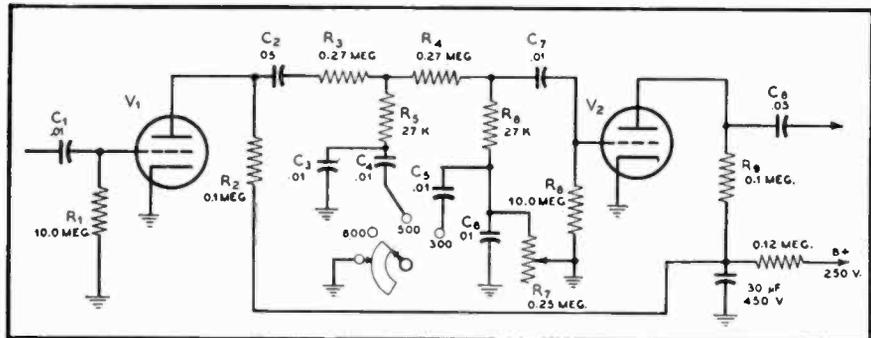
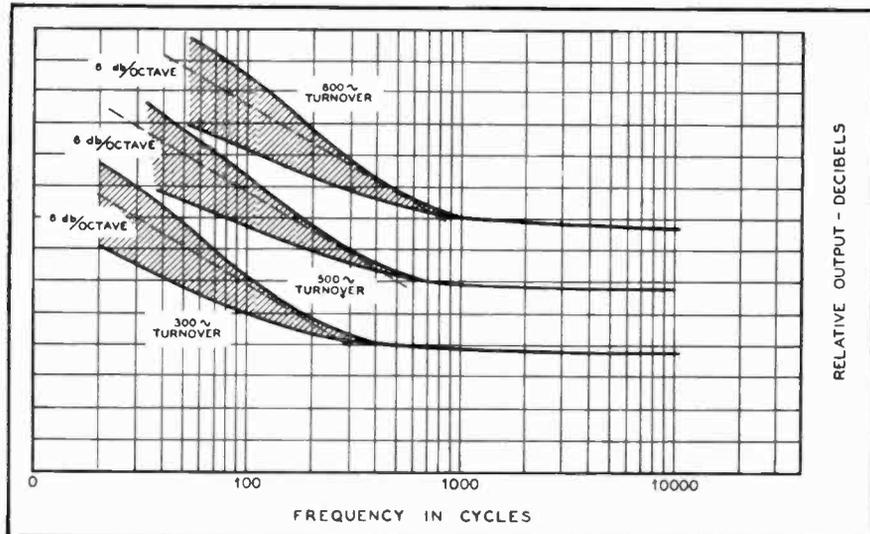


Fig. 1. Equalized two-stage amplifier suitable for low-frequency compensation for modern high-quality, low-level magnetic pickup cartridges.

Fig. 2. Equalization ranges obtainable with circuit of Fig. 1



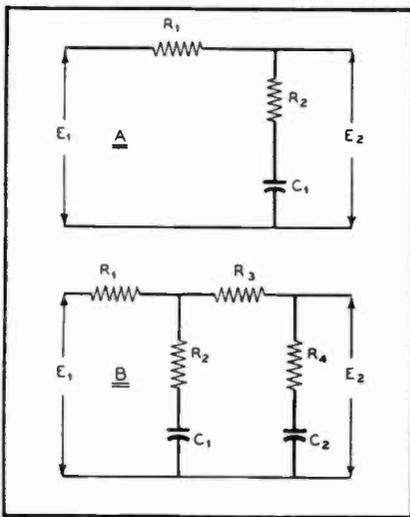


Fig. 3. (A) Single-section RC network used to provide low frequency boost at the rate of 3 db/octave. (B) Two sections are necessary for a boost of 6 db/octave.

be the sections of a 6SL7, 6SC7, or 7F7; or they may be two separate high- $\mu$  triodes such as 6SF5's, or two duo-diode triodes such as 6SQ7's, or 6AQ6's, with the diode plates strapped to the cathodes. The values shown will give a total boost of approximately 7 db/octave, with the switch lowering the transition frequency by adding more capacitance.

Shunting the capacitance of one of the RC sections with the potentiometer  $R_7$  provides a controllable boost ranging from 3 to 7 db/octave. This potentiometer has an audio taper for smoothest action, the connection being made so that clockwise rotation of the knob increases the resistance across the capacitors, and thus increases the bass boost. Figure 2 shows the possible equalizations obtainable with this circuit.

The plate-load resistors  $R_7$  and  $R_8$  may be varied between 50,000 and 270,000 ohms to adjust the total gain of the preamplifier, and the output is intended to feed a 0.5 to 2.0 meg volume control.

### Crystal Pickup Equalization

The problems are entirely different when a crystal pickup is used. These devices normally give a flat output over the frequency range up to the transition point, and droop at the rate of 6 db/octave beyond. The low-frequency response is dependent somewhat on the total load into which the cartridge works, but if this load is greater than 1.5 megohms, the bass may be assumed to be normal.

Since a droop of 6 db/octave corresponds to a loss of 20 db for a frequency ratio of 10:1, the insertion loss must be at least 20 db to compensate for the range from 300 to 3,000 cps for the lowest turnover frequency. An additional 10 db would carry the compensation to 9,000 cps, which is a reasonable upper limit for this type of reproducer. However, most amplifiers used with crystal pickups do not provide sufficient gain to permit the use of this much equalization, so a circuit for a maximum of 20 db will be presented.

If the total insertion loss at low frequencies is to be 20 db, the resistors  $R_1$  and  $R_2$  of Fig. 4 must provide it. Therefore,

$$I. L. (db) = 20 \log \frac{R_1 + R_2}{R_2}$$

Since  $R_2$  may very well be a 0.25 meg potentiometer used for the volume con-

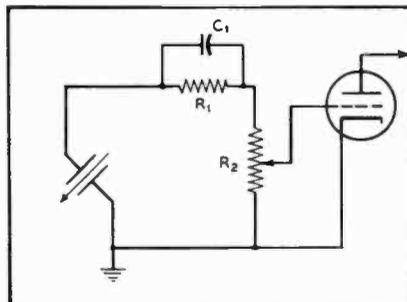


Fig. 4. Type of equalizer used with crystal pickups to provide proper compensation for high-frequency droop inherent in capacitance generator.

trol, and since  $(R_1 + R_2)/R_2$  must equal  $\log^{-1} 1.0$ , or 10.0,  $R_1$  is 2.25 megohms.

The capacitor across  $R_1$  may be calculated for any desired transition frequency. The circuit of Fig. 5 shows a complete equalizer, using a 1.0-meg potentiometer in series with a 1.2-meg resistor for  $R_1$ , thus permitting the control of the amount of high-frequency boost, while the switch  $S_1$  controls the transition frequency.  $S_2$  is used to control the low-frequency response, providing a variety of load resistances for the pickup to work into, with the attendant change in bass output.

### Conclusion

These circuits may be useful to the experimenter, engineer, or record enthusiast for providing a wider range of control for record reproduction. Once the correct settings for the equalizer controls are determined for each individual record or album, they may be marked on the records to facilitate

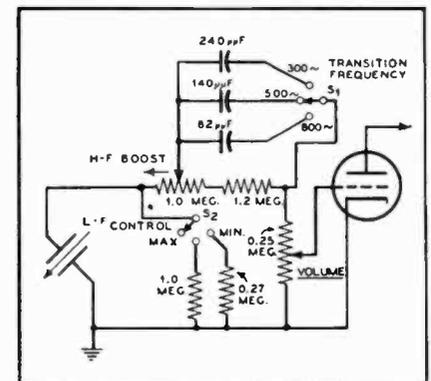


Fig. 5. Complete equalizer circuit to be used between crystal pickup and grid of first tube, making use of both low- and high-frequency compensations.

future use. Proper equalization will do more toward making the reproduction a realistic performance than any other single influence, when used with good quality equipment.

## H-F Equalization for Magnetic Pickups

C. G. McPROUD

Methods of using shunt capacitance to provide low-pass filter action and NAB or LP compensation.

ONCE THE OUTPUT of a magnetic pickup has been equalized so that the response up to the transition frequency is a realistic reproduction of the original recorded material, the listener next turns to the other end of the spectrum in an attempt to eliminate noise or to match the recording characteristic. The

methods of equalizing the low-frequency response were discussed in the preceding article, and such methods are equally applicable to most of the high-quality low-level magnetic pickups that are available. These pickups are of the high-impedance type, designed to work directly into the grid of a vacuum tube. The

electrical characteristics are similar, and the same methods of high-frequency equalization can be applied to all.

In order of importance, the three principal reasons for equalizing the high-frequency end of the audio band are:

1. To reduce surface noise
2. To eliminate distortion

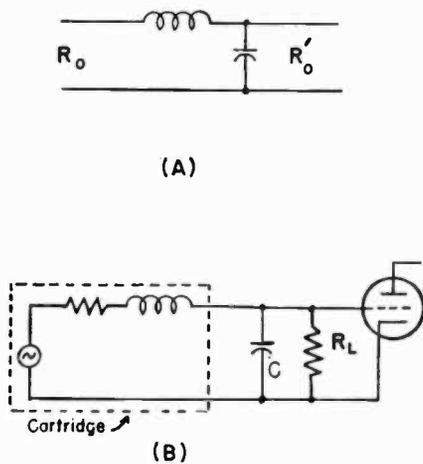


Fig. 1. (A) Conventional L-section low-pass filter configuration. (B) Circuit of magnetic pickup shunted by capacitor  $C$  and terminated by resistor  $R_L$ , feeding grid of amplifier tube.

3. To compensate for the recording characteristic

The reasons for this order will be apparent from the discussion.

The first of these listings is the one most noticed by the layman, who is often the person for whom the equipment is being designed. He is apt to be conscious of noise much before his notice of distortion, and favorite records—usually played more than others—are likely to be the most worn and consequently the greatest offenders.

The second reason becomes more apparent as the ear acquires training. The effect is noticed on many records as a definite "muddiness" and "smearing" in the high-level, full-orchestra passages, probably as a result of cutting at an average level which is too high for the amplifier equipment, the cutting head, or the record material. Consequently, when peak passages are reached, system distortion rises considerably. A higher over-all distortion which remains constant is less objectionable than low distortion throughout parts of the recording, with great distortion on high-level passages.

Compensation for the recording characteristic is likely to be considered by many as the most important of the three listed reasons. Without diverting into a discussion of recording characteristics—which are definitely not standardized among the various manufacturers, or even in one manufacturer's own studios—let it be said that some degree of high-frequency equalization is generally employed in the recording studio, as measured from the microphone to cutting head. However, this does not necessarily mean that the individual record that you buy is going to reproduce with that increased h-f response. It is recognized that one stamper is capable of producing about 1,000 records before it is discarded. The first record may be excellent, but what about the 999th? One more record and the stamper will be discarded. In the

writer's opinion, very few commercial records are on the market today with any excess of high-frequency response—certainly not to the extent that they cannot be compensated for easily by usual "tone control" circuits. Therefore, with the exception of compensation for the NAB characteristic, no "sloping" circuits will be discussed. Reproduction from the record will be assumed to be correct as far as frequency response is concerned, and since both of the high-quality magnetic pickups in growing use are capable of performance up to 10,000 cps, it will be further assumed that the electrical output from the pickup is directly proportional to the velocity of the stylus tip.

**Correction Methods**

The simplest method of reducing the response of any electrical circuit to a desired maximum frequency is by means of a low-pass filter. In proper designs, practically any shape of curve can be obtained. However, in high-impedance circuits the series inductances reach large proportions, and they do pick up hum. Consider a simple low-pass filter such as that shown in Fig. 1A. It consists of a series inductance and a shunt capacitor. The equivalent circuit of a magnetic pickup is essentially a generator in series

with an inductance equal to that of the coil, and with a resistance equal to the resistance of the coil. Thus, if a capacitor is shunted across the output of the pickup, as in Fig. 1B, the net result is that a low-pass filter section is formed. It is only necessary to choose the correct value for the capacitor and the correct value for the terminating resistance, after the cut-off frequency is selected.

**L-P Filter Circuits**

The configuration for the low-pass filter used in connection with these magnetic pickups is now seen to consist of the inductance of the pickup itself shunted by a capacitor, with the combination working into a suitable load resistance. Without considering the actual cut-off frequencies, the general circuit is developed first from the formulas for a constant-k low-pass filter, which will provide the values for  $C$  when the inductance  $L$  and the cut-off frequency are known. The cut-off frequency,  $f_c$ , is that at which the response is down 3 db.

By further rearrangement of the formulas, the value for the terminating resistance is given in terms of  $L$  and  $C$ . To simplify the determination of these capacitance values, they are shown in chart form in Fig. 2. The inductance of the

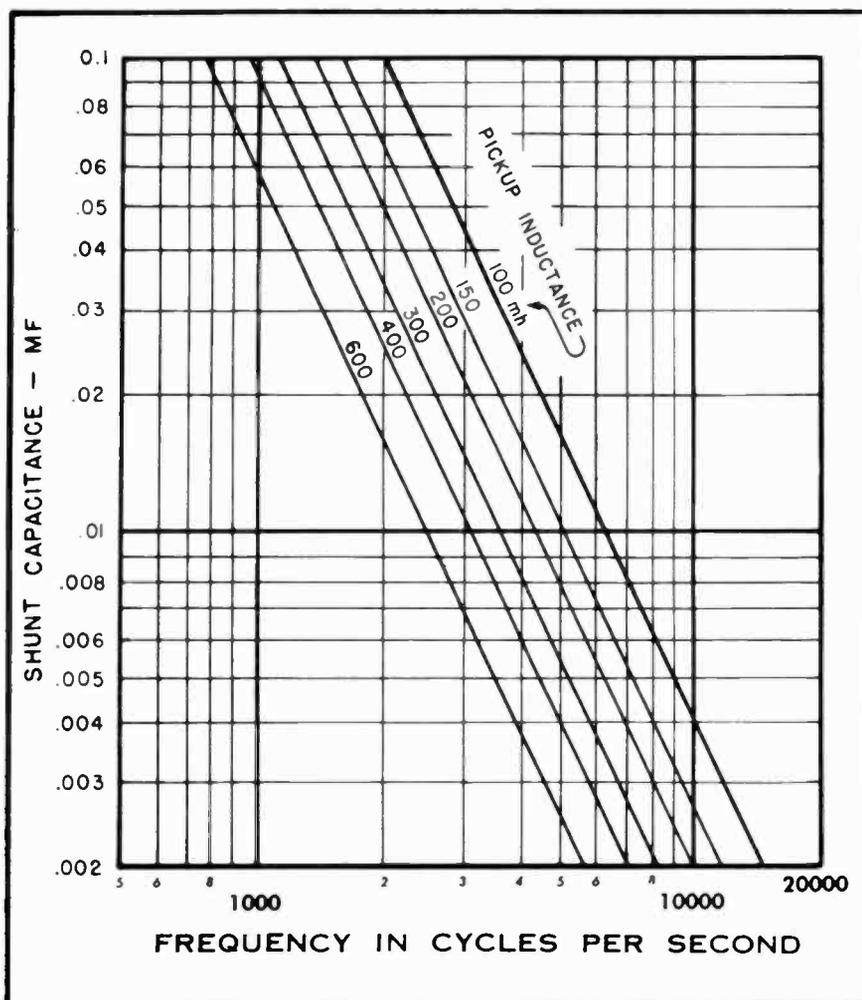


Fig. 2. Chart for determining capacitance to be shunted across magnetic pickup to provide low-pass filter action.

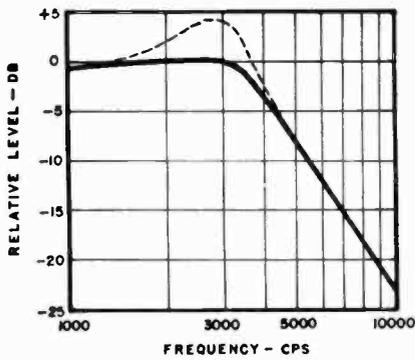


Fig. 3. Shape of response curve for  $f_c = 3850$  cps. Dotted curve indicates peak due to improper terminating resistance.

various available pickups ranges from about 100 mh for the early GE models to approximately 600 mh for some of the more recent designs. For this reason, the curves have been redrawn to accommodate this range of inductances. To use the chart, follow the diagonal line for the inductance of the pickup to the frequency of the desired cutoff, and read capacitance at the left.

#### Response Curves

A typical response curve obtained by the use of a properly terminated capacitance shunt across the output of a magnetic pickup is shown by the solid curve of Fig. 3, while the dotted curve shows the effect of operating the combination into a load resistance equal to three times the optimum value. The smoother response of the solid curve is much to be preferred, and the use of a filter with sharper cut-off characteristics is generally undesirable for high quality audio circuits. Empirical values for the terminating resistance are given by the simple relation

$$R = 1.2 f_c$$

where  $R$  is in ohms and  $f_c$  is the selected cutoff frequency (3 db down). The response is flat up to approximately

80 per cent of the cut-off frequency, is down 3 db at  $f_c$ , and decreases at the rate of approximately 15 db per octave above  $f_c$ .

Any resistance in series with the capacitor does not affect the curve up to a frequency of approximately  $2f_c$ . Above this point, the curve tends to flatten off. Therefore it is recommended that the capacitor be connected directly across the pickup, and shunted by the correct load resistor. For flexibility, and to provide for wide-open response when record surfaces and distortion content permit, a switch may be arranged to connect either of two or more capacitors across the circuit as desired, each capacitor having its own resistor permanently connected across it. Using this arrangement, cut-off frequencies anywhere in the band can be chosen at will. For ordinary use on commercial shellac pressings, a three-position switch providing cut-off frequencies of 3,500 and 6,000 cps, together with one wide-open position, has proven quite satisfactory, although a more elaborate arrangement would obviously give a wider range of control.

#### NAB and LP Roll-Off

Professional users of these pickups often have need for a circuit which will provide the proper roll-off for the high frequencies for correct reproduction of LP records and transcriptions using the NAB recording characteristic. This curve is shown by the solid heavy line of Fig. 4, together with the equalization provided by the low-frequency equalizing circuits previously described, and the simple capacitance droop to correct for the high end. The resulting range of control provides for a response at 100 cps anywhere between +6 and -2db, together with an approximately flat response above 1,000 cps.

The droop resulting from the shunting of a capacitor across a line of a certain impedance is a gradual deviation from flat

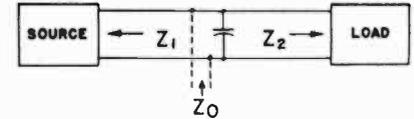


Fig. 5. Impedance  $Z_0$  of any point in a circuit is composed of  $Z_1$  and  $Z_2$  in parallel.

to a loss of 3 db at approximately three times the frequency at the point of deviation, with a further droop at a rate of 6 db per octave above that point. However, this does not apply to capacitor shunted directly across an inductive source such as a pickup. The capacitor must be shunted across a circuit which is essentially resistive. A simple method of determining the correct capacitance to provide equalization for the NAB curve is given by the following:

$$C = \frac{75 (Z_1 + Z_2)}{Z_1 Z_2}$$

where  $Z_1$  and  $Z_2$  refer to the impedances indicated in Fig. 5. This is a modification of the standard formula employed in some broadcasting plants which is  $C = 100/Z_0$ , where  $Z_0$  is the impedance at any given point in a circuit. Thus, the impedance at any point on a 600-ohm line equals 300 ohms, since the source and load—each being 600 ohms—are in parallel. The value of 100 gives a roll-off which is somewhat greater than required to provide flat response.

#### Adaptations to Preampifier

The preamplifier suggested for low-frequency compensation is readily adapted to provide the required roll-off to compensate for the NAB curve. The logical place to insert the shunt capacitor is across the output circuit, at which point the impedance consists of the load resistance shunted by the plate resistance, and both shunted by the grid leak of the following stage. Assuming that the grid leak is 1 meg or more, the impedance at this point is numerically of the order of 40,000 ohms, resulting in a shunt capacitance value of  $0.0018 \mu\text{f}$ .

When the connections from the preamplifier to the succeeding circuits must be fairly long, the capacitance of the shielded lead is apt to cause a droop in the response. This may be corrected easily by the addition of a capacitor across  $R_1$  in Fig. 1 of the preceding article on page 47. With a value of 0.27 meg, for this resistor, the capacitance to be placed across this resistor is equal to  $0.59/f$ , where  $f$  is the frequency at which the response is down 3 db. This same method of equalization may be used to bring the high-frequency response of a poor pickup up to flat at any desired frequency by the application of this formula, if the particular cartridge droops appreciably. The output of the better cartridges is held to a tolerance of  $\pm 2$  db to 10,000 cps, and further correction should not be necessary.

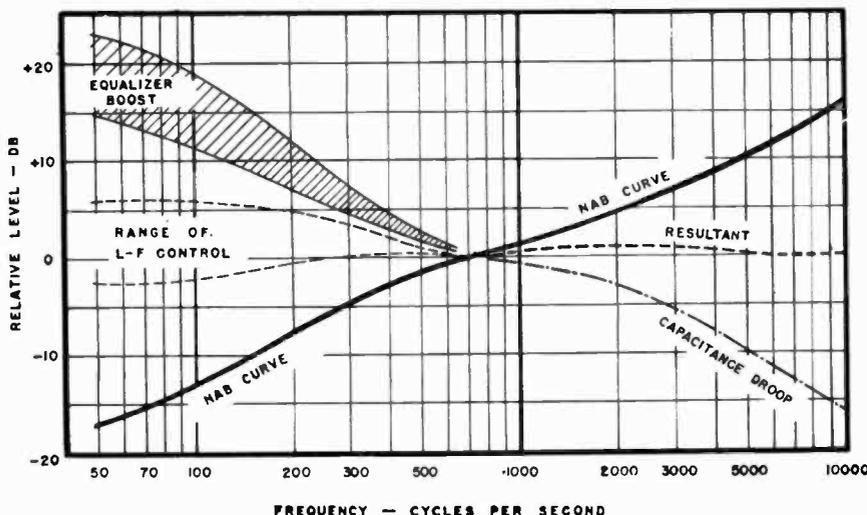
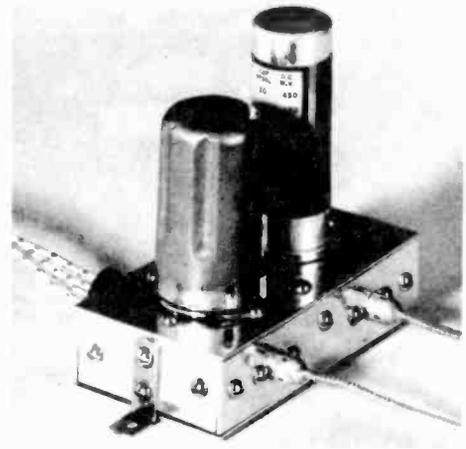


Fig. 4. NAB recording characteristic related to equalized preamplifier and to capacitance roll-off to provide control over low-frequency response and a flat high-frequency response.

# Feedback Preamplifier for Magnetic Pickups

RICHARD S. BURWEN



By using negative feedback for response equalization, this preamplifier design presents many advantages over previous types.

**T**HE PHONOGRAPH preamplifier described herein shows how a feedback circuit can surpass conventional design in four ways and actually cost less to build. These points of superiority are:

1. Very low output impedance enables use of shielded coupling cable without causing severe attenuation of high frequencies.
2. Negative feedback reduces harmonic distortion.
3. Noise and hum originating in the preamplifier are also attenuated.
4. Simpler to build.

Proper reproduction from transcriptions and commercial records with magnetic pickups such as the General Electric and the Pickering requires that the low frequencies be boosted with respect to the middles and, in most cases, that the high frequencies be attenuated to offset the attenuation of lows and emphasis of highs put into the record in the

process of recording. Since the output voltage of these pickups is small, especially at low frequencies, and as it is a good idea to isolate such low level circuits from the power amplifier and a-c supply components, a separate preamplifier unit has been designed which includes this equalization and thereby adapts the pickup to the medium level input of any flat amplifier. Selective feedback accomplishes the equalization.

Designed around the G. E. variable reluctance pickup, the circuit produces the response characteristic shown as the smooth curve in *Fig. 1*. However, it is readily adaptable to other pickups and different degrees of high and low frequency compensation with the aid of formulas developed later on.

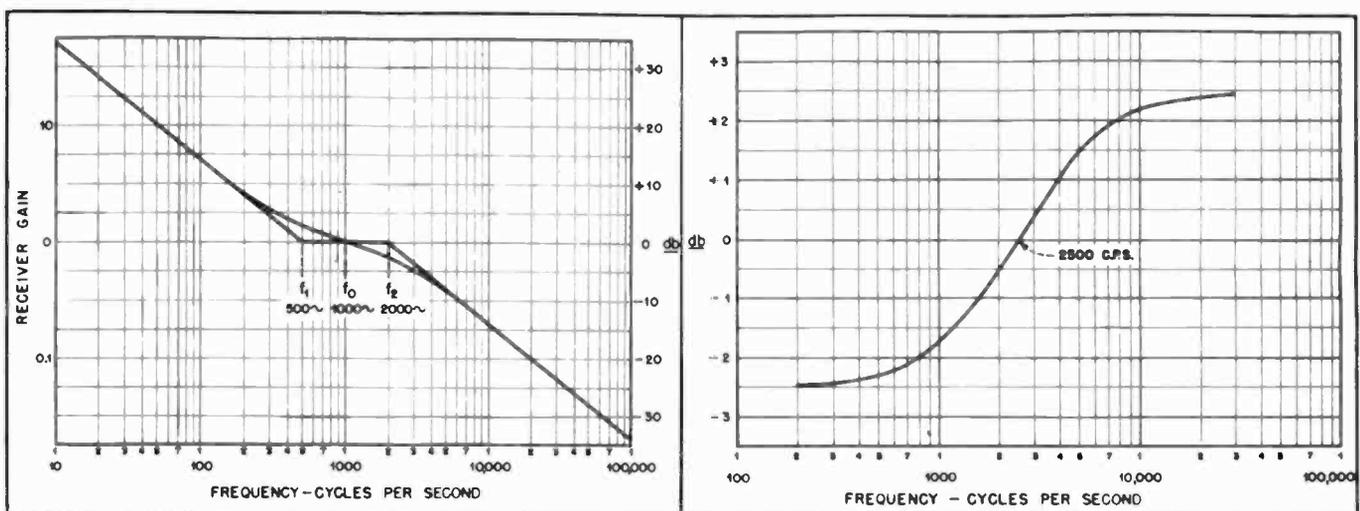
The curve in *Fig. 1* has been chosen as suitable for both transcriptions recorded with the N. A. B. characteristic and commercial records. Variations from the complement of this curve in the record-

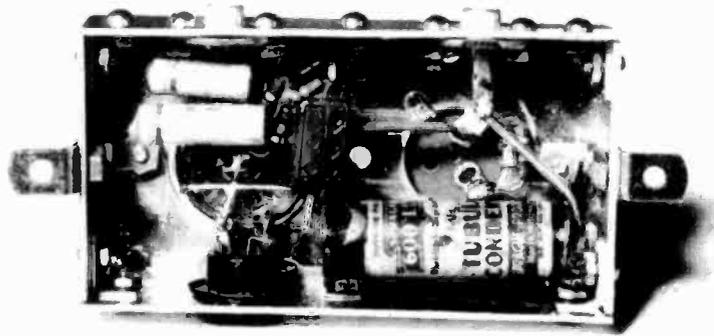
ing characteristics of individual records can be easily taken care of with the usual bass and treble controls elsewhere in the system. By continuing on straight instead of leveling off at 50 c.p.s. it provides 3 db more output at that point than the N. A. B. playback curve; this helps compensate for the deficiencies in many records.

## Design Considerations

The conventional method of accomplishing the job of this preamplifier might consist in using two high- $\mu$  triode stages with a resistance-capacitance equalizer between them. Although this arrangement may be fairly satisfactory provided all the stray capacities are kept to an absolute minimum, we immediately run into difficulty when we try to feed the output voltage through a shielded cable to the main equipment, since shunt capacity in the cable attenuates the high frequencies. We could compensate for

Fig. 1 (left). Play-back curve and its asymptotes for a cross-over frequency of 500 cycles. Fig. 6 (right). Curve showing equalization required to bring magnetic pickup (G. E.) up to flat response on a constant velocity basis.





Under-chassis view of preamplifier.

this loss within the amplifier, but then we would always have to connect the amplifier to the same cable or use a suitable coupling transformer, which would increase the cost.

Another disadvantage with conventional circuits, when the high-frequency roll-off part of the equalizer is inserted between stages, is that while tube hiss generated in the first stage is reduced, that generated in the second stage is not. A simple way of alleviating both these disadvantages would be to roll off the high frequencies at the plate of the second stage by means of a capacitance to ground instead of between stages; but new troubles arise from this expedient because this reactance, effectively across the plate load resistor, will seriously limit the signal voltage handling capability of the tube and greater distortion will result. If the high-frequency roll-off capacitor were connected between grid and ground, we have left a new point where stray capacities and Miller effect can cause unwanted attenuation of the highs.

**Circuit**

All these disadvantages are overcome in the feedback preamplifier circuit of Fig. 3. It uses the same number of parts, but capacitors in the equalizing circuit are smaller; several more advantages are incorporated in the design and construction. Feedback from the plate  $P_2$  of the second triode  $V_2$  to its own grid through the network consisting of  $R_5$ ,  $C_3$ , and  $C_4$  provides the proper frequency compensation. As a result of this feedback, the effective output impedance at high frequencies is of the order of a thousand ohms and the outgoing signal can be fed through a shielded cable or to certain types of tone control circuits that ordinarily require a cathode follower driver without fear of losing the high frequencies. Noise originating in the plate circuit of the last stage is fed back out of phase to its grid and noise and harmonic distortion originating elsewhere are reduced by virtue of the frequency discrimination. Harmonic distortion generated within the second stage is lowered by a fac-

tor of 6 at 30 c.p.s. and considerably more at higher frequencies. Only one point remains where stray capacities can cause appreciable unwanted attenuation of the high frequencies (the plate  $P_1$  of the first stage) instead of two, and the feedback eliminates the possibility of high frequency oscillations due to stray capacitive coupling from plate  $P_2$  to the grid  $G_1$ , rendering the mechanical layout less critical.

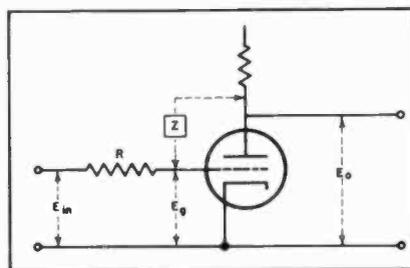


Fig. 2. Detail of feedback stage. Z represents the network  $R_5$ ,  $C_3$  and  $C_4$  of Fig. 3. R is approximately equal to  $R_4$ .

**Construction**

The author's preamplifier was built on a 4 x 2 x 1 inch aluminum chassis with a bottom cover. The tube and the 1 x 3-inch aluminum can filter capacitor were mounted on top and the rest of the parts inside. Filament hum was minimized by completely eliminating filament wires,

accomplished by feeding the power in through a miniature four-prong socket on the side of the chassis that had two of its lugs soldered directly to the filament contacts of the tube socket on the top of the chassis, the types 7F7 and 6SC7 tubes being conveniently designed with the filament prongs adjacent to each other. All the signal-carrying wires were kept down to less than a half inch in length and midget coupling capacitors were used so as to offer as little surface area as possible to the electrostatic field of the filament circuit and to prevent loss of high frequencies through capacity to ground. Hum is thus reduced to a low value provided that the center-tap and not one side of the filament supply be grounded, and this preferably by means of a potentiometer.

**Hum**

It must be pointed out that no amount of care in construction can completely eliminate hum that originates within the tube itself and that where the utmost of fidelity is required the six-volt tube should be replaced by the twelve-volt heater types to reduce current consumption, so the filaments may be heated by a 150 ma direct current supply.

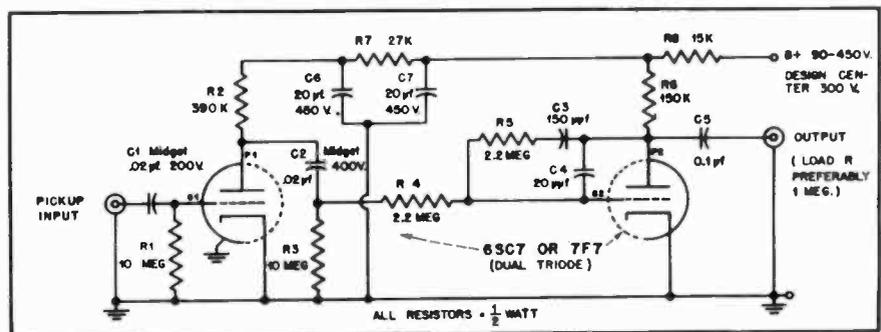
Of the tubes indicated the 7F7 is the one around which this circuit was designed. It was chosen on account of its high gain and low harmonic distortion,\* the total r-m-s distortion for the preamplifier is estimated to be around 0.1 or 0.2 per cent on peaks. The gain is such that with the G. E. pickup, instantaneous peaks seen on an oscilloscope reach five volts on loud records. However, the amplification with the type 6SC7 was found to be nearly the same and the measured plate voltages turned out to be a little closer to the desired values. The 6SC7 is also cheaper and, in the cases of the particular pairs of tubes compared, less microphonic than the 7F7.

**Equalization**

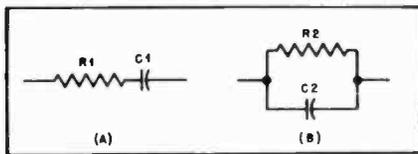
Equalization to flat response from the majority of lateral records and transcriptions requires that the gain rise at a rate

\*Sylvania Electric Products Inc., Technical Manual, Resistance Coupled Amplifier Data

Fig. 3. Complete schematic of the magnetic pickup preamplifier, using negative feedback over the second stage to accomplish equalization.



of 6 db per octave below some frequency which we shall call  $f_1$  and that the gain fall off at a rate of 6 db per octave above some frequency which we shall call  $f_2$ . In addition, there may be a frequency at the low end where the gain begins to level off again which we will consider later. Actually, the bends in the curve at these frequencies are very gradual so that they can be made with simple resistance-capacitance networks and the slope only approaches 6 db per octave at a considerable distance from the bend. The two frequencies  $f_1$  and  $f_2$  vary with different manufacturers, and if we were to construct a network for every combination of  $f_1$  and  $f_2$  in present-day records we would indeed have a large number of networks. The simple way out is to build a single network having  $f_1$  and  $f_2$  representative of a large number of records and leave the rest to be taken care of by ordinary bass and treble balancing controls elsewhere in the equipment. The smooth curve in *Fig. 3* has therefore been chosen with  $f_1 = 500$  c.p.s. and  $f_2 = 2000$  c.p.s. It corresponds closely to the present N. A. B. characteristic.



**Fig. 4.** In (A), network for bass boost only.  $X_{C1} = R_1$  at  $f_1$  (*Fig. 1*). In (B), network for high frequency attenuation only.  $X_{C2} = R_2$  at  $f_2$ .

Mathematically, the simplest way to attain this curve is in two steps, first by boosting the lows and then by attenuating the highs in two separate consecutive amplifier stages. By replacing our  $Z$  in *Fig. 2* with the network *A* or *Fig. 4* we can attain the 6 db per octave slope at the low end. The turnover frequency  $f_1$  is the point at which the reactance  $X_{C1} = R_1$ . A 3 db rise occurs at this point to effect the gradual bend, and the slope approaches nearer and nearer to 6 db per octave as the frequency goes down because the impedance becomes very nearly that of the reactance  $X_{C1}$ , which doubles in every octave. Above  $f_1$  the reactance is small compared to  $R_1$  and so the impedance approaches  $R_1$ , resulting in constant output.

#### High-Frequency Attenuation

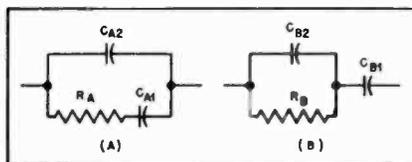
The high frequencies can be attenuated at a rate approaching 6 db per octave above  $f_2$  by replacing  $Z$  with the network *B* of *Fig. 4* where  $X_{C2} = R_2$  at  $f_2$ . Below  $f_2$ ,  $X_{C2}$  becomes large and the impedance approaches  $R_2$ ; above  $f_2$ , where the effect of  $R_2$  becomes small, it approaches  $X_{C2}$ . By passing the signal through a stage containing network *A* and then through a stage containing network *B* we get the combination of these two curves, the same one as in *Fig. 1*. The

straight lines, called asymptotes, show the limiting value of the slope.

If the phonograph turntable produces an excessive amount of rumble, common to the cheap types, it may be necessary to choose a gain reduction factor of about 10 for a leveling off point of 50 c.p.s. and in addition lower the values of all three coupling capacitors  $C_1$ ,  $C_2$ , and  $C_3$ . The N. A. B. playback curve has a leveling off point at 50 c.p.s., but this is usually taken care of by the deficiencies in the recordings and the associated equipment, particularly the loudspeaker.

#### Pickup Response

So far the actual response of the phonograph pickup has not been taken into account. The author tested two G. E. cartridges on the Columbia 10004-M frequency record and on H. M. V. constant tone frequency record numbers D. B. 4034 and D. B. 4035 which according to the label are accurate to within 0.2 db and whose light patterns tend to confirm their accuracy. Agreement was close between the cartridges and fairly good between the Columbia and the British records. The general trend of the curves was that of a roll-off 3 db down at 3000 c.p.s. reaching a maximum dip of 5 to 6 db and then rising again at 10,000 c.p.s. On the Columbia record 10,000 c.p.s. was only 3 db down. Discontinuities in the curve of the British records, which take four sides to change from 8500 c.p.s. down to 500 c.p.s., prevented determination of an exact curve for equalization to flat response, probably on account of the inability of the large point



**Fig. 5.** Network (A) to boost bass and attenuate highs; and (B) equivalent circuit of (A).

radius to reproduce high frequencies as efficiently at the center of the record as at the outside grooves.

So far as the author has been able to determine the resistance-capacitance equalizer curve which comes closest to bringing the output up to flat response is the one shown in *Fig. 6*. In ordinary amplifiers it is achieved by means of the simple 5 db equalizer in *Fig. 7* having a half-loss frequency of 2500 c.p.s. This curve seems to be a happy medium since 10,000 c.p.s. will be up about 2 db at the outside of a record and will be down slightly at the inside.

A word of caution might here be mentioned in connection with the use of long shielded cables between the pickup and amplifier. On account of the 100 mh inductance of the magnetic pickup and the capacitance of the cable, a ten-foot

shielded cable will add another 2.5 db to the output at 10,000 c.p.s.

Fortunately, we can compensate for the pickup characteristic without adding anything to the preamplifier. It happens that if we design the equalizing network for a high-frequency turnover point  $f_2 = 3560$  c.p.s. instead of 2000 c.p.s., the curve of the pickup will subtract from that of the equalizer and produce almost exactly the curve we originally intended in *Fig. 1*. For the two equalizer networks of *Fig. 5* we have.

$$R_A C_{A1} = 318 \text{ microseconds}$$

$$R_A C_{A2} = 52.0 \text{ microseconds}$$

and

$$R_B C_{B1} = 274 \text{ microseconds}$$

$$R_B C_{B2} = 44.7 \text{ microseconds}$$

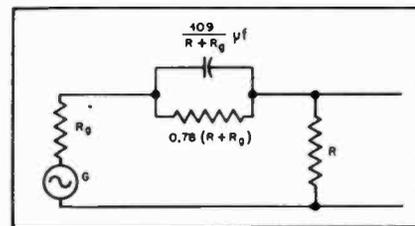
$$\text{The gain } A_o = \frac{Z_o}{R}$$

at  $f_o$ , now 1336 c.p.s., was chosen to be slightly less than unity to permit a gain reduction of at least 50 times and a departure from the asymptote of not more than 0.5 db at 30 c.p.s. for the entire preamplifier. Network *A* was more easily fitted by standard values of resistors and capacitors, making allowance in  $C_{A2}$  for the plate to grid capacitance of the tube and socket.

For use with other magnetic pickups having a higher output than the G. E., it is suggested that the first stage be eliminated and the pickup connected directly from ground to the P1 side of coupling capacitor  $C_2$  in *Fig. 1*. The output can be brought up sufficiently by

selecting a higher value of  $\frac{Z_o}{R}$  provided a

higher leveling off point can be tolerated. Another pickup will of course require different equalization at the high end. A worthwhile addition to the circuit that will make for more pleasing reproduction from worn records would be a switch that would shunt a resistor and capacitor in parallel across the pickup or several pairs so as to cut off the high frequencies fairly sharply at selected points in the manner described in the preceding article starting on page 48. The pre-



**Fig. 7.** Interstage equalizer for flat response from G. E. or similar magnetic pickups.  $R_g$  represents the internal resistance of the driving amplifier.

amplifier can be powered with plate supply voltages other than 300 volts; in fact, there is little difference between operation at 90 volts and that at 450 volts. But, in general, the higher the voltage, the less the distortion.

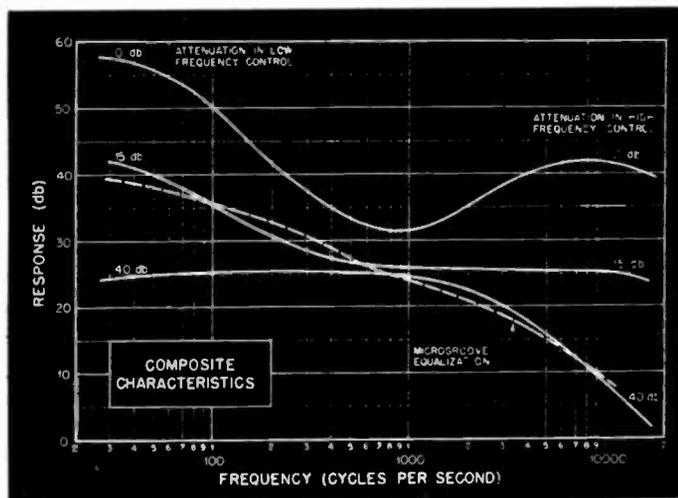


The pre-amplifier schematic is shown in *Fig. 1*. In this particular embodiment, as used in the author's home, the input is a double triode, to accept signals from two variable reluctance phonograph pick-ups simultaneously. After amplification in this tube, the signals are fed to a potentiometer for gain control, and then split into two channels. One of these, the basic channel, goes directly to the summing amplifier. The other, which is further amplified by a triode stage, is subdivided into the two auxiliary channels by low-pass and high-pass RC networks. These networks produce frequency functions suitable for additive equalization.

The feedback amplifier in which the three channels are added has a capacitor,  $C_2$ , across its feedback resistor,  $R_4$ . This, and the capacitor  $C_1$  shunting the plate circuit of the input tube, cause the 12 db per octave high-frequency cut-off of the basic channel. The input resistor,  $R_3$ , for the high-frequency auxiliary channel has a capacitor,  $C_3$ , in shunt for partial compensation of the effect of these elements on the high-frequency auxiliary voltage.

The summing amplifier output is direct-coupled to the grid of a cathode follower, which permits the use of the pre-amplifier at a moderately remote location. The maximum output of the pre-amplifier is not limited by the output stage, which is capable of swinging almost 200 volts peak-to-peak. Overload occurs first at the grid of the auxiliary channel amplifier, which can swing only about one volt peak-to-peak. Since the mid-frequency gain from this grid is unity, the maximum output is also one volt peak-to-peak. This is comparable to

Fig. 4. Limits of available response curves.



the output from a crystal pick-up, insofar as voltage is concerned. Because of the low output impedance of the cathode follower, the output power level of the pre-amplifier is considerably above that of a crystal.

The measured performance of the pre-amplifier is shown in *Figs. 2, 3* and *4*. *Fig. 2* shows the effect of varying the attenuation in the low-frequency channel, with large attenuation in the high-frequency channel. In *Fig. 3* these conditions are reversed. It is seen that the two auxiliary channels overlap in the region of 500 to 1000 cps. The effect of this overlap is seen in the composite characteristics of *Fig. 4*; it is less than 2 db for attenuation greater than 15 db in the auxiliary channels. Smaller attenuation will be used only to compensate unusually poor recordings, or to create special sound effects.

Only three characteristics are shown in *Fig. 4*. Two of these repre-

sent extremes, with either no attenuation, or large attenuation, in both auxiliary channels. Between these extremes is a large number of possible characteristics, of which only one is illustrated.

The required equalization for microgroove recordings is also shown in *Fig. 4*. This characteristic may be achieved within a variation of  $\pm 2.5$  db with attenuation settings of 15 db in the low-frequency channel and 40 db in the high-frequency channel.

The method of equalization presented here is not new. Circuits utilizing the addition of several channels have been described before, but meagre performance data were given. The data shown here are the results of measurements made after the circuit was tailored to produce what were considered to be desirable characteristics. Experience with the pre-amplifier has given convincing evidence that this degree of flexibility is profitable.

# Simple RC Filters for Phono Amplifiers

GEORGE L. ROGERS

The circuits described are designed to enable excellent reproduction from the new magnetic pickups.

**M**ANY AUDIO MEN who are interested in playback of recordings, for themselves or customers, have encountered trouble with "fuzzy" high-frequency reproduction. Some notice it soon after adding a wide-range speaker to the system. The trouble referred to is one which persists after all units of the system seem to be all right. It

shows worst on poor pressings, and even on good pressings after they become a little worn in the loud passages. In both cases the distortion is most apparent in loud passages near the center of the record. Listening tests made with the best available reproducer show that only the best commercial releases have any usable overtones as high as 10 kc, and

on many the surface noise and distortion above 6 kc mask any usable overtones. Some are playable only when the system is cut off above 4 kc. Of course, some of the distortion is due to the inherent characteristics of disc recording and is always present, regardless of the quality of the recording and playback.

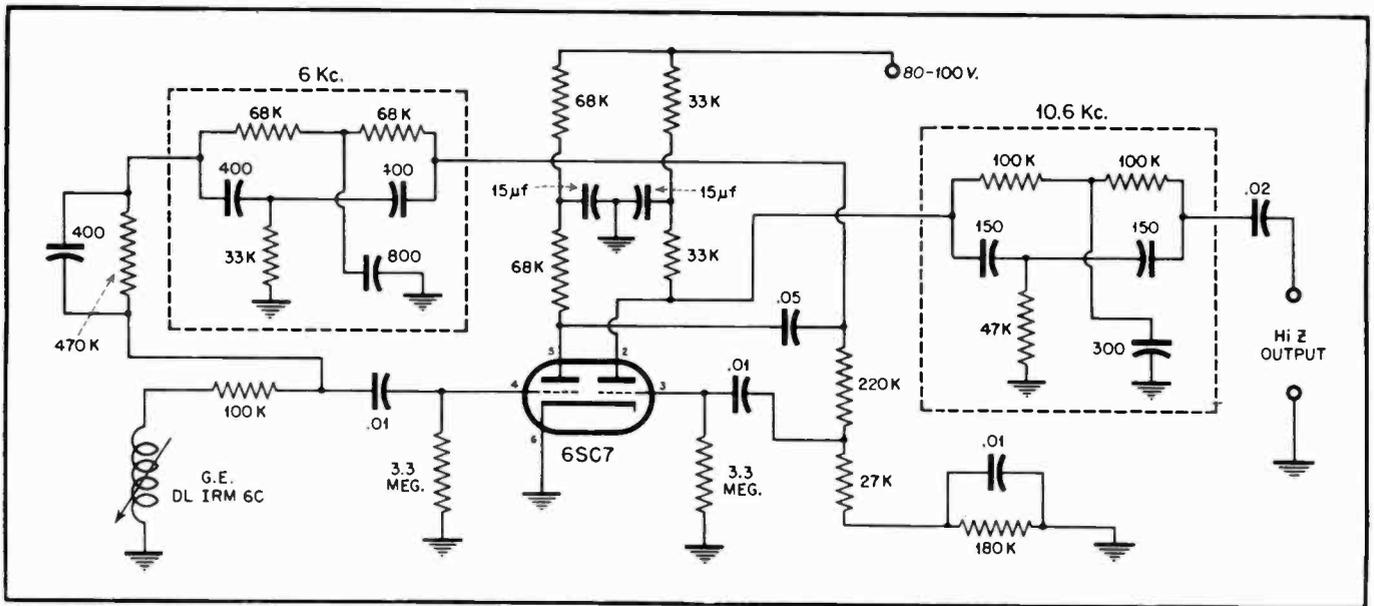


Fig. 1. Schematic of pre-amplifier and filter circuit especially suited for magnetic pickups.

### Frequency Response

However good or bad the system may be, the response of the playback should be cut off sharply at some frequency, for this gives the best reproduction for a given amount of distortion and surface noise. The same applies to any system, whether film, wire, or even live talent through a microphone-speaker system. If the high-frequency end drops off relatively gradually, as is the case with any amplifier without a filter, the distortion and surface noise on average pressings cannot be sufficiently reduced with simple RC tone controls without impairing the medium high frequencies between 3 and 6 kc.

All this is well known to those who have dynamic and variable reluctance pick-ups and filters at their disposal. But these items are not usually available to the average individual, and he usually plays records at high level to make the distortion less noticeable. Even with the proper filter, records sound better at high levels, but distortion is reduced at all levels, the improvement becoming more noticeable as the level is reduced.

The pick-up situation has finally improved with the recent appearance of new units which give smoother high-frequency reproduction than those previously available in the low-price range.

But the cost of filters and filter components is still high and they are not always easy to obtain. Some inexpensive pick-ups use mechanical cut-off to avoid the use of an electrical filter, but this is usually unsatisfactory for one or more reasons. The mechanical filter may lead to excessive needle-talk because of reduced needle compliance, or harmonics generated in the pick-up element may appear in the output even though they

are above the mechanical cut-off frequency. The best results are usually obtained with a wide-range pick-up and an electrical filter to give the cut-off.

One object of this article is to present an inexpensive RC low-pass filter to use with low output variable reluctance pick-ups. The circuit (Fig. 1) is similar to that of preamps recommended for use with this type pick-up, with the addition of two conventional parallel-T networks to provide a low-pass filter action. The 6-kc network in the feedback loop gives a 6-kc peak and the 10-kc network gives cut-off at approximately 10 kc. The resultant response (Fig. 2) shows satisfactory output to 6 kc and fairly sharp cut-off above 7 kc. This cut-off is gradual compared to properly designed LC filters, but gives substantial improvement in reproduction compared to most RC filters of equal cost and simplicity. Naturally feedback

reduces the gain, but the output is still sufficient to satisfy most amplifiers. The high impedance of the output of the filter is a convenience in most cases, but the output should be coupled to the grid of the following stage with short leads to avoid the shunt capacity of long shielded leads. If filament hum is encountered it can usually be cured by grounding the center tap of the filament circuit either at the transformer or through resistors.

The parallel-T feedback loop also is a convenient way to obtain bass equalization of boost without using a choke. The circuit (Fig. 3) uses a network tuned to 80 cps. The amount of the rise may be controlled by varying the amount of feedback, and the frequency of the peak may be shifted by changing values in the network. Response is shown in Fig. 4. This type circuit is preferable to the average RC boost circuit because the 400-cycle region is not lifted objectionably and the response falls off below the peak, giving less turntable rumble for a given boost at the peak. Also there is less "hangover" than with some tuned LC circuits.

Another use of the parallel-T is to cause a dip in the response of the amplifier. In this case the network is used without any tube and the amount of dip is controlled by resistance between the two terminals above ground. This is useful in instances where the speaker or room, etc., have objectionable resonances.

Many variations in the above circuits are possible to vary the frequencies of cut-off or maximum transmission. The network and application in feedback amplifiers have been analyzed in various textbooks and technical publications, but the fundamental formula for determin-

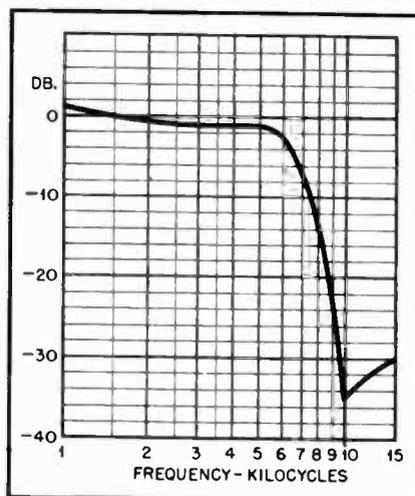


Fig. 2. Response curve of amplifier and filter shown in Fig. 1.

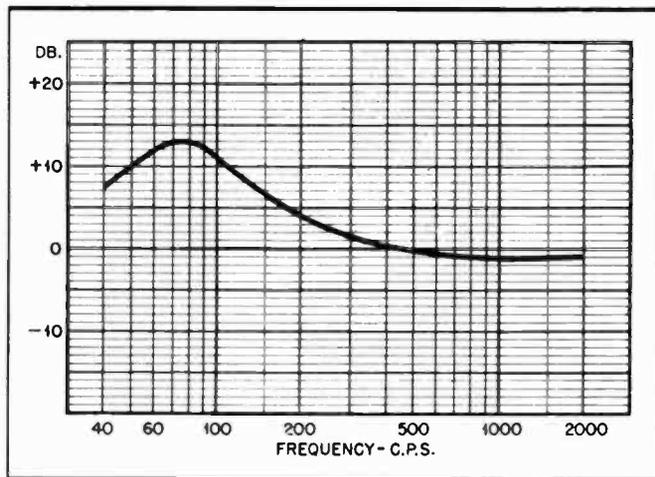
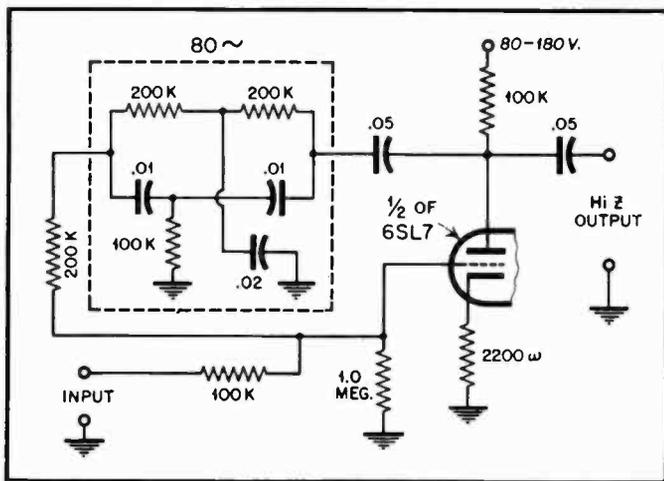


Fig. 3 (left). Circuit to provide bass boost, and Fig. 4 (right), its response curve.

ing the null frequency is repeated here for convenience. In Fig. 5 the null fre-

$$f = \frac{1}{2\pi RC}$$

where  $R$  is in ohms and  $C$  in farads. A more convenient form is  $f = \frac{159,000}{RC}$  where  $f$  is in cps,

$R$  in ohms, and  $C$  in microfarads. The null occurs at this point and transmission returns on each side of this point.

No matter how the constants of Fig. 1 are varied, it cannot be expected to duplicate a good LC low-pass filter. The main use of these filters is to improve results in medium-priced audio systems such as PA and home phonographs. They have the advantage of low hum pick-up, price, and space requirements, and are useful when the results are good enough to be consistent with the quality of the associated system.

### Troubles With Records

Since the above has been non-mathematical and the merit of results largely decided upon by listening tests, some of the inherent distortions in records might be discussed from the same viewpoint.

There is a widespread belief that if record processing and playback were perfected, the results would be nearly life-like. This may be true, but some of the weaknesses in the basic principles of disc recording will make this very difficult to achieve.

If commercial releases are to be improved, several steps are necessary, and widely known and discussed. Probably the most universal offender is the surface. Vinylite pressings have been sold and some show the results expected in reduction of surface noise. Some were disappointing because of impurities in the vinylite or bad pressings. Those with impurities were noisier than the average shellac. On the whole, the average vinylite has not realized the possibilities. However,

when the surface is good, vinylites are superior to competitive shellacs.

Frequency range is an old bugaboo, and the differences in range and recording curves of different manufacturers have been well covered elsewhere. New cutters promise improvement in the manufacturing end. Some new cutters for 78 rpm lateral work employ electrical feedback and multiple windings in the cutter, and lead the way to recordings with the turnover point and the velocity at high frequencies controlled electrically rather than by mechanical damping.

Some of the fundamental reasons for distortion may be shown graphically in a simple manner. The mathematical analyses are necessary for design improvements, but a simple way to illustrate the troubles encountered by the needle in following the groove is to use the imagination a bit. In cutting the master, it can be imagined that the stylus of a high quality cutter, moving in soft wax, can cut a groove which has good playback possibilities.

At any rate, the plow-shaped stylus makes a certain cut and the round needle in riding the groove gives a different motion. The difference between the two motions depends upon the amplitude of the modulation and the frequency. As the amplitude and frequency increase,

the lateral swing of the needle varies more from the original cutter stylus swing, and the vertical component increases. The result is distortion at high frequencies and modulation, and is most apparent at the inner grooves where the groove speed drops, often making the tracking angle error of the tone arm seem more important than it is.

### Graphical Representation

This may be shown in its simplest graphical form by using two sharp pencils. The two pencils are held together so both points touch the paper at the same time and form two parallel lines. Now if the pencils are moved so the lines drawn are separated by the same distance as the points, the lines drawn will simulate the groove cut by the cutter stylus. If the pencils are moved across the paper and deflected transversely similar to the stylus motion in the groove being cut, the lines drawn will roughly portray a modulated groove as viewed from above. With a little practice the lines can be varied so as to show different frequencies and amplitudes. By drawing rough sine waves the lines will be seen to come closer together at high amplitudes, which would "pinch" the round playback needle riding in such a groove. If the process seems worthwhile at this point, it may be carried a step further by adding a diameter equal to the distance between the pencil points. This circle represents the diameter of the playback needle. Now by superimposing the circle on the groove and moving it along the groove, it will be seen that the needle radius cannot follow too high frequencies, and that the needle is pinched by high amplitudes and high frequencies. This means that a smaller diameter is required at such times, and that the needle will be deflected vertically to find the smaller diameter, giving the vertical modulation.

To see what effect this has on the

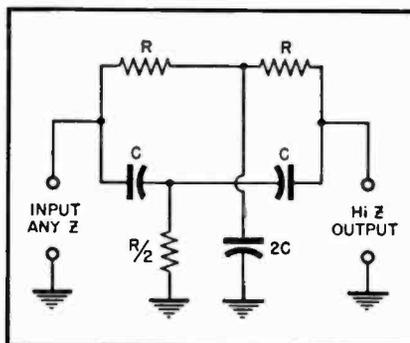


Fig. 5. Fundamental filter circuit.

horizontal component, draw a series of concentric circles on the tracing paper, and at various points along the modulated "groove" fit the appropriate circle and mark the center through onto the paper below. The series of points made by the various centers show the curve of the horizontal movement of the needle. To compare this with the true modulation, trace one of the original outer lines, compare it to the other outer line to be sure they are identical, and then compare to the line of centers. The differences will depend upon the amplitude and frequency, and this dis-

tortion would be present with an ideal pickup.

This method is crude but gives a simple demonstration of lateral recording. It also helps visualization of the limiting factors of frequency response, and distortion at high amplitudes even though the cut does not interfere with the adjacent groove. No matter how these factors are demonstrated, they are present. Pickups which respond to vertical stylus motion (and many do) will give excessive surface noise and distortion. High frequency pre-emphasis will increase distortion.

Even with the above factors present, properly made recordings have a high entertainment value, but the factors make the future of disc recording questionable as far as competition with wire or some other form of recording is concerned. Possibly, when FM is popular, wide-range speakers and amplifiers widely used, and stereophonics practical, the present form of disc recording (vertical and lateral) will become obsolete. Maybe it will be considered a milestone like acoustic recording, but at present it is certainly good business if properly controlled.

# Versatile Phonograph Preamplifier

PAUL W. ST. GEORGE and BENJAMIN B. DRISKO

Design data for an excellent preamplifier.

**A**T THE COST of adding one more article to the mounting list of those intended to solve the phonograph preamplifier-equalizer problem, the writer offers a more detailed analysis and a more versatile solution than has yet come to his attention.

Before launching into a description of the unit, let us consider briefly the purpose of a preamplifier in a reproducing system. This done, we can deal with the design problems more intelligently.

It is our belief that in any A-B test, if a listener finds two units (or two settings of a given unit) which sound different, yet he has no particular preference for either—then it is safe to conclude that neither is the correct answer.

During a long period of experimentation efforts were constantly made to obtain the actual recording characteristics. It was of course obvious from the first that if these are available, the whole design problem is vastly simplified and most of the cut-and-try method—which is cumbersome and time-consuming at best—would be eliminated.

Such data are not easy to obtain, however; but after protracted effort, including tapping all known sources of information, the following table was built up. It is not necessarily accurate but represents the best known estimate of current recording curves:

	Low Freq. Turnover
Columbia 78	300
Columbia 33 <sup>1</sup> / <sub>3</sub>	500
Decca frr	400
R. C. A.	500
Technicord	650
Mercury	300

It is worthwhile at this point to consider the actual purpose of a preamplifier in a reproducing system. A preamplifier should bring the level of a device up to some standard. At this point in the system, switching generally occurs. The output level of the preamplifier should be comparable to that of the other units, such as AM and FM tuners, so that no appreciable change of main gain control setting will be needed on switching from one program source to another, and all sources should be nominally flat at the point of switching.

Rather careful compensation has to be included in a preamplifier to make records sound "right." Listening tests have evolved a series of corrective networks which does a far better job than the first fixed-curve system. With at least semi-certain information to design from, a unit can be built to fit the needs of the day.

Hi Freq. Preemp. db per octave	Freq. at which Preemp begins
6	1590
6	1590
3	3000
2.5	1000
6	2500

Our problem is, then, to use this information to design a simple, inexpensive preamplifier versatile enough to do its work of bringing a low-level pickup output up to some standard level and, at the same time, to produce a nominally flat response curve from the program material. Since a number of curves seem to be in use, the standard single-network preamplifier will not suit our needs. To do this, we place two equalizers, one a low-frequency type with five positions and one a high-frequency type with four positions, in the circuit with an isolating tube between. We now find that by proper choice of constants, we can complement all of the above curves either quite closely or with at most a 2-db error, as *Fig. 1* illustrates.

## Gain Requirements

Fortunately, pickup performance data are easier to obtain than record curves. We find that to bring a Pickering, Clarkstan, or similar cartridge up to about 5-volt level under average conditions requires a gain, in the flat portion, of about 28 db. A GE pickup will require nearer 40.

We want, further, enough extra gain to complement the curve which needs the most boost, and this requires about 30 db more gain. The total is more than high-gain triodes can supply without loss of quality and without the feedback (with its

attendant advantages) falling dangerously low at the bottom end. So we decide on a pair of pentodes.

The 6J7 is an excellent tube for the first stage, having low hum and little noise, yet it is inexpensive and readily available. In the second stage, a 6SJ7 eliminates the gridcap. We choose constants which give high gain consistent with freedom from erratic behavior on changing tubes. We adjust voltages to meet our output requirements and to minimize intermodulation without feedback (as evidenced by d-c plate voltage shift vs. signal level).

Operating the first heater on d.c. was found unnecessary; a variable center-tap and variable positive bias, as shown, suffice. Now we apply feedback to reduce the gain in the flat portion to that desired—28 db in this case. We allow the gain to rise at a 6 db rate by a conventional RC combination. But we supply five such combinations for the five curves. We also use screen bypass values which seem unnecessarily large until we consider that we are bypassing not a screen dropping resistor alone, but it and screen resistance in parallel. Furthermore, we feel that it is desirable not only to make the amplifier stable on the low end but also to achieve critical damping. This requires a 6 db per octave droop below the useful frequency range and on down to the point where the gain approaches unity.

Note that the low-frequency turn-over selector switch is a shorting type. Otherwise opening the feedback loop between switch points results in a sudden and extremely annoying rise in level—about 30 db. The 10-megohm click suppressors are not merely gingerbread; the sharp transients generated in their absence create extremely high peaks in the

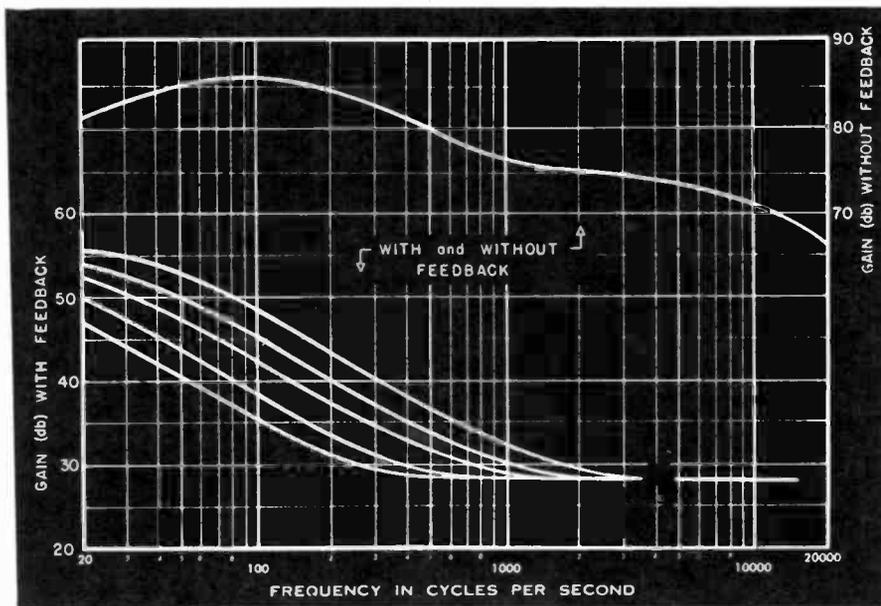


Fig. 1. Response curves of preamplifier with and without feedback.

power amplifier and speaker.

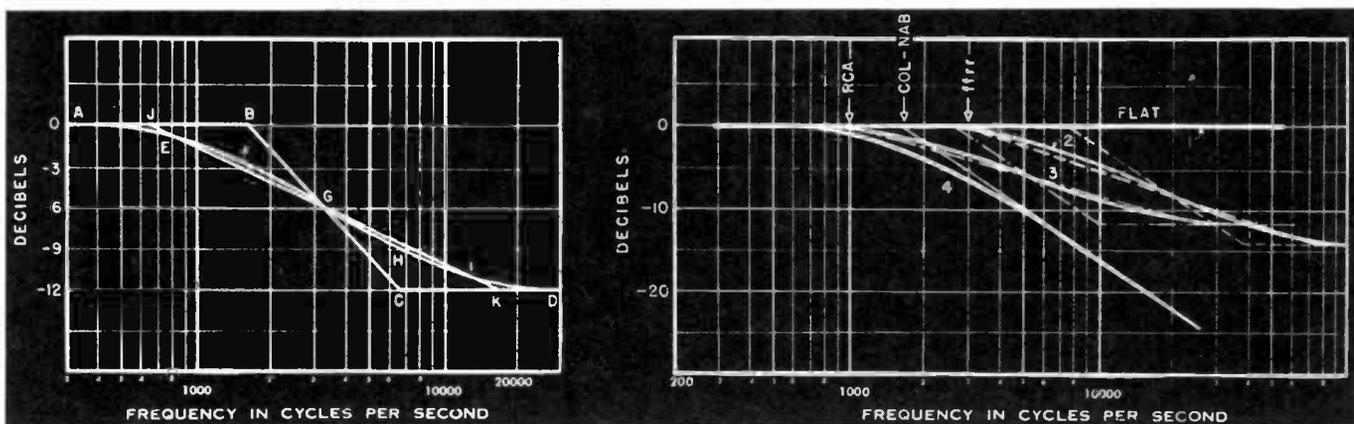
We now have a high-gain, low-noise, low-distortion, low-frequency-compensating preamplifier of simple construction, but one which still does not correct the upper end of the record spectrum. If we applied the feedback approach here, we would run into several difficulties: our gain when on the NAB-type curve would run perilously close to unity at the extreme high end, and our feedback would rise appreciably, with attendant disadvantages.

Further, there is no objection to a conventional lossier in this application. We can feed it directly by the pickup and its design is simplified by being isolated by a tube. For the first we examine the NAB characteristic, which is a simple 6 db/octave rolloff, beginning at 1590 cps. This calls for a simple RC network, with a product of 100 microseconds;

0.1 meg and .001  $\mu$ f is a good combination. With too small an  $R$ , we need to figure in pickup resistance and inductance. Since these vary with manufacturer, and we are designing a universal preamplifier, this is not desirable. At the opposite or high-impedance extreme we may run into hum pickup and tube input capacitance begins to cause some trouble.

A brief examination of the curve produced by the above and all other such networks shows us that at the nominal turnover frequency we are 3 db below the original level, and at exactly one octave either side are 1 db down from the respective asymptotes. The slope of the curve increases at the rate of 6 db per octave. This slope is fixed in character and cannot be changed; however, it can be stopped along its travel. Since we need other curves of less slope (2.5

Fig. 2 (left). Effect of resistor in series with shunt condenser. Fig. 4 (right). Derivation of actual high-frequency roll-off curves from graphical (dotted lines) construction.



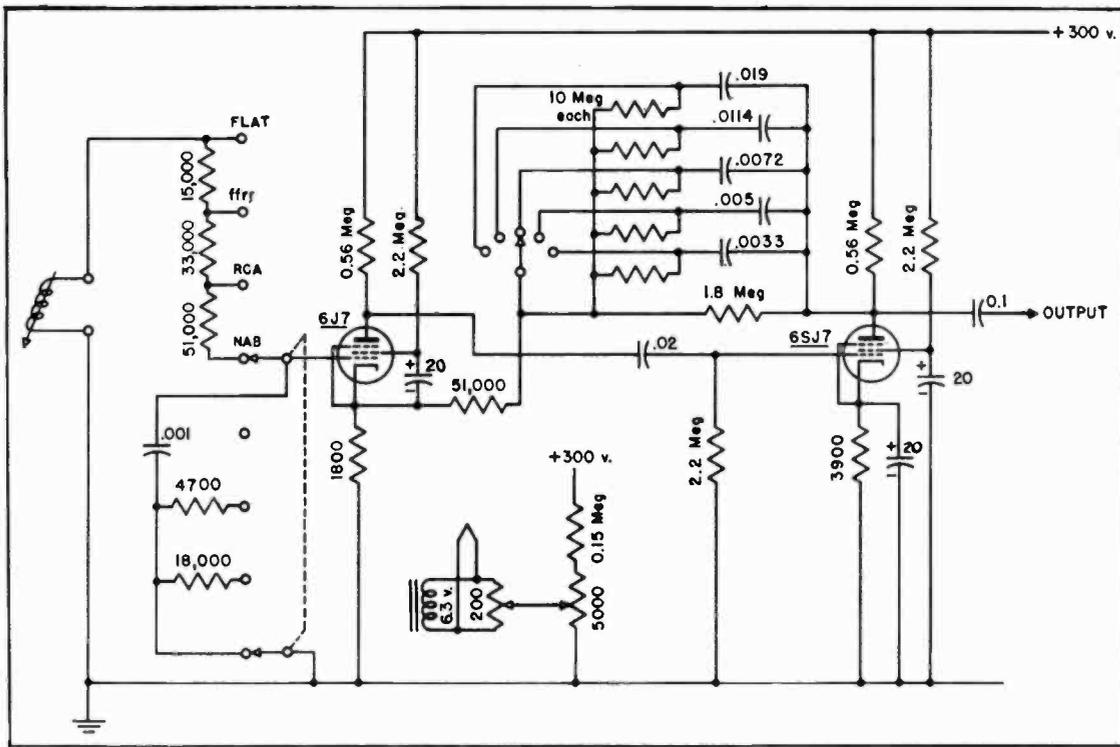


Fig. 3. Complete schematic of pre-amplifier.

db for RCA, 3 db for Decca), they can be obtained by stopping the 6 db curve at some empirically determined point.

Suppose we insert a resistor in series with the shunt condenser to give a second turnover two octaves above the first—as in Fig. 2. The asymptotes are the line *abcd*; as would be expected, the actual curve falls 3 db below *b* and above *c*. The actual curve is *aefghid*; and if we lay a rule along this curve, we find its closest approximation is line *jk*, which has a slope of 2.5 db/octave with respect to a new pseudo turnover frequency which is about 0.39 times the mathematical turnover. It is easily seen that if *b* and *c* were farther apart, the realized slope would have been steeper, and in this manner we arrive at the constants shown in Fig. 3.

Now to the measured performance of the preamplifier. About a half-dozen have been built using 10% tolerance resistors picked at random rather than selected for exact value. Three amplifiers were carefully tested. The best of the lot had 93 db gain without feedback (100  $\mu$ f across the first stage cathode resistor) and 30 db on the flat portion of the curve with feedback. The curves shown happen to be for the worst of the three, as are the performance data. With a Ballantine model 300 meter (0.5 meg) and a Dumont 208 scope (2.0 meg) as load, the maximum output was +20 dbv (db relative to one volt)—a peak-to-peak swing of 28 volts. At 1200-cycle turnover, which

gives maximum bass boost, the output with a 600-ohm attenuator across the input, and no further signal, was -88 dbv. Since the gain at 60 cycles is 52 db this says that the hum voltage referred to the grid is about -120 dbv, or about one microvolt. It is not possible with ordinary test equipment to measure lower, which justifies the hum-bucking circuits used rather than the d-c heater supply with its attendant expense and bulk. The one-microvolt figure was attained in each case with the second 6J7 tried; the first, however, gave only about 10 db more, which is still in the excellent-performance bracket.

The *RC* time constants in the feedback path are, however, rather critical. It is not easy to measure a 1.8 megohm resistor with accuracy; hence the following procedure was followed. The amplifiers were assembled with parts out of the bin; in one case four of the five constants were correct, in another, three, as plotted curves showed. Examination of the curves showed clearly which capacitors to change, and in which direction and amount to achieve the intended result.

Such an equalizer has been in use nearly a year in conjunction with an amplifier-speaker system which we confidently believe to be  $\pm 2\frac{1}{2}$  db from 48 to 13000 cycles and reasonably free from transient defects. The one outstanding change that appeared to coincide with the installation of the equalizer was that listeners started, for the first time, comparing the music to the original

rather than to some other reproducing equipment with which they were familiar.

Even though the curve of a particular record is not known, a little listening experience enables one to select quite readily the proper equalizing characteristic; indeed, in the case of many records, only one setting sounds at all satisfactory, others being very obviously wrong. Occasionally a record turns up for which no settings will provide any illusion of realism.

After having lived with this pre-amplifier for several months, we feel tempted to conclude somewhat as follows: The input data, upon which the whole design was based is certainly in error by much more than 2 db in certain frequency ranges and on certain recordings. The probable error in over-all equalization in the next record played will very likely be well over 3 db, which is about the maximum deviation between characteristics obtainable with the foregoing circuits. This says that further progress is impossible or unwise until more reliable input data is on hand.

The finished product is compact, simple to construct, and inexpensive. It does not demand critical tolerances nor particularly careful physical layout. It will accommodate any current low-level wide-range cartridge and correct any current known or alleged response curve. Its hum, noise, and distortion content are a lot better than most commercially available equipment.

# 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.

**I**N 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

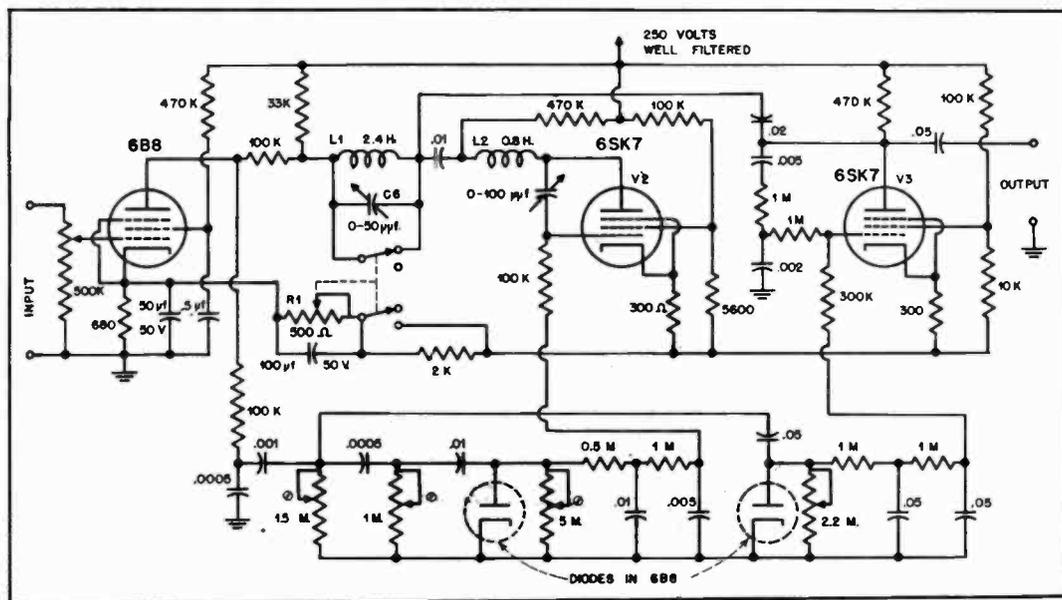
will not preserve all of the frequencies present in every record under conditions of maximum suppression. It *will* improve the signal-to-noise ratio incredibly, and with proper adjustment will preserve the band pass necessary for full range reproduction to a degree that makes it impossible for most 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 reproduction 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

**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.



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

$$(g_m = \frac{\Delta i_p}{\Delta e_g})$$

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 lower-

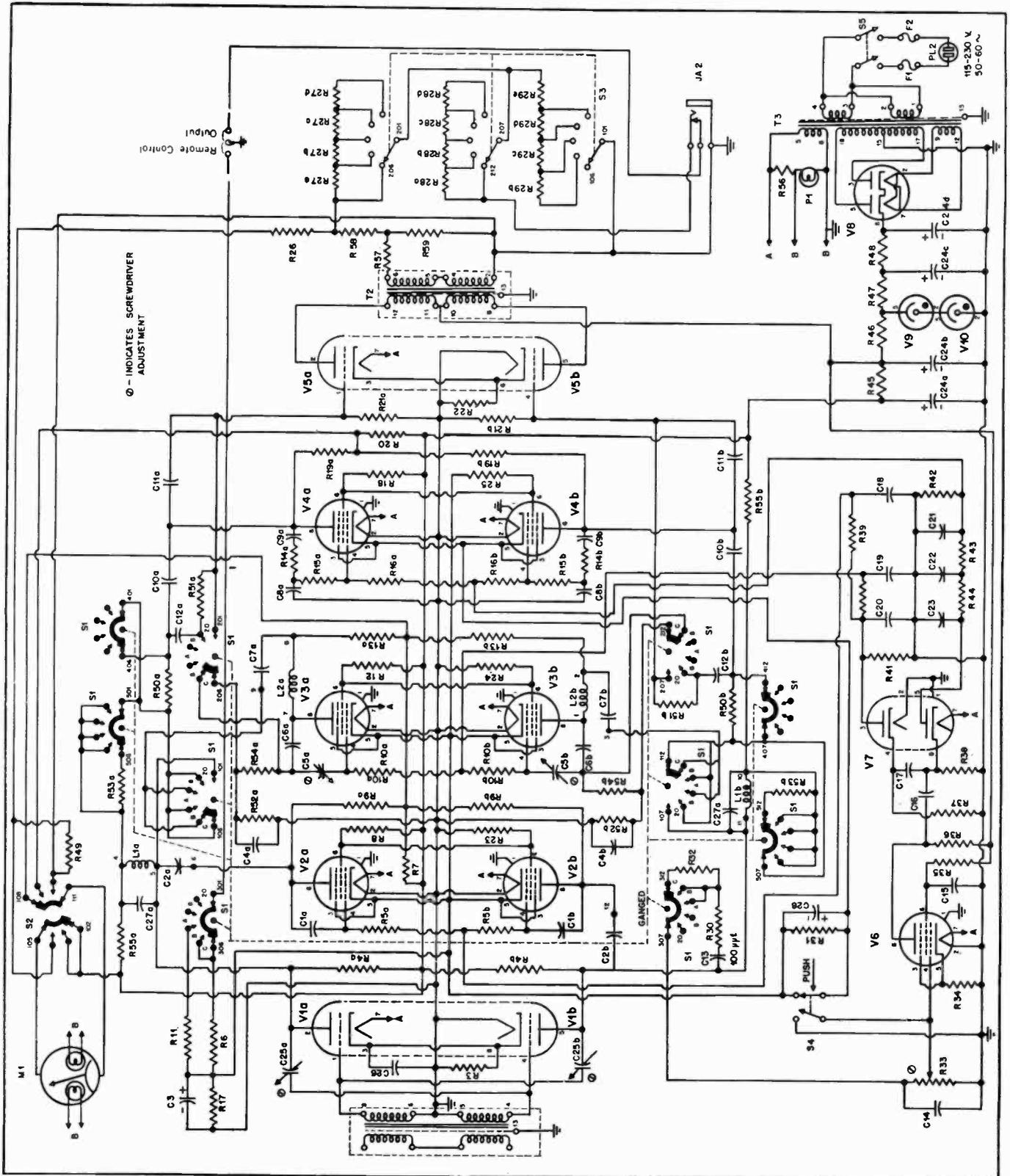


Fig. 2. The 910-A Dynamic Noise Suppressor for broadcast stations. Push-pull circuits conform to standard broadcast engineering practice with usual advantages of push-pull arrangements. Elaborate switching and remote control facilities aid in obtaining optimum conditions on wide variety of records with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

## PARTS LIST

R <sub>1</sub>	600
R <sub>2</sub>	47
R <sub>3</sub> , R <sub>22</sub>	1000
R <sub>4a</sub> , R <sub>4b</sub> , R <sub>8</sub> , R <sub>10a</sub> , R <sub>10b</sub> , R <sub>12</sub> , R <sub>18</sub> , R <sub>30</sub>	0.1 meg
R <sub>6</sub>	22
R <sub>7</sub>	440
R <sub>9a</sub> , R <sub>9b</sub> , R <sub>13a</sub> , R <sub>13b</sub> , R <sub>10a</sub> , R <sub>10b</sub> , R <sub>30</sub> , R <sub>35</sub> , R <sub>37</sub> , R <sub>38</sub>	0.47 meg
R <sub>11</sub>	110
R <sub>14a</sub> , R <sub>14b</sub> , R <sub>15a</sub> , R <sub>15b</sub> , R <sub>21a</sub> , R <sub>21b</sub> , R <sub>30</sub> , R <sub>40</sub> , R <sub>41</sub> , R <sub>42</sub> , R <sub>43</sub> , R <sub>44</sub>	1 meg
R <sub>10a</sub> , R <sub>10b</sub> , R <sub>32</sub> , R <sub>52a</sub> , R <sub>52b</sub>	0.22 meg
R <sub>17</sub>	2200
R <sub>20</sub>	800
R <sub>23</sub> , R <sub>31</sub>	6800
R <sub>24</sub> , R <sub>25</sub>	5600
R <sub>26</sub>	5500
R <sub>27</sub> , R <sub>28</sub> , R <sub>29</sub>	Output T-pad, special
R <sub>33</sub>	0.5 meg
R <sub>34</sub>	550
R <sub>45</sub> , R <sub>48</sub>	1000, 10-watt
R <sub>46</sub>	470
R <sub>49</sub>	3300
R <sub>50a</sub> , R <sub>50b</sub>	10,000
R <sub>51a</sub> , R <sub>51b</sub> , R <sub>54a</sub> , R <sub>54b</sub>	4.7 meg
R <sub>53a</sub> , R <sub>53b</sub>	15,000
R <sub>55a</sub> , R <sub>55b</sub>	0.33 meg
R <sub>56a</sub> , R <sub>56b</sub>	5
R <sub>57</sub> , R <sub>58</sub>	235
R <sub>59</sub>	687
C <sub>1a</sub> , C <sub>1b</sub> , C <sub>3a</sub> , C <sub>4b</sub>	200 $\mu\mu\text{f}$
C <sub>2a</sub> , C <sub>2b</sub> , C <sub>7a</sub> , C <sub>7b</sub> , C <sub>11a</sub> , C <sub>11b</sub> , C <sub>12a</sub> , C <sub>12b</sub> , C <sub>15</sub> , C <sub>22</sub> , C <sub>23</sub>	.01 $\mu\text{f}$
C <sub>3</sub>	500 $\mu\text{f}$ , 25 v.
C <sub>5a</sub> , C <sub>5b</sub> , C <sub>14</sub>	100 $\mu\mu\text{f}$
C <sub>6a</sub> , C <sub>6b</sub> , C <sub>16</sub>	2500 $\mu\mu\text{f}$
C <sub>8a</sub> , C <sub>8b</sub> , C <sub>9a</sub> , C <sub>9b</sub> , C <sub>18</sub> , C <sub>19</sub> , C <sub>20</sub> , C <sub>21</sub>	5000 $\mu\mu\text{f}$
C <sub>10a</sub> , C <sub>10b</sub>	.02 $\mu\text{f}$
C <sub>13</sub>	250 $\mu\text{f}$
C <sub>24</sub>	20 $\mu\text{f}$ , 4 sec.
C <sub>25a</sub> , C <sub>25b</sub>	7 $\mu\mu\text{f}$ , dual unit
C <sub>26</sub>	100 $\mu\text{f}$ , 25 v.
C <sub>27a</sub> , C <sub>27b</sub>	15 $\mu\mu\text{f}$
C <sub>28</sub>	25 $\mu\text{f}$ , 25 v.
M <sub>1</sub>	std. VU meter
T <sub>1</sub>	TRAE-1 transformer
T <sub>2</sub>	TRA4-1 transformer
T <sub>3</sub>	TRIP-1 transformer
L <sub>1</sub>	CHI-A (2 coils)
L <sub>2</sub>	CHI-B (2 coils)
V <sub>1</sub>	6SL7
V <sub>2a</sub> , V <sub>2b</sub> , V <sub>3</sub> , V <sub>4</sub>	6SK7
V <sub>5</sub>	6SN7
V <sub>6</sub>	6SJ7
V <sub>7</sub>	6H6
V <sub>8b</sub>	6X5GT
V <sub>9</sub> , V <sub>10</sub>	OD3

ing 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 control 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 Sup-

pressor 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 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.<sup>1</sup>

#### Pass Band

The Dynamic Noise Suppressor is of the "horizontal" type, providing dynam-

1. Goodell, John D., and Michel, B. M. H., Auditory Perception, *Electronics*, July 1946.

ic 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.

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# Experimental Noise Suppressor

CHARLES D. COLE

Complete data on a noise suppressor developed by the General Engineering Department, American Broadcasting Company.

**A** CHALLENGE, long outstanding in both the broadcasting and home radio industry, is the needle noise or hiss generated in the playback of recordings. Although the ordinary shellac pressing is the worst offender in this respect, the high grade electrical transcription often contains noise components which may prove objectionable, especially

on low level musical passages. Electronic noise-suppressing devices have been designed to overcome this difficulty, but for the most part have proved to be somewhat expensive, especially for the home record player. Below is presented a system for noise suppression which needs little or no maintenance and whose initial expense is quite moderate.

#### Operation

The operation of this suppressor is based, first of all, on the nature of playback noise and its relationship to the recorded program material. It has been found that the amplitude of the noise is approximately constant throughout a recording, irrespective of the amplitude of the recorded program. For high level pas-

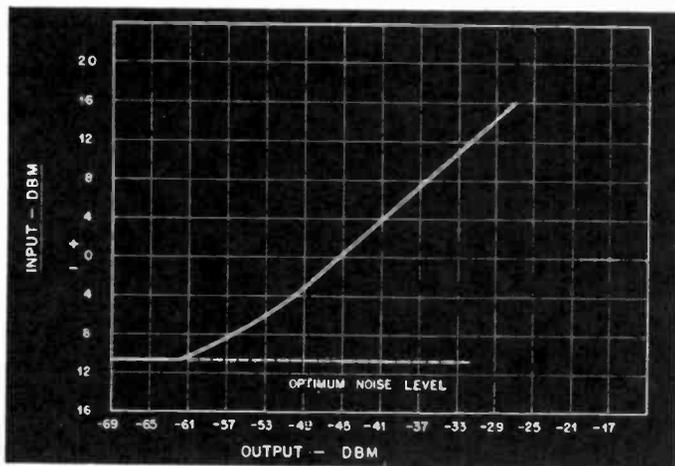
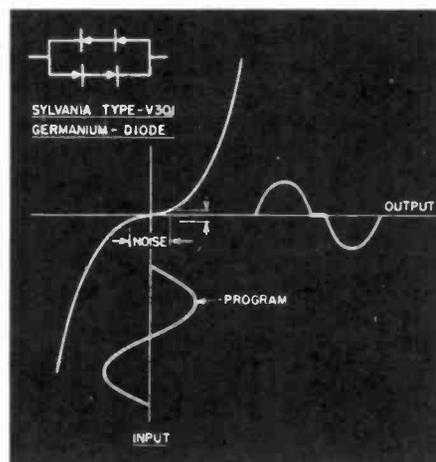


Fig. 1 (left). Suppression characteristic for 1.5 kc to 3 kc channel at 2500 cycles.

Fig. 2. Conduction characteristic of germanium diodes used in the suppressor.



sages the noise is masked, but for moderate to low level passages, which constitute a greater portion of an average recording the noise may become quite objectionable. Secondly, investigation reveals that the greater part of the noise energy lies in the middle and upper ranges of the audio spectrum. Very little noise is found below 1500 cycles. The third factor which enters into the design of this equipment is the relationship between the peak amplitude of the recorded program to the average amplitude of the noise. The program peaks are approximately 40 db above the noise, and this figure was used for design purposes. With these facts concerning the nature and behavior of playback noise, an instrument was designed whose fundamental basis for operation was first set forth by Dr. H. F. Olson of the Radio Corp. of America's Princeton Laboratories. Dr. Olson's proposal and the final instrument evolved from it centered around the characteristics of the germanium diode, and the principle of selective octave filtering.

In a conventional representation of input vs. output voltages for the germanium diode, (see Fig. 1) little attention is given to the load line as it approaches very closely to the origin. The load line for most practical purposes is linear. However, if the portion near the origin

is investigated more closely, it will be found that the load line does not remain linear, but curves tangentially to meet the ordinate axis. The point of tangency (approximately 1 millivolt input) determines the practical minimum level at which conduction can occur in the forward direction. Therefore, an effort to adjust the noise level of the recording so that it would fall in this rejection range was made. For reasons of design it was found more desirable to connect two diodes in series, doubling the rejection level and operating the networks at higher voltages. Two sets of series diodes are connected to give full wave conduction essentially linear except for very small voltages. (See Fig. 2). With the noise level adjusted to fill the dead zone, the program material has a linear excursion some 40 db higher in level.

Although noise is rejected below the point of conduction, program material of a corresponding level is also rejected. This is not especially noticeable to the ear due to the wide dynamic range of most speech and music, except for some elimination of reverberation, or attenuation of sounds inherently low in amplitude such as the spoken letter "S."

#### Distortion

For signals whose amplitudes are 6 to 12 db above the noise level, serious distortion occurs due to clipping and bend-

ing of the load line. It is this distortion that necessitates the use of electric wave filters. Identical filters are used both before and after the diodes. The input filter admits a band of frequencies one octave wide and the output filter passes only this octave and rejects all other frequencies by at least 30 db. The term "all other frequencies" includes not only the program material frequencies, but even more important, the harmonic frequencies or distortion generated by the action of the diodes. The frequency range has been arbitrarily divided into four channels: 0-1500 cycles low pass, 1500 to 3000 cycles band-pass, 3000 to 6000 cycles band-pass, and 6000 to 12000 cycles band-pass. The fact that very little noise occurs below 1500 cycles accounts for this division and subsequent octave relationship. In general, the filters were designed to include the least number of reactors and yet give satisfactory performance. For the low pass filter, an *m*-derived section was chosen. A value of .6 for "*m*" was selected since both input and output terminate in resistive networks. With the low pass section adjusted for satisfactory operation, rejection at the notch is 38 db, least rejection 21 db and rejection at three octaves is 28 db.

The three band-pass filters are constant K networks whose surge impedance is

Fig. 3 (left). Individual filter response curves for experimental noise suppressor.

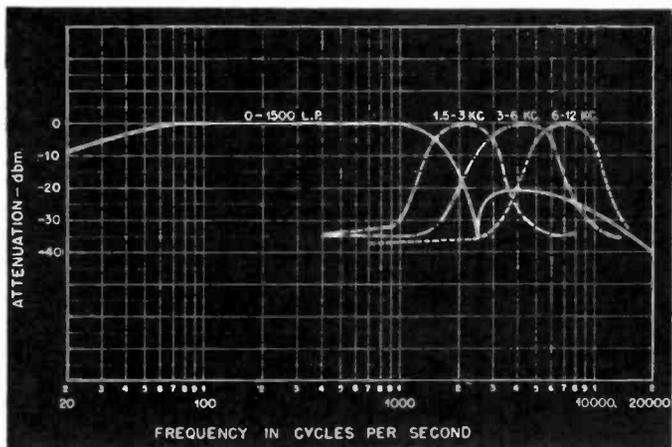
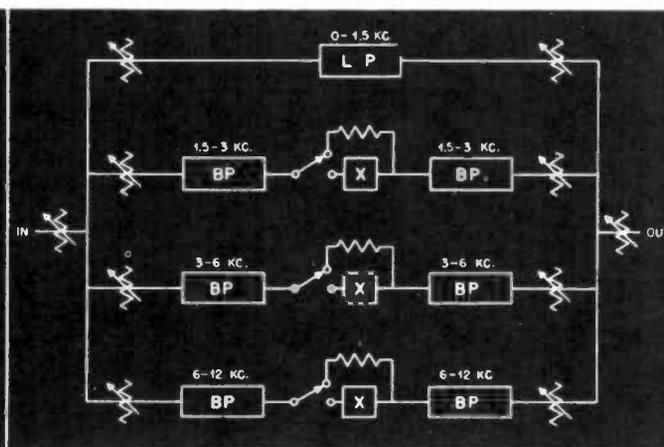


Fig. 4 (right). Block diagram of noise suppressor.



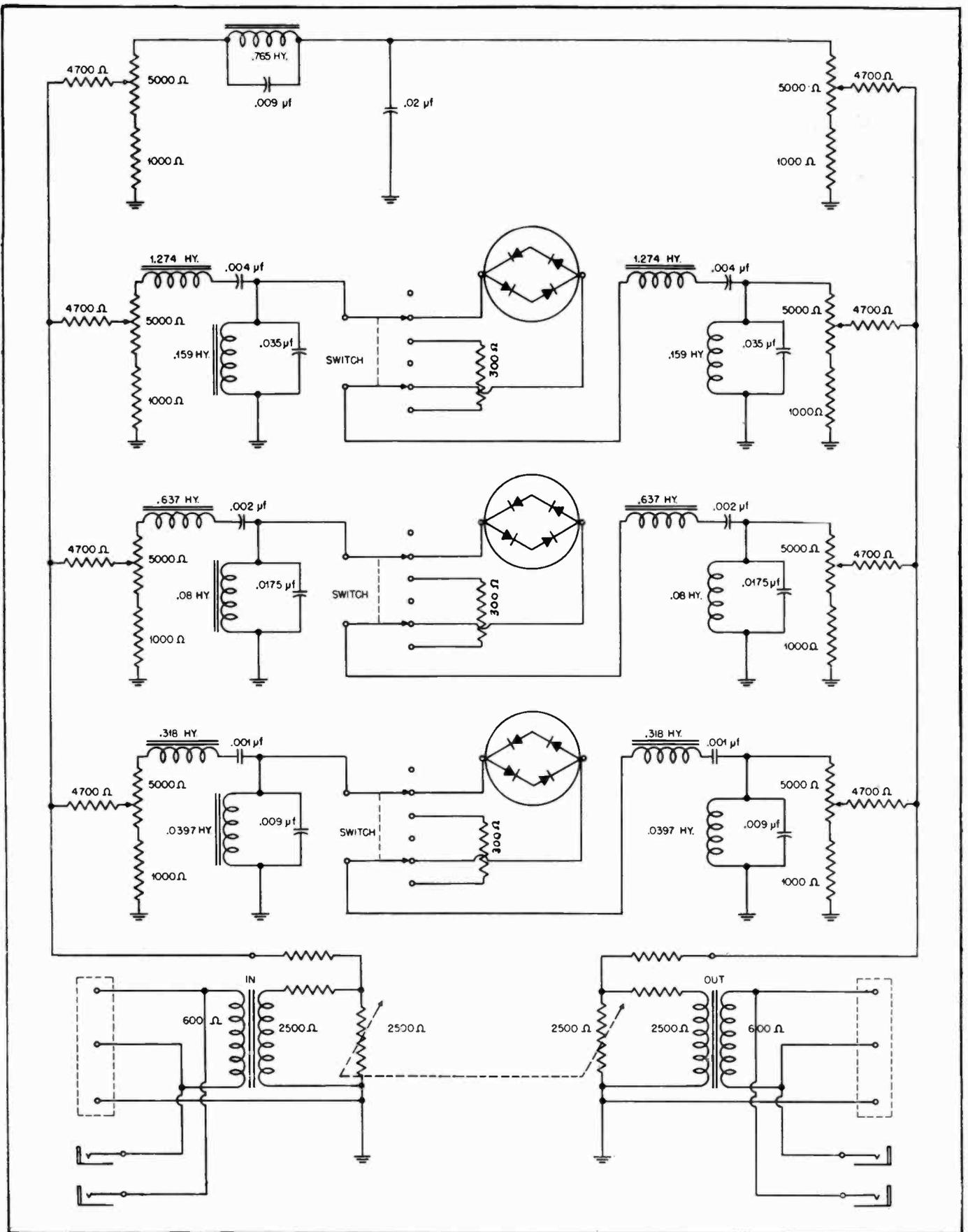


Fig. 5. Complete schematic of the experimental noise suppressor.

equal to six thousand ohms as is the impedance of the low pass "m" derived filter. All three band-pass filters behave similarly so far as attenuation is con-

cerned. (See Fig. 3.) A frequency response curve through both input and output filters shows approximately 30 db attenuation one octave either side of the

roll-off frequency. The surge impedance of 6000 ohms was chosen to represent the best relationship between the high signal and low signal bridging resistance

of the diodes whose resistance varied from several hundred to over 40000 ohms.

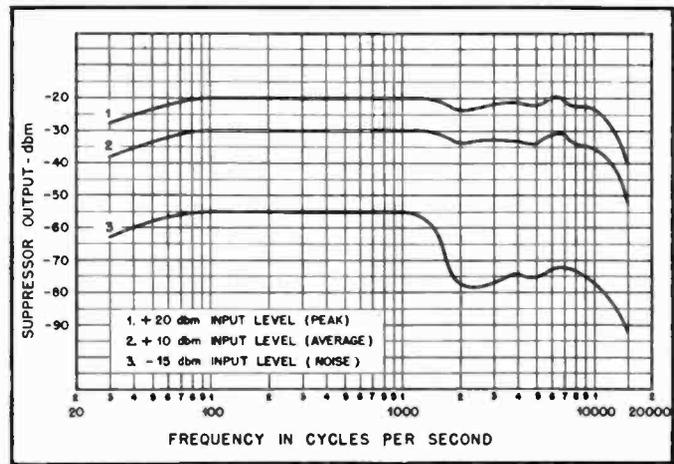
The input and output mixing busses are identical. (See Fig. 4.) All four channels are fed by "L" pads with proper building out resistors which are fed in turn from a matching transformer. A variable "T" pad isolates the transformer from the mixing bus. The output transformer is identical with the input and is also isolated from the output mixing bus by a variable "T" pad. The input and output "T" pads are ganged on the same shaft and are reverse connected so that as the input is raised the output is lowered by the same amount, thus achieving uniform level out while correcting for proper noise level into the diodes. A variation of 15 db input results from this arrangement which has proved adequate for correction of differences in individual recordings. A fixed loss of 15 db results from this arrangement. The input bus, which is composed of potentiometers "L" connected and building out resistors, has a fixed loss of 8.4 db. In general, individual adjustment for each frequency range is made with the corresponding input potentiometer to achieve best signal to noise ratio in that band of frequencies. The output potentiometers are adjusted for uniform response through out the entire band and since some adjustment is required at both input and output bus, a mixing loss somewhat greater than 8.4 db is incurred. Therefore, the total loss in the suppressor is on the order of 35 db. With the constants shown in the diagram, (Fig. 5), the optimum input level was found to be plus 20 dbm and the resultant output, approximately minus 15.

### Crystals

The crystals selected for this equipment are the standard four-element balanced diode modulators mounted in a metal shell provided with an octal base. They are known as the Sylvania type IN40 Germanium-Diode varistor. In the construction of the filters, UTC variable inductors were selected for the reactive elements. These reactors have an adequate range of inductance, satisfactory Q and permit trimming of the final filter to compensate for stray capacities. The capacities encountered in the filter elements are for the most part odd values and parallel connection of two or more standard capacities is required in many instances to achieve correct values. Small postage stamp mica capacitors are used because they lend themselves well to neat parallel installation and occupy a minimum of space.

The input and output coils are selected to match 600-ohm lines to the mixing

Fig. 6. Frequency response curves showing suppressor action at low level as compared with peak and average levels.



buses, whose impedances are approximately 2500 ohms. The mixing bus itself is composed of ordinary composition potentiometers and pigtail composition resistors. Three rotary switches are included on the front panel. Each switch is associated with a pass band. The switches have three positions which selectively switch the diodes into each pass band for suppressor action, replace the diodes with an equivalent resistance for no suppression, or disconnect the pass band completely for test purposes.

The front panel is standard rack width and is  $5\frac{1}{4}$  inches high. (See photo.) The compensated volume control is mounted on the left side of the panel. A double jack for input is located in the lower left corner of the panel and the output jack is mounted directly opposite in the right hand corner. The chassis is No. 16 gauge cold rolled steel. All transformers and reactors are mounted upright on the surface of the chassis and are arranged in rows corresponding to the frequency channels for which they are intended. (See photo.) The potentiometers are also mounted upright through the top surface of the chassis and are located at the input and output of their respective channels. The inductance of the reactors is set with a bridge before being installed in the circuit. After wiring is completed, each individual channel is calibrated separately and trimming of inductors may be necessary to achieve the final band-pass characteristics desired. An audio oscillator is connected to the input terminals and response is read at various points under study with a vacuum tube voltmeter. For a channel response curve through the entire system the input and output pads on all other channels are set to maximum attenuation.

### Signal Source

The Clarkston 16-inch sweep frequency record provides an excellent signal source for final testing of the suppressor. With the aid of an oscilloscope connected across the output of the suppressor, the frequency response can be observed and any

minor equalizing of levels can be accomplished under practical operating conditions. At the same time the pickup and amplifiers associated with the suppressor may be checked for hum noise and proper gain. The optimum input level is plus 10 VU and distortion throughout the suppressor is less than one-half per cent.

In planning a system which is to include this suppressor, several facts must be borne in mind. An input level of plus 10 VU is required which means that a program amplifier must precede the suppressor for practically every application. Secondly, the loss in the suppressor is of considerable magnitude and an amplifier is required to recover the original level before suppression. For example, let us assume that the suppressor is to feed a line at plus 8 VU level. The program amplifier originally driving the line at plus 8 VU must now feed the suppressor at plus 10 VU which presents no formidable stress on the system. However, a booster amplifier will be required to return the output of the suppressor to plus 8 VU the original level feeding the line. The booster amplifier represents additional equipment in this case and must have 30 db gain and be capable of plus 8 VU output.

In addition to its use for suppressing needle noise, this instrument shows considerable promise when used with magnetic tape recorders. Some other applications that seem worthy of consideration are as follows: The suppressor may be used with applause microphones that are located out over the audience. The applause is unaffected, while occasional coughing or room noise is eliminated. Open air concerts often encounter the problem of automobile horns or other undesirable noises which might be eliminated by using this instrument. And finally, moving scenery on television sets while the program is in progress might be accomplished with less unwanted noise by the use of this suppressor.

# Balanced Clipper Noise Suppressor

S. LESLIE PRICE

A simplified noise suppressor with many features. Group listening tests indicate its superiority over conventional filtering methods.

**I**MPROVED PERFORMANCE of pickups and of recording techniques has made it possible to realize a higher frequency response from disc recordings than has been practical in the past. For some years now, increasing numbers of records with the higher frequencies recorded on them have been appearing on the market. In those instances where poor material has been used in making the pressings it has been necessary to tolerate the surface noise or to compromise between frequency response and surface noise. High-frequency pre-emphasis recording lessens the compromise, but this technique seems to be used only in the recording of the leading manufacturers. Also, it must not be forgotten that there are already many recordings in use today which do not embody the pre-emphasis feature. We are, therefore, still faced with the surface noise problem.

The various approaches made in the past to solve this surface noise problem have been watched with much interest and anticipation. Pickups with good response vs. frequency characteristics, low distortion, low transient effects, and low response to displacement of the stylus in other than the desired plane, are now available at a reasonable cost. Amplifiers are not a problem and loudspeakers are continually being improved. However, surface noise remains the stumbling block to the greater enjoyment of much recorded material. The proper place to attack this problem is at its source, i.e., in the manufacture of recordings. However, as already mentioned, existing recordings must also be accommodated.

Pre-emphasis, volume expansion, dynamic noise suppression, and fixed filters

all contribute towards a reduction of surface noise and all leave something to be desired. The author, therefore, set out to see if he could develop something that would be a worthwhile contribution to the solution of the noise-suppression problem.

## Basic Circuit

The noise-suppressor circuit which is described in this article has proven to be effective in reducing the surface noise in record and transcription reproduction, without sacrificing the high-frequency register of the recordings. Observations in the form of listening tests to check the effectiveness of the suppressor were made by small groups of engineers, musicians and laymen. A unanimous preference was shown for reproduction with the suppressor to reproduction using conventional filtering methods.

The basic ideas incorporated in this noise suppressor are similar to those described elsewhere.<sup>1,2</sup> However, this suppressor differs from Olson's in several respects and is capable of covering a wider frequency band while using fewer filter components.

Operation of the suppressor is simplicity in itself. One control adjusts the degree of suppression. Surface noise is reduced while preserving many delicate tones of high frequency such as tinkling bells, triangles, etc., as well as the higher overtones most of which would otherwise be lost.

<sup>1</sup>H. F. Olson, "Audio Noise Reduction Circuits," *Electronics*, Dec., 1947.

<sup>2</sup>Charles D. Cole, "Experimental Noise Suppressor," see page 64.

## Fundamental Requirements

Before experiments were started, careful consideration was given to the factors which would detract from the faithful reproduction of phono recordings and transcriptions. A noise-reducing system for high-quality reproduction of phono recordings and transcriptions should:

1—Be capable of faithfully reproducing, in their proper perspective, all useful signals which are impressed on the recording and should provide wide-range reproduction when wide-range recordings are available.

2—Have various degrees of noise suppression which would introduce a minimum of amplitude, intermodulation and harmonic distortion.

3—Have means of controlling the response vs. frequency characteristic of the system when recordings with objectional distortion are being played through the system.

4—Have automatic control systems or circuits which do not vary the frequency response of the system or the reproduced noise level in such a way that the listener would be conscious of the change.

5—Under average listening conditions, be capable of reducing the noise level of a reasonably good shellac pressing to an unobjectionable level without sacrificing the high frequencies and overtones of the recorded material when used in conjunction with good reproduction equipment.

6—Provide noise suppression during high-level passages as well as low-level passages.

7—Not materially alter the dynamic range of the recording.

8—Not audibly exhibit any delayed action or time constant phenomenon.

## First Experimental Circuit

Incorporation of the above requirements in a noise suppressor become the problem and many of them do not lend themselves to an easy solution. Fig. 1 shows a block diagram of the first experimental suppressor which performed sufficiently well to encourage further investigation with this approach.

The audio spectrum is divided into consecutive bands, the first point of separation being at the highest frequency consistent with reproduced noise that is not of objectionable intensity. Since each

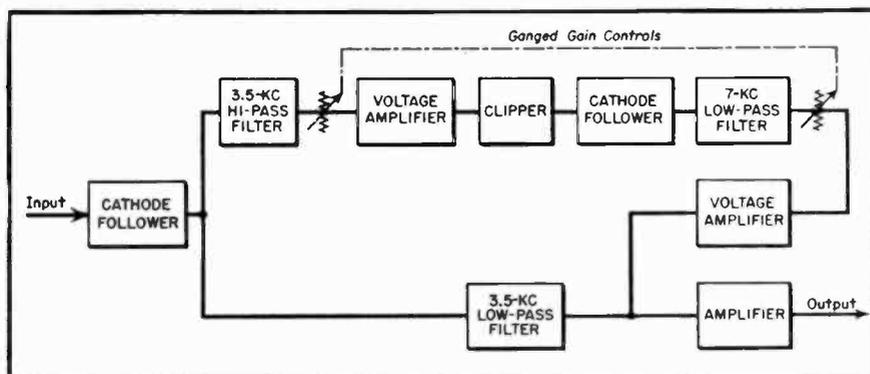


Fig. 1. Block diagram of the first noise suppressor setup, tried experimentally.



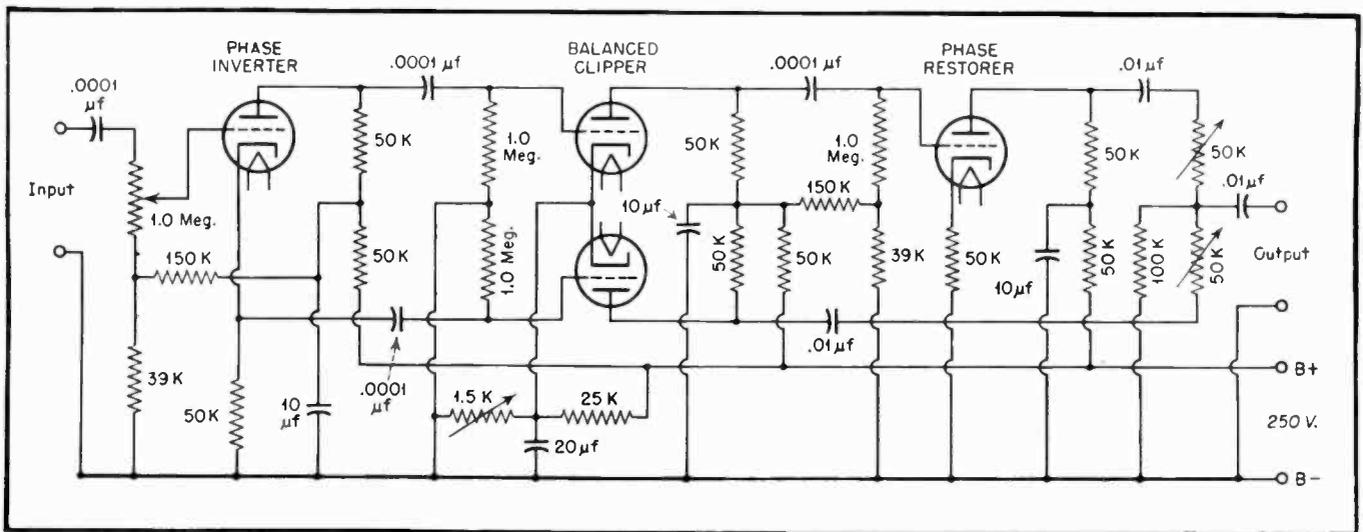


Fig. 4. This circuit may be employed to replace the transformer-terminated design shown in Fig. 3.

proper phase. With the second harmonic reduced to an acceptable level, the next order of distortion will be the third harmonic. The pass bands can then be designed to include slightly less than one and one-half octaves while still providing proper attenuation of the third and high harmonics.

The block diagram, Fig. 2, shows how the second experimental model was arranged to take advantage of this feature. Fig. 3 is a circuit diagram of a suppressor which incorporates these ideas.  $R_1$ ,  $R_2$ , and  $C_1$  adjust the frequency response of the system to give an approximate flat amplitude to the surface noise of shellac records in accordance with Olson's data.  $V_1$  builds the signal up to a level which places the surface noise signals just below the cutoff value of  $V_3$ ;  $V_2$  is a phase inverter;  $V_4$  restores the energy lost in the preceding circuits. The output of  $V_4$  is combined with the output of the low-pass filter in the correct relationship.  $R_3$ ,  $R_4$ , and  $C_2$  restore the frequency response of the system.  $V_3$  in conjunction with  $T_2$  is the balanced push-pull circuit described above.

The frequency response curve of this system is essentially flat up to 8 kc

when adjusted for a normal amount of suppression. Figure 4 is a circuit to replace the transformer-coupled section including tubes  $V_2$  and  $V_3$ .

Cathode followers were used in the experimental models because they conveniently matched the filters which were on hand. Better results will be had when band-pass filters are used preceding and following the clipper circuits. The use of band-pass filters, when used in place of the filters shown, will reduce intermodulation and will increase the effectiveness of the suppressor.

The block diagram in Fig. 5 outlines a circuit which takes full advantage of the ideas brought out above. Note that the first crossover frequency in this arrangement appears at three thousand cycles per second, the second crossover appears at seven thousand cycles per second, and that the third harmonic of seven thousand cycles per second falls beyond the response of many amplifiers and speakers, and is beyond the normal hearing range. It then only becomes necessary to provide sufficient attenuation of the frequencies at the high-frequency limit of the higher frequency channel to guard against intermodulation in this stage.

At the present time, little if any useful signal can be found above the surface noise at frequencies beyond twelve thousand cycles per second on available recordings. The high-frequency cutoff should therefore, appear just beyond twelve thousand cycles per second but can be extended to a higher frequency when recordings and pickups have developed to a point that will justify this extended range.

#### Reproducer Quality

In order that the advantages offered by this type of noise suppressor can be fully realized, high-quality performance of reproducers is essential. The units used for the experimental work and listening tests were of the variable reluctance type and were checked for their performance characteristics before they were put into service. Best results were obtained when compensation for recording characteristics was made before the signal entered the suppressor. Therefore it is not necessary to duplicate the correction equalizers in the suppressor.

Equalizers other than those needed for post recording response-frequency correction are not needed for reproduction of good recordings when the above suppressor is being used. Reproduction of recordings in fair condition is noticeably improved while recordings which are worn or gritty have no place in high quality reproduction systems. The problem of turntable rumble and low frequency noise reduction was not considered in this undertaking inasmuch as the problem has been dealt with elsewhere.

With increased use of materials which result in lower surface noise and with the marked improvements being made in reproducers, loudspeakers, and methods of noise suppression, the realization of higher quality reproduction from recordings is rapidly approaching.

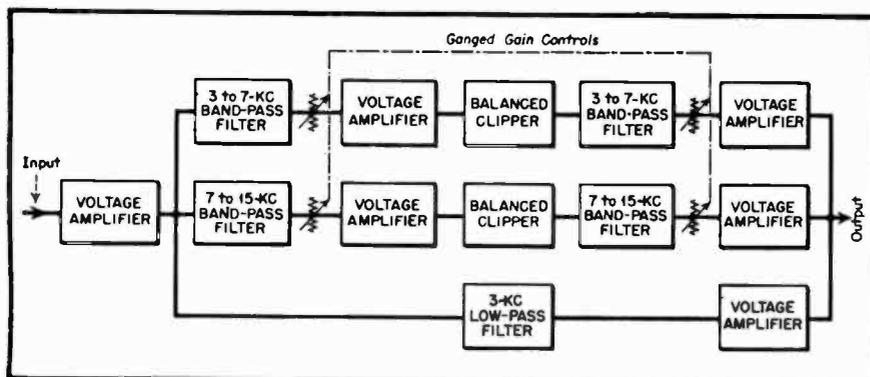


Fig. 5. Block diagram of a circuit with crossover frequencies of 3000 and 7000 cycles.

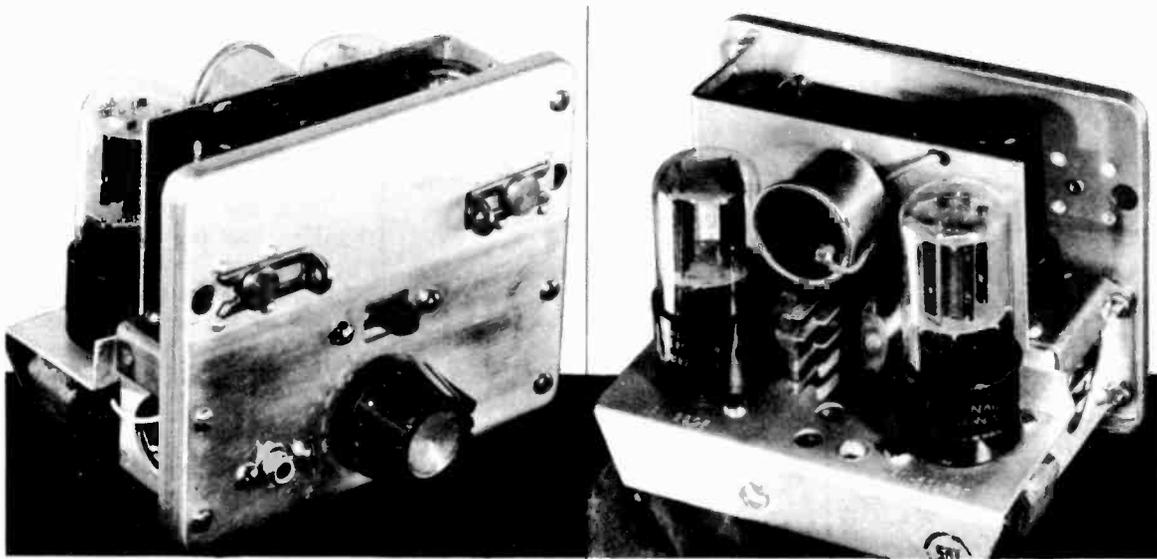


Fig. 1. Equalizer-suppressor amplifier as built on surplus preamplifier chassis.

# Simplified Dynamic Noise Suppressor

C. G. McPROUD

A three-tube preamplifier combining low-frequency equalization and a new type of dynamic noise suppressor for use with magnetic pickups.

ONE of the most outstanding circuit developments in the audio field to be publicized during the last few years is that of the dynamic noise suppressor. While its performance is a definite improvement in record reproduction over fixed filter circuits, it cannot be denied that the circuits are somewhat complicated, and considered by many to be beyond their capabilities for construction.

In attempting to simplify the noise suppressor problem so that the advantages might be enjoyed by more record users, the writer listed a variety of partially related facts and assumptions, from which the circuit to be described was derived.

- 1) To reduce noise, it is desirable to employ a low-pass filter.
- 2) To avoid degradation of musical quality—i.e., a reduction of high-frequency response—the cutoff frequency should be adjustable.
- 3) For convenience, it is desirable to have the cutoff frequency varied automatically by the signal itself.
- 4) Constant-k filters usually “sound” better than filters with a sharper cutoff.
- 5) The qualities of a high-quality magnetic pickup that provide a wider range of

- signal frequencies also cause the reproduction of a higher needle scratch level.
- 6) Magnetic pickups are low-level devices, and require both amplification and low-frequency equalization for use with conventional radio-phonograph amplifiers.
  - 7) Since a magnetic pickup is effectively a generator having an internal impedance which is essentially an inductance, a reasonably sharp cutoff can be obtained solely by the use of a shunt capacitor.

Having these basic premises listed, how can the problem be solved in the simplest manner?

## Design

The use of a magnetic pickup demands a low-frequency equalizer and additional amplification, so the first step in the solution is to set down such a circuit. The high-frequency cutoff may be provided by a shunt capacitor, with the cutoff frequency being varied by changing its capacitance.<sup>1</sup> Suppose, therefore, that the shunt capacitance is supplied by a reactance tube. By varying its grid bias, the cutoff frequency can be changed at will. To make the variation of capa-

citance automatic requires a side amplifier and rectifier, together with some form of manual control.

Putting these separate elements together, the equalizer-suppressor amplifier is now seen to consist of three sections—the equalized preamplifier, the reactance tube across the input, and a control tube consisting of an amplifier and a rectifier. From this point, then, it is possible to design a wide variety of circuits to perform all of these functions.

One such circuit is shown in the schematic of Fig. 2. Built as an accessory unit, it employs a series heater string, a dropping resistor, and a selenium rectifier and filter capacitor, thus operating the heaters from the 115-volt a-c line to avoid overloading the filament windings of the amplifier to which it is connected. Plate current is obtained from a convenient 200-300 volt point in the amplifier with which it is to be used.

The equalized preamplifier consists of the two sections of  $V_1$ , a 12SL7, connected in cascade, and employing

<sup>1</sup>“High Frequency Equalization for Magnetic Pickups,” on page 48.

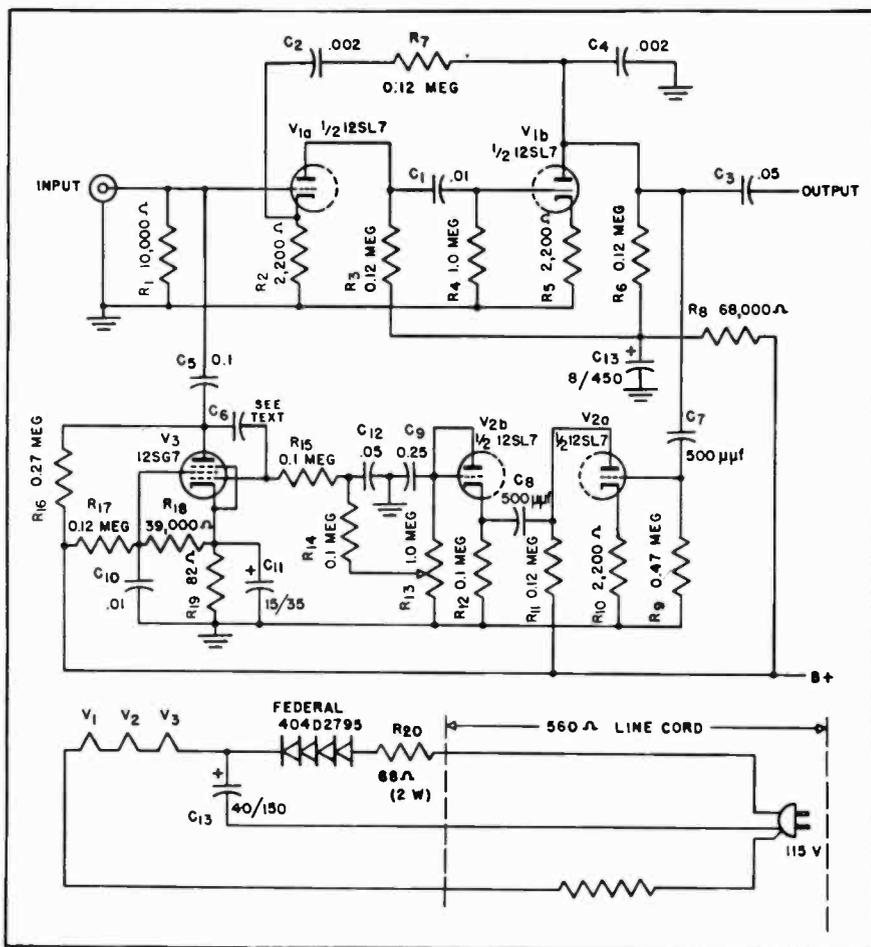


Fig. 2. Complete schematic of dynamic noise suppressor amplifier utilizing single reactance tube directly across magnetic pickup, and furnishing adequate low-frequency equalization to correct for average recording characteristics.

a feedback circuit to provide the low-frequency boost. The voltage amplification of this two-stage amplifier is approximately 25 at 1,000 cps, with a low-frequency boost of nearly 6 db per octave below a transition frequency of 500 cps. This results in a fair compromise for various types of phonograph record characteristics, since the bass tone control on a conventional radio phonograph or amplifier should suffice to make the finer adjustments. However, if desired,  $C_2$  can be made adjustable by a separate switch, using a value of  $0.005 \mu\text{f}$  for a 300-cps turnover, and  $0.0015 \mu\text{f}$  for 800 cps turnover.

The reactance tube  $V_2$  is connected in shunt across the input terminals, isolated by a  $0.1\text{-}\mu\text{f}$  capacitor  $C_5$  to keep d-c voltage off of the pickup. The effective capacitance of  $V_2$  is a function of  $C_6$  and the  $\mu$  of the tube. Therefore, with a given set of operating voltages on this tube,  $C_6$  controls the static cutoff frequency. A value of  $750 \mu\text{f}$  provides a cutoff that is down 3 db at 4,000 cps with the Pickering Cartridge. This value should be increased to  $0.001 \mu\text{f}$  when used with the General Electric variable reluct-

ance pickup. Without going into the operation of a reactance tube circuit, suffice to say that an increase in the negative bias voltage applied to the control grid causes a reduction in the  $\mu$  of the tube, with a consequent decrease in the effective capacitance of the tube.

From the chart shown in the previously cited reference, it is noted that for a 4,000-cps cutoff, the shunt capacitance must be of the order of  $0.02 \mu\text{f}$  for the Pickering, or  $0.03 \mu\text{f}$  for the GE. The effective capacitance of the tube is in series with  $C_5$ , so it is necessary that  $C_5$  be relatively large, as shown. Since the maximum plate voltage is applied across  $C_5$ , its voltage rating must be at least 400 volts. Any high-gain pentode should work in this circuit, but it is desirable to use a tube of the semi-remote cutoff type to obtain smooth operating characteristics. A 12SG7 was chosen for this application.

The side amplifier and rectifier consist of the two sections of a second 12SL7, the first serving as the amplifier, while the second—with the plate and grid connected together—serves as

the rectifier. The coupling circuits between  $V_{1b}$  and  $V_{2a}$  and between  $V_{2a}$  and  $V_{2b}$  are so designed that the response cuts off quite rapidly below 1,000 cps, providing a d-c control voltage which is developed mainly from the upper middle frequency range. Thus, low frequencies do not appreciably affect the reactance tube.

However, when the signal contains frequencies in the upper middle range, the rectifier circuit causes a d-c voltage, negative with respect to ground, to be developed across  $R_{13}$ , a potentiometer used as the manual control.

#### Operation

There are several unusual conditions existing in this suppressor. In the first place, the control signal is obtained from the output rather than from the input. The action, therefore, is aided by the raising of the cutoff frequency, since this raising applies a larger high-frequency signal to the side amplifier. In the second place, there is a small contact potential existing across  $R_{13}$ . This potential is of the correct polarity (negative) to raise the cutoff frequency. Thus the manual control varies the cutoff frequency gradually at the same time that it increases the sensitivity of the automatic control circuits.

In operation, the equalizer-suppressor or amplifier is connected between a magnetic pickup and the usual phonograph input jack. If this connection is followed by the usual RC network to compensate for the high-frequency droop of crystal pickups, this network should be removed. In the schematic,  $C_4$  is used to compensate for the high-frequency boost present in most recordings. The value shown is suitable when the following amplifier and the speaker system is flat. For optimum results, it may be desirable to increase or decrease this capacitance.

When the control  $R_{13}$  is in the maximum clockwise position, the response of the equalizer-suppressor is essentially flat from normal records, with no high-frequency cutoff. This is the result of (1) the contact potential across  $R_{13}$ , and (2) the d-c voltage developed as a result of the residual scratch. Thus, there is sufficient negative bias applied to the 12SG7 to reduce its gain, and consequently to reduce the effective capacitance shunted across the pickup. If the scratch is objectionable, the control may be rotated counterclockwise until the best balance is obtained between musical quality and noise. In the intermediate positions of the control, it will be noted that the noise is reduced during low level passages, but that the noise appears to increase when high level passages—

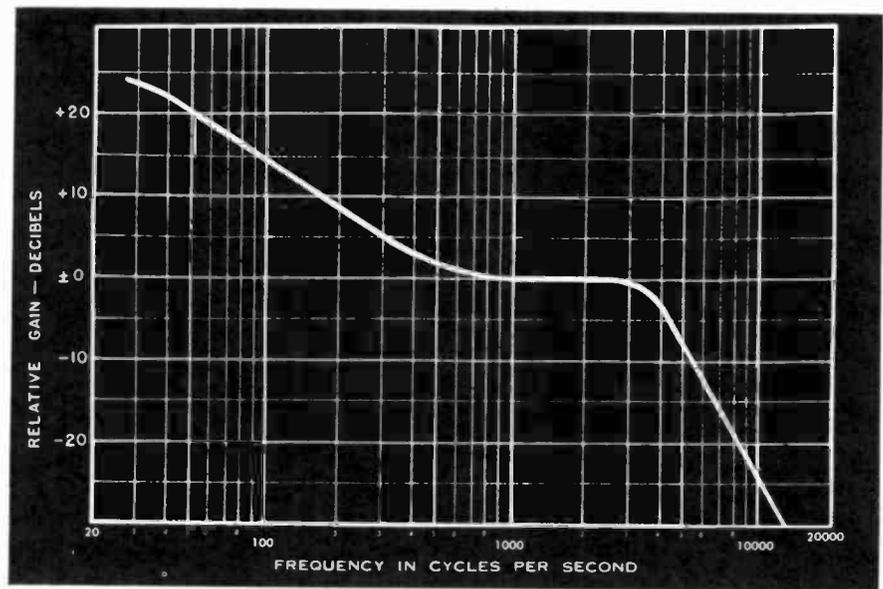
containing high frequencies—are reproduced. The more the control is turned counterclockwise, the greater the reduction of noise. Two actions take place simultaneously as the control is turned: (1) the cutoff is lowered gradually, and (2) the sensitivity of the automatic action is reduced. This results in a single smooth range of control from no cutoff at all to a fixed filter with a cutoff at 4,000 cps, with a gradual increase in dynamic action over the range of the control.

### Construction

The experimental version of this amplifier, shown in *Fig. 1*, was built on a chassis available as a surplus item. However, the amplifier is not limited to such a construction, and any suitable small chassis is suggested. If there is an adequate supply of heater voltage, or if the unit is being constructed as a section of a complete amplifier, 6.3-volt tubes should be used instead of the 12-volt types, and their supply obtained in the usual manner from the power transformer. However, if the equalizer-suppressor amplifier is to be added to an existing radio phonograph, it is advisable to provide a separate heater supply, as shown in the schematic. The connections shown are for use with a 560-ohm line cord and a 200-ma selenium rectifier, furnishing a 150-ma d-c supply for the three 12-volt heaters in series. Since no signal voltage appears across the control potentiometer, it may be mounted on the front panel or the phonograph motor board in a convenient location. Input and output connections should be shielded. The unit shown in *Fig. 1* was built with a switch to cut in or out the suppressor section, but this was found to be unnecessary. The single control is adequate for all operation.

### Parts Hints

The voltage rating of  $C_9$  and  $C_{12}$  need not be over 150 volts, so hearing-aid type capacitors can be used to conserve space. The product of  $R_{14}$  and  $C_{12}$  controls the attack time, which should be short enough to permit operation without appreciable delay, yet long enough to prevent opening the filter circuits on "pops" or "ticks." As shown, the delay is only 5 milliseconds. The release time is the sum of the products of  $R_{13}$ - $C_9$  and  $R_{14}$ - $C_{12}$ , 255 milliseconds in this instance. It appears satisfactory in subjective tests, and since there are no accepted standards for these values, such tests must be relied upon for determination of optimum values. The potentiometer  $C_{13}$  is linear.



**Fig. 3.** Curve of equalized preamplifier with control at minimum, showing cutoff at 4,000 cps and a droop of 15 db per octave above cutoff.

### Operation

Under normal operating conditions, the voltage on the plate of  $V_3$  is approximately 15 volts with no signal and with the control at the maximum counterclockwise position; 40 volts at the maximum clockwise position. In the presence of high-frequency signals with the control at the clockwise position, the voltage at the plate of  $V_3$  is approximately 160. The screen potential is approximately 32 volts with no signal.

The voltages at the plates of  $V_1$  are about equal, approximately 80 volts; the cathode potentials, also about equal, are of the order of 1.0 volt. The voltage at the plate of  $V_2$  is 180; at the cathode, 2.0. A contact potential of 0.7 volts appears across  $R_{13}$  with no signal. The rectified d-c voltages across  $R_{13}$  with signal from a frequency record reach a maximum of about 40 volts at frequencies above 500 cps. These measurements were made with a vacuum-tube voltmeter having an input resistance of 15 megohms. The plate supply was 300 volts, and the potential across  $C_{13}$  was 120 volts.

### Advantages and Limitations

This equalizer-suppressor amplifier has several advantages over the more complicated noise suppressor in that it is much simpler to construct and requires fewer parts. There are no complicated adjustments required in the construction. Since only one control is furnished—or needed—this suppressor is much easier to use effectively. However, it is limited to use with magnetic pickups, and is not usable on radio tuner outputs or with crystal

pickups. While some effect could be obtained, the cutoff rate of 15 db per octave is available only when the source is inductive. It does not have as great flexibility as the H. H. Scott Dynamic Noise Suppressor, but for those who must occasionally compromise cost with performance, it is a definite improvement over the use of capacitors providing fixed cutoff frequencies.

### Additional Notes

The original experimentation with this suppressor was carried out with a Pickering cartridge, which has an output in the vicinity of 50 millivolts, somewhat greater than that available from certain other types. For this reason, those who use this device with GE or Audax cartridges may encounter some difficulties with respect to proper dynamic operation.

Some improvement may be gained by the addition of high-capacitance, low-voltage electrolytics across  $R_5$  and  $R_{10}$ . Values ranging from 20 to 100  $\mu$ f have been tried, with satisfactory results. The addition of these capacitors gives some increase in gain through both the signal circuit and the side amplifier, with the result that a greater signal is impressed on the rectifier,  $V_{2b}$ , thus giving a greater control voltage for the grid of the reactance tube,  $V_3$ . Another change which has given good results is to substitute a 6SJ7 for the 6SG7 indicated for  $V_3$ . When used in conjunction with the two electrolytic by-pass capacitors previously mentioned, satisfactory operation of the suppressor should result.

# Two-Way Speaker System

C. G. McPROUD

A three-part article describing the design and construction of an excellent two-way speaker system.

## PART I

**T**HE 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.

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.

### 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 is discussed in the article on

page 101, and will not be covered 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 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  $\frac{1}{4}$  in. square, since any smaller opening would be difficult to work with. This provides a total throat area of  $8 \times \frac{1}{4} \times \frac{1}{4}$  in., 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 adja-

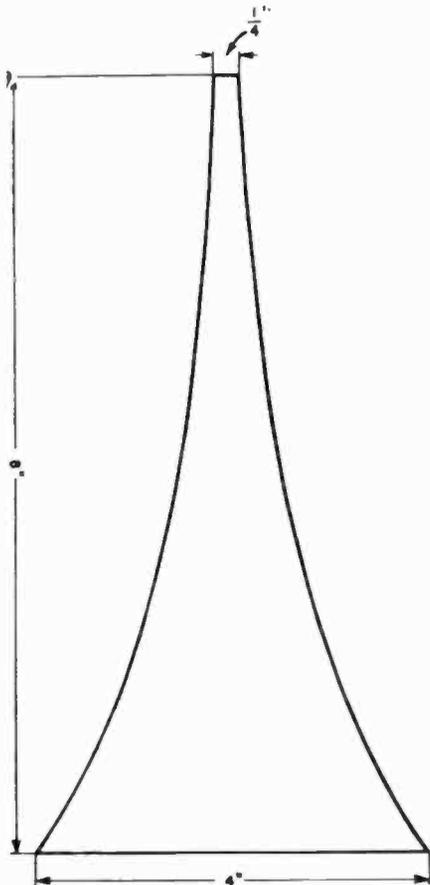


Fig. 1. Developed section of exponential horn, doubling in cross-sectional area for each inch of length along the axis.

cent 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 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  $4\frac{1}{2}$  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  $\frac{1}{4}$ -in. throat at the top and a  $3\frac{15}{16}$ -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  $\frac{1}{4}$ -in. width at the top and a  $4\frac{7}{16}$ -in. width at the bottom. This allows for the 4- and  $4\frac{1}{2}$  in. 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  $\frac{1}{2}$  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  $\frac{1}{4}$ -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

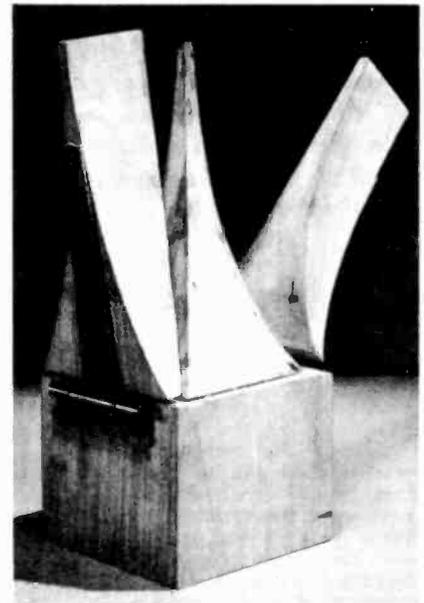


Fig. 2. Wood forming block used for shaping and soldering the individual horn sections to uniform dimensions.

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 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

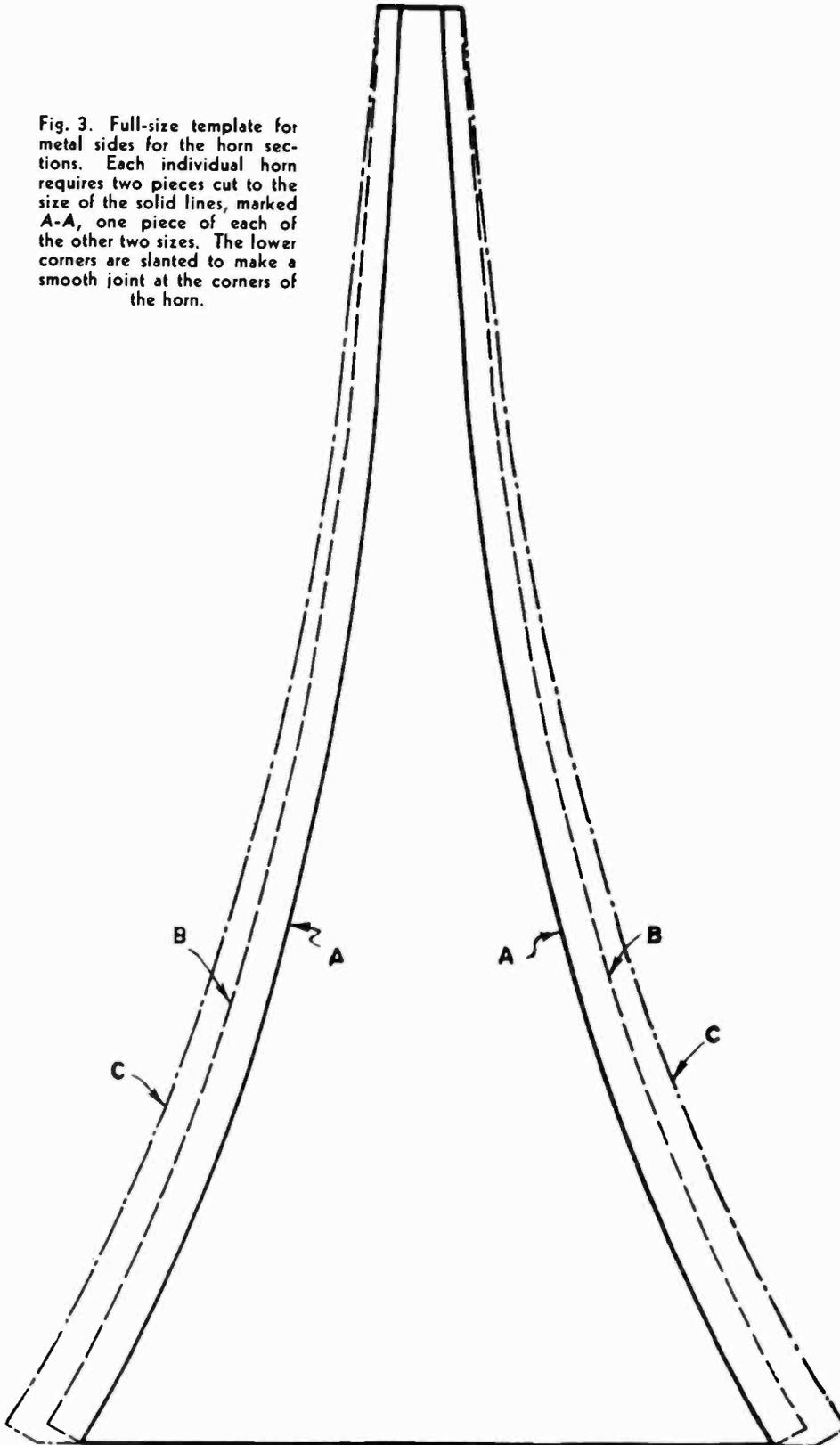
After the eight horn sections are completed, they must then be assembled into pairs. It will be noted that each horn mouth is trapezoidal in shape. Holding a pair of horns together with the long sides of the openings adjacent to each other, and the throat ends aligned, solder the corners of the mouth 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  $\frac{1}{4}$ -in. overlap to bend over the top and bottom covers, and bringing the ends flush with the mouth openings. The throat end should be  $\frac{1}{4}$  in. short of reaching the end of the horns. Before soldering on, bend the upper and lower lips over at an angle of  $90^\circ$ , 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

**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.



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 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

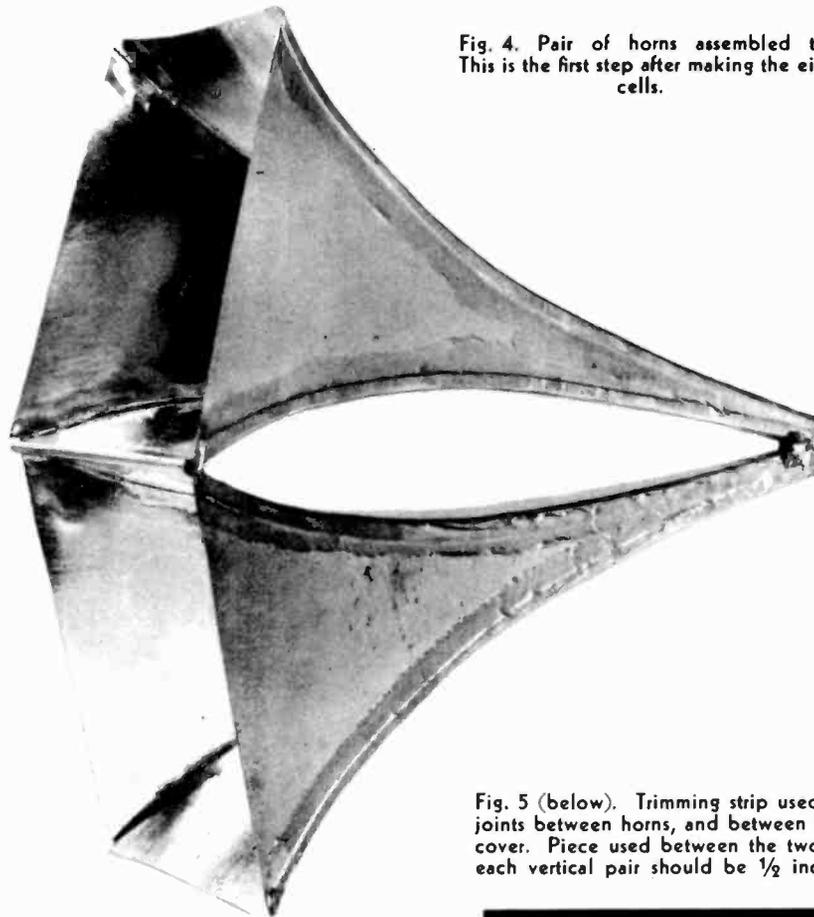


Fig. 4. Pair of horns assembled together. This is the first step after making the eight horn cells.

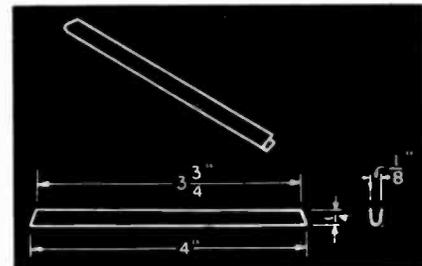
made, and is placed in the sand mould to provide the opening through the casting. It is moulded in a "core box" which is also made of wood to provide a form of proper shape. After the casting is made, it is machined to make a tight joint at the shoulder, and threaded to fit the speaker unit. The castings 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*

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.



*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.



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

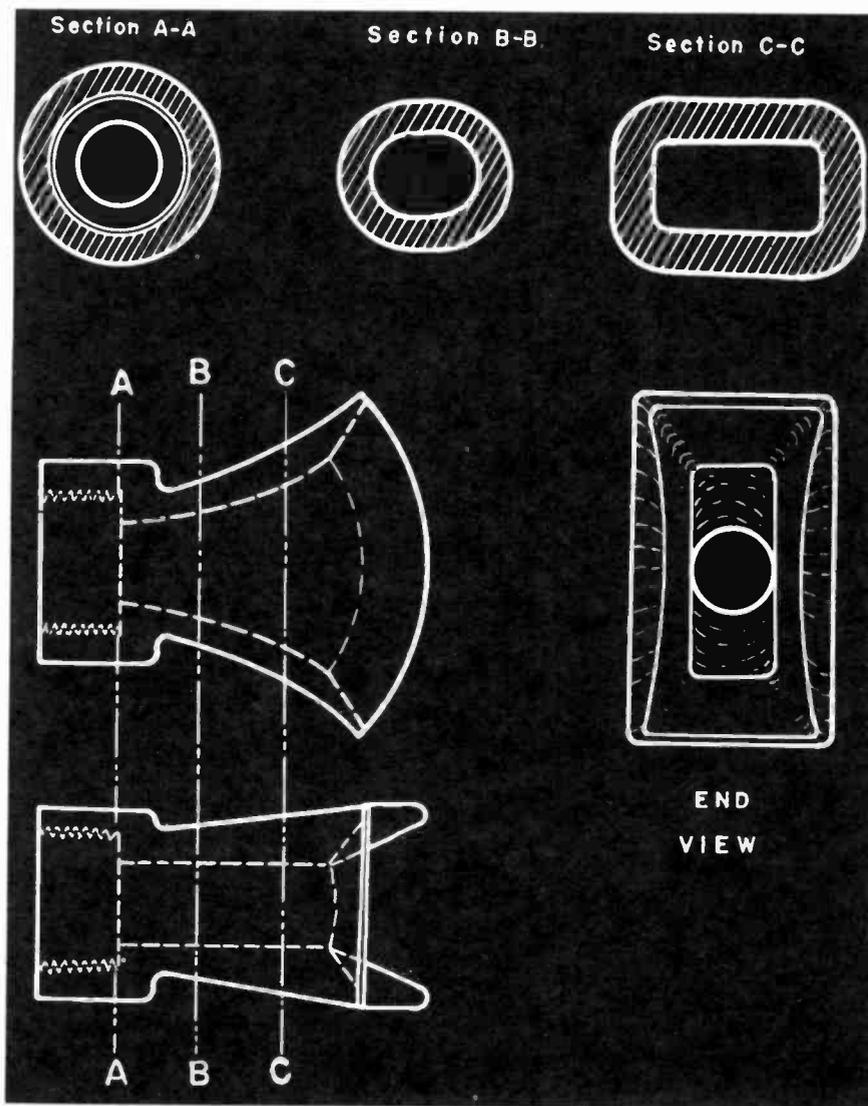


Fig. 7. Plan, elevations, and sections of horn throat coupling, approximately 9/10 full size. Center hole is  $\frac{1}{2}$ -inch diameter.

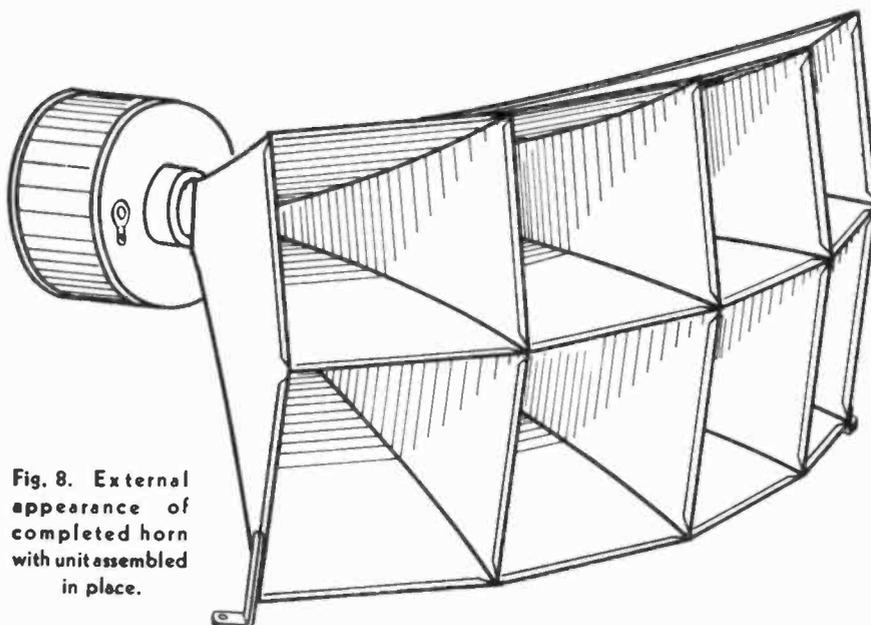


Fig. 8. External appearance of completed horn with unit assembled in place.

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.

## PART II

### The Low-Frequency Section

The low-frequency section of a two-way system may consist of a folded horn, a plain baffle, an infinite baffle, some form of acoustic labyrinth, or a reflexed enclosure. While the best performance can usually be obtained by the use of some form of folded horn, there is some question as to whether the response between about 400 and 900 cps is satisfactory, since the upper frequencies passed by the low-frequency speaker are required to traverse the turns in the horn, and some loss may be expected in that region.

The flat baffle is the simplest arrangement, but it is deficient in the very low register, even if it becomes of large dimensions. The infinite baffle, such as the wall of a room with the speaker itself being in another room, or in an attic, is reasonably satisfactory, and is preferred to the plain baffle.

The acoustic labyrinth type of speaker requires a lot of complicated design and construction, and it is doubtful if it is any better than a cabinet form of mounting, provided some means is taken to reduce the effect of cone resonance. Some speakers are suited for mounting in a completely enclosed cabinet of smaller dimensions than normal, with the stiffness of the enclosed air serving as the damping.

For over-all use, considering the space required, the radiation efficiency of the speaker at low frequencies and the external appearance—which must be considered unless the user is a bachelor—make the reflexed cabinet a logical choice. It should not be assumed that this is the ideal speaker housing for the reproduction of low frequencies, but considering the compromises that must be made occasionally, it has certain advantages.

For these reasons, the low-frequency speaker selected for use with the two-way system described here is of the reflexed type, of a size consistent with the speaker selected.

### Reflexed Cabinet Design

The design of a suitable reflexed cabinet requires the possession of considerably more information than is usually available to the constructor, besides requiring a knowledge of the basic principles of acoustics and the methods of

calculating acoustic compliance, mass, and a variety of other characteristics. Among this data that must be at hand is the resonance frequency of the loud-speaker cone, and in only a very few instances is this figure available from the manufacturer without considerable trouble.

On account of this, a much simpler empirical method of determining the size of a cabinet is suggested. This method is based on the physical size of the speaker, and while it may be far from the classical method of designing a box, it is a slight improvement over one method employed in the design of an early commercial speaker enclosure. The engineer in charge of this design admitted that the box was as big as he could make it and still get it into the back of his sedan.

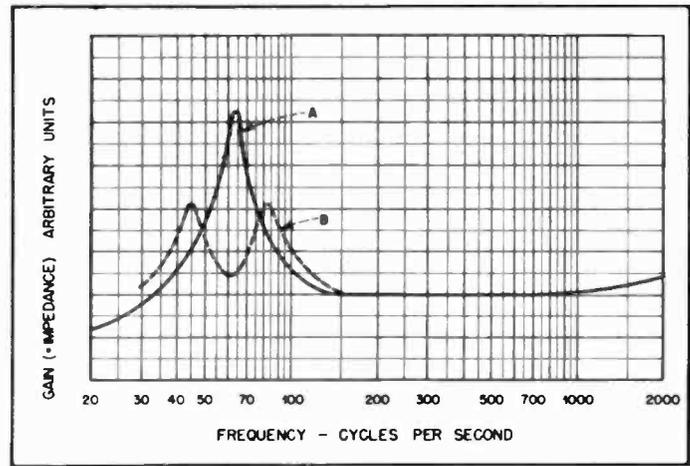
Unfortunately, this particular box was not sufficiently large, and the over-all result of the device was to introduce a peak in the response at about 70 cps, but in spite of that, the speaker was better than any others then available.

The empirical method is based on the following figures: Use a box which has a volume in cubic feet which is numerically equal to the nominal radius of the speaker in inches. Thus, using this method, a 15-inch speaker requires a cabinet volume of 7.5 cubic feet. This is not too far from a reasonable proportion, as has been determined by experiment. The port area is claimed by good designers to be equal to the actual radiating area of the cone, which is somewhat less than that calculated from the nominal diameter of the speaker. For example, a certain 15-inch speaker requires a 13½-inch hole for mounting, but the actual diameter of the conical section of the cone itself is only 12½ inches. Thus the suggested port area should be 123 square inches. Experiments have indicated that this is somewhat too large for the average speaker and enclosure, so resorting to an empirical formula again, the port opening may be taken as 0.8 times the area of the loud-speaker opening, to allow for adjustment to the optimum size.

While the design of a suitable box may be done by means of a number of complicated formulas, the final result—as with anything in the audio field—is how it sounds when completed. Therefore, having decided upon a reasonable port size, it is much simpler to make the box and then experiment with the output until the result is satisfactory. This may be done by simple experimentation with the box, an amplifier, and an audio oscillator, in a method to be described later.

The reflex cabinet serves principally to increase the radiation of a loudspeaker at frequencies below those where it normally loses its efficiency. A secondary reason for the box is to alleviate the effects of cone resonance, which normally occurs at a frequency where it is objec-

Fig. 9. Curves indicating speaker impedance obtained by measurement of gain of an amplifier with pentode or beam-power output tubes. A represents curve of speaker mounted in open back cabinet, showing effect of cone resonance. B represents curve of speaker mounted in reflex cabinet with port adjusted so that both peaks are of equal amplitude.



tionable. If the increased impedance at resonance can be corrected by the design of the cabinet, the response is sure to be better in that region.

### Resonance Effects

The effect of speaker resonance manifests itself as an increase in impedance at the resonant frequency. While this in itself is not particularly important if triodes are used in the output stage, it makes a considerable difference when pentodes or beam tubes are used, unless there is a sufficient amount of inverse feedback to stabilize the output impedance of the amplifier. If a curve of the impedance of a speaker is made with an open-back cabinet, it will be found that it resembles that shown in Fig. 9. Such a curve can be made by measuring the gain of a beam-power or pentode amplifier with the speaker as the termination, and covering the frequency range from about 30 cps to at least 300 cps. The impedance of the speaker affects the gain of the output stage, and consequently the gain-frequency curve represents the variations in impedance.

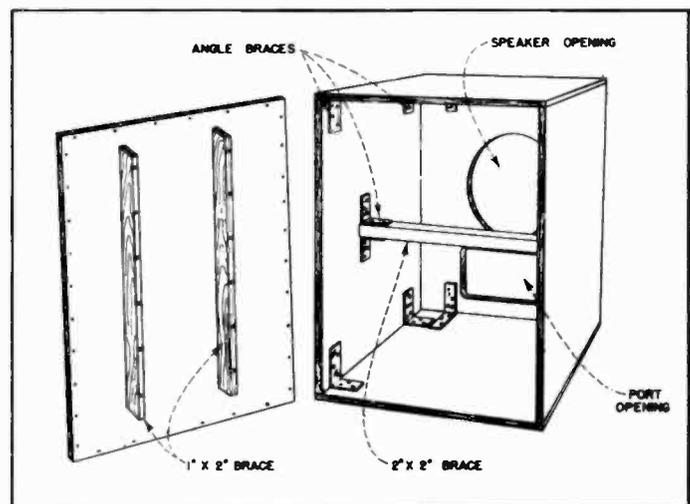
When the back is put on a ported cabinet, the gain-frequency curve exhibits two peaks, one above the resonant frequency, and one below. Optimum

results are obtained when the two peaks are of equal amplitude, as shown at (B) in Fig. 9. The relative amplitude of the two peaks can be varied by changing the size of the port, and the port opening should be so arranged that the two peaks are equal.

With this in mind, it is seen that the entire speaker enclosure may be constructed with a number of empirical formulas, and then by changing the size of the port to obtain the correct response curve, the optimum performance can be obtained. From a construction standpoint, it is advisable to make the port somewhat larger than the anticipated final size, so that the effective opening can be reduced by the use of an additional piece of wood mounted inside the cabinet.

Any good 12- or 15-inch speaker may be used with this system. In order to simplify the dividing network, it is necessary that the impedance of the low-frequency speaker be 8 ohms, since that is the impedance of the high-frequency unit selected for use with this system. While it is possible to design a dividing network which will feed two speakers of differing impedances, it is much simpler to select a low-frequency speaker of the correct impedance in the first place.

Fig. 10. Internal bracing of heavy plywood cabinet used for low-frequency speaker. All joints should be glued, and assembly should be made with wood screws rather than nails. Back must be thoroughly braced, and should have adequate number of screws to hold it securely to prevent vibration.



Other speakers suitable for this type of enclosure are the Jensen P12-N and P15-N in the 12- and 15-inch sizes, or the PMJ-18 in the 18-inch size, although the latter is quite expensive. The General Electric 1201C is an excellent 12-inch speaker, and the Lansing Signature D-130A is preferred by many constructors. Another extremely efficient speaker for this purpose is the Altec-Lansing Model 803, which is used in their Model 800 two-way system shown in *Fig. 11*. However, any of the speakers listed will give good results provided the enclosure is adjusted to work with the unit selected.

Several manufacturers also make suitable reflex enclosures, and one of these may be used instead of going to the trouble of making the housing. However, in order to obtain the best results, the housing selected should be one which is intended for use with the particular speaker chosen for the system.

### Cabinet Construction

The actual construction of the cabinet presents no particular problems. Considering, for example, an enclosure for a 15-inch speaker, it is desired to have a volume of approximately 7.5 cubic feet. The cabinet proportions required to make an over-all structure which is pleasing to the eye are such that the height must be approximately 1.5 to 1.6 times the width. Since the upper section must contain the high-frequency horn and unit, the top 10 inches of the entire speaker must be deducted from the total height in calculating the possible size of the enclosure itself. Thus, assuming a 7.5-cubic foot box the net proportions of the enclosure may

well be approximately 1.2:2 $\frac{1}{2}$ , and from this proportionality the actual dimensions of the box can be determined. Allowing a wall thickness of one inch, the outside of the box for a 15-inch speaker arrives at a size which is approximately 16 x 30 x 35.

The construction should be as solid as it is possible to make it within practical limits. Three-quarter or seven-eighth inch five-ply is suitable, and while somewhat difficult to obtain under present conditions, is recommended for the enclosure. If the plywood panels are assembled with glue and wood screws and braced with steel angle brackets on the edges of sides and front, it will be sufficiently sturdy. It is a good plan to install a 2 x 2 brace through the center of the box from side to side, making it detachable so the speaker may be mounted. Such a detachable brace, however, must be solid when in place, and heavy bolts should be used for attaching it.

The back cover for the box should be arranged to be attached with heavy wood screws, and should preferably be braced with two pieces of 1 x 2 mounted edgewise, running from top to bottom. *Figure 10* shows the suggested bracing for the entire cabinet.

After the box is completed, it should be given a coat of shellac on the inside, and then lined with a sound absorbent material. Rock wool blanketing of 2-inch thickness is especially desirable for this purpose, but if it is not readily obtainable, the box may be lined with two thicknesses of Ozite, the padding material used under rugs. This material should be glued to the cabinet, and tacked with large-headed nails at six-inch intervals. Sufficient clearance should be left around the speaker opening to permit its installation directly against the front of the cabinet, and since the port opening is not yet determined finally, it is suggested that the space between the bottom of the port and the bottom of the front panel be left uncovered.

Before mounting the speaker, a circle of expanded metal should be obtained for installation in front of it as a protection against damage to the cone. The speaker is best mounted with "T"-nuts on the outside, using 10-24 screws through the speaker frame and the front of the cabinet. This type of mounting permits the removal of the speaker, whenever desired, with the greatest ease. When the speaker is in place, with a suitable lead extending through the port, the back may be screwed on.

### Adjusting the Port

Assuming that the port has been cut in rectangular shape with an area of approximately 0.8 times the speaker opening, the speaker is connected to an am-

plifier and a gain-frequency curve made. Assuming also that the amplifier is flat to below 40 cps, the resulting curve is then inspected to determine if the two peaks are of equal amplitude. If they are, the construction is completed, and no further adjustment will have to be made to the box. If, however, the two peaks are dissimilar, experiment with a piece of wood in front of the port until they are alike. Noting the amount of the port area that is covered by the external wood, remove the back and install an equal sized block of wood on the inside of the cabinet over the opening, with another piece of expanded metal covering the port. Then finish the lining of the front panel.

It may possibly be found that the port is too small, and it will then be necessary to enlarge it to obtain the desired result. However, this is a matter of experimentation, and will vary with each speaker used. No specific instructions can possibly be given unless the exact dimensions of the cabinet are known, together with the resonant frequency of the low-frequency speaker.

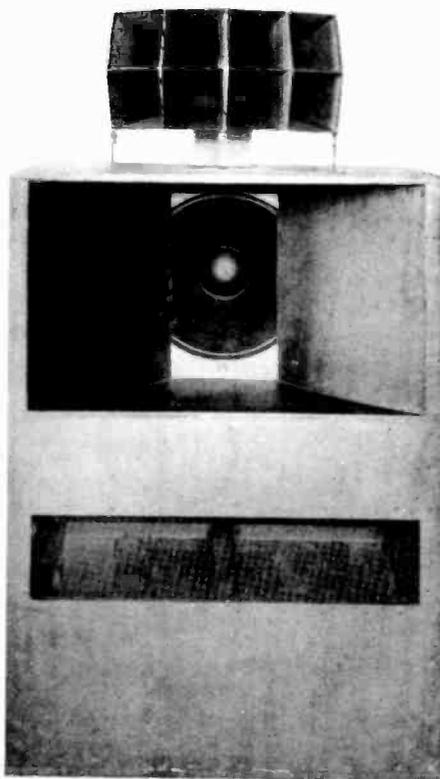
When the port dimensions are completely determined, the exterior of the box may be painted, stained, or finished in any way desired. However, since the high-frequency unit and horn are to be mounted on top of it, it may be that some other housing must be provided in order for the entire speaker to have a passable eye appeal. *Fig. 11* shows a typical arrangement for the complete speaker, without any provision for improving the appearance. Before going further with the steps necessary to complete the external appearance, it is desirable that the final design be determined. The complete speaker is to consist of the reflex cabinet for the low-frequency speaker, the high-frequency unit and horn, and a dividing network.

## PART III

### Dividing Network

The dividing network consists of two coils and two capacitors, and serves to feed the low frequencies to the cone speaker and the high frequencies to the horn speaker. There are several reasons for this—the most important being to prevent the high-frequency unit from being damaged by the high amplitude of the low-frequency excursions which would be set up in the small diaphragm, inadequately loaded for those frequencies. A second reason is to prevent the high frequencies from reaching the cone and causing the inevitable breakup which occurs in that type of speaker unless it is especially designed for the purpose.

The dividing network used with this



**Fig. 11.** Typical commercial form of two-way speaker system employing separate high- and low-frequency speakers.

system is the series filter type, with the circuit shown in Fig. 12. Since both the specified high-frequency unit and the low-frequency cone type suggested for this system were eight-ohm units, the dividing network is calculated for an eight-ohm circuit, and the input to the network should be connected to an eight-ohm winding of the output transformer.

The choice of network circuits is discussed in the article on page 101, to which the reader is referred for further details, but formulas for the four components of the dividing network are as follows:

$$L_1 = \frac{159 R_0}{f_c} \text{ mh}$$

$$L_2 = 0.625 L_1 \text{ mh}$$

$$C_1 = \frac{159,000}{f_c R_0} \mu\text{f}$$

$$C_2 = 1.6 C_1 \mu\text{f}$$

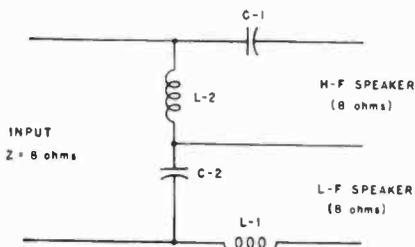


Fig. 12. Basic dividing network circuit used for this speaker system. This is the series filter type.

With a crossover frequency of 900 cps and an impedance of 8 ohms, the values are determined to be as follows:  $L_1 = 1.413$  mh;  $L_2 = 0.883$  mh;  $C_1 = 22.1 \mu\text{f}$ ;  $C_2 = 35.3 \mu\text{f}$ . Taking refuge again in empirical values, it may be stated that the two inductances may be made by winding on forms  $1\frac{1}{4}$ " in diameter and  $\frac{3}{4}$ " in width, using wood flanges to keep the windings in place. This size of form, when used with No. 17 DCE wire—as used in the field coil of Western Electric 555W units—will wind about 13 turns per layer. Referring to the charts shown on page 103, it is determined that  $L_1$  requires 185 turns and  $L_2$  requires 146 turns. For optimum results, these should be adjusted with the aid of an accurate bridge, but with reasonable care in the winding the coils to these specifications, the results should be acceptable.

The accumulation of sufficient capacitance was considerable of a problem until surplus material became available. The writer had an arrangement with a capacitor manufacturer long before the war to provide the values required for dividing networks at eleven cents per microfarad, furnished to exact required values. However, since 10- and 15- $\mu\text{f}$  capacitors are now readily obtainable at six to eight

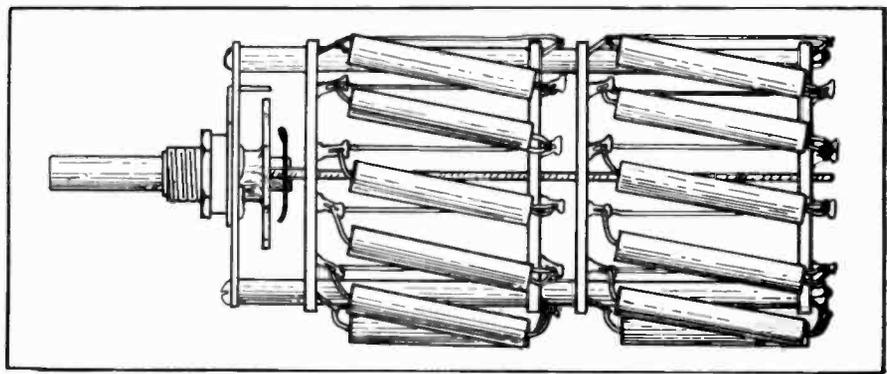


Fig. 13. Method of arranging switch sections to support the resistors for the 1 db/step attenuator in the h-f speaker circuit.

cents per microfarad, it is much easier to use these units. Two 10- $\mu\text{f}$  units and one 2- $\mu\text{f}$  unit should suffice for the 22.1  $\mu\text{f}$  capacitor, while two 15- $\mu\text{f}$  units and 5- $\mu\text{f}$  unit approach the 35.3- $\mu\text{f}$  capacitor. The final values can be built up by the use of smaller units to the correct values, using a bridge for the measurement.

### High-Frequency Attenuator

Since the high-frequency speaker unit is more efficient than the low-frequency cone, an attenuator will be required in the h-f output of the network in order to balance the sound output from the two speakers at crossover frequency. The simplest arrangement is to use a 10-watt adjustable resistor, with a value of 10 ohms, connecting the speaker unit to the tap. However, this does not give an easily variable adjustment, and in general, those who experiment in audio equipment prefer rather more flexibility than is afforded by a semi-permanent adjustment.

The recommended high-frequency attenuator is a step potentiometer, with attenuation values of zero to 6 db in 1-db steps. Such a device is not readily available on the market, but may be constructed on a standard switch. The switch required is a Centralab K-123 index assembly with two "A" decks and two "B" decks. The switch should be assembled with the two "A" decks spaced about  $1\frac{1}{2}$ " apart, followed by the two "B" decks also spaced  $1\frac{1}{2}$ " apart, and with a  $\frac{1}{4}$ " spacing between the two pairs. This construction, shown in Fig. 13, permits the mounting of 1-watt resistors directly on the switch, and parallels two decks for each circuit, thus increasing the current carrying capacity. The circuit of the attenuator is shown in Fig. 14, the resistors being IRC Type BW-1 in the values shown. Fig. 14 also shows the connection of the various switch sections.

A simpler arrangement for the high-frequency attenuator is to employ a 6-ohm L-pad in the circuit shown in Fig. 15. The regular L-pads provide attenuation from zero to infinity in their total rotation, which is more than is desirable for this application. When

connected as shown, the total loss is approximately 12 db—still more than necessary, but an improvement over the usual connection.

### 10-kc Suppressor

When the speaker system is to be used for reproduction of radio programs from a high-fidelity receiver, some trouble may be experienced from the 10-kc inter-channel squeal. A simple suppressor can therefore be installed in the high-frequency horn circuit, and if properly adjusted, it will suppress 10 kc quite effectively without appreciably affecting the remainder of the frequency band.

This suppressor takes the form of a null circuit, shown in Fig. 16. The coil is a 0.5 mh unit, composed of 110 turns of No. 17 DCE wire wound on the same type of form as used for the dividing network coils. The two capacitors resonate with the coil, forming a low-impedance shunt across the h-f speaker circuit. The resistor, 10-ohms, 10-watt adjustable, provides resistance neutralization for the coil, and makes the attenuation curve of the equalizer extremely sharp, as shown

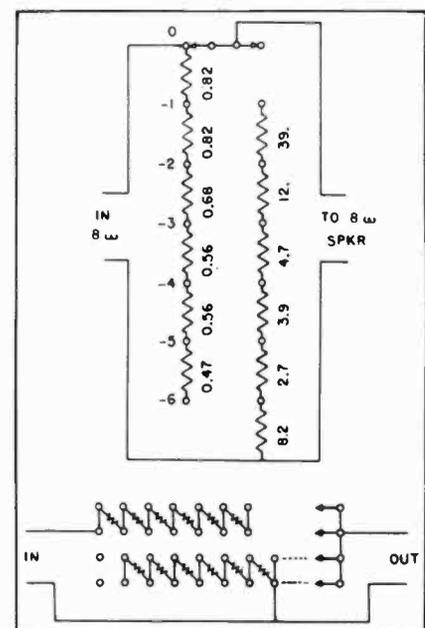


Fig. 14. Circuit arrangement for h-f attenuator, showing resistor values for an 8-ohm circuit.

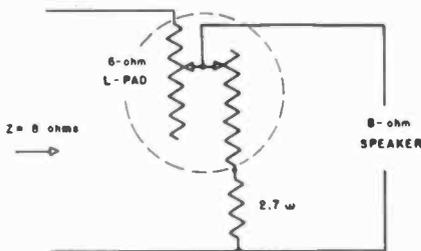


Fig. 15. Alternate arrangement showing the use of a 6-ohm L-pad in an 8-ohm circuit to provide a maximum loss of 12 db.

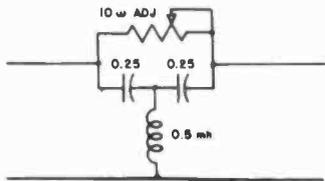


Fig. 16. Circuit of 10-kc suppressor useful for eliminating interstation squeal from wide-range radio receivers.

by the curve of Fig. 18. This suppressor should be arranged for switching in or out of the circuit by means of a switch, Mallory 2006-L push-button type which locks in either position. Since the suppressor causes a loss of approximately 3.5 db throughout the transmission band, an additional fixed pad of this value should be connected across the switch terminals in the filter-out position, to equalize the levels. This pad is shown in Fig. 17 and consists of two resistors, each being a BW-1, 1-watt type.

### The Complete Circuit

The combined circuit of the dividing network, the attenuator, the 10-kc suppressor, and the other connections are shown in Fig. 17. Note that four jacks are inserted between the output of the switching circuit and the speakers, two in each circuit. This permits the insertion of a plug into either circuit for measurement

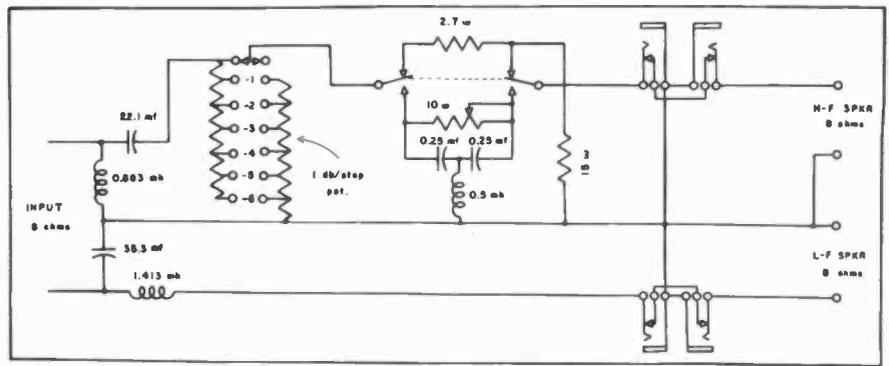


Fig. 17 Complete circuit of dividing network, h-f attenuator, 10-kc suppressor, and jacks to permit access to the various circuits.

purposes, or permits feeding a signal to either speaker without using the dividing network. This provides sufficient flexibility for the most enthusiastic experimenter.

After the completion of the entire switching circuit, it is advisable to make frequency-response measurements of both legs of the dividing network and the 10-kc suppressor. The resulting curves should resemble those of Fig. 18, with the voltage across the two sections being equal at points removed from the crossover frequency, and with both outputs being down 3 db at crossover. The 10-kc suppressor should be adjusted for frequency by adding or removing turns from the coil, assuming that the capacitor values are reasonably close to the specified 0.25  $\mu$ f in each section. After arriving at the correct attenuation peak, the maximum attenuation may be obtained by an adjustment of the 10-ohm shunting resistor. When correctly adjusted, the attenuation at 10 kc should be approximately 40 db, with the response flat at about 9,000 and 11,000 and down 6 db in the vicinity of 9,600 and 10,600 cps.

### Final Assembly

The dividing network, attenuator, and

10-kc suppressor—if used—should be mounted suitably on or in the low-frequency baffle cabinet. If the entire speaker is to be hidden from view, it is possible to mount these parts on top of the cabinet at the sides of the high-frequency unit. In any case, the controls should be accessible, but the special requirements of the physical design may dictate the actual placement of the parts and the controls. It should be remembered that the currents involved in a low-impedance speaker circuit are relatively large—at one watt, for example, the current in an 8-ohm circuit is approximately 0.35 amps. On account of this, the wire used for the connections should be relatively heavy, particularly when higher powers are involved. The choice of switches for such circuits is important—ordinary toggle switches are not satisfactory for speech circuits at these low impedances.

The high-frequency horn and unit should not be permanently mounted until the exact location is determined. To make this determination, it is desirable to have a microphone and an amplifier, together with an output indicator, for best results. With the h-f horn and unit connected to the circuit, and placed on top of the low-frequency cabinet, the two should be energized by a signal at crossover frequency. Then, with the microphone about six feet in front of the combined speakers, move the high-frequency horn and unit back and forth until the maximum output is obtained, as indicated by the output meter on the amplifier connected to the microphone. If the position of the horn is too far forward or too far back of the front of the cabinet, the two leads to the high-frequency unit should be reversed and the procedure repeated. This will give a new position to the horn approximately 7.45 in. from the first position.

The correct position for the h-f unit can be determined roughly by calculation. The dividing network used causes a phase shift of 221° at crossover, which corresponds to 9.2 in. at 900 cps. Thus the two diaphragms must be displaced by that amount, or the phase of the h-f unit may be reversed, requiring a dis-

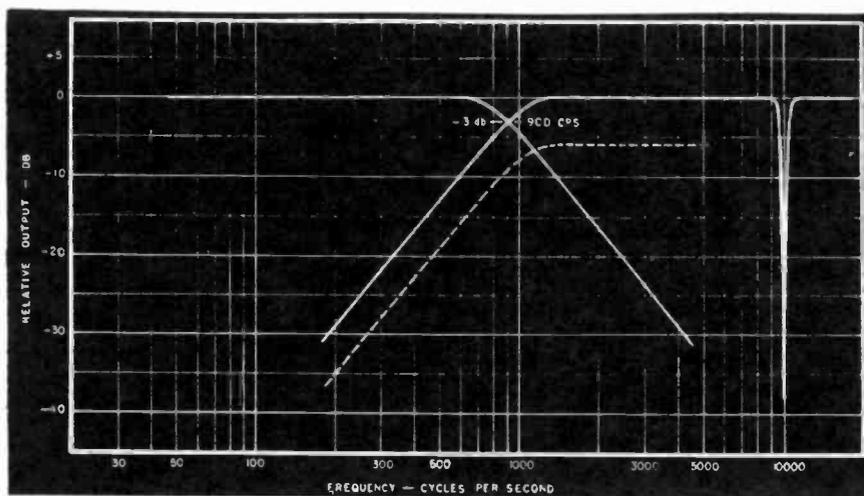


Fig. 18. Curves of typical response from dividing network, and attenuation to be expected from 10-kc suppressor. Dotted curve shows range of adjustment furnished by 1 db/step attenuator.

placement of 9.2—7.45 in. or 1.75 in. However, the diaphragm of the low-frequency cone is not a plane surface, and the exact point of measurement may not be definite, although there is some evidence to support the choice of the voice-coil position as the measuring point. The best method of adjusting the two speakers is by using the microphone, as previously described.

Actually, a trained ear will suffice to select the correct location of the two speakers, although it may require more time and experimenting. It is necessary to play the same selection from a record through the speaker over and over, moving the high-frequency horn slightly each time. The best quality will be heard when the adjustment is correct, and after all, the ear is the best judge of the performance anyhow.

After the correct position of the high-frequency horn is determined to the user's satisfaction, the horn should be permanently mounted.

### Operational Readjustments

One of the bad features of many two-way speaker systems is that the new experimenter is apt to run the system with the high-frequency speaker operating at levels considerably above the correct balance, on the theory—probably—that “now I have a tweeter and you're going to hear the high frequencies, or else.” This is a natural fault, and is generally overcome after using the speaker for some time. The best test for any speaker is how well it “wears” although many authorities advise us that the ear can become accustomed to any speaker, and when it does, any other reproduction sounds wrong. However, if the user makes a point of attending a live concert occasionally in order to keep his perspective on a reasonable balance, he should be able to adjust the operation of a two-way speaker system so that it is capable of giving increased realism over any single-unit speaker available. The “presence” afforded by the reproduction of the frequencies above 900 cps on a small,

well-loaded diaphragm makes listening a pleasure, and the time and effort spent in constructing a speaker of this type will be well repaid.

The use of the attenuator in the high-frequency speaker circuit permits a small variation in the level of the frequencies above crossover, and this will be set at a point that appears to give the correct balance. The average difference between the two outputs is of the order of 4 to 8 db, and with the 3.5 db fixed pad to replace the 10-kc suppressor when it is out of the circuit, this indicates that from 0.5 to 4.5 db will be required in the variable pad. It may be desirable to remove the suppressor from the circuit when using the speaker for reproduction of phonograph records, but in general, the band suppressed is so narrow that it is difficult to detect, and little harm is done by leaving it in the circuit at all times. It does help on AM radio, however, when the output of the tuner contains any of the objectionable squeal which goes with the usual high-quality tuner.

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# TWO-WAY SPEAKERS

## *From Commercial Components*

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

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

### Atlas High-Frequency Speaker

One of the simplest methods of assembling a two-way speaker system is to employ an Atlas HF-1 speaker with an existing cone speaker in its normal baffle. This model of “tweeter” is made in the form of an attractive cabinet, shown in Fig. 1, which is intended to rest on top of a radio or a speaker enclosure. The dividing network is includ-

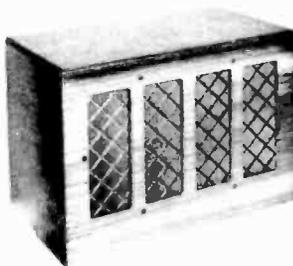


Fig. 1. Atlas HF-1 high-frequency speaker assembly, which includes its own dividing network in the cabinet which is designed to be placed on top of radio or speaker cabinet.

ed in the HF-1 cabinet, and all that is necessary is to remove the present leads from the loudspeaker, connect them to the dividing network chassis, and run another pair of leads from the chassis to the loudspeaker. That is all there is to it. Phasing adjustments may be made by moving the entire cabinet backward or forward, and the balance between lows and highs is made by a control mounted on the dividing network chassis.

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

This is an ideal unit when used with any good low-frequency cone as a “woofer.” Users should be cautioned about the correct balance between the two speakers, since the action of the control is not similar to that of a tone control. Figure 2 shows the output resulting from adjustment of the control. The entire band above the cross-over frequency is raised or lowered, putting a step in the response curve. The dotted line shows the action of a normal tone control. With the tone control of the amplifier or receiver in the “flat” position, the balance adjustment should be set for normal reproduction.

One advantage of the Atlas unit is that it is supplied with the dividing network already built in. While these networks are not difficult to construct if suitable measuring equipment is available, it is at least more convenient to have it already made. Some means for preventing the low frequen-

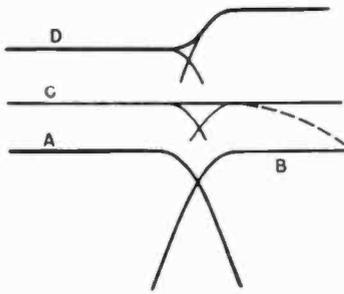


Fig. 2. Curves resulting from various components of speaker system. (A) is output of woofer; (B) is output of tweeter; (C) combined output of system when properly adjusted; (D) output when tweeter gets too much signal. Dotted line indicates action of normal tone control.

cies from reaching the diaphragm of the h-f unit is necessary, and while this may be done by the use of a series capacitor, the network also removes the high frequencies from the woofer, preventing cone breakup and avoiding the possibility of phase cancellation at the higher frequencies.

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

#### University Models

Another equally simple method is to add a Model 4404 University Dual Tweeter to an existing low-frequency speaker and baffle. This unit, somewhat smaller than the Atlas, also comes with its own cabinet, and with a high-pass filter and balancing potentiometer built in. Two small horns are used, each with its own driver, and they reinforce the woofer above about 2,000 cps. Since the filter consists solely of a series capacitor, low frequencies are kept off the h-f unit, and it is adequately protected.

Another University unit, Model 4407, is shown in Fig. 3. It consists of a 12" adapter ring which is used as a spacer between a 12" cone speaker and the baffle, setting the cone back far enough to accommodate a single small horn unit which mounts on a metal strap across the back surface of the adapter ring. To install this unit, it is only neces-



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

sary to remove the cone from its baffle, assemble the h-f unit and horn to the strap, and attach the cone to the adapter ring. The entire assembly is then remounted on the baffle, and a two-way system results. The same type of high-pass filter may be used with this arrangement, or a separate dividing network may be built.

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

#### The Racon Two-Cell Horn

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

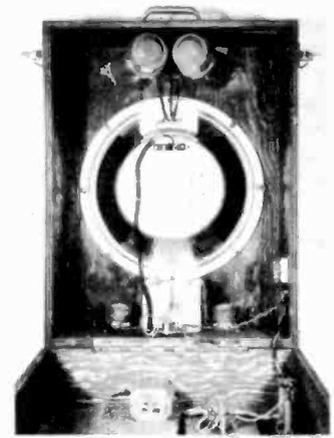


Fig. 4. Experimental assembly of 12" cone and University Model 4402, with dividing network and impedance-matching input transformer.

mounting, such as cutting an opening in the baffle or woofer cabinet.

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

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

#### Dividing Networks

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

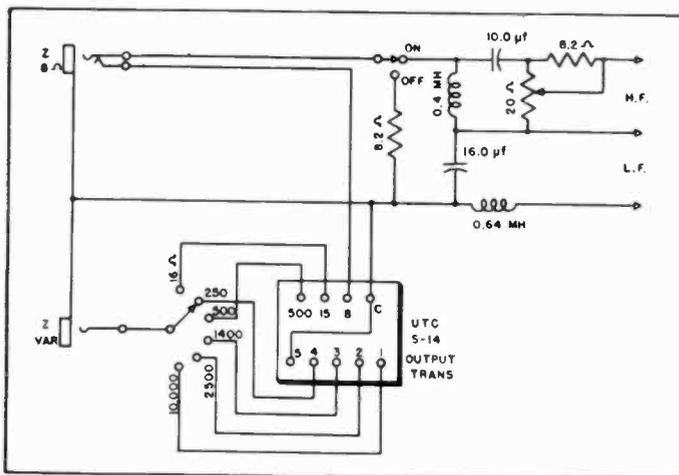


Fig. 5 (left). Schematic of speaker shown in Fig. 4.

TABLE I

DIVIDING NETWORK COMPONENTS

INPUT Z	TWEETERS		$f_c$ cps	L1 mh	L2 mh	C1 uf	C2 uf	R1 ohms
	TYPE	Z						
8	1-UNIV	12	2,000	.636	.398	9.95	15.9	10
	2-UNIV	6"	2,000	.636	.398	9.95	15.9	15
	RACON	15	1,200	1.06	.662	16.6	26.6	10
10	1-UNIV	12	2,000	.795	.497	7.95	12.7	15
	2-UNIV	24	2,000	.795	.497	7.95	12.7	10
	RACON	15	1,200	1.32	.825	13.2	21.1	15
12	1-UNIV	12	2,000	.955	.596	6.64	10.6	20
	2-UNIV	24	2,000	.955	.596	6.64	10.6	15
	RACON	15	1,200	1.59	.995	11.0	17.6	15
15-16	1-UNIV	12	2,000	1.25	.782	5.07	8.11	30
	2-UNIV	24	2,000	1.25	.782	5.07	8.11	20
	RACON	15	1,200	2.08	1.30	8.45	13.5	25

\* UNITS IN PARALLEL

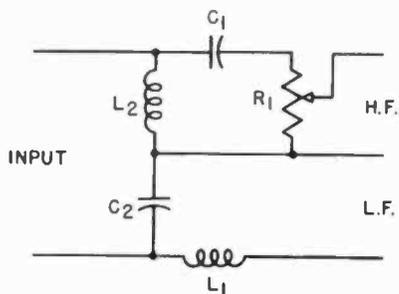


Fig. 6. Crossover network configuration for which values are given in Table I.

extra effort be expended to construct the network because of the improved performance.

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

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

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

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

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

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

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

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

As to making the inductances, close approximations can be obtained if a certain size of wooden spool is used and specific winding directions are followed for a given wire size. For these values of inductance, a suitable spool

size has a core 1-1/4" in diameter and offers a 3/4" winding space. With No. 16 enameled wire, laid thirteen turns per layer, the total number of turns required for the listed inductances are shown in the curve of Fig. 7. No iron should be used in constructing the spools, and they should be mounted with brass screws through the center. One pound of wire will make both of the coils for any combination shown except the last one in the table, which will require almost 1 1/2 pounds.

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

#### Performance Tests

In order to make comparisons of the performance of these systems, two types were assembled. Figure 8 shows a single baffle using the Racon unit with a 15-inch Jensen cone while the dual University tweeter and a 12-inch cone were shown in Fig. 4. For testing, the Racon-Jensen combination was mounted on a 6 cu. ft. enclosure and the response measured using a warble-tone frequency record through a flat amplifier as the source, the sound output measurement being made with an Electro-Voice Model 630 dynamic microphone working directly into a high impedance a-f voltmeter. This microphone has an excellent response up to around 11,000 cps and is quite adequate for measurements of fair accuracy.

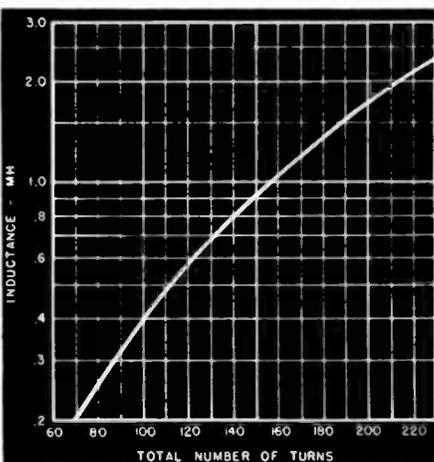


Fig. 7. Chart for determining number of turns to give inductance values necessary for various networks. This chart applies to coils wound on 1-1/4" core of wooden spool to a width of 3/4", using #16 enameled wire with 13 turns per layer.



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

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

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

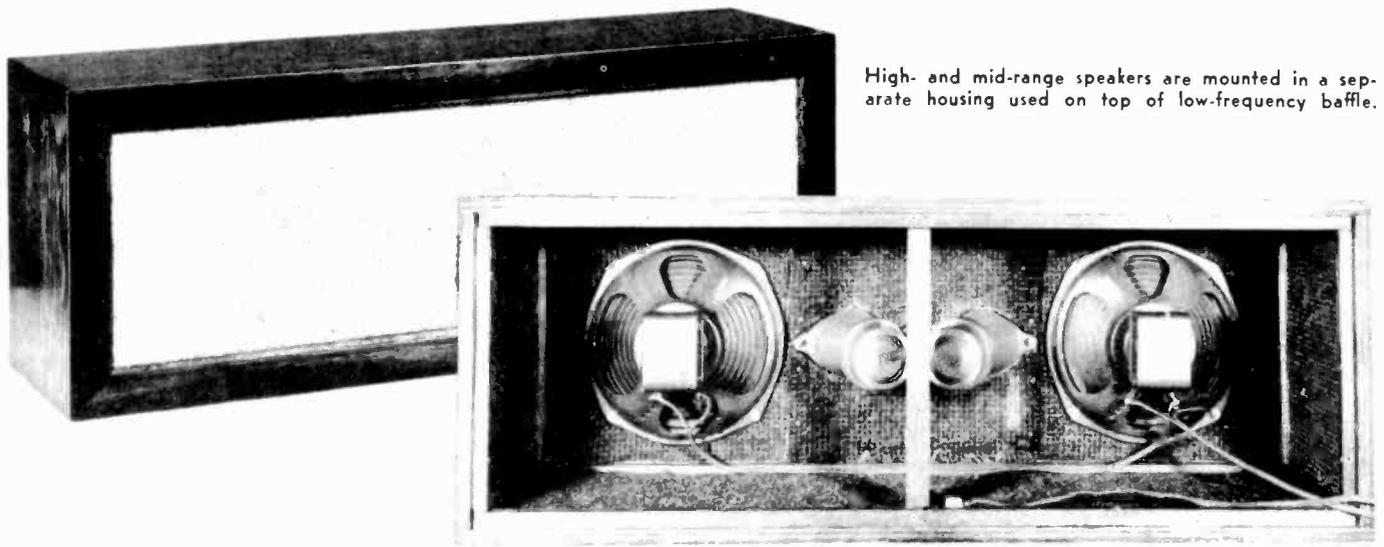
#### Additional Notes

Since this article was originally published, several changes have taken place in the components field, and there is now a greater variety of speakers available with which to assemble a system of this type. The University line has added a new horn designed for a crossover at 600 cps, along with an already-assembled network. A similar network is also available for a 2000-cps crossover, and both consist of a capacitor and an inductance, which is superior to a single capacitor in that it eliminates the high frequencies from the low-frequency cone in addition to eliminating the lows from the tweeter. A new Atlas horn is also available in a design which makes it simple to mount in a baffle. Several new models of l-f cones have been introduced, one of the most notable being the RCA MI-6269A, a 10-inch accordion cone of exceptional bass range.

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AUDIO ENGINEERING



High- and mid-range speakers are mounted in a separate housing used on top of low-frequency baffle.

# Three-Way Speaker System

GEORGE A. DOUGLAS

THE wide acceptance of some form of multiple speaker arrangement as a requisite of high-quality reproduction is responsible for a renewal of interest in three-way systems. The extraordinary strides made by dual speaker systems in the last few years has tended to a state of complacency and has resulted in obscuring somewhat the need for further experiment along these lines. The writer has been unable to find any published data on the three-way system, and has worked at the problem according to his own ideas—which may or may not be consistent with good engineering practice. However, the results attained with the system to be described should warrant serious consideration of the three-way system as a factor in tracking down the elusive will o' the wisp of realism in sound reproduction.

Theoretical discussion is not within the scope of this article other than to point up the desirability of limiting the coverage of single speakers to bandwidths within their capabilities, and utilizing as many as are deemed necessary to overcome as much as possible the shortcomings of each unit as regards mass, cone breakup, and other inherent deficiencies of design contributing to distortion. In a three-way system, these deficiencies are minimized in that each unit is called upon to deliver only within a range of about three octaves, a band well within the ability of any speaker of reasonably good quality. Besides permitting the use of a small unit in the upper band and thereby reducing the effects of mass, a separate middle fre-

quency speaker allows the choice of a very low cross-over without the excessive cost of a comparative dual system, and restricts the low frequency speaker to operation within optimum limits.

## Components

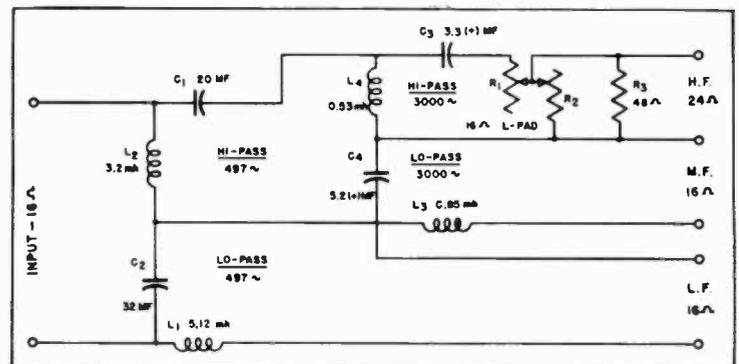
A three-way system was constructed along the same lines as the conventional two-way system, design data being merely extended to include an additional speaker. The existing reproducer, a 15" cone in a large cabinet, became the low frequency unit without modification. Eight inch cones serve in the mid section, two being used to provide sufficient power-handling capacity. In this connection, a horn type speaker was considered, and might have been used if a suitable one had been available. However, the cones were convenient and have proven satisfactory. The upper range is adequately taken care of by the dual horn unit shown in the photograph. It has excel-

lent horizontal and vertical distribution with a manufacturer's rating of 12 watts and response to 15,000 cycles.

Figure 1 shows the manner in which the filters are arranged to provide the cross-overs. Two series type dividing networks are cascaded to form the low-pass, band-pass, and high-pass transmission characteristic of Fig. 2. Economy and the limitations of the speakers were factors in the choice of cross-over points at 500 and 3000 cycles.

The cabinet shown in the photograph was constructed to house the upper and middle frequency speakers. No difficulty due to relative positioning was experienced. At 3000 cycles the distance involved is so small that the units could all be mounted flush and phasing accomplished by observing polarity. The small cabinet, on top of the large baffle, may be placed to advantage with reference to the low frequency speaker.

Fig. 1. Schematic showing circuit of the cross-over networks.



### Constructional Data

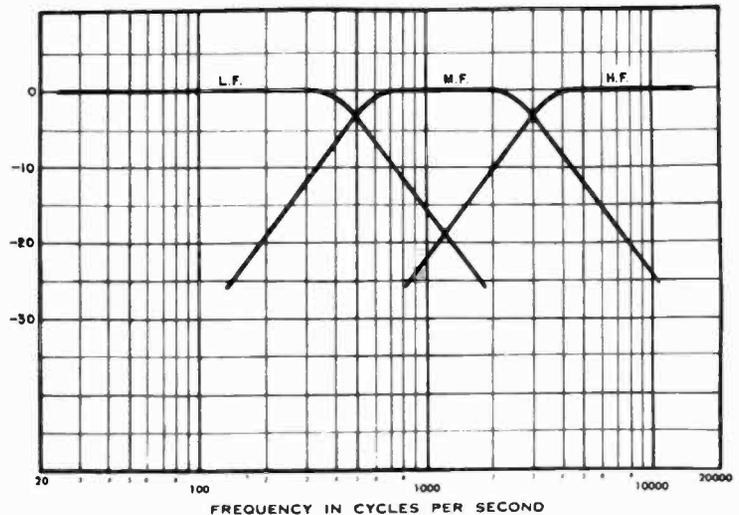
In the design of the dividing networks, the configuration and accompanying data described fully in the article commencing on page 101, were followed. Flanges for the coil forms were made by cutting discs of one-eighth inch masonite with a circle cutter, the edge of the tool being reversed so that the bevel does not appear on the disc. One and one-quarter inch dowel stock in lengths of three quarters and one and one quarter inches served as cores. The flanges are glued to the core, being held in place with a small wood screw until the glue is dry, after which the screw is taken out.

Using No. 17 DCE wire, coil specifications are approximately as follows:

Inductance MH	Winding Space	Turns
5.12	1.25	375
3.2	1.25	315
.85	.75	140
.53	.75	115

Capacitors were assumed to be reasonably accurate and were used to check the coils with the aid of an audio oscillator. It will be noted in the circuit diagram that each pair of inverse reactances are resonant at the cross-over frequency. Connected in series across the output of the oscillator, it was only necessary to alter the inductance until the point of greatest attenuation coincided with the cross-over frequency. Thus the pairs  $L1$  and  $C1$ , and  $L2$  and  $C2$ , are resonant at 500 cycles;  $L3$  and  $C3$ , and  $L4$  and  $C4$  resonate at 3000 cycles. Components were calculated for an  $R_o$  of 16 ohms, and in the case of the lower network, the cross-over point was fixed at 497 cycles to arrive at standard capacitor values.

Fig. 2. Transmission characteristic of the dividing networks shown in Fig. 1.



This slight shift can be ignored and is only mentioned as a point of information. An eight-ohm L pad is connected as shown in the h-f output, the resistor,  $R1$  changing the pad impedance to 16 ohms and at the same time reducing the maximum attenuation possible to 6 db.

The small cabinet is of plywood and was constructed with no particular attention to acoustical considerations other than to build it solidly and of ample proportions, and to partially line it with felt. Open mesh grill cloth covers the entire front panel, enhancing the appearance by concealing the speaker openings. On the rear panel is the h-f level control and two jacks for convenient access to speaker leads.

### Phasing

The pairs of speakers in the upper and mid sections were each connected in

series and phased separately before connection to the outputs of the dividing network. Then, with all speakers in the circuit and the entire system connected to the amplifier, 3000 cycle response was measured with a microphone and VU meter. Maintaining a constant level, the h-f leads were reversed and the two meter readings compared. Maximum indication occurred when the high- and mid-frequency speakers were correctly phased. As mentioned before, the relative position of the upper and middle speakers was found to be approximately correct when both are mounted on the front panel in the usual manner.

Using a 500-cycle signal, the same procedure was followed with the low and mid frequency speakers. In addition it was necessary to locate the small cabinet by sliding it backward and forward for maximum response.

# Getting the Most Out of A Reflex-Type Speaker

BENJAMIN B. DRISKO

Design data, constructional ideas, and simple tests for this type of loudspeaker.

**T**HE so-called "bass reflex" or acoustical phase inverter provides one solution to the loudspeaker mounting problem. To those whose available space or funds do not permit the more cumbersome exponential horn it may be made to provide considerably more uniform response than any simple open-

backed cabinet and, with a little care in the adjusting procedure, will give less hangover than almost any other type of mounting.

The minimum requirements for utilizing the following procedure are: 1. A suitable loudspeaker of the direct radiator type and an accurate knowledge of

its resonant frequency. 2. Some lumber, preferably plywood and a medium amount of carpentry skill. 3. Some acoustical absorbing material; ordinary carpet lining is very satisfactory. 4. A small step function signal generator; a number 6 dry cell is eminently satisfactory and a flashlight cell will do.

## Hangover

If one excites the voice coil of the unmounted speaker with constant current at variable frequency and measures the voltage at the voice coil terminals, one finds usually a pronounced rise at some point, generally between 40 and 100 cycles for ordinary speakers. Also it will be noted that the amplitude of the diaphragm motion is a maximum at this frequency. We are observing the primary resonance between the mass of the diaphragm or cone, the voice coil and other moving parts and the stiffness or restoring force provided by the spider, surround, etc. If the impedance curve is carefully plotted, the frequencies of the half-power points may be noted and the  $Q$  of the vibrating system computed from the quotient of the resonant frequency by the half-power  $b$  and width. It will usually be found that this  $Q$  is between 4 and 8, and in an infinite baffle will usually drop to about half its free air value. Since critical damping corresponds to a  $Q=0.5$  it is easy to see why this type of speaker is prone to hangover.

By analogy we may compare the above state of affairs to a series LC circuit which is said to be resonant. By this same analogy the reflex enclosure, which is really a Helmholtz resonator driven by the back of the cone, is an anti-resonant circuit similar to a parallel LC combination. This is a step in the right direction toward reducing the effective  $Q$  of the system, both by detuning it and by the added radiation resistance of the port of the bass reflex enclosure. If one now repeats the above impedance plot with the speaker in such an enclosure he finds two frequencies where the impedance has a peak on either side of the original resonance peak. If the resonator has been tuned to the same frequency as the diaphragm, the two peaks will be about equal in magnitude and both will have much lower  $Q$ s than the infinite baffle. If the tuning is slightly in error, one of the peaks will be higher than the other and will have a higher  $Q$ , which, for purposes of this paper is undesirable. With no sound-absorbing lining in the box, the damping will still be much less than critical and one is apt to find little sharp peaks in the impedance curve at frequencies for which some inside dimension is a multiple of a half wavelength.

The anti-resonant frequency of the Helmholtz resonator may be computed from the formula  $f=2070 (A^{\frac{1}{4}}/V^{\frac{1}{4}})$  where  $A$  is the area of the port in square inches and  $V$  is the volume of the box in cubic inches. The design is still indeterminate inasmuch as there are an infinite number of boxes of different volumes, each with a suitably sized hole and all

resonant at the correct frequency. One limitation may be placed by the following. The resonator frequency formula is only good where the maximum dimension is small in comparison with the wavelength. If we interpret small to mean between  $1/3$  and  $1/10$  we come out about right. Another rule, this time empirical, is that the area of the port should be between  $1/2$  and 1 times the area of the diaphragm. This results in a 4 to 1 latitude for size and it is doubtful if you could tell the difference between any two within those limits. It is considered good practice to keep the shape of the box somewhere near a cube. If the depth, width and height are in 2-3-4 proportion a fairly satisfactory product results, both acoustically and as a piece of furniture of convenient size.

A sample computation follows: Let's say that we are starting with a 12" speaker whose active diaphragm area is 82.6 in.<sup>2</sup> and whose resonant frequency is 70 cycles. We decide to make the port about  $3/4$  of the diaphragm or .60 in.<sup>2</sup> This gives a 6 x 10 opening which will look well below a 10 $1/4$ " speaker mounting hole. Rearranging the resonator formula

$$v = \left( \frac{2070}{f} \right)^2 \sqrt{A}$$

or, for our hypothetical case,

$$v = \left( \frac{2070}{70} \right)^2 \sqrt{60} = 6770 \text{ cubic inches}$$

If  $2x$ ,  $3x$  and  $4x$  are the depth, width and height respectively of our box, then the volume  $=24x^3=6770$  and  $X=6.56$  and our box comes out to have inside dimensions of .13 $1/8$ " x 19 $3/4$ " x 26 $1/4$ ". This is not an inconvenient size, so let's go on from there.

## Building the Baffle

The next step depends upon your skill as a carpenter. It is probably in order to make some sketches to see how much larger than the above inside dimensions you have to cut the various pieces to get the proper overlap at the corners and also to most economically utilize the available lumber. As to thickness, there is a fair amount of latitude.  $5/8$ " to  $3/4$ " plywood is considered good practice and yet one made out of  $1/2$ " plywood for a dancing teacher who wanted it very portable showed no serious misbehavior. It, of course, had to have corner blocks for structural reasons. While you're in the sketching department, lay out some furring strips, anything from  $1/2$ " x  $1/2$ " to 1" x 1" will do, and sufficient in quantity and sizes so that all six inside faces of the box can be covered with the carpet lining, hair felt, rockwool or whatever you choose, but attached by means of the furring strips so that there is a dead air space

behind each piece of felt, between the felt and the inside of the box and as thick as the furring strip. This takes a bit of doing and may be done in a variety of ways. Be guided by how much the thing will have to be handled and bumped around, inclining toward a minimum volume of furring strips and labor but being sure to get the inside of the box pretty well covered and to have the dead air space between the lining and the box. The purpose of all this is to further reduce the  $Q$  and hopefully to achieve critical damping.

It is also considered good practice to have the port as close to the speaker hole as possible. The reasons are a bit obscure, the principal one being to have as near a point source as possible. At this point it might be well to digress for a moment and discuss the final location of the whole outfit. The details are much too long to go into here, but it has been shown with the very best scientific rigor that a corner of the room is by far the best place, and by corner we mean really in the corner with no air spaces between the sides of the box and the walls and floor. This necessitates some elaborations in case there is a projecting base board. But it pays.

There is, however, another alternative. The whole box may be elevated to the upper corner of the room, between the two walls and the ceiling. Furthermore, since the speaker is fairly directional in the upper register it would be desirable to have it pointing in the general direction of the listeners. This calls for a five-sided platform with the cater-cornered side pointing down the long diagonal of the room. For reasons that are not too clear, by far the best-sounding reflex-type speaker which has come to my attention was of this construction and, furthermore, it was noticeably better up against the ceiling than down on the floor. The port was in top end of the box top when it was on the floor (the bottom when it was up against the ceiling) which brings us to the reason for the digression. If you can talk yourself into a corner location for the cabinet as a whole, then there is some precedent for locating the port anywhere you choose, but if the speaker must sit in the middle of a flat wall, then it may be wise to keep the port on the same side as the speaker and closely adjacent thereto.

## Tests

Let us now suppose that you have the reflex cabinet completed, the speaker mounted and leads coming out at some appropriate spot. Procure the d-c signal generator, namely one 1 $1/2$  volt dry cell and attach one voice coil lead to one terminal. Touch the other lead to the

other terminal and alternately make and break the circuit a few times. A moment's experimenting will indicate whether the lead and the battery terminal are clean enough to give a sharp, instantaneous make and break. This is important. If either make or break sounds scratchy, try scraping the contacts or better still, if you have a telegraph key handy, use it instead. A plain knife switch might do, but a switch that makes an audible click will not do unless you can have an assistant do the clicking for you at some remote point so you do not hear the switch click.

What you should hear on both make and break may best be described as a "bong." It is more than an instantaneous click, inasmuch as it has a recognizable pitch. If there is a piano near-by you could, in a few tries, find the note that matches it. The motion of the diaphragm and/or air is not yet critically damped. In pulse circles, this is called "ringing," a most satisfactory word. This ringing can be cured by application of the proper acoustic resistance to the port. So far as I know, the rules have not yet been worked out—from here on in you're on your own and its strictly cut-and-try.

It may well be argued that you should hear ringing anyhow unless you are in an anechoic chamber, because even if the speaker is perfect, the acoustic click produced by the speaker will excite the room in one or more modes and you will hear the room ring. To these scoffers I hasten to point out that the evidence that it can be done is that it has been done, twice to my knowledge, once with benefit of audio oscillator and v.t.v.m. for plotting impedance curves as above and once without. The oscillator helps, but is not a necessity. If you listen in the opposite end of the room, and if it's a very live room, then you do undoubtedly hear mostly room ringing, but conversely if your ear is a foot or two from the diaphragm and the room is reasonably dead, then you are to all intents and purposes listening to the speaker ringing. One way of evaluating

the deadness of the room in a rough way is to get some people into it and in conversation. Get as far away as possible still within the room however, and alternately plug and unplug one ear. The difference between binaural and monaural listening is much greater in a live room. In other words if you can still understand the conversation perfectly when listening with one ear the room is plenty dead enough for this experiment. It is, however, wise to keep fairly close to the speaker.

So now we're ready to go to work on the port. What we want is something with many small interstices that will cause acoustic losses. If the holes get smaller and smaller and the intervening material gets heavier we finally wind up with a solid rigid member. If you place a piece of board over the port you will still hear ringing, and probably indistinguishable in pitch from that heard with the port open, but somewhere between completely open and completely closed there is a material that has just the right resistance. It might be an old sweater, a turkish towel, a burlap bag or a few layers of window screening. Here's where the oscillator helps. If you have plotted two curves, one with the port open (double hump) and one with the port closed (single hump) then as you try each new material you can tell at a glance whether it is too heavy or too light and what to try next is somewhat simplified. However, it's still cut-and-try anyway you look at it.

The amount of audible ringing depends upon the total amount of damping in the system and a not inconsiderable portion may be supplied by the generator feeding the voice coil. When the battery circuit is closed the voice coil is practically short circuited, i.e., the electrical damping is very high. When the battery circuit is broken the voice coil is open-circuited and the electrical damping is zero. We may therefore in the course of our experimenting find a material such that the sum of the dampings produced by it and the battery together will be critical. In this

circumstance we will hear a "tick" when the circuit is made and a "bong" when it is broken. Make no mistake about the difference between the tick and the bong. You have to listen sharp but it is a very real difference, the tick being cleaner and sharper and without the slightest taint of any pitch whatsoever. So if we hear tick-bong-tick-bong as we close and open the switch we know that we are very close and on the low side and in one to two more moves we should hear tick-tick-tick-tick. In one case we started out with a loosely knitted, light, woolen sweater. One layer did practically nothing. Four layers got us to the tick-bong stage. The only thing we could find at the moment that was heavier than 4 layers of sweater was a piece of the ozite we had used for lining. It was too heavy, giving almost the same curve as a plain board. We then found that the ozite could be split and one half thickness of ozite turned out to be just the right combination. In another instance two layers of coarse burlap did a very good job. In both cases the final results were such that with an assistant to handle the switch it was impossible for the listener to distinguish the difference between the make and the break and both said tick. It helps to have the material stretched or so secured that it doesn't vibrate, otherwise it may contribute reactance as well as resistance.

Unsolicited and uncoached comments regarding the comparison between the reflex type cabinet with and without critical damping as per above might be summed up as follows: On organ music (steady tones), no difference, on percussion or lows with transients, pizzicato, etc., the damped speaker puts out somewhat less total volume of lows but the various low-frequency instruments stand out much more clearly. Several records with what formerly sounded like kettle drums turned out to be plucked strings. Male speaking voices reproduced 20db too loud lose most of the rain-barrel effect.

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# Vented Loudspeaker Enclosures

F. E. PLANER, Ph.D., and I. I. BOSWELL

**T**HE principle of the vented loudspeaker enclosure or reflex type cabinet, first described by Thuras<sup>1</sup>, is now well known, and enclosures of this

<sup>1</sup>A. L. Thuras, Sound Translating Device, U. S. Patent 1869178, July, 1932.

type are to-day widely used in high quality sound reproducing systems. Briefly, a vented enclosure sound system consists of a cone loudspeaker mounted in a felt-lined enclosure, which communicates with the atmosphere via an aperture or

duct in the front panel. The capacitive reactance of the air volume in the enclosure and the inductive reactance of the aperture or duct are arranged to resonate at the bass resonance frequency of the loudspeaker.

## Advantages

One of the main advantages of systems of this type is the improved efficiency at low frequencies, due to the re-radiation of the sound energy from the rear of the loudspeaker diaphragm via the aperture or duct after phase reversal. Other advantages are the improved transient response and reduced voice coil travel due to the additional loading of the diaphragm by the impedance of the acoustic system, as well as the relative independence of the performance from local acoustic conditions, such as the position of the enclosure in the room.

The design principles for vented enclosures, using apertures<sup>2</sup>, as well as ducts<sup>3</sup> as the inductive reactances, have already been treated in some detail in the literature. These notes will be concerned mainly with the discussion of the necessary volume of such enclosures.

The main drawback of the vented enclosure, as compared with alternative methods, such as the infinite baffle or the labyrinth type of enclosure, is the relatively large size required for the reproduction of the lowest audible frequencies. In the course of the design of a new domestic sound system intended to give exceptionally high quality reproduction, the question of cabinet size became of some importance, and an investigation of the relation between size and resonance frequency was made. As a result of this a number of expressions were developed which may be of general interest.

Fig. 1 shows schematically an enclosure comprising an effective air volume  $V_L$  and a duct having length  $l$  and cross-sectional area  $A$ . The two design principles for vented enclosures which have in the past been found to give satisfactory results, state that the resonance frequency of the enclosure itself should be similar to the bass resonance frequency of the loudspeaker, and the aperture or duct area  $A$  should be similar to that of the effective radiating surface of the speaker.

The inductive reactance in the case of an enclosure employing a duct may then, with close approximation, be written

$$x_L = \omega \rho / (\frac{1}{\sqrt{A}} + l/A) \quad \text{mech. ohms... (1)}$$

where  $\omega$  is the (angular) resonance frequency  $= 2\pi f$  and  $\rho$  the density of air.

The capacitive reactance of the air volume is given by

$$x_C = \rho_c^2 / \omega V_L \quad \text{mech. ohms... (2)}$$

where  $c$  is the velocity of propagation of sound in air.

<sup>2</sup>C. E. Hockstra, Vented Speaker Enclosure. *Electronics*, March, 1940, p. 34.

<sup>3</sup>F. W. Smith, Resonant Loudspeaker Enclosure Design, *Communications*, August, 1945, p. 35.

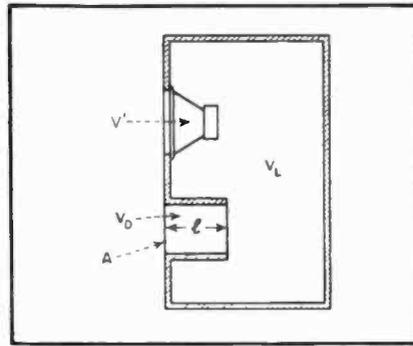


Fig. 1. Cross section through vented loudspeaker enclosure.

Resonance occurs when

$$x_L = x_C \quad \text{(3)}$$

Equating (1) and (2), and solving for  $\omega$  the resonance frequency is found to be

$$\omega = c / \sqrt{V_L (\frac{1}{\sqrt{A}} + l/A)} \quad \text{(4)}$$

### Effect of Duct Length

From this it is apparent that if the length  $l$  of the duct increases, the enclosure volume  $V_L$  may be decreased, other conditions remaining unaltered. At the same time, however, the total volume  $V_T$  which is made up of the effective air volume  $V_L$ , the volume displaced by the loudspeaker  $V'$  and that of the duct  $V_D$ , will not alter at the same rate, owing to the increase in the volume displaced by the duct.

$$\text{Since } V_T = V_L + V_D + V' \quad \text{(5)}$$

and  $V_D = Al$ , the total volume may be written

$$V_T = A (c^2 / \omega^2 (\sqrt{A} + l) + l) + V' \quad \text{(6)}$$

Now, in order to determine the length of duct corresponding to the minimum total volume, the differential of (6) with respect to  $l$  is equated to zero

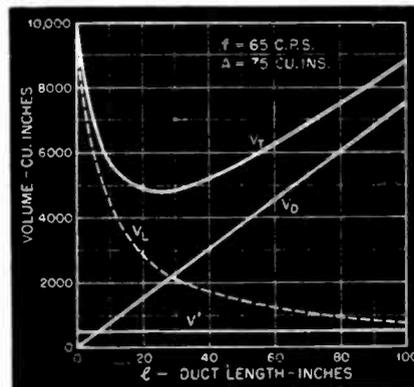


Fig. 2. Relation between duct volume, resonating air volume, total enclosure volume, and duct length for a typical vented loudspeaker enclosure.

$$dV_T / dl = 1 - c^2 / \omega^2 (\sqrt{A} + l)^2 = 0 \quad \text{(7)}$$

and hence

$$l_{\min} = c / \omega - \sqrt{A} \quad \text{(8)}$$

where  $l_{\min}$  is the duct length required to make the total volume  $V_T$  a minimum.

The volume for this condition is found by substituting (8) in (6)

$$V_T \min. = A (2c / \omega - \sqrt{A}) + V' \quad \text{(9)}$$

The corresponding air and duct volumes are given by

$$V_L \min. = A c / \omega, \quad \text{and} \quad \text{(10)}$$

$$V_D \min. = A (c / \omega - \sqrt{A}) \quad \text{(11)}$$

From these results it may be seen that it is generally possible by the correct choice of duct length, to effect an appreciable reduction in the overall size of the enclosure as compared to that required when an aperture only is used.

### Example

To illustrate the effect of the duct length on total volume, let us consider a typical sound system using a 12" loudspeaker with a bass resonance frequency of 65 c.p.s. and an effective radiating surface of  $A = 75$  sq. in. Then, substituting these values in the expression (8) the optimum duct length will be found to be  $24\frac{1}{2}$  inches, assuming the velocity of propagation of sound in air to be  $C = 13,500$  in/sec. If the volume displaced by the loudspeaker is taken as 500 cu. in., then the total volume from (9) becomes  $V_T \min. = 4,800$  cu. in. In comparison, the volume for an enclosure possessing an aperture only, i.e., for the case  $l = 0$ , is found from (6) to be 10,000 cu. ins

The relationship between  $l$  and  $V_T$  for the above example have been plotted in Fig. 2, together with the values of  $V_D$  and  $V_L$ . It will be seen from this that as the duct length is increased there is a rapid fall in total volume, with a minimum at  $24\frac{1}{2}$  in.; thereafter the total volume begins to rise at a more gradual rate. It will be noticed also that the slope of the curve for  $V_T$  is relatively small in the neighborhood of the minimum. The length of the duct may therefore, be made somewhat shorter than the optimum length indicated by expression (8) without an appreciable increase in the dimensions of the enclosure. In actual practice, it is an advantage to reduce the duct length in this manner, as it will then generally be possible to accommodate the duct without folding, thereby rendering the construction simpler and reducing the amount of wood required. Another point in favor of the shortened duct is the smaller volume taken up by the duct walls, a factor which has been neglected in the above calculations. Thus, in the present example the duct length may be

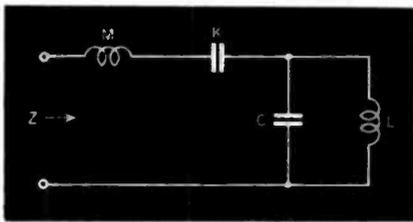


Fig. 3. Equivalent electrical circuit of mechanical system.

reduced to 13 inches with only a 10% increase in total volume above the theoretical minimum.

Apart from its effect on the over-all size of the enclosure, the extension of the aperture into a duct is desirable also for other reasons. Since the vent may be regarded as effectively constituting a second source of sound, it is advantageous to locate the latter as closely as possible to the loudspeaker from the point of view of the combined radiation impedance, as well as for the purpose of concentrating physically the source of sound. While these considerations apply to frequencies in the neighborhood of the resonance frequency of the enclosure, at higher frequencies the vent will tend to reduce the effectiveness of the baffle owing to the air leak created around the diaphragm of the loudspeaker. By the introduction of a duct it becomes possible to maintain the efficiency of the baffle at the higher frequencies due to the increased path length between front and rear of the diaphragm, while at the same time retaining the feature of close proximity between the two sources of sound at the lower frequencies.

In order to investigate the effect on the characteristics of the system when the ratio of duct length and air volume are varied, it is instructive to consider the equivalent electrical circuit of the mechanical system comprising the loudspeaker diaphragm and the acoustic resonator. Fig. 3 is a simplified equivalent network in which the dissipative elements due to radiation and frictional losses have been omitted.

The vibratory system of the diaphragm comprising the stiffness of the suspension and the mass of the moving parts including the effect of air loading is represented by the series tuned circuit having capacitance  $K$  and inductance  $M$ . The acoustic system of the vented enclosure is represented by a parallel tuned circuit having inductance  $L$  and capacitance  $C$ , dependent on the stiffness of the air volume in the enclosure, and the duct and radiation mass, respectively. The impedance  $Z$ , as measured at the terminals of the network is made up of the impedances of the diaphragm  $Z_L$  and that of the enclosure  $Z_E$  in series.

$$\text{Now, } Z_L = (1 - \omega^2 MK) / j\omega K \quad \text{---(12)}$$

$$\text{and } Z_E = j\omega L / (1 - \omega^2 LC) \quad \text{---(13)}$$

The total mechanical impedance, therefore, is

$$Z = (1 - \omega^2 MK) / j\omega K + j\omega L / (1 - \omega^2 LC) \quad \text{---(14)}$$

### Impedance Characteristics

In the absence of dissipative elements, the impedance characteristic of the two coupled circuits, as represented by the expression (14), will possess two points at which the impedance becomes zero, and the admittance infinite. The characteristic of the mechanical system is reflected in the electrical impedance characteristic of the voice coil of the loudspeaker, modified slightly by the electrical constants of the voice coil itself. This electrical impedance will be a maximum when the admittance of the mechanical system is infinite, and the electrical method of measurement, therefore, constitutes a convenient means of analyzing the behavior of the mechanical system.

In order to determine the two frequencies at which the electrical impedance will be a maximum, we substitute

$$MK = LC \quad \text{---(15)}$$

in (14), since the resonant frequency of the enclosure is made equal to that of the loudspeaker, and re-arranging terms (14) becomes

$$\omega^4 - \omega^2 (2K + C) / MK^2 + 1 / M^2 K^2 = 0 \quad \text{---(16)}$$

Solving for  $\omega$ , the two frequencies are found to be

$$\omega_{1,2} = \omega_0 \sqrt{1 + C (1 \pm \sqrt{4K/C + 1}) / 2K} \quad \text{---(17)}$$

where  $\omega$  denotes the resonance frequency of the two tuned systems individually.

The relation (17) is shown in Fig. 4, plotted in terms of the ratio  $M/L$  against frequency. From this it will be seen that as the duct length is increased and the enclosure volume reduced, i. e. with increasing  $M/L$  ratio, the separation between the maxima in the impedance characteristic increases. It will readily be seen that too great a separation is as undesirable as very closely spaced impedance peaks.

A number of loudspeaker enclosures were designed in accordance with the foregoing considerations, and a complete sound system employing these principles was constructed. The loudspeaker enclosure is the central unit and incorporates an 18" exponential cone loudspeaker having a bass resonance frequency of 40 c.p.s., a medium frequency driver unit with multicellular horn, and a special wide-angle distribution electrostatic high-frequency unit together with suitable dividing networks. Allowance has been made in the calculations for the additional volume taken up.

The associated apparatus is housed in the two separate side cabinets. These

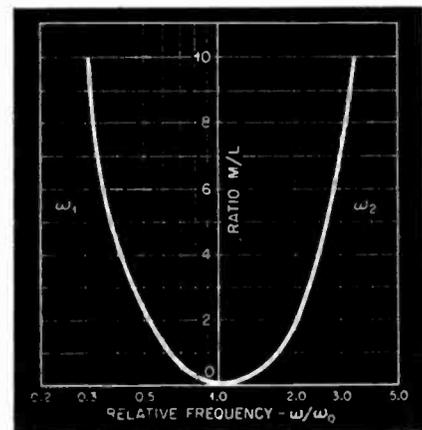


Fig. 4. Separation of the impedance maxima as a function of the ratio  $M/L$ .

apparatus units have been designed so that when they are used together with the loudspeaker enclosure as illustrated, the acoustic performance will be enhanced by the horn loading effect of the exponentially shaped surfaces of the side cabinets.

### Tests

A number of acoustic and electrical measurements have been carried out on enclosures of this type in order to study their characteristics. In the first instance it was necessary to verify that the resonance frequency of the acoustic system coincided with that of the loudspeaker. For this purpose, the voice coil current was measured at various frequencies with the loudspeaker removed from the enclosure and in free air. The baffle opening for the speaker was then blocked, and a corresponding volume introduced to replace that of the speaker. The acoustic system was subsequently excited by means of a separate driver unit coupled to the air volume in the enclosure by means of a tube 3 feet in length and  $3/8$ ths in. in diameter. This precaution was necessary in order to avoid interaction between the vibratory system of the driver unit and the acoustic resonator.

The tube was introduced via the duct, and the driver unit fed from a beat frequency oscillator. A microphone was placed immediately at the mouth of the duct, and together with its associated detector amplifier, served to indicate the resonance frequency of the enclosure. It was found in each case, that the actual resonance frequency of the enclosure was lower by varying amounts, than the calculated value, and an adjustment in the duct length was necessary in order to make the resonance frequencies of loudspeaker and enclosure coincide.

Next the loudspeaker was restored, and measurements were made of the sound output from the duct by means of the microphone and detector amplifier. Fig. 5 shows two typical characteristics (Concluded on bottom of next page)

# A New Corner Speaker Design

C. G. McPROUD

A new speaker cabinet designed to accommodate television in addition to providing high-quality sound reproduction.

*"Trifles make perfection; but perfection is no trifle."*

**W**ITH THIS TRITE SAYING as the guiding maxim in the construction or assembly of a high-quality reproducing system, the experimenter strives constantly to achieve perfection in the whole by working toward perfection in each separate component. In the past few months, the writer has endeavored to delineate the steps taken to arrive at the best possible reproduction from the electrical circuits of a residence radio system. Not that the equipment described is the only solution to the overall problem—far from it—but it is one solution designed to provide the maximum of convenience in operation together with a reproduction quality which leaves little to be desired.

Now that television is firmly established as a home entertainment medium, a complete installation must necessarily contain TV facilities, without sacrificing the superb quality desired for radio and phonograph reproduction. And, of course, no mention

was made in the previous series of the loudspeaker to be used with the residence system. Therefore, solving two problems at once, the TV installation has been combined with the loudspeaker in a form which results in high-quality reproduction, a reasonable compactness, and a piece of furniture which is an eye-appealing addition to a modern living room.

## Basic Design

It has been fairly well established that the most efficient location for a loudspeaker is in the corner of a room. The most outstanding example of this arrangement is represented by the Klipschorn, which consists of a two-way speaker system with both high- and low-frequency units being horn loaded. The cabinet work for the Klipschorn is extremely complicated, and certainly not one which the amateur woodworker should attempt. Some constructors have mounted a multiplicity of medium-quality cone speakers on the two sides of an obtuse enclosure,

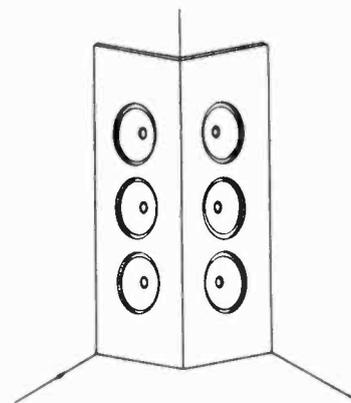


Fig. 1. Mounting a number of ordinary cones on two sides of an obtuse structure for corner use is one solution to the necessity for moving more air.

such as that shown in Fig. 1, and used this arrangement in a corner with excellent results. The corner location is optimum from the standpoint of loading on the speaker, since the radiation is over only half the angle of that from a speaker mounted on a flat wall. With a number of ordinary cones, the result is a means for moving rather a large volume of air without the necessity of having a large cone excursion of a single unit. Thus, better low-frequency response is obtained with speakers which individually would not perform so satisfactorily.

The writer has long used a standard two-way speaker of conventional design, and while the reproduction quality has been considered excellent, the low-frequency output did not compare with that of a good theatre system. Thinking from this point, the next step appeared to be in the direc-

## Loudspeaker Enclosures

*(Continued from previous page)*

obtained for the sound pressure at the mouth of the duct, with the latter tuned correctly, and with the acoustic system mistuned, respectively. (40 cycle enclosure).

Among other measurements the electrical impedance characteristic of the voice coil was determined over the useful frequency range under operating conditions. For this purpose the voice coil was connected in series with a decade resistance box, and the combination fed with signals at various frequencies from a beat frequency oscillator. By adjusting the decade resistance until the voltage

drops across it and the voice coil were equal, the value of the impedance at the particular frequency could be read from the setting of the resistance box. By these means it was in each case verified that instead of the original single peak, the impedance characteristic now possessed two damped resonance peaks, the frequencies of the two maxima being approximately 31 and 55 c.p.s. in the case of the 40-cycle enclosure referred to in Fig. 5. This separation is regarded as quite satisfactory and has been considered to be one of the reasons for the exceptionally smooth response

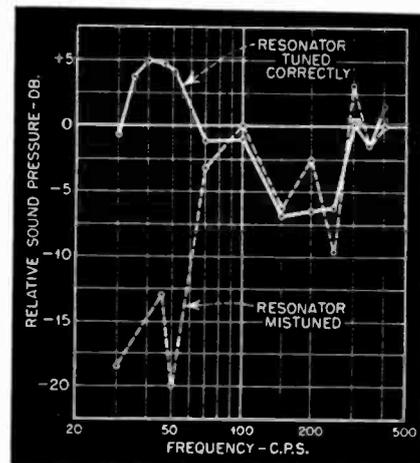


Fig. 5. Relative sound pressure at mouth of duct with the acoustic resonator tuned correctly, and mistuned, respectively.

tion of a corner speaker, yet utilizing the reflex action of a vented cabinet. Basically, therefore, the new design occupies the corner of a room, and is arranged so that the vents are loaded by a horn comprised of the walls and the sides of the cabinet enclosure. The plan view of the cabinet is shown in *Fig. 2*, with the vent openings *A-A'* along the sides. Thus the vents are loaded by the straight-sided horn between the wall and the cabinet.

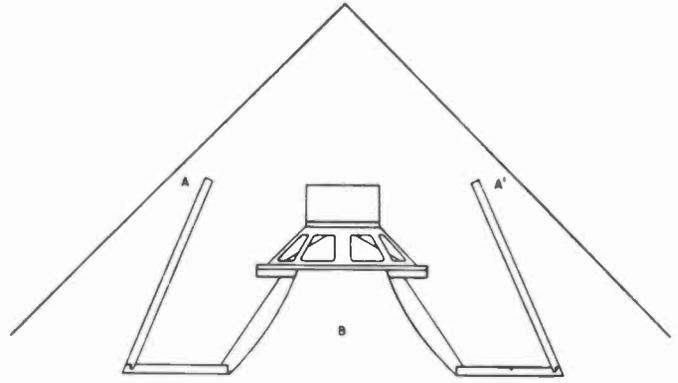
Experience has shown that loading of the vents should be accompanied by a similar loading on the direct radiating side of the low-frequency speaker, so the front of the cone is provided with another horn section, *B*, thus equalizing front and back loading and increasing the radiation efficiency. A top for the cabinet provides an air seal by means of gaskets between it and the wall, and the floor provides the other horn wall for the vented ports. The entire cabinet is open to the back, and utilizes the room corner, although if desirable for use in other locations, a false corner could be constructed to provide the necessary back.

After determining the basic design, any necessary variations can be made to accommodate TV, as has been done in this case. The picture tube is simply enclosed in a wood housing, and doors in the cabinet front cover the screen when it is not being used. The superstructure, shown in *Fig. 3*, houses the multicellular high-frequency horn and unit, and the space behind is large enough to accommodate the TV receiver chassis. With such a construction, the picture tube is between the two speaker sections, and the illusion of sound coming from the picture is considerably better than if the speaker is either above or below, or at the side of the screen.

Development of the practical aspects of the construction is controlled by the units selected for both high- and low-frequency speakers. In order to get the best possible low-frequency reproduction, manufacturers' catalogs were studied, and the cone selected on the basis of power handling ability and natural resonant frequency.

Good speaker performance depends on a number of factors. Among these is a high gap flux, which should be as great as possible. A high field strength ensures good damping as well as the maximum of efficiency. Another important factor is the relative weights of the cone itself and the voice-coil structure. It is considered good practice—for good low-frequency reproduction—to have these two weights as nearly equal as possible. It is also important to have as low a resonant frequency as can be obtained readily.

**Fig. 2. Plan view of the new corner speaker design, showing vents at either side between the cabinet and the wall, and the horn loading provided by the wall and the cabinet sides.**



The low-frequency cone selected for this system is a 15-inch model, rated at 20 watts, and with a resonant frequency of 42 cps. This model is the Stephens P52Lx—the *x* denoting a special model of the more-common P52L—designed for the woofer of a two-way system, and having straight cone sides and a lower resonant frequency than the standard model. Another important feature is the special treatment of the cone rim to prevent reflections from the frame. Good results may be expected from any of the high-quality 15-inch speakers available, such as the Altec-Lansing 803A or the 515, or the Jensen PLM-15A or P15-NLA models, but the lower resonant frequency of the Stephens model governed the final selection for this particular application.

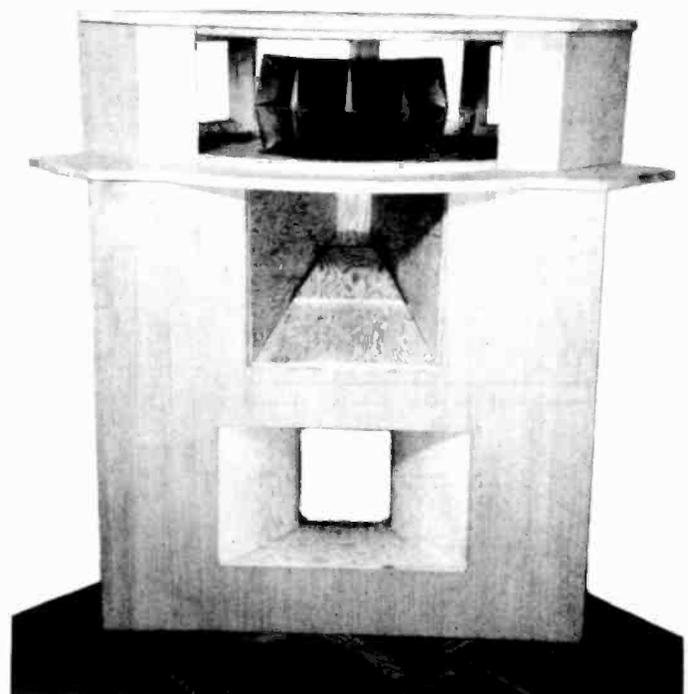
The selection of suitable high-frequency units is slightly wider. Ready-made are the Stephens P15 unit and the 824 horn, with an 800-cps crossover; the Racon RABAT unit with a two-

cell horn designed for a 1200-cps crossover; or the Jensen P8-151 horn with an XP-101 unit. The Atlas HF-1 is also usable, but if this choice is made, the low-frequency cone should have an impedance of 8 ohms to match the dividing network which is supplied. The unit installed in the complete speaker shown is an Altec-Lansing Model 901B, of early vintage, but still quite satisfactory. The horn is a 2x4 multicellular type, built by the writer,

#### Construction Details

Getting down to a specific design, therefore, the cabinet takes the shape shown in *Fig. 2* for a cross section at the plane of the low-frequency cone, and at (*A*) of *Fig. 4* at the plane of the center of the TV picture tube. The top of the low-frequency cabinet has the outline shown in the solid line at (*B*), with the superstructure shown by the dotted lines. The top is 39 inches from the floor, and the corners of the top meet the side wall 36½ inches

**Fig. 3. Front view of the new speaker during construction, showing the LF unit "well" and the TV tube enclosure.**



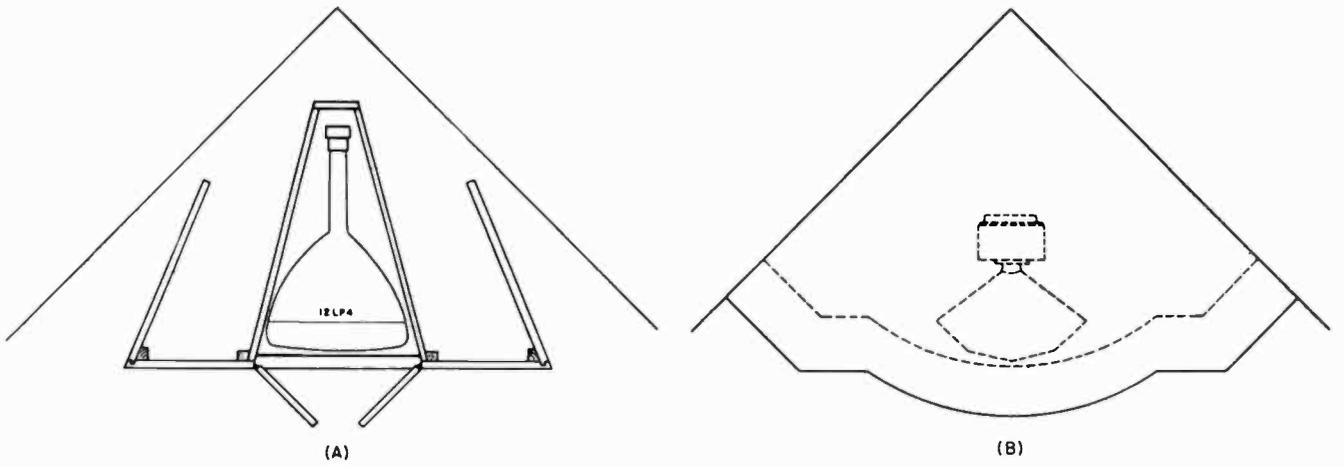


Fig. 4. (A) Cross section of the cabinet at the plane of the center of the picture tube to show location of tube enclosure. (B) Plan of the top (solid lines) and of the superstructure (dotted lines).

from the corner. Allowing for the volume of the speaker well and speaker and of the tube enclosure, the net volume of the cabinet is 8.5 cu. ft. This does not include the vent horns, which are usually included in the volume when vent pipes are used on the reflex ports.

Figure 5 shows the major parts used in the assembly of the low-frequency portion of the speaker, together with the housing for the picture tube. It will be noted that there is a hand hole in

the bottom of the tube enclosure, with a removable cover which serves two purposes: it mounts the deflection yoke, and thus permits adjustment of the TV receiver with the tube removed from the cabinet; and it also permits anchoring the cabinet to the corner of the room by means of a pair of steel cables and two turnbuckles. The cabinet is placed close to the corner, and with the turnbuckles open to their maximum, the cable is looped over a hook mounted on the floor right in the corner. Then

the turnbuckles are tightened up, thus locking the cabinet into the corner with the top tight against the wall. The quarter-inch semicircular groove along the back edges of the top provide space for a gasket to make an airtight seal. When the hand-hole cover is replaced, the structure is airtight except for the vents.

The wood selected for the top and front of the cabinet should be a suitable match (or contrast) for the furniture used in the room where the speaker

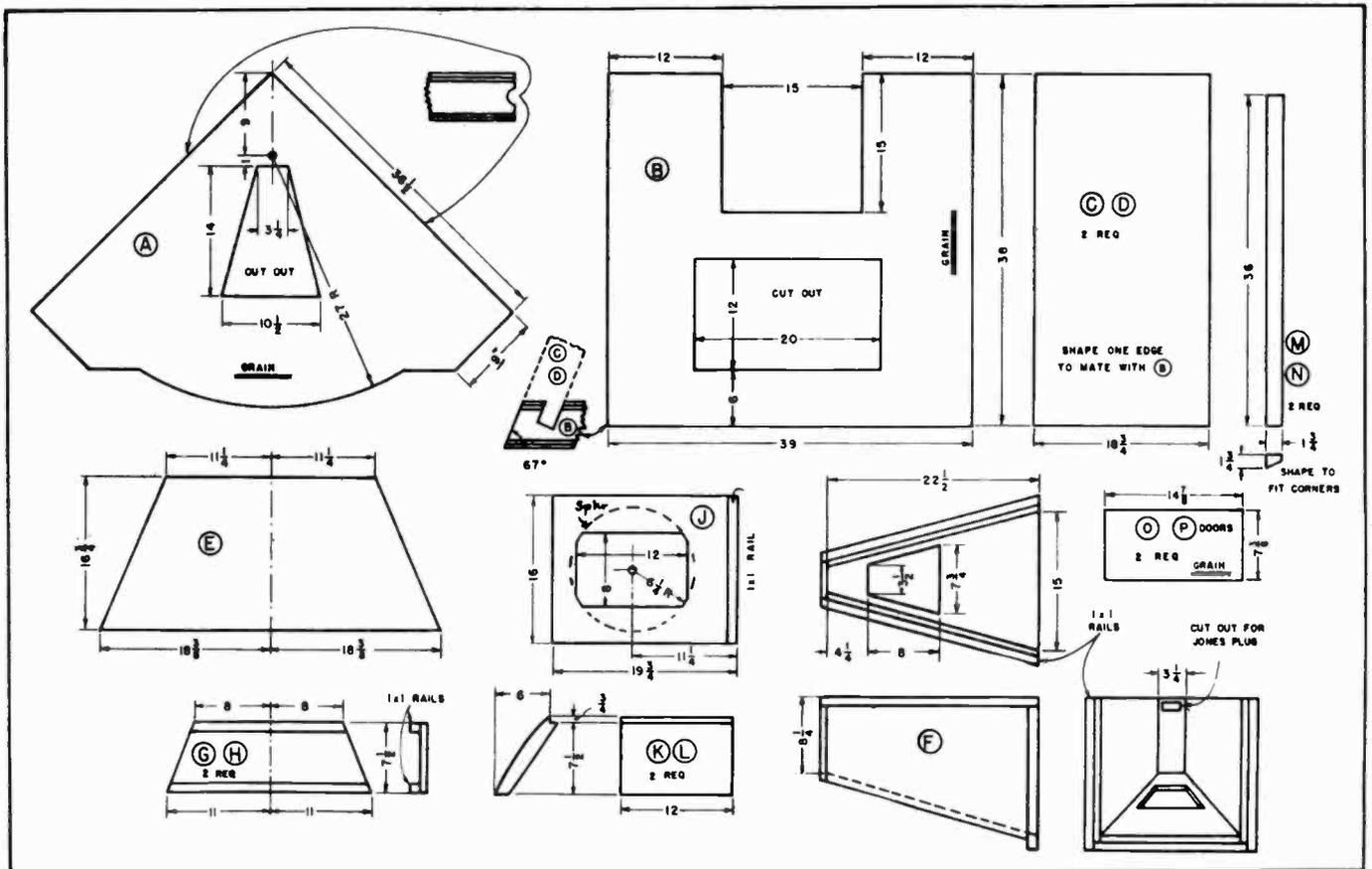


Fig. 5. Details of the pieces which comprise the lower cabinet, speaker well, and tube enclosure.

is located. For solid construction,  $\frac{3}{4}$ -in. material is recommended, with veneered hardwood being used for the top *A* and the front *B*. The doors *O* and *P* should be solid, or else veneered on both sides. The bottom *E*, sides *C* and *D*, speaker baffle *J*, and the tube enclosure *F* can be of less expensive fir plywood, also  $\frac{3}{4}$ -in. thick. The tube enclosure is a part of the acoustic chamber, which accounts for its seemingly over-solid construction.

The sides, *K* and *L*, of the speaker well are shaped from two-inch white pine, and should be fitted to the opening in the panel. The speaker baffle is drilled with eight holes, and T-nuts for mounting the speaker are installed on the front before the "horn" is assembled. In addition to the pieces shown, a number of  $\frac{3}{4} \times \frac{3}{4}$  strips will be needed for corner reinforcement. Parts *M* and *N* are for the acute corners at the front of the cabinet.

The details of the superstructure will be described in the following pages, and the parts are not shown in *Fig. 5*. However, it might be well to plan on another veneered piece nearly as large as the top *A*, since the grain should run parallel with the front of the cabinet, as shown by the shading lines. The two tops will cut readily from one panel of hardwood veneer.

Since this speaker is supposedly "functional," no attempt is made to disguise its appearance. The front of the low-frequency cone is visible in the speaker well, or horn, being protected by a screen of expanded metal. The inside of this horn is finished in dark blue lacquer, as are the sides of the cabinet and the edges of the two tops. The front and the top, together with the superstructure, are bleached oak, as is the tube mask. Lacquer covers the jointing of the speaker well to the panel, as well as the non-veneered edge of the top. If a uniform hardwood appearance is desired, the edges should be veneered—a job best done by the cabinet maker who cuts the pieces out. One caution is necessary—make sure that the top will fit the corner tightly. *Not all rooms have 90-deg. corners.*

#### Assembly

Once all the pieces have been cut out, the next step is that of assembly. Since some of the operations appear to be tricky, it is well to follow a certain procedure to avoid having to put the last few screws in with an offset screwdriver. The first step is to assemble the speaker well, which is a short exponential horn. Parts *G* and *H* are mounted on part *J*, using the shaped sides as spacers. Remember to put the T-nuts in place on the baffle before attaching the other parts. All joints should be glued, preferably with casein

glue, and secured with  $1\frac{1}{4}$ —12 flat-head wood screws, countersunk. This assembly should then be attached to the front panel, also with glue and wood screws. The shaped sides, *K* and *L*, are then fitted into place, also with glue and screws. Every joint in the cabinet is made with both glue and wood screws except that between the top and the lower section. This facilitates moving the entire unit. The top is attached only with screws, so it may be removed to enable the cabinet to pass through a 30-in. door.

After the speaker well is completed, the bottom is attached to the front, using a  $\frac{3}{4}$ -in. strip at the joint. The front extends clear to the floor, to eliminate the extra construction necessary for a recessed base. The bottom is thus inset, since the sides also extend to the floor. After the bottom is attached to the front, it is also secured to the speaker baffle. Next the corner braces are attached to the front, and the strips along the lower edges of the sides are screwed in place,  $\frac{3}{4}$ -in. up from the edge. The sides are then fitted into the groove in the front panel, and all joints screwed together. The tube enclosure is next mounted to the front, and supported at the back with a cross brace. The entire structure should now resemble that shown in *Fig. 6*, which also shows the  $\frac{1}{2}$ -in. square furring strips for the sound-deadening lining.

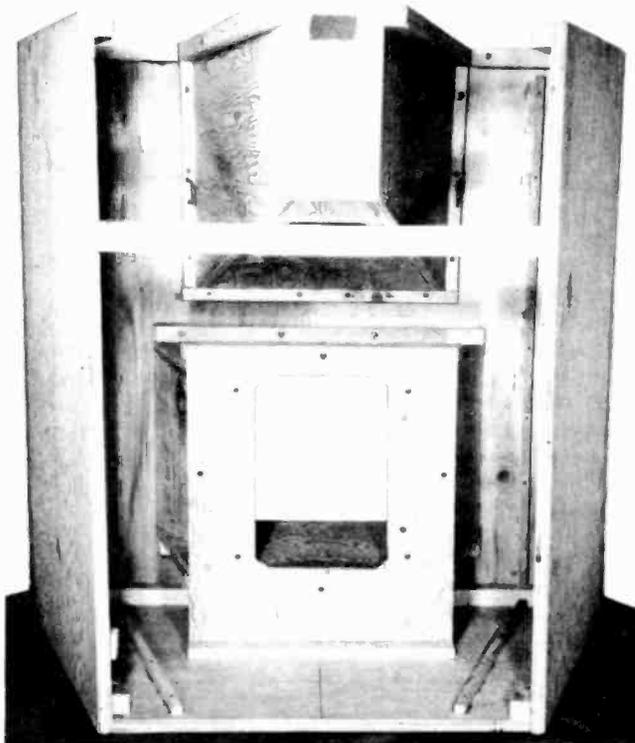
At this point, the doors should be fitted, using  $\frac{3}{8}$ -in. Soss invisible hinges which are mortised into the front and the doors. These hinges are the least

obtrusive of any hinge available, and while they are a little difficult to mount, the final appearance warrants the extra effort.

#### Electrical Connections

To avoid external wires, some provision must be made to introduce the signal and an a-c line to the unit, since it will not be readily accessible once the cabinet is mounted in place. The power circuit is necessary for the TV chassis, as well as for a possible outlet on top of the speaker. Since the speaker is designed to work from a radio-phono system housed elsewhere, the speaker signal must also be fed in. This is done at a small panel located just inside the lower right corner of the cabinet. One three-way male receptacle is used for speech, and a two-way male twistlock receptacle is used for the a-c line. The speech circuit goes to a switch which selects radio-phono in one position, or TV in another, and with an off position—the unused inputs being properly terminated. The output of the switch then goes to the dividing network, mounted on top of the speaker well, and thence to the two speaker units. Access to the high-frequency unit is had through an 8-terminal Jones receptacle, which also receives the input from the TV receiver and carries the a-c line up to the superstructure. This receptacle is mounted at the back of the tube enclosure, and permits removal of the top without disconnecting any wiring. The electrical circuits are shown in *Fig. 7*.

*Fig. 6.* View of the rear of the lower cabinet showing method of assembling the various sections.



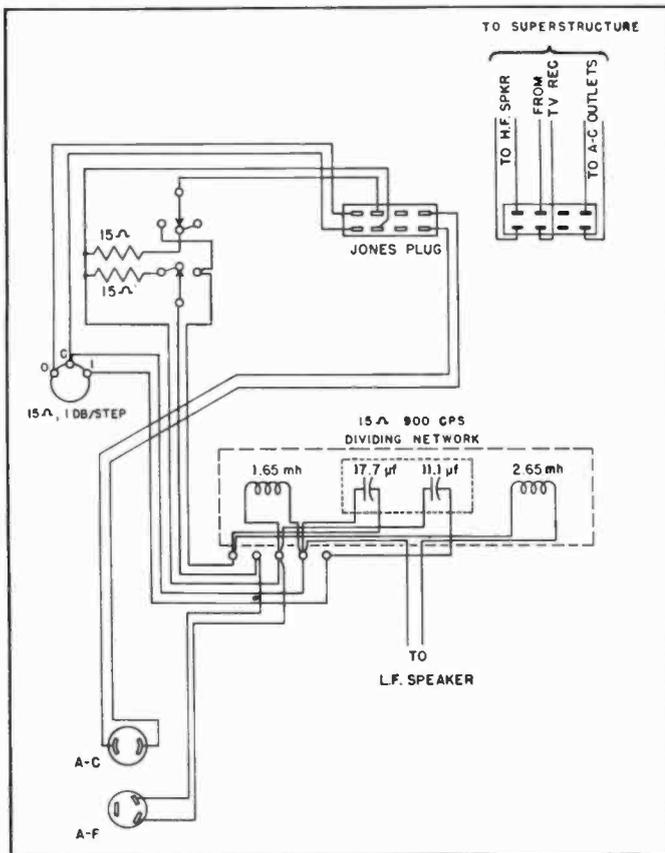


Fig. 7. Wiring diagram of the lower cabinet.

attached to the top, and when the speaker is to be moved, the top and superstructure are detached from the lower cabinet, since the complete assembly will not go through a standard door.

The actual measurements will depend upon the high-frequency horn and unit selected and its placement so as to leave room for the TV chassis. *Figure 9* shows a view of the interior with a standard RCA 630TS blank chassis in place—since the cabinet was designed to house this particular receiver which is available in kit form from Tech-Master Products Company at a considerable saving over the completed model. Since some modifications must be made, it seemed desirable to start from the blank chassis and build the entire receiver—the true experimenter's viewpoint.

In the figure, it will be noted that some of the controls on the rear apron are hard to reach, due to the lower strip of the frame. Therefore, the design shown in *Fig. 10* differs slightly from the photo. Except for the top panel and the two side panels, the material is pine, and is assembled with casein glue and flat-head wood screws. The side panels and the top are removable, the latter by a simple method.

Since it was not considered desirable to have screws showing on the top, the fastening consists of two 5/16-in. rods, threaded on both ends, which extend from the center deck

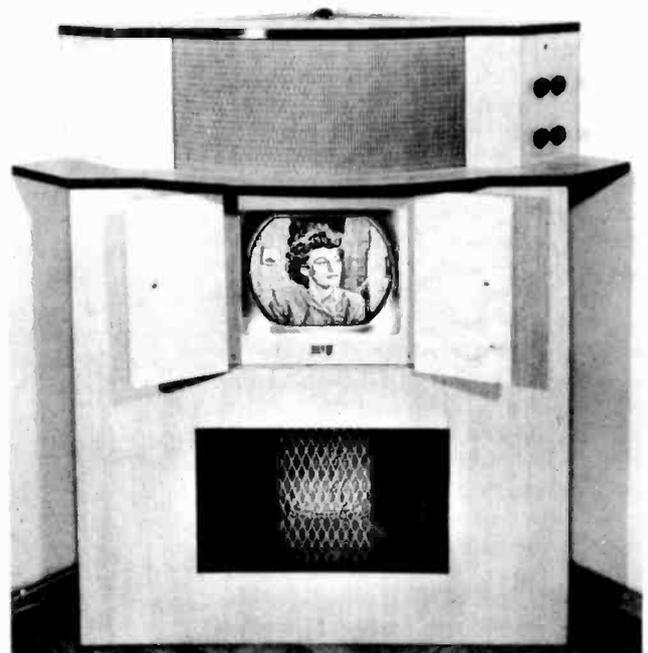
#### Preliminary Finishing

After the lower section is completely assembled, it should receive its first finishing operation. To protect the surface of the wood, the interior and the bottom should be given a primer coat of lacquer or some other undercoat. All cracks in the exterior should be filled with plastic wood, and the rear corners of the speaker well should be rounded out with fillets of the same material. After thorough sanding, the sides and the speaker well should receive a coat of an undercoat such as Firzite, which is an excellent filler for plywood. Finishing of the hardwood exterior should wait until the superstructure is completed in order that the two sections match as well as possible. Since most of the work on the lower section is now complete, the padding may be tacked in, using large-headed nails to prevent tearing out. Ordinary rug padding, such as Ozite, appears to be satisfactory for this purpose, although rock wool or Fibreglas is recommended by some constructors. The possibility of the fine glass shredding around a speaker cone argues against the use of either of the latter insulating materials, and the Ozite appears to do a satisfactory job of deadening without this risk. It is desirable to use two thicknesses over the larger areas, though the furring strips provide a good absorptive covering since there is an air space behind the padding.

#### The Superstructure

All of the cabinet above the top—lettered (A) in *Fig. 5* on page 94—is called the superstructure. It is primarily an ornament, since it serves no function except to enclose the high-frequency unit and horn and the TV chassis. This section is permanently

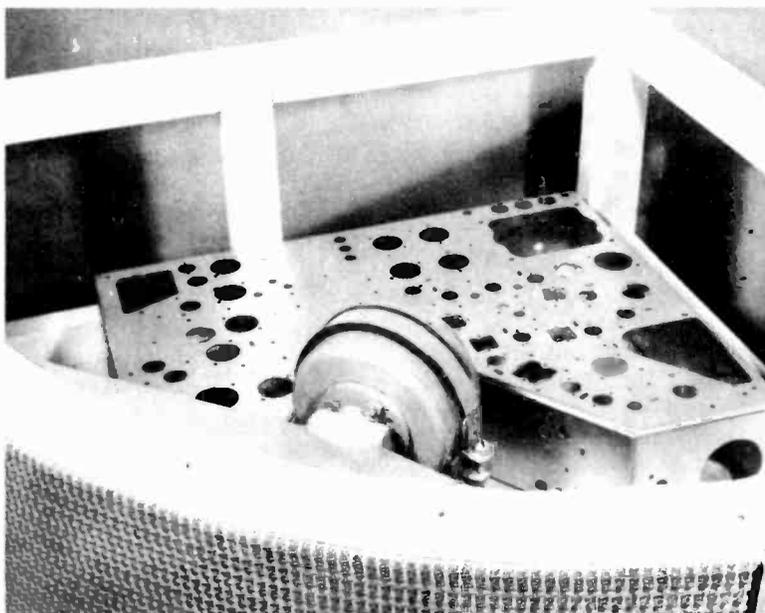
Fig. 8. The completed two-way speaker system in modern corner cabinet, with a 12-inch TV tube in optimum location for best illusion.



and engage two T-nuts which are set into the top panel and fastened with flat-head wood screws. Wing nuts are threaded onto the lower ends of the rods and peened on so they serve as handles, as shown in *Fig. 11*. The cotter keys keep the rods from dropping out of the hole in the upper framework, so there is no difficulty in engaging the T-nuts when the top is put in place.

The triangular cut-out is filled with a perforated metal grille to serve as ventilation for the TV receiver, which draws nearly 300 watts, and consequently needs free circulation of air. The two side panels are assembled of  $\frac{3}{8}$ -in. veneer, of the same wood as the tops and the front; they are simply screwed to the grille support. The TV controls are brought through these panels—the channel switch, fine tuning, picture and sound-volume controls at the right—by means of shaft extensions. The brightness and hold controls vary d-c voltages, and are extended from the chassis with long leads and placed in symmetrical positions on the left side panel.

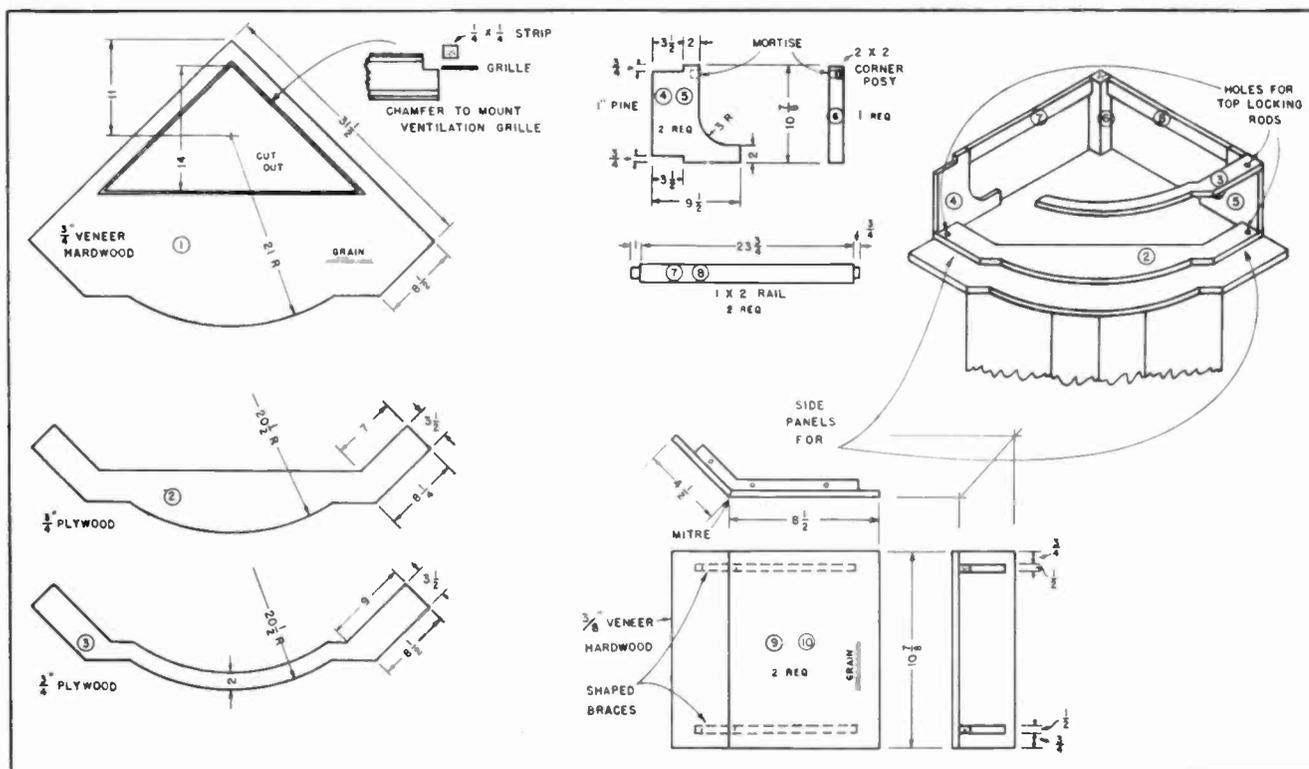
Since the two controls on the right use concentric dual knobs, the method of extending these shafts is shown in detail in *Fig. 12*. This requires long rods and conventional shaft couplings for the center shafts, and a brass tube with bushings for the outer control in order to clear the couplings, as shown.



**Fig. 9.** Interior of the superstructure, showing the positions occupied by the television chassis and the high-frequency horn and unit.

It may be desirable to include an a-c outlet on the top so as to accommodate a lamp or clock. It is equipped with a length of wire to plug into a dual outlet on the TV tube cover, which also mounts the high-frequency speaker unit, and a plug which receives the output of the TV receiver. All connections to the top section are carried through an 8-terminal Jones plug and receptacle, the latter being mounted on the back of the tube housing.

The screen in front of the high-frequency horn consists of another piece of perforated metal, covered with a piece of plastic grille cloth. Since the color obtainable did not appear entirely suitable with the finish of the woodwork, a piece of black buckram was placed between the metal and the plastic cloth to darken the over-all effect. This effectively hides the high-frequency horn, although both the perforated metal and the horn should be painted a



**Fig. 10.** Details of the various parts comprising the superstructure.

light color so as to avoid the appearance of a dark mass behind the cloth. Although it is claimed that the speaker is functional and not too much effort is expended to disguise its appearance, the grille in front of the high-frequency horn was added as a concession to over-all eye-appeal.

#### H-F Speaker Mounting

It is necessary to provide access to the tube compartment, so the high-frequency horn and unit are quickly demountable. Jumbo banana plugs were mounted on the front corners of the horn, and on the mounting block for the unit, and jacks were set into the framework for the front pair, and into the tube cover for the rear ones. The electrical connections for the high-frequency unit are carried on the latter two, and the entire h-f speaker may be lifted bodily from the jacks when necessary, without the need for watching phasing. The a-c outlets and a receptacle for the TV output are on the same tube cover, and all connections are carried through a short jumper cable to the male Jones plug.

Needless to say, the high-frequency speaker must be phased correctly before its final position is determined. This is best done by feeding a tone at crossover frequency to the speaker and reversing the high-frequency leads until the greatest output is obtained from the entire system, preferably measuring the output by a microphone and another amplifier with a volume indicator at its output. Then the high-frequency speaker is moved backward and forward until the maximum output is obtained. If the additional equipment is not available, put a tone at the crossover frequency on the system, and listen to first one speaker and then the other, moving the ear up and down in a plane parallel with and about 18 inches from the front of the cabinet. If the speakers are correctly phased, there will be a continuous tone heard from one speaker to the other. If not, there will be a null somewhere between them. At the crossover frequency (900 cps for the constants shown in Fig. 7) it should not be possible to detect any

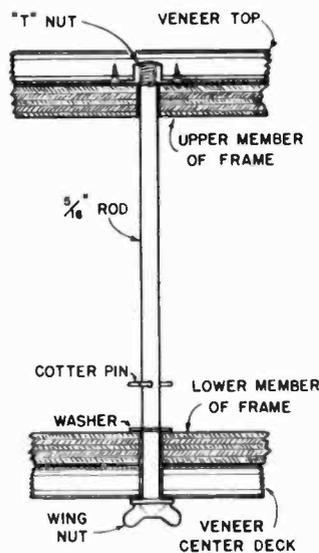


Fig. 11. Method of fastening the top to the superstructure without exposed screws to mar the appearance.

difference between the two speakers as the head is moved up and down. Try moving the high-frequency speaker backward and forward until there is no difference between the two sound outputs, then check by listening to male voices on the complete system. Optimum results can be obtained without instruments if necessary, and the lack of them need not deter anyone from assembling a satisfactory system. It is suggested, however, that after deciding upon the correct position and phasing for the high-frequency speaker, this position be marked carefully. Then, live with the system for a week or so before making the mounting permanent. Try shifting the unit back and forth while listening to a familiar record or program. Make sure that the optimum position is determined before the job is considered complete. Using the network shown in Fig. 7, the front of the high-frequency horn will just about coincide with the front of the cabinet when phasing is correct.

#### TV Receiver Placement

The entire superstructure was planned to house the Tech-Master kit receiver, since this model has an

excellent reputation for performance. Other models could undoubtedly be fitted into the space if desirable, but this chassis has a cutout for the tube which fits around the high-frequency unit with efficient utilization of the space. With the chassis selected, however, it is certain that performance with 12 or even 16-inch tubes will be satisfactory. If the kit is to be built for the purpose, the controls normally mounted on the rear apron might be relocated on the main chassis deck in the area normally occupied by the focus and deflection coils.

Connections to the picture-tube socket should be made through a 5-prong socket and plug, and those for the focus and deflection coils should be made with an octal socket and plug. It is necessary to carry a ground connection to the brackets which mount the two coils so as to have a ground for the outside coating of the tube, as well as for protection during adjustments. For the high-voltage connection, it is suggested that a banana jack be installed on the Bakelite insulating strip where the lead normally leaves the high-voltage compartment. This permits the chassis to be removed for servicing without unsoldering any connections.

The picture tube is a Sylvania 12LP4, selected because it is directly interchangeable—except for size—with the 10BP4 normally used with the 630TS receiver. The requirements for deflection angle are nearly identical, and the d-c and signal voltages necessary are the same. The Sylvania tube has an essentially flat screen, and features an ion-trap type of electron gun. For best results, these two characteristics are important, and they governed the choice of the tube. The screen size with this tube is  $8\frac{1}{2} \times 11$  in.

The focus coil and the deflection yoke are mounted on the hand-hole cover in the tube compartment, us-  
cover in the tube compartment. The tube is centered in the compartment, with padding all around for protection. The leads are brought up through notches along the side of the tube compartment cover. The positioning of the tube is shown in Fig. 13. A plastic mask is mounted on a  $\frac{3}{8}$ -in. oak front panel, providing both protection for the face of the tube and masking for the picture area.

#### Woodwork Finishing

In the cabinet shown, the front and tops are oak veneered, treated in the blond "rift" finish. This is a

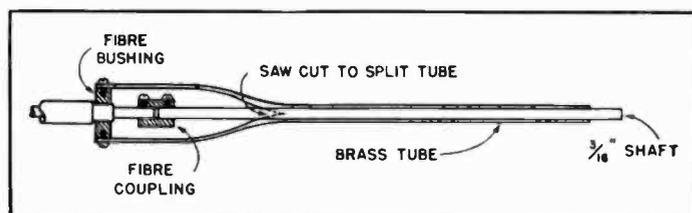


Fig. 12. Manner in which the television receiver controls are extended to the control panel by extension shafts.

simple operation, consisting solely of painting the well-sanded oak surface with one coat of Firzite, allowing it to dry for about five minutes, and then wiping it off. The white remains in the grain of the wood, and gives an interesting finish. After the paint dries for about 24 hours, the surface should be waxed thoroughly. The speaker well, the sides, and the edges of the tops are lacquered with two coats of ensign blue Larcoloid, which gives a glossy finish from a brush coat.

This particular treatment may not appeal to everyone, but it is modern and attractive, and requires a minimum of work. Since the entire design is essentially modern, it is felt that the surface treatment should also be modern.

After the cabinet is completed, it should be permanently installed as previously outlined. Two rings are mounted at the sides of the tube compartment. To these are attached 8-in. lengths of chain, which are connected in turn to 4-in. turnbuckles. Another length of chain is attached to the other end of each turnbuckle, and looped as closely as possible over a large screw hook inserted in the floor right in the corner of the room. The turnbuckles are then tightened up, sealing the top against the wall. Originally it was planned to use a steel cable, but the difficulty of attaching the hooks and turnbuckles to the cable ruled it out after a few trials. The chain is much simpler. It may be desirable to deaden the chain with a cloth sleeve over it, or by lacing venetian blind cord through the links. The gasket used for the top is a 5/16-in. braided clothesline, tacked on the ends and glued into the groove for its entire length. The sides of the cabinet are spaced from the wall by ordinary door stops—adjusted to the required 1½ in. by selecting the point at which they are attached to the cabinet.

#### Performance

Subjectively, this speaker system seems to be the answer, in the opinion of the writer and of several others who have heard it. It gives the feeling of a wide source of sound, as would be expected since the separation between low- and high-frequency speakers is approximately 33 inches from center to center, and the very low frequencies come from the side vents. The overall width of the speaker from wall to wall on the plane of the front is 65 inches. This wide-source effect is pleasant in the extreme and until it is experienced,

Fig. 13. Mounting of picture tube in compartment, showing relative position of various components.



the listener may doubt its advantages. A similar effect may be obtained for a trial by connecting two or three speakers to the output of an amplifier, and placed well apart.

Figure 14 shows the measured sound output with an Electro-Voice Model 630 microphone and a sensitive a-f voltmeter, the microphone being at a distance of six feet from the front of the cabinet. The curve is corrected for the measuring equipment as completely as possible. The tones above 200 cps were obtained from a warble-frequency record, while those below were obtained from an audio oscillator.

The signal input was held constant at the input to the 6AS7G main amplifier, which is flat from 20 to 13,000 cps, so the output represents the actual operating characteristics of the speaker with its normal driving amplifier.

The efficiency is evaluated by comparing the setting of the high-frequency attenuator with that used when the identical components were assembled with a 7-½ cu. ft. reflexed cabinet of conventional design. With the old cabinet, balance was obtained with 6 db in the h-f attenuator, while only 4 db is required with the corner speaker. This indicates that the low-frequency speaker is 2 db more efficient in this cabinet than it was with the standard box.

This is the second speaker built using the same principle of design, and the results in both cases certainly seem to justify the expectations. Performance compares favorably with medium-size theatre systems, and it has "presence"—the intangible characteristic of realism which gives the desired feeling that the performer is actually in the room.

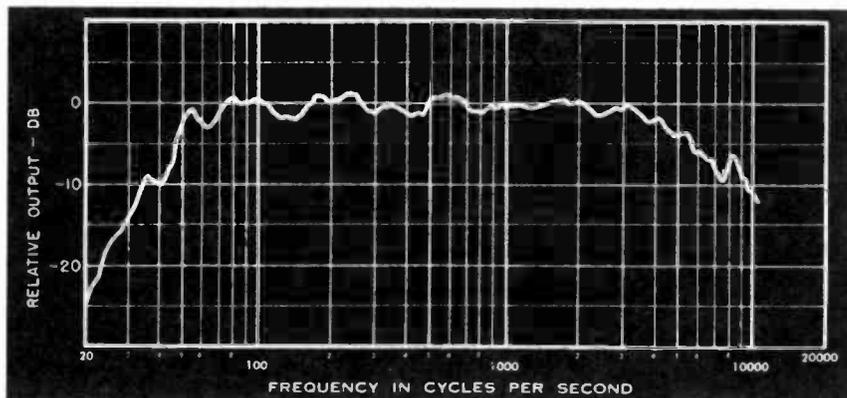


Fig. 14. Measured response curves, showing performance

# Corner Speaker Cabinet for 12-inch Cones

**T**HE DESIGN for a corner cabinet for 15-inch cones, first shown in the January issue, is followed here by a similar design adapted for the use of a 12-inch cone, either with or without a high-frequency unit and horn. This design has been worked out on the basis of a 7-cu. ft. enclosure, and without the TV picture tube compartment, it is considerably smaller in size than its predecessor.

The dimensions shown in the drawing below are planned for the use of 1/2-inch 5-ply panels, although only slight changes would be necessary if 3/8-in. plywood were to be used instead. The mounting strips, 3/4-in. square, are not shown for most of the construction, but for solid work it is desirable that all joints be reinforced by these strips. All points should be glued, and the assembly should be made by means of wood screws.

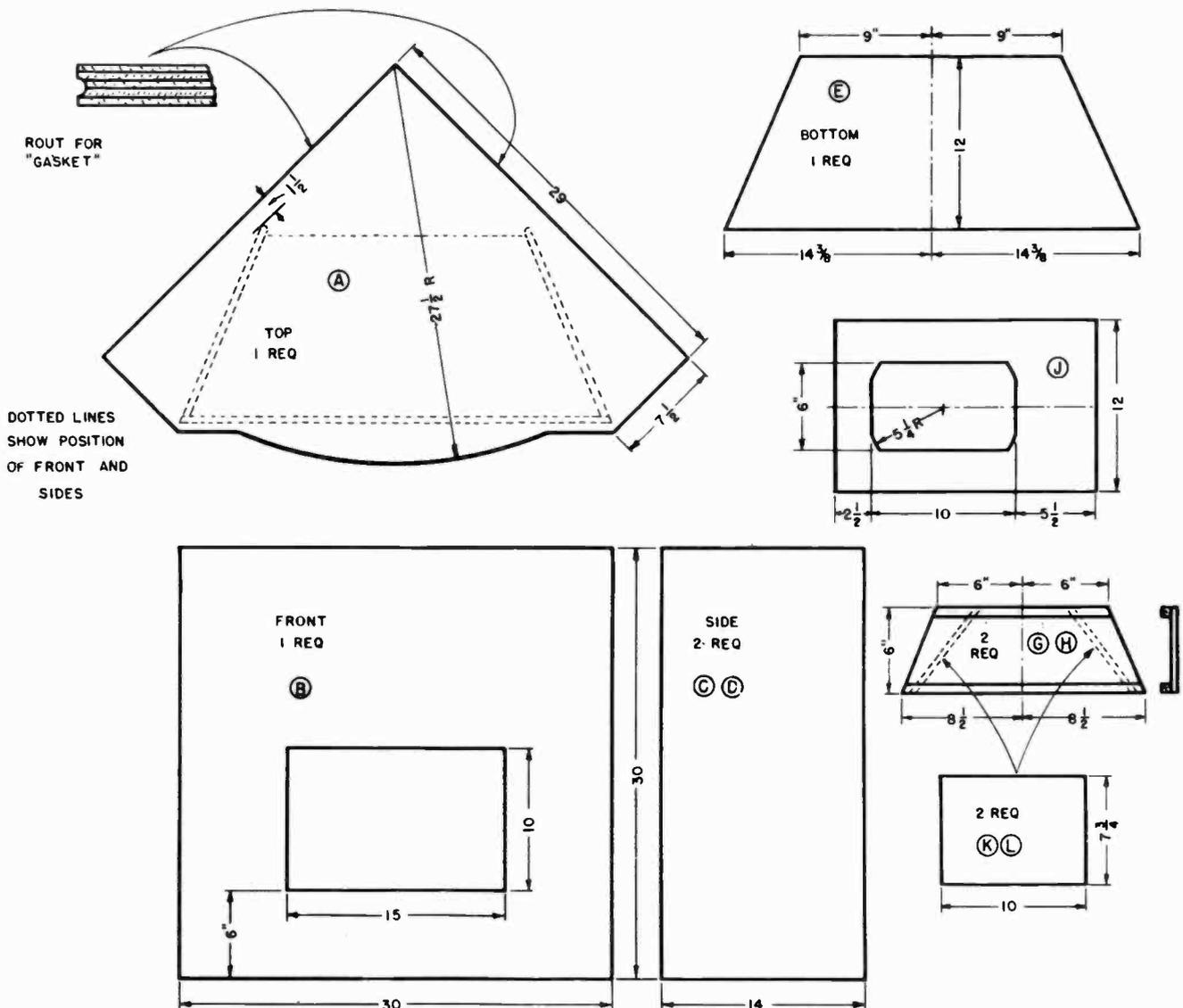
Because of the fact that there are only two parallel surfaces in the enclosure, it is not actually necessary that the entire inside of the cabinet be padded. The two parallel surfaces are the top and the bottom, and these should be padded with Ozite or some similar material. The two sides should be braced across the back after the cone is installed. The top must be gasketed against the wall to ensure an air-tight joint. The sides of the speaker-well are straight, and 1-in. material is recommended.

If a two-way system is desired, an h-f horn and unit may be installed above the cone opening. Either the Ricon two-cell horn and unit, the University dual tweeter, or the Atlas HF-1 horn will mount flat against the front panel, and any of these units should be used with a suitable dividing network.

To give a finished appearance to the cabinet, a grill cloth can be placed over the speaker-well before it is attached to the front panel, and some similar treatment should be given to the high-frequency horn, if one is used.

This cabinet will ensure good loading of the cone down to about 35 cps, and the relatively small floor space required for a corner unit is a definite advantage.

Details for assembling this model of the corner speaker are similar to those for the 15-inch model described on the preceding pages. The individual parts of the cabinet are lettered to match those of the larger model, and by referring to the photos on pages 93 and 95 the builder should encounter no difficulty. It should be noted that part J is shown in a horizontal position. It is to be mounted with its longest dimension in the vertical direction.



# Design and Construction of Practical Dividing Networks

C. G. McPROUD

Instructions for making loudspeaker dividing networks without using laboratory equipment.

**A**LTHOUGH THE SUBJECT of dividing networks has been covered with considerable thoroughness in the literature over the past few years, much of this material has made it necessary for the builder to make a number of choices as to the circuit used. Few writers on the subject have made definite recommendations, most of them providing details for the calculation of all types of network configurations. In addition, the builder who does not have a suitable bridge available for measurement of the required inductances is handicapped because of the difficulty in obtaining ready-wound coils for this application. It is the purpose of this article to describe one particular type of dividing network, the choice having been made on the basis of a number of listening tests. Furthermore, constructional details are furnished which make it possible to wind acceptable coils for these networks without the need for measuring the inductance values on a bridge. Provided the coils are wound exactly as described, the resulting inductance should be within 5 per cent of the required value; this may be of some aid to those who do not have access to the necessary measuring equipment.

## Network Design

Dividing networks are designed by two entirely different methods. The most common is based on the filter principle, and the characteristics of the sections resemble those of any other type of filter. The other method is based upon obtaining a constant-resistance characteristic of the sections, and while the configurations are similar, the method of calculating the values for the components differs considerably. Although the idea of maintaining a constant-resistance load on the output of an amplifier, with a consequent stabilization of the source impedance out of which the speakers work, is one which appears to be in the realm of good design, it cannot be denied that no loudspeaker offers a truly constant impedance to its source. For this reason, the filter type cannot be eliminated from consideration, since the impedances resulting from the use of this network do not vary more than 20 per

cent throughout the range for which they are designed. More important, however, is the fact that critical listening tests have usually resulted in the selection of the filter network. It is, therefore, the one selected for this article. The principal advantage to be gained from the use of the constant-resistance network is one of simplification of manufacture, since both inductance elements are of the same value, and both capacitors are of the same value. For the constructor who has occasion to make but one or two, this is not an important consideration.

The next step in selection of a network type is based on the required attenuation beyond the crossover frequency. Single L-section networks provide an attenuation of approximately 12 db per octave outside the transmitted band, and if greater attenuation is desired, a full T- or pi-section is used to give a loss of about 18 db per octave. The use of properly designed high-frequency horns will aid in the attenuation of the low frequencies provided the horns are constructed so that there is a natural falling-off of output as the frequency is lowered. Thus, with a crossover at 1,000 cps, for example, the horn itself should be designed to cut off around 750 cps, so that additional attenuation is not necessary. This same consideration applies to some extent with the low-frequency speaker, although unless special speakers are used, they may supply considerable energy well above crossover. Again resorting to the results of listening tests, it is usually sufficient to em-

ploy single L-section networks, with their 12-db/octave attenuation. One other consideration in this selection is the loss due to the use of additional reactive elements. If there were no resistive component in the coils and capacitors used in networks, any number of sections could be used without any additional loss. However, with practical coils and capacitors there is always some loss, and even when relatively large wire is used in the inductances, a loss of 0.5 to 1 db must be expected through any network.

## Arrangement

The final choice in the network selection is that of the arrangement of the sections. Either parallel or series connection may be employed, with differing results. The characteristic of the parallel network provides that the impedance outside the transmitted band rise to infinity, thus offering little or no damping to the speaker unit at these frequencies, even though they are not in the transmitted band for that particular speaker. The objection to this circuit may be questioned, but again basing the selection upon the all-important listening test, the series network is chosen for this discussion. The impedance of each section of the filter network falls to zero outside the transmitted band.

The final form for the network under consideration is shown in Fig. 1, together with the formulas for the calculation of the various components. The constructor will have selected the desired crossover frequency based upon the efficiency of either or both speakers throughout the audio band. Crossover frequencies below about 800 cps require the use of larger and longer horns, if the multi-cellular type of horn is used. If the low-frequency speaker is of the folded-horn type, the transmission may be limited to as low as 500 cps, requiring the lower crossover. In general, it is usually considered advisable to make the crossover frequency as low as possible in order to get as much of the mid-range frequencies out of the high-frequency unit, particularly if a metal-diaphragm unit is being used with a multi-cellular horn. Very satisfactory results have been obtained with crossovers ranging from 800 to 1,000 cps, while if the space permits,

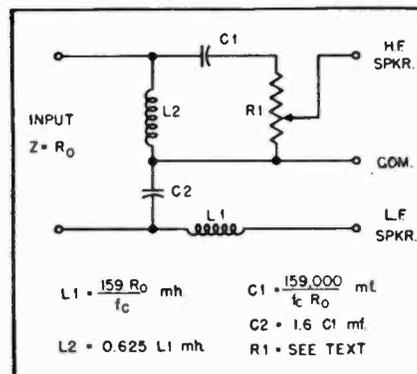


Fig. 1. Schematic of dividing network discussed in this article.

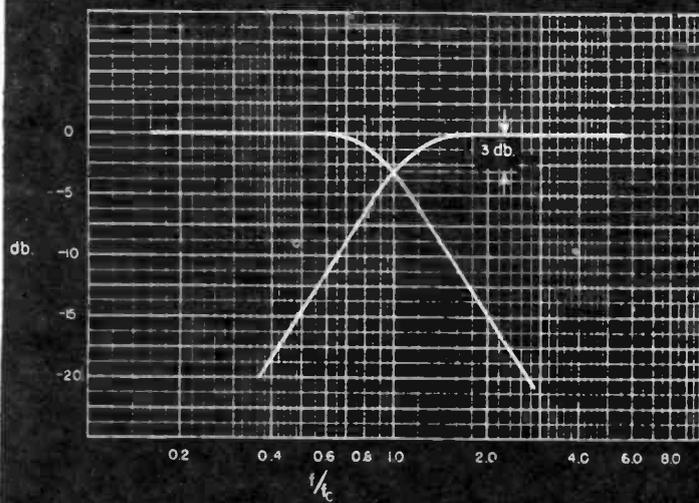


Fig. 2 (left). Frequency-response characteristics of dividing network of Fig. 1.

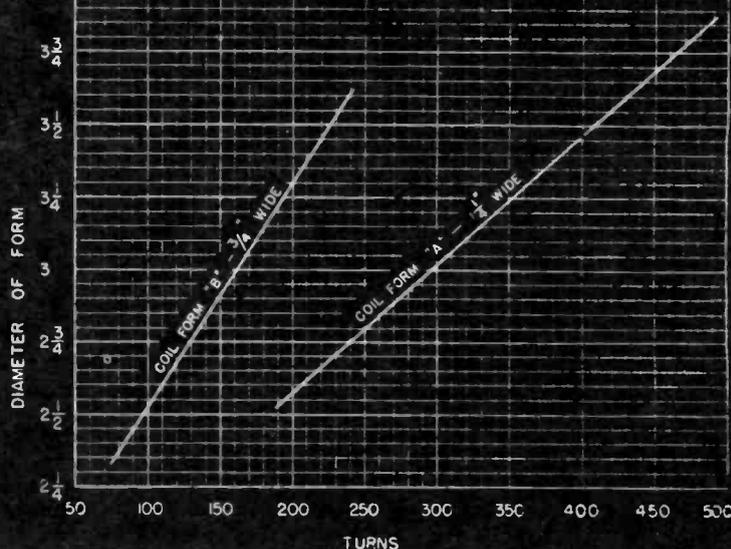


Fig. 4 (right). Flange diameter required for coil forms for coils having given number of turns of No. 17 DCE wire.

still better results can usually be obtained from a crossover at 500 cps. Some high-frequency units are designed for crossovers of 1,200, 2,000, and 3,000 cps, and while the output in the upper ranges is improved by the use of a two-way system with the higher crossover, greater intelligibility and naturalness is obtained with the middle speech-frequency range (1,000 to 2,000 cps) supplied by the metal-diaphragm unit. These factors must all be considered in making any choice.

### Configuration

The network configuration arrived at through these choices is shown in Fig. 1, which is an L-section series network. It is designed to feed two speakers of the same impedance, at any crossover frequency desired, with the components calculated against these parameters. If the high-frequency unit is of a different impedance from that of the low-frequency speaker, the additional efficiency of this unit may be used to permit the matching to the impedance of the low-frequency speaker by the proper choice of value for  $R_1$ , as will be described later.

Since odd values may often be ob-

tained from the formulas for  $C_1$  and  $C_2$ , a simple artifice may make it possible to use standard, commonly available values for the capacitances. Suppose, for example, that the formulas yield values of 14.2 and 22.7  $\mu\text{f}$  for the two capacitors for a crossover frequency of 800 cps. Capacitors having even values can be substituted for the calculated units, maintaining the same ratio between the two values, so that 15.0 and 24.0 are usable, resulting in a change of the crossover frequency from 800 cps to 845 cps. This figure is then used for calculations for the inductances, and little change in the performance of the system should be observed. The advantage of such a change lies in the use of capacitor values more easily obtained, without the necessity of building up the required total capacitance with a number of small units.

It is considered good practice to be able to radiate power from a loudspeaker to at least half an octave beyond the crossover point in order to provide satisfactory operation in the crossover region. Thus, if the crossover is to take place at 1,000 cps, output from the high-frequency unit is necessary down to 1.5

(0.5 x 1,000) or 750 cps. Any shift in the predetermined crossover frequency should take this into account.

Figure 2 shows the transmission curves for both sections of the network of Fig. 1. It will be noted that at crossover each circuit has a loss of 3 db, making the total output power at crossover equal to the input power. This may necessitate some compensation for the increased efficiency of the high-frequency unit, and  $R_1$  is provided for this purpose. For nominal differences in efficiency of the two speakers, of the order of 6 db, the value of  $R_1$  may be chosen at 1.5 times the impedance of the network or speakers, assuming that both speakers are of the same impedance. This will allow a reasonable margin for adjustment of the relative level fed to the high-frequency unit, without greatly disturbing the load impedance on the h-f branch of the network. For more critical applications, the resistor  $R_1$  can be replaced by an L-pad, or with fixed resistors calculated from the desired attenuation and the impedance of the speaker.

When the high-frequency unit has an impedance differing materially from that of the low-frequency speaker, as is often the case when 8-ohm speakers are used for the low-frequency unit and 16-ohm high-frequency units are employed on a horn, the difference in efficiency permits the adjustment of impedances by the use of a simple shunt resistor at  $R_1$  without the use of a tap. Thus, if the efficiency differs by 6 db, the 16-ohm unit should be shunted by 16 ohms, and fed from an 8-ohm network, when used with the 8-ohm low-frequency speaker.

### Coil Construction

The principal objective of this article is to simplify coil construction to the point where reasonably good results may

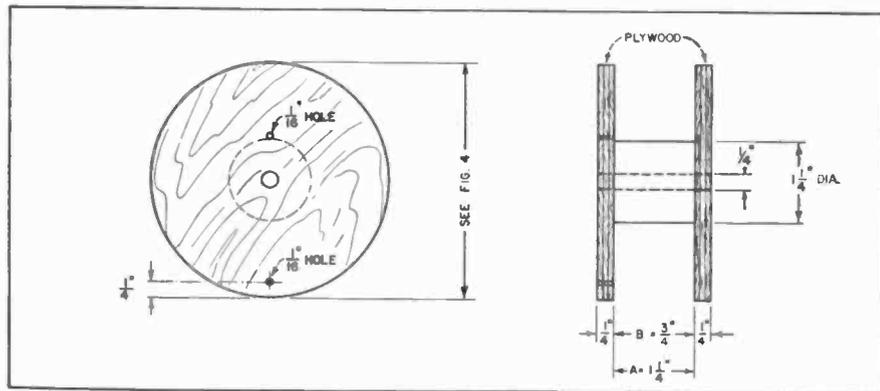


Fig. 3. Coil forms used in winding inductances in accordance with curves of Figs. 4 and 5.

be obtained without resorting to a bridge for the measurement of inductance values. With this in mind, therefore, the coil forms shown in Fig. 3 are used as the standards for these data, two sizes being provided to take care of a wide range of inductances while maintaining a reasonable form factor for the coils. Therefore, it is recommended that coil form "A", with a winding space of  $1\frac{1}{4}$ " be used for coils ranging from 2.0 to 8.0 mh, and that coil form "B", with a winding space of  $\frac{3}{4}$ " be used for coils of 0.5 to 2.0 mh. Assuming that the constructor has determined the inductance values required for a network and selected the form most suitable for winding the coils, he may determine the maximum outside diameter for the flanges from Fig. 4. These values, together with the information provided in Figs. 5 and 6 are predicated upon the use of No. 17 DCE wire.<sup>1</sup>

Returning to Fig. 4, if the flanges for the coil forms are cut to the dimensions shown, they will allow suitable winding space with about  $\frac{1}{4}$ " overlap, thus giving coils of uniform appearance. The curves of Figs. 5 and 6 may be read directly to determine the number of turns of wire to be wound on each coil form for the required inductance value. Note that twenty-three turns are wound per layer on the large form, and that thirteen turns are wound on each layer of the small form. If wire of other size must be used, comparable values for the number of turns required will be obtained if the number of turns per layer is held to either of these values. This will, however, require a recalculation for the width and diameter of the forms. The curve of Fig. 5 covers inductance values from 1.0 to 8.0 mh, while the curve for Fig. 6 covers values from 0.2 to 2.0 mh. When carefully layer-wound by hand or on a lathe, Q values of the order of 20 to 25 should be obtained, which are adequate for dividing network coils.

It goes without saying that no iron should be used in the construction of these coil forms but that all parts of the form are of wood or plastic. It is recommended that they be made by gluing the flanges to the core. A  $\frac{1}{4}$ " hole through the center may be used for mounting the forms by means of large brass wood screws. Some constructors prefer to wind coils on a demountable form, and after winding to remove the coil from the form and tape it up. This is somewhat of a refinement, and when only one set of coils is required, it is

<sup>1</sup>This may appear to be an unusual size, but many experimenters may have the field coil from a discarded Western Electric 555 unit, which is wound with wire of this size. However, if either #16 or #18 wire is used, and wound with the indicated number of turns per layer, the inductance values obtained by following the charts will be sufficiently close for practical purposes.

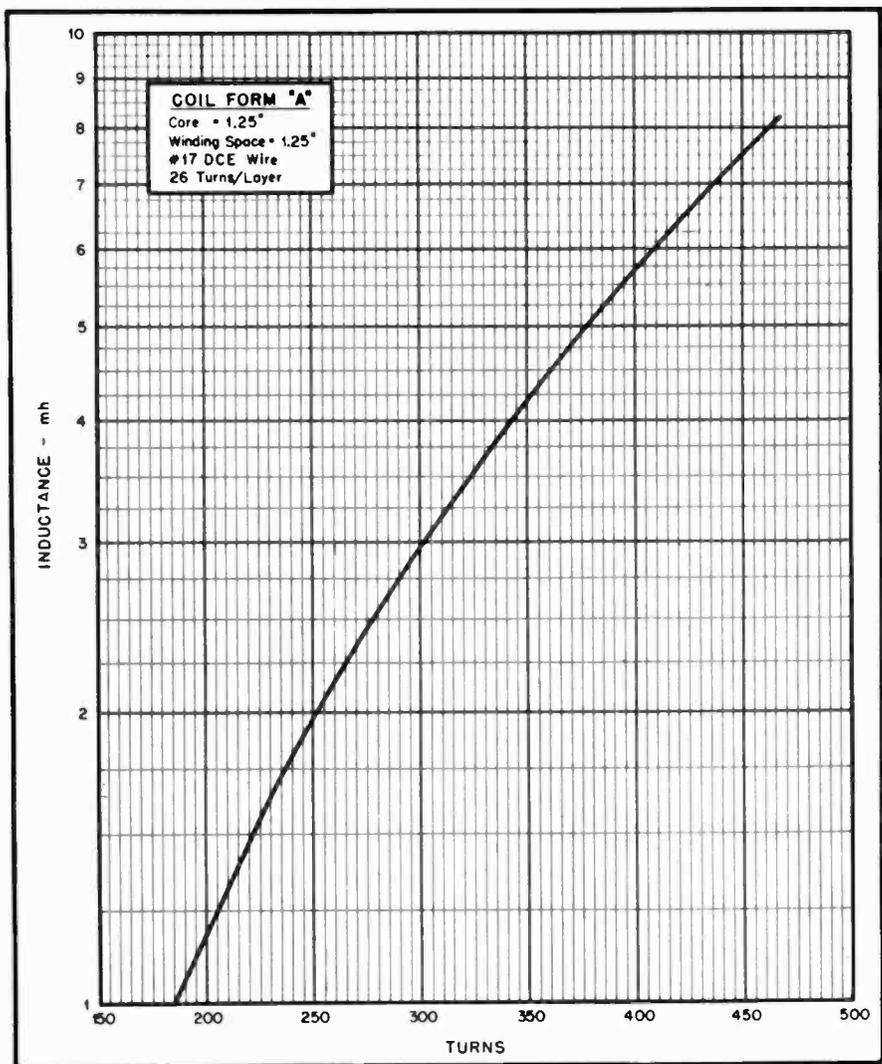
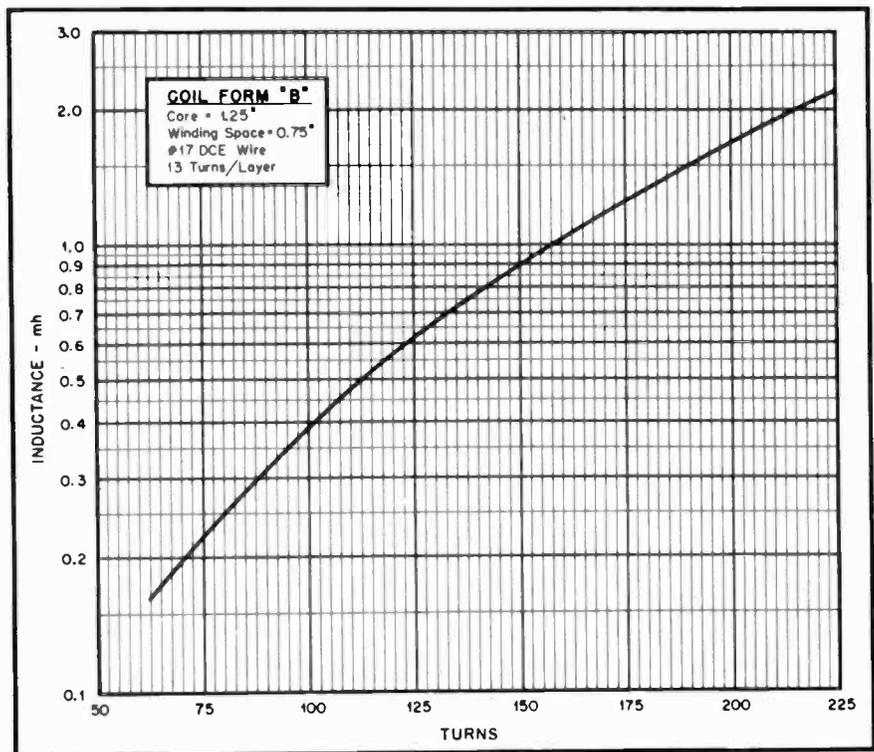


Fig. 5 (above). Turns necessary to obtain various inductance values on coil form "A".

Fig. 6 (below). Turns for various inductances on coil form "B".



simpler to wind the form with the required number of turns and leave the winding in place. The start and finish of the winding may be brought out through holes in the flange and attached to suitable terminals. Two or three coats of lacquer will furnish a protective coating for the coils, but if they are to be used in humid climates, they should be baked out and impregnated with varnish.

After winding the coils, the network should be assembled on a wooden base, making sure that the two coils are not mounted closer than six inches from each other, and that their axes are at right angles. If it is desired to enclose the entire network in some fashion, a wooden box is recommended; the use of metal in proximity to the coils tends to reduce the  $Q$ , and may affect the over-all performance.

### Tests

If an audio oscillator and gain set is available, the constructor will wish to make measurements of the transmission characteristics through each section of the network, the other section being suitably terminated. If such equipment is not available, it is possible to make a reasonable adjustment solely by means of the ear, which is the ultimate instrument for which the speaker system is designed. For the constructor who makes several speaker systems, the test unit shown in Fig. 7 will prove an aid in balancing and phasing.

The two-pole, three-position switch permits the feeding of either speaker independently, or both together. When either section is cut off, the circuit is terminated properly. The 1-db/step pot provides for the attenuation of the high frequency unit for determination of the required amount which must be introduced by the resistor  $R_1$  in Fig. 1. The DPDT switch in the leads to the high-frequency unit provides for reversing the

Loss db	Series Arm (2 required)	Total Shunt Arm
1	0.58	86.7
2	1.14	43.0
3	1.71	28.4
4	2.26	20.9
5	2.80	16.4
6	3.32	13.4
7	3.82	11.2
8	4.30	9.45
9	4.76	8.13
10	5.20	7.02

phase. In operation, this test unit is set for transmission to the low-frequency speaker, and the acoustic output at the crossover frequency is measured using a microphone and amplifier with an output meter. The switch is then thrown to the high-frequency position and the pot is adjusted to obtain the same acoustic output from the high-frequency horn. Then, with the switch in the center position, both speakers are energized at the crossover frequency and the DPDT thrown from one position to the other to obtain the maximum output as measured by the microphone and amplifier. Fine adjustment of phasing is then made by moving the high-frequency unit and horn back and forth relative to the low-frequency unit to obtain the maximum output from the combination. The relative positions of the two speakers are then noted and final assembly of the speakers is arranged to maintain these positions. If maximum output should happen to be obtained with the high-frequency horn extended beyond the front of the low-frequency baffle, this condition may be corrected by reversing the DPDT switch and sliding the horn back one-half wavelength at the crossover frequency. Table 1 gives values

suitable for constructing the 1 db/step "scaling hook" pot for an impedance of 10 ohms. For any other impedances,  $Z$ , the values given should be multiplied by  $Z/10$ .

Such an elaborate device is not necessary, however, for making final adjustments to two-way speaker systems. The acoustic outputs of the two speakers can be balanced close enough by ear, using the tap on the resistor  $R_1$  of Fig. 1 for adjusting the relative levels.

### Phasing

Phasing adjustments between the two units may be arrived at by listening, preliminary settings being obtained by a simple reversal of the leads to the high-frequency unit, and final positioning being determined by continued listening. It will be noted that when the optimum point is located, the maximum realism will be heard from a two-speaker combination. With crossover in the vicinity of 500 to 1,000 cps, a reversal of the leads will give the effect on speech of jumping back and forth between the two speakers, completely eliminating the illusion of a single source of energy.

Final adjustments on any two-way speaker system should be made with whatever screening material is to be used in place, for two reasons. Most important, there is certain to be some attenuation of the high frequencies, and this should be compensated by adjustment of the resistor  $R_1$ . In addition, however, the illusion of a single sound source is often destroyed when both units are separately visible, and the covering with suitable grill cloth will eliminate this effect and blend the two sources into one apparent source, provided the phasing is correct. It is suggested that no adjustments to a system of this type be considered final until the listener has "lived with" the speaker for several days, and it should be expected that gradual improvements may be obtained in the over-all performance for the first two or three months after its installation as the user becomes familiar with its characteristics.

With this simplification of dividing network design and coil construction, it is believed that the average high-fidelity enthusiast should be able to obtain reasonably good results with a minimum of equipment being necessary to make adjustments. A loudspeaker system is designed primarily as a medium for transmitting sound to the ear, rather than to a microphone and a group of measuring instruments, and while no depreciation to the value of measurements is intended, it must be realized that the ear is actually the final judge of the performance of any loudspeaker system. If the constructor is able to obtain satisfactory results from the information contained in this article, its function is fulfilled completely.

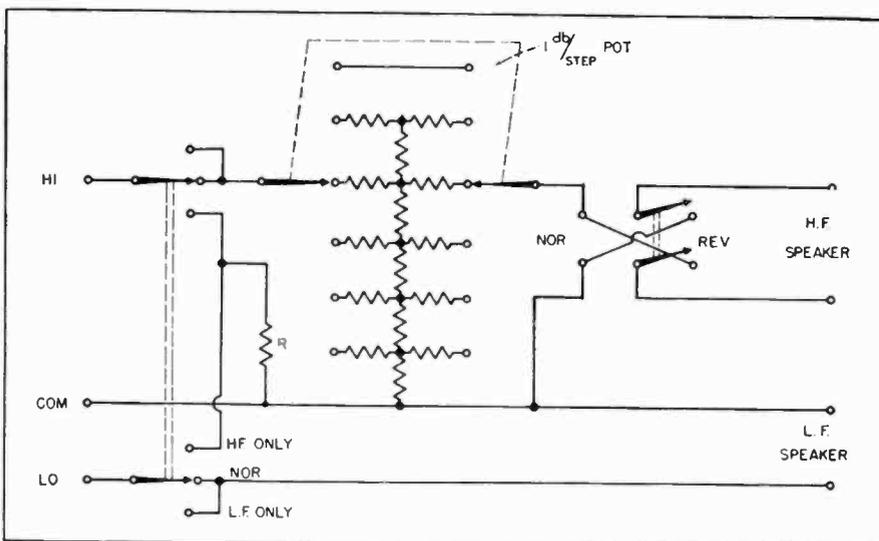


Fig. 7. Test unit for simplifying the balancing and phasing of l-f and h-f speakers.

# Multiple Speaker Matching

**I**N PUBLIC ADDRESS systems using a number of loudspeakers, it is considered good practice to regulate the amount of power delivered to each speaker in the system in accordance with its requirements. There are a number of ways in which this can be accomplished, but the simplest is by proper choice of impedance for the transformers matching the speaker to the line. With this method, no power is lost in resistance attenuators, and at high power levels every watt of power must be utilized most efficiently.

For most systems, the usual connection of the output lines puts the various loads in parallel across the output transformer. This simplifies the wiring, since all distribution lines branch out from the amplifier, and one pair of wires feeds each speaker or each line of speakers. When the speakers are arranged in a long loop, however, it may sometimes be more economical to feed them from a series circuit, so that a single lead can be used for the wiring. These two methods are shown in Fig. 1 and Fig. 2, together with the formulas for calculating the required primary impedances. In these formulas,  $P_t$  represents the total available power, with  $P_1, P_2, P_3 \dots P_n$  representing the power delivered to each speaker; and  $Z_s$  represents the source impedance — which is the output impedance into which the amplifier is to feed — with  $Z_1, Z_2, Z_3 \dots Z_n$  representing the reflected primary impedances of the individual speaker transformers.

## Slide Rule Method

Although the calculations for either type of connection are quite simple once they are reduced to these formulas, a recent attempt to simplify the problem even more by means of a chart indicated a slide-rule method for determining the correct impedances.

For the case where the parallel connection is to be used, let us consider the following example: The system in a small auditorium is capable of putting out a maximum of 15 watts at an impedance of 600 ohms; the stage speaker line requires 12 watts, a speaker in the lounge requires 2 watts, and a speaker in the manager's office requires 1 watt. What is the required primary impedance of each of the speaker matching transformers?

For the solution, each circuit is handled separately, using a slide rule for each computation. Opposite 15 on the "D" scale, representing the total available power, set 12 on the "C" scale,

representing the power required for the stage speaker. Under 500 on the "C" scale, representing the output impedance of the amplifier, appears 625 on the "D" scale, which is the impedance of the primary of the speaker matching transformer, the secondary value being matched to the speaker itself. Similarly, for the lounge speaker, an impedance of 3,750 ohms is indicated; and for the manager's office, an impedance of 7,500 ohms is indicated. All three of these primary impedances are connected in parallel across the output circuit of the amplifier, and the resulting impedance is given by

$$Z = \frac{1}{\frac{1}{625} + \frac{1}{3,750} + \frac{1}{7,500}}$$

$$= 500 \text{ ohms.}$$

The calculations for the series speaker circuit are performed in a similar manner, except for the position of the two impedances on the slide rule. For the same problem as before, the settings for the power available ("D" scale) and for the individual load power required

("C" scale) are used, but the impedances are read from the opposite scales. Thus, the output impedance of the amplifier is on the "D" scale, and the required impedance for the matching transformer is directly above this value on the "C" scale. For the example, then, the three impedances are 400, 66.7, and 33.3 ohms, which, when connected in series, match the 500-ohm output of the amplifier.

The decimal point must be watched carefully in these calculations, as in all other slide rule uses. As an aid in this regard, it is simple to remember that all impedances are higher than the source for the parallel connection, and all are lower than the source for the series connection. Furthermore, it will be noted that if any speaker load is to receive more than one-half of the total power, the transformer impedance will be between one and two times the source impedance for the parallel connection. For the series connection, the transformer impedance for any line receiving more than one half of the total power will be greater than one half the source impedance.

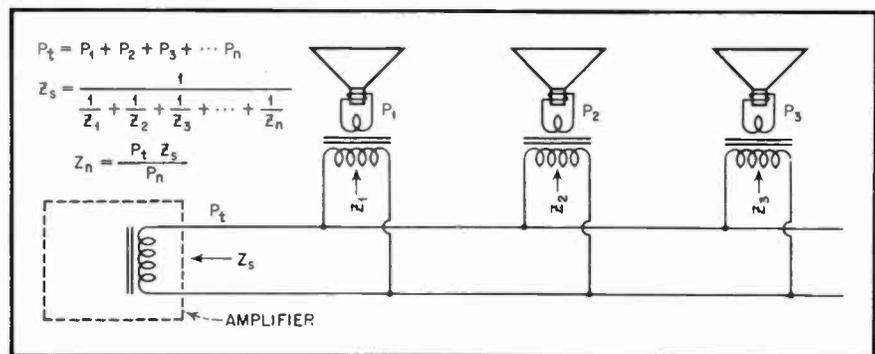


Fig. 1. Typical distribution system for number of speakers fed in parallel from the output of an amplifier. By adjusting the primary impedance of the transformers, the percentage of the total power fed to any individual speaker can be regulated.

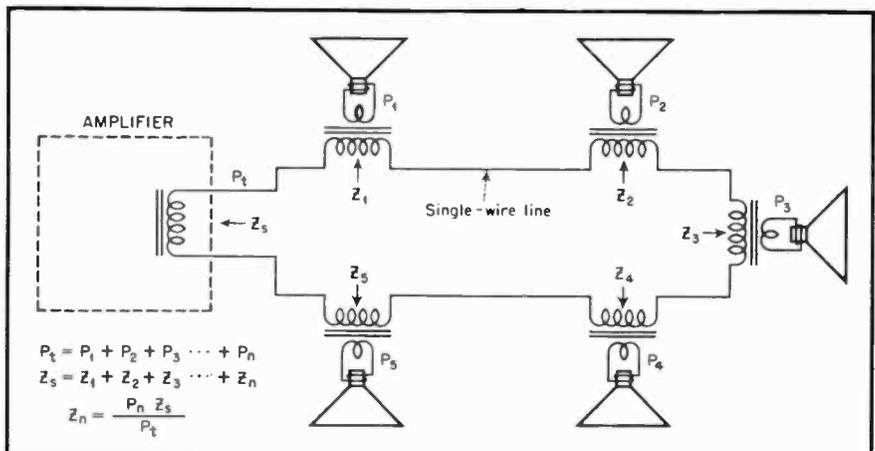


Fig. 2. Alternate method of feeding speakers by means of a series circuit. The use of a single line proves economical where the speakers are distributed in a loop, such as around a race track.

# Residence Radio Systems

C. G. McPROUD

Part I. A push-button operated, fixed-tune FM receiver serving as a high-fidelity source of signal for home radio systems.

**W**HILE the audio engineer is interested primarily in amplifiers, speakers, microphones, and other equipment more closely related to the audio spectrum, it cannot be denied that one reason for high-quality amplifiers and speakers is for the reproduction of radio programs. Thus, the audio engineer must also consider the equipment used to provide the source signal, be it AM or FM.

By far the largest percentage of reproducing equipment is used for home entertainment, and there is some evidence to indicate that many listeners are sufficiently interested in good quality to go to the expense of installing an elaborate system as a semi-permanent part of the house. Such installations come under the heading of Residence Radio Systems, and while they include the same elements as is contained in a modern AM-FM-phonograph combination, the physical construction is usually quite different.

The reasons for this are fairly obvious. It is understood that a good speaker and amplifier are a necessity, and if of excellent quality, they might well be utilized for many years, although changes in the phonograph or in the radio tuners might be made more often. Therefore, in most such installations, the equipment is composed of a number of individual units, each serving a specific purpose, and all tied together by a control system to provide sufficient flexibility.

#### Typical System

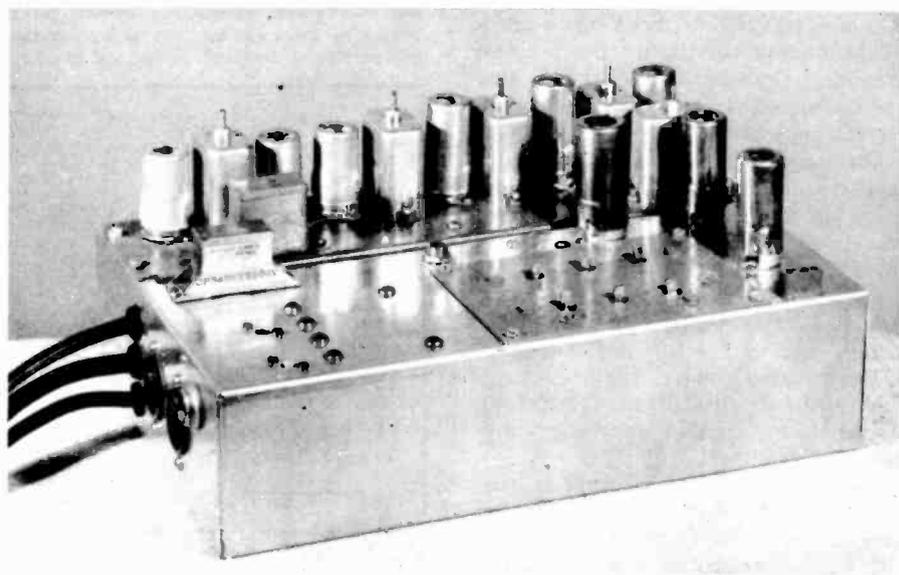
A typical system, therefore, will contain a speaker, a "main" amplifier, a preamplifier—possibly with a scratch suppressor—a record player or changer, one or more radio tuners, depending upon the availability of programs, and a power supply. These sections are basic, and although various sec-

tions may be combined physically, all are represented in most instances. In the more elaborate installations, push-button controls may be employed, arranged for remote operation or not, as the owner desires. Some years ago the writer obtained a clock which is capable of completely controlling a radio, turning it on or off, and selecting any of six stations at any desired fifteen-minute interval for a 24-hour period. With such a device, some form of electrical push-button tuning is necessary.

Because programs on either AM or FM may be desired, the switching can become complicated if usual means are employed, so the system to be described has been assembled with the ultimate aim of obtaining good reproduction in the home from FM, AM, or phonograph, with adequate flexibility and with as few controls as possible. The complete system consists of a two-way speaker, a 6AS7G amplifier, a power

supply, the preamplifier and scratch suppressor unit, and two tuners, each arranged for push-button station selection. Altogether, this equipment provides a complete home entertainment system; it may be either more or less elaborate than another user might wish. It is offered as an indication of what *can* be done in the way of a home entertainment system.

In the New York area, a total of ten stations appeared to be sufficiently popular to warrant inclusion in the system. Six of these are FM stations; four AM. To simplify switching, it was desired that any push button would select its associated station, whether AM or FM, without any other control operation. The clock made one other feature useful, since it can serve as an alarm clock. However, when set to turn off the set after its owner retires for the night—providing a restful lullaby—it would come on in the morning at too low a level to serve as an



Front view of FM tuner chassis.

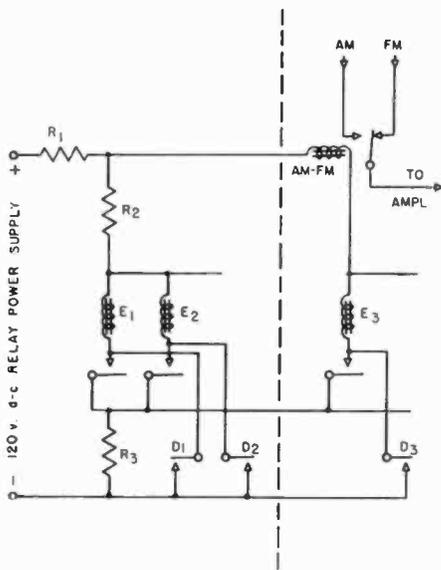


Fig. 2. Basic schematic of self-holding and automatic releasing relay circuits for station selectors.

alarm clock. Therefore, a muting arrangement was included which would reduce the level by about 20 db for the sleeping music, yet, as soon as the clock shut the set off for the night, the system would be restored to normal, and the morning level would be sufficient to serve as an alarm clock.

Thus the problem is stated, and the FM tuning unit is the first to be described. It includes the AM-FM switching relay and the muting circuit, as well as the tuner proper. Relays are used throughout for the various operations, and in spite of the high frequencies involved, the entire unit performs quite satisfactorily.

#### The Tuning Circuits

The front end of the FM tuner is fairly conventional—electrically. However, the antenna and r-f circuit are tuned with variable air trimmers and left set—the antenna circuit peaked at 101 mc and the r-f circuit at 97 mc. The r-f stage is a 6AG5, with its

grid fed with avc voltage from the first limiter. The converter tube is another 6AG5 operated with the oscillator signal fed to an un-bypassed cathode resistor.

The oscillator circuit is a little unusual, and provides the advantages of automatic frequency control, almost a necessity with a fixed-tune unit. From the schematic Fig. 1, it will be seen that the oscillator is of the electron-coupled type, using one-half of a 12AT7 for the oscillator and the other as a reactance tube for the afc circuit. The tuning of the oscillator circuit is accomplished by the use of six variable trimmers, the desired one being selected by a relay. This all seems perfectly straightforward but, on account of the frequency range at which it operates, some care is necessary in its construction. Each relay is electrically self-holding, and once a push button is depressed, the associated relay holds in until another button is depressed, at which time the previous one releases.

This circuit is quite simple, yet requires only one lead to the push button, plus one lead common to all. Referring to the portion of Fig. 2 to the left of the dotted line, the operation may be explained as follows: Suppose push button  $D_1$  is momentarily depressed; current flows through  $R_1$ ,  $R_2$ ,  $E_1$ , and  $D_1$ , energizing the relay which pulls in its armature and completes the circuit through its contact and  $R_3$ . The relays are all 24-volt, 300-ohm surplus units, and require about 60 ma to close and 30 ma to hold. Thus the relay is energized, and holds in with less current than is required to operate it. To change the station, another button  $D_2$  is depressed, causing a greater voltage drop across  $R_1$  and  $R_2$ , so  $E_2$  operates (since  $R_3$  is out of its circuit) but  $E_1$  releases. Thus, the operation of any push button causes the associated

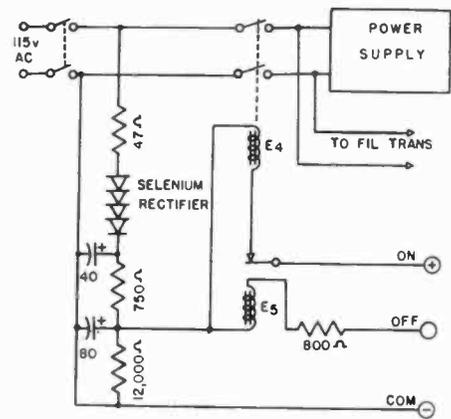


Fig. 3. Power supply for relay circuits, showing main power relay and arrangement for switching on and off.

relay to operate, and releases any other that may have been energized.

When an AM station is to be selected, a button such as  $D_3$  is depressed, and  $E_3$  is operated, releasing  $E_1$  or  $E_2$ . But the current for  $E_3$  flows through the coil of the AM-FM relay, switching the a-f amplifier to the output of the AM tuner unit. Simultaneously, it switches the plate supply on another set of contacts. The operation is reliable, once the correct values are determined for the resistors. Although not built into the FM tuner unit, the power supply for the relay circuit is straightforward, and provides the means for turning on or off the entire system. Thus, when the set is off, it requires only the pressing of a station selector button to turn the set on and select the station. The relay power supply, shown in Fig. 3, is self-explanatory. The current for operating any of the station selector relays passes through the power relay  $E_4$  to close the primary circuit of the power supply and the filament transformers. To turn off the system, relay  $E_5$  is energized, breaking the holding circuit through its contacts.

#### Tuner "Front End"

Considerable flexibility is possible in the front end of the FM tuner. In the unit shown, the six tuning capacitors and the associated relays were mounted in a separate small box 3x3x4 inches, with a slot in the bottom to permit the adjustment and cleaning of the relay contacts. The rotors of all the capacitors were connected together, using #16 bus wire, and the stators were connected to the relay contacts. The capacitors were mounted on a piece of linen Bakelite, and the whole enclosed in a sheet metal box, with holes in the top to permit access to the adjusting screws. The capacitance of each trimmer is  $25\mu\text{mf}$ , somewhat

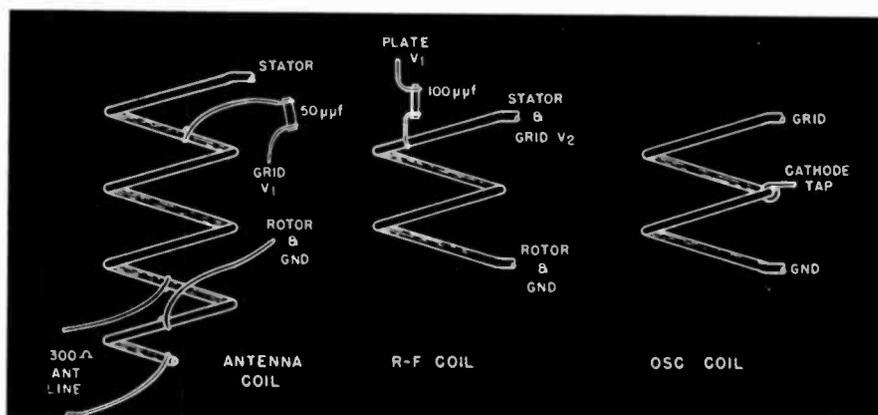


Fig. 4. Coils for front end of the FM tuner. All are wound on  $\frac{5}{8}$ " mandrel of No. 14 copper bus wire.



greater than necessary to tune the band, so the tuning section was connected across one-half of the coil, the band being fixed by a capacitor across the entire coil. This provides a "band-spread" feature, and makes tuning easier. The three coils are shown in Fig. 4, all of them being wound of #14 bus wire on a  $\frac{5}{8}$ " mandrel. The antenna coil has a total of three and a half turns, the ground connection being made at one-half turn from the end. The end and one turn from the end provide the connection for a 300-ohm line, which is effectively balanced to ground by this arrangement. The coil is mounted directly on a crystal socket which serves as an antenna terminal strip. The capacitor coupling to the grid of  $V_1$  is tapped on at one-half turn down from the top.

The r-f coil consists of two full turns, soldered directly to the terminals of the air trimmer, and with the capacitor coupling to the plate of  $V_1$  connected about three-eighths of a turn from the high end. The grid is connected to the stator of the tuning capacitor, with a tip jack provided to facilitate connection to the grid for alignment.

The oscillator coil may take a little more time, but it was found to oscillate perfectly with two turns, with the cathode tap at one turn from the bottom. The tuning capacitors—selected by the relays—are also connected at one turn from ground, as is the capacitor coupling to the converter cathode. The fixed band-setting capacitor connects across the entire coil.

In aligning the oscillator, it will be found necessary to set its frequency on the correct side of the signal, or else the afc circuit will always pull it away from the station. Setting procedure consists of inserting a plug into the afc jack, with leads to a vacuum-tube voltmeter, and adjusting the oscillator to get zero d-c voltage from the discriminator. Then, when the plug is removed, the afc will hold the setting quite accurately. If the oscillator is tuned to the wrong side of the carrier, the afc will always move off the station.

It is preferred to tune the oscillator to a frequency lower than the signal, since a reduction of even 20 mc is an appreciable amount to make the oscillator more stable. Once this is determined, the polarity of the afc voltage may be reversed, if necessary, by reversing the leads from the discriminator transformer to the plates of the 6AL5. Reducing the discriminator heater voltage by the use of the 2.7-ohm resistor aids in preventing "hunting" of the afc circuit.

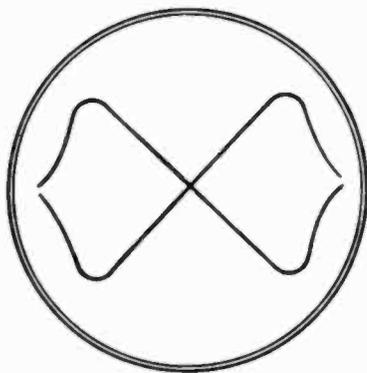
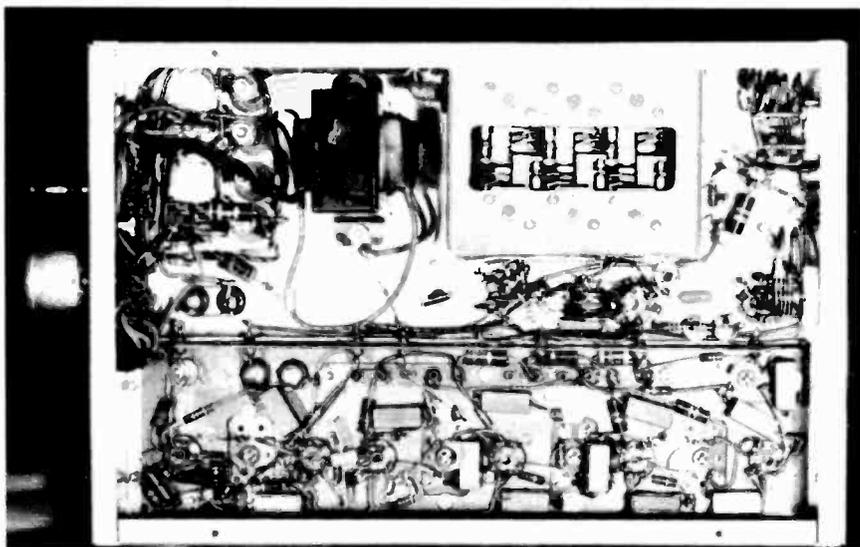


Fig. 5. 'Scope pattern obtained during alignment of i-f amplifier, using 400 kc sweep from frequency-modulated signal generator.

#### I-F Strip

The i-f amplifier and discriminator for this tuner is built on a separate chassis, three inches wide and ten inches long, which mounts in an opening on the main chassis. This construction was followed for two reasons: it is much simpler and more convenient to work on, and in case the relay-tuned front end did not work, it could have been removed and used with a more conventional front end. However, since the entire unit functions satisfactorily, there has been no need to change it. The circuit follows the Meissner 9-1091 AM-FM tuner almost exactly, with the addition of a filter circuit for the afc voltage. The transformers are apparently of the low-impedance type, and it will be noted that three i-f stages and two limiters are used. From the photo it will be noted that mica by-pass capacitors were used, surplus items at five cents each, thus cheaper than paper types. This construction is straightforward and fairly easy to align.



Under the chassis of the FM tuner. Relay box is at upper right.

#### I-F Alignment

The alignment of the i-f section must necessarily follow the construction work. Feed a 10.7-mc signal to the grid of  $V_4$  through a .01- $\mu$ f capacitor, and connect a vtvm to the ave line at "A" on the schematic. Adjust both top and bottom cores of  $T_4$ ,  $T_3$ , and  $T_2$  for maximum voltage on the ave bus. Then shift the output of the signal generator to the grid of  $V_2$ , still using the .01- $\mu$ f capacitor, and adjust the cores of  $T_1$  for maximum. Due to the low impedance of the grid circuit of the converter tube to 10.7 mc, it may be necessary to increase the output of the generator somewhat. However, keep the output to as low a voltage as will cause sufficient deflection of the vtvm.

Then connect the vtvm to a plug and insert into the afc jack, and adjust the top core on  $T_5$  for the maximum voltage, either positive or negative. Adjust the bottom core to obtain zero voltage again. This alignment should be checked several times, because the minimum distortion is obtained only when the adjustments are correct.

If a frequency-modulated signal generator and an oscilloscope are available, the usual procedure is followed for visual alignment except for the discriminator transformer. Connect the vertical input of the 'scope to the output of the tuner, and feed the signal to the grid of the converter as before, using a 400-kc sweep at a 60-cps rate. The sweep circuit in the 'scope should be set for a 120-cps linear sweep.

The pattern on the 'scope will resemble Fig. 5. Adjust the top core on the discriminator transformer for

maximum amplitude of the peaks of the pattern. Then adjust the lower core for maximum straightness of the two crossed lines. The point of cross-over should be in the center of the pattern. These adjustments should be repeated until the peaks are at the maximum, and the lines as straight as it is possible to get them.

#### Construction Hints

Connections to the i-f strip were made with Kovar bead seals, though any type of terminal strip would suffice. A small shield is mounted between the two limiter sockets, and between the r-f stage and the converter tube socket. All resistors are one-half watt types, unless otherwise marked on the schematic. The two chokes—one in the heater circuit of

the oscillator and one in the heater circuit of the discriminator—may be made by winding a 1-meg resistor full of #32 enameled wire, soldering the ends of the wire to the resistor leads.

The muting relay is actuated by a separate push button, and when once set, the station selector relays cannot be changed without switching the set off first. In addition to lowering the level by about 20 db, the mute also changes the frequency characteristic, giving a 3-db/octave boost below 200 cps to compensate for the hearing curve at the lower level. Power connections to the unit are through attached cables, as is the output signal. Separate cables are used for the rectified line voltage for the relay circuits, and for the plate and heater supply

voltages. The filament transformer for the tuner is mounted on its chassis, and 115-volts is fed from the power supply, and through to the AM tuner. A nine-terminal socket provides connections for the push buttons, an octal socket furnishes power to the AM unit, and a microphone-type socket receives the output from the AM tuner.

No trouble should be encountered in this construction. It is desirable to employ a steel plug in the miniature sockets when wiring to them to avoid misaligning the contacts, with the possibility of damage to the tube when it is later inserted. However, make sure the plug is removed before applying power. This information is offered as the result of experience, which necessitated replacing the 6AL5 heater choke.

## Part II. Details of single-channel AM receiver chassis construction. With each station having its own receiver, optimum operation is obtained.

**A**N ideal AM tuner for a Residence Radio System may naturally take one of many forms, but the quality enthusiast is most certainly likely to insist on a t-r-f tuner to ensure complete listening satisfaction. Wide range t-r-f tuners are complex instruments, and while it is not difficult to design such a tuner, the execution of the design may be somewhat more involved. To be sure, several such tuners are available commercially, either as complete units or in kit form. However, for the system covered by this series, a different arrangement was desired.

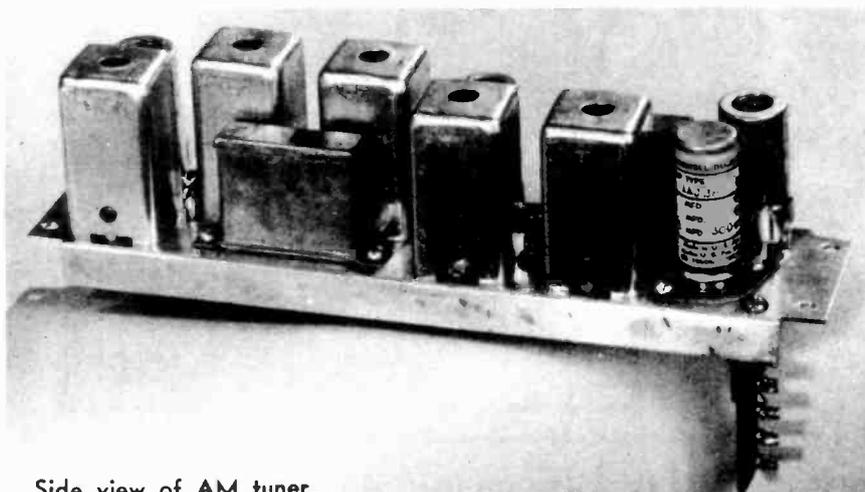
The reasons for the development of the fixed-station tuner units to be described are several. Primarily, the particular installation planned by the writer required that four AM stations

be accommodated, and that they be selected by push buttons which actuated relays. Additionally, fixed station tuners permit a large residence installation to be made with provision for feeding an audio signal from each desired station throughout a home, with a selector switch and an individual amplifier and speaker at each listening point. This allows a number of listeners to choose their own program fare without disturbing others' choices. A still further application for the fixed tuner is for recording studios which need high-quality receivers for off-the-air recordings. A separate tuner may be set up for each station frequently recorded, with the assurance that the audio signal is optimum for each.

A single-station tuner has at least

two advantages over a conventional receiver. It may be tuned carefully and accurately, with the band-width set to the best value for the conditions involved, and it is always ready for instant service simply by switching its output to the recording or audio amplifier. The principal disadvantages are the cost and the space required, but the design shown reduces the cost considerably, and the units are small enough that a six-station assembly can be mounted behind a single 5¼-inch rack panel with ease. For recording studios which must record several programs simultaneously, the strip-tuner arrangement is actually less expensive, since the conventional arrangement would require a complete receiver for each recording channel, which actually duplicates the facilities. For example, ten strip tuners will provide service from ten stations to any number of recording machines at greatly reduced cost over an equal number of conventional receivers which would be necessary to record the ten stations simultaneously.

The quality requirements, together with those of simplicity, dictate the use of a t-r-f circuit for the single-station tuner, and an arrangement using two r-f stages and an infinite-impedance detector serves adequately for the purpose. Because no more than one station is to be received at any time, no audio stage is included, and the switching from station to station is done by switching the plate supply



Side view of AM tuner.

from unit to unit, as will be described. This method eliminates complicated switching arrangements, and functions perfectly. For good quality, the interstage coupling is band-pass, using low-impedance capacitance coupling. The circuits of Fig. 6 represent two methods of band-pass coupling, using capacitance for the common impedance. The tuned circuits are both adjusted to the frequency of the desired station, and the coupling capacitance  $C_m$  adjusted for optimum—or the maximum usable—band width. Actually, with this form of coupling, a change in the value of  $C_m$  causes the two resonant peaks to move apart, but not equally from the previous single peak. However, they may be set properly during alignment, and no trouble arises from this effect.

The response curve of either circuit of Fig. 6 is shown in Fig. 7, with  $\Delta f$  being the separation between the peaks. The usable band width can be shown to equal  $\sqrt{2} \Delta f$ , and  $\Delta f$  is approximately equal to  $Kfo$ , where  $K$  is the coefficient of coupling. For the circuit of (A) in Fig. 6,  $K$  is roughly equal to  $C_m/C_o$  when the two inductances have equal values of  $Q$ . Since a 20-kc band width at 1,000 kc gives a  $\Delta f$  of  $20/\sqrt{2}$ , or 14.14 kc,  $K$  becomes  $14.14/1,000$  or .014, and  $C_m$  is a relatively small value of capacitance when practical values of  $L$  and  $C_o$  are used.

For the circuit of (B) in Fig. 6,  $K$  is approximately equal to  $C_o/C_m$ , so  $C_m$  becomes considerably larger, and thus easier to adjust easily. This type of coupling is known as low-impedance capacitance coupling, and is employed in the tuner strips.

In order to reduce the effect of the valley between the two peaks, a single tuned circuit is employed for the antenna coupling, as shown in Fig. 8. The coils used have adjustable cores, and the tuning capacitances are fixed. Since a number of tuners are to remain connected to the antenna at all times, the coupling to the antenna is such as to load the circuit only at the frequency of operation, rather than throughout the entire band. Thus a series resonant circuit is used for tuning the input, and the peaks of the over-coupled circuits are flattened out appreciably. The tuning circuits are seen to consist of a peaked series resonant circuit ahead of the first tube, a band-pass circuit between the two r-f stages, and another band-pass circuit between the second r-f stage and the detector.

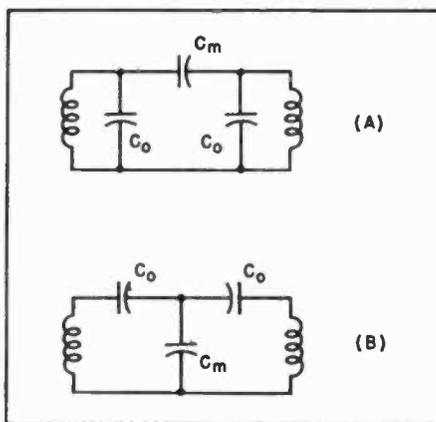


Fig. 6. Two methods of obtaining band-pass coupling by the use of two tuned circuits and an impedance common to both. (A). High-impedance capacitance coupling; (B). Low-impedance capacitance coupling

#### Station Switching

The method used to select the desired channel is quite simple, involving only the switching of the plate supply. This is made possible by using an infinite-impedance detector for each channel, and making a part of the cathode-ground resistance of each detector common to all channels, as shown in Fig. 9. Then, when the plate voltage is switched, the detector in the active channel passes current, with the common portion of the cathode-ground resistor serving to develop the signal voltage. This voltage is then fed to the audio amplifier. Since the value chosen for the common section is 18,000 ohms, while the remaining section is 82,000 ohms, the shunting effect of the remaining channels is negligible. Holding relays of the type described in Part I are used to switch the plate supply in the four-channel unit being used in the complete system.

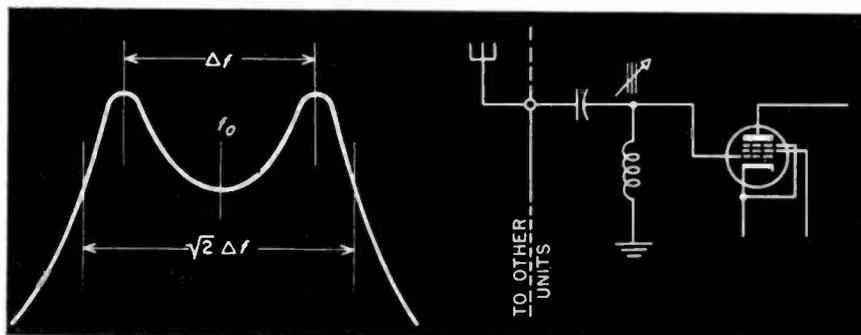


Fig. 7. (left). Curve resulting from over-coupled circuits of Fig. 6. The usable band width is represented by the limits  $\sqrt{2} \Delta f$ . The valley between the peaks can be filled by using a single-tuned circuit in conjunction with the double-tuned circuits. Fig. 8. (right). Method of coupling the antenna to each of the tuner strips. This enables tuning the antenna circuit for each station and provides adequate signal input to the first stage.

The audio output level is fixed with respect to the detector, and since no avc is used, the gain is adjusted by varying the bias on the r-f stages. Each r-f stage has an isolating minimum bias resistor, with the common section being selected for suitable signal to the detector.

#### Construction

Convenient mounting of a number of strip tuners demands that they be reasonably small, with a  $2\frac{1}{2} \times 10$  in.

TABLE I  
Turns of wire to be removed for various ranges

Freq. Range	$C_o$ $\mu\mu\text{f}$	Turns Removed
550—800	50	0
800—1,000	50	40
960—1,175	50	70
1,075—1,300	50	90
1,300—1,550	25	80
1,500—1,700	25	100

chassis being about the minimum for the necessary parts. Five coil shields are required, together with space for three tubes and a filter capacitor for the detector plate supply. The necessary bypass capacitors are in two inverted cans, each containing three 0.1-mf sections.

The complete schematic is shown in Fig. 10. For the r-f coils, CTC<sup>1</sup> permeability-tuned coils type LSM are used. These coils are made in several inductance values, the 1-mc coils being used for this application. Since the inductance variation is not large, turns must be removed from the coils in the broadcast band. The number of turns to be removed are shown in Table I for use with a 50- $\mu\mu\text{f}$  capacitor for  $C_o$  from 550 to 1,300 kc, and 25- $\mu\mu\text{f}$  capacitors for  $C_o$  for frequencies above 1,300 kc. The relative small

<sup>1</sup>Cambridge Thermionic Corp., 445 Concord Ave., Cambridge 39, Mass.

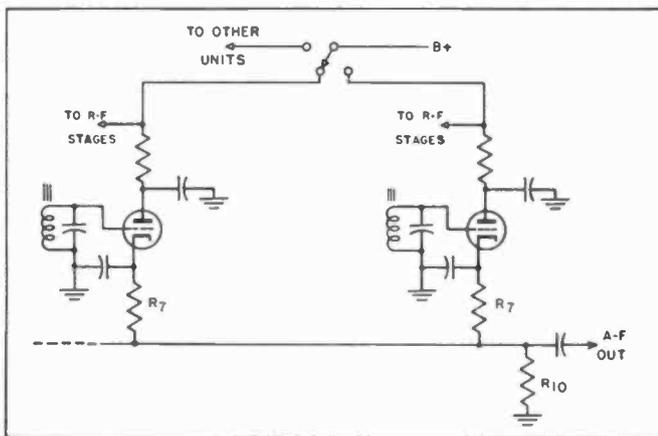


Fig. 9. Switching between channels is accomplished by simple switching of plate supply. Since  $R_{10}$  is common to all channels, the a-f output is developed across it for the one channel which is provided with plate supply.

tuning capacitances provide a high LC ratio, and the gain is fairly constant for all the bands. The approximate values for  $C_m$  for a 20-kc band width are shown on the curves of Fig. 11, and for convenience in adjusting, it is suggested that  $C_m$  be composed of a fixed mica capacitor of suitable value paralleled with a 320-to-1,000  $\mu\mu\text{f}$  mica padder. This will permit a reasonable adjustment of band width.

The coil shields are 1- $\frac{1}{8}$  in. square by 2- $\frac{1}{8}$  in. high, and are sufficiently large to allow for good performance from the coils. The construction is straightforward, and after laying out the chassis, should offer no difficulties. There is no reason why several such tuners should not be built on a single conventional chassis since only one operates at a given time, and there is no chance of interaction between them.

**Adjustment**

Satisfactory alignment almost demands the use of a frequency modulated oscillator and an oscilloscope if proper adjustment of band width and correct flat-topping of the response curve is to be obtained. The second coupled pair of coils should be aligned first, feeding the signal to the grid of the second r-f stage. After the desired response is obtained, shift the oscillator to the antenna terminal and align the coils between the two r-f stages. For convenience, a potentiometer should be connected in place of the gain-setting resistor,  $R_9$ . The antenna coil should be adjusted when all the individual tuners are connected together and to the common antenna terminal, though for flat-topping it will be necessary to make preliminary adjustments on the single chassis. If an f-m oscillator and 'scope are not

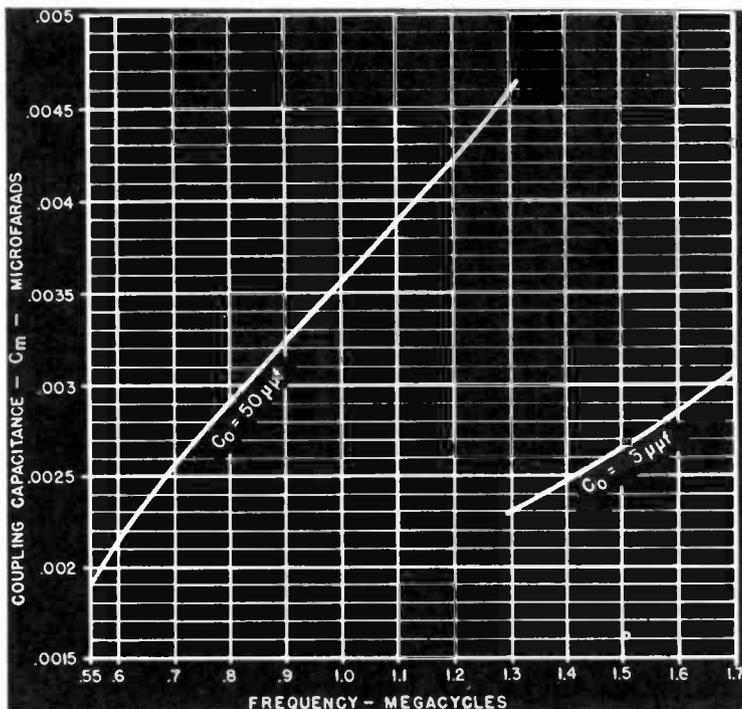


Fig. 11. Curve showing values of  $C_m$  for 20-kc band width with the coils recommended, and the values given for  $C_o$ .

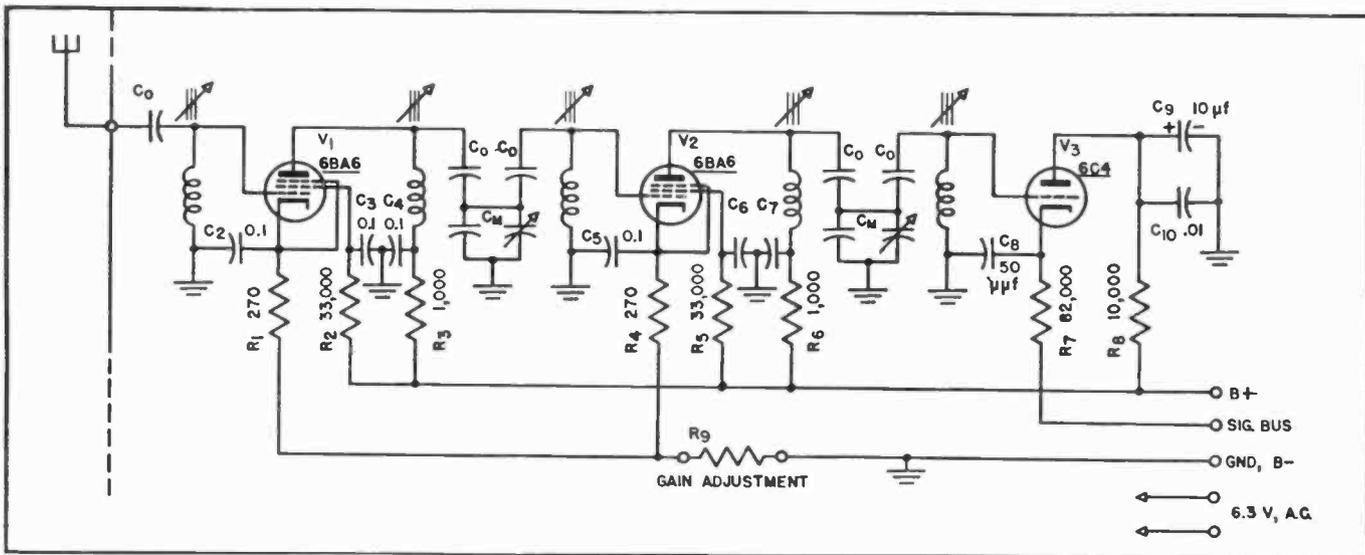
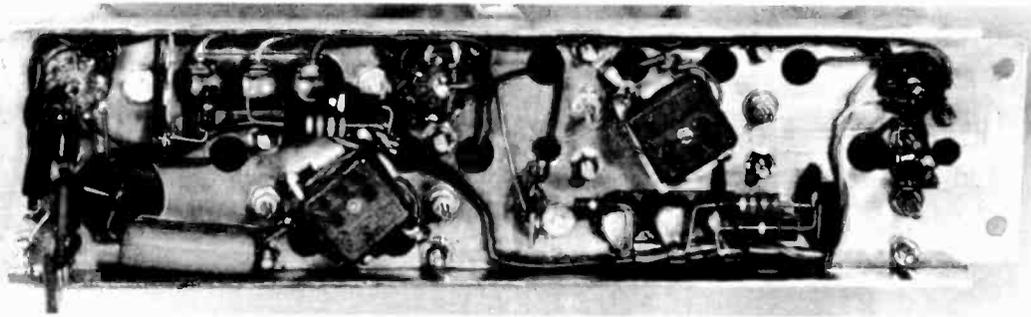


Fig. 10. Over-all schematic for the AM tuner strip. Values for  $C_o$  and  $C_m$  are given in the text, Table I, and Fig. 11.

Bottom view  
of  
AM tuner.



available, the tuner can be aligned approximately by ear, although it is doubtful if the response curve will have the ideal flat-topped shape. However, if no other means is available, fair results can be obtained by this method of alignment and the quality should be at least as good as that of a conventional receiver.

After the channels are all aligned and installed, they may be adjusted to equal audio output levels by selecting a suitable value for  $R_9$  in each. Probable values for use in metropolitan areas will range from 1,200 to 2,700 ohms. Too low a value may cause oscillation, but the proper point at which the detector should work is the important parameter, and it is doubtful if it will be necessary to reduce  $R_9$  below 1,000 ohms. Each r-f stage is essentially a tuned-grid, tuned-plate arrangement, and such a circuit is prone to oscillate if external feedback is sufficient or if the gain of the stage increases too greatly. The d-c voltage at the cathode of  $V_3$  should be from 15 to 20 volts for a 200-volt plate supply.

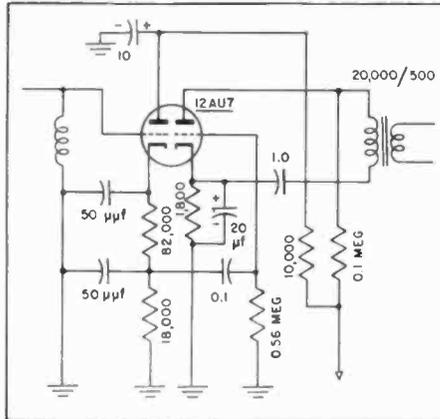


Fig. 12. Alternative output arrangement suitable for use in recording studios where several channels may be used simultaneously. Transformer matches output tube to 500-ohm line.

and this value is dependent upon the r-f signal applied to the grid.

With a number of these channels it is possible to select stations by means of a single-gang rotary switch, all filaments being heated whenever the system is turned on. For use in

recording studios, however, an additional stage should be added, together with a tube-to-line transformer, as indicated in Fig. 12. One section of the 12AU7 twin triode is used for the detector, while the other is the audio output stage. Some such arrangement is required if it is necessary to record more than one station simultaneously.

For a large residence installation, the low-impedance outputs of several fixed tuners may be fed at approximately zero level throughout a home, with a selector switch and an amplifier at each room where a speaker is desired, or any of a number of switching arrangements may be employed. There is practically no limit to the applications to which the fixed tuner strip may be put, and their use will depend upon the ingenuity of the constructor. The cost for each tuner should be somewhere around \$10.00, without the transformer, and for those who wish convenience and quality, they are considerably superior to more conventional receivers.

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

IN THE FIRST TWO ARTICLES of this series, both AM and FM tuners were discussed as suitable input sources for a specific type of Residence Radio system. These units were designed with two primary considerations—quality of reproduction and convenience in use. Tuners alone do not make a radio receiver, and this article covers the next stage in the system. The phonograph preamplifier and the noise suppressor are grouped into one unit known as the control amplifier, so called because it is the only point in the system where any control over volume or frequency response is exercised. It includes a number of circuit sections, but is constructed as a single unit. Throughout this entire system, professional practices are fol-

lowed to a large extent, and the output of the control amplifier appears as a 600-ohm line feeding the power amplifier, which has a transformer input. This practice has the advantage of separating the various sections into units which may be changed individually if a desire for modification should arise or if new developments make it advisable, thus avoiding scrapping a complete system as would be necessary if it were built in one unit. It is not economical from the standpoint of tubes or chassis or plugs and cables, but it does provide a high degree of flexibility.

The control amplifier consists of a phonograph equalizer with two turn-over points, a preamplifier, switching facilities for selecting various inputs,

a Scott-type noise suppressor, and a compensated volume control. Since the suppressor can serve as a low-pass filter, and since the volume control furnishes almost complete compensation for the Fletcher-Munson hearing curve, no tone controls are employed. If they are ever considered necessary, they may be installed in the 600-ohm line to the power amplifier without disturbing either chassis, and constant-impedance equalizers can be used in accordance with professional practice. So far, no such controls are deemed necessary.

The physical arrangement is shown clearly in Fig. 13. The chassis is a standard 7x15x3 radio chassis, but all the tubes and some other components are mounted on one of the 3x15 sides,

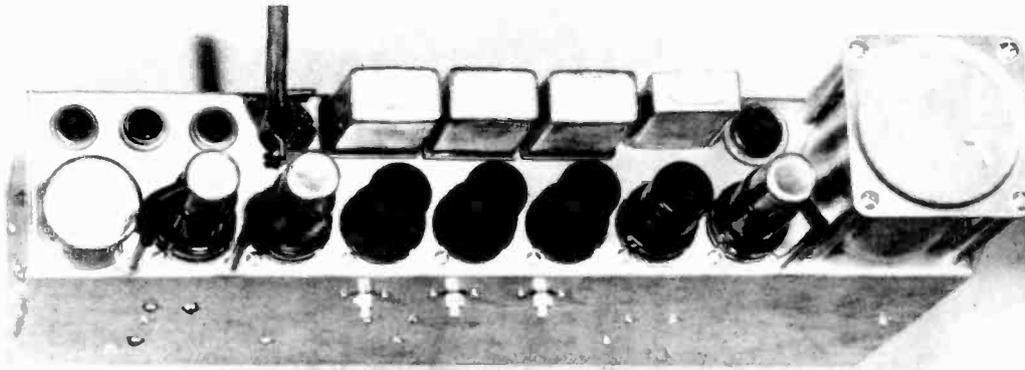


Fig.13. External view of the control amplifier, showing the unique arrangement of the components. Operating controls are on the opposite side of the chassis.

with the controls on the opposite one. This makes parts accessible and provides a desirable chassis arrangement. The plate and filament power are obtained from the main power supply which furnishes 225 volts, regulated, for the plates, 12 volts d.c. for six of the heaters, and 6.3 volts a.c. for the seventh. *Figure 14* shows the inside of the unit before completion, with some of the resistors and capacitors mounted on the strips. Note the use of solid copper tubing for shielding leads that must be run any great distance. These tubes are shaped and soldered in place, and the wire pulled through afterward. Tubing provides better shielding and is self-supporting.

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

#### Phonograph Equalizer

A more accurate low-frequency equalization curve, with lower losses, can be obtained from a magnetic pickup equalizer by the use of low-impedance elements. The unit used here consists of the elements to the left of  $T_1$  in *Fig. 15*, the over-all schematic. To adjust such an equalizer requires the use of an audio oscillator, but once the principles are understood, an ideal equalization curve can be obtained.

The basic equalizer circuit is shown in *Fig. 16*. The series inductance  $L_1$  is tuned by the capacitor  $C_1$  to the turnover frequency and with  $R_2$ , serves to adjust the response in that region. The shunt elements,  $R_3$  and  $C_3$ , control the low-frequency rise. The value of  $R_2$  is somewhat critical, as shown at (A) in *Fig. 17*. The easiest method of adjusting  $C_1$  and  $R_2$  is to reduce the latter to about 2,000 ohms, and then to vary  $C_1$  to obtain a dip at the turnover frequency. Then increase  $R_2$

gradually until the curve is flat above the turnover. This will result in a smooth but fairly sharp turnover. The over-all schematic shows two capacitors,  $C_1$  and  $C_2$ , together with two resistors,  $R_2$  and  $R_3$ . These provide two turnover frequencies, one at 800 cps and one at 400 cps, a compromise between the common 300-and and 500-cps points.

The resistor  $R_4$  adjusts the slope of the curve below the turnover, as shown at (B) of *Fig. 17*. In order to make measurements on this type of circuit, the oscillator should be connected with a series inductance equivalent to that of the pickup to be used. A simpler procedure is to connect the actual pickup between the oscillator and the input terminals, as at (C). The input transformer provides a voltage step-up to the grid of the first tube. Any good quality, well shielded input transformer should suffice, with the primary intended to work from a 250 to 600-ohm source. Resistor  $R_1$  across the input terminals is used to flatten the response over the higher end of the spectrum, and a switch on the phonograph motor board shunts an additional 1800-ohm resistor across the pickup to provide a roll-off almost equivalent to that required for NAB transcriptions, and also necessary with the now quite popular LP records.

This type of equalizer is more efficient than RC types, and has a total loss of only 20 db, not counting the gain in the transformer. Actual measurements on this unit show a voltage of .023 on the grid of  $V_1$  for an input of .016 volts at 1,000 cps. The transform-

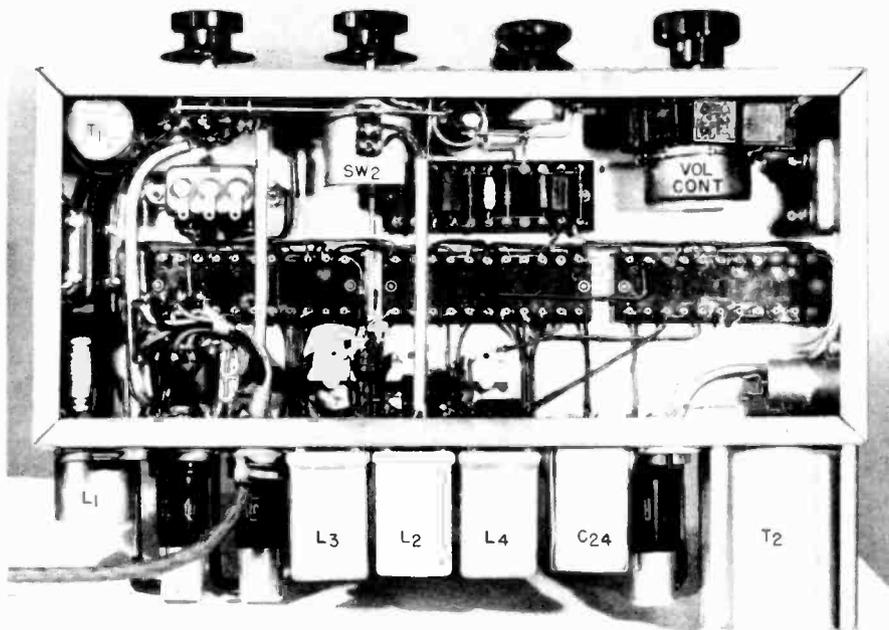


Fig.14. Internal view of the chassis prior to completion. The input transformer is at the upper left corner, and resistors and capacitors will be mounted on the Bakelite strips. Note use of copper tubing for shielding long leads.

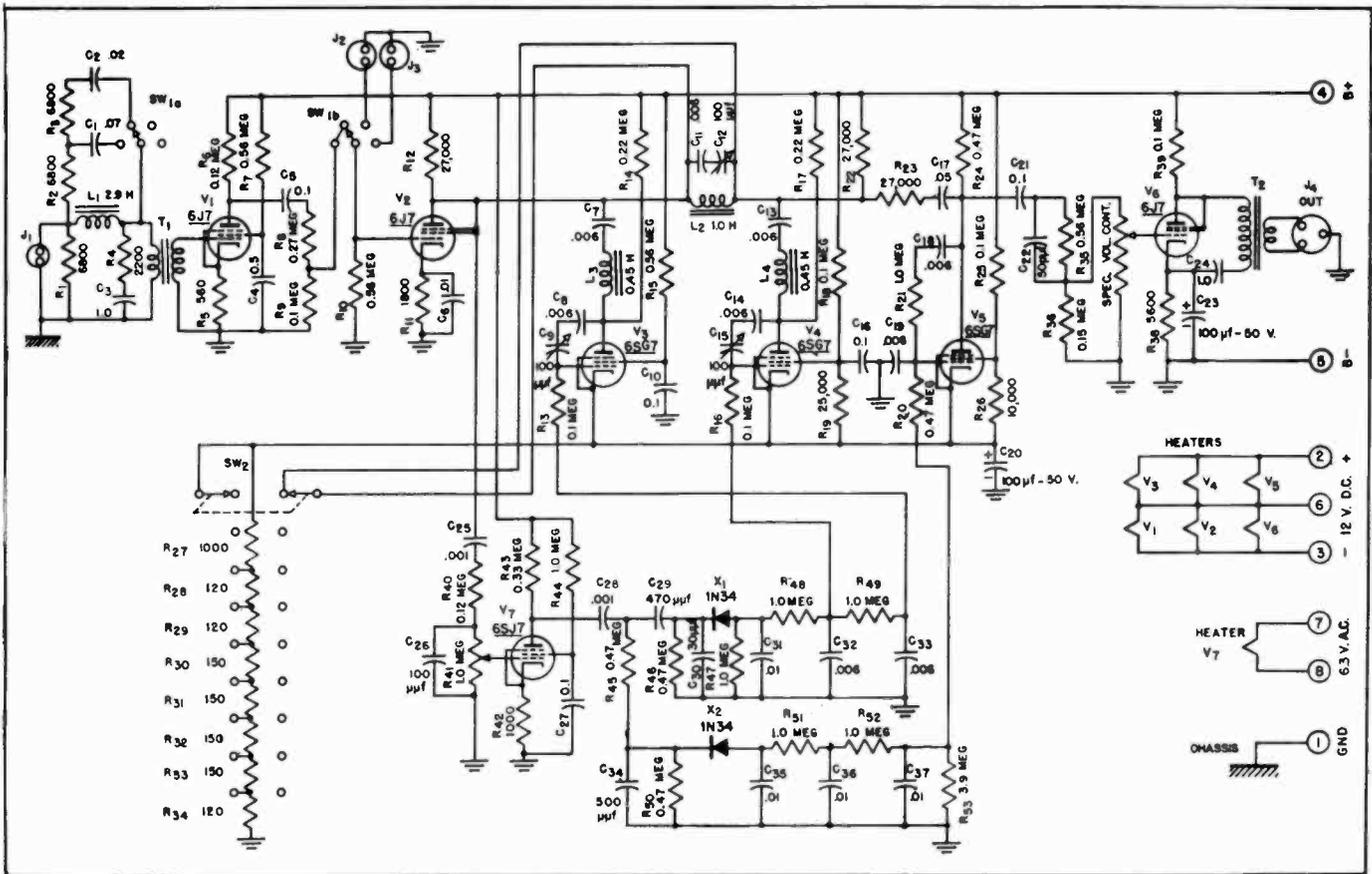


Fig. 15. Complete schematic of the control amplifier and noise suppressor.

er has an impedance ratio of 1:120, or a voltage step-up of approximately 11, yet the equalization realizes the total of 20 db. Typical RC equalizers lose up to 40 db for complete correction, and in addition, the curve is considerably more rounded at the turnover point. The response curve of this equalizer is shown in Fig. 18. The preamplifier tube,  $V_1$ , provides a voltage gain of approximately 31 as used in this circuit.

**The Noise Suppressor**

The input to the second stage,  $V_2$ , is preceded by a selector switch. Two positions are used for records, the ganged section,  $SW_{1a}$ , adjusting the turnover frequency. One other position is used for radio input, and the fourth is a spare.  $R_8$  and  $R_9$  serve as

a voltage divider to balance the output of the preamplifier to the radio input.

The noise suppressor section is similar to that previously described by the writer<sup>1</sup>. However, some modifications have been made, and the reasons therefor will be discussed.  $C_6$  across  $R_{11}$  is used to equalize the frequency response, providing a gradual increase in the high frequencies to compensate for all the circuits "hanging" on the high-impedance line from the plate of  $V_2$  to the volume control.  $C_{11}$  isolates the plate voltage from the stator of the trimmer  $C_{12}$  so the latter will not be "hot" to chassis.  $C_8$  and  $C_{14}$  perform similar functions.

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

Filter circuits which employ high-Q coils are capable of extremely sharp cut-off, and as a result, it is not uncommon for the response to rise 2 to 3 db above normal just before the steep droop. This peak usually results in a poor-sounding output, and its presence is readily recognized by a trained ear. By way of digression, the

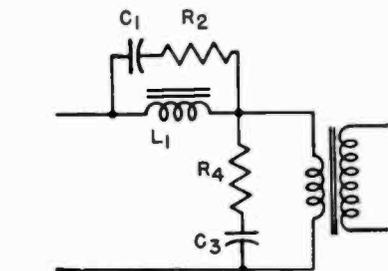


Fig. 16. Simplified circuit of the phono-graph equalizer.

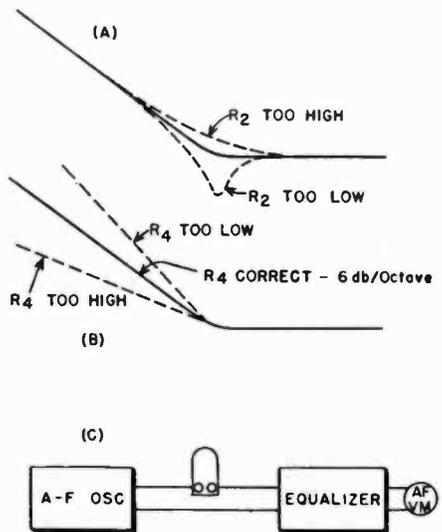


Fig. 17. Phonograph equalization curves at A and B show the effect of resistor values. C shows method of connecting oscillator to obtain correct results.

<sup>1</sup>"General Purpose 6AS7G Amplifier," See page 10.

filter circuits used in this type of noise suppressor are actually of the band-elimination type, and only the lower portion is normally measured. If the measurements were carried up to 40 or 50 kc, the signal voltage would rise again, assuming the other circuits would pass the higher frequencies. Such a filter is normally designed to work between specific impedances. As the suppressor operates, the various tuned circuits must be altered in order to remain correct for a given source impedance. However, in the suppressor, the capacitance of the shunt circuits—that of the reactance tubes—is the only element that is varied. Also, the coil  $L_2$  in the series leg and its tuning capacitors are not changed.

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

Another characteristic which is also objectionable is occasionally observed. That is the return of the response curve after the resonant point of the filter. For example, a filter of this type will have a response similar to that of the solid curve of Fig. 19, where  $f_r$  is the frequency of the dip due to the shunt circuits,  $L_3-V_3$  and  $L_4-V_4$ , and  $f_\infty$  is the frequency of the dip due to  $L_2-C_{11}$ ,  $C_{12}$ . This curve also shows, at A, the rise before the dropoff. The return, at

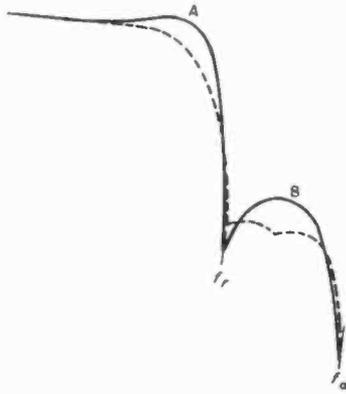


Fig. 19. Curves resulting from noise suppressor. Solid line represents initial measurement before correction, showing peak at A and return at B. Dotted line shows curve after applying corrective measures described.

B, often rises as much as 10 db above the dip at  $f_r$ , and this too does not sound well. Therefore, in this suppressor,  $L_4-V_4$  is tuned to one frequency and  $L_3-V_3$  is tuned to a frequency midway between  $f_r$  and  $f_\infty$ , giving a curve like that of the dotted line in Fig. 19. Now, while the dip at  $f_r$  is not as great as before by about 6 db, the response does not rise appreciably above  $f_r$ , and results are more pleasing to the ear. The problem now is to keep the dip at higher frequency midway between  $f_r$  and  $f_\infty$  during the shift of  $f_r$  by the action of the gate control circuits.

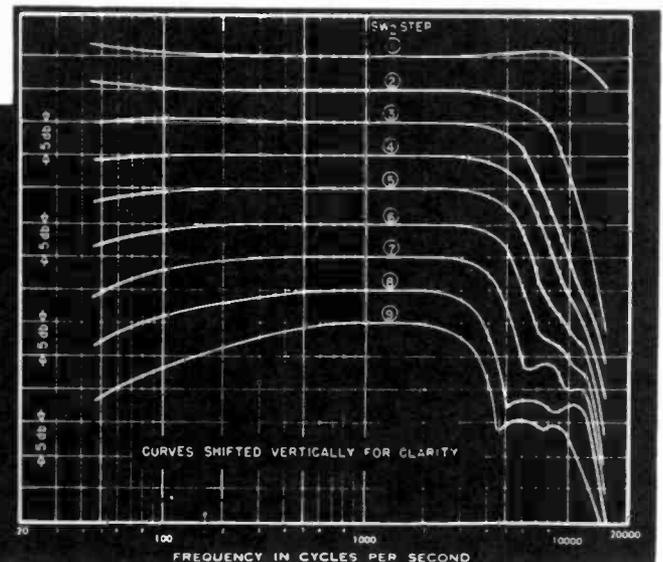
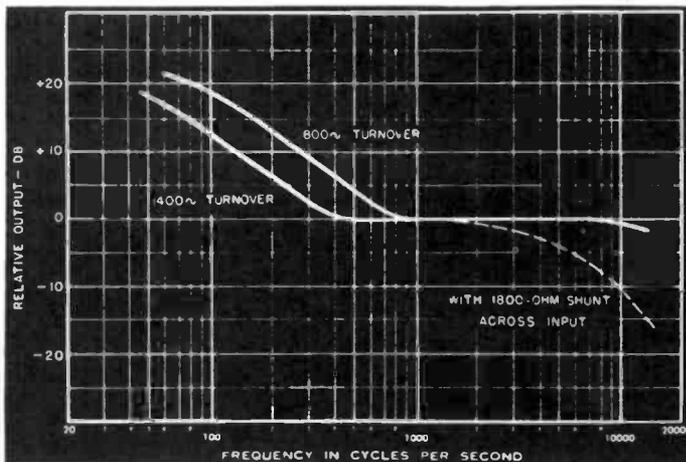
This can be accomplished readily by proper choice of the operating characteristics of the two reactance tubes.  $V_4$  is tuned to 4,500 cps at maximum suppression, and its screen voltage is supplied from a voltage divider,  $R_{18}-R_{19}$ .

This tube is connected to the side amplifier-rectifier circuit to operate more quickly than does  $V_3$ , so the lowest-frequency dip moves upward quickly in the presence of high frequencies in the signal.  $V_3$ , however, has its screen supplied by a series dropping resistor, thus changing the  $\mu$ -bias characteristics of the tube. With this connection, the effective capacitance of  $V_3$  varies less slowly with a given change in bias than does that of  $V_4$ , and the second dip in the curve remains approximately midway between  $f_r$  and  $f_\infty$ , as  $f_r$  is increased by a shift of grid bias.

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

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

Fig. 18. (below). Final response curves of phonograph equalizer with two positions of the selector switch. Fig. 20 (right). Over-all response curves of noise suppressor for nine positions of control switch.



### The Compensated Volume Control

Following the suppressor section is the volume control. This unit is completely compensated for the Fletcher-Munson curve, and was originally described in these pages<sup>2</sup>. Since it is the only volume control in this system, it has more steps than the original model, but the operation is similar. This unit was assembled on an IRC commutator-type attenuator, with the capacitors mounted externally with Kovar bead seals leading through the shield case. The complete schematic of the volume control is shown in Fig. 21.

Until a listener becomes familiar with this type of volume control, he is certain to be somewhat amazed by its characteristics. It is necessary to adjust the over-all gain of the system so that the control is operated about five steps below maximum for normal room volume. This gives a slight boost to the low frequencies, but it is nearly correct for normal room volume when compared to the original sound source. As the control is turned, however, the apparent balance between low and high frequencies remains so nearly constant that it is sometimes difficult to determine whether or not the level has been changed. It is this control that relieves the necessity for tone controls, and the writer is convinced that this circuit is one of the outstanding contributions to the audio art in some time. The combination  $R_{35}$ ,  $R_{36}$ , and  $C_{22}$  serve to adjust the total gain to a suitable amount to take advantage of this type of control, and the response curves of Fig. 22 represent the output at various settings.  $C_{22}$  again adjusts frequency response occasioned by the residual capacitance of the volume control structure.

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

In this amplifier, 6J7's were used because of the grid connection separation. The only tube with a-c filament supply is  $V_7$ , which is not actually in the signal circuit. The other heaters are wired in a series-parallel arrangement, with the center tap brought out to a terminal. Normally operating from a 126-volt d-c supply, terminals 2, 3, and 7 of the terminal strip may

<sup>2</sup>"Loudness Control for Reproducing Systems," David C. Bomberger, page 37.

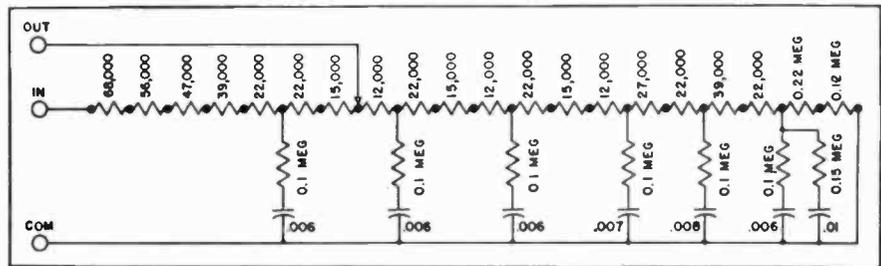


Fig. 21. Schematic of the volume control compensated for loudness contours.

be connected together, and 6 and 8 strapped together, and all tubes fed from a 6.3-volt a-c or d-c supply. The regulated 225-volt plate supply for this unit comes from the main amplifier-power unit. Since 6SG7's have a semi-remote cutoff, they are more desirable for the reactance tubes than either the 6SJ7 or 6SK7 types.

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

### Components

Resistor and capacitor value are all shown on the schematic, Fig. 15, but the characteristics of the transformers and inductances used are not so obvious. In the unit shown  $L_1$  is 2.9 H;  $L_2$  is 1.0 H;  $L_3$  and  $L_4$  are 0.45 H. For these coils, UTC VIC adjustable inductances are suitable and readily obtainable. The types chosen should be

used as near the minimum inductance setting as possible, to insure the highest Q of the coils, which will then be around 15. Therefore,  $L_1$  should be a VI-C15;  $L_2$  should be a VI-C13; and  $L_3$  and  $L_4$  should be VI-C11 units.

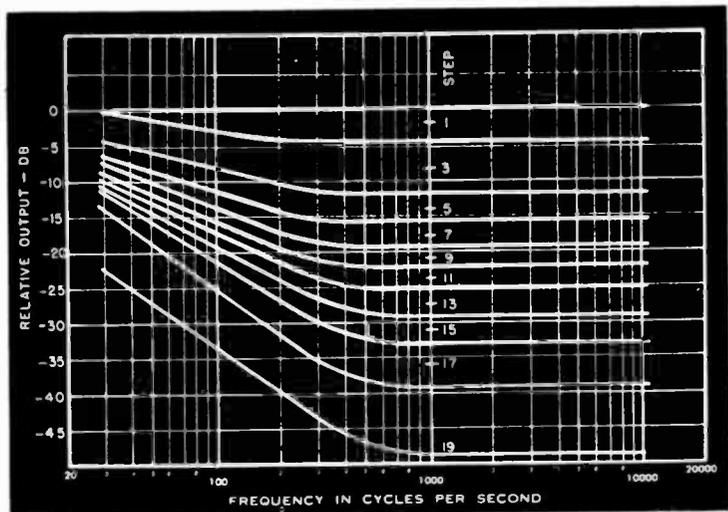
The input transformer used is a Langevin 401B, but suitable substitutions would be UTC A10, ADC 215A, Thordarson 20A05, Chicago BI-1, or Stancor A-4351. The output transformer used is a Western Electric 132C, but UTC LS-27 or HA-113, ADC 215A, Thordarson T-22S92, Chicago BO-1, or Stancor A-3315 should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

### Modifications

It should be understood that the use of the low-impedance equalizer is not necessary to the satisfactory operation of this circuit, but that any of the many pre-amplifiers described in various other articles on preceding pages could be substituted completely for all of the elements ahead of  $SW_{10}$  in Fig. 15.

Another optional change would involve resistance-capacitance coupling from the plate of  $V_6$  to the grid circuit of a power amplifier, such as the "Musicians" described on pages 33-36. Experimenters rarely duplicate any suggested circuit exactly, but often make minor changes to fit their own particular conditions. However, this should not affect performance appreciably, provided these changes are made with care.

Fig. 22. Curves obtained with various settings of compensated volume control.



## Part IV. A description of the main amplifier and the power supply units, concluding the series.

**I**N ANY SYSTEM for radio and phonograph reproduction, there are a number of individual circuit entities which may or may not be physically separate units. The system being described in this series happens to be built in unit form—each section being essentially a separate unit—which has certain advantages for an experimenter's installation. It also has some disadvantages.

The principal advantage of unit construction lies in its flexibility. Suppose, for example, that a complete system were constructed on a single chassis. The wiring is simpler, no plugs or cables are necessary, and it is more economical, both of chassis space and of cost. But if the builder or owner should happen to want a major change in some specific section—and what experimenter is satisfied to leave well enough alone?—he must rebuild completely, or must try to adapt the new design to an already-cut chassis. However, with the unit system, a single section can be replaced easily, without the necessity of disturbing those portions which are working to the owner's complete satisfaction.

In the system being described, there are five separate units: the FM tuner; the AM tuner assembly; the control amplifier consisting of the phonograph preamplifier and equalizer, the noise suppressor, and the volume control; the main amplifier, and the power supply. The first three have already been discussed; this article covers the last two, also built as separate units.

### The Main Amplifier

The final amplifier in this system provides the voltage gain and output power to raise the signal level from the control amplifier sufficiently to drive the loudspeaker. Since the entire system is used only for home entertainment, and the loudspeaker itself is quite efficient, the power requirements are relatively low. The general design of the amplifier follows closely that of the 6AS7G amplifier previously described by the writer.<sup>1</sup> The output is approximately 6 watts at one per cent harmonic distortion, and the average power used with the present speaker is less than one watt.

It will be remembered that the control amplifier terminates in a tube-to-line transformer, and that a 600-ohm line is used to feed the signal to the main amplifier. Thus an input transformer is required for the latter. Briefly, the main amplifier consists of a step attenuator, the input transformer, two stages of amplification using 6J7's as triodes, and employing 17.5 db of feedback, the push-pull interstage transformer feeding the 6AS7G, and the output transformer. The complete schematic is shown in *Fig. 24*.

The input transformer,  $T_1$ , is designed to work across a terminated line, which means that the apparent source impedance is one-half the nominal line impedance. Referring to *Fig. 25*, (A) shows a transformer working from a

<sup>1</sup>"High Quality Amplifier with the 6AS7G", beginning on page 5.

600-ohm line, without a resistive termination. This arrangement does not terminate the line correctly for many circuits, and often results in a frequency response of the preceding equipment which differs from that measured into a resistance load. Some transformers, especially those at the input of a microphone preamplifier, are designed to operate from an open circuit, and similarly most microphones are intended to work into an unloaded transformer winding. Other transformers are designed to work with a resistance termination on the secondary, as at (B), but this often causes a reduction in high-frequency response. The transformer used in this amplifier was designed to work across a resistive termination, so the apparent source impedance is one-half the line impedance, as shown at (C).

This permits the use of a simple arrangement for a step potentiometer, since the transformer offers no load to the line, and it is not necessary to use a T-pad. The main requirement is that the transformer "looks back" at 300 ohms. In order to adjust the gain of the main amplifier to a value which permits the compensated volume control (in the control amplifier) to operate over its optimum range, it is desirable to have discrete steps of attenuation, and 5 db is a suitable value for these steps. Thus the input attenuator consists of an L-pad designed to offer a constant 600-ohm load to the line, and to offer a 300-ohm source to the transformer when the 600-ohm line is connected. The entire attenuator is assembled on a Centralab #1404 switch, with one of the rotor contacts removed, and with the lugs bent back so as to provide a number of tie points for the resistors as shown in *Fig. 26*. The shunt resistors,  $R_1$  to  $R_5$ , are selected from RMA values to provide a total resistance of approximately 600 ohms (actually 605) with 5-db steps. The resistance at point X, for example, is equal to  $(150 + 82 + 47 + 56)$  in parallel with  $(605 + 270)$ , or 242 ohms; 300 less 242 equals 58 ohms for  $R_6$ , and the 56-ohm RMA value is sufficiently close. The same type of calculation is used to determine the values for  $R_7$ ,  $R_8$ , and  $R_9$ .

### Input Transformer

The input transformer has an impedance ratio of 300:90,000 which represents a voltage step-up ratio of 17.3. The secondary feeds the grid of  $V_1$ , a triode-connected 6J7, which is resist-

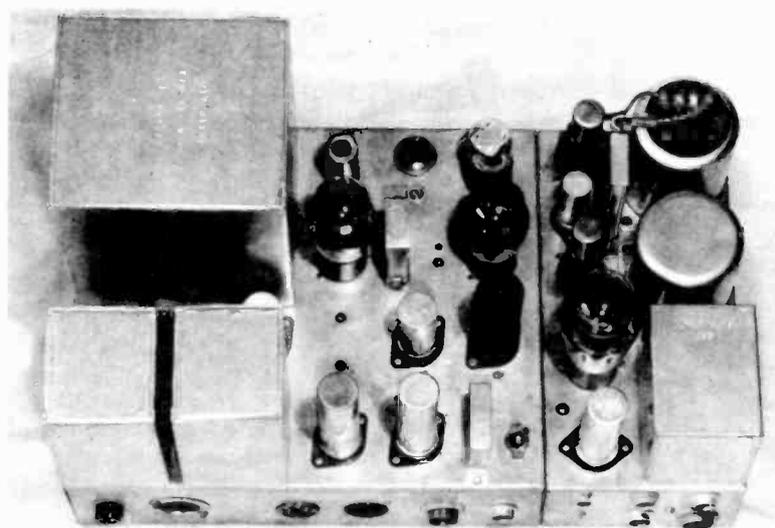


Fig. 23. Top view of the main amplifier and power supply units bolted together as a single chassis.

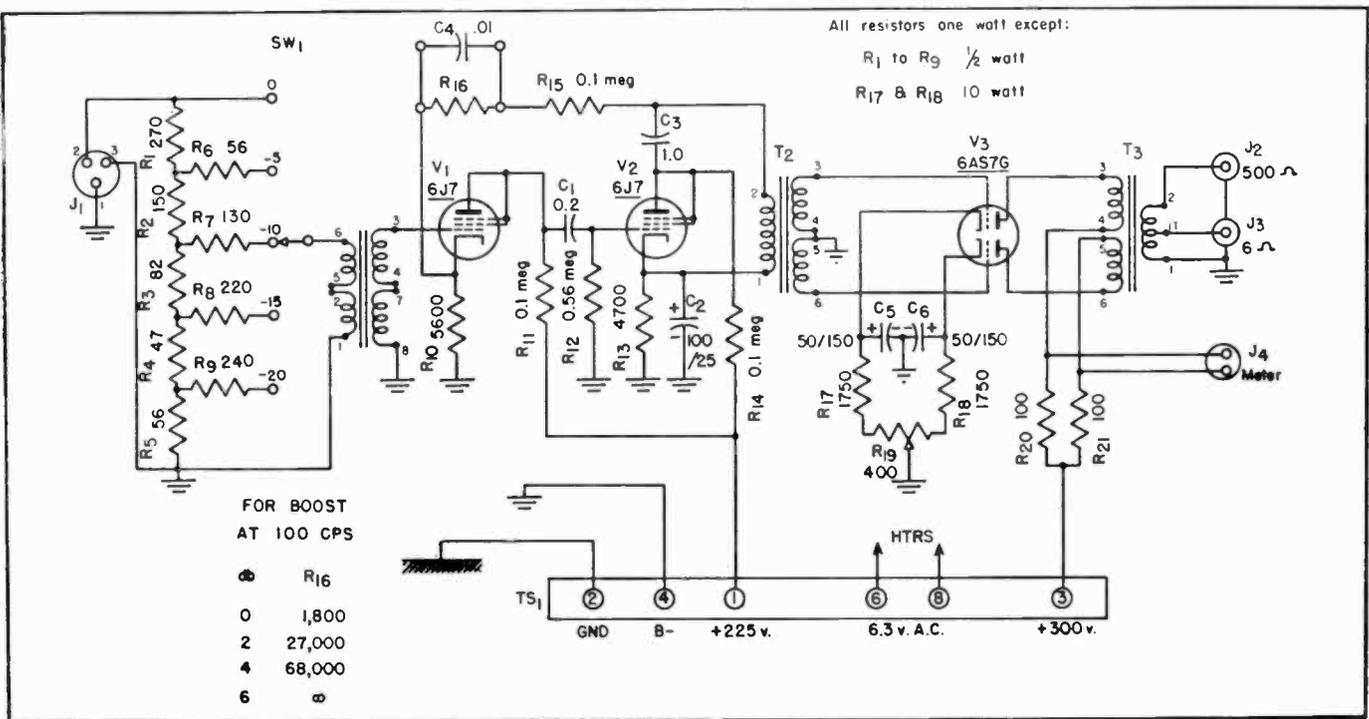


Fig. 24. Schematic of the main amplifier, closely following the original 6AS7G amplifier described back on page 5.

ance coupled to a second 6J7, also triode connected. The choice of 6J7's was dictated by two conditions: it permitted mounting the selected input transformer in its normal position, with the terminal board on top and with a short lead to the grid cap, thus keeping the grid connection well removed from the heater leads; and according to the amplifier tables in the RCA Tube Handbook, the triode-connected 6J7 is capable of a higher output voltage than the 6J5, which would appear as a logical choice. The 6J7 also appears to be better than the 6N7 previously employed, both with

respect to distortion and for a lower hum level.

Inverse feedback is employed on the first two stages, since the requirements from the second 6J7 are rather severe, and since the output tube has low gain and a low plate impedance and thus does not actually need feedback. Applying feedback around two transformers is also likely to introduce troubles which are difficult to eliminate, unless the transformer is designed specifically for such use. Parallel feed is used to keep d.c. out of the transformer primary, with the capacitor  $C_3$  isolating the plate voltage from the transformer. Normally it is considered more desirable to place the capacitor between the low end of the primary and the cathode, as shown at (A) of Fig. 27, since this arrangement constitutes a bridge which balances out hum components in the plate supply. Referring to (B),  $R_p$  represents the plate resistance of the tube,  $R_L$  the shunt-feed resistor,  $C_c$  the coupling capacitor, and  $C_r$  a decoupling capacitor. The hum component of the plate supply appears between A and C, and if  $C_c$  and  $C_r$  are chosen so their reactances are proportional to  $R_p$  and  $R_L$  respectively, the bridge will be balanced, and no hum voltage from the plate supply will appear between points B and D, to which the primary of the transformer is connected.

isolate the plate voltage from the feedback circuit. Feedback is applied to the cathode of the first stage through the network consisting of the resistor,  $R_{15}$  in series with the capacitor  $C_4$  which is shunted  $R_{16}$ . This connection provides a quick means for varying the low-frequency response readily. The entire amplifier is flat with 1,800 ohms across  $C_4$ . For boosts of 2, 4, or 6 db at 100 cps, the value of  $R_{16}$  is 27,000, 68,000, and  $\infty$ , respectively. This is not intended as a variable tone control, but is a fixed adjustment which is set for a given speaker system. In general, this method of varying low-frequency response is not desirable, since it reduces the feedback at the low frequencies where it is most useful. However, the amplifier is in use with the 1,800-

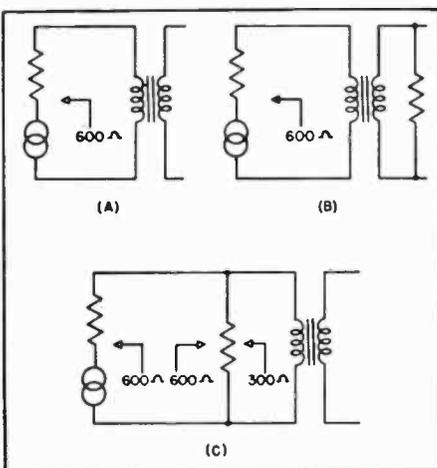


Fig. 25. Differing methods of operating transformers: (A) across an unterminated generator of 600 ohms impedance; (B) with a termination across the secondary; and (C) across a terminated line. Note that the transformer "sees" an impedance equal to one-half the nominal line impedance.

However, the plate voltage for this stage is practically humless, since it comes from a regulated supply, so this connection was not considered necessary. Therefore, the coupling capacitor  $C_3$  is top-connected, and also serves to

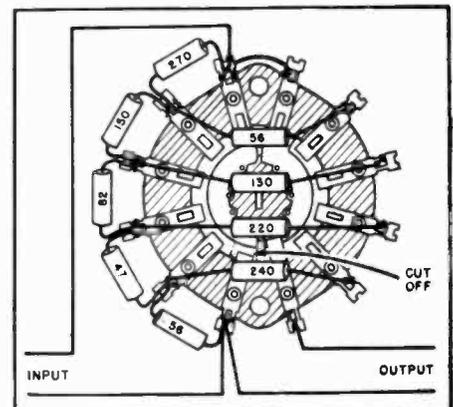


Fig. 26. Method of constructing input attenuator on a single switch deck. The contact fingers on one half are bent back or cut off, and the lugs serve only to mount the resistors.

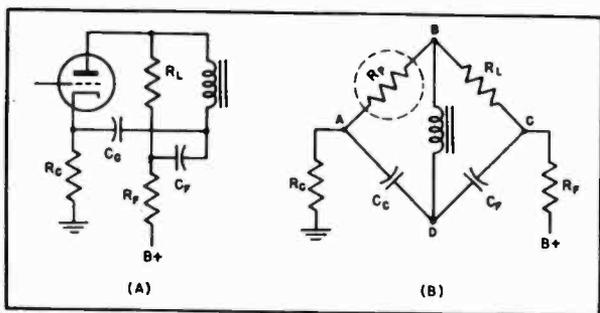


Fig.27. Bottom-connected isolating capacitor (A), and rearrangement of elements in bridge form (B) to show hum balancing effect.

ohm resistor across  $C_4$ , so this problem is not encountered in practice. Even with 6 db of boost, the effect is not particularly important with program material, since there is a falling off in peak power requirements below the most probable peak at 350 cps<sup>2</sup>. It would, however, show up on constant-frequency measurement methods.

### Output Stage

The output stage is conventional for the 6AS7G, using separate cathode resistors for each of the triode sections, and by-passing them heavily. The potentiometer  $R_{19}$  serves as a balance for the plate currents. The output transformer has a split primary, and a 100-ohm resistor is connected in series with each. A 150-0-150 microammeter is connected to the two junction points. When the plate currents in the two halves of the primary are balanced, the drops across the two 100-ohm resistors are equal, and the meter indicates a balance with no current through it. For initial adjustments of  $R_{19}$ , it is suggested that a resistor be used in series with the meter, to avoid possible overloading. The connections for the meter appear on a receptacle, as do the 6 and 500-ohm outputs of the amplifier.

<sup>2</sup>"Powers Produced by Musical Instruments", John C. Steinberg, 9-23, Electrical Engineers Handbook, Pender & McIlwain, (Wiley)

The overall gain of this amplifier is 50 db, with the input attenuator providing additional gain settings of 45, 40, 35, and 30 db. Power output at one per cent harmonic distortion is 6.2 watts, and the response is flat within  $\pm 1$  db from 24 to 17,000 cps. The output stage works with a plate supply of 300 volts, and the bias is 87 volts. The first two stages are fed from a regulated supply at 225 volts, and the filaments are heated from a 6.3-volt winding on the power transformer.

### The Power Supply

Aside from the relay system and the d-c filament circuits, the power supply is conventional. Surplus parts were used when their characteristics were suitable for the purpose, and no similar units are available from jobber stocks. However, equivalent voltage and current ratings may be obtained by using an additional filament transformer with a heavy-duty power transformer. Referring to the schematic of Fig.28, it is seen that the a-c line first passes through a fuse and the main power switch  $Sw_1$  thence to the contacts of the on relay,  $E_1$ , which are paralleled by another switch,  $Sw_2$ , which permits the use of the unit without relay operation if desired.

The relay system, described in Part I of this series, consists of a 100-ma selenium rectifier and an RC filter sys-

tem. Whenever any of the station selector relays is actuated by depressing its corresponding push button, the current flows through the relay circuits, first passing through  $E_1$ , and the normally closed contacts of  $E_2$ . The selector relays have holding contacts, and as long as any one is actuated,  $E_1$  remains closed, applying power to the transformer primary. Depressing the off button energizes  $E_2$ , which operates and breaks the holding circuit;  $E_1$  releases, and disconnects power from the transformer primary.

The power transformer has a total of six filament windings, of 6.3 and 5 volts. Two of the 5-volt windings and one of the 6.3-volt windings are connected in series, and feed a 1-amp bridge-connected selenium rectifier, with  $R_5$  being used to adjust the d-c output voltage. The filter consists of  $L_1$ ,  $C_3$ , and  $C_4$ , and the output is adjustable from 10 to 14 volts at 1-amp drain.

The high-voltage supply uses a 5V4G rectifier with a two-section choke input filter, furnishing 300 volts to the 6AS7G. The regulating circuit consists of a 6Y6G as the series tube, with a 6SL7 as the control amplifier and a 5651 for the voltage reference. The circuit is that recommended by RCA, and the output voltage is adjustable from 180 to 240 volts. It is normally set at 225 volts, and the regulation holds the output constant within two volts over a current range from 10 to 75 ma.

The metering circuit for balancing the 6AS7G output stage is fed to the power supply, and with another circuit consisting of ground and a series resistor from the regulated B+ bus is fed to a four-terminal receptacle for the meter. Either of these two circuits is selected by a 3PDT switch, not shown on the schematic, with the third position connecting to a jack on the front panel. This jack is used when aligning the FM tuner by inserting a patch cord between it and the jack on the tuner chassis. The discriminator balance is read directly on the meter, and the job of alignment is made as simple and accurate as possible.

Power for the control amplifier appears on an octal socket, and that for the main amplifier appears on a terminal block, mounted under the chassis. The FM tuner requires plate supply and 115-volts a.c. for the primary of the filament transformer, and this fed through a 5-p socket. The AM tuner obtains its power through the FM tuner, as indicated on the block schematic, Fig. 29,

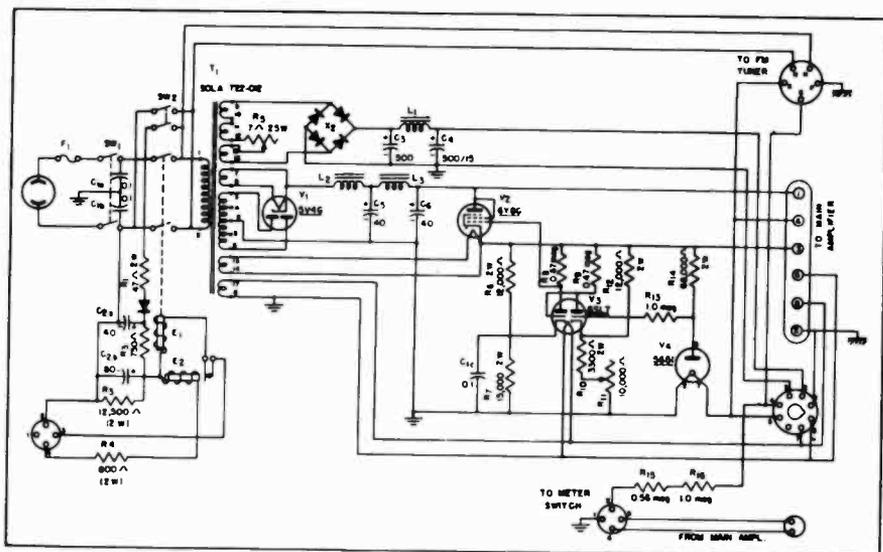


Fig. 28. Schematic of the power supply unit.

### Construction

The main amplifier is built on a 5"x10" chassis, and the power supply is on a 12"x10" chassis. As shown in the photos, Fig. 23 and Fig. 30, these two chassis are bolted together, with short leads connecting their terminal blocks. These two sizes were chosen so that they could be used together as a single unit, 10"x17", and thus fitting on a standard relay rack; or so they can be used separately if desired. The use of terminal blocks for each section permits interconnection with short leads when the chassis are used together, or with a cable as long as necessary when they are used apart from each other. This construction provides an effective stiffener for the large chassis dimensions, as well as some shielding between the two sections.

The transformers in the main amplifier are all Western Electric types—a 247J for the input, a 264C for the push-pull input, and a 166B for the output. The latter is designed for push-pull 300A tubes, and has an impedance ratio of 4130 to 500 and 6 ohms. It is normally used with a 16-ohm load circuit, and reflects a somewhat higher load upon the output tubes than its nominal value. However, since the load desired for the 6AS7G is of the order of 4,000 ohms, the 6-ohm winding feeds a 16-ohm speaker and the 500-ohm winding feeds a remote speaker load of 1,300 ohms, thus reflecting approximately 4,000 ohms to the tubes. The interstage transformer has an impedance ratio of 18,000:100,000, and is designed for shunt feed. Suitable jobber types of transformers are listed in Table I.

The power transformer has seven secondary windings: 400-0-400 at 250

Fig. 29 Block schematic of the entire system described in this series.

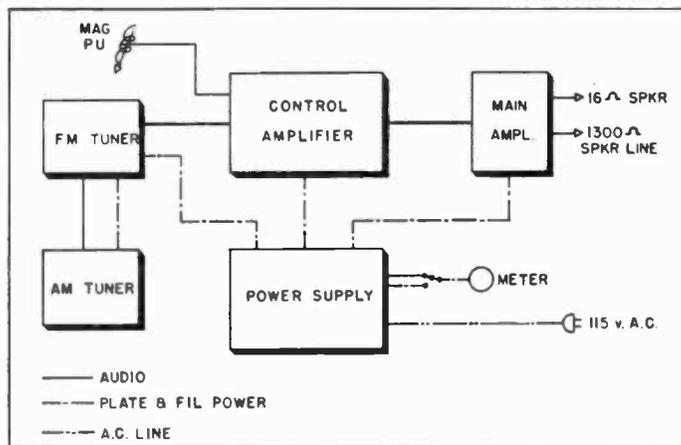


TABLE I

POSSIBLE TRANSFORMER SUBSTITUTIONS				
	Input	Interstage	Output (V.C.)	Output (Line)
Acro			T0-240	
ADC	215A	215C		315C (& v.c.)
Freed			F-1951	F-1950
Peerless	K-251-Q	G-212-Q	S-240-Q	
Triad	HS-1	HS-25	HS-84	HS-85
UTC	LS-10	LS-21	LS-57	LS-55 (& v.c.)

ma; three of 5 volts, 3 amps; and three of 6.3 volts at 5, 3, and 1 amps respectively. The two chokes are 10 henry, 200 ma units, and the d-c filament supply choke has an inductance of 22 mh at 1 amp. The total d-c filament drain for the control amplifier is 0.9 amps. The a-c drain for the main amplifier is 3.1 amps, and this is supplied by the 5-amp winding, which also feeds the 6SL7 in the power supply and the 6SJ7 in the control amplifier, making a total of 3.7 amps. The 3-amp winding

is used for the heater of the 6Y6G in the regulator circuit. Although all the transformers and chokes in the power supply are surplus items, they may be replaced by standard items, using a multiple-filament transformer in addition to a combined plate and filament transformer.

### Operating Characteristics

The quality of reproduction from this entire equipment is considered somewhat above average. Hum level is approximately -41 dbm—note that this is not 41 db below the maximum output as most commercial amplifiers are rated, but actually 79 db below the 6-watt maximum at one per cent distortion, about 20 db quieter than the average. This includes the control amplifier in the measurement, and was made with the input attenuator on the main amplifier set at 40 db and the volume control at maximum.

The most outstanding single feature of the equipment is the completely compensated volume control, which provides the correct aural balance to suit the output level. Most of the quality of reproduction is credited to the use of high grade components in a simple straightforward design without any shortcuts or tricks. The results appear to prove the advantages of this form of construction, and while considerably more expensive initially, the continued performance of equipment built in the professional manner will ensure lasting satisfaction.

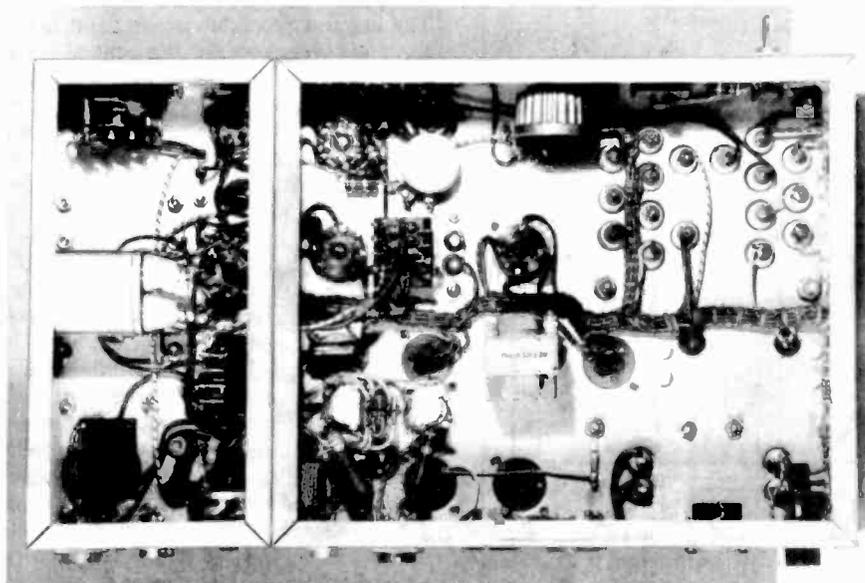


Fig. 30. Bottom view of two units bolted together, showing cabled wiring and use of resistor mounting strips.

# High-Fidelity Volume Expander

NORMAN C. PICKERING

Presenting for the first time a volume expander in which all forms of distortion are reduced to negligible proportions.

**A**LMOST ALL devoted listeners to recorded music, after they have gotten their playback systems to the point where the sound is pleasant to hear, feel that the loudness range on records is much too limited. To the listener who is concerned primarily with the aesthetics of the matter, the compromise can be made in either of two ways: the volume level can be set high enough to achieve satisfactory fortissimos—at which setting the pianissimos are too loud and too noisy; or the volume can be adjusted for maximum tolerable surface noise—in which case frustration sets in when the crescendo level off at a feeble mezzo-forte.

A considerable improvement in loudness range can be made by the intelligent use of a well-designed volume expander. The great difficulty is, of course, in the design of the expander itself, which is far from being a simple job. There have been

many circuits offered in the past, all of which have at least one of the following defects:

1. The distortion is excessive
2. The expansion is not smooth
3. The rate of action is too slow
4. There is no positive limit to the amount of expansion.

The development of the present circuit was instigated by David Hall, the eminent authority on records and sound reproduction. The equipment had to pass the most stringent tests from the standpoints of both technical and musical listeners. This it seems to have done very satisfactorily, and it is believed that this is the most musically satisfying volume expander yet developed.

In order to keep the distortion to substantially zero level, and at the same time provide a definite limit to the possible increase in gain, it was decided to attack

the problem by decreasing the gain of a high-quality low-distortion amplifier stage in a controlled manner, by a method which, in itself, would not introduce distortion.

The use of variable- $\mu$  tubes was abandoned at the outset, as soon as inherent intermodulation distortion of the order of 15% was discovered. From the standpoint of low distortion there is still nothing better than a triode of low plate resistance.

The gain of an amplifier tube is given by the following equation\*:

$$\text{Voltage across load} = \frac{-\mu e_s Z_1}{R_p + Z_1}$$

where  $\mu$  = amplification factor  
 $e_s$  = input signal  
 $Z_1$  = load impedance  
 $R_p$  = plate resistance.

Since we have already ruled out the advisability of a variable- $\mu$  tube, we must find another way to vary the voltage across the load for a constant input signal,  $e_s$ . It is apparent that variations in the load impedance  $Z_1$  will change the output, especially if  $R_p$  can be varied in the opposite direction at the same time.

Referring to the circuit diagram of Fig. 1, it can be seen that the controlled amplifier is essentially a single stage of push-pull amplification. In addition to the plates of the amplifier tubes (6J5) the output transformer supplies current to the plates of the control tubes (6P5). The amount of plate current, and hence the plate resistance, of the control tubes can be determined easily by the instantaneous grid potential. At the same time, the additional plate current flowing through the common cathode resistor causes the plate resistance of the amplifier tubes to increase because of the increase in grid bias.

The actual operation is shown in the curves of Fig. 2. The output transformer is loaded with its nominal load of 500 ohms, which reflects a normal load of 15,000 ohms across each amplifier tube (control tubes cut off). This is indicated by the load line A-A across the curves. The quiescent operating point is shown by P in the diagram. The heavy line, of which P is the center, shows the region of actual operation. When a plate current of approximately 10 ma per tube is allowed to flow through the control tubes,

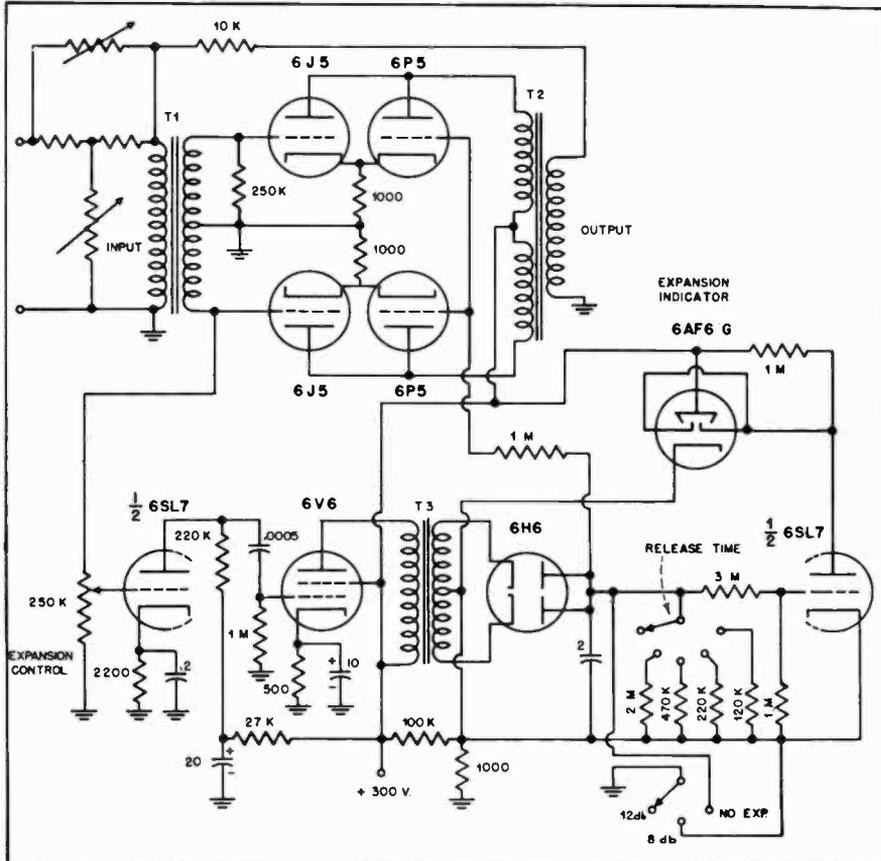


Fig. 1. Schematic diagram of the high-fidelity volume expander.

\*Terman—Radio Engineering, p 173.

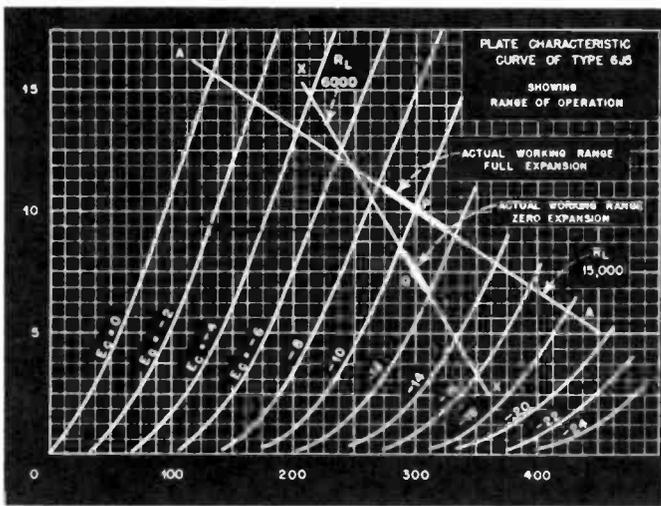


Fig. 2. Curves showing operating range of expander tubes.

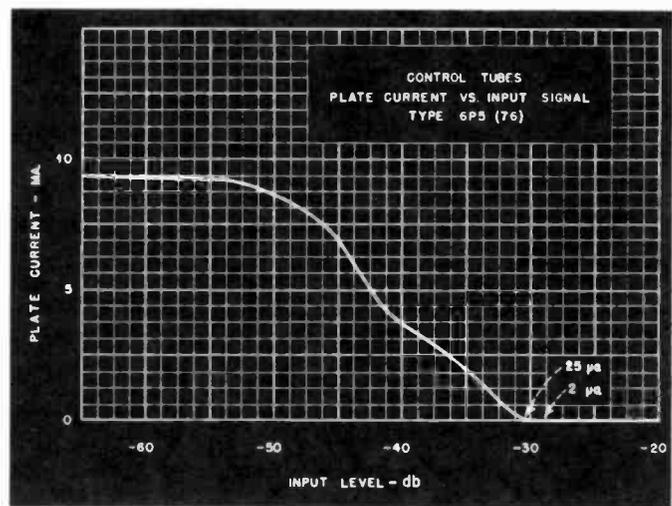


Fig. 3. Plate current vs. input signal curves of control tubes.

their plate resistance drops to 10,000 ohms. This causes the effective load to be:

$$\frac{R_p Z_l}{R_p + Z_l} = \frac{10,000 \times 15,000}{10,000 + 15,000} = 6000 \text{ ohms}$$

At the same time the effective increase in cathode current causes the operating point to drop to point Q. This condition of operation is indicated by the load line X-X passing through point Q.

The net result of the two effects is to reduce the amplification as the plate current of the control tubes is increased. The only thing which limits the amount of gain reduction possible is the approach of point Q to the non-linear region of the plate characteristic, with resulting increase of intermodulation distortion. It is interesting to note that the limitation in permissible distortion is on the low-level or unexpanded condition of operation. There is no possibility of running into distortion when the amplifier is wide open.

With the control tubes cut off, the gain of the amplifier stage is approximately 20 db. This will be the maximum possible gain, regardless of the amount of expansion used. A large number of listening tests indicated that, if smoothly applied, a maximum of 12 db expansion is permissible on well-monitored records. On records where there are noticeable "steps"

in the volume changes due to too-rapid changes of gain during recording, 8 db is the maximum tolerable amount of expansion. All of these results were based on the premise that the listener must not be able to detect the actual operation of the volume expander.

With the foregoing information as a basis, the type of control tube was chosen, and the operating point was selected to give a loss in gain of approximately 8 db when the control grids were at zero potential relative to ground. It was then found that a grid voltage of -21 volts was sufficient to reduce the plate current to 2 microamperes. A positive potential (relative to ground) of 3 volts, applied to the control grids, decreased the gain by an additional 4 db. It is necessary merely to switch this positive bias on and off in order to change the expansion range from 8 db to 12 db. An additional switch position is provided to ground the control grids, making the unit a fixed-gain amplifier with zero expansion.

#### Time Constants

The next problem is that of the attack and release time constants of the system. Again it was necessary to resort to listening tests to decide on the optimum conditions. It was quickly decided that an

instantaneous attack was necessary to provide unnoticeable operation. In other words, a loud sustained tone reaches its full level at once, and doesn't build up over a noticeable time interval, as so often happens with volume expanders. On the other hand, it is necessary to delay the release time so that the expander will not follow individual cycles at low frequencies. Furthermore, the release time adds to the normal decay of the sound, and, if too rapid, will effectively decrease the amount of reverberation. For this reason it is necessary to have at least 2 seconds delay for normal program material. In the laboratory model of the expander, values of 0.5, 1, 2, 5 and 10 seconds were provided. It was found that 5 seconds gave good results on almost every type of record.

Inasmuch as it is necessary to charge the 2-mfd capacitor in a matter of 1 millisecond or so, it is apparent that a considerable amount of power is required. A 6V6 is used to supply power to the rectifier. The rectifier, type 6H6, is momentarily called upon to supply several times its peak rated current of 10 ma, but because of the extremely short duty cycle, this has no adverse effect on the tube life.

It was necessary to do a considerable amount of testing to determine the opti-

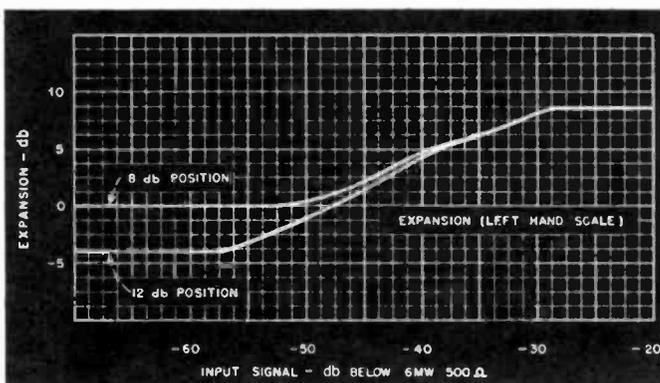


Fig. 4. Input signal level vs. expansion curves.

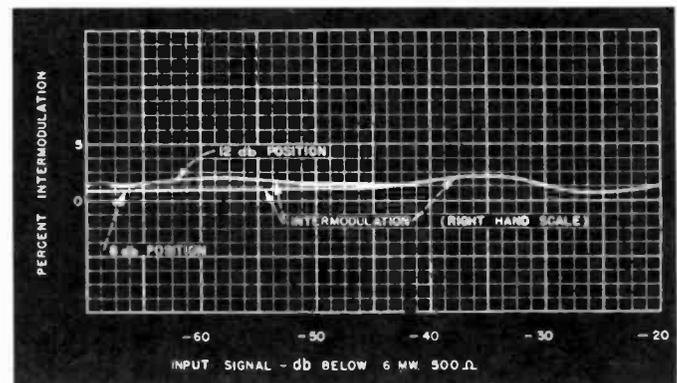
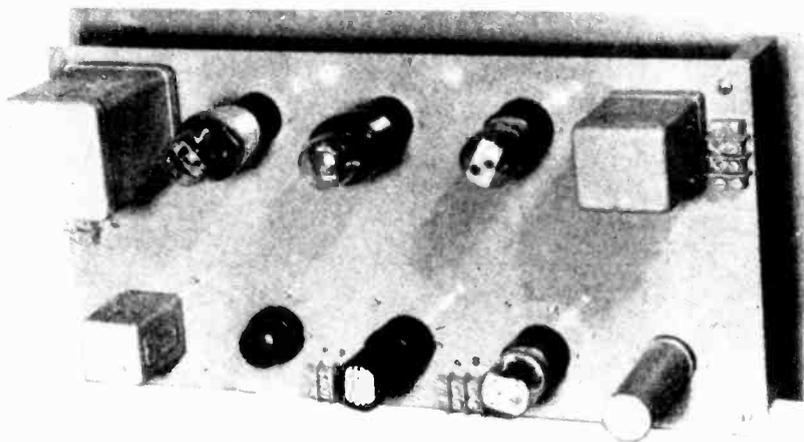


Fig. 5. Intermodulation distortion at various input signal levels.



Rear view of high-fidelity volume expander.

imum frequency response of the amplifier used to supply power to the diode. If a wide-band—30-10,000 cps—frequency range is used, it is apparent that the control voltage will be influenced by the strongest signal currents within that frequency range. Furthermore, low-frequency rumble and thumps, and high-frequency surface noise and clicks, will produce false control and consequent gain increase. It is best, therefore, when using a single band of frequencies from which to derive the control signal, to limit the response to from 500 to 3000 cycles. The effective loudness is determined by this band of frequencies, anyhow, so it is logical to use the same region for the controlling voltage.

An interesting effect was achieved by splitting the control amplifier into two channels, each supplying a separate diode. The outputs of the diodes were connected in series, and the individual amplifiers were arranged so that they would overload when supplying half the necessary control voltage. One amplifier passed the frequency range from 100 to 500 cycles; the other from 600 to 3000 cycles. Either amplifier alone could only

produce half the total expansion, regardless of the energy in the pass band. It required energy in both bands to produce full expansion. This condition prevails in full orchestra, organ, or band music, at which time full expansion is required. This system prevented "blasting" when a single instrument or voice momentarily overbalanced the full ensemble. The results were very good, but on most records the improvement over the single-channel system of restricted frequency range did not justify the circuit complexity.

#### Expansion Indicator

A 6AF6 electron-ray tube is used as an indicator of the amount of expansion. It has a considerable advantage over a pointer-type meter, because of its freedom from dynamic error. It is adjusted so that the eye is just closed when the 6P5 control tubes are cut off.

The last important problem is the linearity of expansion versus input signal. It is very important not to have any "steps" in the expansion control. The plate-current variations with input signal, with the control voltage derived from the previously described rectifier system, are

shown in Fig. 3. The change in plate resistance is not exactly the type of curve needed to produce a linear expansion. This condition was greatly improved by making the total range of expansion about 6 db more than desired, and reducing the over-all gain by means of inverse feedback. Since the gain reduction is a function of the amplifier gain, it is apparent that the amount of feedback will vary as the amplifier gain is varied. The gain will be reduced more at full amplification than at low levels. This has the effect of straightening out the expansion curve and reducing its slope, thereby accomplishing the desired end. This is shown in Fig. 4. The feedback also contributes somewhat to the over-all low distortion obtained with this device. The curves of intermodulation distortion versus input signal level are shown in Fig. 5. The harmonic distortion is barely measurable.

Fig. 6 shows the main amplifier circuit arranged to be inserted in a high-impedance amplifier. The action is identical with that of the low-impedance unit.

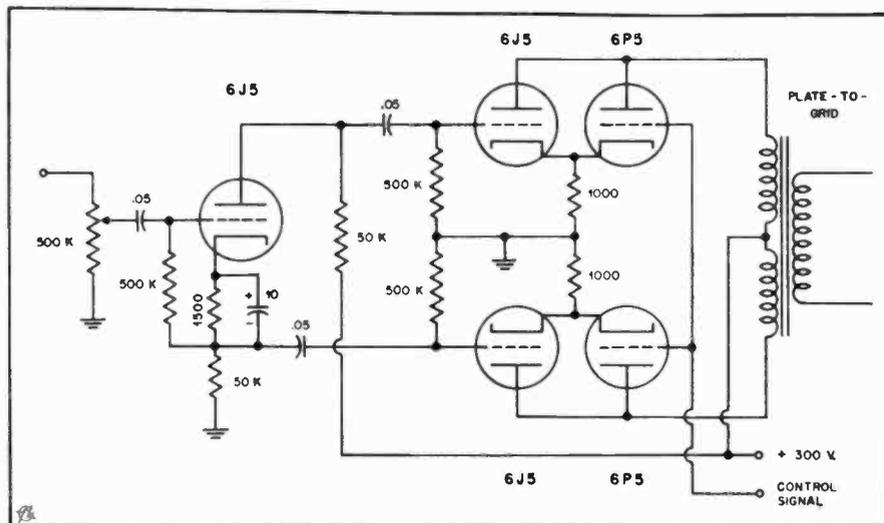
It will be noticed that this circuit can be used as a remotely operated gain control. It is merely necessary to substitute for the control signal a 22½-volt battery and potentiometer.

A little care must be exercised in using the volume expander. It should never be used on program material where the source of sound is inherently incapable of a volume range of more than 20 to 30 db. This applies to solo instruments (other than piano and organ), solo voices, string quartets, and so on. On orchestral, choral, and organ music it can be used on almost any record with excellent effect. The actual manner in which the original recording was controlled determines whether 8 db or 12 db of expansion can be used. Paradoxically enough, the wider the volume range on the original recording, the more expansion can be tolerated.

In Fig. 4, it will be noticed that in the 12 db position the input signal necessary to cover the entire range of expansion is about 29 db. This is about the volume range of a good modern recording. When playing such a recording it is best to set the expansion control so that the eye of the indicator tube fully closes on average peak levels. The expansion will then be completely off on very low-level signals. On records of more restricted range, it is best to set the expansion control so that surface noise just does not operate the indicator. This will then give the maximum increase of volume range.

It is good practice to install the expander with a gain control following the unit. The expansion control can then be left in the full-on position, and the input gain control used to adjust for input signal peaks. The output gain control then controls loudspeaker volume, and all of the output peaks will be at the same level, regardless of the actual level on the recording.

Fig. 6. Modifications needed for high-impedance input circuit. Input signal should be about 2 volts rms, maximum.



# Audio Amateur

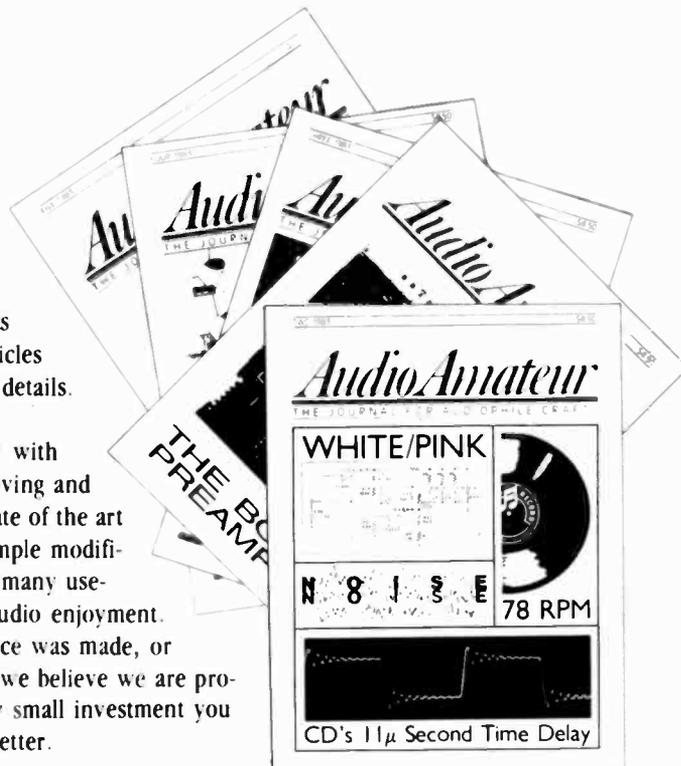
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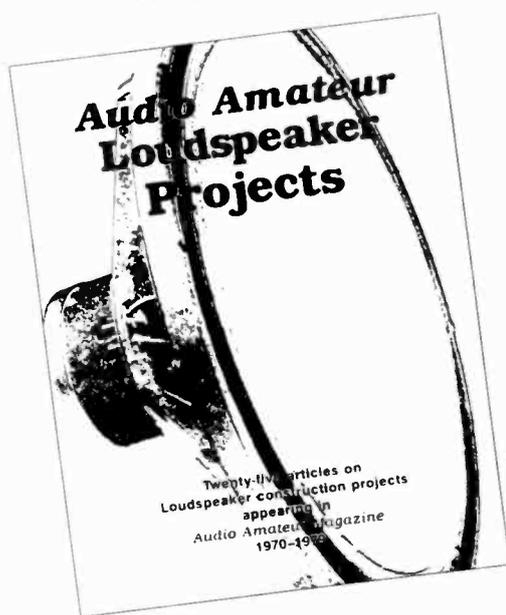
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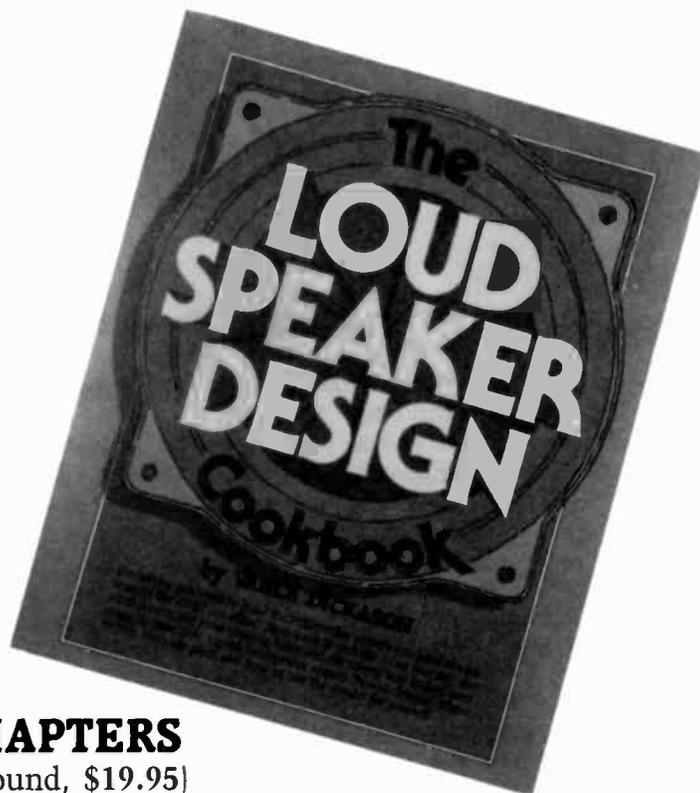
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