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(1944)

DIRECT-COUPLED FM-AM

AMPLIFIER MANUAL

A. C. SHANEY

DIRECT-COUPLED FM-AM

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BY

A. C. SHANEY

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Printed in the United States of America

FOREWORD

THE AUTHOR has compiled this series of articles to meet the urgent need for information on push-pull direct-coupled amplifiers, based upon practical laboratory experience, rather than pedagogical theories.

The material was assembled primarily for technicians and laymen, who are no longer able to obtain, or refer to, the original published material.

It is hoped that this booklet will disperse the shroud of mystery surrounding the performance of the modern direct-coupled amplifier which has found universal acceptance in leading commercial, industrial, and governmental research laboratories, as well as in professional recording studios, broadcast stations, and in the homes of many discriminating music lovers.

The author is profound in his belief that it will only be a question of time before this new type of direct-coupled amplifier will over-shadow in popularity, the already waning transformer-coupled circuits as well as conventional resistance-coupled amplifiers

Grateful acknowledgment is made for the cooperation of Frank Kosinski who laboratory checked all designs, and who also materially contributed to the perfection of the equalized self-balancing inverter circuit.

Acknowledgment is also made to those few individuals, who have throughout the period of development of these amplifiers, given of their time, thought and energy. To these few, particular thanks are due: Paul Samoluk and Otto Kriz.

The author is also indebted to R. D. Washburne, Managing Editor, and N. H. Lessem, Associate Editor, of Radio Craft Magazine, at whose insistance, the original direct-coupled laboratory amplifier was perfected for universal application.

In addition, acknowledgment is also made to Radcraft Publications, for permission to reprint material, by the author, from Radio Craft.

New York, N. Y.
March, 1941

A. C. SHANEY

W. Morris
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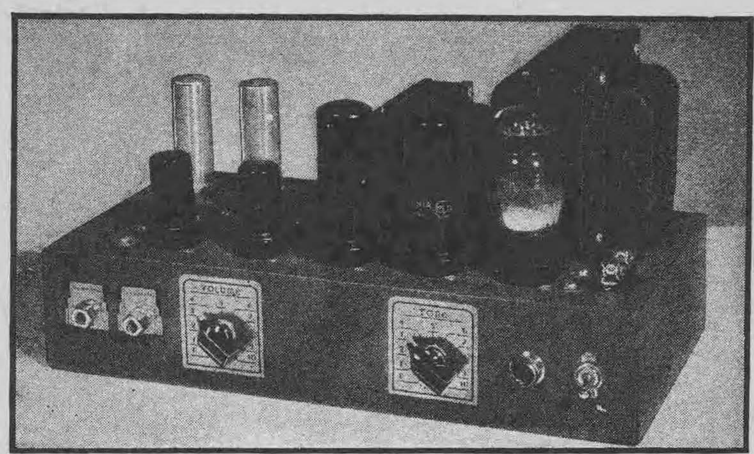
CONTENTS

*Radio Craft Magazine.



The 10-W. All-Push-Pull Direct-Coupled Amplifier. Note its simplicity, and compactness. A view underneath the chassis will amaze you with its scarcity of components.

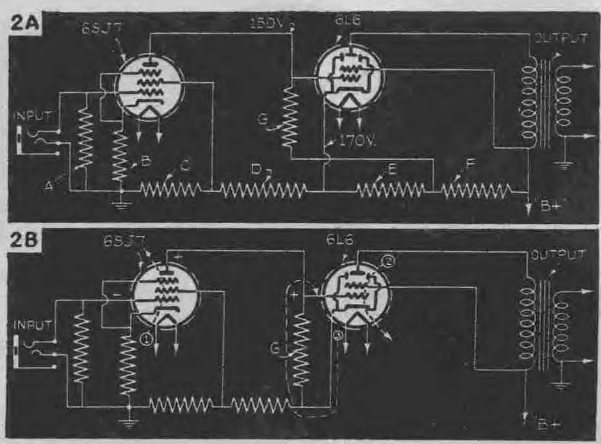


Fig. 2. In (A) is shown the basic diagram of a direct-coupled circuit; and in (B), old method for obtaining grid bias for output tube.

How To Design
ALL-PUSH-PULL DIRECT-COUPLED

An engineer takes you into the laboratory to show you how ideal amplifier per-tubes give new impetus to the direct-coupled amplifier *made famous by Messrs.*

> **Dear Editor:**
>
> The results of our laboratory measurements made on the All-Push-Pull Direct-Coupled Amplifier, which is described in the accompanying article, have amazed me and my associated engineers.
>
> You will note that I have intentionally failed to supply you with a frequency response curve of the essential amplifier, inasmuch as I want to avoid the slightest possibility of being accused of placing a straight edge upon a graph sheet and drawing a straight line.
>
> *(Continued on page 30)*

WITH the present-day high development of engineering skill and manufacturing technique, it is possible to build into an amplifier an exaggerated characteristic along almost any line that one can name. Such an accomplishment, as a matter of fact, is not merely so difficult, as the designing of a product in which there is a well-balanced relation between the various aspects of performance, dependability, power output, weight, economy, and simplicity.

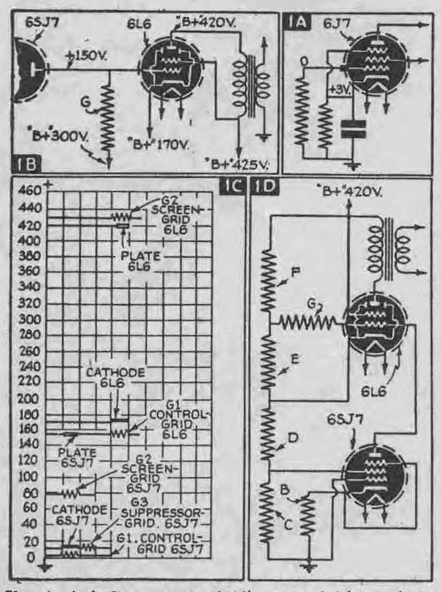


Fig. 1. (A) The actual distribution of bias voltage in a conventional amplifier stage; (B) voltage distribution system in a fundamental direct-coupled circuit; (C) voltage distribution, graphically illustrated; and (D) direct-coupled amplifier circuit graphically illustrated.

WHY THE "DIRECT-COUPLED" AMPLIFIER?

If the ideal attributes of a perfect amplifier were to be carefully tabulated and checked against features offered by various amplifier circuits, it would be found that a direct-coupled amplifier will fit all of the conditions and will lead, by a wide margin, the very finest resistance-coupled, impedance-coupled, or transformer-coupled units.

The advantages of Direct-Coupled Amplifiers were known for a great many years, but the inability to attain a simple and practical *direct-coupled inverter, in order to achieve push-pull output and its many attendant advantages,* offered a serious handicap to the popular use of this circuit. Although a successful Direct-Coupled Inverter Circuit was developed by the writer about 2 years ago (see March, April, and July, 1937 issues of *Radio-Craft*), it required a special output transformer, and did not follow extreme simplicity of design. *A new circuit (see Fig. 3) removes the last objection to the popular use of the ideal Direct-Coupled Amplifier for all applications; at the same time, ALL-PUSH-PULL operation is obtained.*

It is also well known, that *direct-coupling* overcomes objectionable characteristics of transformer coupling (core saturation, magnetic lag, and transformer resonance) and resistance coupling (short-circuiting of weak signals and grid blocking of strong signals). Furthermore, it is unsurpassable from a simplicity standpoint (Only 9 resistors and the usual output transformer plus filter supply are required to attain extraordinary results!).

The question may well be raised, "If this type of a circuit is so extraordinary, why is it not more popular?" The an-

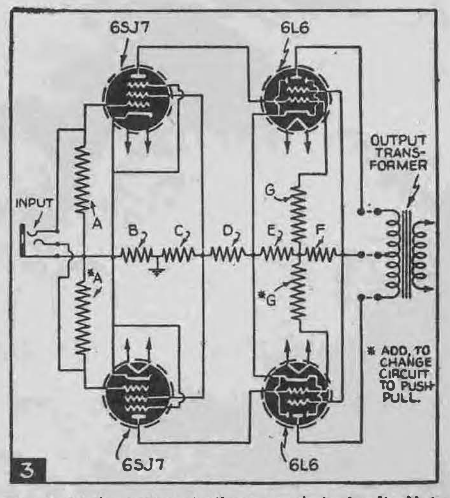


Fig. 3. Basic push-pull direct-coupled circuit. Note that only 2 extra resistors are required to change the circuit from a basic single to basic push-pull.

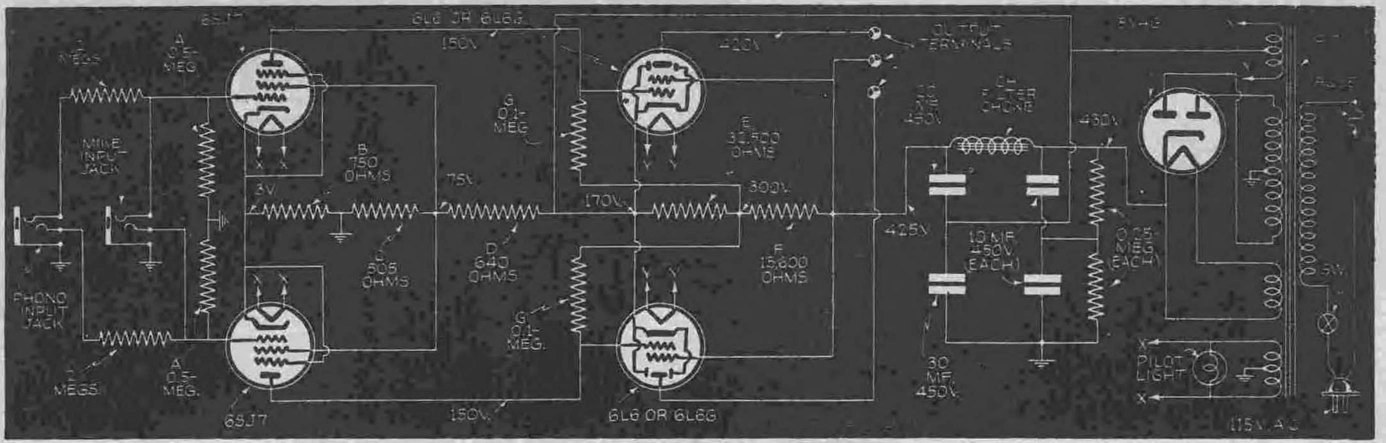


Fig. 5. Complete schematic diagram of the 5-tube, 10-W., All-Push-Pull Direct-Coupled Amplifier illustrated on the opposite page.

An Inexpensive 10-WATT AMPLIFIER

formance is attained in a practical and economical twin-channel unit. New Loftin-White; and make possible a phase inverter-less ALL-PUSH-PULL circuit.

swer is found in the usual objections offered to direct-coupled circuits, plus the fact that because of its unusual arrangements it had not received deserved attention from design engineers.

The usual objections to the Direct-Coupled Amplifier are:

(1) Tricky circuit.
(2) Instability.
(3) High voltages required.
(4) Critical hum-balancing adjustments necessary.
(5) Variations of characteristics in similar-type tubes affect voltage distribution within the amplifier.

All of these objections have now been completely eliminated. In fact, an understanding of the design principles involved (which are covered in detail in this article) in engineering an amplifier of this type, will convince the greatest skeptic, that its performance, dependability, economy, and simplicity could not be surpassed.

Applications

- High-Fidelity P.A. Amplifier
- High-Fidelity Phono Amplifier
- Laboratory Standard Amplifier for comparing microphones, speakers, pickups, etc.
- Twin-Channel Amplifier
- Constant 2-Way Communicator
- Switchless Recording and Playback
- Amplification in auditory perspective
- Reproduction of artificial echo and reverberation
- Replacing amplifier section of radio receivers, where high-fidelity performance is desired
- Amplification of musical instruments
- General replacement of obsolete amplifiers

SOLVING PROBLEMS

In the 1st place, no circuit can be more fundamentally simple than direct-coupling. Anyone who objects to tricky circuits, confesses his lack of understanding of the circuit operation. It is a well known fact, that some of the greatest feats of magic are amazingly simple, once their operating principles are understood. A study of the design principles involved, and which are given here, will prove this point about Direct-Coupled Amplifiers.

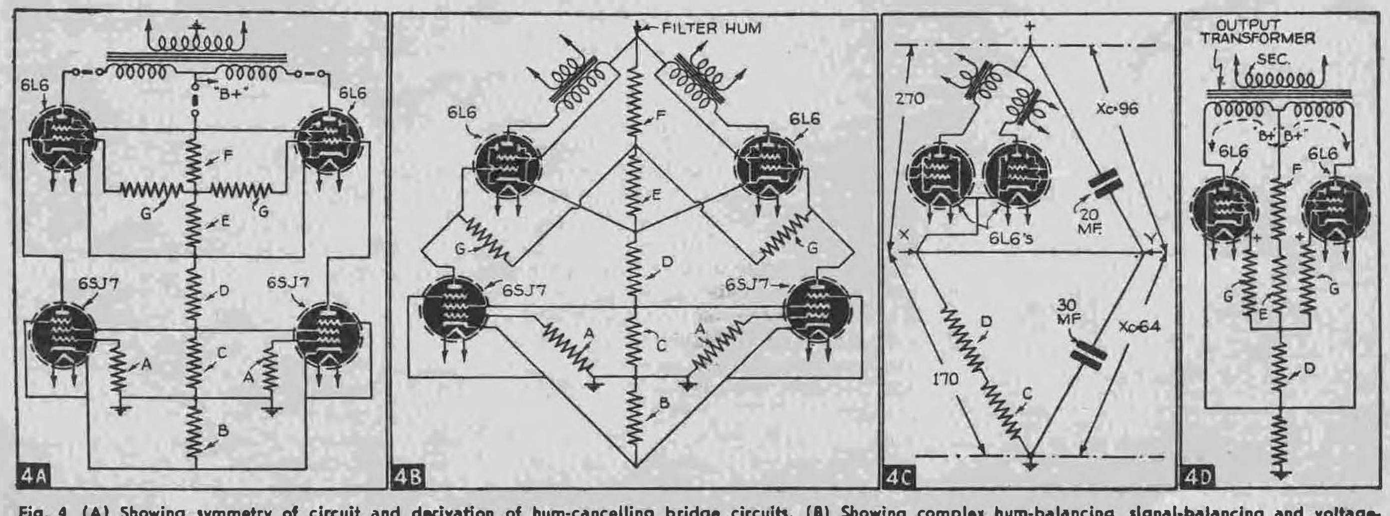


Fig. 4. (A) Showing symmetry of circuit and derivation of hum-cancelling bridge circuits. (B) Showing complex hum-balancing, signal-balancing and voltage-balancing circuits. (C) Hum-balancing filter network. (D) Showing how hum is balanced in push-pull output transformer.

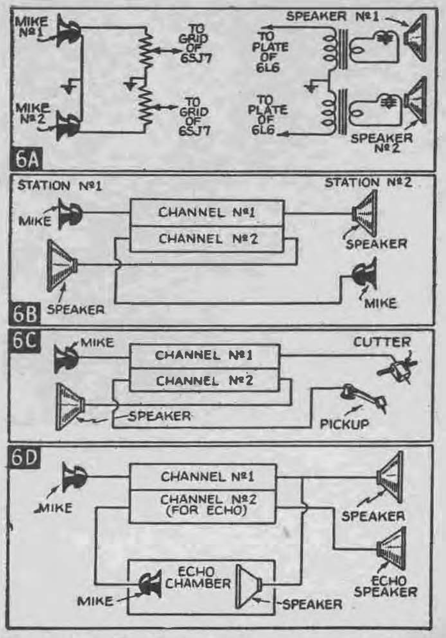


Fig. 6. (A) Circuit for auditory perspective. (B) Two-way communication system. (C) Switchless recording and playback system. (D) Artificial reverberation (echo) system.

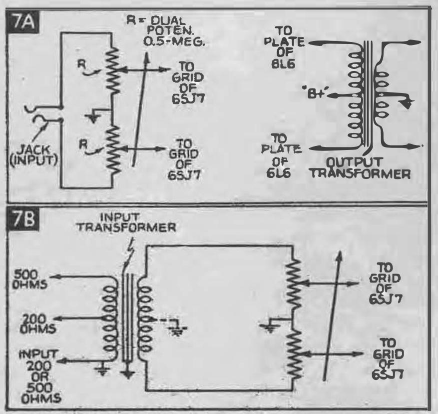


Fig. 7. (A) Push-pull input and push-pull output of a direct-coupled amplifier system. (B) Push-pull input connection.

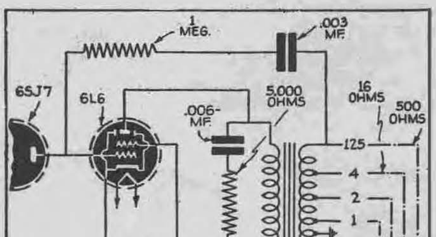


Fig. 8. Push-pull inverse feedback and compensating network for high fidelity results with economical transformer. The inverse voltage is derived from the 500-ohm taps on the secondary of the output transformer (which, incidentally, except for the power transformer is the *only* transformer in this amplifier). The center-tapped primary of this transformer connects to the 3 terminals in Fig. 5 marked OUTPUT TERMINALS.

In the 2nd place, complete stability of this circuit is attained by applying stabilized and self-balancing voltages to control elements.

In the 3rd place, the highest voltage required for this particular amplifier is approximately 430 volts. It is indeed simple, to safely handle this voltage, by using two 450-volt condensers connected in series. Such a combination will handle 900 volts, and makes available a safety factor of 100%, which exceeds, by far, safety factors employed in usual commercial amplifiers.

In the 4th place, critical hum-balancing adjustments are no longer required in a truly all-push-pull direct-coupled amplifier, inasmuch as all filter hum voltages automatically cancel themselves. This is further clarified in the design principles which follow.

In the 5th place, variations in characteristics of similar-type tubes will not detrimentally affect the distributed potentials within the amplifier, as any such unbalance between corresponding tubes will automatically tend to produce equivalent unbalance in its adjacent channel.

FUNDAMENTAL DIRECT-COUPLED CIRCUIT

To really understand the operation of a direct-coupled amplifier, it is necessary to realize that its basic principle depends upon the direct connection of a plate of an input tube to the grid of an output tube. Both of these elements have the same applied potential, but suitable corrections are applied to the output tube so that the effective bias and plate voltages are in conformance with standard ratings. To understand this condition, let us analyze a conventional bias circuit (as shown in Fig. 1A) in a 6J7 pentode tube. For a negative bias of 3 volts, a resistor is usually inserted in series with a cathode circuit, so that a positive potential is developed at the cathode. In actuality, there is a zero potential at the grid (as measured from ground), and a plus 3 volts from ground to cathode.

We say that a negative bias of 3 volts is applied to the grid. However, if an analysis of this circuit is made, the following conditions are apparent:

If we look into the tube from the cathode to the grid, we "look down 3 volts," so that the voltage distribution within the tube is of such a nature, that it may be construed as –3 volts on the grid (as compared with the cathode). If, however, we look into the tube from the grid to the cathode, it may be construed as +3 volts on the cathode (as compared with the grid). This might appear to be a tricky circuit to one who is unfamiliar with this type of biasing. The average radio man, however, takes this circuit for granted, and probably gives it no thought.

In the same way, voltages are distributed within a direct-coupled circuit (as illustrated in Fig. 1B). It will be noted that 150 volts is applied to both the plate of the input tube, as well as to the grid of the output tube, but 170 volts is applied to the cathode of the output tube so that the effective bias (looking from cathode down to the grid) is 20 volts. Although the plate potential is 420 volts (from ground) its effective potential is only (420-170) 250 volts, as measured from cathode to plate.

FOR THE FIRST TIME IN ANY RADIO PUBLICATION

INITIAL DESIGN CONSIDERATIONS

The first step in the design of the Direct-Coupled Amplifier, is to determine (a) power output required, (b) highest voltages desired in the filter supply, and (c) the necessary gain. Let us assume that our specifications call for the following conditions:

Power Output: 10 watts with less than 2% total harmonic distortion;

Filter Supply Voltage: Not to exceed 450 volts (to avoid excessively high voltages, and assure adequate safety factor of any filter design);

High Gain Input: 90 db. (to operate in conjunction with medium-level microphones);

Medium Gain Input: 70 db. (for crystal pickup or radio set).

A cursory examination of available tubes would lead us to select two 6L6's for the output stage, operating with 250 volts on the plate and screen-grid, which according to standard ratings, will develop approximately 14 watts at 2%.

Allowing for a 5-volt drop in the filter choke, a 5-volt drop in the output transformer, plus a 250-volt drop in the output tube, and an additional 20-volt bias drop to grid, there is available approximately 150 volts for the plate of the input tube. A 6SJ7, operating as a pentode with approximately 150 volts on the plate and 75 volts on the screen-grid, will satisfy our conditions for gain.

For medium gain, an additional attenuator is placed in the input circuit to drop the input signal 20 db., so that a crystal pickup can be easily accommodated.

If we list the tubes and their corresponding applied potentials, we have the essence of our Direct-Coupled Amplifier.

Electrode	*Operating Conditions For* *6SJ7*	*6L6 (or 6L6G)*
Plate Volts (Ep)	150	250
Control-Grid (Ec1) . . .	–3	–16
Screen-Grid (Ec2)	75	255*
Suppressor-Grid (Ec3)	0	——
Av. Plate Current (Ib)	1.5ma.	65ma.
Screen-grid Current (Ic2)	0.5ma.	6.5ma.

(*We anticipate a 5-volt drop in the output transformer, so that the screen-grid potential will actually be 5 volts higher than its plate. This normal condition does not affect the performance of the amplifier in any manner.)

Although we will finally develop a push-pull amplifier, the element potentials are the same as a single-ended job, in accordance with the voltages listed above.

Figure 1C shows a graphic voltage distribution of the amplifier. The ordinates are plotted at the right of the various elements which have been arranged in order of their applied potentials, in accordance with the above tabulation. Here, too, it will be graphically noted, that although approximately 150 volts are applied to the plate of the 6SJ7 (*and to the control-grid of the 6L6*), a negative bias is applied to this grid by making the cathode approximately 20 volts higher (at 170 volts from ground). All other element potentials are likewise distributed. Figure 1D shows the fundamental circuit arrangement to obtain the potential distribution plotted in Fig. 1C. Each resistor used, has been identified so as to make it easy to follow its position during the step-by-step development of the amplifier. If Fig. 1D is redrawn to conform with standard circuit design, Fig. 2A results.

It will be noted, that resistors E and F are used across the high "B+" and cathode of the 6L6 to obtain the plate potential for the 6SJ7's. This simple expedient avoids objectionable "trigger action," which was predominant in early direct-coupled amplifier designs. Inasmuch as the grid potential of the 6L6 is lower than the cathode potential, the original designers were tempted to obtain this voltage directly from the cathode, as illustrated in Fig. 2B. This circuit is greatly susceptible to "trigger action," because of the following sequence of events:

(1) When an instantaneous negative potential appears on the grid of the input tube, less plate current flows, and a smaller voltage drop takes place in the plate resistor G, so that the plate potential of the 6SJ7 tends to rise. Naturally, the grid potential of the 6L6 also rises, which in turn, decreases the effective bias of the output tube, and (2) increases its plate current, so that higher potential appears at the cathode, (3) which in turn raises the potential (through resistor G) on the output grid.

This cycle of events continues until plate current becomes excessive and the tube is thrown off its Eg—Ip curve, and maintains itself in a blocked position. This effect is popularly known as "trigger action." By employing resistors E and F (Fig. 2A), the plate potential of the input tube is independent of the plate current of the output tube.

BASIC ALL-PUSH-PULL DIRECT-COUPLED CIRCUIT

The extreme simplicity of making an amplifier push-pull *throughout*, suggests itself as an ideal and simple manner to attain push-pull direct-coupled amplification without the use of any additional expensive components. In fact, *only two additional resistors are required*, as illustrated in Fig. 3, which is composed essentially of the basic direct-coupled circuit of Fig. 2A drawn with its "stereoisomer" (or mirror image) of the single-ended circuit.

The remarkable simplicity and symmetry of this circuit can best be observed by redrawing Fig. 3 as it appears in Fig. 4A. Here we see the essential components of a high-fidelity amplifier which is composed of 9 resistors (less the output transformer). This basic circuit is capable of developing 15 watts with less than 5%, and 10 watts with less than 2% of total harmonics. While the circuit looks extremely simple from a construction and wiring standpoint, and it really is, there are a number of complex hum-balancing, signal-balancing, and voltage-balancing bridge circuits contained therein, which become apparent, only when the circuit is redrawn as shown in Fig. 4B.

Here you will find we have a complex diagram composed of 4 interlinked bridge circuits. These circuits all contribute to circuit stability independent of tube variations, and elimination of hum without the necessity of using critical hum-balancing adjustments. If a large filter hum is introduced at the apex of the circuit (marked "Filter Hum"), it will be noted that all hum potentials will be evenly distributed between each half of the bridge circuit, so that for any hum potential introduced in one-half of the circuit, there will be an identical potential (equal in phase and amplitude) in the other half of the circuit. As long as this condition exists, cancellation will take place in the output transformer. This is further clarified in Fig. 4D.

If a hum potential is applied at the junction of the resistors E-D and passed through resistor G to the respective grids of the output tubes, both output plate circuits will behave identically. If an instantaneous positive value is assumed during the hum voltage cycle, both grids go positive at the same time. More plate current flows in each of the tubes, so that a voltage drop takes place at each plate terminal. Inasmuch as the primary winding of the output transformer is in opposite direction (which is a standard procedure for all push-pull output transformers), this hum voltage cancels itself in the primary, and no voltage appears in the secondary. This phenomenon, however, does not take place when a signal voltage is applied, for under this latter condition, one grid goes positive, while the other goes negative.

If you will refer back to Fig. 4B and select any resistor in the "B+" or filter hum voltage network which may induce undesired hum into any grid circuit of the amplifier, it will be found that the same hum voltage is applied to its push-pull mate, and ultimately cancels in the output transformer.

It is for this reason, that the hum level of the amplifier can be brought down to –70 db. below maximum output, without the use of hum-balancing adjustments.

AUTOMATIC COMPENSATION FOR VARIATIONS IN TUBES

A large number (of the same type) of tubes were interchanged in the amplifier without noting any appreciable difference in performance. The apparent reason for this is evident by the additional study of Fig. 4B. Reasonable variations in plate or screen-grid currents of the 6L6 output tube cancel at their cathodes. Variations in the input 6SJ7 tube are likewise cancelled at the junction of their cathodes. It is obvious, of course, that any tube which will not operate satisfactorily in a standard amplifier should not be used in this unit.

Another existing hum-cancellation bridge is noted in Fig. 4C which is the output stage and its associated filter condensers re-drawn in a bridge circuit form. Here, it will be noted that the capacitative reactance of the 20 mf. condenser (which is approximately 96 ohms) and the 30 mf. condenser (approx. 64 ohms) is approximately proportional to the hum distribution in this portion of the filter network. The hum distribution may be considered proportional to the D.C. voltage distribution. This type of an arrangement insures against excessive hum at points X and Y, regardless of the variable effects of the mu of the 6L6 screen-grids.

CALCULATION OF RESISTOR VALUES

The design procedure necessary to calculate the values of the important 6 resistors required, makes use of an elementary application of Ohm's Law. There are only two design precautions which must be kept in mind, and these are:

(1) The voltage drop in the plate resistors G, should be made equal to the voltage drop in the plate circuit of the input tubes, i.e., 150 volts; which means that the voltage applied to the high-potential side of the G resistor should be 2 x 150=300 volts. This voltage should appear at the junction of resistors E and F.

(2) The bleeder current through resistor F should be exactly equal to the plate and screen-grid currents required by both input tubes, i.e., (1.5+0.5)X2=4 ma.

With these points in mind, it is extremely simple to calculate the values of all the resistors based on a voltage drop across, and current through, each one. The following tabulation indicates the formulas used:

RESISTOR

(B) $\dfrac{3}{0.004} \doteq 750$ ohms

(F) $\dfrac{125}{2[2(1.5+.5)]\text{ ma.}} = \dfrac{125}{0.008} = 15{,}600$ ohms

(E) $\dfrac{130}{2(1.5+.5)\text{ ma.}} = \dfrac{130}{0.004} = 32{,}500$ ohms

(D) $\dfrac{95}{2(65+6.5)+1+4\text{ ma.}} = \dfrac{95}{0.148} = 640$ ohms

(C) $\dfrac{75}{148-1} = \dfrac{75}{147} = 505$ ohms

(G) $\dfrac{150}{0.0015} = 100{,}000$ ohms

(A) May be a conventional ½-meg. grid resistor

The watts rating of each of these resistors should be between 2 and 3 times its actual watts dissipation, so as to provide a minimum safety factor of 50%.

TWIN-CHANNEL OPERATION!

Figure 5 is a completed basic circuit of the All-Push-Pull Direct-Coupled Amplifier. The output transformer, volume control and tone control have been intentionally omitted, as there is a large number of possible variations in these 3 elements, depending upon the final application of the amplifier. If a separate output transformer is used for each side of the circuit, and separate input signals are applied, *twin-channel amplification* is effected, making this unit admirably adapted for reproduction of sound in "auditory perspective."

If each one of 2 microphones is independently fed into each one of the 2 input circuits, as Fig. 6A, and 2 speakers are correctly placed in an auditorium, so as to bear the same relative positions as the microphones, amplification in auditory perspective will take place. For this application, two independent half-meg. potentiometers replace the (A) resistors. This same input and output set-up will also enable 2-way communication between any two remote points, without the use of talk-listen switches (see Fig. 6B).

The amplifier can also be used for recording and playback, or any other, similar duplex arrangement, without the necessity of switches. In the usual recording amplifier

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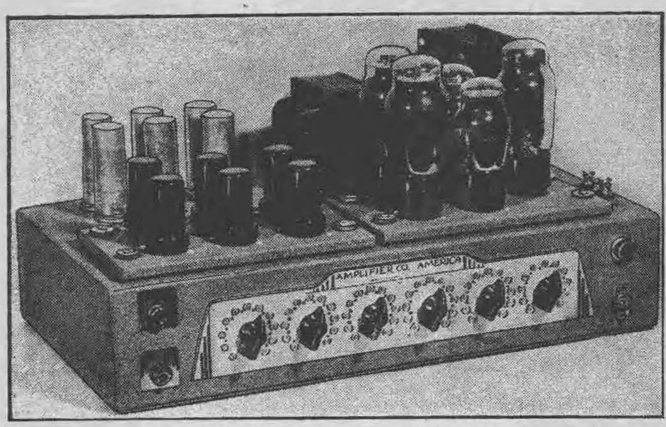


Fig. A. View of the 30-W. direct-coupled amplifier with variable expander feature included.

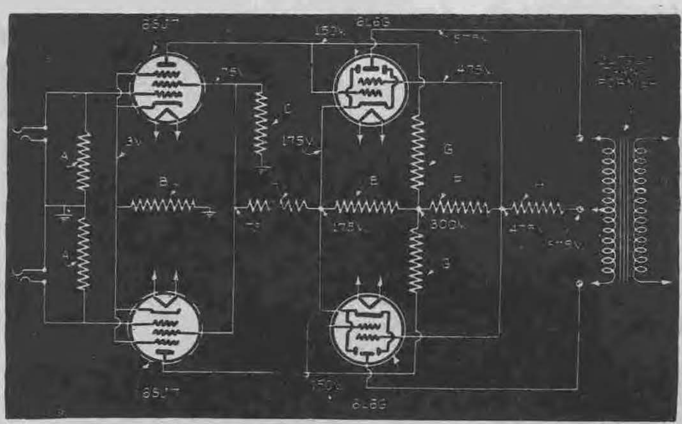


Fig. 1. Schematic diagram of the fundamental circuit of the direct-coupled 30-W. amplifier.

How to Design

ALL-PUSH-PULL DIRECT-COUPLED

Here's basic design data on a versatile power amplifier which delivers Features may be added to suit individual preferences. A new scratch

THE enthusiasm which greeted the writer's all-push-pull and all-direct-coupled 10-watt amplifier (completely described in the July 1939 issue of *Radio-Craft)* proves beyond any question of doubt that a truly flat amplifier has a universal appeal.

The opinion expressed by many readers definitely pointed towards the desirability of developing an amplifier with a number of additional features including:

(1) High power output.
(2) Built-in push-pull expander.
(3) Bass-accentuating attenuation control.
(4) High-frequency accentuating and attenuation control.
(5) High overall gain (approx. 130 db.).
(6) Automatic volume control.
(7) First-stage inverter for accommodating single-ended signals.
(8) A non-frequency-discriminating scratch suppressor.
(9) Remote volume control.

THE IDEAL AMPLIFIER (TO END ALL AMPLIFIERS)

A careful analysis of the design principles involved in constructing a direct-coupled amplifier will readily indicate the inadvisability of attempting to make one amplifier having all features! Furthermore, no one individual (of all those who wrote for special amplifiers) requested all features.

The plausible solution to this problem was quite evident. It merely revolves about the design of a basic high-power (30-watt), direct-coupled amplifier which could have any one or more features added to it, in order to fill special requirements.

The prime purpose of the following description is to place before the average constructor or dealer sufficient information to build or buy a versatile amplifier which retains all of the valuable features of a direct-coupled amplifier in addition to any special features that he may require.

The overall function and performance of the amplifier will of course depend upon the number of special circuits incorporated. The essential difference between this unit and commercial amplifiers is that it offers the possibility of combining a number of features, not available at any price!

Flexibility PLUS . . .

You can make more than 100 different kinds of amplifiers by combining any one or more of the various features which may be incorporated into this 30-watt direct-coupled amplifier.

Without any basic design changes, you can add any one or all of the following features:

- Variable speed expander.
- Individual low-frequency boost or cut.
- Individual high-frequency boost or cut.
- High-, medium- or low-gain preamplification.
- Automatic peak limiter.
- Automatic volume compressor.
- Automatic volume control.
- Degenerative low-gain inverter.
- Degenerative high-gain inverter.
- Non-frequency-discriminating scratch-suppressor.
- Remote control.
- One or more high- or low-impedance microphone inputs.
- One or more high- or low-impedance phono or line inputs.

ADVANTAGES OF SPECIAL FEATURES

High Power Output. It is obvious that a high power output will not only enable its use in a wide variety of applications, but will also provide for the production of lower levels with correspondingly reduced distortion. In addition, the high available output will prevent overload when either the expander or any of the frequency-accentuating circuits are employed.

Built-In Push-Pull Expander. Although much has been written about expanders, few have covered the advantages of a *Push-Pull Expander*.

Many fine receivers and high-priced amplifiers have offered a conventional expander utilizing a single pentode or pentagrid mixer. But while many buyers of the finest equipment are aware of the fact that 2nd-harmonics cancel in a push-pull stage, few consider the 2nd-harmonics generated in a single-ended stage. And that these spurious frequencies, of course, carry on through any subsequent push-pull stage. This condition can be eliminated by the use of a push-pull expander.

There are a number of other features

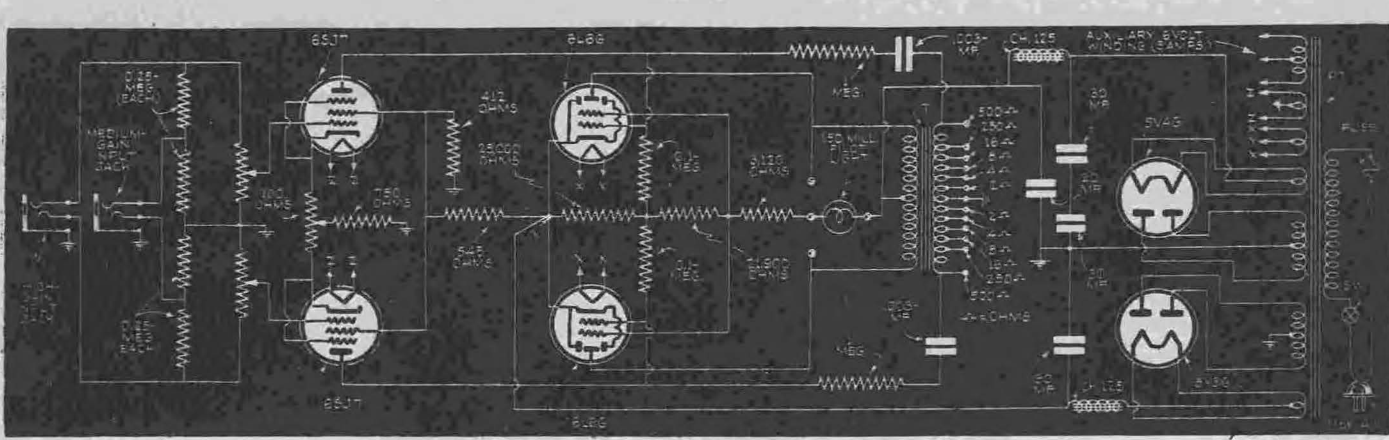


Fig. 2. Complete schematic diagram of the 30-W. all-push-pull direct-coupled amplifier.

a Flexible 30-WATT AMPLIFIER

30 watts at about 3 per cent distortion, from 2—6L6G's in class AB1. filter without frequency discrimination is described by its originator.

of this expander which are highly desirable such as a *Variable Time Delay Control* so as to enable the proper playback of both slow and fast music.

The additional metal tubes shown in Fig. A (2—6SQ7's and 2—6SJ7's) comprise the expander circuit, and the additional 5Y3G rectifier supplies plate voltage to this portion of the circuit. The controls on the front skirt of this amplifier are as follows: **Jacks**: upper-left, Radio Input; lower-left, Phono Input. **Knobs**, left to right: Radio Volume, Phono Volume, L.F. Control, H.F. Control, Expander Degree Control, Expander Speed Control. Upper-right corner: On-Off Pilot Light; lower-right, On-Off Switch.

Bass-Accentuating Attenuation Control. The Bass Accentuating Control is desirable in compensating for many known existing deficiencies in either the speaker, microphone, radio, phono pickup, etc. The type of bass boosting employed is not only effective, simple and economical, but also insures against a rising hum level with increasing bass accentuation.

High-Frequency Control. This circuit provides for the simple accentuating or attenuation of all frequencies above 1,000 cycles. Maximum effect takes place at 20,000 cycles.

High Overall Gain. The value of increased gain is self-evident since it not only enables the attainment of full power output from low-level input devices, but also provides for maintaining peak power operating level under any degree of high- or low-frequency attenuation. Total gain is about 130 db.

Automatic Volume Control. The use of *Automatic Volume Control* has gained considerable popularity during the past few years. A fine distinction exists between many of the control circuits employed; some of them are of the compressing type, others of the peak-limiting type. Few are true automatic volume controlling circuits since they do not tend to raise the power output when the input signal level drops; and inversely, few tend toward maintaining a constant power output when the input signal level rises.

The effectiveness of any of these circuits is a function of its controlling range and harmonic production. The type of push-pull circuit to be described has an exceedingly wide operating range and, because of its push-pull action, introduces a negligible amount of distortion.

Inverter Input Circuit. Although it is desirable to maintain push-pull operation throughout the entire amplifier, there are a number of input devices which do not lend themselves to an input push-pull circuit without the use of a conventional transformer. Because of this condition, a degenerative type of inverter is made available so as to accommodate any type of single-ended input signal.

Non-Frequency-Discriminating Scratch Suppressor. This appears to be an impossibility. The usual method of suppressing scratch was based upon the fallacy that scratch frequencies lie between 1,000 and 3,000 cycles. This band was therefore suppressed. Of course, this type of treatment not only greatly alters the character of the reproduced music, but it also fails to completely eliminate scratch, because of the fact that scratch frequencies lie over a much wider band. They may be found below 300 cycles and well above 3,000 cycles. In order to effectively suppress scratch without altering frequency response, it becomes necessary to differentiate between scratch, and music or voice. A scientific approach to this troublesome problem reveals the following interesting situation:

In normal signal-to-scratch ratio records, scratch is not objectionable at high-level outputs. It gradually becomes more and more objectionable as the signal level decreases, simply because the signal-to-scratch-ratio has increased. It is therefore evident that *an effective method of decreasing objectionable scratch is to automatically lower the overall gain of the amplifier at low-level outputs*. This furnishes us with a form of "inverted expansion," a condition which maintains an average volume level at high output levels, but effectively decreases objectionable scratch at low levels.

An interesting test which demonstrates the effectiveness of this type of scratch suppressor circuit may easily be made by listening to the scratch at the end of a record under the "normal" and "expander" conditions. When the circuit is changed from *normal* to *expansion* the scratch level *actually drops 10 db.!*

When the same change is made during normal signal levels, no change in volume takes place. The overall result is both an apparent and actual reduction of scratch to less than 1/16th its original value! Worn records with excessive scratch become tolerable. New records with normal surface noise become scratchless.

By choosing the proper circuit constants for the inverse expanding circuit so that a flat response is maintained between 20 and 20,000 cycles, the scratch-suppressing circuit becomes non-frequency discriminating.

If we bear in mind that the operating principle of this scratch suppressor is based on an important differentiating characteristic between objectionable and tolerable scratch (signal-to-scratch ratio), we can mentally appreciate its effectiveness. Nevertheless, like many other aural phenomena, it must be heard to really be appreciated. (Insofar as the author is aware this new principle in scratch filters has never before been described in print.—*Editor)*

Remote Control. Like all of the other features mentioned, ***Remote Control*** may under certain conditions become a prime requisite in an amplifier installation. Many types of remote control circuits have been developed — some good, others bad; but few indeed have been incorporated into push-pull circuits. The control of gain by applying a variable bias (via the cathode circuit) to a variable-mu tube has met with wide approval. Its distortion characteristics, at various attenuation levels, are usually overlooked because subsequent distortion introduced in the following stages usually masks that of the input stages.

The remote control circuit employed with this amplifier introduces no 2nd- and a negligible amount of odd-harmonics. Its smooth attenuating range, freedom from microphonics, minimum distortion, ability to use ordinary tubes, and extreme simplicity mark it as a distinctive contribution to remote control circuits.

Flexibility of Input Circuits. The wide variations of applications to which a general amplifier may be subjected makes it imperative to design a flexible input circuit whereby any number of high- or low-impedance inputs may be made available to accommodate any combination, or type of input signal devices. Such a flexible input system is described, for use in conjunction with the basic 30-watt direct-coupled amplifier.

Flexibility of Gain. By making available a choice of a high-, medium- and low-mu input stage (or any combination) it becomes possible to provide a wide choice of overall gain which may be judiciously selected in accordance with the type of input device employed. To augment the input stage a high- or low-gain inverter is offered, for special applications to further increase the gain range.

TAILORING THE AMPLIFIER TO FIT REQUIREMENTS

Packaged Engineering. The great variety of variables that can be introduced, added or selected for use with this basic amplifier places into the hands of the layman the ability to construct or select any specific type of amplifier regardless of the complexity of its overall circuit. Furthermore, it is no longer necessary to purchase or build a complex amplifier, having a number of unnecessary features in its initial design just because one or two of its many features are really necessary.

The simple expedient of providing an isolated, well-regulated additional rectifier enables the addition of as many as 10 extra low-level input tubes without upsetting the voltage distribution balance within the amplifier. An extra heater winding which may or may not be used provides optional heater current for ten 0.3-amp. tubes.

The Design of the Fundamental Amplifier. The design considerations involved in this amplifier follow closely the principles laid down for the 10-watt model previously described. The essential difference being however that the output tubes are operated with a plate voltage of 400 and screen-grid voltage of 300. As the circuit is of the class AB1 type, no grid current is drawn during any part of the input cycle.

If we list the tubes and their corresponding applied potentials, we have the essence of our new 30-W. Direct-Coupled Amplifier.

	Operating Conditions	
ELECTRODE	6SJ7	6L6G
Plate volts (Ep)	150	400
Control-grid (Ec1)	3	25
Screen-grid (Ec2)	75	300
Suppressor-grid (Ec3)	0	...
Plate current, av. (Ib)	1.5	77 ma.
Screen-grid current, av. (Ic2)	0.5	12 ma.

Although the voltages and current indicated above are for a single-ended amplifier, the push-pull model utilizes the same voltages and twice the screen-grid and plate currents of 1 tube.

Figure 1 shows the basic circuit diagram of the amplifier. It will be noted, by comparison with the basic 10-Watt Direct-Coupled Amplifier (see page 17, July 1939 issue *Radio-Craft*) that only 1 additional resistor is required. Resistor H, which is used to drop the 400 volt plate voltage to 300 for the screen-grids of the 6L6G tubes.

Calculation of Resistor Values. The design procedure necessary to calculate the values of the 7 important resistors (B, C, D, E, F, G, H) simply involves the application of Ohm's Law. If we remember that the voltage drop in resistors G should be made equal to the voltage applied to the plate of the input tubes, it is evident that the voltage at the junction of the resistors E-F should be 300 volts (off ground). In order to apply an effective screen-grid voltage of 300 volts to the screen-grids of the output tube, 475 volts as measured (from ground) should appear at the junction of resistors F-H.

Knowing the voltage drop desired in each resistor, and as well as the current flowing through it, its value is calculated as shown on the opposite page.

The power supply utilizes 2 separate sets of rectifier circuits to avoid the use of high voltages. One rectifier delivers 400 volts for the output plate voltage while the other furnishes the 175 volts required for the 1st stage and any auxiliary circuit which may be added. Excellent regulation is maintained in this latter circuit by maintaining a comparatively large "bleeder" current across its output. This so-called "bleeder" current (of approximately 168 ma.) is actually the plate and screen-grid current of both output tubes after they have done their useful work in the power output stage.

The completed circuit diagram is shown in Fig. 2. It will be noted that a number of refinements have been added. The "tube balancer" placed in the cathode circuit of the input stage enables the matching of what would ordinarily be considered "mismatched" tubes. It also provides for a single hum-balancing adjustment.

A 150 ma. pilot light is placed in series with the plate supply of the power output stage to avoid damage to the output tubes, should the bias rectifier be accidentally removed. Complete circuit details and design factors for all of the available auxiliary circuit features for this 30-Watt Direct-Coupled Amplifier will be described

CALCULATION OF RESISTOR VALUES

$$\text{Resistor H} = \frac{100}{2(I^*_{c_2}) + 4(I_b + I_{c_2})} = \frac{100}{2(.012) + 4(.0015 + .0005)} = 3{,}125 \text{ ohms.}$$

$$\text{Resistor F} = \frac{175}{4(I_b + I_{c_2})} = \frac{175}{4(.0015 + .0005)} = 21{,}900 \text{ ohms}$$

$$\text{Resistor G} = \frac{150}{I_b} = \frac{150}{.0015} = 100{,}000 \text{ ohms}$$

$$\text{Resistor E} = \frac{125}{2(I_b) + 4(I_{c_2})} = \frac{125}{2(.0015) + 4(.0005)} = 25{,}000 \text{ ohms}$$

$$\text{Resistor D} = \frac{100}{2(I^*_b + I^*_{c_2}) + 2(I_b) + 4(I_{c_2})} =$$

$$\frac{100}{2(.077 + .012) + 2(.0015) + 4(.0005)} = 546 \text{ ohms}$$

$$\text{Resistor C} = \frac{75}{2(I^*_b + I^*_{c_2}) + 2(I_b + I_{c_2})} =$$

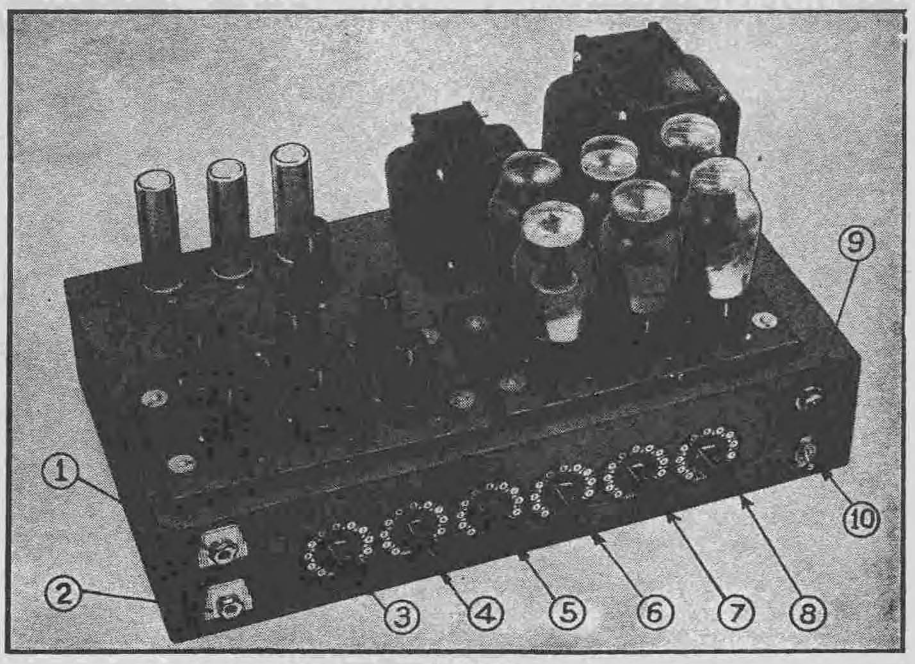
$$\frac{75}{2(.077 + .012) + 2(.0015 + .0005)} = 412 \text{ ohms}$$

$$\text{Resistor B} = \frac{3}{2(I_b + I_{c_2})} = \frac{3}{2(.0015 + .0005)} = 750 \text{ ohms}$$

Resistor A = 0.5-meg.

*Indicate currents for output (6L6G) tube.

Many Radio-Craft *readers have requested a really modern audio amplifier utilizing triodes. The multi-feature amplifier here described is a high-fidelity semi-triode job which answers these requests. Power output is about 15 W. with under 1.5% total harmonic distortion from a pair of 2A3's; at 5% total harmonic distortion the amplifier will deliver 30 W.*



The Triode Direct-Coupled 30-W. Amplifier. The controls, etc., are identified as follows: (1) phono input, (2) radio input, (3) radio volume control, (4) phono volume control, (5) expressor, (6) timing, (7) low-frequency equalizer, (8) high-frequency equalizer, (9) pilot, (10) master switch.

A TRIODE DIRECT-COUPLED AMPLIFIER

With Non-Frequency-Discriminating Scratch Suppressor

THIS unusual amplifier has been developed for a group of music lovers and technicians who are profound in their belief that triode power output amplifiers are incomparable in quality.

A discussion of the comparative quality obtained from pentodes, tetrodes, and triodes would bring up a number of questions which are both varied and complex. Notwithstanding academic objections (which are advanced by beam power enthusiasts) a 2A3 offers a number of very desirable circuit features, including low plate resistance, which makes it particularly adaptable for feeding output loads which have a pronounced impedance-changing characteristic with variable frequency. Its single-control-grid construction avoids regulation problems encountered in multi-grid output tubes. Its unusual power handling ability provides for the attainment of 15.6 watts at 1.4% total harmonic distortion, from a pair of 2A3's, under ideal conditions (which are rarely attained).

In order to carry the triode idea out within the basic direct-coupled circuit, a pair of 6SJ7's operated as triodes, are used as push-pull voltage amplifiers.

ADDITIONAL FEATURES

In designing this amplifier it was decided to build-in a number of desirable features which would enable the music lover to derive maximum enjoyment from his audio equipment. The following features were therefore incorporated:

(1) Push-Pull Expansion, with Variable Time Delay.

(2) Non - Frequency - Discriminating Scratch Suppression.

(3) Independent Low-Frequency Equalization.

(4) Independent High-Frequency Equalization.

(5) Choice of Optional Push-Pull or Single-Ended Input Circuit.

(6) Dual Grid Inversion.

THE PUSH-PULL EXPANDER

A contemporary reviewer of the advancement in the electronic art, said for the year of 1934 "volume expansion in radio sets came in with a boom and went out very quietly."

There is no doubt that volume expansion has been subjected to a considerable amount of misuse which may have led it into disrepute through careless design or operation. Unquestionably, many of the circuits developed in the rush to present volume expansion to the public presented a number of objectionable circuit characteristics which overbalanced the benefits of volume expansion. A number of tricky circuits involving microphonic tubes were presented. Many experimenters indiscriminately added these circuits to existing amplifiers and found, among other things, overload distortion was being caused by expander circuits. This proves, beyond any doubt, that an expander to really "do its stuff," must be specifically designed into an amplifier, and cannot be indiscriminately added to any existing unit.

There is no better argument for volume expansion than the fact that the Bell Telephone Laboratories, in demonstrating their stereophonic recordings at Carnegie Hall in April, 1940, introduced a new technique of "enhancing" music, which among other things, involved nothing more nor less than exaggerated expansion. It is true that some technicians will advance the theory that a properly recorded selection does not require expansion. While such may be the case, *few selections are properly recorded* and nearly all can be "enhanced" by use of the correct type of expansion.

Though no expansion standardization exists in the present stage of the art, the writer believes it is only a question of time before recordings are offered to the public with a coefficient of compression notated on each record. In playing the record back, the playback equipment would be set to a coefficient of expansion equal to the coefficient of compression so that the ultimate rendition would be heard in the same dynamic range as it was originally played. Actual experience has shown that many records may be improved when properly expanded. Furthermore, listeners who condition their hearing to expanded music, will never again enjoy "flat" or unexpanded renditions.

The technical problems involved in designing an acceptable expander are comparatively simple. They are mainly centered about the selection of suitable circuit constants so as to provide minimum distortion, wide-range variable time delay, and suitable precautions for avoiding overload at maximum expansion.

Minimum distortion may be maintained by utilizing push-pull expansion. Wide-range variable time delay is made available by the use of push-pull signal voltage amplification and full-wave rectification for obtaining control voltages for the expander circuit.

The simplest way of avoiding overload under conditions of maximum expansion is to incorporate a loss, when the expander is placed in the circuit, which is exactly equal to the greatest expansion encountered, so that under conditions of maximum expansion, the rated power output of the amplifier is not exceeded.

NON-FREQUENCY-DISCRIMINATING SCRATCH SUPPRESSOR

When the Non-Frequency Discriminating Scratch Suppressor developed in our laboratories was first described, for the first time in any radio magazine, in the October, 1939, issue of *Radio-Craft*, it created a considerable amount of interest. It was therefore decided that additional work be done on this project until a highly-perfected circuit could be made available. The circuit as finally employed is virtually an inverted expander and operates on the principle that

(Continued on page 15)

How to Add 1 to 14

ALL-PUSH-PULL DIRECT-COUPLED

"Packaged engineering" applied to a basic sound amplifier now enables or more of a large number of important features—90% of which, Mr.

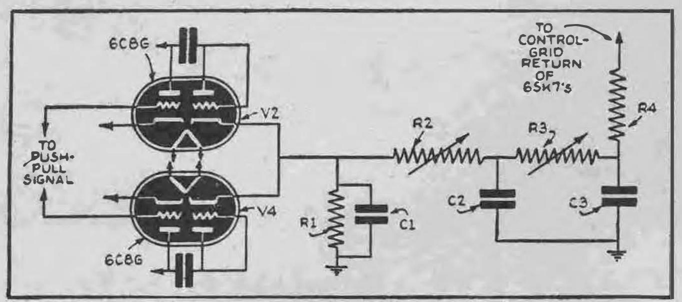


Fig. 1. Full-wave diode rectifier of expander.

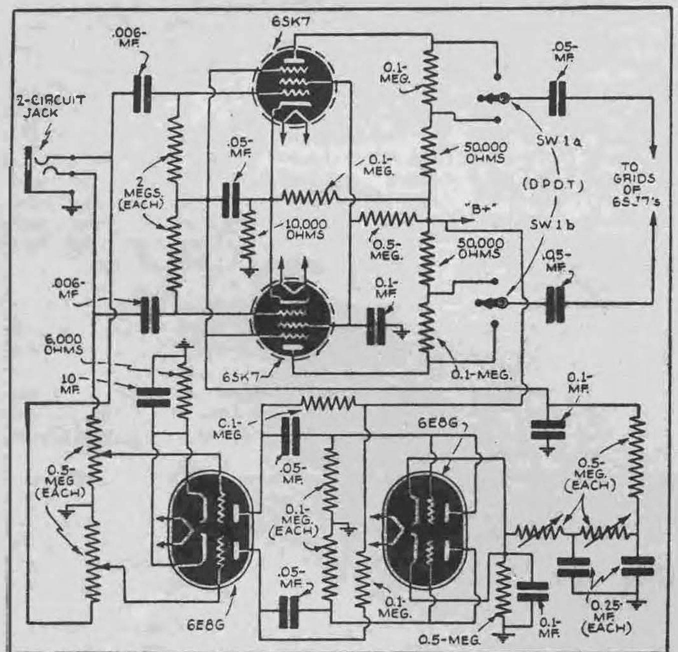


Fig. 3. Non-overloading push-pull expander with variable time delay.

THESE features provide for the first time, a practical system of "packaged engineering," whereby any one or more of the following 14 available features can be added to the popular 30-Watt Direct-Coupled Amplifier.

(1) Variable Speed Non-Overloading Push-Pull Expander
(2) High-Frequency Accentuation and Attenuation
(3) Low-Frequency Boost and Cut Control
(4) Push-Pull Automatic Volume Compressor
(5) Push-Pull Automatic Volume Limiter
(6) Push-Pull Automatic Volume Control
(7) Non-Frequency-Discriminating Scratch Suppressor
(8) Additional Low-Gain Single-Ended Input
(9) Additional High-Gain Single-Ended Input
(10) Additional Low-Gain Push-Pull Input
(11) Additional High-Gain Push-Pull Input
(12) Audio Spectrum Control
(13) Calibrated Volume Indicator
(14) Remote Control

A review of the design of the Direct-Coupled Amplifier described on page 202 of the October, 1939, issue of *Radio-Craft*, will coordinate the description of the features to follow; although this amplifier is being used as a basic design, these features may also be added to most other amplifiers—depending, in part, upon the individual feature selected, and the available amplifier.

WHY NOT DIRECT-COUPLED THROUGHOUT?

The question was raised in our laboratories as to the advisability of designing the additional features in such a manner as to permit direct-coupling between the auxiliary feature and the amplifier proper. The design problems encountered, however, in following such procedure, make it obviously impossible to economically maintain the ease with which these auxiliary features can be added through a standard resistance-condenser coupling method. If the auxiliary features were to be direct-coupled into the main amplifier, a different design would have to be worked out for each particular combination. *As there are over 14 individual features, which make a total possible combination of nearly 16,000 different variations, the inadvisability of designing so many different types of amplifiers is quite apparent.*

With the conventional method of coupling between the auxiliary features and the amplifier proper, it becomes

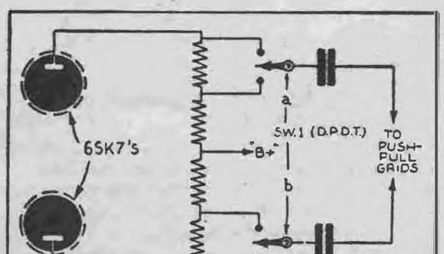


Fig. 2. Losser circuit in expander plate output. Note that the switch sections (A&B) of Sw.1 operate simultaneously, either towards both plates (no loss) or towards "B+" (12 db. loss).

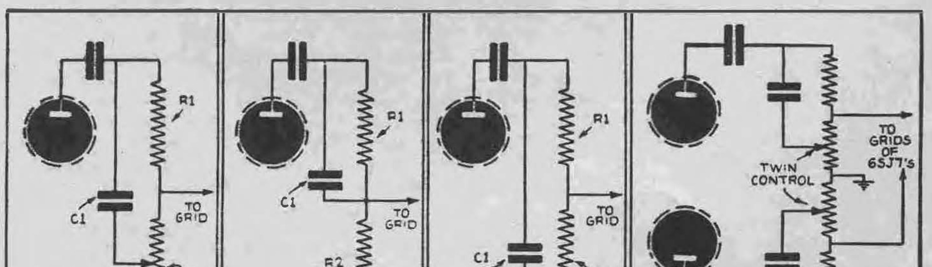


Fig. 4. At A—basic H.F. circuit. B—H.F. accentuation. C—H.F. attentuation. D—Push-pull H.F. equalizer.

Modern Features to the

30-WATT P. A. AMPLIFIER

any public-address specialist to incorporate in a suitable amplifier one Shaney believes, have never before appeared in any radio magazine!

PART II

feasible to standardize on the 2-stage direct-coupled amplifier unit, and provide suitable heater and plate voltages for any one or more auxiliary features. The extraordinary flexibility of this arrangement more than compensates for any advantages in maintaining direct-coupling throughout the input stages. Furthermore, it is comparatively simple to design a wide-range low-level input stage. As these stages or auxiliary features are to be designed for push-pull operation, it becomes a relatively simple matter to outperform any existing types of standard single-ended circuits.

THE VARIABLE SPEED NON-OVERLOADING PUSH-PULL EXPANDER

During the past few years, volume expansion has received a considerable amount of attention from design engineers. Its practical application, however, has not attained the wide popularity that it deserves, because of the following 4 reasons:

1.—Individuals who have never heard volume expansion, find it very difficult to imagine this type of circuit operation, and unless it is actually demonstrated, it is rather difficult to sell.

2.—The usual type of volume expansion incorporates a *fixed* time delay. This limitation makes it definitely unsuited for the wide variety of recordings that may be used, because of the fact that expansion for reproduction of slow and rapid tempo music, requires different timing.

3.—Expanders which have been successfully applied, invariably produce overload distortion because the peak power of the amplifier is incapable of adequately handling the full power output at points of maximum expansion.

4.—Nearly all commercial expanders are of a single-channel type, and because of their pentode-type construction, introduce an appreciable amount of even-order harmonics, which are carried on through the output of the amplifier.

The expander developed for this 30-Watt Direct-Coupled Amplifier, eliminates all of these objections and places before the music lover an outstanding development to enhance all types of recordings.

The difficulty of making people understand the value of expansion is most effectively circumvented by an actual demonstration. A complete description of *why* an expander should be used, will be found in the April, 1938, issue of *Radio-Craft*, page 76, under the paragraph heading "Why An Expander Should Be Used."

The value of incorporating a *variable* time delay is quite apparent. The usual practice has been to use a delay of approximately ¼ of a second. While this may be a good average setting, it is certainly inadequate for proper expansion of "jitterbug" records. On the other hand, a slow symphony with slowly rising swells, should have a slower acting expander to smooth the rate of expansion and more nearly complement the recording engineer's monitoring.

THE TIME DELAY CIRCUIT

By referring to Fig. 1, it will be noted that the resistor-capacity filter system feeding from the diode rectifier has a variable resistor network. As the time constant of this filter is a function of the variable resistors R2 and R3 it

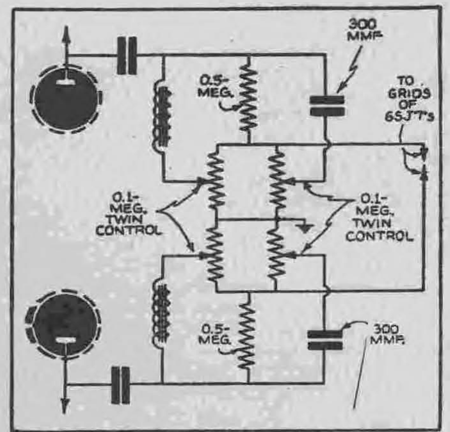


Fig. 6. High- and low-frequency equalizers.

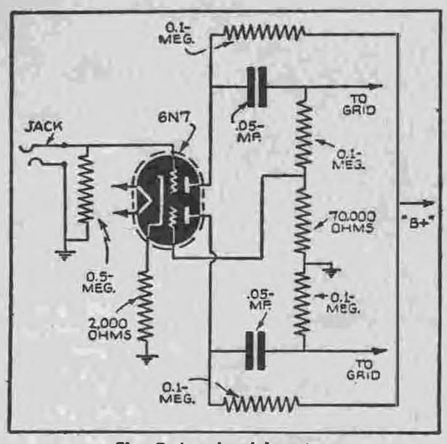


Fig. 7. Low-level inverter.

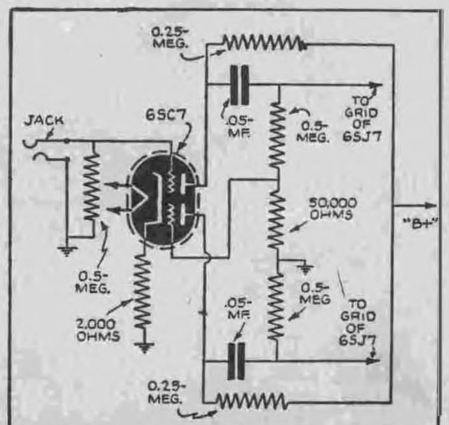


Fig. 8. High-gain inverter.

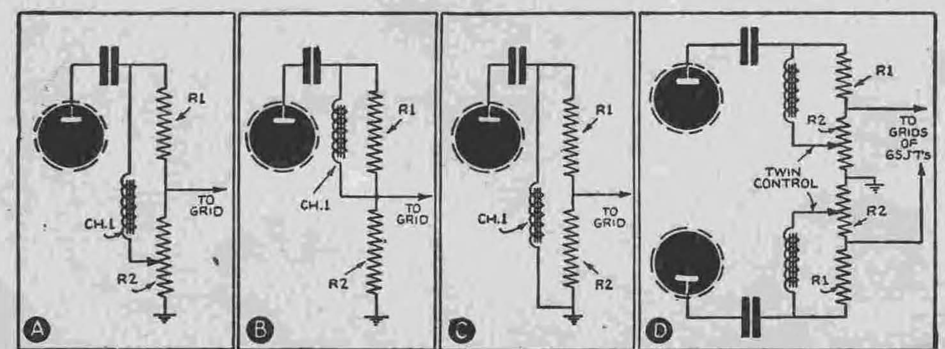


Fig. 5. At A—basic L.F. circuit. B—bass boost. C—bass cut. D—Push-pull L.F. equalizer.

becomes possible to vary the expansion time from 0.05-second to 0.5-second (an overall time ratio of 100).

One of the additional important advantages obtained in using push-pull expansion is that full-wave rectification takes place at V2 and V4 after the expander amplifier V1 and V3 have amplified the incoming signal. This push-pull rectification cuts hash down considerably and enables the use of comparatively low values for R1, R2, R3 and R4 (filter resistors) which in turn provide for a wide range in time delay settings without the introduction of hash into the amplifier proper.

WHY USE A 30-WATT AMPLIFIER?

One of the most important contributing factors in the development of overload distortion in expander amplifiers is that when a predetermined degree of expansion is employed, insufficient attention is given to the clean power output rating of the amplifier. Few technicians realize that when 12 db. expansion is employed, in an amplifier delivering an average level of 1.5 watts, a total of 24 watts is developed at peak expansion. (A simple method of calculating this, is to remember that for every 3 db. the amplifier power is doubled; 12 db. progressively doubles the output 4 times from 1.5 watts to 3, 6, 12, and finally 24 watts. Likewise, if the original average signal is 2 watts, 12 db. expansion will bring it up to 32 watts!) Needless to say, a 10- or 20-watt amplifier will not properly handle this degree of expansion, and here lies the true reason why a 30-Watt Amplifier is not excessive for home use.

It has been the writer's experience to note a considerable degree of surprise when a 30-watt amplifier (with expansion) was recommended for home use. The usual astonished reply was, "What! 30 watts! I will drive all my neighbors out!" It is to be borne in mind, however, that the 30-watt level may only be maintained *for a fraction of a second* during the rendition of the entire recording. Furthermore, amplification of low frequencies at this high level, will not produce half the amount of the estimated disturbance. Of course, the amplifier need not always be operated at a 12 db. expansion setting, nor need it be operating at a 2-watt average.

OVERCOMING OVERLOAD DISTORTION

In order to overcome the possibility of overloading this amplifier under any operating condition, a losser circuit is incorporated in the plate circuits of the 6SK7's, which operate in conjunction with a D.P.D.T. switch Sw.1, Fig. 2, to introduce a loss of 12 db. As the amount of expansion will not exceed 12 db. when the expander switch is snapped into the circuit, at any predetermined level, 12 db. is automatically subtracted. It is therefore impossible to exceed the previous normal level at peak expansion. This type of circuit insures against overload distortion.

The objections of a single-channel system are completely overcome by the use of the push-pull expander circuit illustrated in Fig. 3. This type of a circuit completely cancels all even-order harmonics and makes available *for the first time,* a high-quality phono amplifier circuit. It will be noted that the usual 1612 pentode grid converter, or its more microphonic prototype the 6L7, is not employed, because of the high even-order harmonics introduced with this tube at comparatively low input level.

HIGH-FREQUENCY ACCENTUATION AND ATTENUATION CIRCUIT

The desirability of using frequency equalization cannot be disputed by any amplifier technician (unless, of course, the amplifier is to be used for some fixed laboratory setup). One of the most effective types of high-frequency equalizing circuits is illustrated in Fig. 4A.

In Fig. 4A, the resistor network, R1, R2, introduces a fixed loss into the grid circuit of V1, dependent upon the ratio of R2 to R2 + R1. If a 10 db. accentuation is desired at 10,000 cycles, the network is designed to produce a 12 db. loss at 1,000 cycles.

When the potentiometer, R2, is at the "boost" position, Fig. 4B results. Condenser C1 will then bypass resistor R1 and effectively shunt this resistor to a negligible value. This effect, however, will only take place at high frequencies, so that a slowly rising characteristic is obtained. By proper design, the rise can be started at 1,000 cycles. The slope of the degree of accentuation is dependent upon the setting of R2.

If, on the other hand, R2 is set for high-frequency attenuation, Fig. 4C results. Under this condition, the high frequencies are simply shunted. The slope of this attenuation circuit is likewise dependent upon the selection of the resistor and condenser values. An intermediate setting between 4A and 4B will result in a normal (straight line) response.

The adaptation of this circuit for push-pull operation is relatively simple. It merely involves the use of a twin volume control and duplicate set of resistors and condensers, as is illustrated in Fig. 4D. This type of tone circuit makes available, *for the first time,* a true push-pull high-frequency equalizing system.

THE LOW-FREQUENCY BOOST AND CUT CONTROL

No equalizing circuit can be considered really complete unless it provides for individual control of both the high and low frequencies. For ideal conditions, the design of this type of circuit should provide independent control of both the high and low frequencies, without any interaction. The most effective "mate" circuit for the high-frequency equalizer is basically illustrated in Fig. 5A.

It will be noted that this circuit is virtually identical to the high-frequency equalizer, with the exception that a choke (Ch.1), is used in place of the condenser, C1. As is well known, an ideal choke exhibits a diametrically opposite characteristic to frequency response when compared to a condenser. In other words, the impedance of a condenser increases with a decrease in frequency, and decreases with an increase in frequency. A choke, however, reacts in an opposite manner. That is, its impedance increases at high frequencies, and decreases at low frequencies. Based upon this well-known phenomenon, it can easily be noted, how Ch.1 boosts bass frequencies when R2 is set for maximum bass boost.

See Fig. 5B. Here Ch.1 effectively shunts resistor R2, *but only at low frequencies.* Therefore, the ratio of R2 to R2 + R1 changes to increase bass response.

When the control is set for bass cut, see Fig. 5C, the choke effectively shunts the output of the preceding circuit, *but only at low frequencies.* Naturally, this shunting effect is inversely proportional to frequency. That is, it is greater at lower frequencies.

The adaptation of this circuit to push-pull follows conventional procedure, and is illustrated in Fig. 5D.

A COMPLETE EQUALIZER CIRCUIT

By combining both of these basic circuits, a unique and highly effective push-pull individual high-frequency and low-frequency accentuating and attenuating circuit is made available. Inasmuch as both circuits are virtually identical, it becomes a relatively simple matter to combine them by merely using 2 twin controls, of such a value, so that their effective shunt resistance is equivalent to the value of R2. A completed tone-compensating circuit is illustrated in Fig. 6.

OPTIONAL LOW-GAIN SINGLE-ENDED INPUT

As many experimenters, technicians, and users of this type of an amplifier have expressed difficulty in obtaining a push-pull signal to be fed into the push-pull amplifier, a low-gain inverter tube, the 6N7, is provided as an optional input circuit. See Fig. 7.

It will be noted that this circuit follows conventional design with the exception that plate load resistors having very low values are used to insure quiet operation. This in-

(*Continued on page* 30)

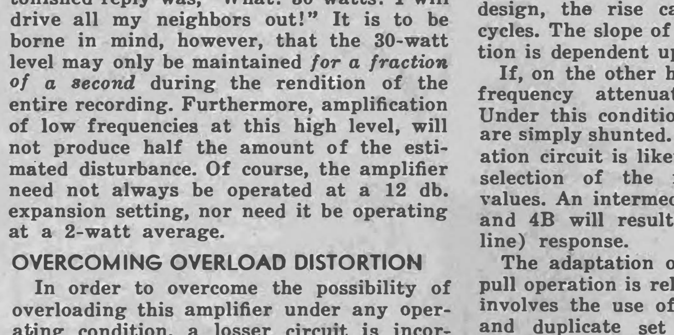


Fig. 9. Low-gain push-pull input.

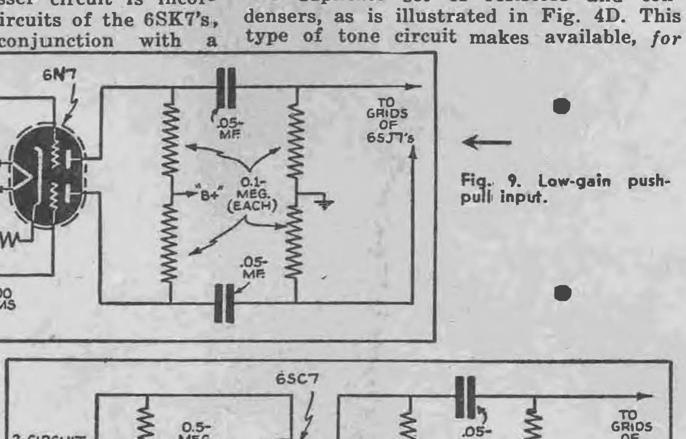
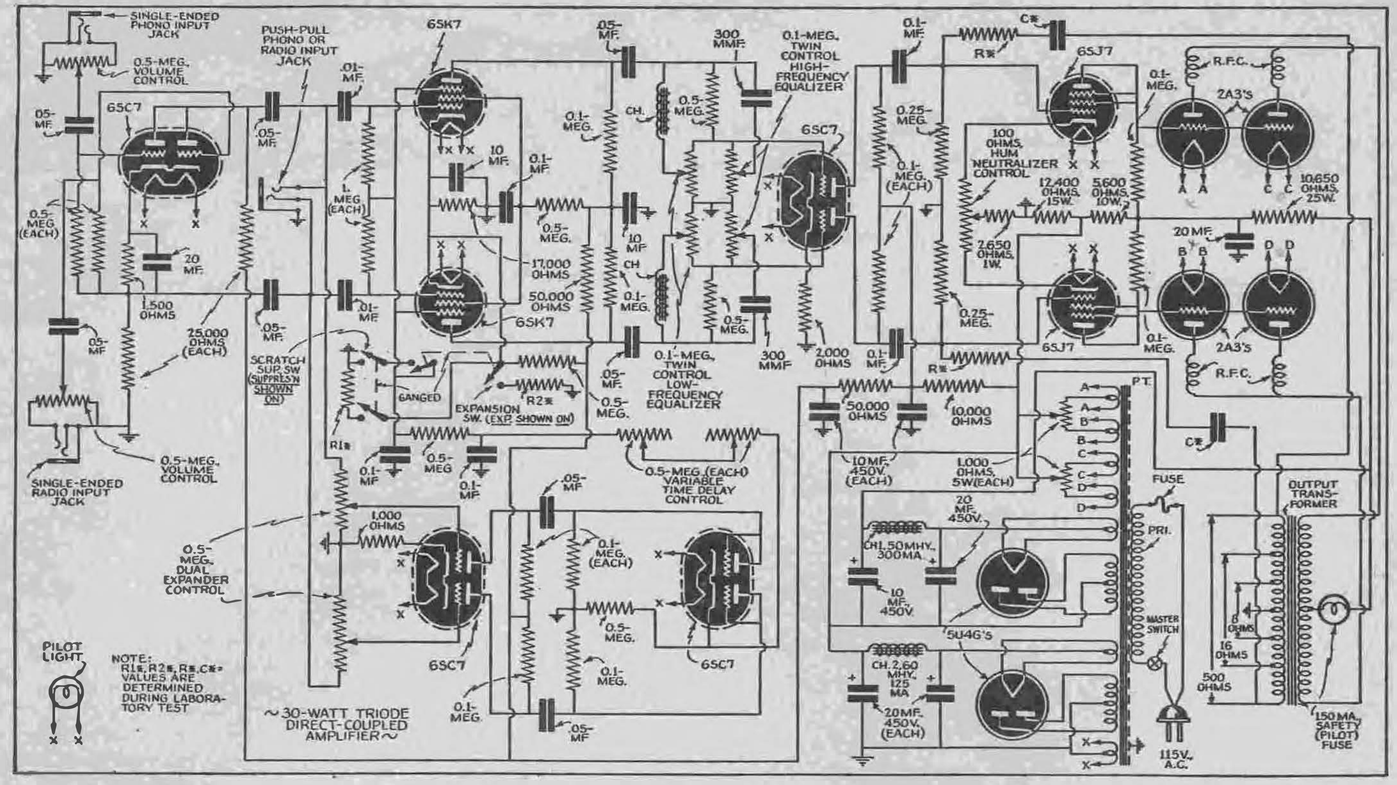


Fig. 10. High-gain push-pull inp t.

Fig. 11. P sh-pull detector circuit.

(Continued from page 11)



scratch may be separated from signal by the difference of the average level of each.

If we were to design an ideal scratch suppressor, it would be necessary to find a number of differential characteristics which would provide for the simple separation of scratch from signal. The fallacy of cutting high frequencies in an attempt to suppress scratch is of course obvious. When the upper audio spectrum is effected, it alters the higher harmonics of any tone. Some so-called scratch filters eliminate, or greatly attenuate, all frequencies above 3,000 cycles. While it is true that the ear may be conditioned to this type of reproduction, unconditioned listeners would immediately notice the limited range of such a system. It is therefore apparent, that in order to separate scratch from signal, frequency response should not be altered.

In attempting to find methods for suppressing scratch, the following 2 solutions presented themselves:

(1) Inasmuch as scratch is a conglomerate of indiscriminate frequencies, and music is characterized by discrete frequencies, it seemed possible to develop a set of discrete and indiscrete filters, each feeding into control-grids of a *differential* amplifier arranged so that when a preponderance of indiscrete frequencies were present, the volume would drop, and when a preponderance of discrete frequencies were present, the volume would maintain its average level. This effect would ultimately produce lower levels when scratch was present and normal levels when definite signals were being reproduced.

(2) Scratch is not objectionable at high-level outputs, but gradually becomes more and more objectionable as the signal level decreases, only because the signal-to-scratch ratio has decreased. It follows that an effective method of decreasing objectionable scratch is to automatically lower the overall gain of the amplifier at low levels and maintain its average level at normal levels.

The latter method seemed to offer a simpler solution. In the process of the development of this circuit, it was found that the functions of both the expander and the suppressor could be combined in the same pair of tubes, it being only necessary to increase the level above predetermined average for expansion and to decrease it for scratch suppression.

It was further found, that the *speed* of scratch suppression needed adjustments for different selections, because of the fact that the sensitivity of the ear does not change instantaneously with changes in level. To provide a wide degree of time delay control, a full-wave rectifier is employed, which couples to a separate push-pull control voltage amplifier. The use of full-wave rectification (which doubles the frequency of the rectified voltage) provides for small capacitative filters which thereby enable the use of high-speed control circuits without introducing hash into the signal control-grids of the expander - scratch-suppressor circuit. The degree of expansion and suppression is controlled by the "Expressor" control (which is a contractual abbreviation for expander - suppressor).

THE LOW-FREQUENCY EQUALIZER

Even-order harmonics introduced in single-ended circuits are, as is well known, canceled in push-pull stages. It was therefore decided to utilize push-pull low-frequency equalization so that all even harmonics introduced into the circuit by equalization would be canceled at the output.

The Low-Frequency Equalizer employed, has been completely described in the Nov., 1939, issue of *Radio-Craft* (Page 269). It is designed to provide a 15 db. cut and boost at 40 cycles, with a gradual tapering of control up to 1,000 cycles. Its overall range is 30 decibels, which is more than adequate for compensation of existing deficiencies in speakers, phono pickups, microphones, and radio tuners.

HIGH-FREQUENCY EQUALIZATION

The High-Frequency Equalizer follows the Low-Frequency Equalizer principles. It is also employed in a push-pull fashion. Its operating principles were also described in the Nov., 1939, issue of *Radio-Craft*. The revised circuit, however, provides for a 15 db. cut and boost at 15,000 cycles, providing a tapering control down to 2,000 cycles. For special equalization applications, these equalizers may be replaced by push-pull audio spectrum controls, the operating principles of which were described in the Dec., 1939, issue of *Radio-Craft* (Page 342).

THE BASIC AMPLIFIER

The basic amplifier is composed of four 2A3's to provide for minimum distortion under maximum expansion. A pair of 6SJ7 triodes provide more than adequate voltage to drive the four 2A3's in a strict class A condition. One 5U4G furnishes voltage for the output power tubes, while another supplies fixed-bias for the driver tubes and plate voltage for the expander - suppressor tubes as well as the dual-grid inverter.

Independent balancing controls are provided for matching plate currents of the 2A3's. Individual anti-parasitic filters are inserted in the plates of all output tubes to prevent the slightest trace of transient distortion under any condition of operation. Push-pull inverse feedback is employed to insure an extended low-frequency response down to 20 cycles.

THE BALANCED FEEDBACK CIRCUIT

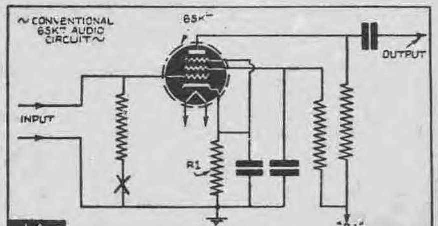
Although feedback is generally used for reduction of distortion, hum, and output plate resistance, none of these corrective measures is necessary in this amplifier. A balanced frequency-discriminating feedback network is employed, however, to compensate for odd-order harmonics introduced in the output transformer and for frequency correction. The inverse feedback circuit employed is of a distinct frequency-discriminating type. The value of the resistors and condensers in this circuit are dependent upon the design and construction of the output transformer employed. They are therefore not given. It is important, however, that a balanced output transformer be used in order to attain a balanced inverse feed-

(Continued on page 30)

How to Add 1 to 14

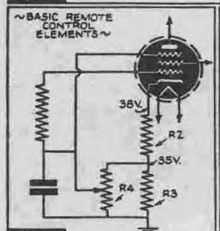
ALL-PUSH-PULL DIRECT-COUPLED

"Packaged engineering" applied to a basic sound amplifier now enables or more of a large number of important features 90% of which, Mr.



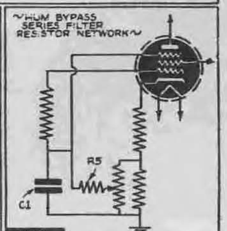
~CONVENTIONAL 6SK7 AUDIO CIRCUIT~

1A



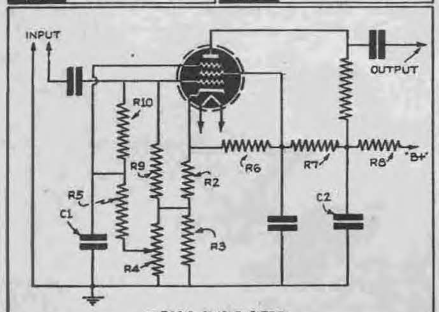
~BASIC REMOTE CONTROL ELEMENTS~

1B



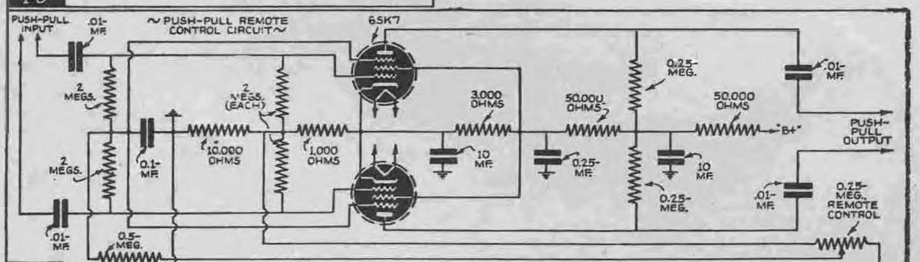
~HUM BYPASS SERIES FILTER RESISTOR NETWORK~

1C



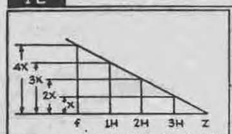
~BASIC SINGLE-SIDED REMOTE CONTROL CIRCUIT~

1D



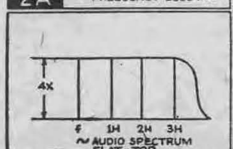
~PUSH-PULL REMOTE CONTROL CIRCUIT~

1E



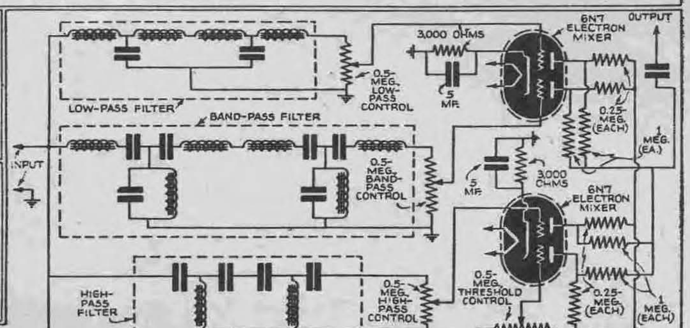
~CONVENTIONAL SLOPING FREQUENCY BOOST~

2A



~AUDIO SPECTRUM FLAT-TOP FREQUENCY BOOST~

2B



~AUDIO SPECTRUM CONTROL CIRCUIT~

2C

ANY one or more of the following 6 optional features may be added to the flexible, All-Push-Pull 30-Watt Direct-Coupled Amplifier, originally described in the October, 1939, issue of *Radio-Craft*, with or without the 7 auxiliary features described in the November, 1939, issue of *Radio-Craft*. Lack of space prevents the description of the Non-Frequency Discriminating Scratch Suppressor, which will be covered in a subsequent issue, if readers are interested.

The features to be described in this issue are:

(1) Remote Control
(2) Audio Spectrum Control
(3) Calibrated Volume Indicator
(4) Push-Pull Automatic Volume Compressor
(5) Push-Pull Automatic Volume Limiter
(6) Push-Pull Automatic Volume Control

REMOTE CONTROL

Remote control circuits have been widely adopted during the past few years for many special applications; particularly in those installations requiring control of volume at some point remote from the amplifier proper. Naturally, the usual grid circuit volume control would be impractical because of hum pick-up in this high-impedance line. Although low-impedance lines and suitable attenuators may be employed in conventional coupling circuits, this is undesirable because of the increased costs involved, and the difficulty of maintaining flat frequency response with economical matching transformers.

The popular procedure of controlling volume remotely is to employ a pentagrid converter tube (such as the 6L7 or its prototype), and varying the bias of both control-grids. As these tubes were designed primarily for R.F. mixer service, they do not readily lend themselves to audio work because of inherent hum, microphonics, and wide variations in characteristics between similar tubes of the same type.

The conventional remote control circuit furthermore employs a single-sided stage which, in itself, introduces an appreciable amount of distortion. Naturally, the practical solution to this problem is to utilize a push-pull circuit employing tubes of a relatively non-microphonic nature, low inherent hum, and of comparatively stable characteristics.

Figure 1A. The 6SK7 lends itself admirably to this type of application. Figure 1A shows the conventional manner of using a 6SK7 with a series dropping screen-grid resistor. This type of circuit provides a number of interesting and valuable secondary features, such as the change in both screen-grid and plate voltage corresponding to their respective plate and screen-grid currents, which in turn, are controlled by the control-grid voltage.

For remote control purposes, it is important to design the circuit, so that minimum distortion takes place, regardless of the transconductance of the tube. By introducing a variable voltage at the point X, the transconductance of the tube may be changed from 2,000 to 10 micromhos. To further increase the controlling range of the variable voltage, the screen-grid should be disconnected from the cathode and connected to the low end of the grid-return resistor. This, in effect, provides twin-grid control of the electron stream.

If the cathode resistor R1 is made large enough to obtain a bias of –35 volts, the tube will be operating at minimum gain. If the ground end of the grid-return resistor is slowly varied across the cathode resistor R1, the bias

Modern Features to the

30-WATT P. A. AMPLIFIER

any public-address specialist to incorporate in a suitable amplifier one Shaney believes, have never before appeared in any radio magazine!

PART III

will be decreased, and its gain increased. A limiting resistor should, of course, be inserted at the high end of the cathode, so as to avoid bringing the bias to below –3 volts, as this would bring the tube off its recommended grid-voltage plate-current curves.

Figure 1B. This figure shows a revised cathode and grid control circuit to obtain the desired effect. It will be noted that the cathode is 38 volts off ground, while the low end of the limiting resistor R2 is 35 volts from ground. The difference between the two is the effective grid voltage applied to both the control grid and suppressor grid at maximum gain.

Figure 1C. When the center arm of R4 is brought towards ground, the bias is gradually increased, until cut-off is approached. If R4 was placed at some remote point, the bias (and gain) of the tube could easily be controlled remotely.

In order to by-pass hum voltages picked up within the three-wire cable, a series filter resistor network should be employed (R5, C1), as illustrated in Fig. 1C. The time constant of this combination should be of such a nature, so as to effectively bypass 60 cycles.

Figure 1D. In order to avoid excess distortion, the plate and screen-grid voltages should automatically adjust themselves to different control-grid biases. This is affected by obtaining both of these voltages through a common series dropping resistor (R8), as illustrated in Fig. 1D.

It can readily be seen that the voltage at the junction of R6 and R7 as well as the junction of R7 and R8 will be a function of the drop in R8. The condenser, C2, removes the audio component from this D.C. voltage and provides the optimum plate supply voltage, which, in turn, likewise controls an optimum screen-grid supply voltage.

In order to provide a smooth-acting control, an additional resistor, R9, is added. With resistor R10, this provides a bridged-type control circuit.

Figure 1E. The circuit of Fig. 1D may be adopted where single-sided operation is desired. Fig. 1E, however, shows the push-pull derivation of our basic remote volume control circuit.

As the resistor, R4, is across a comparatively low voltage, and as very little current flows through the remote control, its lines may be extended indefinitely. It is unnecessary to shield these cables, regardless of their length, because of the hum filtering circuit employed.

AUDIO SPECTRUM CONTROL

The theory and operation of the Audio Spectrum Control was completely described in the December, 1937, issue of *Radio-Craft* (page 346), and in the June, 1938, issue (page 797).

Many music lovers utilizing amplifiers having both bass boost and cut control, have invariably complained of the production of "false bass" when boosting low frequencies. An analysis of this condition has disclosed the fact that the usual form of low boost controls *does not raise the fundamental frequency and all associated harmonics equally.*

Figure 2A. The reason for this condition is illustrated in Fig. 2A. Sloping line YZ typifies the usual bass boost response circuit. If F is some predetermined frequency, and 1H, 2H, and 3H, are its 1st, 2nd, and 3rd harmonics, respectively, it will be noted that the fundamental frequency is accentuated approximately 4 times as much as its 3rd harmonic, twice as much as its 2nd harmonic, and 1 1/3 times as much as its 1st harmonic. This change in the relative value of associated harmonics produces a decided change in the character of the fundamental tone.

As is well known, all middle A's in the standard pitch have the same fundamental

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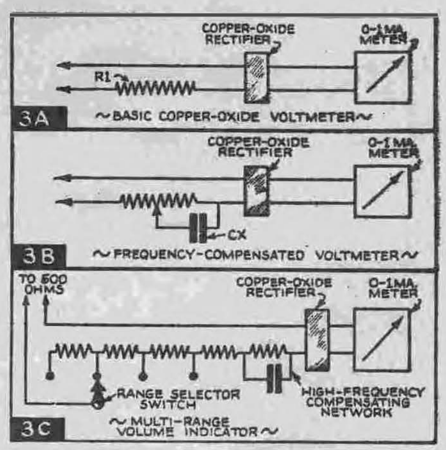
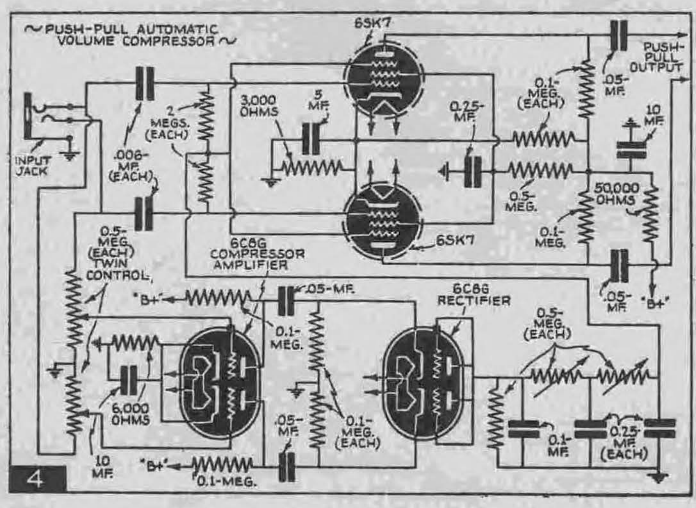
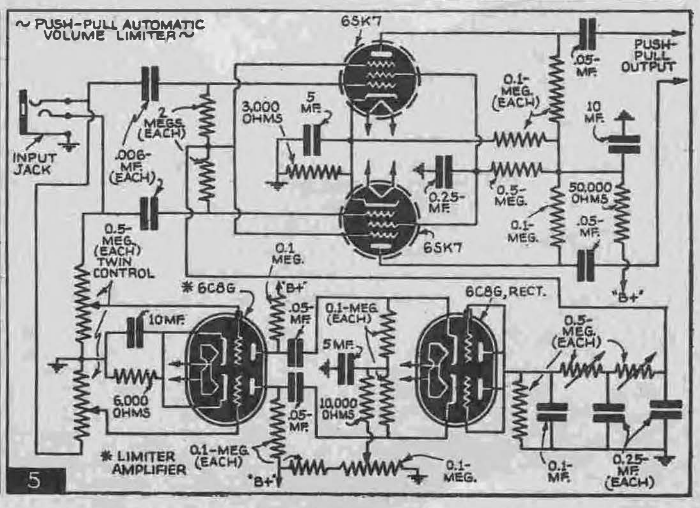
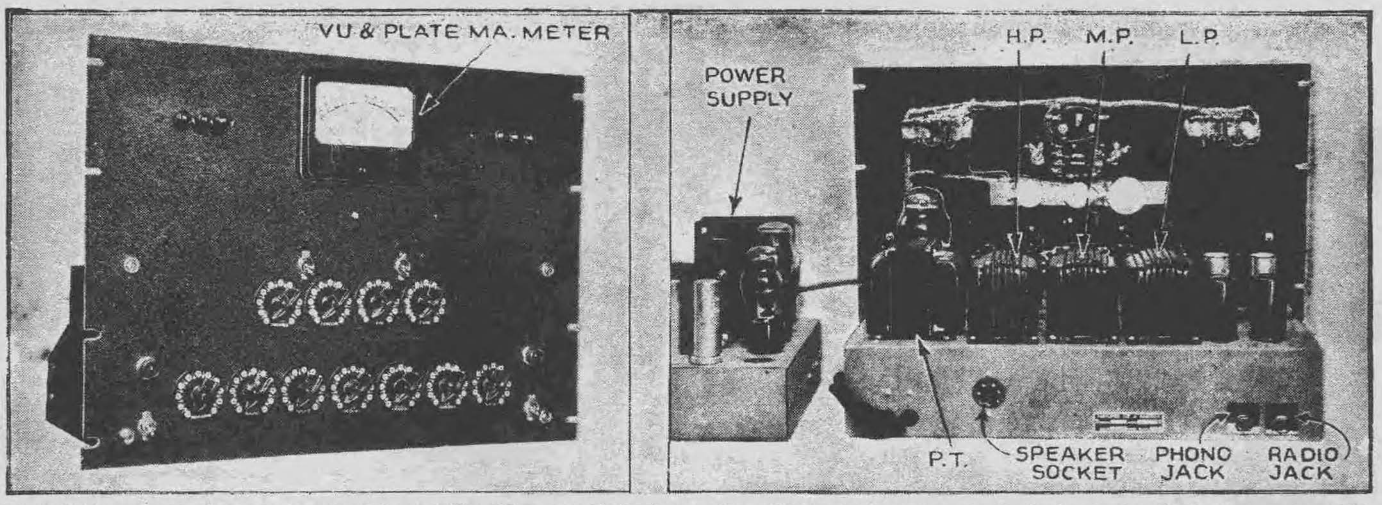








Front (left) and rear views of the Direct-Coupled Recording and Playback Amplifier. All the equipment in the front view is identified, reading from left to right, as follows: The pushbuttons at left of meter are marked P1, P2 and OFF; at right, OFF, +30 and +40. Immediately below the meter are the EXPANDER and SUPPRESSOR switches. Next, 4 controls that read THRESHOLD, SUPPRESSOR, TIMING, and AUXILIARY H.F. xtreme-lower-left, STANDBY pilot light and, below, STANDBY switch. Controls: PHONO, RADIO, LOW PASS, MEDIUM PASS, HIGH PASS and AUXILIARY L.F. Extreme right, MASTER pilot light and, below, MASTER on-off switch. The Auxiliary Controls are for *constant velocity* equalization; the Low, Medium and High Pass controls are for *constant amplitude* equalization.

RECORDING-PLAYBACK AMPLIFIER

Has Direct-Coupled Output Circuit and Equalized 30-Watt Output

A semi-technical discussion of an amplifier specifically designed to fill the requirements for the high-fidelity recording and playback field. A novel feature includes 2 independent equalizer circuits for both constant-velocity recording and constant-amplitude recording.

THE requirements involved in the design of a semi-professional high-fidelity Recording and Playback Amplifier offer a distinct challenge to the designing engineer, for here it is necessary to carefully balance performance against cost. If these considerations are not kept in mind during the design, the cost may soar, or the performance may be inadequate for universal application.

It was therefore decided to develop a medium-priced amplifier capable of providing more than sufficient flexibility of operation required to cover all unusual recording applications. The desirable features in a truly flexible amplifier of this type should include at least the following:

- Variable Equalizer for Constant-Velocity Recording
- Variable Equalizer for Constant-Amplitude Recording
- Variable Push-Pull Expander
- Variable Expander Delay
- Variable Time-Delay Control
- Non-Frequency-Discriminating Scratch Suppressor
- Feedback Push-Pull Output Amplifier (20 to 20,000 cycles, ±1 db.)
- Low Hum and Noise Level (–30 V.U.)
- Adequate Reserve Power (30 watts)
- Correct Level Indicator Meter
- Output-Stage Plate Current Meter

FEATURES—AND WHY

Variable Equalizer for Constant-Velocity Recording.—In *constant-velocity* recording, the amplitude of the recorded signal varies inversely with frequency. That is, higher frequencies have a lower amplitude swing. This condition is brought about by maintaining a constant velocity of the cutting needle. It is obvious, therefore, that when the number of oscillatory motions are increased, the amplitude must be decreased so that a constant distance is covered by the cutting needle within a given time.

This arrangement, however, is usually restricted at the lower frequencies (below 350 cycles, which is known as the "crossover frequency"), so as to avoid overcutting within low-frequency ranges. As most magnetic pickups (which is the type commonly employed for constant-velocity recording) have existing frequency response deficiencies which vary directly or inversely with frequency, it is advisable to employ a type of equalizer which will most rapidly compensate for such deficiencies.(*)

Variable Equalizer for Constant-Amplitude Recording.—In *constant-amplitude* recording, the velocity of the cutting needle is directly proportional to the recorded signal. Under these conditions, the amplitudes of all frequencies are recorded of equal magnitude.

Where equalization is desired for normal deficiencies in the recording head, recording blank, or associated equipment, it is desirable to equalize with a constant-amplitude equalizer which is nothing more than the "audio spectrum equalizer" familiar to *Radio-Craft* readers.(†) This type of equalizer accentuates or attenuates a given frequency band, equally.

Variable Time Delay.—In automatic volume expansion (A.V.E.) and automatic volume control (A.V.C.), the attack and release time, or *time controls*, of these circuits are of critical importance in order to avoid detrimental effects.

For example, in voice recording, it is imperative that the attack time be reduced so as to prevent clipping of initial syllables. For music, fast or slow selections may require individual adjustments for ideal results. Recordings that have both low *and* rapid tempos should have the timing control set for some arbitrary average.

Non-Frequency-Discriminating Scratch Suppressor.—The *scratch suppressor* is highly desirable in order to effectively reduce background recording noise. The circuit employed decreases scratch by 10 db.(**)

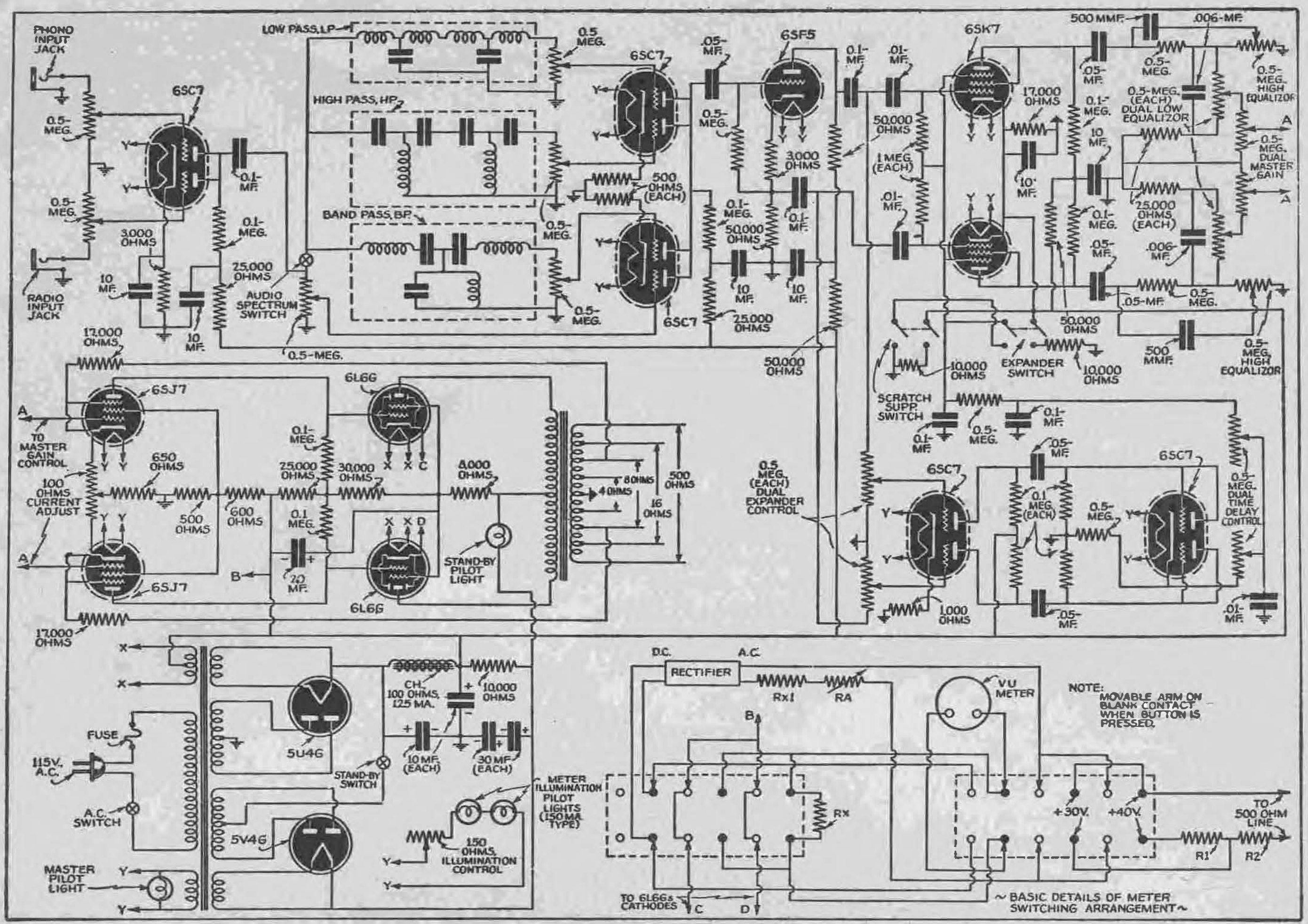
Feedback Push-Pull Output Amplifier.—The use of a balanced type of feedback provides for looping the output transformer, push-pull driver, and push-pull output stage. This *feedback circuit*, plus a generously-designed output transformer, provides for a flat frequency response from 20 to 20,000 cycles ±1 db., which exceeds the most stringent requirements of professional recording equipment. This type of a circuit also insures low hum, noise and distortion levels.

Adequate Reserve Power.—Although the amplifier is rated at 30 watts, it should normally be used as a 20-watt unit at which level it produces less than 1.2% *total* distortion. At recording levels up to +30 V.U., its distortion is practically unmeasurable!

Correct Level Indicator Meter.—This meter is of the new V.U. type designed in accordance with specifications set up by N.B.C., Bell Labs., and C.B.S.

(*)This type of equalizer has been completely described in the November, 1939, issue of **Radio-Craft**, pg. 268.
(†)Audio Spectrum Control was described in the December, 1939, issue of **Radio-Craft**, pg. 42.

(**)The operating principles involved were completely described in the July, 1940, issue of **Radio-Craft**.



Complete schematic diagram of the 30-W. Direct-Coupled Recording and Playback Amplifier.

It is equipped with a variable illuminating control so that the intensity can be adjusted for any given set of conditions without undue eye-strain when a program must be constantly and accurately monitored. In order to insure against output tube unbalance, a switching arrangement is employed whereby the meter is switched into the plate circuits of each of the output tubes for plate current balancing adjustments. The switching circuit is arranged so that the meter cannot be damaged when pushbuttons are indiscriminately operated.

ELECTRICAL CIRCUITS

A 6SC7 dual triode tube is employed for the phono and radio input. Electronic mixing is most easily accomplished in the manner indicated in the circuit diagram. For microphone work, an additional stage should be added. This may be a conventional 6SJ7. This tube should preferably have D.C. applied to its heater (use 12SJ7) which may be obtained from the common return of the high-voltage winding.

The components employed for the audio spectrum circuit follow conventional design. The audio spectrum is split up into the following 3 bands: (1) 20 to 200; (2) 200 to 2,000; and (3) 2,000 to 20,000 cycles. The audio spectrum switch enables the inclusion or exclusion of this portion of the circuit. Electronic mixing of the audio spectrum signals is accomplished through the following two 6SC7 tubes which are in turn coupled to a 6SF5 degenerative inverter. The output of the inverter in turn couples into a pair of 6SK7s which are used for expansion in conjunction with two 6SC7 tubes, one of which is used as a push-pull expanding amplifier and the other as a full-wave expander rectifier.

The direct-coupled portion of the amplifier follows conventional design which has been described in past issues of *Radio-Craft.*

For simplicity of circuit arrangements the values and the circuit of the complete V.U. meter have been omitted. The attenuator employed in the V.U. meter is of the "T" type. Resistors R1 and R2 represent "T" attenuators to provide any degree of attenuation desired. Resistor Rx is a shunt across the meter for changing the basic movement to read the high plate currents of the output stage. Resistor Rx1 is a series resistor of approx. 3,600 ohms and RA is a variable 500-ohm adjustment for absolute calibration of the V.U. meter.

The power supply is composed of 2 individual supplies. A 5U4G supplies bias voltage for the preamplifier, voltage amplifier and driver amplifier, while the 5V4G supplies voltage for the output stage. This latter voltage is added to the bias voltage in order to attain the high voltages required in the direct-coupled portion of the circuit. A standby pilot, in series with the plate supply of the output stage, insures against damaged output tubes should one of the driver tubes be thoughtlessly removed during operation. The power supply is built on a separate chassis to minimize inductive hum pick-up between the power transformer and the audio spectrum circuits.

Although this amplifier has been constructed for rack panel mount (its front panel measures 14½ x 19 ins.) it may be built in any standard cabinet for semi-professional use or home use.

RECORDING EQUALIZING TECHNIQUE

As there are a number of variable factors which tend to prevent the attainment of perfection in recording work, it is desirable to have a suitable number of compensating circuits to offset as many detrimental conditions as possible. If we assume that the amplifier itself will not introduce any frequency discrimination, we have the following additional elements to consider:

(1) Microphone—its type, response, and placement.

(2) Cutting Head—its type and associated driving mechanism.

(3) The Recording Material—its type (degree of hardness) and its surface cutting speed.

As an infinite variety of response variables may be introduced by the 3 elements mentioned it is naturally desirable to locate and classify major frequency discrepancies within the recording system. This task is comparatively simple when modern laboratory facilities are available. For the recording enthusiast, however, who has limited facilities, the relative quality of the finished record may easily be checked against its original sound source, and noticeable discrepancies can easily be detected by trained listeners.

If the recorded signal can not be easily differentiated from the original signal, it is safe to assume that no noticeable discrimination occurs within the entire reproducing system. If noticeable discrimination is present, various "shades" of equalization (utilizing either the constant-velocity or the

(Continued on page 32)

A New A.F.-Drift Correcting, Signal-Balancing, Direct-Coupled

F.M. AUDIO AMPLIFIER

This circuit achieves remarkable results (Frequency Response—13 to 30,000 cycles ± 1db.; Noise Level—at least 75 db. below rated power output; Distortion—1% total harmonics at average working level). It includes a novel D.C. balancing arrangement, A.C. balancing circuit and push-pull balanced feedback, all of which provide marked reduction in tube noise and hiss, and a wide range response, as well as sufficient clean power output to provide distortionless high- and low-frequency amplification beyond *requirements set for F.M. transmitters.*

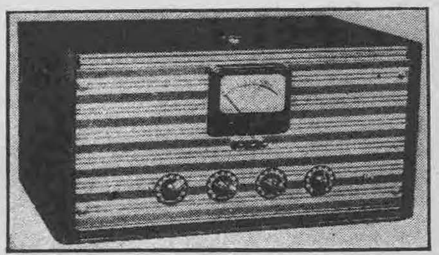
PART I

FOREMOST among problems presented by Frequency Modulation is the design of an amplifier which will not prove to be the "bottle neck" of the entire system. The new standards set by the Federal Communications Commission for designing F.M. transmitters, that should be taken into consideration when designing an audio amplifier for F.M. receivers, briefiy follow:

(1) The transmitter and associated studio equipment shall be capable of transmitting a band of frequencies from 50 to 15,000 cycles within 2 decibels of the level of 1,000 cycles. In addition, provision shall be made for pre-emphasis of the higher frequencies in accordance with impedance frequency characteristics of a series inductance - resistance network, having a time constant of 100 micro-seconds.

(2) The noise in the output of the transmitter in the band 50 to 15,000 cycles shall be at least 60 decibels below the audio frequency level represented by a frequency swing of 75 kilocycles (100% modulation).

(3) At any frequency between 50 and 15,000 cycles at a swing of 75 kilocycles the combined audio frequency harmonics generated by the transmitting system shall not be in excess of 2% (root mean square value). This means, simply, that the transmitter should be capable of passing a band of 50 to 15,000 cycles ±2 db. of the 1,000-cycle reference; it shall have a combined hum and noise level at least 60 db. below full power output; and, it should not generate more than 2% total harmonics at any frequency within its transmitted band.



The completed Frequency Modulation Audio Amplifier. Controls, left to right: Radio Volume, Phono Volume, H.F. qualizer, L.F. Equalizer. On the right side of the controls is the On-Off switch; on the left side, the pilot light. The 3 pushbuttons underneath the VU Meter are for the Meter for Attenuator Ranges.

F.M. A.F. AMPLIFIER STANDARDS

In setting up standards for an F.M.-receiver audio amplifier the natural reaction would be to use the standards set for the F.M. transmitter. Careful consideration, however, will reveal specific disadvantages for such an arrangement.

It is obvious that for ideal performance, the amplifier at the receiving end should have an effectively flat frequency response, introduce no distortion and have no inherent noise. With such an ideal amplifier, the *full* benefits of frequency modulation will be obtained.

Any discriminating characteristics inherent within the receiving amplifier will, of necessity, introduce additional detrimental conditions, which are added to existing deficiencies within the transmitter to provide an overall result far below a desirable ideal. For example, let us assume that the transmitter is down 2 db. at 50 cycles. The receiving amplifier (which was built in accordance with the standards set for F.M. transmitters) is also down 2 db. at 50 cycles. The overall result will be a 4 db. loss at this low frequency, which is sufficient to change the character of many types of music. Similarly, an amplifier which introduces 2% distortion (say at an average level of 1 watt) will provide an ultimate program having a combined distortion of more than 2% (which we can assume was produced by the transmitter). It therefore follows that the amplifier should be definitely better than the transmitter.

In addition to this, it is also feasible to assume that additional improvements will be made in F.M. transmitters, and F.C.C. regulations may tighten their specifications. If this occurs, an amplifier which has been built to existing standards may not pass on to the listener all the benefits of future improvements in F.M. transmission. ***The present specification covering the width of the audio band is unbalanced,**** and it is reasonable to assume that, in time, the lower portion of the band will ultimately be extended to at least 26 cycles to produce a balanced spectrum.

Proof of this line of reasoning can be found in new F.M. transmitters, which are being constructed to exceed the F.C.C.'s F.M. requirements. For example, one of

*See "Balanced Audio Spectrums," *Radio-Craft*, Sept., 1940, pg. 164.

A Letter from the Author

Dear Editor:

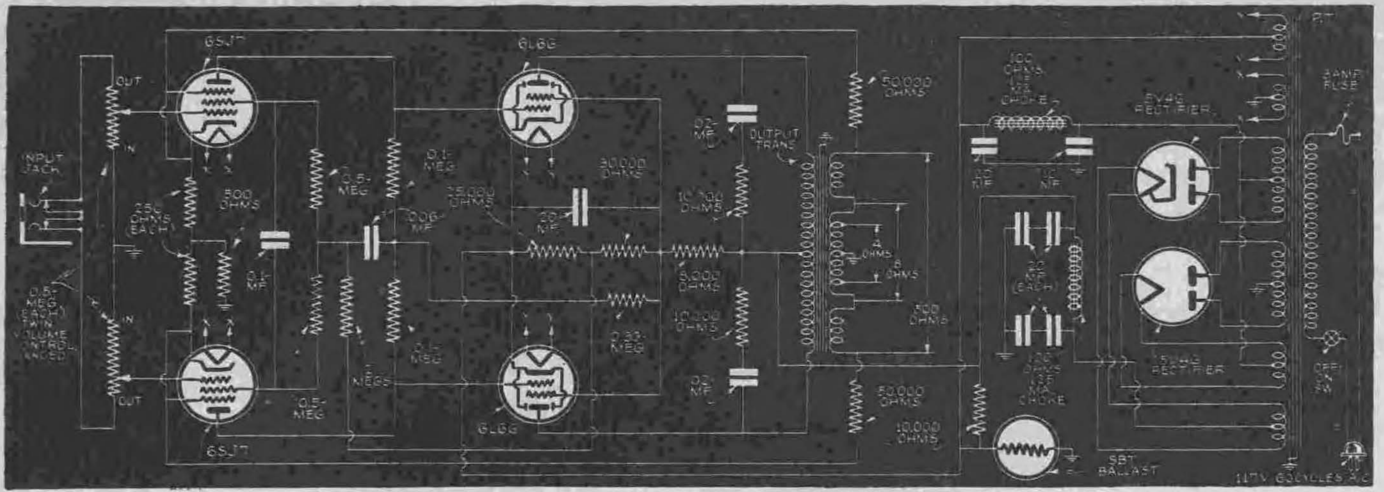
The development of this stabilized push-pull Direct-Coupled Frequency Modulation Amplifier has convinced all technicians who have studied, and checked the performance of the circuit, that we have finally removed the last obstacle for universal application of Direct-Coupled Amplifiers. In fact, our development (patent applied for) has over-shot our desire to make the stability of this model at least equal to standard resistance-coupled circuits.

In a conventional push-pull resistance-coupled amplifier, signal unbalance between each side of the circuit is carried through and finally cancelled in the output transformer. This condition introduces an unbalanced push-pull action and is usually encountered to a varying degree, in all standard resistance-coupled amplifiers. In our attempt to balance the amplifier for variations of plate current in push-pull tubes, we found that we had also developed a circuit which would stabilize for variations in tube gain. The revolutionary circuit arrangement provides for balancing of the signal circuit in the preamplifier stages long before it reaches the output transformer.

Aside from the advantages gained by an extended frequency response range, and very low noise and hum levels, this A.C. balancing circuit makes this general type of amplifier far superior to any standard resistance-coupled unit.

A. C. SHANEY

P.S.—Although this particular amplifier was designed for F.M. applications, your readers should not construe this as limiting the application of the unit for this purpose only. Its exceptionally fine response, low noise level, and no effective distortion, makes it admirably adapted for any other application which would normally require a high-quality laboratory amplifier.



Schematic circuit of the Push-Pull Direct-Coupled Frequency Modulation A.F. Amplifier. It incorporates balanced negative-feedback and novel A.C. and D.C. balancing circuits.

the largest manufacturers of transmitters guarantees the following audio characteristics:

(1) Frequency Response—Flat ±1 db. from 30 to 15,000 cycles.
(2) Noise Level—70 db. below full modulation.
(3) Distortion—Less than 2%, total harmonics.

It was therefore decided to anticipate a reasonable amount of improvement and design this F.M. amplifier so as to prevent obsolescence. The following tentative specifications were set:

(1) Frequency Response—±1 db. from 13 to 30,000 cycles.
(2) Noise Level—At least 75 db. below rated power output.
(3) Distortion—1% (at average working level), total harmonics.

With an amplifier of this type, it was felt no ultimate consumer would ever have to worry about having the "bottle neck" of an F.M. program in his audio amplifier equipment.

Furthermore, reasonable improvements in F.M. transmitters (based on similar improvements which have taken place in A.M. work) will provide direct benefits to the listener.

SELECTING THE FEATURES

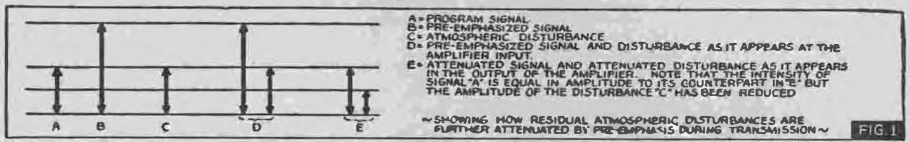
The Equalizer

Offhand, it would appear that an F.M. Amplifier should be built to meet ideal requirements and have unvarying characteristics. In other words, the amplifier should be devoid of high-frequency or low-frequency controls. Referring to the requirements set by the F.C.C., it will be noted that provision *must* be made in every F.M. transmitter to pre-emphasize high frequencies. This means that high frequencies will be accentuated during transmission. The purpose of this pre-emphasis is to attenuate residual atmospherics.

As disturbing effects of atmospherics are predominant in the higher audio frequencies, it is logically assumed that accentuation at the transmitter and attenuation at the receiver will ultimately result in a flat overall response and at the same time, materially attenuate atmospherics. This is graphically illustrated in Fig. 1.

If we assume that a high-frequency program signal has a level of +20 VU and it is pre-emphasized to a level of +23, this signal will be received along with an atmospheric disturbance of say +20. Hence, without pre-emphasis, the original program signal and the atmospheric will be of equal intensity. On the other hand, pre-emphasis has already made the program signal appreciably higher than the atmospheric. By attenuation in the receiver, the program signal is brought back to its original level of +20 VU, and the atmospheric is reduced 3 VU. The degree of attenuation of disturbances is a function of the pre-emphasis at the transmitter.

From a casual study of this operating procedure, it would appear that a high-frequency attenuator is the only required control of the receiver. A study of existing deficiencies in present records, however, will



~SHOWING HOW RESIDUAL ATMOSPHERIC DISTURBANCES ARE FURTHER ATTENUATED BY PRE-EMPHASIS DURING TRANSMISSION~ FIG. 1

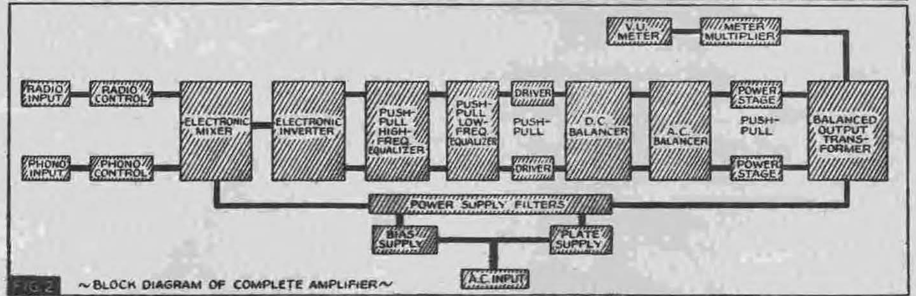


FIG. 2 ~BLOCK DIAGRAM OF COMPLETE AMPLIFIER~

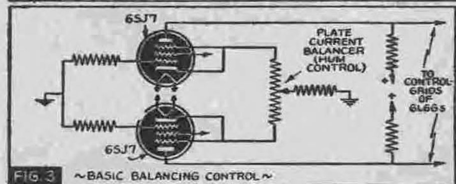


FIG. 3 ~BASIC BALANCING CONTROL~

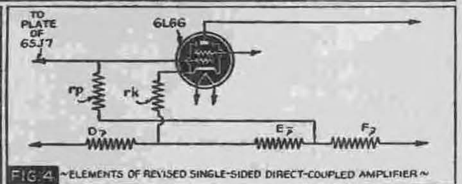


FIG. 4 ~ELEMENTS OF REVISED SINGLE-SIDED DIRECT-COUPLED AMPLIFIER~

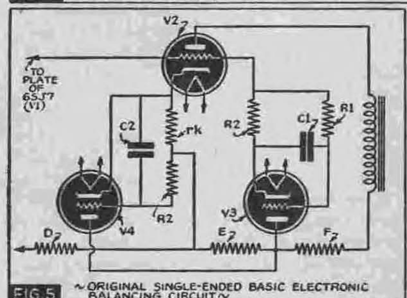


FIG. 5 ~ORIGINAL SINGLE-ENDED BASIC ELECTRONIC BALANCING CIRCUIT~

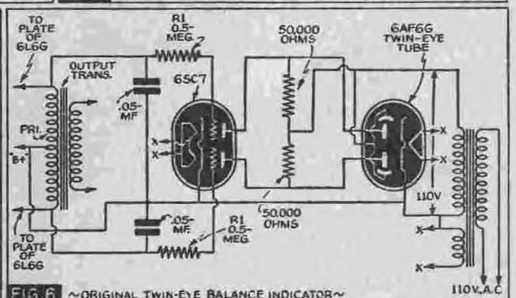


FIG. 6 ~ORIGINAL TWIN-EYE BALANCE INDICATOR~

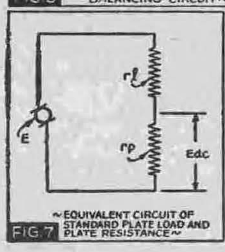


FIG. 7 ~EQUIVALENT CIRCUIT OF STANDARD PLATE LOAD AND PLATE RESISTANCE~

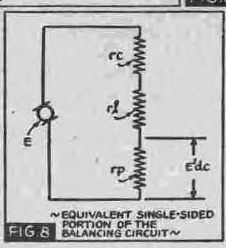


FIG. 8 ~EQUIVALENT SINGLE-SIDED PORTION OF THE BALANCING CIRCUIT~

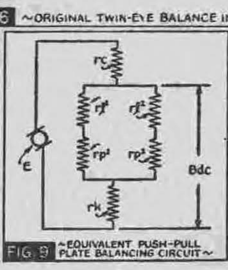


FIG. 9 ~EQUIVALENT PUSH-PULL PLATE BALANCING CIRCUIT~

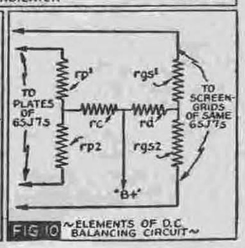


FIG. 10 ~ELEMENTS OF D.C. BALANCING CIRCUIT~

clearly indicate that both the high and low frequencies should be independently controlled, and the control range should provide for both attenuation *and* accentuation. Another very desirable characteristic in the equalizer circuit is to have it exactly complement the equalizer used at the transmitter or in the recording studio (for recorded programs). The equalizer should not introduce harmonics, hum, or resonant peaks in any portion of the spectrum.

The VU Meter

It was also considered desirable to have a visual monitoring arrangement so as to indicate normal, average, and peak levels of the program. This auxiliary feature is highly desirable when it is required to avoid overload of either the amplifier or the loudspeaker. Low-frequency speaker overload is usually judged from a distortion viewpoint, because the intensity of the signal cannot be accurately judged in view of the fact that the ear is comparatively insensitive to low frequencies. Only critical listeners, therefore, will detect overload at low frequencies. The use of the meter, however, makes it possible for any average individual to adjust the intensity of the program level so as to definitely prevent overload at any frequency. Furthermore, it becomes relatively simple to detect just what actual effect the various settings of the equalizer controls have upon the overall program level.

Dual-Channel Input and Electronic Mixer

In order to extend the usefulness of this Direct-Coupled F.M. Amplifier, it was considered desirable to incorporate an additional input circuit so that phonograph records, in addition to F.M. transmissions, may also be enjoyed.

A dual circuit input could most economically be employed by the use of a changeover switch, but inasmuch as the average volume level of the radio program and the recorded program may be different (and therefore necessitate a continual change), it was thought more desirable to incorporate an electronic mixer. This provides 2 entirely independent input channels with independent controls so that each level may be set for ideal results. Furthermore, the use of the electronic mixer insures complete isolation of both controls, so that they do not affect either the volume or the frequency response characteristics of its associated channel.

Details covering the design of these 3 features will be described in Part II of this article. A block diagram which shows the relative position of the various features is given in Fig. 2.

THE AMPLIFIER

In order to more fully understand the advanced design principles incorporated in this unusual Direct-Coupled F.M. Amplifier, it is suggested that the reader refer to the previously-published data.*

As all of the several 10-, 20- and 30-Watt Direct-Coupled Amplifiers previously described in this magazine have been designed around an effective drift-correcting circuit, no immediate improvement in stability seemed apparent. Subsequent investigation, disclosed that *unusual* difference in plate resistances of the input tubes affected the performance of direct-coupled amplifiers more than resistance-coupled units. This difference in effect was to be expected to be noticeable because of the increased efficiency, improved response, and lower noise level characteristic of direct-coupled amplifiers. Upon further investigation, it was found that manufacturers of tubes had not set close standards for plate resistance of preamplifier and voltage amplifier tubes.

Although normal variations in tubes produce a measurable difference in the performance of the resistance- and transformer-coupled amplifiers, they have been found to produce another effect in direct-coupled amplifiers. For example, an unbalanced pair of input tubes would unbalance the plate current of the output tubes sufficiently to increase residual hum and require readjustment of the hum-balancing adjustment. It was therefore decided that 2 self-correcting networks would be incorporated in this new amplifier; one to automatically balance for difference in the plate resistance of the driver tubes and the other to automatically balance for difference in gain of the driver tubes. As a further requisite, it was decided that these circuits should provide for superior results in the direct-coupled amplifier as compared to a standard resistance-coupled amplifier with a given set of greatly unbalanced (or even defective) tubes.

THE D.C. BALANCING CIRCUIT

During the development of the 30-Watt Direct-Coupled Amplifier,* it was found that a normal variation between tubes could be compensated-for by correcting the bias on the input tubes. The basic portion of this manual balancing circuit is illustrated in Fig. 3.

Fortunately, when an unbalance of more than 10 milliamperes occurred in the output stage, the hum level came up. It therefore became a relatively simple matter to balance the input tubes by adjusting for minimum hum. With a change of input tubes, it was sometimes necessary to readjust the initial setting. It was found, however, that some of the ultimate users of these amplifiers would insert greatly-unbalanced tubes, without attempting to readjust for balance. It was therefore believed highly desirable to provide some automatic means for balancing. The first method of attack which presented itself was to use a tube in place of the load resistance of the voltage amplifier and arrange for automatic compensation for variations in plate resistance of the voltage amplifier. Another tube was to be used to augment the bias of the output tubes, so as to compensate for variations in output plate current.

In Fig. 4, which shows the elements of a revised single-sided direct-coupled amplifier, Rp is the plate resistance of the voltage amplifier and Rk is the partial cathode resistor of the power amplifier.

Figure 5 shows the basic balancing circuit originally conceived to automatically compensate for both variations in plate resistance of the input tube V1 and the output tube V2. It will be noted that V3 is used as a plate load resistor for V1. The bias applied to V3 through R1 depends upon the plate current flowing through its cathode resistor R2. The time-delay constant of R1, C1, prevents signal frequencies from affecting a change in the plate resistance of V3, and limits automatic adjustments only for "steady state" or average conditions; V4 was to be used as a shunt across Rk, so as to keep the bias across Rk constant. This circuit is likewise made responsive only to steady state or average unbalance, by inserting a time lag through the resistor-condenser network R2-C2.

Inasmuch as the final amplifier was to be push-pull throughout, 4 additional tubes would be required for this balancing action. The added expense and complexity of this circuit inspired additional research to produce a simpler and more economical circuit to achieve the desired results.

A side project was started to adapt the use of the twin indicator (6AF6G) through a twin-triode amplifier (6SC7), so arranged as to measure the voltage drop across the balanced primary winding of the output transformer. A special transformer was wound so that both sides of the primary were of equal D.C. resistance (and equal A.C. impedance). The idea behind this development was to provide a partially visual check on the plate current of the output tubes so that should greatly unbalanced tubes be used, it would become immediately visible, and the tube would provide for readjustment. It was found, however, that the indicator with its associated amplifier was too insensitive for the average user to adjust within a 10-ma. balance. This circuit was therefore abandoned, but it is given in Fig. 6 for the benefit of some readers who may have other applications for this particular type of indicator. The condenser-resistor network R1-C1 provides a time delay to prevent A.C. potentials from having any effect upon the twin-eye indicator. A novel portion of the circuit is that raw A.C. is applied to the plates of the indicator. The flicker is not observed because of the persistence of vision of the eye which will tolerate interrupted images down to about 16 cycles before flicker becomes visible.

The easiest way to understand the action of the final D.C. balancer is to substitute a resistor (rl) for the plate load and another (rp) for the plate resistance of the tube. If a D.C. voltage E (as indicated in Fig. 7) is applied across this network, the voltage Edc is the effective voltage applied to the plate of the tube and is dependent upon the voltage drop across rl. Thus, if rp is varied from zero to infinity, the voltage will vary proportionately. The ratio of voltage change will depend upon the ratio of $\frac{rp}{rl + rp}$. If rl is made large in comparison to rp, the ratio of change will be small. If an additional resistor (rc) is inserted in series with both rl and rp, as indicated in Fig. 8, then the effective voltage E'dc would be equal to $\frac{rp}{rc + rl + rp}$. The push-pull version of this circuit is indicated in Fig. 9. If we neglect rk (which is very small) the voltage which appears across rl' + rp', is equal to Bdc which can be calculated from

$$Bdc = \frac{\dfrac{(rl^1 + rp^1)\,(rl^2 + rp^2)}{rp^1 + rl^1 + rp^2 + rl^2}}{rc + \dfrac{(rl^1 + rp^1)\,(rl^2 + rp^2)}{rp^1 + rp^2 + rl}}$$

If rl^1 is 100,000, rc is 500,000, and rp^1 varies from 800,000 to 120,000 (which represents a ± variation of approx. 20%), it will be found that the percentage of change at Bdc is 1.9% as compared to a 4% change which would take place under conditions of Fig. 7. In other words, a 50% correction is affected. If the same type of network is applied to the screen-grids of the driver tubes, as indicated in Fig. 10, still more correction is affected.

The practical value of this self-balancing circuit can best be indicated by referring to laboratory data compiled during its development. A total of 100 average 6SJ7 tubes were checked for the maximum deviation they produced in the output plate circuit of the 6L6G's. Two sets of the worst combination produced the following results:

(Continued on page 32)

A New A.F.-Drift Correcting, Signal-Balancing, Direct-Coupled

F.M. 24-WATT AUDIO AMPLIFIER

This wide-range amplifier, which was partially described in a preceding issue, incorporates a new A.C.-D.C. balancing circuit. It is the perfect auxiliary for use with Frequency Modulation tuners, as it passes 13 to 30,000 (±1 db.) cycles, and has a noise level of 75 db. below rated output. Distortion is only 1% total harmonics at average working level! Push-pull balanced negative-feedback is incorporated.

PART II

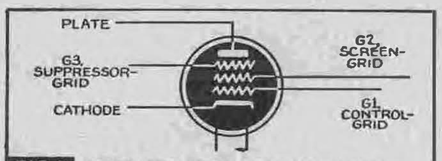


FIG. 1 ~FUNDAMENTAL SCREEN-GRID ELEMENTS~

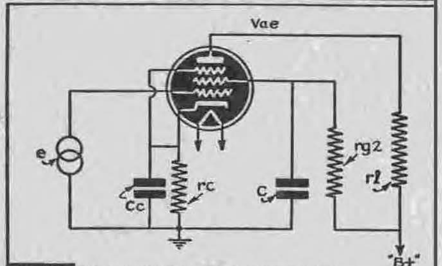


FIG. 2 ~ STANDARD PENTODE CIRCUIT~

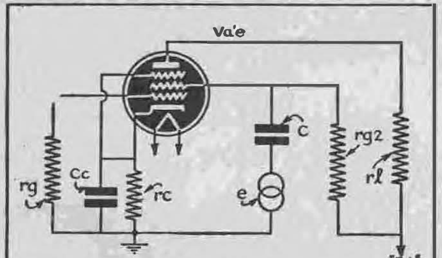


FIG. 3 ~SCREEN-GRID USED AS A CONTROL-GRID~

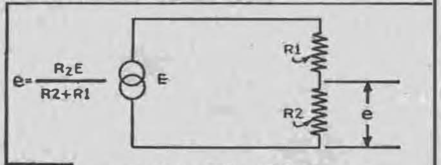


FIG. 4 ~BASIC DIVIDER NETWORK~

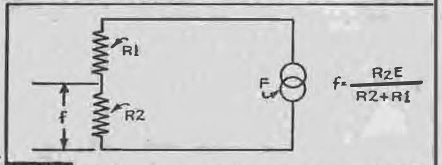


FIG. 5 DIVIDER NETWORK OPPOSITELY PHASED TO FIG.4 CIRCUIT

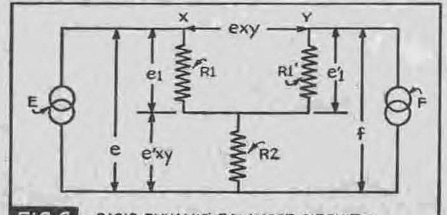


FIG. 6 ~BASIC DYNAMIC BALANCER CIRCUIT~

IN an effort to surpass the stability of conventional transformer- and resistance-coupled amplifiers, a combined D.C.-A.C. balancing circuit was developed. The *D.C. balancer* provides for automatic audio-drift correction under static conditions. The *A.C. balancer* provides for automatic signal balancing under dynamic conditions.

Stated in other words, the static balancer (D.C. corrector) automatically compensates for a very wide variation in plate resistance characteristics of input tubes. It prevents unbalance in the output stage with change of emission characteristics within the input stage. The dynamic balancer (A.C. corrector) automatically compensates for a very wide variation in voltage amplification of input tubes. It prevents the application of unbalanced signals to the control-grids of the output stage.

I think the dynamic and static balancing circuit described in this article is one of the most interesting circuits I've ever developed. It shows what unconventional things can be done with conventional tubes. This circuit should find wide application in all standard all-push-pull resistance-coupled circuits as well as direct-coupled jobs. What it does for hum reduction and extension of the audio range is almost unbelievable.

A. C. SHANEY.

THE DYNAMIC BALANCER (A.C. CORRECTOR)

The easiest way to understand the operating principles of this unusually effective circuit is to analyze the basic operating principles of the screen-grid tube. This tube is normally used in a conventional manner, i.e., by applying a control voltage to the control-grid; a "B+" voltage, adequately bypassed to the screen-grid; a "B+" voltage, through the load resistor to the plate; and, its suppressor-grid connected to cathode.

If these elements are viewed fundamentally as diagrammed in Fig. 1, it will be noted that all of the grids are in the electron stream. This means that any one of them can be used as a control-grid. Naturally, the further away the grid is from the emitter (cathode), the less control it has upon the electron stream. If the grids are labelled G1, G2, and G3, in order of their distance from the cathode, these notations will correspond to control-grid, screen-grid, and suppressor-grid, respectively.

Figure 2 shows a standard circuit, wherein the input signal (e) is applied to the control-grid, a signal Vae will appear at the plate. This voltage will be out-of-phase with the input signal. If the screen-grid bypass condenser, C, is disconnected, and the voltage e is applied in series with the condenser, as illustrated in Fig. 3, a voltage Va′e will appear on the plate. Va may be defined as the control-grid to plate voltage amplification. Va′ may also be defined as the screen-grid to plate voltage amplification. It is therefore obvious that the screen-grid can be used as a control-grid. This particular application is important as it plays a prime role in our dynamic balancer.

The suppressor-grid may likewise be used as the control-grid in which case, the voltage which appears at the plate would be equal to Va″e (wherein Va″ may be looked upon as the suppressor-grid to plate voltage amplification).

PRINCIPLES OF DYNAMIC BALANCING

With the above phenomenon kept in mind, a review of fundamental balancing circuits will further simplify the operating principles of the dynamic balancer. If an A.C. generator E is applied to a series resistive network R1, R2, as illustrated in Fig. 4, the voltage (E) appearing across R2 is equal to $\frac{R2E}{R2 + R1}$. Expressed mathematically, it becomes

$$e = \frac{R2E}{R2 + R1}.$$

If another identical generator F is connected to a similar resistance network R1, R2, the voltage (f) which appears across R2 is likewise equal to $\frac{R2E}{R2 + R1}$. If both circuits of Figs. 4 and 5 are connected together, so that R2 becomes a common return, Fig. 6 results. If the generators E and F are so adjusted as to be equal in potential but opposite in phase, and $R_1 = R_1'$, the following voltage conditions are present:

(1) The voltage across E (e) is obviously equal to the voltage across F (f).

(2) The voltage across R_1 (e_1) is equal to the voltage across R_1' (e_1').

(3) As the voltages are out-of-phase, it is also obvious that the voltages across R2 will cancel, and equal 0.

(4) The voltages across X and Y (e xy) will also cancel and be equal to 0.

The above conditions are prevalent only when the generators are opposite in phase and of equal potential. If we assume, however, that one of these generators drops in voltage, let us say to 50%, of its original value, it is apparent that the total difference will be equal to e—f or e xy. With an unbalance in the generators it is further apparent that complete cancellation will not occur across R2. In fact, some of the larger voltage will appear at this point. This voltage (e′ xy) is equal to

$$E \frac{R2}{R2 + R1} - F \frac{R2}{R2 + R1} = e'xy$$

An examination of this formula shows that as R2 is increased, more of the unbalanced voltage appears across it. If this voltage unbalance (e′ xy) is applied back to F, so as to increase its voltage output, it is obvious that some balance will automatically be obtained.

THE DYNAMIC PLATE BALANCER

How this is actually done in the amplifier can best be indicated by redrawing Fig. 6 and replacing E and F by their respective tube circuits, as indicated in Fig. 7. In this circuit, the push-pull generator EF, takes the place of the original generators E and F. R1 becomes the independent plate loads of both tubes, while R2 becomes the common degenerative resistor. If both tubes A and B have identical voltage amplification characteristics, the voltage which appears across R2 will be 0. On the other hand, if A has twice the voltage amplification of B, then a portion of this difference will appear across R2.

A typical example is given in Fig. 8, wherein the plate load resistors R1 equal 100,000 ohms each, the common degenerative resistor R2 is equal to 400,000 ohms. If we assume that the voltage amplification of one tube (A) is 20, and the other tube (B) is 10, and if a balanced push-pull signal (grid-to-grid of 2 volts) is applied to the input of the circuit, the voltage which appears at the plate of A is equal to say, +20 volts (the voltages indicated are instantaneous A.C. voltages). The voltage which appears at the plate of B is equal to —10 volts. If these signals are out-of-phase, there will be a total voltage difference between both plates of 30 volts (for ideal conditions, there should be a total voltage difference of either 20 volts [if both plates have 10 volts each] or 40 volts [if both plates have 20 volts each]).

The portion of the voltage developed by plate A which appears across the 400,000-ohm resistor, is equal to

$$+20 \times \frac{400,000}{400,000 + 100,000} = 20 \times \frac{4}{5} = +16 \text{ volts}$$

The portion of the voltage developed by plate B which appears across the 400,000-ohm resistor is equal to

$$-10 \times \frac{400,000}{400,000 + 100,000} = -10 \times \frac{4}{5} = -8 \text{ volts}$$

The cancellation which occurs across the 400,000-ohm resistor is equal to 16 — 8 or + 8 volts. This instantaneous value of + 8 volts is fed back to the screen-grids of both tubes to affect further automatic connection. Before considering the balancing action of this voltage, let us briefly look into the screen-grid circuit.

THE DYNAMIC SCREEN-GRID BALANCER

The fundamental principles involved in the dynamic screen-grid circuit are virtually identical with those for the plate dynamic balancer. There are, however, 2 important exceptions.

In our conventional circuit of Fig. 2, it will be noted that the screen-grid was bypassed to ground through C. If this condenser is entirely removed, a voltage will appear at the screen-grid, which is equal to Va‴e (Va″ being the control-grid to screen-grid voltage amplification). If the rest of the circuit of Fig. 2 remains unchanged, it will be found that the voltages Vae and Va‴e will be in-phase. The voltage Vae however will be decreased. This is caused by the degenerative action of the voltage which appears at the screen-grid. Its degenerative action can best be analyzed by referring again to Fig. 1. If a positive instantaneous voltage is applied to G1, the electron stream is increased. The increased current through G2, produces a drop across its supply resistor. This, in turn, decreases the applied potential of C2 to retard the flow of electron streams to the plate. As the control-grid to screen-grid voltage amplification increases, the control-grid to plate voltage amplification decreases. Very large signals can easily be handled by the screen-grid under this condition.

Figure 9 shows the elements of the dynamic screen-grid balancer circuit, arranged to simulate the plate dynamic balancer of Fig. 8. It will be noted, however, that an essential difference is the inclusion of the condenser Cg2. If the control-grid to screen-grid characteristics are identical in both tubes, complete cancellation of the voltages which appear at both screen-grids will take place, as discussed for the conditions illustrated in Fig. 6. Let us assume for a moment however, that the control-grid to screen-grid characteristics of tube B, are lower than that of A. This naturally means that complete cancellation will not take place across both screen-grids and a residual potential will appear at the screen-grid of tube A. This voltage will then drive the screen-grid of B in a very conventional resistance-coupled circuit, which can easily be perceived by redrawing Fig. 9, as indicated in Fig. 10.

Here it will be noted that the screen resistor R3 of tube A acts as an equivalent "plate load". Condenser Cg2 assumes the role of the common coupling condenser. The screen-grid of B acts as the control-grid. The voltage which appears at the plate of B will be out-of-phase with that which appears at the plate of A because of the following reasons: when the control-grid of tube A is being used as a driver, and it becomes instantaneously positive, both the plate and its screen-grid become instantaneously negative. The negative screen-grid of tube A is coupled to drive the screen-grid of tube B negatively. This in turn produces an instantaneous positive potential at the plate B.

With correct selection of values, this circuit may be made to operate as a perfect inverter, and shows how complete balancing may be attained even though the control-grid of tube B is entirely inoperative. In actual practice, however, such a condition is rarely encountered. What usually happens is the control-grid to screen-grid voltage amplification of both input tubes are not always equal. This coupling circuit equalizes the difference within the first stage so that practically equal but oppositely-phased voltages appear at the push-pull output plates of A and B.

In addition to the dynamic screen-grid balancer and the dynamic plate balancer, there is an auxiliary regenerative balancer which comes into play when the common coupling resistor of the plate supply, R2 is coupled to the common coupling resistor of the screen supply, R4, through condenser Cg3, as indicated in Fig. 11. If we redraw this schematic again so as to take the form of a more familiar coupling circuit, we have Fig. 12. Here, it will be noted, the full potential difference which appears across R2 (400,000-ohm resistor of Fig. 8) is applied through Cg3 and through both R3 resistors directly to the screen-grids of both tubes. If we assume that the control-

(Continued on page 32)

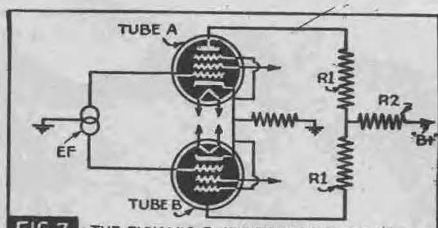


FIG. 7 ~THE DYNAMIC PLATE BALANCER CIRCUIT~

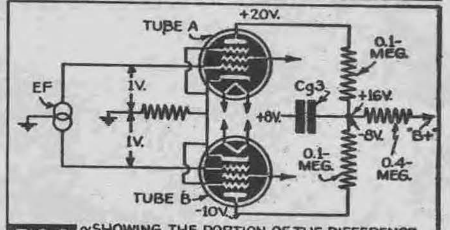


FIG. 8 ~SHOWING THE PORTION OF THE DIFFERENCE VOLTAGE WHICH APPEARS AT Cg3~

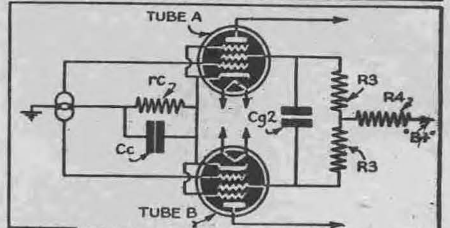


FIG. 9 ELEMENTS OF THE DYNAMIC SCREEN-GRID BALANCER

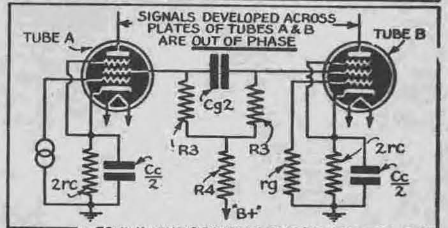


FIG. 10 ~EQUIVALENT CIRCUIT OF FIG. 9 (ONE CONTROL-GRID RECEIVES NO SIGNAL)~

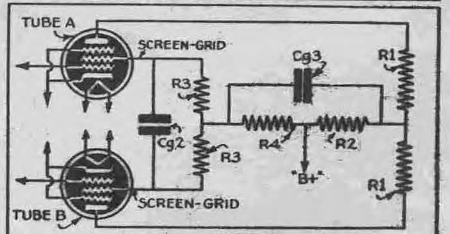


FIG. 11 ~THE AUXILIARY REGENERATIVE EQUALIZER. DIFFERENTIAL VOLTAGE ACROSS R2 DRIVES BOTH SCREEN-GRIDS THROUGH CONDENSER Cg3~

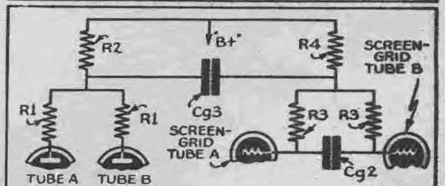


FIG. 12 ~SIMPLIFIED VERSION OF THE DIFFERENTIAL DRIVER CIRCUIT OF FIG. 11~

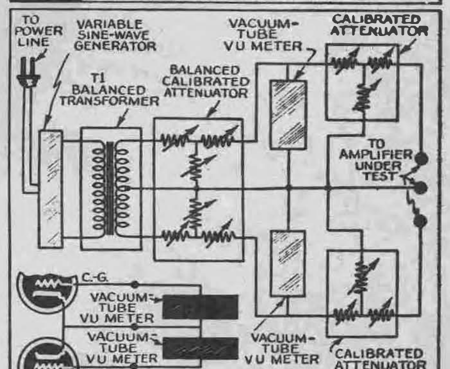


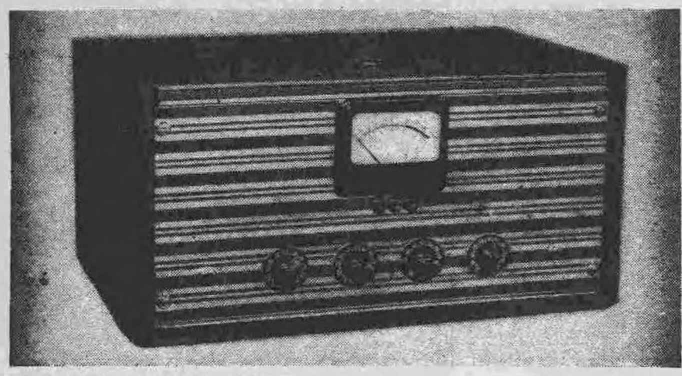
FIG. 13 ~BLOCK DIAGRAM OF LABORATORY SET-UP TO CHECK DYNAMIC BALANCING CIRCUIT~

A New A.F.-Drift Correcting, Signal-Balancing, Direct-Coupled

F.M. 24-WATT AUDIO AMPLIFIER

This wide-range amplifier, which was partially described in a preceding issue, incorporates a new A.C.-D.C. balancing circuit. It is the perfect auxiliary for use with Frequency Modulation tuners, as it passes 13 to 30,000 (±1 db.) cycles, and has a noise level of 75 db. below rated output. Distortion is only 1% total harmonics at average working level! Push-pull balanced negative-feedback is incorporated.

PART III



Appearance of the completed F. M. Amplifier. This novel amplifier, designed at the suggestion of R. D. Washburne, employs a basically new dynamically and statically balanced direct-coupled output stage, as well as an equalized self-balancing inverter, and D.C. supply for preamplifier and inverter tubes. All necessary precautions have been taken in its design and construction to avoid obsolescence.

A NUMBER of questions have been repeatedly asked of the writer since the initial article describing this F.M. amplifier appeared. Among these were:

"Why is it necessary to extend the range of the amplifier from 13 to 30,000 cycles?"

"Why is a 24-watt amplifier required for reproduction of phono or F.M. programs in an average home?"

Both of these questions are answered in this article, after the technical description has been completed.

THE BALANCED FEEDBACK CIRCUIT

The voltage which appears in the balanced 500-ohm winding of the output transformer is fed back to the cathode circuit of the 6SJ7 drivers through a bridged circuit. This particular feedback circuit can best be studied by redrawing the original circuit, which appeared on Page 352 of the Dec., 1940 issue, as shown in Fig. 1, below.

An analysis of this bridge circuit will show that, under normal conditions, no voltage will appear from cathode to cathode of the input tubes. The A.C. voltage across rk2 will be equal to 0. If the feedback resistors rb, rb1, or the cathode resistors rk, rk1, or the feedback windings, are unbalanced, a voltage will be present across rk2. As the input tubes are operating in push-pull, the voltage which appears across rk2 must be in-phase with one of the cathodes, and out-of-phase with the other. It therefore degenerates with the cathode circuit with which it is in-phase, and regenerates with the circuit with which it is out-of-phase. This action, in turn, tends to further balance the voltage across the plates of the driver tubes. Its overall effect greatly increases the overall dynamic stability of the amplifier.

The advantages of running the feedback loop from the secondary of the output transformer back to the cathodes of the input tubes are as follows:

(1) By embracing the output transformer, the feedback loop corrects for frequency discrimination.

(2) Most effective circuit stability is attained by coupling the balanced output feedback circuit directly to the push-pull drivers.

If the feedback voltages were taken directly from the plates of the output tubes, it is apparent that compensation for discrimination within the output transformer would not be effected. During the development of this unit, a tertiary feedback winding was checked, and it was found that a distinct phase shift occurred between the primary of the transformer and the tertiary winding. This latter winding was not always in-phase with either the secondary or the primary. Such a condition naturally results in feedback regeneration at some frequencies. This confirmed a long-standing theory that tertiary windings are not ideal for feedback purposes.

By slight adjustments of feedback resistors, they may be coupled directly to any one of the output taps, so that any variations in the coefficient of coupling between the used output terminals and the 500-ohm line (if these terminals are not used), will not have any effect upon the desirable action of the feedback loop.

THE EQUALIZED SELF-BALANCING INVERTER

One of the major problems in developing an ideal inverter is to be able to obtain equal voltages (out-of-phase) from each side of the inverter output. Reasonable variations in tubes should not produce objectionable unbalance. A basic circuit for a popular self-balancing inverter is given in Fig. 6. In this circuit, balancing action is obtained by including a common grid-return resistor (rg2) in the push-pull stage following the inverter. Balancing action for variation in the amplification factors of V1 and V2 is obtained by applying the differential voltage which appears across rg2 back to the grid of V2. While this action is very effective, it does not provide a perfect balance when V1 and V2 are reasonably matched. In fact, normal operation of this circuit provides an unbalanced signal at

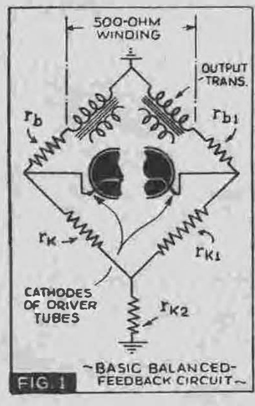


FIG. 1 ~BASIC BALANCED-FEEDBACK CIRCUIT~

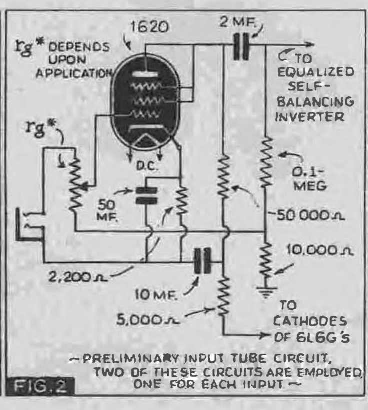


FIG. 2 ~PRELIMINARY INPUT TUBE CIRCUIT. TWO OF THESE CIRCUITS ARE EMPLOYED, ONE FOR EACH INPUT~

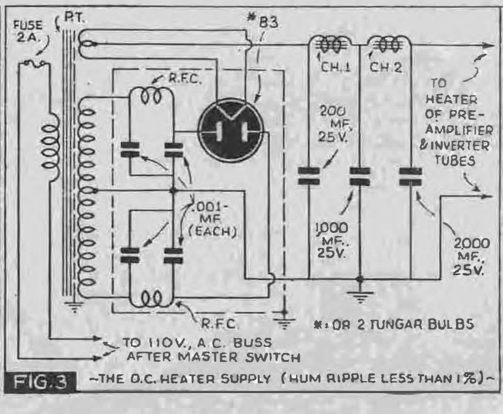


FIG. 3 ~THE D.C. HEATER SUPPLY (HUM RIPPLE LESS THAN 1%)~

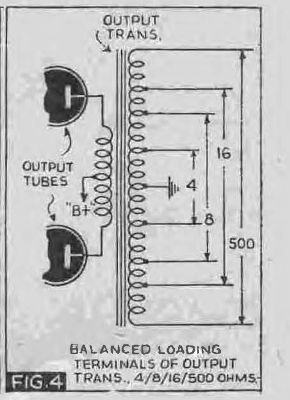


FIG. 4 BALANCED LOADING TERMINALS OF OUTPUT TRANS., 4/8/16/500 OHMS.

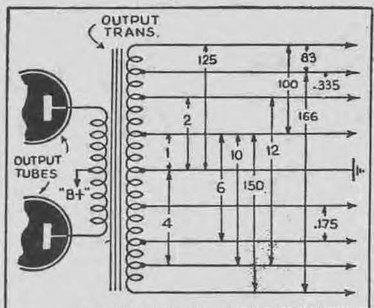


FIG. 5 ~UNBALANCED IMPEDANCE COMBINATIONS, .175/.335/1/2/4/6/10/12/83/100/125/150/166 OHMS

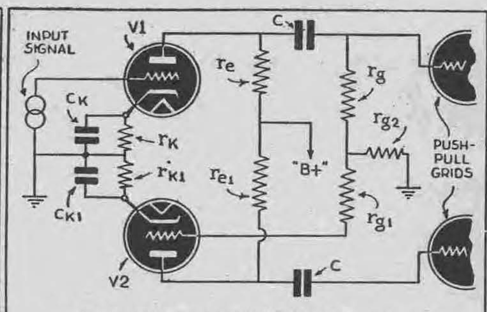


FIG. 6 ~STANDARD SELF-BALANCING INVERTER. NORMALLY FURNISHES A 10% UNBALANCED SIGNAL TO THE PUSH-PULL GRIDS.

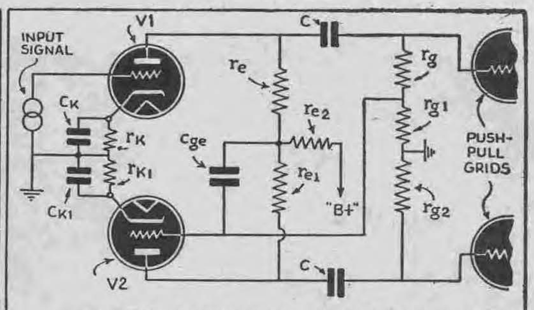


FIG. 7 ~EQUALIZED SELF-BALANCING INVERTER NORMALLY FURNISHES A PERFECTLY BALANCED SIGNAL TO THE PUSH-PULL GRIDS.~

the grids of the output tubes. While this unbalance may not be serious, it nevertheless, introduces distortion. To correct this condition, the equalized self-balancing circuit, illustrated in Fig. 7, was developed. In this circuit, a common plate load resistor is inserted in series with both plate loads of V1 and V2. Any unbalance in output voltage appears across r2 and is coupled through condenser cg to the grid of V2. If this unbalance is opposite in phase to the signal being impressed upon the grid of V2 through the dividing network rg and rg1, then degeneration takes place so as to decrease the output of V2, which in turn, equalizes the signals appearing on the grids of the output stage. On the other hand, if the residual voltage appearing across r2, is in-phase with the voltage being impressed to the grid of V2 through the dividing network rg and rg1, then, regenerative coupling takes place, to increase the output of V2. The great advantage of this equalized self-balancing inverter over the standard self-balancing inverter is that 100% balance is normally attained.

In order to ascertain the relative effectiveness of both circuits, the amplification factor of V2 was altered. In one case the output voltage of V2 was normally adjusted to produce twice that of V1, and in the other case, it was adjusted to produce one-half of V1. Both of these adjustments were made without either balancer in the circuit. Then both balancers were incorporated, and the following data tabulated: (In order to evaluate the effectiveness of the balancing actions, the percentage of unbalance is given under various conditions).

Condition	*Percentage of Unbalance at P.P. grids* Self-balancing Inverter	Equalized Self-balancing Inverter
Balanced Tubes	10%	Perfect balance
V2 Unbalanced +50%	20%	22.5%
V2 Unbalanced −50%	9%	11%

It should be noted that while the standard self-balancing inverter provides a slightly better (by 2.5%) balancing action under widely unbalanced tube conditions, the equalized self-balancing inverter provides for better balancing under normal operating conditions. It should be remembered, that the data indicated, was obtained by unbalancing V2 50% in either direction. This represents a far greater change than ever encountered in actual experience.

SELECTION OF THE INPUT TUBES

As the input tubes and their associated circuits determine the residual noise level within the amplifier, it was decided to carefully check all prevailing tubes and standard circuits in an effort to attain a condition which would provide the highest gain-to-noise ratio.

In conducting these tests, a wide variety of tubes were set up in standardized circuits. Both the gain and their noise were measured under a wide range of operating conditions, so as to obtain the optimum gain-to-noise ratio. One of the preliminary acceptable circuits is given in Fig. 2. It will be noted that the type 1620 triple-grid amplifier tubes are used. It is not to be construed, however, that these are the only desirable tubes. As a matter of fact, a number of other types may be used, depending upon the ultimate application of the amplifier.

In checking residual noise, it was found that in many tubes, hum constituted a substantial portion of noise. By substituting a storage battery supply, many "hummy" tubes had their overall noise reduced from 8½ to 10 db. It was therefore decided that a D.C. supply would be incorporated within the amplifier to heat the preamplifier and inverter tubes.

THE D.C. HEATER SUPPLY

By connecting the 1620s in parallel, and then in series with the 6N7s, a 12-volt 600 ma. supply was required. Work done in our laboratory about 2 years ago, utilizing a type 83 to deliver 1A. at 24 V. proved the advisability of building an "A" supply around this tube. Our lab records had shown 1,000 hours' test without any measurable decrease in emission. The "A" supply rectifier system was therefore built around this type of tube circuit, which is illustrated in Fig. 3.

The writer feels that many readers will think that the 83 would be considerably over-worked in this circuit, in view of the fact that its published D.C. output current rating is 225 ma., maximum. Its peak plate current, however, it will be noted, is 675 milliamperes, maximum. This rating, though, is applicable to a 450-volt condition. It appears from empirical data, that when lower voltages are applied to the plates of the 83, such as 30 or 35 volts, a much higher current can be drawn, and still obtain reasonable life from the tube. A number of photocell exciter lamp supplies, incorporating four 83s and delivering 10 volts at 4 amperes, have proven the dependability of this type of circuit. For sceptical readers however, the amplifier can easily be redesigned to accommodate standard tungar bulbs in place of the 83 rectifier.

It will be noted that a pair of R.F. chokes and bypass condensers are employed in the filter circuit to avoid any disturbances from interfering with A.M. tuners, should they be used with this amplifier.

PLATE AND BIAS SUPPLIES

A study of the original power supply circuit will indicate that 2 rectifiers are employed in a tandem power supply circuit. A 5U4G supplies plate voltage to the drivers, preamplifiers, and inverter tubes, while a 5V4G supplies plate voltage to the power output stage. As a 5V4G is a slow-heating rectifier, plate voltage cannot be applied before the full bias appears at the control-grid of the output stage.

Thus by carefully designing the power transformer and its associated filter this circuit affords increased life of power output tubes as compared to circuits employing 2 rapid-heating or 3 slow-heating bias and plate supply rectifiers.

THE BALANCED OUTPUT CIRCUIT

Although the original circuit showed a balanced output transformer equipped with 4/8/500-ohm taps, this transformer can be supplied with any variation of impedances. It has been standardized, however, with 4/8/16/500-ohm windings. The balanced nature of the transformer provides a wide variety of impedances, which are obtained either by balanced or unbalanced loading. A balanced loading circuit is illustrated in Fig. 4. Figure 5 shows the unbalanced output terminals available, ranging from 0.175-ohm to 166 ohms. It will be noted that a total of 16 impedance combinations are available, ranging from 0.175-ohm to 500 ohms.

Although the transformer may be loaded in an unbalanced fashion, true balanced feedback and push-pull action throughout the driver and power output stage still takes place. An analysis of the unbalanced loading circuit diagram, will clarify this point. Regardless of where the load is applied, the voltage from either terminal of the 500-ohm line to ground, would be identical. If any variation does exist, it would be caused by a difference in the coefficient of coupling from the loading portion of the secondary to the 500-ohm terminal on the same side. It is a relatively simple matter, however, to over-design the output transformer so as to provide a unity coefficient factor under any conditions of normal unbalanced loading.

POWER OUTPUT RATING OF THE AMPLIFIER

In rating the power output of an amplifier used for F.M. applications, the reader should dis-associate himself from conventional P.A. amplifier ratings, as unfair evaluation will take place, if this factor is not taken into consideration.

Ordinarily, a P.A. amplifier can safely be rated up to 5% or 7%. In most P.A. applications, this amount of distortion would not be readily detected. In F.M. work, however, it is imperative that the amplifier be operated at not more than a total of 1% distortion. This precaution must be taken, as originally outlined, to prevent the amplifier from becoming the bottle-neck of distortion in the entire F.M. transmission-reception chain.

Although the amplifier delivers a maximum output of 30 watts, it has been rated at 24 watts for 1% total harmonics. It is intended, however, to be normally operated at an output level of 12 watts which pro-

(Continued on page 32)

"AUDIO-SPECTRUM CONTROL"

Study this new "A.S.C." method of frequency accentuation without the use of shock-excited resonators, equalizers or lossers. It provides a degree of A.F. response control heretofore unattained in sound systems.

Fig. A. Locations of A.S.C. components; unit TV is mounted under chassis.

FOR THE FIRST TIME IN ANY RADIO PUBLICATION

AUDIO SPECTRUM CONTROL—or, as we will usually refer to it hereafter in this article, "A.S.C."—may briefly be defined as a system for controlling, within wide limits, the relative response characteristics of the high, low and middle spectrums of an audio-frequency sound reproducing system.

The A.S.C. (Audio Spectrum Control) system accomplishes a number of useful purposes, as follows:

(1) Restores the high frequencies lost in recording.

(2) Suppresses natural resonating frequencies of audio transformers, loudspeakers, microphones and pick-ups.

(3) Eliminates boominess.

(4) Squelches feedback.

(5) Accentuates weak low, middle, or high frequencies without the use of shock-excited resonators, equalizers, or lossers.

(6) Provides a degree of flexible A.F. response control heretofore unattained.

The need for such a system has long been felt. Proof of which may be found in a hundred-and-one attempted methods for securing this flexibility of control.

GRAPHS SHOW ADVANTAGES OF BROAD-BAND AUDIO-SPECTRUM CONTROL

Conventional tone control circuits provide for a degree of high- or low-frequency compensation as illustrated in Fig. 1A. Regardless of whether either the "highs" or "lows" are accentuated it will be noted that, because of the slope of the accentuation curve, all frequencies are not amplified equally. This condition introduces *amplitude distortion*. With a broad-band control system such as here described the entire low-frequency spectrum may be evenly accentuated, as shown at 1 in Fig. 1B (over any pre-determined band) so that all low frequencies and their corresponding harmonics are prevalent in the same proportion as those present in the original signal. This phenomenon also applies to the band-pass and hi-pass filter circuit controls (2 and 3, in Fig. 1B).

The degree of accentuation of any one of the 3 bands may be varied by the corresponding band control so as to provide an unlimited degree of flexibility in controlling the over-all response curve. This is clearly illustrated in Fig. 1C. A typical, composite curve at given settings of the 3 controls is illustrated in Fig. 1D.

By manipulating the Threshold Volume control (T.V., in the representative 32-W., class A, beam amplifier, Fig. 2), the amplitude of the complete audio spectrum (having any band of any predetermined composite curve) may be varied by altering a single control. This master control, once adjusted, may remain unchanged; in fact, it is shown in Fig. 2 as a fixed 10-to-1 ratio voltage divider.

Heretofore, distortionless tone control systems were limited to a very narrow band equivalent to the resonating frequency of the tuned high- or low-pass circuit.

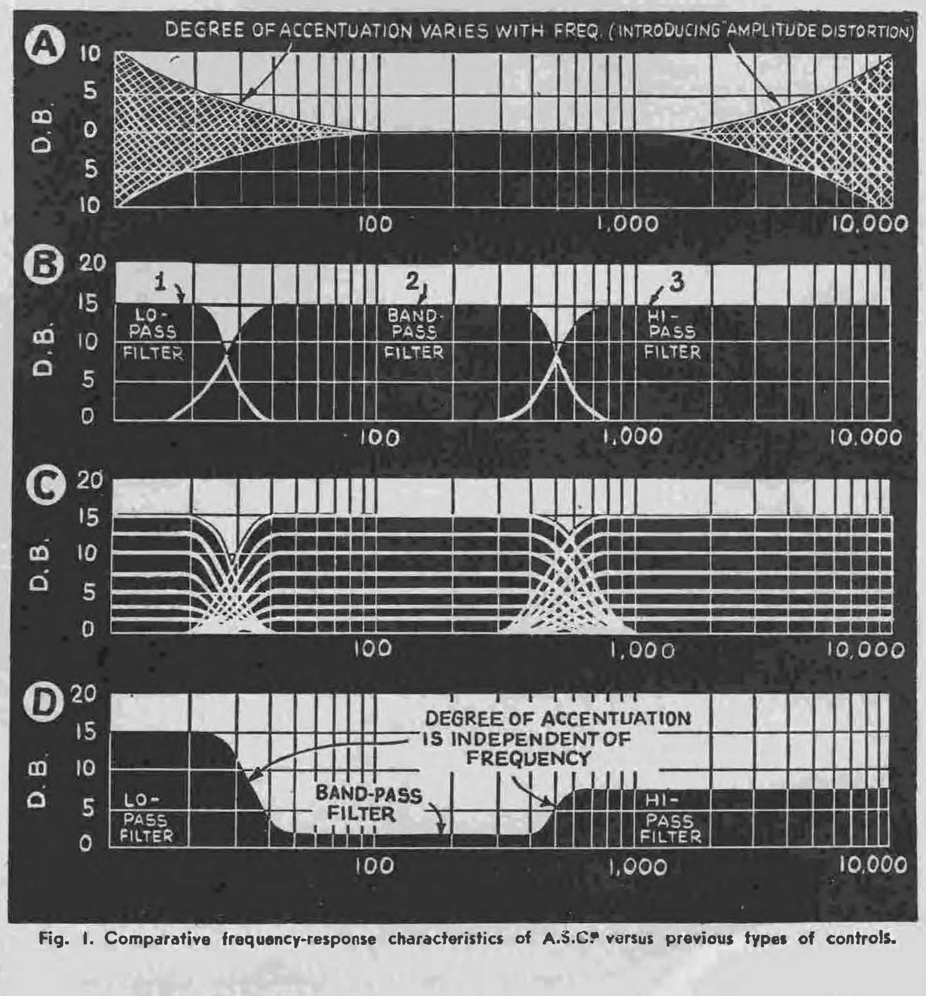


Fig. 1. Comparative frequency-response characteristics of A.S.C.* versus previous types of controls.

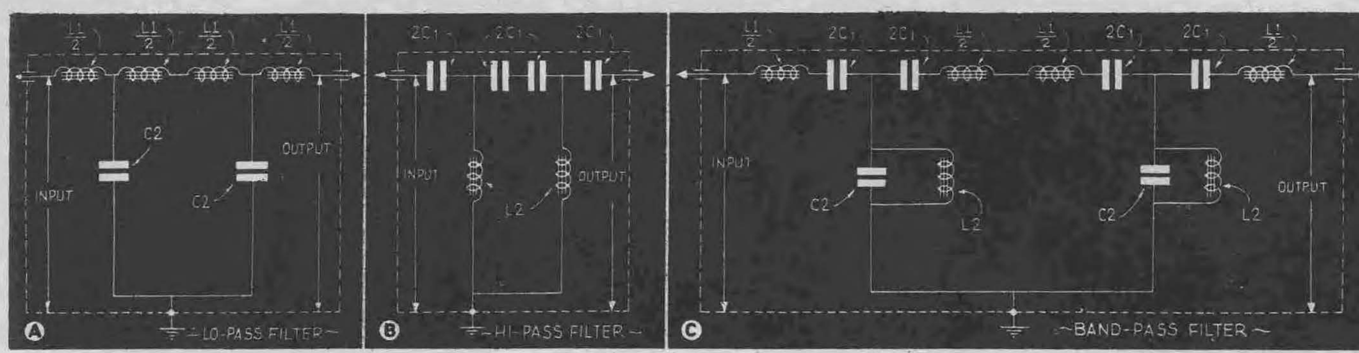


Fig. 3. Schematic circuits of the A.S.C. filter elements shown pictorially in Fig. A. Formulas for finding values appear in text.

LIMITATIONS OF PREVIOUSLY AVAILABLE OR "NON-A.S.C." CIRCUITS

The predominant disadvantages of each type of circuit becomes readily apparent when one analyzes the effects produced by the following existing types:

(a) **Tone Control**—Random high frequencies are (usually) cut off.

(b) **Bass Booster**—Maximum bass compensation takes place at some natural (resonant) frequency of the tuned system. Flat response cannot be maintained over a given band.

(c) **Equalizers**—Cannot be controlled within wide limits. Flexibility of control and response is definitely limited by the pre-assigned characteristics of the equalizer.

(d) **Compensators**—Provide for varying frequency response characteristics at different levels. Operation control is limited and subject to conditions outlined under Bass Booster.

(e) **Lossers**—Utilize an anti-resonant circuit providing maximum attentuation at the anti-resonant frequency. Flat response losses cannot be maintained over a wide band.

(f) **Resonators**—Are as a class similar in circuit, design and performance to bass boosters, compensators, equalizers and lossers. These are all subject to *shock excitation*—a condition which brings about the persistence of resonant signal frequencies after the original signal dies away. This effect adds boominess to bass and is often referred-to as "false bass."

(a) Tone controls, (b) bass boosters, (c) equalizers, (d) compensators, (e) lossers and (f) resonators, have contributed a great deal toward the improvement of frequency-response characteristics encountered in commercial amplifiers and associated accessories; however it has become evident that there are definite limitations in the application of each type circuit which greatly restrict its usefulness in practice.

Fundamentally, the major limitation is that no method has heretofore been perfected which will provide for adequate audio-frequency control over a continuous, predetermined band.

From the foregoing, it is evident that the problem of adequate frequency control has not been solved in commercial amplifiers.

WHY USE A FREQUENCY-RESPONSE CONTROL IF AMPLIFIER RESPONSE IS "FLAT"?

Why should it be necessary to provide a high-fidelity amplifier with a suitable method for controlling its response characteristics?

The answer—in 5 parts—can be found in searching for the reason which necessitated the use of the amplifier and its associated accessories—i.e., for the reproduction of sound in either a natural or pleasing manner.

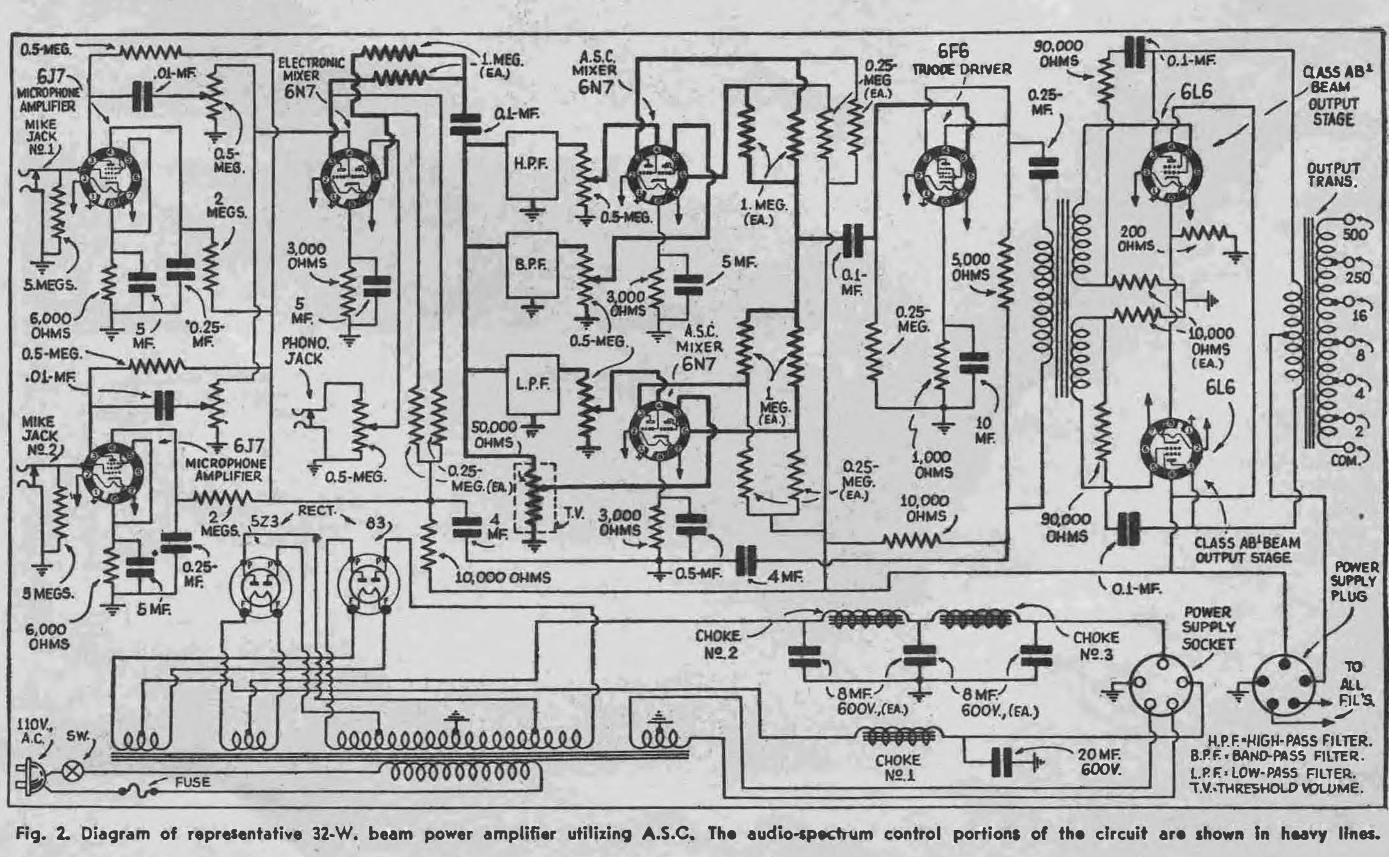


Fig. 2. Diagram of representative 32-W. beam power amplifier utilizing A.S.C. The audio-spectrum control portions of the circuit are shown in heavy lines.

(1) For the ideal recreation of recorded audio signals, it is to be assumed that all deficiencies of the recording system will be compensated-for during reproduction. Obviously a wide range of frequency control is required to neutralize all frequencies attenuated or accentuated during recording and playback.

(2) Assuming for a moment that present recording systems are ideal, the frequency response characteristics of microphones, speakers and other accessories still contribute frequency distortion.

(3) Further extension of our supposition that all auxiliary components have attained an ideal state of perfection, the problem of studio acoustics will still affect the frequency response of our ideal system.

(4) Even if all studios were s ndardized acoustically, the physiological reaction of individual listeners will not be alike. In fact, the response characteristics of each ear of an individual listener will sometimes vary enough to offset all efforts exerted in the design of a flat-frequency-response audio system.

(5) On top of all this, when one realizes that the response characteristic of the average ear is far from a straight line in nature (maximum sensitivity lies between 1,000 and 5,000 cycles), it becomes readily apparent that some flexible method of controlling the frequency response characteristics of our reproducing system is not only desirable but a practical necessity particularly if pleasing reproduction is desired for all types of signal amplification under any local acoustic condition.

Therefore, a desirable method of control should provide for the accentuation of a series of predetermined audio bands each of sufficient width so that additively they cover the entire audio range. As each band width would require a separate control, the spectrum should be divided into a minimum number of bands.

For simplicity of manipulation, a *master control* should be incorporated so as to avoid the necessity of resetting all band controls for each different volume level. In operation the master volume control should have the same effect as equally increasing or decreasing all band controls.

WIDE-RANGE THRESHOLD CONTROL

As complete suppression of any one band would prove to be detrimental to quality reproduction a wide-range threshold control should be incorporated so as to vary the minimum accentuation of all band controls. This feature is particularly desirable if any unit in the audio system resonates at some fixed frequency. Proper adjustment of the threshold control will prevent undue overload distortion at a given resonating frequency when the band control (in which the resonating frequency lies) is turned to maximum.

As the ear response varies most at both extremes of the hearing range it was decided to arbitrarily divide the audio band into the following 3 sub-ranges: (1) Low Frequency (16-250 cycles); (2) Middle Frequency (250-5,000 cycles); and (3) High Frequency (5,000-20,000 cycles). This empirical division need not be followed, since any other ratio of division may be utilized depending upon the type of compensation required for any given condition.

THE CIRCUIT IN A NUTSHELL

Essentially, the audio spectrum control circuit, as shown in heavy lines in Fig. 2A, splits the input signal into 4 branches. The first 3 branches are fed through a "low-pass filter" (for low-frequency accentuation), a "band-pass filter" (for middle-frequency accentuation), and a "high-pass filter" (for high-frequency accentuation), respectively. The remaining branch passes through a wide-range control, T.V., for controlling the operating threshold of all band controls.

This type of circuit has the distinct advantage that it does not utilize "losser" circuits and that it does not utilize resonating-type accentuating circuits which have control over a limited and narrow audio band. Of course, the usual boominess, caused by the shock excitation of tuned resonating circuits, is absent. The ability of being able to add-to or accentuate any portion of the audio spectrum, and to maintain that accentuation *flat* within the selected portion of the A.F. range, brings a new type of fidelity control to the audio field.

DESIGNING THE FILTERS

The general design follows conventional engineering practice. In order to attain a flat-line response curve, when all filters are in the circuit, it is important that the slope of the cut-off characteristic of the filters complement each other. The following formulas are included only for the technician who wishes to design his own high-pass, low-pass, and band-pass filter, shown in detail in Fig. 3, to cover any desired frequency range.

Lo-Pass Filter. The fundamental circuit of the *lo-pass filter* is illustrated in Fig. 3A. The inductances are determined by the following formula:

$$L_1 = \frac{Z}{\pi F} \text{ henrys}$$

Inasmuch as the circuit employed is of the π type, the value of the inductance above calculated, should be divided by 2. The value of the shunt condensers may be found from the following formula:

$$C_2 = \frac{1}{\pi F Z}$$

F = cut-off frequency
C = capacity in farads
L = inductance in henrys
Z = the iterative (That is, the same impedance is reflected from either end of the filter.) impedance of the filter

Hi-Pass Filter. The *hi-pass filter* follows the same design principle and is similar to the circuit shown in Figure 3B. The condenser and inductance values may be found from the following formulas:

$$L_2 = \frac{Z}{4\pi F} \text{ henrys} \qquad C_1 = \frac{1}{4\pi F C}$$

Band-Pass Filter. The circuit employed in the *band-pass filter* is similar to the schematic shown in Fig. 3C. Constants for the various condensers and inductances may be found from the following formulas:

$$C_1 = \frac{F''-F'}{4\pi(F' \cdot F'')Z} \qquad L_2 = \frac{(F''-F')Z}{4\pi(F' \cdot F'')}$$

$$L_1 = \frac{Z}{\pi(F''-F')} \qquad C_2 = \frac{1}{\pi(F''-F')Z}$$

F' = lower cut-off frequency
F'' = higher cut-off frequency

THE MIXER AND AMPLIFIER CIRCUITS

It will be noted from the schematic circuit, Fig. 2, that the mixer arrangement provides for independently varying the level of the various signals passing through their respective filters. The unlimited degree of flexibility offered by this type of audio spectrum control provides for the addition or subtraction of frequencies in any one or more of the bands to any recording or other program passing through the amplifier.

The circuit of the amplifier also follows conventional high-fidelity design. It will be noted that 2 microphones may be mixed with a phono pickup. The 2-position input preamplifier utilizes two 6J7s which are in turn fed into the 6N7 mixer which supplies the band-pass system with its signal energy. The two 6N7s utilized in the output of the filter feed the 6F6 driver which utilizes a parallel-resistance-feed plate circuit to keep direct current from the primary of the driver transformer. Inverse feedback is employed in the output stage in a conventional manner so as to reduce the output hum and distortion caused by resonance effects in the speaker load circuit.

THE POWER SUPPLY

In order to minimize any inductive hum pick-up effects, between the power supply and the band-pass filter units, a separate power pack is utilized. It will be noted from a casual observation of the power supply circuit, Fig. 2B, that a stabilized voltage supply system is employed in order to insure constant screen-grid voltage to the beam-type power output tubes. This feature eliminates screen-grid circuit distortion.

ALL-PUSH-PULL DIRECT-COUPLED 10-WATT AMPLIFIER

(Continued from page 7)

(with single input and single output) it is necessary to switch a number of circuits before playback can take place. In the twin-channel amplifier (see Fig. 6C) a microphone feeding into the 1st channel can operate the cutter (connected to the output of the 1st channel), and a crystal pickup (for playback) can be connected to the input of the 2nd channel. A speaker connected to the output of the second channel completes the playback system.

This same arrangement can also be used for introduction of "artificial reverberation," or *echo*, simply by providing a time delay in one of the amplifier circuits. This is illustrated in Fig. 6D. With this arrangement, part of the original amplified signal from channel No. 1 is sent through an echo chamber or other acoustic time delay unit, such as a long pipe. This sound is picked up and sent through channel No. 2, and is ultimately reproduced along with some part of the original signal, so that the effects of reverberation or echo are obtained.

For conventional push-pull operation, it is necessary to use a good push-pull output transformer, together with a twin half-meg. potentiometer, as illustrated in Fig. 7A. In order to obtain push-pull operation of the input tubes, it is necessary to feed a push-pull signal into the input of the amplifier. This is obtained by removing one of the phono pickup or microphone leads from ground, and feeding in through a 2-wire shielded cable. Microphones and pickups are easily attainable for this type of cable connection.

INPUT AND OUTPUT TRANSFORMERS

If it is impossible to isolate one of the leads of the input signal from ground, or if a low-impedance (200- or 500-ohm) input device be connected to the amplifier, an input transformer must be used as per Fig. 7B. In order to attain true high-fidelity reproduction, this unit should employ an "electric metal" core, and should match the input device to the input of the amplifier (100,000 ohms, grid-to-grid). Hum-balancing construction should be utilized to avoid excessive hum pick-up.

Needless to say, it is impossible to design or construct transformers which will have a response comparable to that of the amplifier. In fact, a wide-range output transformer capable of passing 10 to 20,000 cycles with less than 1 db. variation, would coat approximately $20. In order to enable the use of a low-priced output transformer with this unusual amplifier, a special push-pull inverse feedback and compensating network circuit was employed, which is illustrated in Fig. 8. This circuit enables a $3.50 transformer to equal the performance of a $20 unit!

TONE CONTROL CIRCUIT

It appears to be sacreligious to add a frequency discriminating arrangement to this ideal amplifier. Nevertheless, existing deficiencies in speakers, transmission lines, microphones and pickups necessitate such an adaptation.

The frequency discriminating network should be connected in the input circuit across both grids of the 6SJ7 as per Fig. 9, or in series, or shunt, with the input device, depending upon the type of equalization desired. A standard tone control (of the high-frequency cut-off) is indicated in the List of Parts, although this type of circuit need not be followed, as any other form of equalization may be effectively employed.

Dear Editor . . .

Nevertheless, *its response is flat from 1 cycle per* minute *to 20,000 cycles per* second.

Unfortunately, there are no signal generators to produce such a low frequency, and there are no output meters available capable of measuring this low frequency response. We therefore had to adopt a new method for measuring these unusually low frequencies, by introducing a small varying voltage into both grids. This is done by connecting a dry-cell into the input circuit and slowly varying the volume control from 0 to full setting.

The output voltage is measured by a cathode-ray oscilloscope in the same manner that D.C. would be measured.

The high-frequency response seems to be unlimited, as there are unusually low distributed capacities present in the tubes employed. Although we made no measurements above 20M cycles, I believe that its response can be extended out to radio frequencies, which makes this unit also admirable for television application at the video frequencies.

A. C. S.

How to Add 1 to 14 Modern Features to the

ALL-PUSH-PULL DIRECT-COUPLED 30-WATT P.A. AMPLIFIER

(Continued from page 14)

verter circuit adds approximately 20 db. to the overall gain of the amplifier. With this particular circuit, it is possible to dispose of the twin volume control of the main amplifier and insert a standard audio grid taper control in the input grid circuit of the inverter tube.

OPTIONAL HIGH-GAIN SINGLE-ENDED INPUT

Where a gain of more than 100 db. is required, it is recommended that a high-gain single-ended input circuit be incorporated, as diagrammed in Fig. 8. This unit may take the place of the low-gain inverter (of Fig. 7) or it may be used as an auxiliary high-gain input.

It will be noted that in this inverter circuit, a 6SC7 is used with (comparatively) high-resistance plate loads, so as to insure maximum gain. Approximately 30 db. is added to the overall gain of the Direct-Coupled Amplifier, bringing its total up to 120 db. If both inverters are to be added, their outputs are simply paralleled. With such an arrangement, 2 individual single inputs are made available; one having a gain of 110 db. and the other of 120 db.

ADDITIONAL LOW-GAIN PUSH-PULL INPUT CIRCUIT

For some applications, it may be desirable to maintain the push-pull arrangement throughout the amplifier. This type of a circuit will require a twin triode unit. The 6N7 is admirably adapted for such use. The circuit follows conventional design as diagrammed in Fig. 9. It is to be borne in mind, that a 3-way plug should be used to bring the signal into the low-gain push-pull input. This circuit adds approximately 17 db. to the overall gain of the amplifier, making a total of 107 db.

ADDITIONAL HIGH-GAIN PUSH-PULL INPUT CIRCUIT

If both a higher gain *and* a push-pull input circuit are required, the 6SC7 may be used, with values as diagrammed in Fig. 10. This circuit adds approximately 27 db. to the overall gain of the amplifier, making an overall total of 117 db. available.

The following remaining features, which may be incorporated into this amplifier, will be completely described in the next issue:

Push-Pull Automatic Volume Compressor
Non-Frequency Discriminating Scratch Suppressor
Audio Spectrum Control
Push-Pull Automatic Volume Limiter
Calibrated Volume Indicator
Remote Control
Push-Pull Automatic Volume Control

Addenda: **Balanced Push-Pull Detector Circuit.** A great many requests have been received for a suitable *balanced push-pull detector circuit* for use with the All-Push-Pull Direct-Coupled Amplifier. Such a circuit is given in Fig. 11.

It will be noted that the load resistor of the diode circuit is divided into 2 equal parts, and the center is grounded. This type of a circuit will deliver 2 signals, which are equal and exactly 180 degrees out of phase. The 2 resistors, of course, are equal sections of a twin volume control. It is important to note that each of the sections is individually filtered. This circuit has a distinct advantage in developing a balanced push-pull output signal under widely varying tube characteristics, because the output is automatically balanced as long as both sections of the twin volume control are of the same value.

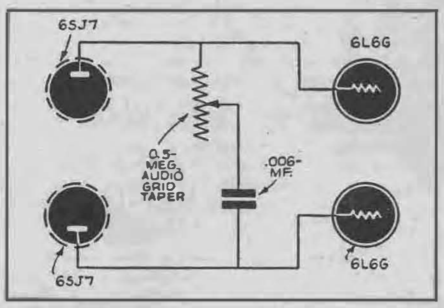


Fig. 12. Tone circuit for 10-W. direct-coupled amplifier.

(Continued from page 15)

Triode Direct-Coupled Amplifier

back signal. Otherwise it would be impossible to provide for balanced frequency equalization.

PUSH-PULL SINGLE-ENDED INPUT

A push-pull signal may be coupled directly into the grids of the 6SK7 tubes. For single-ended signals, however, such as a grounded phono pickup or radio tuner a 6SC7 inverter is added. This is of the degenerative type. A dual-grid tube is used so as to provide for electronic mixing of 2 independent input circuits. Each grid is equally efficient in providing an inverted signal to the push-pull 6SK7 tubes.

The overall gain of the amplifier, with controls set for flat response and no expansion, is 75 db. For microphone work, an additional preamplifier stage should be added.

POWER OUTPUT RATING

This amplifier is rated at 22.5 watts at 2.5% total harmonics. It will deliver 30 watts at 5%.

How to Add 1 to 14 Modern Features to the

ALL-PUSH-PULL DIRECT-COUPLED 30-WATT P.A. AMPLIFIER

(Continued from page 17)

frequency, regardless of what instrument is being played. Each tone's *quality*, however, is determined by the quantity and amplitude of its associated harmonics. If these harmonics are to be changed in *intensity*, a characteristic change takes place in the quality of the tone. Therefore, if any tone is amplified and its associated harmonics are altered in relation to the fundamental, a change in character of tone is to be expected. This accounts for the popular expression of "false bass," which is pronouncedly present in the usual bass-boosting circuits. This same condition exists when low frequencies are cut or when high frequencies are accentuated or attenuated.

Figure 2B. A typical response curve of an Audio Spectrum Control Circuit, Fig. 2B, shows how the fundamental frequency and its associated harmonics are equally accentuated, thereby introducing no change in the relative amount of harmonics present in the original signal. The Audio Spectrum circuit operates similarly in both the high- and middle-frequency bands.

Figure 2C. As this circuit employs a type of triode which has been carefully selected for minimum even-order harmonic contributions push-pull operation is not required. The circuit for the complete Audio Spectrum Control is illustrated in Fig. 2C.

CALIBRATED VOLUME INDICATOR

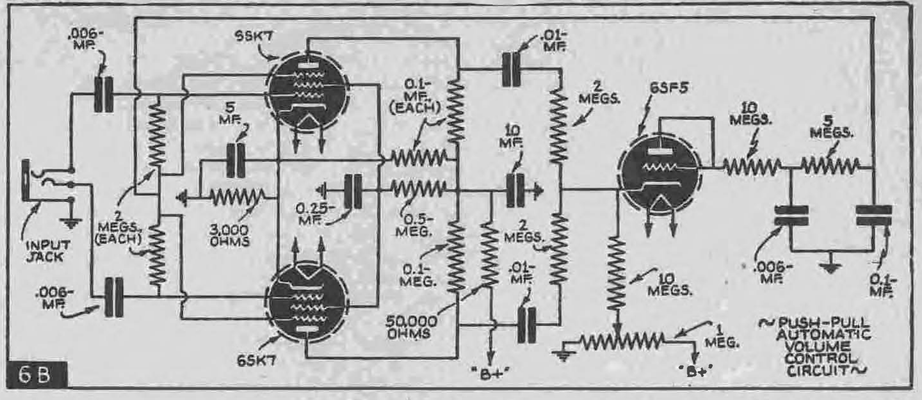
The use of a calibrated volume indicator across the output of the amplifier is a highly desirable auxiliary feature, for it enables an accurate check on the power output delivered by the amplifier. The important point in constructing a volume indicator is to check the accuracy of indication over *all* audio frequencies. Any good 0-1 ma. meter utilizing an external copper-oxide rectifier may be employed with excellent results. As most copper-oxide rectifiers are comparatively inefficient at the higher frequency, a compensating network is required.

Because of the wide variations in both meter and rectifier construction, it would be difficult to offer a fixed method of compensation. However, a general procedure for adjustment will enable the average experimenter to obtain satisfactory results.

Figure 3A. In Fig. 3A is illustrated the basic copper-oxide rectifier arrangement. This arrangement, however, will usually fall off at the high frequencies. If R1 is selected to obtain a desired voltage at 1,000 cycles, and a potentiometer substituted in its place, compensation can easily be attained by utilizing a suitable condenser across the center arm and one end of the potentiometer.

Figure 3B. The most rapid procedure would be to introduce a condenser decade box at point CX, Fig. 3B, and adjust both the capacity of the condenser, and the setting of R1 until a flat response is maintained out to the highest frequencies to be measured. During these adjustments, constant references should be made to 1,000 cycles. Under any condition, the compensating network should not change the 1,000-cycle reading unless the response of the meter and rectifier falls off at this frequency.

Figure 3C. For multiple ranges, additional series resistors should be included, with a suitable switching arrangement, as illustrated in Fig. 3C. The exact voltages of the series resistors will depend upon the db. range desire. Many charts are available which indicate the voltage developed across a 500-ohm line at any output level in db.



PUSH-PULL AUTOMATIC VOLUME COMPRESSOR

The *volume compressor* acts directly opposite to the *volume expander*. A review of the volume expander circuit (described in past issues of *Radio-Craft)*, will facilitate understanding of the compressor circuit.

For expansion, the 6SK7's are negatively biased for low transconductance. The expanding amplifier takes a portion of the input signal, amplifies it, then passes it on to a full-wave rectifier, which develops a positive voltage to oppose the normal high negative bias. The overall result is a reduction in negative bias and an increase in transconductance. This effect produces expansion. For compression, the input 6SK7's have a low negative bias for a high transconductance. The compression amplifier amplifies a portion of the input signal and applies it to the full-wave rectifier, which develops a negative bias. This negative bias increases the original bias to decrease the transconductance (and lower the gain).

Figure 4. Only 2 basic changes are necessary to convert the push-pull expander into the push-pull compressor, namely, (1) a decrease in the 10,000-ohm cathode resistor to 3,000 ohms, and (2) a reverse connection of the rectifier, so as to develop a negative instead of a positive bias. This revised circuit is illustrated in Fig. 4.

PUSH-PULL AUTOMATIC VOLUME LIMITER

The fine distinction between *volume compression* and *volume limitation* is that volume compression may take place without regard to the output of the amplifier, whereas volume limiting must take into consideration the power output developed by the amplifier.

Fundamentally, the volume limiter circuit is similar to the volume compressor, with the exception that the signal which feeds the compressor amplifier, is taken from the 500-ohm output instead of the input which feeds the 6SK7's. This signal is then further amplified, rectified, and applied back as a negative bias to decrease the transconductance of the input 6SK7 tubes. An auxiliary bias voltage is introduced so as to prevent effective rectification from taking place until some predetermined volume level is attained.

Figure 5. This circuit, therefore, acts as a "peak limiter" and simply removes excessive volume surges to keep the output from exceeding some predetermined volume limit. The complete circuit is illustrated in Fig. 5.

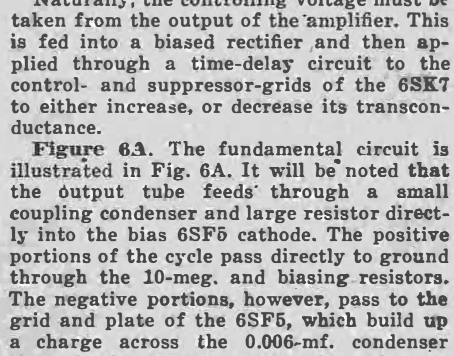
PUSH-PULL AUTOMATIC VOLUME CONTROL

This unique circuit is characteristically different from either the volume compressor or the volume limiter in both its circuit and performance. The *automatic volume control* automatically maintains a given volume output. That is, it actually increases the gain of the amplifier when the input level drops; and conversely, decreases the gain of the amplifier when the input level increases.

Naturally, the controlling voltage must be taken from the output of the amplifier. This is fed into a biased rectifier and then applied through a time-delay circuit to the control- and suppressor-grids of the 6SK7 to either increase, or decrease its transconductance.

Figure 6A. The fundamental circuit is illustrated in Fig. 6A. It will be noted that the output tube feeds through a small coupling condenser and large resistor directly into the bias 6SF5 cathode. The positive portions of the cycle pass directly to ground through the 10-meg. and biasing resistors. The negative portions, however, pass to the grid and plate of the 6SF5, which build up a charge across the 0.006-mf. condenser through the 10-meg. plate-grid series resistor. The negative charge across the 0.006-mf. condenser is then applied through a time-delay circuit (which may be made variable) composed of a 5-meg. and a 0.1-mf. condenser. By adjusting resistor R1, the range of control may be varied. At maximum setting, constant output is maintained with a 30 db. variation of the input signal. This unusual controlling range offers a wide latitude of movement to orators employing "fixed position" microphones.

Figure 6B. The push-pull version of the automatic volume control circuit is illustrated in Fig. 6B.



Recording-Playback Amplifier

(Continued from page 19)

constant-amplitude equalizers, or both) should readily correct any noticeable discrepancies. It should be borne in mind that a constant-amplitude cutting head may require constant-velocity equalization, because the frequency losses encountered in cutting a soft material will vary proportionately with frequency. When a crystal cutter is cutting high frequencies in a relatively soft material, all of the material is not cut away. A good portion of the needle displacement may simply be pushing the material from side to side. This "pushed" material nearly completely returns to its original form because of its resilient nature. This phenomenon, in itself, introduces an appreciable high-frequency loss. Under these conditions the constant-velocity equalizer should be placed into the circuit so as to provide a gradual high-frequency boost. A greater degree of equalization will be required, as the needle approaches the center of the record, because of the decreased surface record speed. For 33 1/3 r.p.m. recording a similar type of equalizer is required in order to maintain good high-frequency response. The degree of equalization should, of course, be gradually increased as the cutting mechanism approaches the center of the record.

Automatic electronic equalizer circuits may be incorporated into this amplifier so as to provide for automatic equalization once the exact losses have been established in some standard recording material.

FM Audio Amplifier — Part I

(Continued from page 22)

Unbalanced Tube Numbers	*Output Unbalance*
1 and 2	61 ma.
1 and 3	68 ma.

When these same tubes were inserted into the balancing circuit, the following results were noted:

Unbalanced Tube Numbers	*Output Unbalance*
1 and 2	8 ma.
1 and 3	8 ma.

As the D.C. balancer becomes an integral part of the A.C. balancer circuit as well, it was necessary to select optimum resistor values which would provide a minimum D.C. unbalance and minimum A.C. unbalance.

FM Audio Amplifier — Part II

(Continued from page 24)

grid to screen-grid voltage amplification of both tubes is equivalent (for simplicity of explanation), then the residual instantaneous +8 volts of Fig. 8 is applied directly to both screen-grids without any additional cancellation. This instantaneous positive voltage also acts as a driving voltage to the screen-grid of tube B to further increase the negative swing of its plate. In actual practice, circuit values can be adjusted to automatically correct for any desired range of variation between tubes. Laboratory tests, however, simplify the determination of optimum values for maximum D.C. static correction, maximum A.C. dynamic correction and minimum loss of overall gain.

LABORATORY TEST SET-UP

For checking the degree of balance obtainable, the laboratory equipment indicated in Fig. 13 was used. The coupling transformer T1 was used to obtain a push-pull signal. Two vacuum-tube VU meters were used across each half of the push-pull input signal to enable exact adjustments of input voltages. Individual calibrated attenuators were used to vary the amount of input signal fed into each half of the push-pull stage.

It was found that when full signal was fed into one grid and no signal into the other, a 50% balance occurred. In other words, one output grid developed a voltage 50% of the other and exactly 180° out-of-phase. With a 50% variation in input signal, 80% balancing occurred. In other words, when half as much signal was fed into one input grid, as compared to the other, its associated output grid had 4/5 of the voltage which appeared on the opposite push-pull output grid. This signal was also exactly 180° out-of-phase. Both of these conditions represent extreme abnormalities. Over 100 combinations of input tubes were checked for variations in voltage amplification. It was found that the greatest variation of tubes produced a difference of less than 5% between both output grids.

FM Audio Amplifier — Part III

(Continued from page 26)

vides less than ½ of 1% total distortion. These unusually low ratings are advocated so as to virtually eliminate distortion considerations from the amplifier. It should be borne in mind, however, that if the unit is operated at an average level of 3 watts to produce unmeasurable harmonics, transient increases of level of 9 db. will bring the power output up to 24 watts with its intended 1% of total harmonics. Many so-called de-luxe F.M. radio receivers employ relatively low power output stages to effect appreciable economies, particularly when large quantities of receivers are involved.

WIDE-RANGE RESPONSE

The development of an amplifier having a response of from 13 to 30,000 cycles ± 1 db., obviously increases its overall cost, and sometimes raises the question, "Why should I buy an amplifier with such a wide-range response, when F.M. broadcasts only run from 50 to 15,000 cycles? Furthermore, the average human being can not hear 30,000 cycles."

To answer this question intelligently, we must first acknowledge the fact that the latest findings amongst young listeners with acute hearing clearly indicate that 30,000 cycles CAN be perceived! Furthermore, fundamentals and sub-fundamentals, should be reproduced, in order to avoid destruction of original tone qualities. This can easily be proven by a difference in quality of response of bass drums or organ programs when fundamentals are cut off. In view of the fact that the response range of amplifiers has been continually increasing the writer believes that it is only a question of *time* before the ultimate amplifier will extend out to the *outermost limits of human hearing,* and, if this can be accomplished now, why shouldn't the amplifier be removed as a restricting link in the chain of reproduction?

We note that in the past, loudspeaker manufacturers have consoled themselves for restricted response by contending that no "program or amplifier can reproduce more than 5,000 cycles." Record manufacturers complained that no phono pickup could reproduce more than 6,000 cycles, and pickup manufacturers contended that no amplifier passed more than 8,000 cycles. This vicious circle naturally hindered projected improvements in any one branch.

These illogical assumptions really have no place in modern communication equipment. If F.M. stations *are forced by the F.C.C.* to provide 50 to 15,000 cycles, the writer believes it is only a question of time before this spectrum will be balanced* and some of the better stations will eventually extend this range to the very outer limits of human hearing.

In providing this extremely wide range within the present amplifier, any possibility of early obsolescence is completely eliminated, for further extension of the range is obviously unnecessary, unless the human race during the process of evolution will acquire an extended hearing range.

CONCLUSION

The writer wishes to caution readers not to compare this amplifier with conventional public address units, by checking power output, distortion, hum, or tube components, as a number of essential features have been included in its design, which are highly desirable, in order to attain the full benefits from F.M. broadcasting.

Because of its technical excellence, it can of course, be used in any other application requiring the ultimate in design and performance.

* See "Balanced Audio Spectrums," Radio-Craft, September, 1940, Page 164.

Corrections in Amplifier Articles

In the article, **"How to Design a Flexible All-Push-Pull Direct-Coupled 30-W. Amplifier,"** by A. C. Shaney, in the October, 1939, issue of *Radio-Craft*, an error appeared in the schematic diagram, Fig. 2, pg. 203. Both power supplies should be shown connected in series. That is, the center-tap and associated filter condensers of the high-voltage A.C. winding which supply the plates of the 5V4G should be connected *directly* to the cathodes of the 6L6G output tubes.

In the article, **"How to Add 1 to 14 Modern Features to the All-Push-Pull Direct-Coupled 30-W. P.A. Amplifier,"** Part III, by A. C. Shaney, an error appeared in the input-jack circuit of the schematic diagram, Fig. 5, pg. 343. The entire diagram is reprinted below in its corrected form. Terminals X connect to the push-pull output leads. (Dec., 1939, issue.)