High-Quality, Low-Power Stereo Amplifier

JOSEPH GIOVANELLI

Our own Mr. Giovanelli describes how he converted his system to stereo, and the reasons for making his decisions.

RECENTLY I SWITCHED TO STEREO. Like many others I had what I considered to be a fine monophonic system and, therefore, could see no need to scrap the equipment. In order to achieve the stereo effect, I merely duplicated my original monophonic system. At that time there was very little choice of stereo preamplifiers, integrated preamplifier-amplifier units, and the like.

It was not long before I became annoyed with the balancing problem. Since all equipment was duplicated, balance was achieved by turning the volume control knobs, one on each amplifier. It would not have bothered me much if, after having achieved a good balance between channels, the system stayed in balance from tape to tape and, later, from disc to disc. The whole business became a real production when I wished to raise or lower the volume of both channels simultaneously. This necessitated rebalancing. (It should be said here that in my case balancing was difficult for an additional reason—because the equipment cabinet was located much nearer to one speaker than to the other one. Of course, with a little practice one can learn to make this adjustment even under such conditions, but it was annoying nevertheless.)

I did not mind the duplication of the remaining controls because it gave me a great deal of flexibility. I could listen to a monophonic recording and at the same time record another program by means of the unused channel. I could match the two channels to suit the room acoustics.

When I had time I tried to figure out just what I wanted my amplifier to do. It should be self-contained with both amplifier channels and the power supply mounted on a single chassis. Only aluminum should be used because of the ease with which it can be worked and because of its non-magnetic properties. It should be at least as small as *one* of my original power amplifiers. It should be kept as cool as possible and be made as light in weight as possible. Also, the amplifier should be inexpensive to build, making use of as many parts from the junk box as possible.

The sound produced by this amplifier must be good. Intermodulation must be under one per cent at full output. Above all, this is the most important.

Since I very much liked the preamplifiers I was using, I did not wish to scrap them, so balancing would have to be accomplished in the power amplifier circuit itself. Yes, while I was about it, why not paint the lily and build the balance circuit in such a way that balancing could be accomplished remotely from any part of the room? (This last sounds like a tall order, but, as can be imagined, it is far more comfortable and convenient to balance a stereo system while seated in a favorite listening position.)

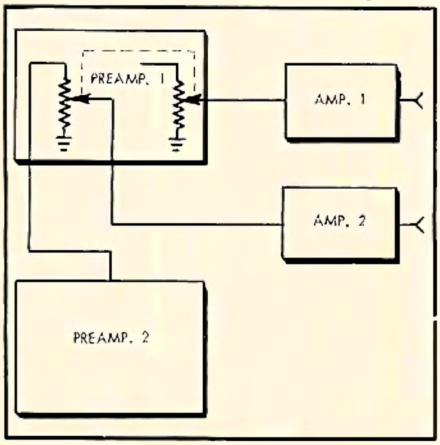

Fig. 1. Circuit arrangement for controlling the gain of both channels simultaneously.

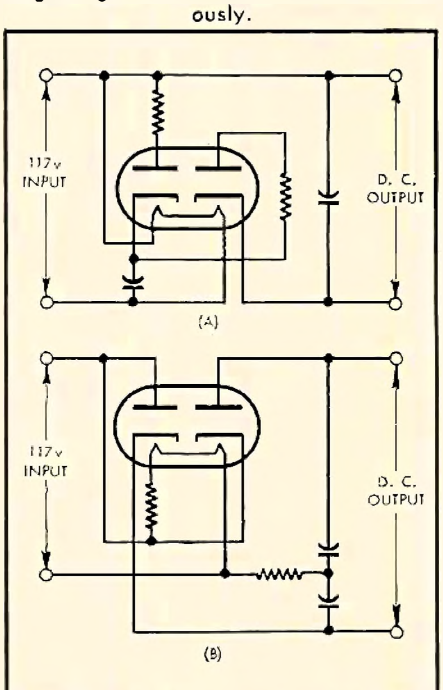

Fig. 2. Voltage doublers: (A) half wave, and (B) full wave.

How to meet these requirements? That was the big question. I will tell you how my reasoning went, how I rejected some ideas and kept others. When I am through, why not try building the finished amplifier?

Balancing and Volume Considerations

As previously noted, controlling the volume of both channels simultaneously was a problem.

I overcame this by means of ganged potentiometers in one of the preamplifiers, rather than by using the single pot which normally served as the volume control. The first gang was connected in the normal manner and served to adjust the volume of this preamplifier. The second gang controlled the output of the other preamplifier. This preamplifier was connected in the usual manner. The output of the unmodified preamplifier was connected across the second gang of the potentiometer of the modified preamplifier, and the input of the power amplifier for that channel was connected between the arm of that second gang and ground. This circuit arrangement is shown in *Fig.* 1.

This allowed me to adjust the volume of each channel with the turning of one knob. The control on the unmodified preamplifier was turned up fully. The fact that this worked proved that the preamplifiers are of good design since turning up the gain of the unmodified preamplifier could result in overloading its output circuit if that circuit were not properly designed.

The first preamplifier had to be turned up so that it would provide sufficient

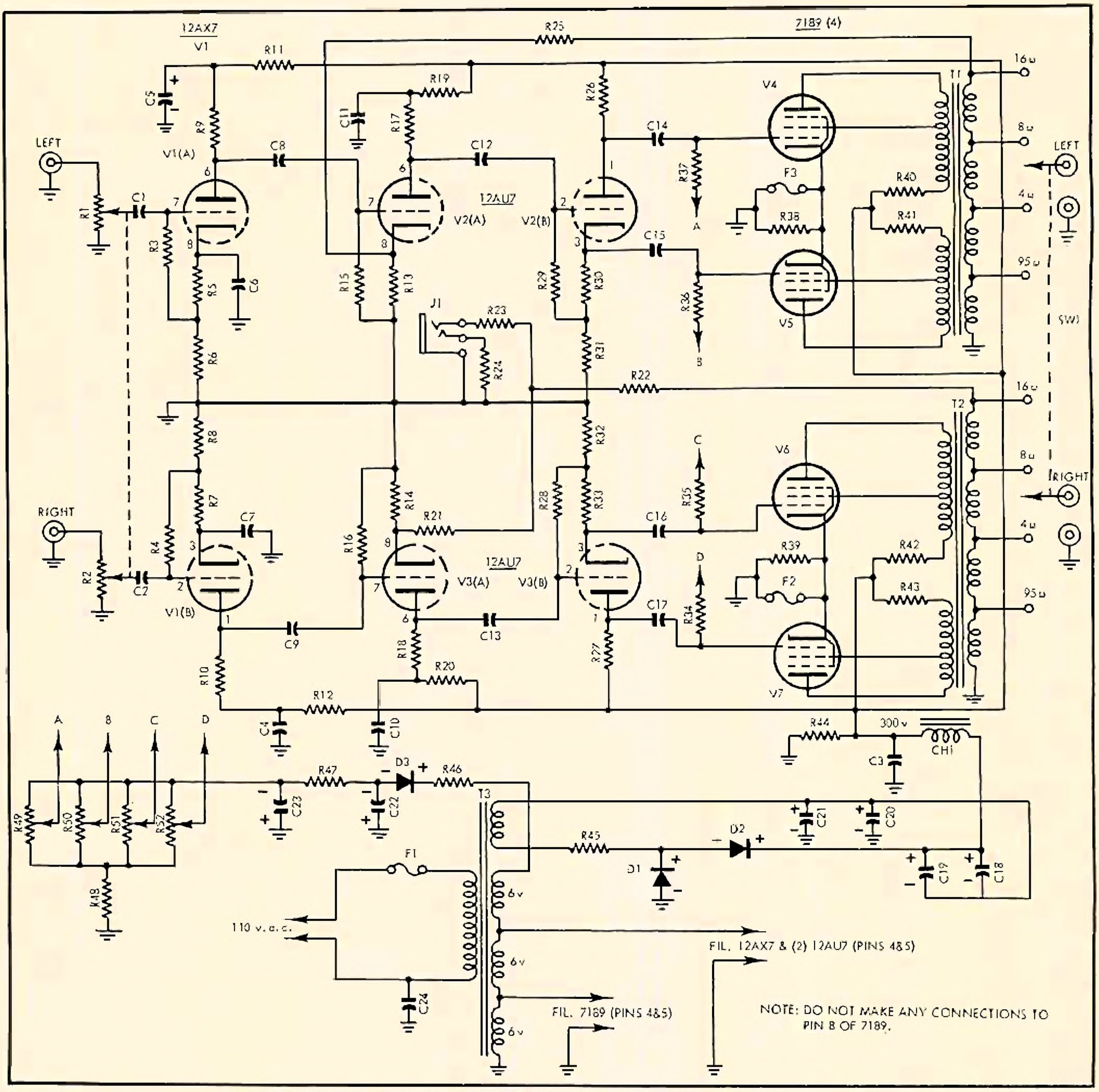

Fig. 3. Above, Schematic of amplifier, and below, the complete parts list.

PARTS LIST

RESISTORS

NOTE: UNLESS SPECIFIED, ALL RESISTORS ARE ONE WATT

R1, R2 0.5 MEG EACH, GANG.
R3 1 MEG
R4 1 MEG
R5 2200 Ω
R6 8200 Ω
R7 2200 Ω
R8 8200 Ω
R9 100 k
R10 100 k
R11 20 k
R12 20 k
R13 5600 Ω
R14 6800 Ω
R15 0.5 MEG
R16 0.5 MEG
R17 100 k
R18 100 k
R19 20 k
R20 20 k
R21 6800 Ω
R22 100 Ω
R23 100 Ω
R24 125 Ω
R25 10.8 k
R26 51 k
R27 51 k
R28 0.5 MEG
R29 0.5 MEG
R30 3600 Ω
R31 47 k
R32 47 k
R33 3600 Ω
R34 270 k
R35 270 k
R36 270 k
R37 270 k
R38 130 Ω, 5 w
R39 130 Ω, 5 w
R40 10 Ω, 2 w, 1%
R41 10 Ω, 2 w, 1%
R42 10 Ω, 2 w, 1%
R43 10 Ω, 2 w, 1%
R44 50 k, 5 w
R45 10 Ω, 10 w
R46 20 Ω
R47 6200 Ω
R48 3300 Ω
R49 50 k, 2 w, WW, POT.
R50 50 k, 2 w, WW, POT.
R51 50 k, 2 w, WW, POT.
R52 50 k, 2 w, WW, POT.
R53 1000 Ω, POT. (REMOTE)

CAPACITORS

C1 0.05 μf, 600 v.d.c.
C2 0.05 μf, 600 v.d.c.
C3 120 μf, 450 v.d.c., ELECTROLYTIC
C4 40 μf, 450 v.d.c., ELECTROLYTIC
C5 40 μf, 450 v.d.c., ELECTROLYTIC
C6 330 pf
C7 330 pf
C8 0.05 μf, 600 v.d.c.
C9 0.05 μf, 600 v.d.c.
C10 40 μf, 450 v.d.c., ELECTROLYTIC
C11 40 μf, 450 v.d.c., ELECTROLYTIC
C12 0.05 μf, 600 v.d.c.
C13 0.05 μf, 600 v.d.c.
C14 0.1 μf, 600 v.d.c.
C15 0.1 μf, 600 v.d.c.
C16 0.1 μf, 600 v.d.c.
C17 0.1 μf, 600 v.d.c.
C18 150 μf, 150 v.d.c.
C19 100 μf, 150 v.d.c.
C20 150 μf, 150 v.d.c.
C21 100 μf, 150 v.d.c.
C22 50 μf, 50 v.d.c.
C23 50 μf, 50 v.d.c.
C24 0.05 μf, 600 v.d.c.

MISCELLANEOUS

V1 12AX7
V2 12AU7
V3 12AU7
V4 7189
V5 7189
V6 7189
V7 7189

D1, D2 SARKES-TARZIAN M 500
D3 SARKES-TARZIAN M 150

F1 1.5 AMP., SLOW-BLOW
F2, F3 0.1 AMP., SLOW-BLOW

CH1 8h, 250 mA, 50 Ω d.c. RES.

SW1 SINGLE POLE 4 POSITION FOR EACH CHANNEL.

T1 PARTRIDGE 5201
T2 PARTRIDGE 5201
T3 TRIAD R-63-B

output to balance the modified preamplifier. When the modified preamplifier is turned up fully, *all* the signal feeds into its power amplifier. Since the two preamplifiers are identical save for the modification of one of them, the unmodified preamplifier would have to be turned up fully in order to transmit equal output to its power amplifier. As the volume of the modified preamplifier is turned down, that of the unmodified one will be attenuated also.

One thing is very important: *The two gangs of the potentiometer must track.* That means that for a given rotation of the shaft, the change of resistance of one section should be the same as the change of resistance of the other section. If the pot does not track perfectly, the volume of the two channels will not be changed equally, and this in turn will cause the system to shift from a balanced to an unbalanced state with various settings of the potentiometer.

One word of caution is in order here. If the leads between this added volume control gang on the modified preamplifier are fairly long—two to three feet—loss of highs may result. When this occurs, shunt the section of the volume control between the arm and the "top" with a small capacitor. The value of this capacitor will be in the vicinity of 50 or 100 pf. The use of low-capacitance shielded cable may eliminate the necessity for the added capacitor.

This preamplifier arrangement does allow for a crude balance between channels which can be achieved as follows: Suppose that the modified preamplifier is connected to the right channel and the non-modified preamplifier to the left. Suppose further that the controls are set as suggested, with the left control up full and the right one set to any convenient volume. We are now listening to a stereo tape and we notice that the left channel is too prominent. All we need do is turn the volume down on the left channel. Suppose, however, that the right channel is too prominent. If we turn down the right channel, the left channel will be attenuated at the same time, and no balance can be established. If the right channel has level adjustments for all inputs, they can be set slightly lower than the same settings of the same controls of the left channel. This would give us a situation in which the left channel is always prominent, and balance can be effected by merely adjusting this channel as necessary.

(Note that many preamplifiers do not have level controls for all inputs, especially the phono input, although such a control is very much to be desired, especially when the phonograph stage is driven by high-input cartridges. Because of this lack of level controls, it was also necessary to assume here that in such a case the balancing must be done from the power amplifier.)

Power Supply

No matter how good the circuitry, the limiting factor will be the output transformer. A good one is likely to be heavy, and two are needed. That means that power supply components should be as light as possible in order to counterbalance the weight of the output transformers. Of course, since you cannot select an output transformer without knowing the tubes with which it is to be associated, that is the next thing to consider. The choice of tubes hinges considerably upon the power supply voltage and current which is to be available. (Remember, I said that the power supply would have to be light.) Since power supply components *can* be very light if a.c.-d.c. circuitry is employed, I began to explore the possibility of this type of circuit arrangement.

If the filtering is good enough, there probably will not be much hum. Series filament strings, however, can be hum producers, but of course there will be feedback and this will counteract this source of hum considerably. On the other hand, I don't like to use feedback for anything but the minimizing of distortion. Often when feedback is used to minimize hum, the hum appears anyway in the form of a background, modulated with the signal. Since this is a power amplifier, it therefore need not possess a great deal of gain: ½ volt is probably more than adequate sensitivity. It will take a lot of doing to develop sufficient hum voltage to bother such an insensitive circuit, but there still will be a possibility of some residual hum. Further, it would be hard to find suitable tubes. The 50C5 would not give us the ten watts we wanted because the plate voltage cannot be made high enough without damaging the tube.

Another disadvantage of this type of circuit is that it is sometimes advisable to ground the preamplifier to a radiator or waterpipe in order to eliminate hum. This could cause trouble with a "hot" chassis of the type associated with a.c.-d.c. equipment if the plug is accidentally placed in the wall socket in the wrong way.

There is, however, a cure for that. Simply remove one prong from the plug, the prong which connects the plug to the chassis side of the line. The ground could be established by connecting the wire which formerly went to this prong to the radiator as mentioned, and the plug inserted into the wall socket. If the polarity is correct, the remaining prong will make contact with the "hot" side of the line and ground will be taken from the radiator. If the polarity is reversed, the "hot" terminal will be connected to ground, and no current will flow.

Since the voltage provided by a.c.-d.c. circuits is too low for a power tube, why not use a voltage-doubler circuit? There are two types, a half-wave, and a full-wave. Each has some advantages. The half-wave type shown in *Fig.* 2 has the advantage that one side of the line can be grounded, making it possible for us to use the scheme with the radiator, but the ripple frequency of such a doubler is equal to the line frequency, and this means that it will be harder to filter than would be true of the full-wave type of doubler. However, the full-wave type has the advantage that the line frequency is doubled, making it as easy to filter as with conventional a.c.-operated power supplies of the type found in most high-fidelity equipment. This is an advantage because the filter components can be smaller, and this can mean a lighter power supply. This type of circuit, however, has the disadvantage that the line cannot be returned directly to ground.

Naturally, I wished to use the full-wave type, so I decided that if there was hum of the type that could be cleared up only by connecting the chassis to the ground system, I would use an isolation transformer.

Why not use a filament transformer as well? (Many of you may say at this point that if I was going to use both a filament transformer and an isolation transformer I might just as well use a conventional power transformer in the first place. The answer to this is simply that the B current is quite high—215 ma or thereabouts—and a power transformer capable of delivering this amount of current would be both costly and bulky.) This would allow me to use better tubes than can be used with the series string. Actually, almost any tubes can be hooked in series-parellel, but I really do not like that kind of string since it does give rise to hum. It is likely that the filament transformer would be small enough to be mounted underneath the chassis, and there is usually quite a bit of space around the power supply section which might otherwise be wasted. (In the final version of this amplifier, the filament transformer is combined with the plate transformer.)

As it finally wound up, I used a power transformer having the requisite filament windings. The final design involved fixed bias for the output stage. I derived this from the filament windings, as shown in *Fig.* 3: three 6.3-volt windings are connected in series to give 18 volts to the bias supply.

The other winding on this transformer gives the voltage and current required by the voltage doubler so that the output from the power supply would be 325 volts under load.

Well, my light power supply did not end up particularly light, but I was

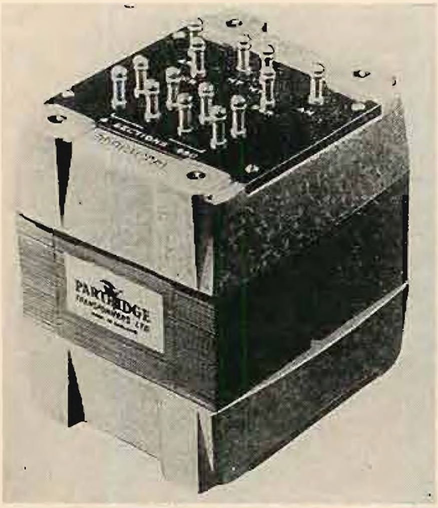

Fig. 4. Partridge transformer, Model 5201.

glad to sacrifice that for the sake of good voltage regulation.

Of course, not everybody who may wish to build this amplifier will care about compactness, so the conventional power supply can be used if the requisite voltages and currents are available. Your power supply may be dictated by whatever parts are available.

Heat must be taken into account, therefore the rectifier tubes were replaced by silicon diodes. Since the circuit was to be a voltage doubler, two such diodes were necessary, and I used the type which can be mounted in a dual holder.

Because this is a voltage doubler, the regulation depends upon the two capacitors associated with the diodes, $C_{18,19,20,21}$. They are rather large. Since they were, the diodes had to be protected against the surge currents which would occur at the instant that the equipment received a.c. A small limiting resistor of surprisingly high wattage rating was placed in series with one side of the line feeding the diodes in order not to ruin them and not to overheat the capacitors. This resistor is R_{36} in the complete schematic of the amplifier, *Fig.* 3.

Output Stage

What type of circuitry should be used? At first a 12AX7 used as a voltage amplifier and split-load phase inverter driving the output tubes seemed to be the most compact idea. The output tubes might be EL84's or the like. Then the thought struck me that there should be a dual power pentode, one tube envelope for push-pull output. A search disclosed the 6360, a 9-pinner which would be compact indeed. This tube, however, is designed primarily for radio-frequency work. The next thing was to ask the manufacturers of tubes what they had in the works, which would be suitable and up came the General Electric 7189, a single power pentode, which necessitated using two per channel.

So far, then, at this point we have decided to use a 12AX7 driving the 7189's. However, we must still design the circuitry for these tubes, a task which cannot be done until we know the nature of the output transformer. Actually, some of the circuitry cannot be designed at all; it must be worked out by the good old trial-and-error method. This is most true of the feedback circuit. The feedback is dependent on both the circuit gain and the characteristics of the output transformer.

(By way of an aside, note that I like to connect speakers in parallel when possible in order to improve the damping of the speakers. The parallel hookup will allow each speaker to act as a load for the others. In addition to this the amplifier will act as a load for all speakers.

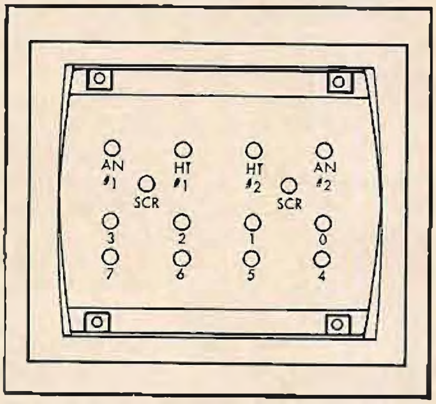

Fig. 5. Connections of Partridge transformer.

When the speakers are connected in series, the total impedance is increased over the value of the impedance of a single speaker, and hence, the loading cannot be as great, which, in turn, will render the damping less effective.

(Of course, when speakers are connected in parallel, the impedance needed to drive them becomes lower than the impedance needed to drive any one speaker. It is obvious that if there are a great number of such speakers connected in this way, the output transformer must have a very low-impedance tap in addition to the standard taps provided on most transformers.)

I wanted to find a transformer with lots of iron, and one wound with heavy wire. I guess I am still old fashioned enough to believe that a good grade of iron and lots of it can go a long way in making a good transformer. Of course, this can be carried too far, but by and large I have found that a lightweight transformer simply will not give the lows with any respectable power level. The heavy wire keeps the d.c. resistance low compared to the inductance, which makes it possible to have an amplifier with as low a source impedance as possible, thereby improving the damping characteristics. As pointed out earlier, I needed one with a low-impedance tap.

I selected the Partridge 5201 output transformer. Its primary impedance is such that it will match 9000 to 12,000 ohms. This transformer is equipped with screen taps which are employed in this circuit to lower distortion. In addition to the 3.8, 8.5, and 15-ohm secondary taps, the unit is arranged for 0.95-ohm loads, ideal for multiple speaker installation or for unique requirements which may not always be foreseen. Of course, the 7189 requires a load of 14,000 ohms plate-to-plate. This transformer is rated at 12,000 ohms; not enough of a mismatch to make me want to use a different transformer. The transformer is shown in *Figs.* 4 and 5.

It is clear that this transformer is ideal for use with the General Electric 7189 when this tube is set up for 10-watt operation. It is also ideal in terms of my other criteria—iron and copper. In addition to all of these good qualities, it possesses extremely low leakage inductance, which means a greater power yield.

Biasing

Once having decided upon the components to be used in the output stage, the design proceeded along conventional lines.

The output stage is wired as a fixed-bias stage, and potentiometers $R_{49,50,51,52}$ are used as fixed-bias balancing controls. $R_{60,61,62,63}$ are incorporated in the B leads to enable measurement of the balance of this stage. (Measurement can be made in the cathode circuit rather than the B circuit. Such a modification should be made as a safety measure. This was not done in my model since it is a throwback to a design using a dual pentode output tube in which the cathode

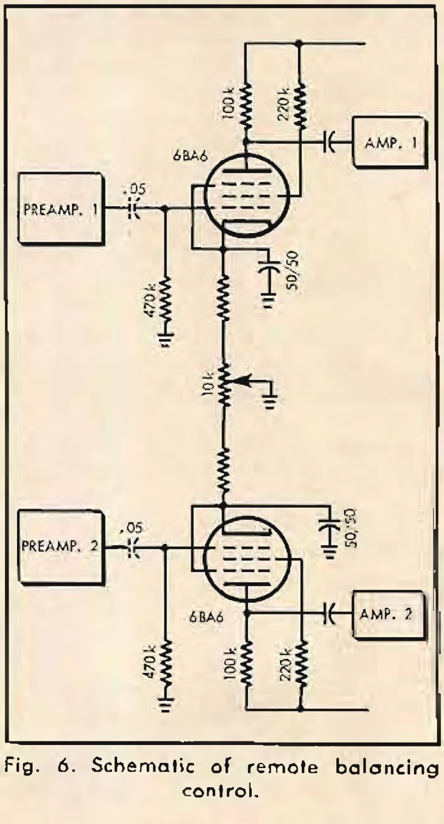

Fig. 6. Schematic of remote balancing control.

was common to both sections.) The transformer primary is a split winding, making it possible to feed B voltage separately to each plate. Balance is obtained when a voltage of 0.43 volt is read across each metering resistor. Since the value of the resistor is 10 ohms and we need a current of 43 ma in each plate circuit, 0.43 volt will be developed across these resistors when the correct amount of current it flowing. You will notice that there is some interaction between the balance pots, so balance must be repeated several times before the stage is truly adjusted.

Be sure that the potentiometers are set for maximum bias before voltage is applied to the stage so that the amplifier will not draw excessive current prior to balancing. However, failure to do this probably will not result in damage to the 7189's because the bias circuitry was designed so that the voltage does not go completely to zero when the pot is set to the minimum voltage positions. This was accomplished by raising the low ends of the pots off ground and returning them there via a fixed resistance voltage divider, R_{48}. Be sure that no "hot" portions of your voltmeter come in contact with the chassis of the amplifier and that you do not touch the chassis and the meter while balancing the stage.

The question might well be asked at this point: "Why make your measurements in the B leads when all you needed to do was to measure the voltage from the arms of each balance pot to ground?" This could be done if tubes were made exactly as specified on the data sheets. Unfortunately, equal bias voltages do not necessarily indicate equal plate currents. Equal plate currents are very important in an output stage in order to minimize saturation of the iron in the output transformer. Assuming that the tube is within tolerance at its installation, it is nearly certain that it will not remain right on the nose. As filament emission falls off, each pentode section will almost certainly age differently from the other. This is not a flaw in the design of the particular tube; it is simply the nature of vacuum tubes to perform in this manner. With all measuring done in the plate circuits, however, both the amount of current and the equality of that current in each tube section can be checked.

Driver Stage

Originally I had hoped that the input circuit for this amplifier would consist basically of a 12AX7 driving the 7189's. However, the 12AX7 stage did not work so well as I had hoped, and I tried a 12AU7, which gave me the distortion characteristics I wanted, but did not provide sufficient gain. Therefore, a 12AX7 dual triode was added ahead of the 12AU7, one triode used for each channel. In order that the feedback should remain stable, I did not include the 12AX7 triode in that loop. However, it was obvious that this triode gave more gain than was needed. I realized that I could use a 12AU7 in place of the 12AX7, but it seemed more practical to use the 12AX7 and reduce the gain by means of some cathode degeneration, thereby reducing the distortion produced by this stage. This explains the reason for the input circuitry of each channel.

The cathode-to-ground circuit provides the degeneration required for improved distortion characteristics plus some frequency compensation at the upper end of the spectrum beyond 40,000 cps. I elected to place the frequency compensation at this point in the circuit rather than in the main feedback loop. Otherwise changes in the setting of the balance control would cause changes in frequency response in the main loop of the right channel.

Fig. 7. Remote balancing control with cable.

Remote Balancing

There was one requirement which had not been met as yet; namely, balancing the channels from a remote location.

One possible method was to balance the circuit at the speakers by means of a ganged potentiometer circuit, but this method was too wasteful of power. A friend suggested that I balance the circuit in the cathodes of remote cut-off pentodes. The balance control would not have to handle any signal, just d.c., and low-voltage d.c. at that. The scheme is shown in *Fig.* 6. The values are arbitrary.

I never did build the circuit, but it does seem reasonable. If you have a stereo system and have no means for balancing it, this circuit can be made to work very well with a bit of experimenting. Just be sure that you keep the signal level to the input of this balance network to less than one volt in order that distortion can be held down to reasonable levels. It should be connected between the amplifier and the preamplifier and mounted close to the amplifier. If it must be located somewhere remote from the power amplifier, I recommend that you follow the balancing stage by a 12AU7 cathode follower, one section feeding from each 6BA6. Since most preamplifiers are equipped with cathode follower outputs, it is not particularly important that this circuit be located near them.

The use of this arrangement called for two additional tubes and I did not like that, especially because the distortion would probably be higher in view of the nature of this circuit. I never built this arrangement, as I have said, and so did not verify this fact. Of course, I realized that I could omit the 12AX7 because of the gain provided by the 6BA6's. However, I elected to use a simpler system.

An examination of *Fig.* 3 will show that the two channels are nearly identical except with regard to the main feedback circuit. One of them is conventional, save for the fact that there is no feedback capacitor shunting the resistor. The characteristics of the output transformer used here are so exceptionally good that stability was excellent well beyond the 20 db of feedback employed in this amplifier. By eliminating this capacitor it was possible to obtain the high-frequency performance which is shown in the PERFORMANCE DATA section of this article.

The right channel is not like this one in terms of the feedback network, although it still does eliminate the feedback capacitor. In order for me to explain the differences, look again at the conventional feedback loop used with the left channel—the channel not containing the balance jack. This channel was adjusted to have a feedback of about 20 db. I decided that the balance could easily be achieved by slightly varying the feedback of the right channel. A variation of 6 db of feedback would be all that would be required to account for differences in balance between channels of the program sources used, all other things being equal. Therefore, the feedback of the right, or balance channel would have to have a feedback loop whose gain is adjustable from 17 db to 23 db, which gives us a gain of plus or minus 3 db with respect to the other channel. Therefore, I juggled the values of cathode and feedback resistors in the unconventional channel so that when no plug is inserted in the balancing jack, the feedback of this channel is equal to that of the left channel. All that was required now was to find a means for adjusting the feedback in the balance channel, and of course this would have to be done with a potentiometer in order that any desired amount of feedback between 23 and 17 db could be obtained.

Suppose we take the 16-ohm tap of the output transformer through 100 ohms to the feedback loop. From the junction of the 100 ohms and the feedback resistor we take another resistor also of 100 ohms and connect it to ground. Notice that this is the same as saying that the feedback is connected to the tap on a voltage divider. Because the value of each of the resistors in the divider are equal, the voltage between the tap and ground is half that of the voltage across the entire divider. Half the voltage represents a loss of 6 db at this point. This, therefore, represents a loss of feedback of 6 db. Now suppose we lift the low side of the grounded 100-ohm resistor and connect this to one side of a variable rheostat, and ground the other side of this rheostat. When this rheostat is at maximum resistance (assuming that the resistance of this rheostat is reasonably high—1000 ohms) the voltage divider action will be virtually nonexistent and full feedback will be applied to the input. (Naturally, the values of cathode resistor and feedback resistor are adjusted to provide 23 db of feedback under these conditions.) However, when the rheostat is at its zero resistance, the 100-ohm resistor is again grounded, as has been shown earlier, and the feedback has been reduced by 6 db. Intermediate settings of the rheostat, or potentiometer wired as a rheostat, will produce intermediate degrees of feedback.

In the present circuit this potentiometer is not connected directly to the circuitry but is applied through a closed circuit phone jack. When this phone jack is inserted, the potentiometer is connected as described. However, when the potentiometer is removed (by removing the plug from the jack) the tip of the jack is grounded via an additional 100-ohm resistor. The purpose of this arrangement is to reduce the feedback by approximately 3 db so that when the balance jack is removed from the circuit, the gain in each amplifier channel will be about the same.

When you do use the balance control, you can sit in your favorite listening position and adjust from there, and that will eliminate most of the annoyance of having to balance stereo sources. It might well happen that you would not want to use the control at all times. Portable operation might be the reason for this, or it might be that the wires on the floor would interfere with your teenagers having a dance, or who knows what. Then all that is required is to remove the plug from the balance jack and the amplifier will operate in the normal manner.

For protection and ornamentation the balance pot should be enclosed in a container, such as the Bud CU2100 Minibox used with my instrument, *Fig.* 7. When mounting the pot and cable, be sure to provide an anchor for the cable so that it cannot be pulled from the lugs of the pot. A cable clamp is a good method since, in addition to preventing the cable's accidental detachment by a sudden pull, it prevents twisting, which would eventually break the connections.

Comments

Good grade components should be used for this will keep maintenance to a minimum. Should trouble develop, however, the voltage and resistance chart in Table II may be used as a check list.

Performance Data

Naturally, once I completed the unit described here, I hoped it would "measure out." Let's not kid about it; the very first model was not a success, but the one shown here was definitely a success as can be seen from the table of performance data obtained with fine measuring equipment. A Heath AG-9 audio oscillator was used in conjunction with a General Radio Wave Analyzer and Tektronix 'scope.

TABLE I

Performance Data

Interchannel cross-talk	60 db
Signal-to-noise ratio at one watt output	60 db
Voltage regulation from zero to maximum drive—both channels	1.4 per cent
D.c. transient response	One cycle of ringing
Squarewave response	Excellent—0.1 μf across the output made no significant difference here
Frequency response	Flat 20 cps to 40,000 cps; ± 1.5 db from 10 cps to 90,000 cps
IM distortion using 60 cps and 7000 in 4 : 1 ratio	0.3% at 14 watts
Harmonic distortion (14 watts output)	20 cps, 1%; 50 cps, 0.2%; 1000 cps, 1%; 20,000 cps, 1.5%
Voltage sensitivity for maximum power output	0.8 volt
A.c. supply requirements	110–120 volts at 0.8 amp

One of the tests shown here may not be familiar to you and should therefore be explained. This is the d.c. transient response figure for an amplifier. Most of you are already familiar with the transient response at high frequencies. It is desirable that an amplifier reproduce the starts and stops of program material with no lag and no overhang added to the signal by virtue of shortcomings in the amplifier circuitry. The frequency response of the amplifier should be at least five times that of the highest frequency to be reproduced if good high-frequency transient response is to be obtained. There is a phenomenon similar to this which occurs at the low frequencies, and is observed on occasions such as when the tuning dial of an FM tuner is adjusted. Some amplifiers do not respond to this adjustment. The effect is that the signal is turned off or reduced in volume during the tuning operation. It is often necessary to reduce the bass response during this adjustment. Obviously, tuning is not as easy as it should be. Even when this type of distortion is not severe enough to make tuning difficult, it can muffle the true character of a really low bass note. This distortion is often produced by too large a time constant in some of the coupling circuits. When a signal is applied to such a circuit, the grid to which it is connected goes negative and the tube draws less plate current. This continues until the charge has had time to leak off through the grid-return resistor. Defective output tubes can also cause the same condition.

The object of the test is, naturally, to see if such overloading exists. It is performed as follows: A load of the

TABLE II

Voltage and Resistance Values

PIN	1	2	3	4	5	6	7	8	9
V1	150k 210	1 MEG 4.7	9500Ω 7.4	0.2	0	150k 210	1.1 MEG 4.8	10k 7.8	5
V2	100k 210	500k 72	50k 94	0.2	0	150k 118	500k 0	3800Ω 5.8	6
V3	80k 210	550k 72	50k 94	0	0.2	150k 118	500k 0	3600Ω 5.8	
V4	300k -11.6	300k -11.6	3.4 0.6	0	0	50k 310	50k 305	0 0	50k 310
V5	300k -11.6	300k -11.6	3.4 0.6	0	0	50k 310	50k 305	0 0	50k 310
V6	300k -12.4	300k -12.4	0.8 0.6	0	0	50k 310	50k 305	0 0	50k 310
V7	300k -13.6	300k -13.6	0.8 0.6	0	0	50k 310	50k 305	0 0	50k 310

LOW POWER STEREO AMPLIFIER

proper type is connected across the output terminals of the amplifier, and this load is shunted by the 'scope. So far this is, of course, normal test procedure. The input is connected to a d.c. source which can be interrupted by some sort of switch. A telegraph key is excellent for this purpose. The input d.c. voltage is adjusted to that value which was found to give maximum power output from the amplifier commensurate with low distortion. This voltage is alternately applied and removed by means of the switch, and the trace from this is observed on the 'scope face. There should be only a slight, sharp blip on the tube's face each time the current is applied and removed —in the ideal amplifier. Some amplifiers will show several cycles, each one of which has a lower amplitude than its predecessor. The amplifier described in this article showed approximately one such cycle, which is excellent.

I believe that it is also in order to mention that the power output was measured using an ohms reading. No attempt was made to measure with "music" output; I believe that this entire concept is misleading. If the amplifier is of fine quality, there will be no difference between the "music output" rating and the conventional methods of measuring power output.

You may ask what the reason for this is. Well, the music output is taken as follows: The amplifier is set to read an output of x per cent distortion. A reading is then taken of the plate voltage, with the signal still applied. Before the signal is introduced, the plate voltage is measured. If there is any difference in plate voltage, the difference is then made up and a new test is run. Naturally, somewhat more output can be obtained in this manner. The procedure is then repeated until the absolete maximum is reached. This kind of thing can make a poor amplifier have the same power output as a fine one. A fine amplifier will exhibit little plate voltage fluctuation with or without signal, while the poorer amplifier will exhibit a marked voltage fluctuation. This difference is compensated for during the test, and that is why I said the poorer amplifier is given a better chance than it deserves.

To be prefectly fair about it, the proponents of this test procedure believe that the voltage will not sag during actual use of the amplifier. They contend that the amplifier draws its peak power in program peaks, and the charge on the capacitors will be sufficient to handle such peaks. Whether this is true or not, the amplifier is given the benefit of the doubt in this test, and it does allow a claim for a higher power rating than would otherwise be possible. Perhaps this is all right as long as everyone plays the game, but the poorer amplifier nonetheless has an advantage always which it could not and probably should not enjoy.

For this reason I included the voltage regulation of the amplifier, measured with both channels operated at full power output: 14 watts per channel.

All other tests were standard and will not be discussed for that reason.

Experiments were performed with the channel containing the balance feature, and it turned out that the distortion figures for that channel were the same as for the one which was a straightforward circuit, and this was true regardless of the setting of the balance control, from maximum to minimum.

It should also be mentioned that the signal-to-noise ratio was taken at a power level of one watt so that a more realistic picture of the amplifier performance would be obtained. A somewhat higher figure could have been obtained at the maximum output of 14 watts, but I assume that average listening level is one watt or less, so used that figure in the tests.

An amplifier capable of this kind of performance can be used with the most inefficient speakers available. An amplifier with higher power, but at the cost of higher distortion, would probably not be used to supply more than this 14 watts of relatively distortionless output to the speaker. Some of the speakers used in association with this amplifier are quite inefficient, but can be easily operated with this amplifier.

Well, there it is. Build it and use it well. I have gone into considerable detail in this article to show you some of the workings of a designer's mind. After following the various steps in my reasoning in building the unit, you probaly will appreciate the new stereo power amplifier kit you were thinking of buying, and can send a few good wishes to some of the engineers whom most of us never think of when we put the plug into the wall socket and listen.

This amplifier was designed with my own special needs in mind, but I did take into account its general use once I decided to write it up. In the same manner, a manufacturer is faced with the problem of trying to please everybody and do it competitively. Some people want simplicity while others wish elaborate flexibility and myriad input connections. I hope that you can look upon an amplifier or whatever with the knowledge that much work went into its development. Maybe there is something about it that you do not like, but that something may be just the thing for which someone else bought that same piece of equipment.

That is where the fun of building your own unit comes in. What I have given here are my own ideas of how a unit might be conceived and built. Perhaps there are features which you want added. Perhaps you wish to take away something from the unit. Maybe you have a preamplifier with a stereo focus control, and so you do not need the balance feature. In that case you can adjust each channel to have 20 db of feedback. J_1 can be removed then.

Perhaps you will not want a level control in each channel. You can use a fixed load resistor instead, and with a voltage divider you can provide two input connectors, one for full sensitivity of the amplifier and one for reduced sensitivity.

Certainly you may wish to change the power supply system used here. Perhaps you have a power transformer from an old TV receiver. This will work very well, but you will have to allow for the extra space in the layout required by this type of transformer and its associated components. The diodes probably will have to go unless you use several of them in series in each branch of the full-wave rectifier. If you elect not to use the diodes, you must revert back to the tube-type rectifiers and the heat from them must be taken into account.

Perhaps some of you are not interested in stereophonic reproduction. In this case you need build only one of the channels.

These are but a few of the possible variations of the simple amplifier circuit. If some one of them appeals to you, try it. It seems to me that an article is merely a springboard for one's own thinking. More can be learned by adapting it than can be learned by adopting it. Æ