A Transistorized Stereophonic Control Unit

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Stereo controls and their functions have been a source of confusion for the audiofan. The following analysis dispels some of the confusion and leads to the design of a high quality preamp.

STEREOPHONIC SOUND arrived suddenly, perhaps unexpectedly, transported by the media of compatible disc recordings, FM multiplex, and simultaneous AM–FM and AM–TV transmissions, and multitrack magnetic tape recordings. While each of these techniques requires a different method of conversion from electromagnetic or acoustical information to an electrical signal, all have the common characteristic that two similar but separated audio channels are required for compensation and amplification. This article will discuss the design of a stereophonic control unit by examining the problem in two parts: first, the philosophy of stereo reproduction and the functions of the various controls which are necessary to such reproduction; and second, the actual circuit design and construction of such a control unit, incorporating transistors for increased reliability, low hum and microphonics, and compactness.

After a brief examination of the stereophonic control units on the market, one would be forced to the conclusion that the transition from monophonic to stereophonic reproduction entails an increase in the complexity of controls by a factor of at least two-to-one, and perhaps greater. This observation is supported by the initial approach of most manufacturers of stereo equipment, namely, to provide two of everything plus some peculiar additional functions. While "two-of-everything" is certainly a straightforward solution, it does not reflect very favorably upon the engineering "know-how" of these manufacturers, since such a solution does not imply anything more than a very superficial analysis of the problems of stereophonic reproduction. In lieu of any further criticism of what has not been done, let us proceed instead with a first-order analysis of what should be done, and reflect this analysis in the design of a control unit intended for stereo, not merely a double-barreled approach to monophonic sound.

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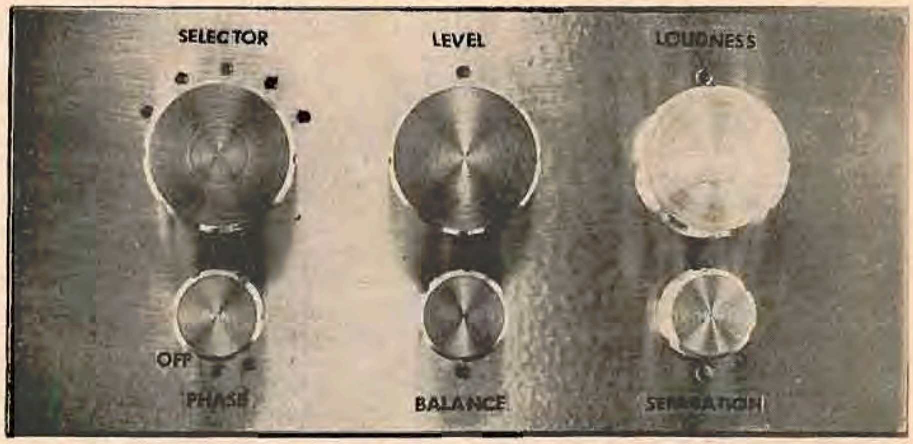

Fig. 1. Front panel of control unit.

Audio Controls in General

Audio control functions in a stereophonic system may be divided into two general categories: first, those functions which are common to all types of audio reproduction in general, and hence can be controlled in both channels simultaneously; and second, those functions which are peculiar to stereo, and which may require some definition before they can be appraised. In the first category we find: (1) input selector switching, (2) phonograph record and magnetic tape playback equalization, (3) master level control, (4) loudness contour compensation, (5) tone control, (6) scratch and rumble filtering, and (7) power switching.

It is often convenient to incorporate the first two functions, input selector switching and equalization compensation, into a single control, especially since most modern monophonic disc recordings and virtually all stereophonic discs are recorded with the RIAA characteristic, and the NARTB recording curve has become standard for tape recording at the 7½ and 15 ips speeds. The selector-equalizer should thus control at least three low-level inputs: a high-impedance microphone input, a magnetic phono input which incorporates RIAA equalization, and a magnetic tape head input with NATRB equalization. At least two high-level inputs are also necessary, one for a tuner, and one to be used for pre-equalized tape playback or a high-level phono cartridge. While level control of each input would be a pleasant luxury, it is desirable to limit the number of level pots to one per preamp channel, so as to be able to adjust the output of the low-level preamp to approximate the signal at the high-level inputs, thus avoiding sudden changes in volume when switching from any of the low-level inputs to a high-level signal.

Level control needs little discussion, except to point out that such a control should be ganged to both channels, avoiding the inconvenience and added complexity of separate concentric controls and unreliable mechanical clutches which are supposed to permit operating the two knobs in a ganged fashion.

Loudness contour compensation and tone controls may usually be considered simultaneously, since in the ordinary listening environment either will produce a satisfactory result, and in installations where both are present it is common that the loudness control produces such satisfactory results that the tone

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quence of the isolating resistors R_{30} and R_{32}, so that the unit should work into an impedance of at least 50,000 to 100,000 ohms. If a really low output impedance is desired, and this may often be the case, the present outputs may simply be followed by emitter-followers, one for each channel, identical in design to the Q_3 or Q_5 stages. For a typical installation where long cable runs are not necessary, the unit is quite acceptable without additional emitter-followers, and the treble loss and hum pickup will be negligible.

The phase-reversal feature, as discussed in part I, has been retained for separate treatment. The circuit is a "split-load" or "matched-load" phase splitter of a configuration which is common in vacuum-tube power amplifiers. When this circuit is inserted in ONE channel just ahead of Q_3 then the output at the collector of the reverser corresponds to an inverted signal with the same amplitude as the input, and the output at the emitter is a signal identical to the input. The Z_{in} of this circuit is the same as that of the Q_3 stage, namely 100,000 ohms. The design of this reverser is identical to that of a grounded-emitter stage with a voltage gain of 1, taking into account the fact that the equivalent load and unbypassed emitter resistors change, depending upon which output is being used. The operating point of Q_6 is:

$$V_{ce} = -10 \text{ vdc.}$$
$$I_c = -1.1 \text{ ma.}$$

The section of S_2 which selects either output, is one deck of the power switch. Thus S_2 is a 2-pole, 3-position switch, and the three positions are: (1) OFF, (2) ON, PHASE NORMAL, and (3) ON, PHASE REVERSED.

One additional feature bears a passing examination; the "third-channel" output. Because of the construction of the separation control R_{33}, there is always a signal available which is the sum of the signals in the two channels, and hence is monophonic, representing the sound from the "middle." This signal may be used to provide a "phantom" or third channel by the application of a third amplifier and speaker; however, the impedance at this point is very high, and any loading of less than 1 megohm will reduce the sum-signal level and disrupt the operation of the separation control. Therefore, if this third channel is to be used, it would be advisable to follow it with a very high input impedance emitter-follower or a cathode-follower, and to implement this, a schematic of such an isolation amplifier is appended as *Fig.* 7.

The performance of the high-level portion of the control unit may now be tabulated:

(A) *0-db Loudness contour:*

- Frequency response flat ±1 db. from 20 to 20,000 cps.
- Sensitivity 100 millivolts for 1 volt output.
- Intermodulation Distortion 0.6% for 1 volt output.
- Harmonic distortion not measurable at 1 volt output.
- Noise level more than 65 db. below 1 volt output.
- (This noise is characteristically below 30 cps.)

(B) *−35 db Loudness contour:*

- Follows Fletcher-Munsen contour ±3 db.
- All other applicable specifications are equalled.

Construction

The circuitry of the author's control unit was laid out on three 2¾ by 6¾ in. perforated bakelite resistor boards; one channel on a board, except that the

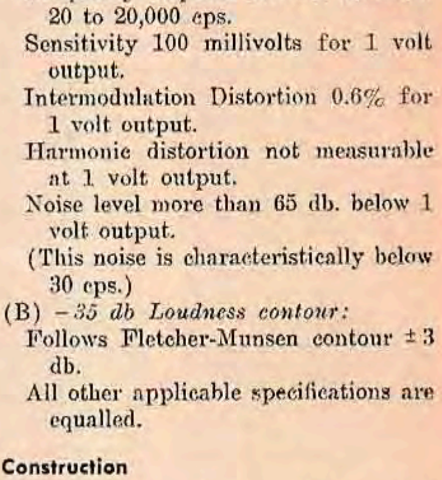

Fig. 6. Schematic of the control unit power supply.

PARTS LIST

C_1, C_2, C_3	100 µf, 50 v, electrolytic
CR_1	10-ma, 75-PIV rectifier, selenium or silicon
F_1	3-amp fuse
K_1	DPST relay, 26-volt a.c. coil, 3-amp contacts (Potter & Brumfield GA-11A)
L_1	20-H, 15-ma choke (Stancor C1515)
R_1	500 ohms, 2-watt potentiometer, 10%
T_1	Transformer, 26.5 volts, 0.5 amps (Thordarson 21F27)

equalization networks and loudness-control components, and the entire phase-reversal stage, were placed on a third board. It is interesting to note that printed circuits would be ideal for this application, and could probably be made using some of the kits which are presently available. The author's three resistor boards were mounted inside a 3 × 6 × 7 in. aluminum chassis as shown in *Fig.* 4, with one 3 × 7 in. panel serving as the front and the other as the terminal panel, *Fig.* 5. To achieve a professional-looking unit, a 3½ × 7½ in. brass front panel was added, rubbed to a satin finish with crocus cloth and then steel wool, sprayed with Krylon clear plastic and then letter decals added, followed by about six more coats of plastic spray until the surface was smooth over the decals. *Figure* 1 shows the finished appearance. The author used machined Dural knobs to further enhance the appearance, but these are inordinately expensive to purchase, and any knob would suffice.

The schematic of a simple remote power supply, *Fig.* 6, is included without lengthy explanation. The supply is switched on and off by controlling the current in the coil of relay K_1, so that there is no 115-volt a.c. power inside the

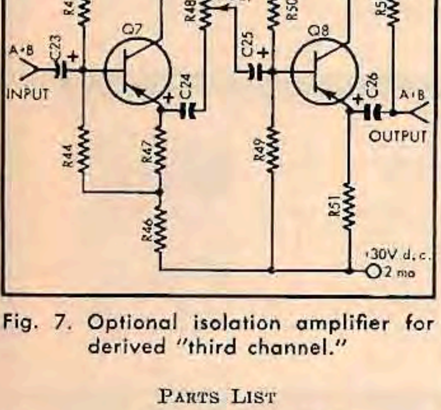

Fig. 7. Optional isolation amplifier for derived "third channel."

PARTS LIST

C_{23}	0.22 µf, 100 v, paper
$C_{24, 25}$	2.5 µf, 25 v, electrolytic
C_{26}	25 µf, 25 v, electrolytic
R_{44}, R_{50}	180k, ½ watt
R_{45}	470k, ½ watt
R_{46}	18,000 ohms, ½ watt
R_{47}	10,000 ohms, ½ watt
R_{48}	100k, potentiometer, linear
R_{49}	220k, ½ watt
R_{51}	27,000 ohms ½ watt
R_{52}	1 meg, ½ watt
Q_7, Q_8	PNP transistor, (GE 2N265)
	(All resistors 10%)

control unit at all. This supply will furnish +30 volts d.c. at the total current drain of approximately 7 ma required for the control unit; the output voltage is adjustable by R_1, and the total ripple is quite low, well below 1 mv rms. The supply voltage could be obtained in any of a number of alternative ways, the

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only restrictions being that the ripple must not exceed 1 mv, and the supply voltage must never exceed +35 volts, on pain of a damaged transistor. These requirements are so slight that the voltage could be obtained by a simple bleeder from the power supply of a power amplifier, or even from a battery.

The author's control unit has been in operation for twelve months at the time of writing, and has performed entirely satisfactorily. There is a complete absence of hum, hiss, or microphonics, and no audible distortion. The controls have justified the time spent in selecting them, in that the system has proved entirely satisfactory for both stereophonic and monophonic applications. The over-all cost of all parts, including the power supply described, was about $75, which is certainly comparable in price with the stereo preamplifier kits on the market—and there is no doubt that the kits could be assembled in quantity for a smaller price. Whatever the cost, the convenience of a high-performance control unit with only six knobs is one which cannot be bought on the market, at any price.

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controls soon become dusty with disuse. For the purposes of a compact installation, then, the tone controls may be eliminated from the control unit and made a part of the power amplifier, wherein they are adjusted to compensate for unusual room acoustics and then left set; in many cases tone controls may be eliminated from the system entirely. In any case, the loudness contour is a necessity, and should be capable of reproducing the Fletcher-Munsen equal-loudness contours from the 0 db, or full room volume, curve to a fairly low listening level, say -35 db. It is also possible to consider scratch and rumble filtering under the general heading of tone control, but logical consideration of this topic soon leads to the conclusion that the audiophile who is interested in building his own control unit is probably using a professional turntable or high quality changer, and is concerned almost entirely with the reproduction of high fidelity program sources, thus predicating the proposition that the inclusion of filters designed to compensate for the shortcomings of lower quality equipment and low fidelity sources, is a waste of effort in an ostensibly high fidelity system. Even the enthusiast who may wish to play badly worn or scratched discs which are collector's items isn't likely to find much utility in such controls. He is more apt to dub these irreplaceable recordings onto tape using special filters as a part of the recording system, and then play the tape to save wear and tear on the originals. The conclusion to be drawn from this argument is that the elimination of rumble and scratch filters from a stereophonic control unit intended primarily for the reproduction of new, high quality recordings with professional quality equipment will hardly impose any hardship by limiting the flexibility of the control unit.

Power switching is a subject which could be expanded into a volume by itself, but in this control unit such switching will be confined to controlling the d.c. power for the unit itself, plus one additional circuit which may be used to control the coil of a power relay, and this in turn controls all the a.c. power to amplifiers, tuner, turntable, and tape or other auxiliary equipment. This is a more desirable situation than trying to mount a switch to handle as much as half a kilowatt within the control unit itself, and if the d.c. power is used in the relay control circuit, then there will be no a.c. power within the control unit except signals, with attendant advantages in hum pickup reduction.

The preceding analysis has thus reduced the original seven monophonic functions to four: (1) a combined selector-equalizer, (2) a master level control, (3) a loudness contour control, and (4) power switching. In each case, where there was doubt about the inclusion of a function, assumptions of high quality equipment and logical function were advanced in order to determine whether such a function was a necessity or merely a luxury.

Stereo Controls in Specific

A second set of control functions becomes necessary with the advent of stereophonic reproduction. Such terms as "balance," "separation," "phase reversal," and "channel reversal" have begun to appear in the profusion of recent literature on the subject of stereo, and the consequent confusion about their meaning warrants a brief definition of each, as well as an examination of their importance.

Balance means simply a comparison of the relative volume levels of the two audio channels; if the inputs are assumed equal, as is usually the case, the term "balance" describes the gain ratio of the A and B channels. Since the overall gain of each channel must necessarily include everything from the cartridge to the acoustic output as it reaches the listener's ears, it seems obvious that even if all the reproducing elements and amplifying elements are exactly balanced, the position of the listener within the room may be such that the sound does not seem balanced to him. Moreover, even the assumption of balanced inputs is unrealistic, thus the necessity for a control to adjust balance is compelling. Such a control usually operates so as to vary the gains of the two channels differentially; that is to say, to increase the volume in one channel while simultaneously decreasing that in the second channel.

"Separation" is a term which defines a more subjective phenomenon of stereo; that is, the separation of the two halves of the apparent source of sound. The two limiting cases of this effect are easy to imagine: on the one hand, a total lack of separation would cause the apparent source of sound to seem to be located at a point midway between the two speakers; the other extreme would occur when the speakers were located so far apart that there seemed to be a "hole-in-the-middle" of the sound emanating from the left and right hand speakers, giving an exaggerated stereo or "ping-pong" effect. Since this effect depends upon the electrical channel isolation, on speaker placement and room acoustics, and, in fact, on the microphone placement in the original recording studio, it would seem almost a necessity that some means of continuous adjustment of the separation should be included. One means by which this adjustment can be achieved electrically is by adding a portion of the signal from one channel to the signal in the other, and vice-versa. Each channel would then contain some information which is common to both and hence is monophonic in character, serving to fill the "hole-in-the-middle"; each channel would also contain some information which is peculiar to that channel and hence retain its stereophonic character. Using this control it would be possible to attain a "curtain-of-sound" spread across the space separating the two speakers, regardless of variation between recordings, room acoustics, and so on. It is scarcely necessary to mention, however, that the adding type of control can only reduce the separation of the two channels, since if it could increase the separation beyond that of the recording it would be possible to make monophonic recordings sound stereophonic! There have been circuits designed which use a matrix to take advantage of the separation already inherent on the stereo recording and improve upon it, but such circuits are nearly impossible to construct, except on paper, because of the extreme precision required.

The third term mentioned was channel reversal, which means simply the reversal of the two channels in left-right orientation, and is a purely aesthetic effect which assumes prior knowledge of the subject arrangement. An illustration of this effect would be a recording of a symphony orchestra, which has standardized so that the strings are to the left of the conductor, the percussion to the right, and so on. In the event that this is reversed in the reproducing system, it is a simple matter to correct, since the two channels may be interchanged at any point in the system, from the pickup leads to the loudspeakers. In view of this fact, a separate control to accomplish this function is not felt to be important enough to merit inclusion on any save a very elaborate control unit, and such a control will not be included in this case.

The fourth term, "phase reversal," is perhaps the most difficult to describe. The need for attention to this function arises because of the relative youth of the stereo art, so that there is no universally observed standard as to what direction of excursion of the phonograph stylus shall produce a signal of a given polarity; or in the case of magnetic tape, what state of magnetization of the oxide layer on the tape shall produce a signal of the same given polarity. In a purely stereophonic system, (i.e., the channels are assumed to have perfect isolation), the result of improper phase orientation will be cancellation of some of the signals. In the case where there is electrical adding of the signals, as in a separation control, the out-of-phase components will cancel and reduce the overall signal level; and in the case where the signals are not added electrically there will still be acoustic interference of the same sort which arises in an improperly phased public address system. The remedy is simple in principle: just reverse the direction of excursion of the speaker cone in one channel for a signal of a given polarity, providing that no electrical adding takes place. In the case where there is adding, then the phase of the signals in one channel must be reversed before the addition, either by interchanging the pickup leads for one channel, or by inserting a stage with a voltage gain of −1 in the voltage amplifying stages of the preamp; and since the introduction of a switch to reverse pickup leads is more than likely to introduce hum, and may not be possible with a three-lead system, a stage with a gain of −1 is by far the better solution.

The Control Unit

We have thus arrived at a realistic complement of controls for a flexible yet compact stereo control unit, see *Fig.* 1. The front panel contains six controls, four of which operate upon both channels simultaneously to cover the important monophonic functions, and the remaining three control the essential stereo functions. (This totals seven, but the phase reverse and power switch functions are combined in one control.) The remainder of this article will describe the design and construction of the control unit, incorporating all the electronic refinements which are consistent with high quality, reliable performance. This design takes advantage of the freedom from hum and microphonics which characterize the transistor, an attribute which provides a convenient solution to the problem of the low-level output of most stereo cartridges. The small size and weight, and the lack of heat generation of the transistor will also serve us in good stead so far as a compact design is concerned, and the reliability of a properly designed transistor circuit is such that it should practically never require service.

DESIGN DATA FOR THE TRANSISTOR CONTROL UNIT

The first portion of this article has discussed the philosophy of a simple but adequately flexible stereo control unit. After examination of the requirements for such a control unit, a set of functions was specified which would meet the needs of nearly any audiofan. The design problem now becomes one of electrical realization of the functions thus specified, and of high-quality, high-reliability audio equipment design in general. In considering each channel separately we must observe the performance criteria which have become standardized in spe-

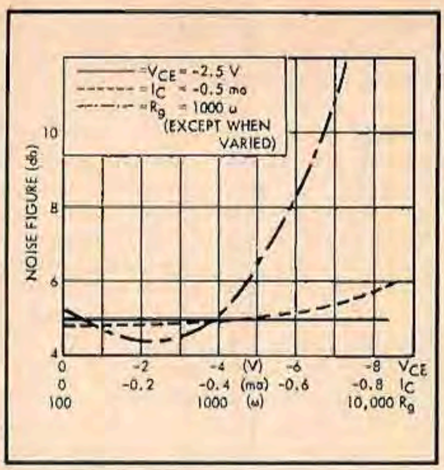

Fig. 2. Typical noise characteristics of a low-noise germanium transistor.

cifying the performance of monophonic equipment; the design is specialized only insofar as the peculiarities of stereo are concerned.

At the time that the idea for this control unit was conceived, it was decided that such a unit should be transistorized. Many reasons for such a decision may be advanced, and several of these merit examination. First, when a transistor is operated within its conservative ratings, it has an extremely long lifetime, or "mean-time-to-failure." For a high-quality transistor this lifetime approaches that of a passive device such as a resistor or capacitor, and even for the lower-priced transistors which we shall use, months or years will elapse before a typical transistor will require replacement. Second, the transistor and its associated circuitry are small and generate almost no heat. This means that the advent of stereo, requiring almost twice the equipment that monophonic reproduction entails, will not necessarily increase the size of the system by a factor of two to one. Transistor eqpipment can be made very compact, it does not require much power to operate, and the lack of heat generation means that such equipment may be mounted in the cabinetry without much thought to ventilation. Third, the transistor is virtually free from hum and microphonics, and this means that the hum level in the preamplifier—a problem because of the extremely low output voltages of some of the popular stereo cartridges—will be almost unmeasurable. In addition, we shall see that "transistor noise," which has manifested itself as a hiss or "frying" sound in early units, can be reduced to a vanishingly low level if proper design techniques are applied.

The discussion to follow will be somewhat different from the ordinary construction article. Because of the novelty of transistor circuits compared to their vacuum-tube counterparts, some detailed information will be included concerning the operating points of the transistor stages, and the equations used to calculate stability *vs.* temperature change, feedback resistors, and gain will be presented. While such calculations are necessarily tedious, it is hoped that by their presentation the reader will gain a better understanding of the approximations and computations associated with the design of transistorized equipment.

The circuitry of the control unit, *Fig.* 3, may be divided into two sections: the low-level preamplifier, where noise figure and exact frequency compensation are the important factors, and the high-level control stage, where signal-handling capabilities are important. The low-level preamplifier circuit consists of two low-noise transistors Q_1 and Q_2, connected in the grounded-emitter configuration. The first transistor, Q_1, is operating at a low collector voltage and very low collector current to minimize the noise figure of the transistor, and the load resistor for this stage is a deposited-carbon type for the same reasons. The bias configuration is the "H" type, with the temperature-stability factor (defined as $S = \Delta I_c / \Delta I_{co}$) chosen to have a numerical value of less than 4. This value for S is chosen mostly from experience, which has shown that this is a reasonable criterion for reliable class A operation over the rated temperature range of a germanium transistor. The "H" bias configuration depends upon a large resistance in the emitter circuit to achieve stability, and then this resistor is bypassed with a large capacitor to maximize a.c. gain. In proceding with the design of this stage, the value of the collector-to-emitter voltage, V_{ce}, and the collector current, I_c, are first chosen from a graph of the noise characteristics of a typical transistor versus V_{ce}, I_c, and the generator impedance R_g. (*Fig.* 2) the values thus selected are:

$$V_{ce} = -8 \text{ vdc.}$$
$$I_c = -0.35 \text{ ma.}$$

Before resistor values can be calculated from the operating point, there are several restrictions on the parameters of the stage which must be taken into account. First, the voltage gain, G_v, and the input impedance Z_{in} must conform to external considerations. A survey of the popular stereo cartridges shows that an input impedance of 50,000 ohms is sufficient for all the cartridges requiring high-impedance, low-level inputs, and a lower Z_{in} may always be achieved by loading the cartridge with a resistor. It must be remembered that high values of input impedance are difficult to achieve in transistor stages which also have voltage gain, and for this reason we shall not attempt to make Z_{in} any higher than necessary. The compensated closed-loop gain of the low-level preamplifier should be about 40 db ($G_v = 100$) so that a 10-mv input will produce a 1-volt output. Since about 20 db for bass boost is required for the RIAA curve, the preamplifier will require an open-loop gain of 40 plus 20, or 60 db ($G_v = 1000$). This means that each stage within the loop should have a gain of $60/2 = 30$ db ($G_v = 31.5$).

We are now in a position to calculate some resistor values for the first stage. With the "H" bias configuration and the large amount of feedback present, the input impedance of the first stage will be primarily determined by the parallel combination of R_2 and R_3; we shall call this equivalent resistance R_b', and we can then say that approximately:

$$Z_{in} \cong R_b' \qquad (1)$$

The next restriction on the values of the resistances in this stage is the S factor. For a grounded-emitter stage using a transistor with a high grounded-emitter current gain Beta (β) and small internal base and emitter resistances compared to the circuit values, S can be given by:

$$S \cong 1 + \frac{R_b'}{R_e'} \qquad (2)$$

where R_e' is the total external emitter circuit resistance, $R_4 + R_5$. But we have specified R_b' at 50,000 ohms, and S can have a maximum value of 4, so we can easily calculate the minimum value of R_e' necessary to achieve this stability factor, and this turns out to be 16,000 ohms. Allowing an ample margin of safety, we will let R_4 equal 22,000 ohms, and the voltage drop across R_4 is $I_eR_e = 7.7$ volts. Making one additional assumption, that there is a small and nearly constant voltage difference between the base and emitter terminals of the transistor, amounting to about 0.3 volts, we know that the voltage across R_2 is $7.7 + 0.3 = 8$ vdc. We can similarly deduce that the voltage across R_3 is the supply voltage to the stage less 8 volts. In our case the power supply voltage is 30 volts, selected because experience has shown that this is a convenient voltage for audio work in transistors. To make certain that there is no cross-coupling between stages, there are two decoupling resistors in the supply line, R_7 and R_{19}, and the supply voltage to the Q_2 and Q_3 stages is actually about 28 volts, and that to the Q_1 stage about 25 volts. This shows that there are $25 - 8 = 17$ volts across R_3, and now that we know that the ratio of R_2 to R_3 is 8 : 17 (assuming that he current in the base of the transistor is negligibly small) and that the parallel equivalent of R_2 and R_3 is 50,000 ohms, we can easily calculate their values, which are 73,000 ohms and 156,000 ohms, respectively. Increasing these to the nearest 10 per cent EIA values gives: $R_2 = 82,000$ ohms, and $R_3 = 180,000$ ohms, and recalculation of R_b' now yields 56,000 ohms, still sufficiently close to the target figure of 50,000 ohms.

We have only two resistors left to calculate, the load resistor R_L (R_6), and the small unbypassed emitter resistor R_5, which controls the voltage gain. R_6 is actually already determined since we know the collector current (which equals the emitter current, approximately) and the voltage at the collector (which is the supply voltage less the V_{ce} and I_eR_e drops), and the calculated value is 26,600 ohms, the closest value in a deposited-carbon resistor being 26,100 ohms. Once the load resistor is known, we can calculate the value of the feedback resistor R_5 from the equation for gain with feedback:

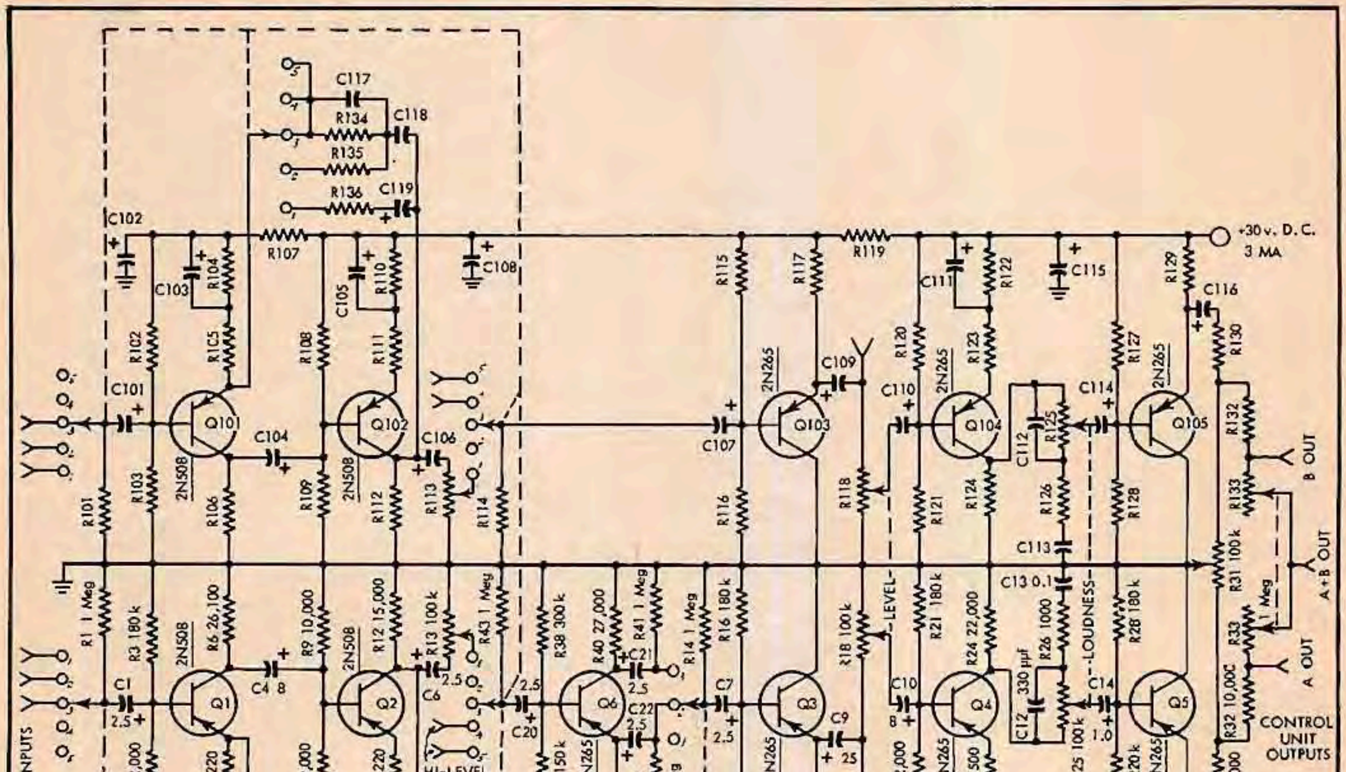

Fig. 3. Complete schematic of the transistorized control unit.

$$G_v \cong \frac{(\beta R_L')}{(h_{ie} + \beta R_5)} \qquad (3)$$

where R_L' is the equivalent load imposed by the load resistor and any subsequent circuitry. H_{ie} is the grounded-emitter input impedance of the transistor, as specified by the manufacturer. It is interesting to note that this is the first time the parameters of the transistor have entered the equations explicitly; all the previous calculations have simply assumed some range of values.

Choice of Transistors

At this point, then, it will be necessary to select the specific transistor to be used in the circuit. A survey of the inexpensive, commercially available germanium "p-n-p" transistors with low noise-generation characteristics results in the selection of any of two or three types, all of which are pretty much interchangeable. The GE 2N508, the Philco 2N535B, (a newer version of the 2N207B), and the Raytheon 2N422 will all perform well in the preamplifier, and from this point forward we shall assume the characteristics of the 2N508. This transistor has a β of 100 and an h_{ie} of 3000 ohms, and a typical noise figure of less than 6 db (0 db = 1 μv.). Using this data, and assuming a loading by the following stage of 15,000 ohms, we can calculate from Eq. (3) that the G_v will be approximately 33 for an R_5 of 220 ohms. For the remaining control unit circuits, where noise is not so critical but larger signals entail larger collector voltages, we select a transistor with the same general characteristics but an increased maximum V_{ce}; such a transistor is the GE 2N265 or the Philco 2N534.

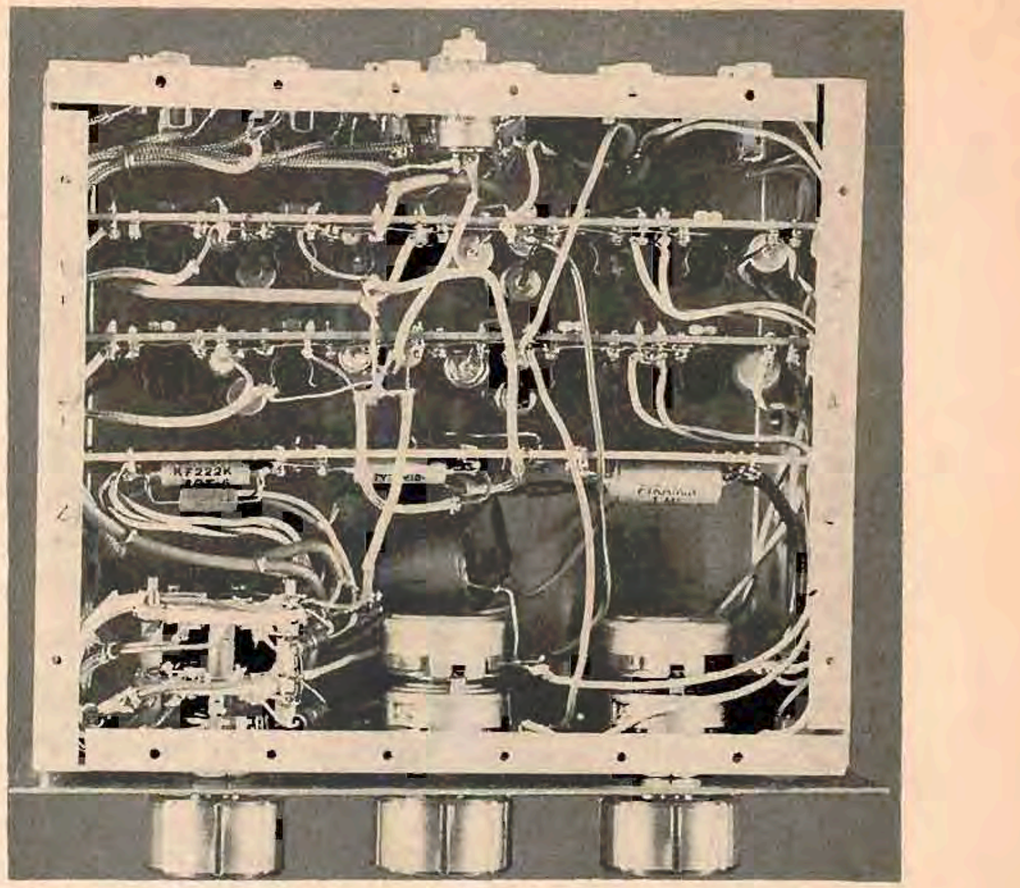

Fig. 4. Top view of the completed preamplifier unit. The upper two resistor boards are identical; the lower contains the equalization networks, loudness-control components, and the phase-reversal stage.

The preceding discussion has covered all the computations which are peculiar to the design and stabilization of a transistor stage with a specified voltage gain and input impedance. The values of coupling and bypass capacitors are calculated in the same manner as those for vacuum tube stages, that is, by making the time constant of the equivalent circuit small compared to the period of the lowest frequency we wish to pass or bypass. The technique just illustrated may now be repeated for the Q_2 stage, using the conditions and approximations just discussed, and assuming the loading of subsequent circuitry to be 50,000 ohms. Once this stage is completed, we are ready to calculate the values of the feedback and compensation resistors and capacitors, R_{34-36} and C_{17-19}.

When the selector switch S_1 is in position 1, R_{36} and C_{19} are the elements in the feedback loop. This corresponds to the "microphone" position, and we are desirous of a flat frequency response and a closed-loop gain of 4 db (G_v = 100). The value of R_{fb}, (R_{36}), in the condition where the presence of this resistor does not upset the open loop gain excessively, is:

$$R_{fb} \cong -\frac{(R_5)\ (G_v)}{(K-1)} \qquad (4)$$

where K is the ratio of the open loop gain G_v to the closed loop gain G_v'. This equation gives a value of 24,000 for R_{36}, but since this would load the output of the second stage and change the open loop gain slightly, (we assumed a loading of 50,000 ohms) the corrected value of R_{36} becomes 27,000 ohms. In this case, C_{19} is merely a coupling capacitor with

Parts List

*R_1, R_{14}, R_{41}, R_{43}, R_{44}	1 megohm, ½ watt
R_2	82,000 ohms, ½ watt
R_3, R_{16}, R_{21}, R_{28}	180k ohms, ½ watt
R_4, R_{30}	22,000 ohms, ½ watt
R_5, R_{11}	220 ohms, ½ watt, 5%
R_6	26,100 ohms, ½ watt, 1% deposited carbon
R_7	15,000 ohms, ½ watt
R_8	39,000 ohms, ½ watt
R_9	120k ohms, ½ watt
R_{10}, R_{20}, R_{32}	10,000 ohms, ½ watt
R_{12}	15,000 ohms, ½ watt, 5%
R_{13}, R_{21}	100k ohms, ½ watt
R_{15}, R_{27}	220k ohms, ½ watt
R_{17}, R_{29}	27,000 ohms, ½ watt
R_{18}, R_{25}	100k-ohm dual potentiometers, linear, 10%
R_{19}	2200 ohms, ½ watt
R_{22}	4700 ohms, ½ watt
R_{23}	1500 ohms, ½ watt, 5%
R_{24}, R_{39}	22,000 ohms, ½ watt, 5%
R_{26}	1000 ohms, ½ watt
R_{25}	1-megohm potentiometers, linear, 10%
R_{34}, R_{36}, R_{40}	27,000 ohms, ½ watt, 5%
R_{35}	5600 ohms, ½ watt, 5%
R_{37}	150k ohms, ½ watt, 5%
R_{38}	300k ohms, ½ watt, 5%
*C_1, C_6, C_7, C_{20}, C_{21}, C_{22}	2.5 μf, 25 v, electrolytic
C_2, C_3, C_9	25 μf, 25 v, electrolytic
C_4	100 μf, 6 v, electrolytic
C_5, C_{10}, C_{16}	8 μf, 25 v, electrolytic
C_8, C_{11}	100 μf, 10 v, electrolytic
C_{12}	330 μμf, 300 v, mica
C_{13}	0.1 μf, 100 v, paper
C_{14}	1 μf, 6 v, electrolytic
C_{15}	100 μf, 50 v, electrolytic
C_{17}	.0027 μf, 100 v, paper
C_{18}	.012 μf, 100 v, paper
C_{19}	5 μf, 12 v, electrolytic
S_1	6-pole, 5-position switch, shorting
S_2	2-pole, 3-position switch, shorting
*Q_1, Q_2	"p-n-p" transistor, (GE 2N508)
Q_3, Q_4, Q_5, Q_6	"p-n-p" transister, (GE 2N265)

* Values given for subscripts from R_1 to R_{30} (except R_{31}) apply also to resistors with subscripts R_{101} to R_{128}; values given for capacitors C_1 to C_{19} apply also to capacitors with subscripts C_{101} to C_{119}; transistor types listed for Q_1 to Q_6 apply also to transistors Q_{101} to Q_{106}.

All resistors 10% tolerance unless otherwise specified.

negligible reactance at the frequences from 20 to 20,000 cps.

The RIAA phonograph curve is realized by a low-frequency boost with a characteristic time constant of 300 microseconds, and a high-frequency rolloff time constant of 75 microseconds. Since the midfrequency gain is still supposed to be 40 db, R_{34} is 27,000 ohms. The bass-boost capacitor C_{18} will have a value of 0.012 μf, the boost time constant divided by R_{34}. Similarly, the rolloff capacitor C_{17} will be the rolloff time constant divided by R_{34}, and this yields a value of 0.0027 μf.

The NAB tape curve is realized by a single low-frequency boost with a time constant of 67 microseconds, where the high-frequency gain is considered the closed loop gain G_v'; thus the computation in this caste will be somewhat different. The output from a tape reproducing head is generally lower than that from a phono cartridge, so that it would be desirable to have the highest over-all gain possible with the equalization required. The total equalization between high and low frequency is 32 db, and since the open loop gain is 60 db, this means the maximum high-frequency gain (in this case the closed loop gain), will be 28 db ($G_v' = 25$). For this value and the known value of C_{18}, we calculate the corrected value of R_{35}, which is 5600 ohms. If we now calculate the gain at 500 cps (which is the crossover frequency for the NARTB curve), this midfrequency gain is 150, or about 43 db. This means that a tape head output of only 3.5 mv will produce a 0.5 volt output, while a 5-mv phono or microphone input is required to produce the same output.

At this point, it is timely to interject a note concerning the behavior of the large values of capacitance necessary in R-C coupled transistor stages. In order to maintain flat frequency response down to the lowest audio frequencies, the values of coupling capacitors tend to become very much larger than those in vacuum tube circuits, with values as large as 25 μf not uncommon. These are usually low-voltage midget electrolytics, and as such as subject to leakage currents considerably larger than those in the typical paper tubular capacitor. Therefore, it is important that d.c. return paths be provided for these leakages at inputs and outputs, and that the variable controls be arranged in such a way that the operation of the control does not change or reverse the d.c. voltage on the electrolytic, lest the resulting nonlinear discharge generate an audio signal not unlike a Bronx cheer.

The over-all characteristics of the low-level preamplifier stages may now be tabulated:

(A) *Flat Equalization:*

Frequency response flat ± 0.5 db from 20 to 20,000 cps.

Sensitivity 5 mv for 0.5 volts output.

Intermodulation Distortion 0.25 per cent at 1 volt output.

Harmonic distortion not measurable at 1 volt output.

Noise level more than 65 db below 1 volt output.

(This noise is characteristically below 30 cps.)

(B) *RIAA Equalization:*

RIAA equalization ± 0.5 db from 20 to 20,000 cps.

All other specifications equalled where applicable.

(C) *NAB Equalization:*

NAB equalization ± 0.5 db from 20 to 20,000 cps.

Sensitivity 3.5 mv for 0.5 volts output.

All other specifications equalled where applicable.

High-Level Section

The input to the high level section is made through the selector switch S_1, resulting in the choice of either the preamplifier output or a high-level input. The signal is then fed to the base of the emitter-follower Q_3. With the emitter-follower configuration we are able to realize a high input impedance and low output impedance at the cost of unity voltage gain, much in the same manner as the cathode follower in vacuum-tube circuitry. The output of Q_3 is labelled the "monitor output," and may be used to drive a tape recorder for dubbing, headphones, or almost any impedance greater than a few hundred ohms; this output is equalized to have a flat frequency response and constant amplitude, independent of the settings of the level and loudness controls R_{18} and R_{25}.

There is little to describe in the design of the emitter-follower stage because it is an extremely simple configuration, and there is no gain calculation to make. Suffice, then, that the operating point of the transistor is chosen so that:

$$V_{ce} = -12.5 \text{ vdc.}$$
$$Ic = -0.5\text{ma.}$$

and the S factor is again less than 4.

The Q_4 stage, like the Q_1 and Q_2 stages, is connected as a grounded-emitter amplifier with voltage gain determined by emitter degeneration. The smallest input anticipated at the high level jacks is 0.1 volt, and a 1-volt output of the control unit is sufficient to drive most power amplifiers to full output, so that an over-all voltage gain of 10 will satisfy the needs of the high-level portion. If each of the emitter followers has an over-all gain of 0.95, the gain of the Q_4 stage must be 12. Making the assumption of 50,000-ohm loading by the subsequent stages, and using exactly the technique described for the Q_1 stage and an operating point of:

$$V_{ce} = 11.5 \text{ vdc.}$$
$$I_c = 0.7 \text{ ma.}$$

the result is a gain of 12.5 for a load resistor of 22,000 ohms and an unbypassed emitter resistance of 1500 ohms.

The output of the fourth stage drives the loudness control R_{25} which, despite its simplicity, will approximate the Fletcher-Munsen loudness contours reasonaly closely from the 0 db to the −35 db curves. The output of this control is connected to the fifth transistor, which is an emitter-follower circuit identical to the third stage. The output of Q_5 drives the balance control, R_{31}, and the separation control, R_{33}. The balance circuit is capable of an infinite range of attenuation, but the important range near balance is spread out so that it occupies a large portion of the range of rotation of the control. With this arrangement the balance can be adjusted very finely for nearly equal signals, or one channel can be eliminated from the output completely, by a 150-deg. rotation of the control. The separation control is likewise nonlinear in operation, making only about 6 db of variation in the separation through the first 100 deg. of rotation, and then decreasing the separation rapidly to zero at full rotation.

The output impedance of the control unit is about 20,000 ohms as a conse-

(*Continued on page 82*)

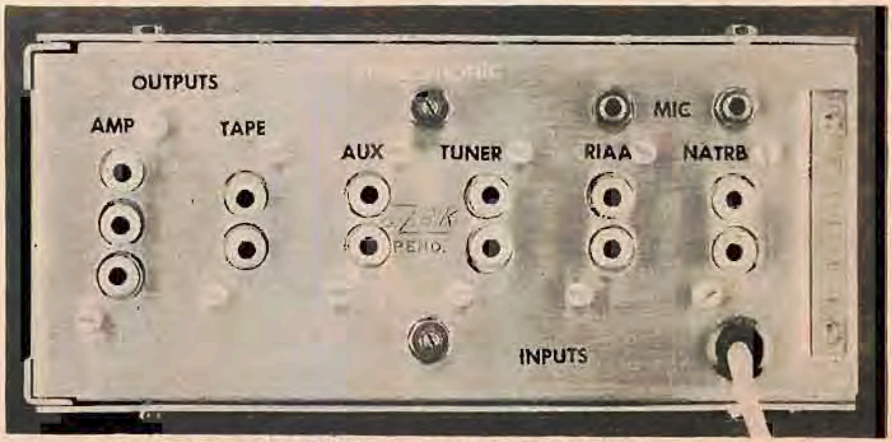

Fig. 5. Rear view of the completed transistorized preamplifier.