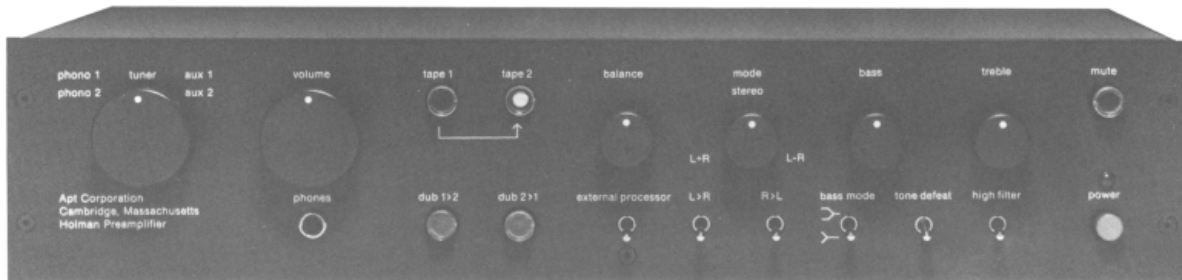


**The Apt/Holman Preamplifier
Service Manual**

Apt Corporation
Box 512, 147 Sidney Street
Cambridge, Massachusetts 02139 USA
Telephone 617 492-5145



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1 Introduction

The Apt/Holman Preamplifier has been designed to provide a control center for home music systems which allows maximally flexible use of other equipment. It features filtering to avoid interaction from other signals outside the audio band (including RFI), a phono cartridge amplifier which allows optimum matching to a wide range of cartridges, newly developed control characteristics to allow precise adjustment of stereo depth, tonal balance and accurate loudness compensation, buffered outputs to all external components, as well as noise, distortion, and frequency response characteristics which represent the state of the art in audio equipment.

Despite the amount of complex circuitry required to perform these tasks optimally, the circuitry and switching have been almost entirely confined to three fiberglass circuit boards, which require no structural support other than the steel case. Hence, all test points are accessible as soon as the case is removed, and virtually all components may be reached by freeing one circuit board. A power transformer operated at low flux density and careful attention to magnetic design have allowed inclusion of the power supply on the main board, avoiding the cumbersome external power supply required by some other designs.

2 Specifications

- 2.1 Test Specifications.** All data are valid for any combination of inputs and outputs, unless otherwise noted. Data are valid for 20 Hz to 20 kHz unless otherwise noted. Voltages in dBV are referred to 1.0 Vrms. Line inputs under test are driven by a generator with source impedance of 1.0 kohm. Outputs under test are loaded by 10 kohms in parallel with 1000 pF. Measurements are made with a Hewlett-Packard 239A oscillator, a Radford ANM2 rms ac voltmeter, and a Hewlett-Packard 3580A Spectrum Analyzer, as specified. Apt reserves the right to make changes as technical progress warrants, but we will not engage in capricious updating. Specifications are subject to change without notice.
- 2.2 Mechanical Data.**
- 2.2.1** Outside dimensions of the case are 7.92 × 38.20 × 20.80 cm (3.12 × 15.04 × 9.32 in). The overall dimensions are 8.38 × 38.20 × 23.67 cm (3.30 × 15.04 × 9.32 in). The knobs protrude by 1.42 cm (0.56 in); connectors and cables require up to an additional 4.44 cm (1.75 in) depth from the rear surface. The unit's weight is 4.8 kg (10.5 lb); the packed weight is 5.4 kg (12 lb).

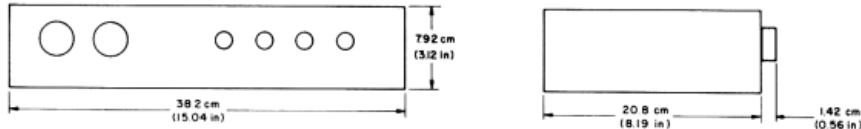


Fig. 2-1. Case Dimensions.

- 2.2.2** The front and rear panel finish is baked instrumentation gray polyurethane enamel with permanent baked-epoxy white markings; the cover finish is neutral gray wrinkle baked enamel.
- 2.2.3** All operational modes are indicated visually by means of pilot light, rotary switch knob, pushbuttons with indicator windows, tallying pushbuttons, or toggle switch levers.
- 2.3 Power Requirements.**
- 2.3.1** Input line voltage is 120 Vac nominal, 240 Vac with internal wiring change, 50-60Hz. The range of normal operation for the 120 Vac connection is 95-135Vac (190-270 Vac connected for 240 Vac operation). Low line ("brownout") conditions below 95 Vac activate the muting circuit to prevent spurious signal from reaching the amplifier or loudspeakers. Power consumption is less than 15 watts.
- 2.3.2** The switching capacity for the convenience outlets marked "switched" is 1000 watts (US model).
- 2.4 Levels, Gains, Impedances.**
- 2.4.1** All inputs and outputs are unconditionally stable under any conditions of source or load impedance.
- 2.4.2** All inputs and outputs have identical polarity; i. e., a positive-going signal at any input will produce a positive-going signal at any output.
- 2.4.3** The output level for rated specifications is 2.0 Vrms (+6.0 dBV).
- 2.4.4** The maximum undistorted output is at least 7.0 Vrms (+17 dBV).
- 2.4.5** The Phono 1 input impedance is switchable between 47 kohm or 100 kohm resistive in parallel with 50 pF, 100 pF, 200 pF, or 400 pF; the parallel RC equivalent circuit is a complete and accurate representation of the input impedance. The Phono 2 input impedance (without the optional pre-amplifier) is 47 kohm in parallel with 50 pF. Changes to the input impedance to accommodate unusual termination requirements may be made by a qualified service technician or by means of a factory-supplied impedance module.
- 2.4.6** Phono 1 or 2 input level for rated output is 5 mVrms (1.25 mVrms for 0.5 Vrms output).
- 2.4.7** Phono gain to tape monitor and external processor outputs is 36.5 dB at 1 kHz. The right channel gain of each phono input is adjustable by ±2 dB to compensate for imbalance in the phono cartridge. As shipped, the gain is precisely matched to the left channel.
- 2.4.8** The input impedance for all high level inputs is 50 kohms resistive in parallel with 330 pF. Unselected inputs are terminated by 2.2 kohm.

- 2.4.9** The source impedance at the Tape and External Processor outputs is less than 100 ohms; the minimum load impedance for those outputs is 10 kohms in parallel with 1000 pF.
- 2.4.10** The maximum gain for any high level input to the main output is 18 dB; the line input sensitivity for rated output is 320 mVrms (80 mVrms for 0.5 Vrms output).
- 2.4.11** The source impedance at the main output is 330 ohms; the minimum load impedance for the main output is 5 kohms in parallel with 3000 pF.
- 2.4.12** A load impedance of 2 kohms on any output will limit the maximum available voltage at that output to 3.0 Vrms.
- 2.4.13** The input overload for the phono inputs is greater than 100 mVrms at 1 kHz and follows the inverse RIAA function in the audio band. Infrasonic warp signals will not overload the input or intermodulate with audio band signals for any known playback system.
- 2.4.14** The input overload for the line-level inputs is greater than 10 Vrms (+20 dBV, +22 dBm).
- 2.4.15** The tracking accuracy of the Volume control is within 1 dB over the full range of attenuation.
- 2.5. Frequency and Group Delay Response.**
- 2.5.1** The frequency response from the line-level inputs to the main output with the infrasonic and ultrasonic filters engaged is ± 0.5 dB, 20 Hz to 20 kHz.
- 2.5.2** The infrasonic filter response is within 0.5 dB at 20 Hz and above, -3 dB at 15 Hz, and more than 30 dB down at 5 Hz, with 18 dB/octave slope. Phono and line level inputs are all passed through this filter to prevent infrasonic noise from adding distortion in tape recorders, power amplifiers, or other external components. The group delay of the filter is 4 milliseconds at 50 Hz, while the limit of perception has not been found to be lower than 100 milliseconds at that frequency. With the filter defeated the -3 dB frequency is 8 Hz.
- 2.5.3** The ultrasonic filter response is within 0.5 dB of flat at 20 kHz and below, -3dB at 40 kHz, and -18 dB at 100 kHz, with 12 dB/octave slope. All inputs are processed through this filter to prevent ultrasonic noise or distortion products from being detected (i. e., causing transient intermodulation distortion) in external components with inadequate slew rates or other susceptibilities to ultrasonic components. The group delay of the ultrasonic filter is 7 microseconds at 20 kHz; the audible limit for a perceptible change of timbre at any frequency is many orders of magnitude larger.
- The differential group delay between channels is less than 0.5 microseconds; the audible limit for a localization difference is 10 microseconds. With the filter defeated (through Tone Defeat) the -3 dB frequency is 150 kHz.
- 2.5.4** The RIAA phono equalization error is within ± 0.2 dB from 30 Hz to 15 kHz.
- 2.5.5** The response error due to phono impedance interaction error is less than 0.2 dB.
- 2.6 Distortion.**
- 2.6.1** Leinonen and Ojala (1977) have shown that no one distortion test fully characterizes all known distortion mechanisms. Therefore, a combination of tests is necessary. Their work shows that the combination of T.H.D., S.M.P.T.E. I.M., Difference Tone I.M., and T.I.M. tests will reveal all known distortion components. We specify the performance of the Holman Preamplifier on the basis of all of these tests. Also given is a specification for slewing induced distortion, caused by the same mechanism as T.I.M., but specified in a different manner.
- 2.6.2** All distortion measurements (with the exception of those which require greater bandwidth) are made with a Hewlett-Packard Model 3580A Spectrum Analyzer.
- 2.6.3** A total-harmonic-distortion test consists of applying a pure sine wave to the input and examining the output for the presence of distortion products at $2f$, $3f$, The total harmonic distortion is the rms sum of all such components. Total harmonic distortion (T.H.D.) at rated output level and lower is less than 0.01% and consists of pure second harmonic or a mixture of second and third harmonics at all levels and frequencies. For example, at 1 kHz and rated output the distortion typically consists of pure second harmonic down 85 dB from the fundamental (0.0045%).
- 2.6.4** The S.M.P.T.E.-intermodulation-distortion test consists of applying to the input 60 Hz and 7.0 kHz sine waves mixed with an amplitude ratio of 4 : 1 and examining the output for intermodulation products spaced at 60 Hz intervals about the 7.0 kHz tone. The distortion percentage is the rms sum of all such sidebands compared to the amplitude of the 7.0 kHz signal. The S.M.P.T.E. intermodulation distortion at rated output level or lower is less than 0.01% and consists of first order sidebands only.
- 2.6.5** A difference-tone-intermodulation test consists of applying to the input two high-frequency sine wave signals mixed with an amplitude ratio of 1 : 1 and examining the output for the difference product $f_1 - f_2$. The distortion percentage is the ratio of the distortion products compared to the amplitude of the sum of the high-frequency sine waves. With 19 kHz and 20 kHz mixed 1 : 1 at rated output level or lower, the difference-tone-intermodulation is less than 0.005%.
- 2.6.6** A transient-intermodulation-distortion (T.I.M.) test consists of applying to the input a symmetrical square wave signal at 3.18 kHz and a 15 kHz sine wave, mixed with an amplitude ratio of 4 : 1 peak-to-peak, low-pass filtered at 6 dB/octave at 100 kHz (T.I.M. 100), and examining the output for

the presence of any intermodulation products in the audio band. The T.I.M. 100 distortion is less than the measurement residual of -84 dB (0.006%) at rated output level or lower.

- 2.6.7** All stages meet the Jung-Stephens-Todd (1977) criteria for negligible measurable and audible slewing-induced distortion. The criteria are that all amplifiers have a linear-transconductance input stage, symmetrical slewing, and adequate speed.

2.7 Noise and Crosstalk.

- 2.7.1** Unless otherwise indicated, all noise measurements are made with a true rms reading ac voltmeter with psychometric weighting for the annoyance value of noise (ASA A-weighting).

- 2.7.2** For the phono inputs the equivalent input noise with a standard cartridge (defined by IHF Standard 3.12 in IHF-A-202 as 500 mH $\pm 10\%$ in series with 1 kohm $\pm 10\%$, this network in parallel with 125 pF $\pm 10\%$) connected and A-weighted (ANSI Standard S1.4) measures less than 74 dB below the IHF reference level of 5 mVrms at 1 kHz. (The measured noise is typically 76 dB below 5 mV.) The noise of a standard cartridge plus its standard load is 78 dB below 5 mV, which yields a maximum noise figure of 4 dB (typical 2 dB). (Noise figure is a comparison between the actual measured noise and the theoretical minimum noise due to the resistance of the cartridge and its load. In other words, the phono preamp stage adds only about 2 dB of noise to the lowest possible noise attainable. Noise measurements made with a shorted input are misleading, since they do not account for the noise generated by the phono cartridge nor by the amplifier's own input impedance under the intended operating conditions, i. e., with a cartridge connected.)

- 2.7.3** For the line-level inputs the equivalent input noise is less than -106 dBV, A-weighted.

- 2.7.4** The residual output noise with the Volume control down is less than -95 dBV A-weighted.

- 2.7.5** The output noise with the Volume control set for -20 dB attenuation (a typical listening level) is less than -95 dBV A-weighted for the line-level inputs, and -90 dBV A-weighted for the phono inputs.

- 2.7.6** The 60 Hz hum component and its harmonics are typically equal to or below the wideband noise spectrum.

- 2.7.7** Other weighting schemes may be applied to the data above. For example, CCIR/ARM weighting will typically increase the noise measurement by 2 dB. DIN Audio Band weighting (flat, bandlimited to 20 Hz to 20 kHz) will typically increase the noise measurement by 2 dB. DIN Audio Band weighting (flat, bandlimited to 20 to 20 kHz) will typically increase the noise measurement by 3 dB.

- 2.7.8** Crosstalk between program sources, including tape monitor, is less than 90 dB to inputs terminated with 1 kohm or less from 20 Hz to 20 kHz.

- 2.7.9** Crosstalk between channels of a single program source is less than 65 dB at 1 kHz and 45 dB at 20 kHz.

2.8 Dynamic Range.

- 2.8.1** From the data above, it follows that for line level inputs the dynamic range is 123 dB from the input overload point to the weighted noise level. This means that the full dynamic range of any medium now available or contemplated is greatly exceeded, including the forthcoming digital media. With the greatest dynamic range system available to professional recordists, level mismatches up to 30 dB can be tolerated without degrading the signal-to-noise ratio.

It likewise follows that the dynamic range for the phono inputs is typically 94 dB from the phono overload point to the A-weighted noise, which greatly exceeds the limits imposed on the phono playback system by the thermal noise of the cartridge at the low end and the recording geometries at the high end of the dynamic range.

3 Test Instrumentation

3.1 General Advice. The specified parameters for frequency response, distortion, and noise of the Apt/Holman Preamplifier are beyond the range or accuracy of much commonly used test equipment; hence, no repairs or adjustments which may in any way affect these parameters should be undertaken unless adequate instrumentation is at hand.

3.2 Recommended Equipment.

3.2.1 Oscillator. The test oscillator must be flat ($< \pm 0.1$ dB) from 20 Hz to 20 kHz and must show no distortion products or noise at greater than -85 dB relative to 20 mV at 1 kHz nor relative to 150 mV at 16 kHz.

*Recommended: Hewlett-Packard 239A
Sound Technology 1701A*

3.2.2 Oscilloscope. 15 MHz or greater bandwidth, preferably dual trace, with dc input.

*Recommended: Tektronics 922
Hewlett-Packard 1220A*

3.2.3 Volt-Ohmmeter. 20 kohm/volt or better.

*Recommended: Simpson 260
Triplet 302*

3.2.4 Ac Voltmeter. This meter, used for establishing frequency response and noise levels, must be flat within $< \pm 0.1$ dB 20 Hz to 20 kHz, should compute a true rms average, should allow A-weighted noise measurements, and should read accurately to -105 dBV ($6 \mu\text{V}$). Since no meter currently available completely meets all of these requirements, each of the meters listed will require some calibration or modification, as indicated.

*Recommended: Radford ANM2. Calibrate frequency response, Section 3.3.1.
Sound Technology 1701A or 1710 Distortion Measurement System with 1200A Stereo Test Panel. Correct for ac averaging, Section 3.3.2.
Sennheiser UPM 550; RV 55. Add input preamp, Section 13.A; see Sections 3.3.1, 3.3.2, 3.3.3.
Hewlett-Packard 400 F; 400 GL. Add input preamp, Section 13.A; see Sections 3.3.1, 3.3.2, 3.3.3.
Heathkit IM-5238. Add input preamp, Section 13.A; see Sections 3.3.1, 3.3.2, 3.3.3.
Leader LMV-185; LMV 181A. Add input preamp, Section 13.A; see Sections 3.3.1, 3.3.2, 3.3.3.
Digital Meter Research DMR-100. Add input preamp, Section 13.A.*

3.2.5 Spectrum Analyzer/Distortion Meter. Authentic measurements of harmonic distortion, independent of noise and spurious signals, can only be carried out properly on a spectrum analyzer (See IHF Standard A-202, Section 2.9.6). Nevertheless, in most cases adequate measurements can be made with a high quality distortion meter capable of measuring to -85dB (.006 %) in the range 100 Hz to 50 kHz and which is equipped with low and high pass filters (e. g., 400 Hz and 80 kHz).

Recommended: Hewlett-Packard 3580A Signal Analyzer. A most desirable and versatile instrument.

Sound Technology 1700B; 1701A; 1710A. See Section 3.3.4.

Hewlett-Packard 339A. See Section 3.3.4.

3.2.6 Variable autotransformer, 120 Vac (or 240 Vac), at least 0.1 A, equipped with ac milliammeter, 0–250 mA. Since a 10 percent deviation in drain current can be significant, the milliammeter is a most useful diagnostic aid. A rugged moving-vane meter equipped with a fuse for 1 amp is recommended (for servicing power amplifiers or receivers, change fuse to 5 amperes and shunt meter for larger current range).

3.3 Calibration and Use.

3.3.1 Frequency calibration. Using the 1.0 Vac range of your voltmeter, set the output of your signal generator to read 0.0 dB at 1.0 kHz. Record the readings at 10, 20, 25, 30, and 300 Hz, as well as at 3, 10, 15, and 20 kHz. Save this calibration table for correcting measurements of frequency response with this generator—meter set-up.

If, for any reason, you desire an absolute calibration of your signal generator, a good, wide-band, dc-coupled oscilloscope such as the Tektronix T922 may be used as a standard. Set the trigger mode to X-Y, position the trace in the middle of the screen, and adjust the amplitude to obtain full-screen vertical deflection. An output variation of 0.1 dB corresponds to a variation of 1 percent in peak-to-peak amplitude and will be clearly visible on the oscilloscope.

Averaging vs. rms meters. Averaging ac voltmeters are calibrated to give the reading on a sine wave input that a true rms meter would give, and for measuring sine waves they are just as accurate. However, for other waveforms this calibration will no longer be valid (since the arithmetic average computed by the averaging meter will not equal the root mean square computed by the rms meter). With A-weighted white noise, the averaging meter will read about -1.1 dB relative to an rms meter, even though sine waves will give the same reading. (Because the sensation of loudness itself approximates a power, or rms response in most cases, the rms measurement is the preferable basis for comparison.)

If you are using an averaging ac voltmeter, add 1.1 dB to the A-weighted noise measurement.

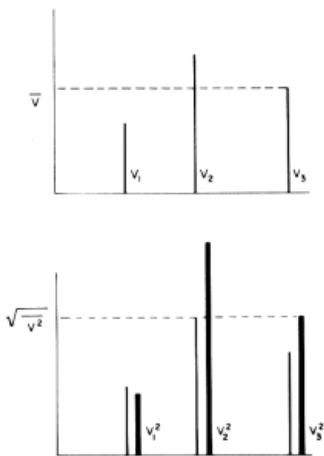
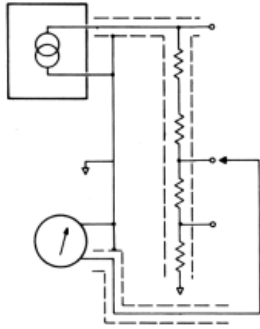


Fig 3.1. Arithmetic vs rms Mean.

3.3.3 Noise Calibration. The “Volume down” noise test requires reliable measurement of levels down to -105 dBV ($6\ \mu\text{V}$). In addition, psychoacoustically meaningful measurement requires a weighting function such as the ASA “A” curve. If your ac voltmeter is not capable of reading these voltages accurately, or if it does not allow for A-weighting of the input, refer to the construction project in the Appendix 13.A.

Even very good voltmeters sometimes show some non-linearity at very low levels (often because of internal noise adding to the input signal). Check your voltmeter’s lowest range with a low level signal fed into a well-shielded string of four 1 kohm resistors.



Relative to the unattenuated signal the center tap of the string should give -6 dB; the $\frac{1}{4}$ division, -12 dB.

When the A-weighting filter is inserted, no change in level should be observed at 1 kHz relative to the unweighted measurement.

-10-

Fig. 3-2. Linearity Test.

4 Circuit Description

4.1

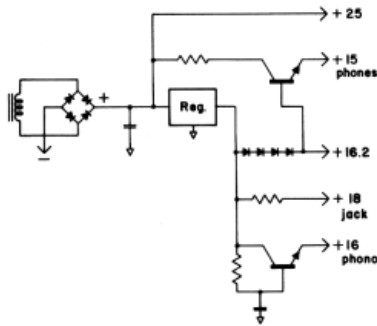


Fig. 4-1. Power Supply Levels.

4.2

Power Supply. The power supply is designed to eliminate induced hum and supply ripple which might later appear in the preamplifier output. A specially designed power transformer operating at low flux density rests in a steel shielding bracket, and its orientation as well as the shape and materials used in the construction of the preamplifier case have been chosen to minimize the hum induced in low-level signal paths. Three sets of dual supply voltages are required, ± 25 V, ± 18 V, and ± 16 V. The first of these is used directly only to supply the mute relay circuit and the pilot light, D13. As shown in Fig. 4-1, IC regulators reduce this unregulated supply to +18 V, typically with 60 dB of ripple and noise rejection. Regulated +18 V is supplied to the +18 V jack (through 1 kohm) to operate external devices, such as microphone preamplifiers. Diode strings reduce the regulated supply to ± 16 V for the IC operational amplifiers. A second ± 16 V supply with additional filtering is obtained for the phono input stage through transistors Q8 and Q9; this comparatively high voltage allows the stage to meet its overload specification. Because of its high current requirements the headphone amplifier does not draw current directly from the regulated supply. Instead, a pair of transistors, Q20 and Q21, whose bases draw negligible currents at the regulated 16 volt level, allow the headphone amplifier to draw large currents directly from the rectifier capacitors at a highly filtered 15 volt level, without degrading the quietness of the regulated supplies.

Mute Circuit. The mute relay provides a turn-on delay of 4 to 6 seconds. Until it is actuated, the preamp output jacks are connected to ground through 330 ohms, so that transients arising in the preamplifier or in other components will not be passed on to the power amplifier. If the unit is turned off, or if power is interrupted or drops below 95 volts (120 V connection), the output jacks are again returned to ground and the same 5-second delay elapses once normal line voltage is restored.

Diode D12 and capacitor C65, fed directly from one side of the secondary of the power transformer, form a fast charging supply to feed the RC timing network R85-C66. When the voltage on capacitor C66, divided by the divider string C69-R87, exceeds the 0.6 V turn-on threshold of Q17, the collector of Q17 pulls down the base of Q18 and turns it on. (R86 continues to bias Q17 after turn-on.) Resistor R88 provides positive feedback to give a sharp, bi-stable turn-on action. When Q18's emitter is pulled down, capacitor C67 supplies momentary base current to transistor Q19, which saturates and provides a surge of turn-on current for positive pull-in of the relay. Following this current surge, resistor R90 continues to supply a smaller hold-in current. At turn-off, diode D13 and resistor R84 rapidly discharge C66; this accounts for the difference between the 5-second turn-on time and the nearly instantaneous turn-off.

4.3

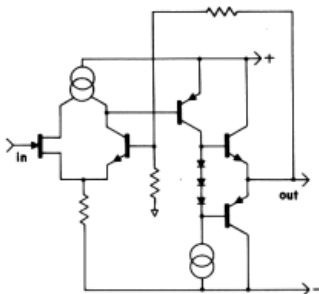


Fig. 4-2. Input Stage.

Design of the Phono Input Stage. The phono cartridge input stage is the section which sets the most important limits on the performance of the preamplifier as a whole, and its design reflects an effort to find the most audibly advantageous compromises among a number of competing objectives. It resembles a class-A complementary-symmetry amplifier with a differential input. This results mainly from bandwidth and noise considerations; in particular, from the general principle that low noise is possible only at low operating impedances, which require relatively high currents. Because every resistance is an inescapable source of noise, the ultimate noise floor of the input stage is set by the inherent Johnson noise of the phono cartridge. For the standard phono cartridge impedance defined by IHF Standard A-202, Section 3.12, this noise has the value -127.3 dBV. Although a lower shorted-noise level could be obtained at the preamp input by using a bipolar transistor biased at several mA of collector current, the source impedance noise of the cartridge cannot be lowered. Hence nothing is lost (and much is gained) by using a FET as the input device, which in this circuit typically exhibits a shorted-input noise of -126 dBV. However, once this is done it is important to keep the noise contributions from other devices in the circuit as low as possible; to do this, the input resistance to the opposite transistor needs to be relatively low, as do a number of other resistances in the circuit. In particular, the noise level in the other half of the differential amplifier is largely

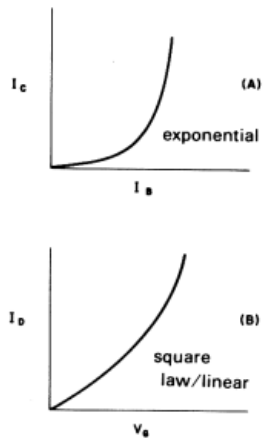


Fig. 4-3. Characteristics of Bipolar (A) and Field Effect (B) Transistors.

set by the effective resistance at the base of Q2. The chosen value of the source impedance on this side (about 360 ohms) guarantees a noise level considerably below that of the FET, but it requires correspondingly low values throughout the rest of the RIAA feedback network. At high frequencies, where the impedance of capacitors C15 and C16 approaches zero, the series impedance of the feedback network approaches 1.4 kohms. Only a complementary-symmetry output stage can drive such a low impedance at levels of many volts without consuming unmanageably large amounts of current and (in changing to higher dissipation transistors) limiting the open-loop bandwidth of the stage. Nevertheless, the inherent, open-loop distortion of this stage is held low by the large emitter resistors, R14 and R15, and by liberal biasing which ensures essentially class-A operation. This configuration also largely eliminates the electrolytic capacitors necessary to maintain dc biasing levels in other circuits, which can introduce non-linearities as the voltage on the capacitor changes.

A FET has been chosen as the input device also because of its relative immunity to radio frequency interference (RFI). The main advantage of the FET lies in its greater linearity relative to a bipolar transistor. Distortion arising in the input device of a differential amplifier is not compensated for as adequately as that arising in devices lying entirely within the feedback loop. As a consequence, a bipolar input device, with its exponential characteristic, is more likely to 'detect', that is, to develop audible distortion products from RF input signals than is the FET with its more linear, square-law characteristic (Fig. 4-3). Sufficient high frequency stability has been achieved to allow placing the open-loop unity gain crossing, set by the Miller capacitor, C14, at 30 MHz. The high open-loop gain-bandwidth product thus obtained also contributes to this stage's immunity to RF interference by ensuring that signals well outside the audio band will be followed linearly, without detection, before they are stripped from the signal by subsequent filtering.

To obtain accurate frequency response the response-setting elements must, of course, be of high accuracy. R16 and R17 are 1 percent; C15 and C16 are 2.5 percent precision parts. The RIAA standard provides for a response pole at 50.05 Hz, a zero at 500.5 Hz, and a pole at 2122.07 Hz. This last pole should give a response falling at 6 dB/octave ad infinitum, but because a negative feedback network must contain a resistor like R10 to develop the feedback voltage, the gain cannot fall below unity. It is nevertheless highly desirable to follow the falling RIAA response well out into the inaudible high frequency region, since this suppresses high frequency noise which could adversely affect the operation of later stages. A single-pole RC filter composed of R20 and C18 continues to follow the high frequency roll-off at around 36 kHz, where the feedback-controlled response levels off.

4.4

Operation of the Phono Input Stage. Both Phono 1 and Phono 2 inputs are terminated immediately by 50 pF at the jack (25 pF disc capacitor plus 25 pF obtained through the circuit board) and additional termination capacitance is added at switch S3 (for Phono 1) to accommodate various phono cartridges and cables. For optimum grounding of RF interference, the ground for this capacitance selector switch is obtained directly through a lockwasher to the rear panel of the case, rather than through the circuit board foil. The input resistance for Phono 1 may be switched between 100 kohms (set by R6) and 47 kohms; the latter resistance is obtained by adding 91 kohms (R1) through switch S2 in series with R6. The FET, of course, does not contribute measurably to the input resistance.

Q1 and Q2 form a differential pair at the input to the stage. Offset voltage for the stage is set by the offset voltage of Q1, and the FET is selected in production for both low offset voltage and low noise. The topology chosen largely eliminates the charging and discharging of capacitors (and possible attendant non-linearities), but turn-on is nevertheless not instantaneous: it takes a couple of seconds to charge capacitor C13 far enough to turn the circuit on. If the offset voltage of Q1 or the capacitance of C13 is too high, the turn-on time can thus exceed the turn-on time of the muting relay.

The drain of Q1 and the collector of Q2 are actively loaded by the network composed of R7, R8, Q3, and D1. Q4 provides high voltage gain; its collector is sunk through diode string D2, D3, D4 to the current source Q5 and drives a complementary-symmetry output stage formed by Q6 and Q7. RIAA equalization is obtained from the feedback loop, R16, R17, C15, C16, which drives the amplifier's inverting input. These are 1 percent and 2.5 percent parts, respectively. Open-loop compensation is provided by Miller capacitor, C14,

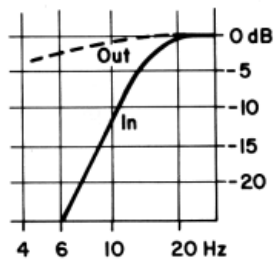


Fig. 4-4. Infrasonic Filter Characteristics.

4.5

and further high frequency roll-off, as required by the RIAA specification, is provided beyond 30 kHz by the filter R20, C18. Supplementary power supply filtering, required to maintain the low noise level of this stage, is obtained through Q8 and Q9, whose bases are shorted to ac ground through the capacitors C19 and C20; R21 and R22 supply base current.

The Phono 2 inputs feed the selector switch through a series of jumpers, numbered 1 through 4 on the circuit board. These bypass the connector for the pre-preamplifier and must be cut or removed if a pre-preamplifier or special termination network is installed (see Section 9.1).

Trimpots R121 and R122 are alternately inserted in the feedback loop of the right channel to trim its gain in the Phono 1 and Phono 2 settings and equalize imbalances which may be present in the cartridges used. The middle section of the selector switch, S1A, grounds the right channel feedback loop through one or the other of the trimpots.

Infrasonic Filter. Capacitors C21, C22, and C23, with resistors R22, R23, and R24 form a 3-pole, 18 dB/octave active high pass filter which removes signals below 20 Hz. When the Infrasonic Defeat Switch S4 is closed (out position), the response of the stage is set by the filter C23-R24, which is down 3 dB at 5.5 Hz.

4.6

External Loops. The direct signal path from IC1 to the Volume control input buffer, Q10, can be interrupted by any or all of the switches S5, S6, and S9. When a tape monitor button is engaged, the signal path is interrupted and the signal passed on to the Main Output is taken from the selected "in from" jacks. The preamplifier input signal (e. g., Phono, Tuner) is still available at the "out to" jacks, buffered by IC2, IC3, or IC4. Each of these buffers has a 100 ohm build-out resistor, and feedback to the inverting input is applied through 1 kohm. A 0.01 μ F capacitor connecting the output directly to the inverting input compensates for reactive loading of the buffer and ensures that it will remain completely stable. The "out to" signal, when returned through the "in from" jacks, reenters the direct signal path through the third section of the tape monitor switch. Open contacts of the switches are grounded to avoid capacitive crosstalk. The Dub switches send the signal from one set of "in from" jacks to the opposite "out to" jacks (to allow dubbing from one tape recorder to another), isolating it through the "out to" buffer.

4.7

Volume Control. Instead of forming a variable attenuator in the signal path, the Volume control potentiometer actively changes the gain of an inverting amplifier. This novel configuration circumvents the noise problems associated with conventional volume controls. At intermediate levels a conventional passive attenuator presents to the input of the following stage a relatively large resistance whose Johnson noise is *not* attenuated. As a result, equipment which exhibits excellent signal-to-noise characteristics in the volume down and volume up settings very often suffers a quite audible degradation in its signal-to-noise performance in the middle range of the volume control. In the configuration used here the thermal noise contribution is limited to the parallel resistive sum of R39 and R40, which is always less than 1 kohm.

Q10 forms a common-collector buffer, loaded by the current sink Q11, capable of driving the low, 1 kohm impedance which the Volume control presents at maximum gain. A low-resistance potentiometer has been chosen to give the lowest practical thermal noise; in addition, a bipolar differential pair is placed in front of the IC amplifier's input in order to obtain an optimum noise match to the potentiometer. This additional circuitry amounts to nothing more than an extension of the IC amplifier's differential input stage. Because of the additional stage of open loop gain this adds to the total configuration, however, the open loop gain of the Volume control is quite high, and additional stabilizing components, R41, R42, C36, C37, must be added.

An inverting operational amplifier, upon which the control is based, has the basic topology of Fig. 4-5. Since the positive input is grounded, the amplifier will produce at its output whatever voltage is necessary to bring the voltage at its negative input back to zero (i. e., it tries to keep the difference between the two inputs close to zero). Suppose $R_1 = R_2$, and that we apply a positive voltage to the input. The amplifier will produce an equal but opposite voltage at its output so that the voltage at the midpoint of the divider R_1 - R_2 remains zero. If R_2 is increased, the amplifier will have to produce a larger voltage to maintain zero volts at its positive input.

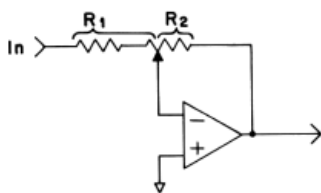


Fig. 4-5. A Variable-Gain Inverting Amplifier.

In the Volume control, the 1 kohm resistor R39 along with part of the potentiometer form R_1 , and the remaining section of the potentiometer forms R_2 . Hence the gain at full volume is given by the ratio R_2/R_1 , or 10 kohm/1 kohm, which gives 20 dB of maximum gain. (The contributions of other resistances around the control actually reduce this to about 18 dB.) The input signal to the control appears at one end of the potentiometer; the output appears at the opposite end; and the wiper (held at zero volts by the amplifier) actually shows no signal at all.

The attenuator steps are not all equal, since finer gradations are more important in the usual listening range than at the lower extreme of the control. Starting from the fully clockwise, maximum output position, the stepwise attenuation is

Steps	Attenuation
1-15	1.5 dB
15-21	2 dB
21-25	3 dB
25-31	4 dB

From step 31 to step 32 is, of course, infinite, since the bottom step represents no output.

4.8

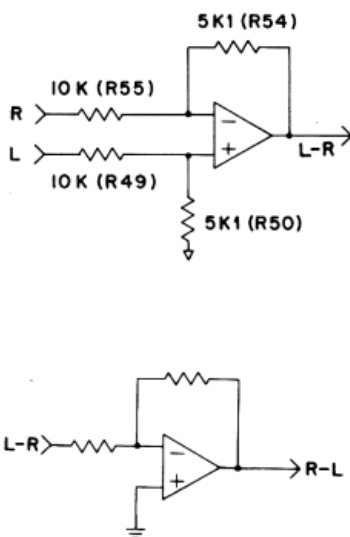


Fig. 4-6. L - R and R - L Amplifiers.

Mode Control. This control (patent pending) blends the two stereo channels present at its input to give either a monaural output (a sum of the two signals), a stereo output (the control is effectively disengaged), or a difference output (one channel is inverted and added to the other). Intermediate positions allow precise control of depth and spaciousness in the stereo image formed at the loudspeakers.

Right and left channel signals from the Volume control are fed to the center taps of the two ganged potentiometers, R52 and R152, in the Mode control; the output passed on in both channels is taken from the two control wipers. In the detented, "stereo," position, the control wiper contacts the tap coming from the Volume control; the signal from each channel passes on directly to the balance control, and the Mode control is in effect ignored.

Consider the left channel alone: the left channel signal from the Volume control is applied to the center tap of the control, and the signal passed on is taken from the wiper. At the monaural, "L + R," end of the control appears the monaural signal $(L + R)/2$, so that as the wiper is advanced toward this end, the signal passed on in the left channel is increasingly the sum of R and L, i. e., a monaural signal. At the opposite end of the control ("L - R") appears the signal $(L - R)/2$, so that as the wiper is advanced in this direction, the signal passed on is increasingly the difference between L and R.

Returning to the counter-clockwise end of the control, we see here the signal $(L + R)/2$. This signal results very simply from the fact that both gangs of the control are connected together at their counter-clockwise ends. A signal from one Volume control output will pass to the center tap of one control, through the 5 kohm left-hand section, to the corresponding 5 kohm section of the other potentiometer, and out to the virtual ac ground formed by the low-impedance output of the other Volume control. Hence, both the R and L signals will appear at half their original amplitudes at the counter-clockwise ends of both potentiometers as $L/2 + R/2$. For a monaural input voltage to both channels, $v = v$, a monaural signal will emerge with the same amplitude as the input ($v/2 + v/2 = v$); for a monaural input to one channel only, the output at both channels will be $v/2 + 0$, in other words, 6 dB down from the input.

The difference signal is derived through a difference amplifier in a somewhat unusual configuration (Fig. 4-6). The difference signal applied at the clockwise ("L - R") end of the control has the amplitude $L/2 - R/2$. This signal is derived as follows: the divider R49-R50 divides the signal from the left Volume control by 3; this $L/3$ signal is applied to the non-inverting input of IC6A. This amplifier does not work at unity gain; for signals applied at the non-inverting input, R51 and R55 form a negative feedback loop which sets the gain of the amplifier at 15 kohm/10 kohm, or 3/2. (Note that R55 does not

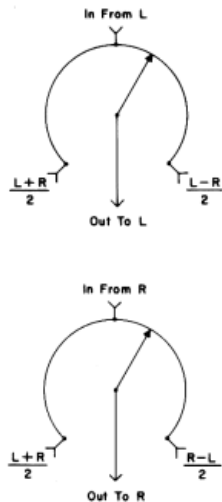


Fig. 4-7. Addition Modes in Mode Control.

actually return the feedback signal to ground, but to the output of the right channel Volume control. Because the output impedance of the Volume control is very nearly zero, its output presents a virtual ground to ac.) Thus the $L/3$ signal applied at the non-inverting input is amplified by $3/2$, and the left channel signal appearing at the output of IC6A is given by $L/3 \times 3/2 = L/2$.

The right channel arrives at the output of the amplifier in a different way. Looking at IC6A from the right channel, one sees an inverting amplifier with a gain of $-(R51/R55) = -(1/2)$. Hence, the right channel signal appearing at the output has the amplitude $-(R/2)$, and the total signal at the output is thus $L/2 - R/2$.

This signal, however, is sent only to the left channel potentiometer. To obtain proper signal addition in the right channel, the difference signal must be inverted, to give $R/2 - L/2$. IC7A is a unity-gain inverting amplifier which applies the inverted difference signal to the right channel potentiometer. Referring to Fig. 4-7, consider the wiper position to be given by the quantity a , where $-1 \leq a \leq +1$. The signal passed to the left channel output is then

$$(1 - a)(L/2) + [a(R - L)]/2$$

and the signal to the right channel output is

$$(1 - a)(R/2) + [a(R - L)]/2.$$

Note that in the "L - R" position, only the left channel shows $L - R$; the right channel actually shows the inversion of this, $R - L$.

4.9

Balance Control. The signals leaving the Mode control can be attenuated in the right or the left channel by rotation of the ganged potentiometers, R56 and R156. As elsewhere, 10 kohm pots have been chosen for minimum noise (so that the maximum thermal noise here is that from 2.5 kohm). IC6B, IC7B, and IC8 provide input and output buffers to prevent interactions with other controls.

4.10

Tone Controls. When the Tone Defeat switch is activated, IC10 is connected as a simple, unity-gain inverting amplifier which feeds the preamplifier outputs. With the tone controls engaged, the input to IC10 is through the high filter and tone control circuits.

Before entering the tone control section, the signal passes through a low pass filter which strips it of undesired high frequency components. The input signal to the tone control passes through a common collector stage formed by Q14 and its current sink, Q15. A 2-pole Butterworth filter, placed in a positive feedback loop around Q14 (modeled on a Sallen-Key active filter) provides 12 dB/octave rolloff beyond 40 kHz; when the High Filter switch is engaged, the low pass knee shifts to 8 kHz to suppress noise from poor record surfaces or other noisy program sources.

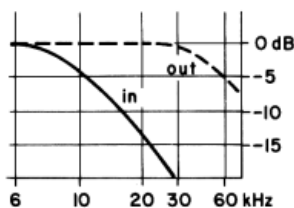


Fig. 4-8. Ultrasonic Filter.

IC9 functions as a buffer, passing the signal to the input of the tone control amplifier.

This section has the basic configuration of a Baxandall feedback tone control, but shows some important modifications (patent pending). The controls are inserted in the negative feedback loop of IC10. At mid-band frequencies the feedback path is mainly through the two 8.2 kohm resistors R68 and R71; the 1 : 1 ratio of these resistors sets the gain of the stage at unity. The emitter-follower Q16 provides a low enough source impedance to override the contributions of other resistances to the feedback signal. At low frequencies the reactance of the 0.47 μ F capacitor, C49, begins to eliminate the signal furnished by the fixed divider, R68-R71, so that the feedback signal is returned mainly from the Bass control, R64. Hence, at low frequencies, the gain varies according to the ratio of the two resistances on either side of the control wiper. (The use of a single capacitor to generate the low frequency characteristic in this design eliminates the mid-band response ripples usually associated with the Baxandall circuit.) A second capacitor, C48, may be added to alter the Bass control characteristic (non-shelving mode). At high frequencies the feed-back signal from R63 can pass easily through the 0.01 capacitor C50, through R72 and R73 to ac ground at the emitter of Q16, the Bass control being in effect isolated. Thus, at high frequencies the amplifier gain is set by the ratio of the two resistances on either side of the control wiper of R63. The 100 kohm trimmers R62 and R64 allow precise calibration of each control to ensure absolutely flat response

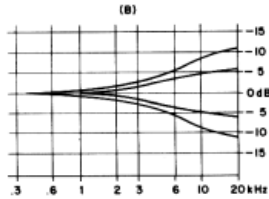
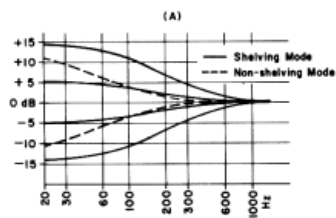


Fig. 4-9. Bass (A) and Treble (B) Control Characteristics

at the mid-rotation, detented position.

Note that there is no output buffer to the main outputs. However, under normal conditions the tone control amplifier is isolated from the output to a considerable extent; much of the output current is actually supplied by IC11, the negative conductance. This amplifier has a gain of about 1 dB, set by the negative feedback loop, R98-R99. The output of the tone control amplifier, IC10, thus looks into a signal 10 percent greater than itself in amplitude through 220 ohms (R100), so that the effective load on the output is reduced.

4.10

Headphone Amplifier. A separate headphone amplifier output at a higher level is provided by IC12, a non-inverting amplifier with 10 dB of gain and high current capability. Separate regulators supply this amplifier so that large amounts of current can be supplied without affecting the quietness of the regulated supplies to the rest of the unit. Q20 and Q21 draw collector current directly from the unregulated supply, from C59 and C62; only the base current comes from the IC regulators.

5 Block and Level Diagrams, Functional Tests

- 5.1** Before disassembling the unit, it is advisable to check quickly all of the preamplifier's functions, both to make sure that the problem you wish to repair has been correctly identified and to isolate the section of the unit in which the problem occurs (see Fig. 5-1). In many cases you will find that, because of the number of controls, much of your initial troubleshooting can be done before the case is even removed. If you can identify a problem as belonging to the switchboard (areas in dashed lines on the schematic), you will not need to remove the unit's top cover; releasing the switchboard by its four mounting screws will suffice (see Section 6.4).
- 5.2** Apply a 13 mV, 1 kHz signal to the Phono 1 input, left channel. At each tape out L and at external processor L observe 0.87 V, undistorted. Maximum rotation of the Volume control should give 7 V at Main Out 1 and Main Out 2, L. Left channel Phones should show clipping at 28 V peak-to-peak (10 V rms). Repeat for right channel. (See Level Diagram, Fig. 5-2.)
- 5.3** Use the Block Diagram, Fig. 5-1, to isolate the problem. Note, for example, that a suspected problem in the muting relay can be identified by examining the output of the headphone amplifier. Likewise, a problem suspected in the tone controls can be positively identified as such by activating Tone Defeat.

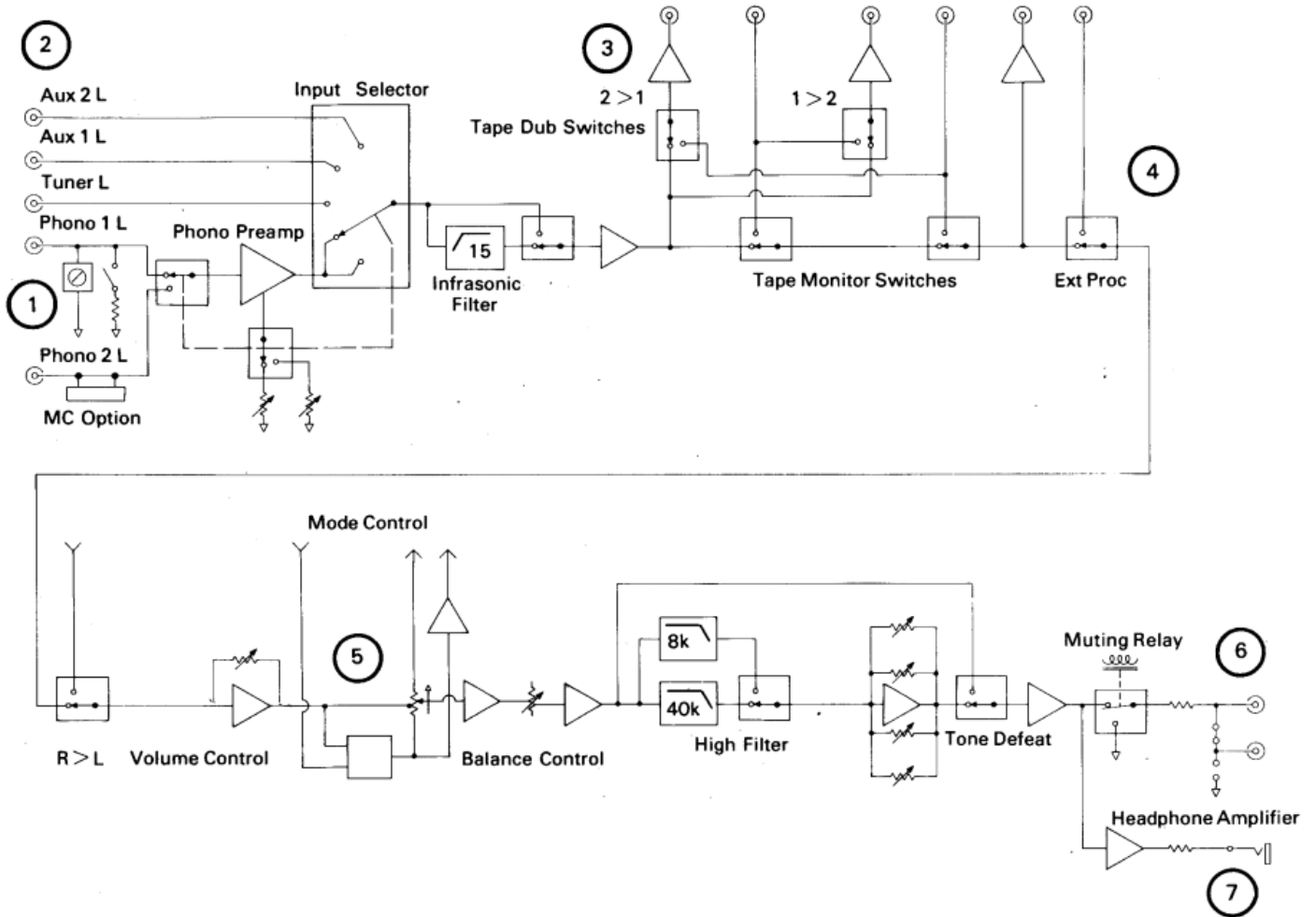


Fig. 5-1. Block Diagram.

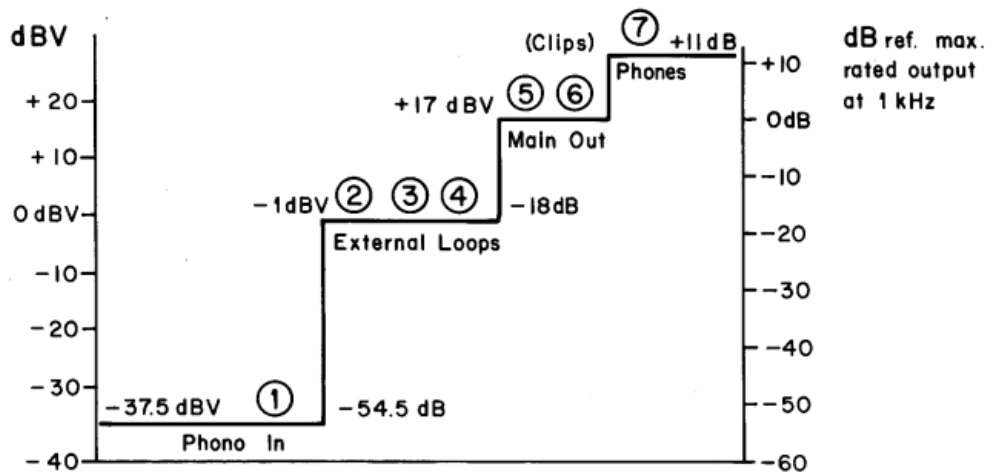


Fig. 5-2. Level Diagram.

6 Mechanical Disassembly

- 6.1 Tools:** Number 1 Phillips screwdriver
Number 2 Phillips screwdriver
2 mm hex driver
1/16 in. hex driver
3/16 in. flat screwdriver
5/16 in. hex nut driver
7/16 in. hex nut driver

6.2 Bottom Panel. Remove the nine number 6 sheet metal screws with a number 2 Phillips driver. If you have determined from functional tests that the problem to be repaired is located on the switchboard, you may wish simply to remove the switchboard.

6.3 Switchboard and Jackboard. Remove the four 4-40 machine screws marked S in Fig. 6-3 and lift up the rear of the board (toward the power transformer). Avoid flexing flat cables any more than is necessary. Nearly all parts are accessible once the board has been lifted. When reinserting this board, pull all of the toggle switches up to allow the toggle levers to enter the front panel holes easily.

For easy access to a few parts, you may want to remove the jackboard. Remove the screws marked J in Fig. 6-1.

6.3 Top Cover. Remove the five no. 6 sheet metal screws in the back panel marked C in Fig. 6-1 with a no. 2 Phillips driver. Remove the five 4-40 flat head socket screws marked C in Fig. 6-2 with a 1/16 in. hex driver. Spring the two ends of the case slightly away from each other and lift off the cover.

Further disassembly is rarely necessary.

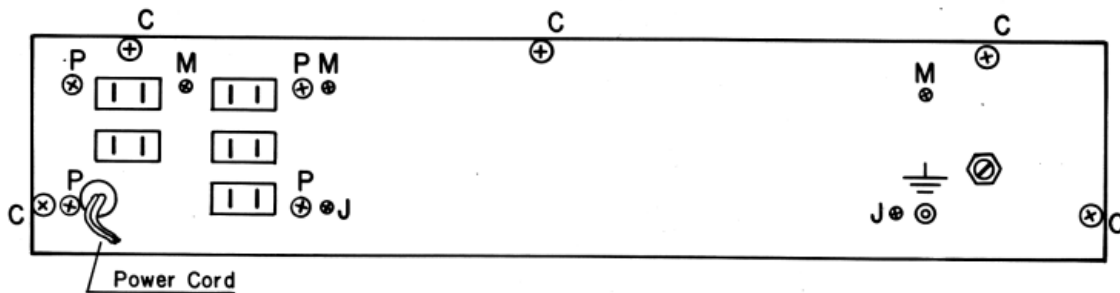


Fig. 6-1. Back Panel.

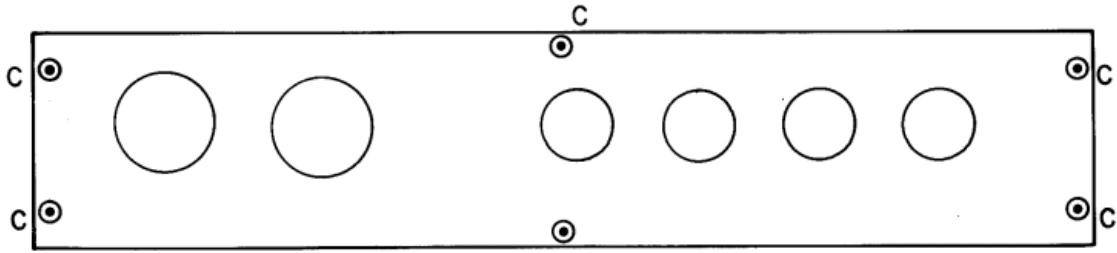


Fig. 6-2. Front Panel.

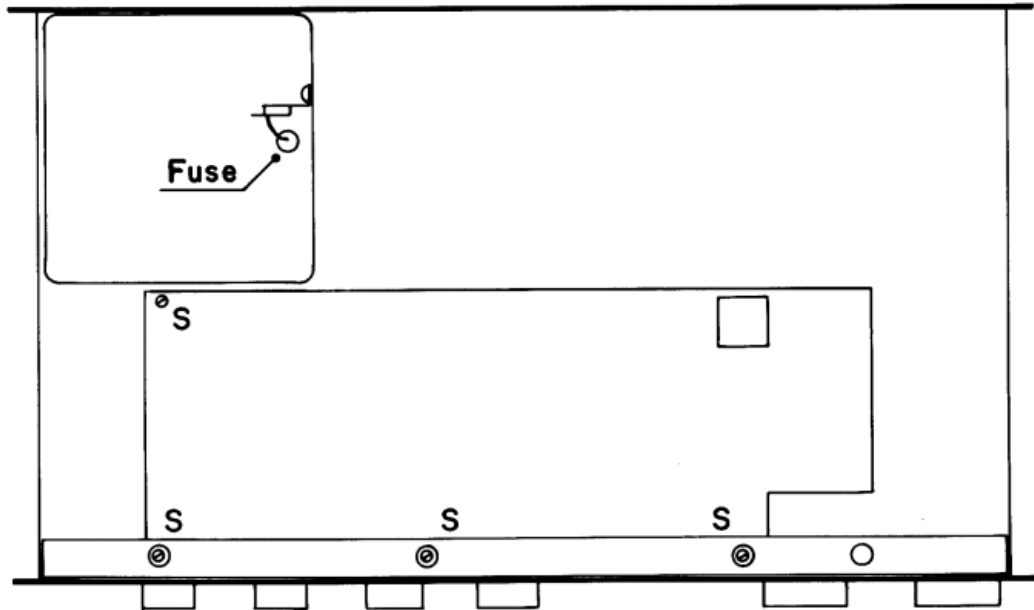


Fig. 6-3. Switchboard Mounting.

- 6.5 **Rear Panel.** After removing the case, remove the no. 6 sheet metal screws marked P (power supply bracket) in Fig. 6-1 with a no. 2 Phillips driver, and the machine screws marked M (main board) and J (jack board) with a no. 1 Phillips driver. Remove the 8 mm hex nut on the Load: Capacitive switch. This will allow you to pull the rear panel loose; for complete removal, it will be necessary to unsolder the line cord, the 120 Vac outlets, and the +18 V jack.
- 6.6 **Front Panel.** Remove the knobs with a 2 mm hex driver. Remove the sixth 4-40 flat socket machine screw (unmarked, Fig. 6-2) with a 1/16 in. hex driver. Remove the control nuts and washers, and set aside the nut and washer from the Volume control. (This nut and washer are 9 mm; all others are 8 mm.) The front panel will now slide off.

6.7 Sub Front Panel. This plate mounts the headphone jack, the power switch, and the LED. It is secured between the control potentiometers and the front panel by the control hex nuts and is mounted to the front panel by the unmarked socket screw in Fig. 6-2.

The LED, D13, is mounted as follows: with front panel and sub front panel securely mounted in the unit (cover removed), insert the new LED far enough to let the rounded lens project from the front panel, but do not let the cylindrical section show. Secure the LED with hot melt glue (or in a pinch, with silicone rubber adhesive), surrounding the diode case entirely.

The headphone jack mounts with a 3/8 in. flatwasher and 3/8 in. hex nut in the well provided in the sub front panel.

The power switch mounts on standoffs with two machine screws, 4-40 × 3/8 in.

6.8 Power Supply and Power Switch. Except for the power switch, all line voltage wiring is contained inside the steel power supply bracket. After the rear panel is taken off, removal of five more 4-40 × 1/4 in. machine screws (four in the main circuit board, one in the switchboard, marked S in Fig. 6-3) will release the bracket.

The pigtail fuse is soldered to the lug bracket; it can be replaced, however, without removing the power supply bracket by simply clipping the leads of the blown fuse and carefully soldering a new one to the lugs.

The power switch assembly must be replaced by removing the mounting screws on the sub front panel and unsoldering the leads at the power outlets on the rear panel.

7 Test Points and Troubleshooting

- 7.1 General.** Familiarize yourself with all of the front panel controls and with the Block Diagram (Fig. 5-1). Much can be learned very quickly from intelligent use of the controls, and a problem can often be more quickly localized in this way than by hunting for test points.

Because the preamp uses IC operational amplifiers to a great extent, there are no large-scale feedback loops and, in general, failures can be quickly isolated as belonging to a particular functional stage (e. g., Infrasonic Filter, Balance control). Signal tracing with a *dc-coupled oscilloscope* of a signal injected at phono or an appropriate high level input is generally the best procedure, since it allows you to check both signal and dc voltages with the same probe. Checking for correct dc operation is an indispensable key to troubleshooting all IC circuits.

An abnormal load such as a short may cause apparent failure of a stage which is in fact functioning properly. Likewise, slewing distortion (triangular output from sine wave input) may be a sign not of an amplifier fault but of overloading by the following stage. If in doubt, always disconnect the following stage to be sure you have located the fault properly.

While all parts have been chosen for reliability, the JFET is necessarily a delicate part and can be damaged by high static voltages. The TL072 IC operational amplifier and the 2SC1345 transistor have shown every evidence of being extremely reliable.

The Volume control and Mode control circuits will be somewhat unfamiliar configurations for most technicians, and the explanations of these should be read carefully (Sections 4.7 and 4.8). A problem occurring in both channels is probably power supply related. Lack of full channel separation, for example, can result from a fault in a supply regulator which causes the regulator to present a tangible ac resistance to ground, which in effect inserts a resistance common to both channels in some part of the signal path.

- 7.2** Before starting to troubleshoot, bring the unit up to line voltage on a variac to avoid potential damage. The unit should draw 100 mA \pm 10 % at 120 Vac (50 mA at 240 Vac).

7.3 Power Supplies.

You should observe the following voltages at the test points indicated below.

S1 +25 V	(C59)	<i>Unregulated supply</i>
S2 -25 V	(C62)	
S3 +18 ± 0.4 V	(IC12)	<i>Regulated supply</i>
S4 -18 ± 0.4 V	(IC13)	
S5 +16 V		<i>Op amp supply</i>
S6 -16 V		
S7 +15 V	(Q20)	<i>Input stage supply</i>
S8 -15 V	(Q21)	
S9 +16.5 V	(Q8)	<i>Headphone amp supply</i>
S10 -16.5 V	(Q9)	

Hum or noise appearing in both channels will frequently indicate a noisy power supply rail. You should measure the following noise levels (shorted inputs):

<i>Test Points</i>	<i>Unweighted (DIN Audio Band)</i>	<i>A-weighted</i>
S3—S8	< -60 dBV	< -106 dBV
S9, S10	< -75 dBV	< -120 dBV

Note that the IC regulators show considerable variation in noise and hum rejection; the negative regulators are often 10 dB quieter than the positive regulators (the values given above are maximum, not typical, values).

7.4 Troubleshooting for Hum. Once you have determined that the power supply rails do not exceed their nominal noise levels, grounding problems should be suspected. Because of the levels involved, locating a minor hum problem may require that you take factors less commonly associated with hum into account. Supply ripple, for example, may be introduced by resistive grounding, e. g., by faulty grounding of the 1000 μ F supply capacitors, C59 and C62. Inspect possible foil ground problems. Check the grounding of the main board foil at point G1 (through the screw marked G1 in Fig. 6-1) to the bracket which fastens the rear panel; this must be a firm, low resistance contact. The steel power supply bracket should connect to foil ground only through the rear panel and rear panel bracket at G1. Observe that the grounds to phono inputs are *not* obtained through the foil passing along the rear panel; the capacitor C1 obtains RF ground only, and if shorted can cause hum susceptibility in phono inputs.

7.5 Mute Circuit. Despite the tightly-fitting cover, dirt can get on the relay contacts. If a VOM shows greater than zero resistance across closed contacts, do not attempt to remove the cover to clean them (this will usually damage the relay); install a new relay. If the relay fails to operate, first test the relay itself by finding the +25 V supply pin and then shorting the opposite pin, test point R1, to ground. This should cause the relay to pull in strongly.

Test point voltages after relay pull-in should be:

R1 +16 V
R2 +1.4 V

7.6 Phono Input Stage. For troubleshooting purposes, the phono input stage can be regarded as a differential input power amplifier. The following voltages should be observed:

- PL1, PR10.0 V (exactly)
- PL2, PR2 +15 V
- PL3, PR3 0 V \pm 1 V
- PL4, PR4 +15.3 V
- PL5, PR5 +3 V
- PL6, PR6 +2 V

The voltage across the emitter-follower resistors, R15 and R115, should be close to 0.4 V. The voltage at PL1 and PR1 should be *exactly* 0.0 V; a FET which shows any voltage at its gate should be replaced.

A repair involving either the FET or capacitor C13 may occasionally cause the turn-on time of the phono stage to exceed that of the mute relay (particularly if the mute relay has a fast turn-on time), since the turn-on time rises proportionally to the offset voltage of the FET and the value of C13.

- 7.6.1** Phono Frequency Response and Impedance. Since both of these parameters interact, always check both before diagnosing a problem as one or the other (follow the tests in Sections 8.6 and 8.8). If a problem is suspected of being a genuine frequency response error, the unit should be returned to the factory so that frequency response can be ascertained to the specified degree of accuracy.
- 7.6.2** Noise. The most sensitive components are the FET (Q1), the diodes D1, D5, D6, and the 3.3 kohm resistor, R9. Noise appearing in both channels may be caused by supply rejection transistors, Q8 or Q9. If replacing these does not reduce the noise, check the connections to their bases. Collector-emitter leakage can occasionally allow the transistor to operate without an external source of base current. High frequency oscillations can also appear indirectly as noise.
- 7.6.3** Distortion. Again, high frequency oscillation or ringing may show up as distortion. Examine the output of the phono input stage with a wide-band oscilloscope (15 MHz or more) before the 10 μ F output capacitor, C17. Check all voltages and currents. Diodes D2, D3, D4 should each show close to 0.7 V.

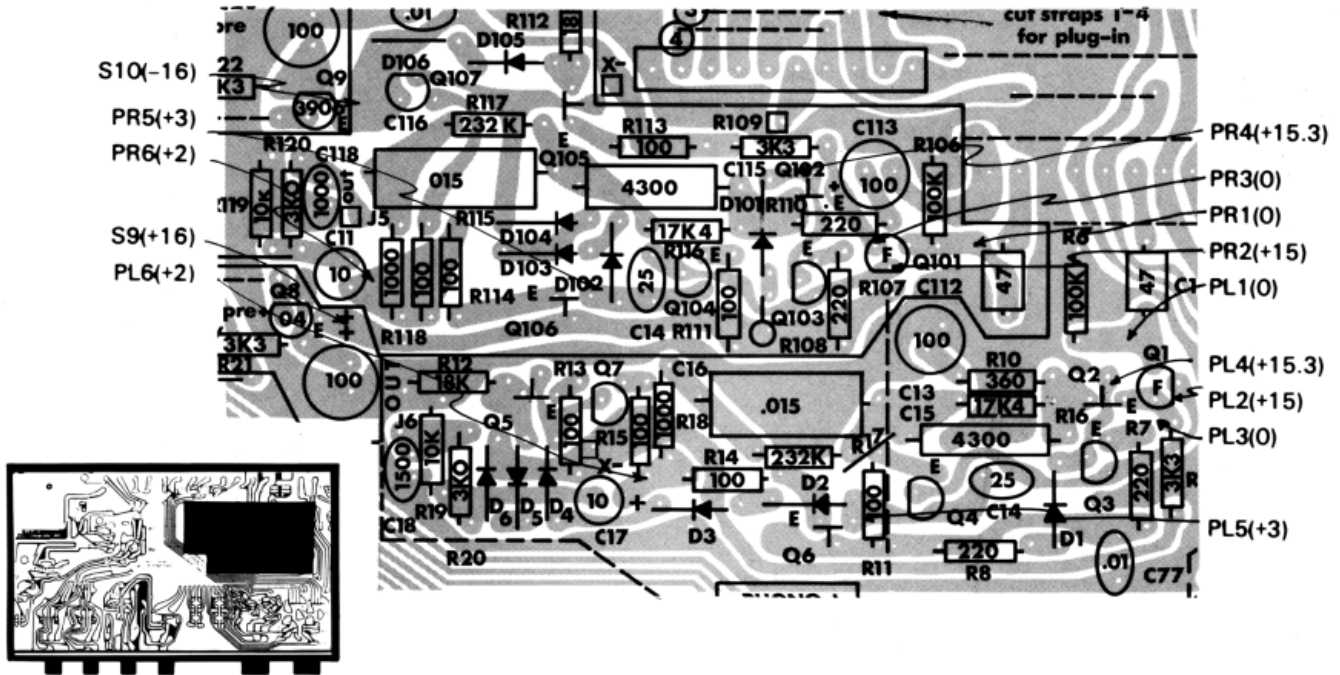


Fig. 7-3. Phono Input Stage Layout, Component Side.

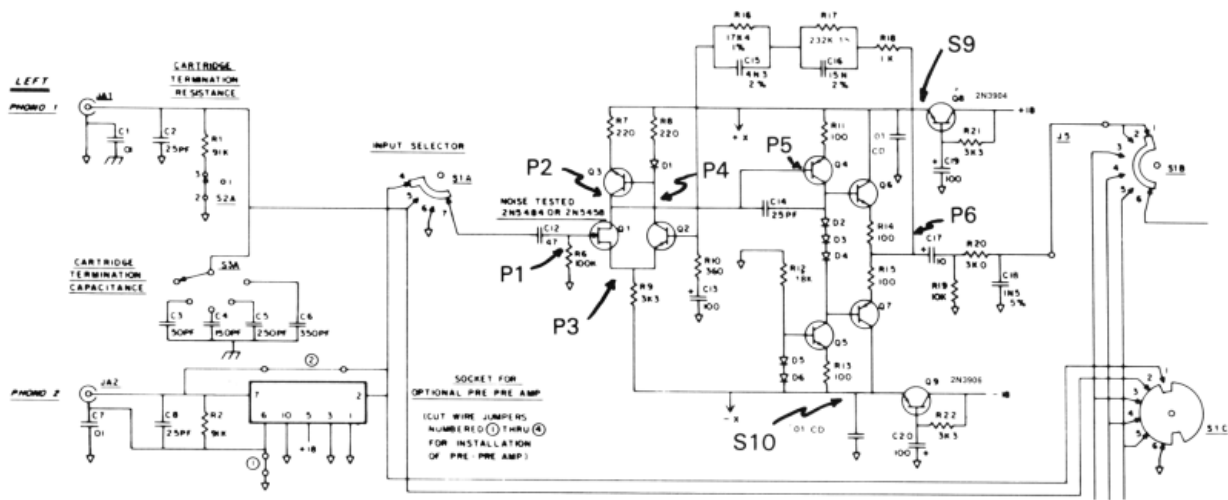
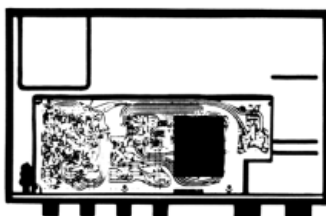
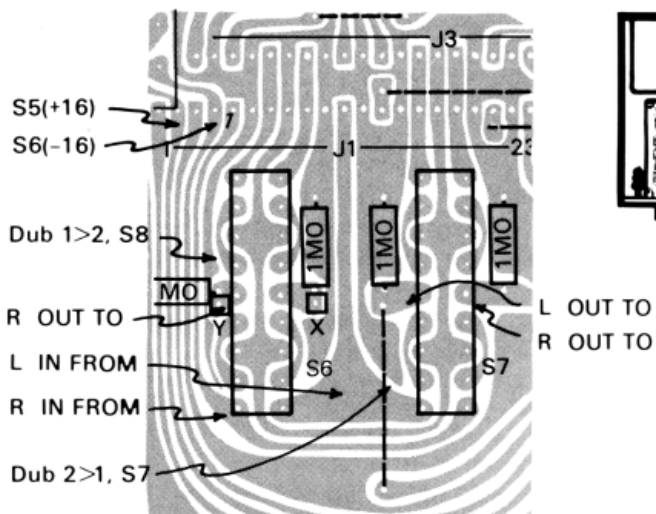


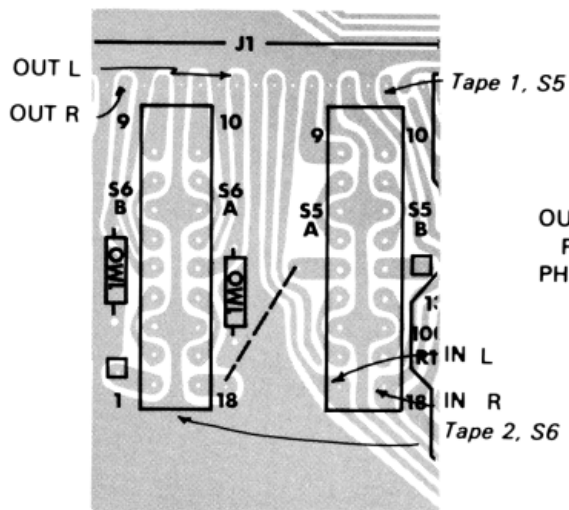
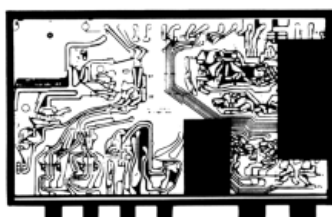
Fig. 7-4. Phono Input Stage Schematic.

7.7 Switching and External Loops. Note that the selector switch (S1) is provided with three decks; the deck closest to the rear switches between inputs Phono 1 and Phono 2 as well as between the two phono balance trimpots, R121 and R122. The middle deck shorts all inputs not being used (including phono) to ground, and the deck closest to the front selects the input passed on to the infrasonic filter, IC1. Note that loss of an input can occur because of improper shorting in the middle deck as well as because of open connections.

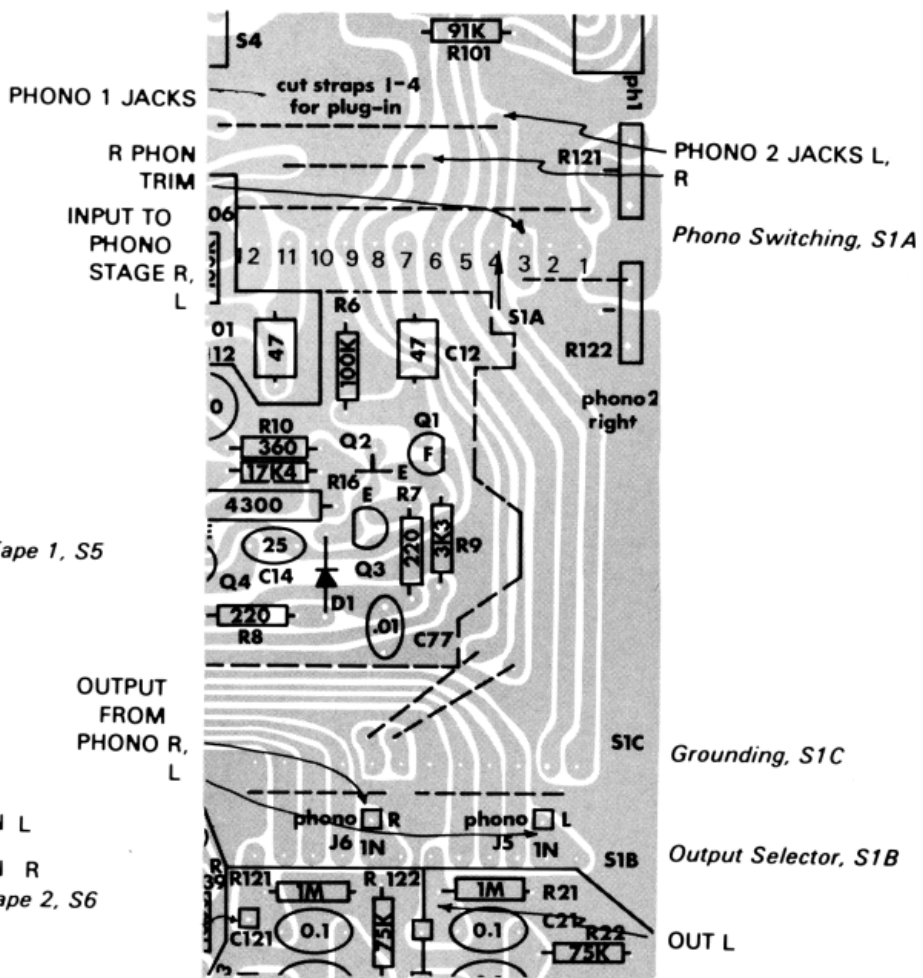
Test points are indicated in Fig. 7-5 for the Selector, Tape, and Dub switches. Note that the flat cables between main and switch boards carry only half as many live conductors as they appear to, since every second conductor is grounded to provide electrostatic shielding.



Dub Switches



Tape Switches



Selector Switch

Fig. 7-5. Selector and External Loops, Layout, Component Side.

7.8 Volume, Mode, Balance Controls. Note that the Volume control presents an unconventional configuration, consisting of an input buffer (Q10) to a variable gain inverting amplifier (see Section 4.7). The control potentiometer employs a highly reliable, multi-wiper construction; however, in the event that all wipers should lose contact with the film, the signal will not drop out, as it does in voltage dividers, but will instead rise to a high level set by the ratio of the 101 kohms series resistance of R41 and R42 to the output impedance of IC5. Note that the inverting amplifier functions in such a way here as to try to maintain the wiper of the Volume control at zero volts; a properly operating control will therefore show no signal on the wiper.

Abrupt increases in gain as the Volume control is turned will indicate a faulty potentiometer. Contact cleaning sprays may restore the control, but they tend to wash out lubrication, and should be used sparingly, if at all.

7.8.1 Hum in the Volume control. Make sure that the control bushing makes electrical contact with the sub front panel, which in turn must make electrical contact with the case (which must be its only ground connection). Shielding of all the potentiometers is *only* through the ground contact to the sub front panel.

7.8.2 Test Points:

- VL1, VR1 Input to Volume control
- VL2, VR2 Output of IC5A, IC5B
- M1 Left minus right output (Mode control)

7.8.3 Mode control. Refer to Section 4.8 for theory of operation.

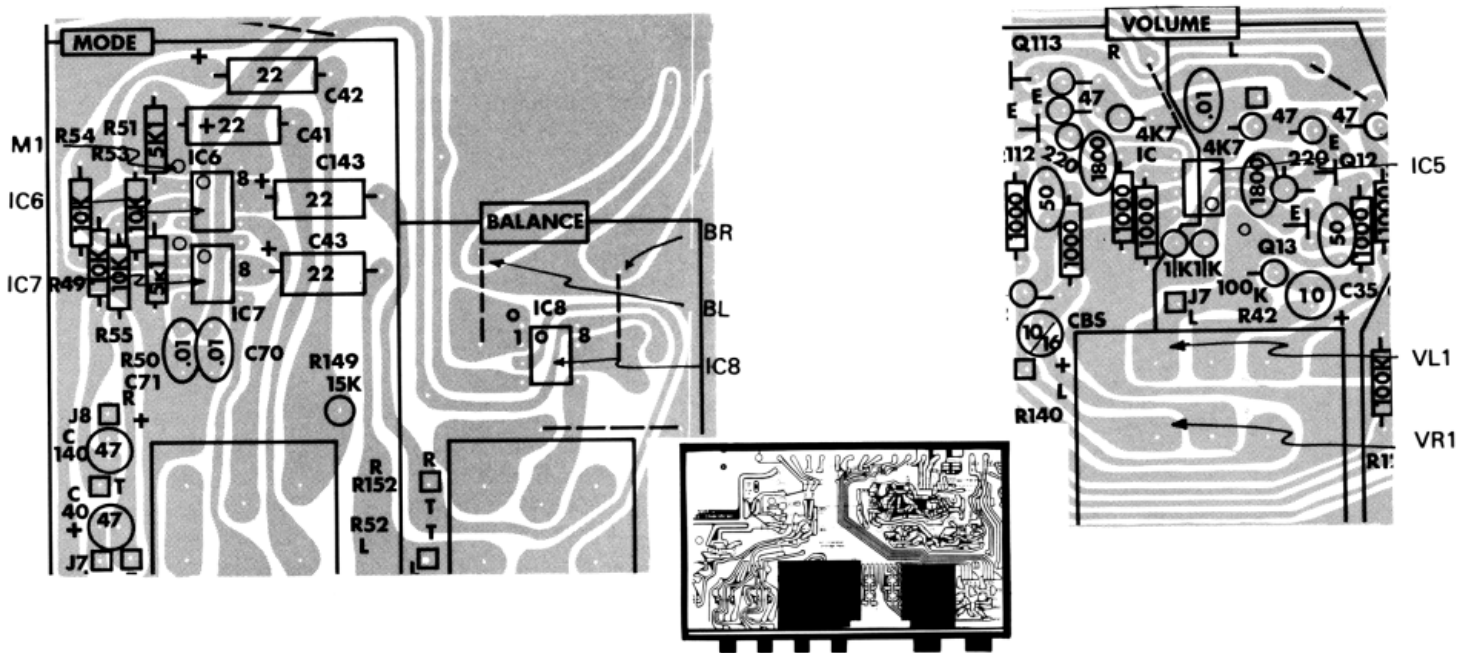


Fig. 7-6. Volume, Mode, Balance Controls, Layout, Component Side.

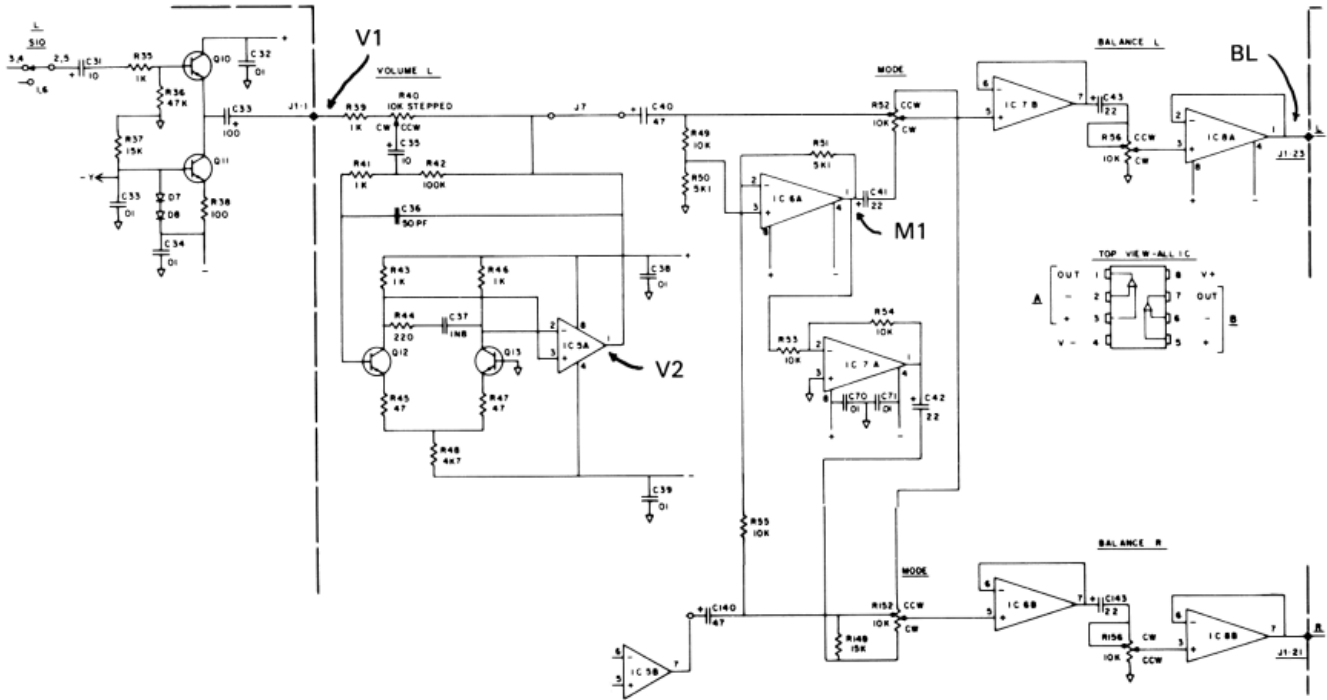


Fig. 7-7. Volume, Mode, Balance Schematic.

- 7.9 Tone Controls.** Observe that the tone control is a somewhat modified Baxandall configuration. Operation of the Bass control depends on the impedance transformer Q16. An apparent fault in the treble control may result from a fault in the high filter (in the base-emitter feedback around Q14).
- 7.9.1 Intermittent or Faulty Toggle Switches.** Spray the switch with contact cleaner spray (*not* with a lubricant like WD-40) and immediately apply a soldering iron to the solder connections of the switch, while flipping the switch some 10 to 20 times. Replace the switch if this does not completely restore sure electrical contact.

Test Points:

- TL1, TR1 Driver output, dc level -0.6 V
- TL2, TR2 Emitters of Q16, Q116, dc level -1.2 V

- 7.9.2 Distortion.** Many factors may contribute to low level distortion—slight RF instabilities, for instance, arising from a fault in an RF stabilizing component, may show up as abnormally high second harmonic distortion. Failure of the negative conductance, IC11, may raise the distortion level at the Main Out jack to as much as -76 dB or .016 percent. Small amounts of distortion may also be introduced by dirt in control or trimmer potentiometers (which must *not*, however, be cleaned or reset without consulting the factory, since their setting influences the frequency response of the tone control section).
- 7.10 Headphone Amplifier.** You should observe a gain of 11 dB relative to the Main Out jacks. Clipping, unloaded, will come at about 10 V rms (28 V peak-to-peak).

Tone Control Section

Headphone Amp

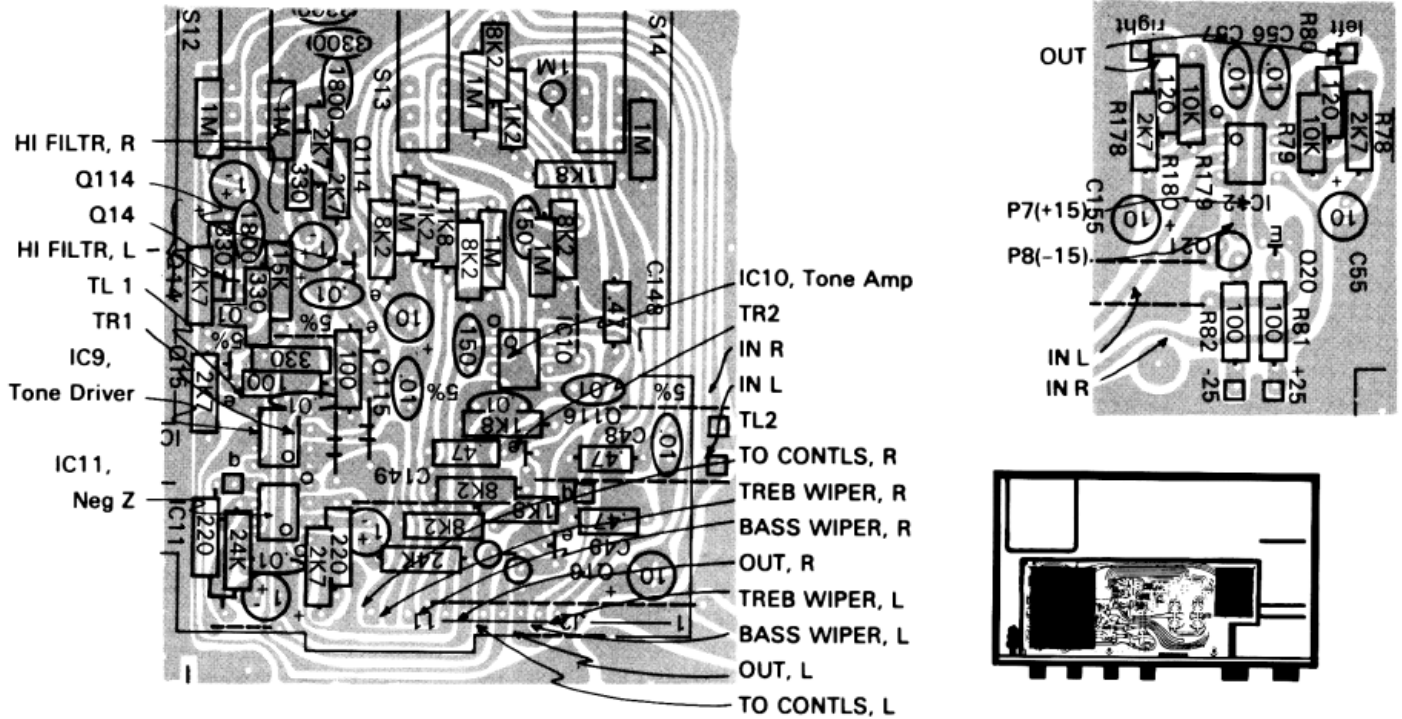


Fig. 7-8. Tone Control and Headphone Amplifier Layout, Component Side.

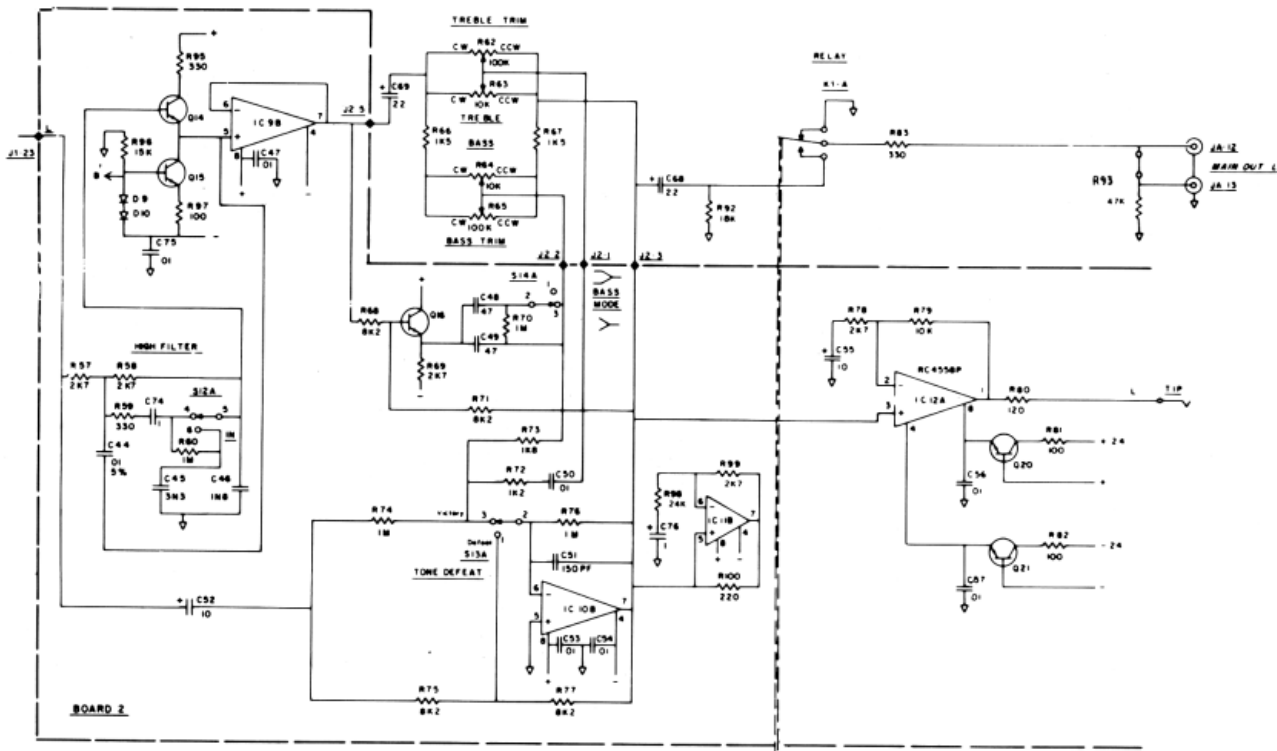


Fig. 7-9. Tone Control and Headphone Amplifier Schematic.

8 Final Test of Specifications

- 8.1** The following procedure should be performed after all repairs or adjustments have been completed to ensure that the preamplifier still meets all of its published specifications. It is essentially the same set of tests which generates the Final Test Report packed with the new unit.
- 8.2** **Power and Relay.** Set all front panel controls as follows: small-knob controls at detent, program selector at Tuner, Volume control at maximum gain, pushbutton switches disengaged (window black). Set voltmeter to 10 Vac, oscilloscope to 1 mS/cm sweep, dc coupled, test oscillator off. Connect cable from inverse-RIAA amplifier (see Appendix 13.B) to Phono 1, oscillator output to Tuner, voltmeter and oscilloscope to Main Out 1. Connect a pair of headphones to Phones output. Connect power cord to ac outlet and push Power switch to on. Count the interval until the mute relay pulls in; it should be between 4 and 6 seconds, and it should *follow* the turn-on noise of the phono input stage heard in the headphones. Push and release the mute button a few times and observe clean operation of the mute relay.
- 8.3** **Clipping Level.** Turn on test oscillator, setting frequency to 1 kHz and level to about 2 Vac. Raise level until you observe clipping on the oscilloscope; voltmeter should read greater than 7 V rms (typ. 8 V). Change channels and observe same clipping level. Switch to Tuner input and repeat test, observing same clipping levels.

- 8.4** **Inputs and Outputs.** Switch oscillator to 0.2 V output; voltmeter to 1 Vac range. Check the functioning of the Aux 1 and Aux 2 inputs in both channels. Use three stereo patch cables to connect outputs to inputs for Tape 1, Tape 2, and External Processor (5.1 kohm in parallel with 680 pF may be added across each of these to include the driving capability of the output buffers in the following test). Check operation of the external loops by engaging Tape 1, then Dub 2 to 1; disengage Dub 2 to 1, disengage Tape 1, engage Tape 2, engage Dub 1 to 2, disengage Dub 1 to 2, disengage Tape 2, engage and disengage External Processor. Note that a latch-up will occur in this test if both Dub buttons and either Tape button are depressed simultaneously (because of a closed feedback loop this condition creates). Change channels and repeat. Observe that the signal remains present at all times and that there is no 'bounce' on the oscilloscope, i. e., no dc present at inputs or outputs. Headphones should reveal no roughness or intermittent contacts in the switches; each should function cleanly, with no evidence of other dc or ac voltages appearing during the travel of the switch armature, and the signal must drop out very briefly at the middle of the armature travel.

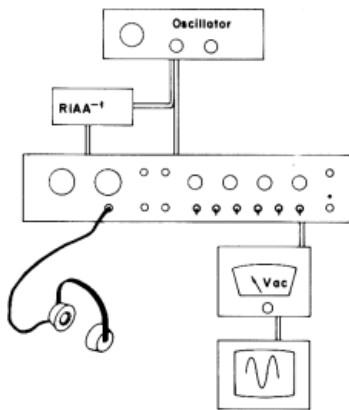


Fig. 8-1. Basic Test Set-up.

Test for board integrity and microphonics by inserting the 1 kohm shorting plugs into Phono 1, R and L and listening to phono noise through the headphones. Set selector to Phono 1, Volume to maximum gain. Check that all three external loops are engaged. Using a rubber mallet, tap the switchboard in various places, then the jack board, and finally the main board. The switch and jack boards should exhibit only very low level microphonic effects; the phono section of the main board will, of course, be more sensitive to tapping, particularly near the 0.47 μ F mylar input capacitors. Nevertheless, you should hear no clicks, sharp snaps, or intermittencies while tapping. Such sounds point to faulty connections, which must be located and repaired.

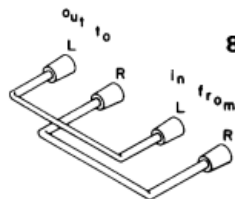


Fig. 8-2. External Loop Patching.

- 8.5** **High Level Frequency Response.** Calibrate oscillator and ac voltmeter according to Section 3.3.1. Adjust Volume control to give about 0 dBV reading on 1 Vac reading of voltmeter and trim for exactly 0.0 dBV with oscillator output trimmer. Sweep left channel from 20 Hz to 20 kHz, recording readings at 20 Hz, 25 Hz, 30 Hz, 15 kHz, and 20 kHz. Change channels, trim to exactly 0.0 dB at 1 kHz, noting less than 0.7 dB interchannel balance error. Repeat response test. Switch oscillator to 10 Hz; observe about -12 dB on both channels. Disengage Infrasonic Filter (out position) on rear panel and observe about -1 dB relative to 1 kHz on both channels. If a response error ($>\pm 0.5$ dB) is observed, return unit to the factory so that frequency response can be measured and restored to published specifications.

8.6 RIAA Equalization Accuracy. Set oscillator to 1 kHz. Change selector, inputs and outputs to Phono 1, left. Observe that the level rises 0.5 ± 1 dB (assuming the inverse RIAA amplifier to have a gain of -36 dB). Retrim the oscillator output for exactly 0.0 dB reading. Sweep from 20 Hz to 20 kHz, recording levels at 30 Hz, 70 Hz, 300 Hz, 3 kHz, 10 kHz, and 15 kHz. Change to right channel. Reset oscillator to 1 kHz. Trim right phono gain on the 250 ohm trimmer, the one closer to rear panel at the selector switch end of the main board. Repeat response test and record readings as before. If a response error is observed ($> \pm 0.2$ dB), return the unit to the factory.

Change to Phono 2, left channel, and set level at 1 kHz to exactly 0 dB. Change to right channel and set right channel trimmer (closer to front panel) for 0 dB.

Note that this is a critical test and that only an extremely accurate, calibrated voltmeter, such as the Radford ANM 2, will give useful results.

8.7 Tone and Volume Controls. Connect oscillator output to Tuner L input. Run Volume control through its range, watching for irregularities on meter and oscilloscope. Amplitude should increase smoothly, without jumps, from one detent to the next. Set oscillator to about 0.2 V output, 1 kHz; voltmeter to 0.2 or 0.5 Vac range. Rotate Volume control to obtain about -20 dBV reading, then trim oscillator output to obtain exactly -20 dB. Switch oscillator frequency to 100 Hz. Rotate Bass control fully clockwise, then fully counter-clockwise, and observe the range of the control to be ± 7.5 dB ± 1 dB. Switch Bass Mode switch up ("shelving position") and observe that the range is now ± 11.5 dB ± 1 dB. Return Bass control to the detent. Change oscillator frequency to 10 kHz. Rotate Treble control through its full travel and observe its range to be ± 8 dB ± 1 dB. Leave it at full boost ($+ 8$ dB) and switch in High Filter. Observe a drop in level of 5 dB ± 1 dB. Engage Tone Defeat. Observe a further drop in level to the reference point of -20 dBV. Disengage Tone Defeat and High Filter; return Treble control to detent. Switch oscillator to 1 kHz; voltmeter to 1 Vac range. Rotate Volume control to obtain about 0 dBV. Make sure all toggle switches are in the down position and all small knobs at their detent positions. Engage the L to R switch and rotate Mode control to the L - R position. Observe a drop in level of more than 30 (typ. 40) dB. Return Mode control to detent. Check action of Balance control for left channel. Change oscillator cable to right channel. Disengage L to R switch and engage R to L. Observe that the signal reappears. Repeat all other preceding tests for the right channel.

8.8 Phono Impedance. Connect the special impedance test cable with its 10 Mohm series resistor (see Section 13.C) at the Phono input end of the cable. Set oscillator output to about 5 V at 20 kHz. Set the Load: Capacitive switch on the rear panel to 50 pF. Apply the 20 kHz signal to Phono 1, L, and adjust the Volume control to give about -20 dBV on the 0.2 or 0.5 Vac range of the voltmeter. Trim oscillator for exactly -20 dBV. Actuate the Load: Resistive switch on the rear panel to obtain 100 kohm input resistance. Observe a rise in signal level of 5 dB ± 0.5 dB. Return to the 47 kohm (flush) position. Turn Volume control up to give -10 dBV (0 dB scale reading). Rotate the Load: Capacitive switch through its travel, observing that the signal falls to the levels below:

100 pF -11 dBV (drop of 1 dB)

200 pF -13.5 dBV

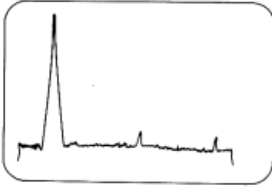
300 pF -16 dBV

400 pF -18 dBV

All levels $\pm 10\%$

Repeat for right channel. Change to Phono 2 and observe that the values above are the same. (If you are testing the unit without its case, this test will be particularly susceptible to hum because of the high source impedance. Try applying a finger to the ground lug and cupping your hand around the circuit board under the selector switch to provide electrostatic shielding. If there is still a considerable amount of hum visible in the oscilloscope trace, set the Mode control to the "L-R" position. This will cause cancellation of the hum common to both channels, leaving only the single channel, 20 kHz signal. Since the L-R signal is 6 dB lower than L or R alone, you will need to compensate by raising the Volume control gain.)

8.9 Distortion. Reset Mode control to Stereo. Check that all small knob controls are in detent position and all toggle switches are down. Use stereo patch cables (preferably



f_1 f_2 f_3

Fig. 8-3. 16 kHz Test.

8.9.1

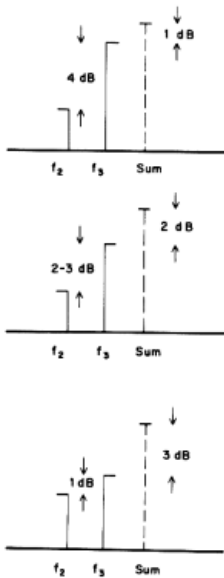


Fig. 8-4. Rms Sums.

8.9.3

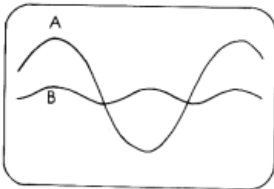


Fig. 8-5. Distortion Input (A) and Output (B)

8.10

8.10.1

with parallel 5.1 kohm resistors to load the outputs) to connect each Tape and External Processor output to the corresponding input. Set selector to Phono 1; connect oscillator directly to Phono 1, L. Engage Tape 1, Tape 2, and External Processor switches (so that the signal path is now through all of the external loops).

Measurement with Spectrum Analyzer. Set oscillator to produce 20 mV at 1 kHz. Connect voltmeter to Main Out 2, L, and spectrum analyzer to Main Out 1, L. Adjust Volume control to give about 2 V of output.

Set the gain of the spectrum analyzer to produce a peak of 0 dB for the injected 1 kHz signal, with amplitude mode set to log display, 10 dB/division. Adjust to perform a sweep from 0 to 10 kHz with 30 Hz resolution. You should observe a noise floor at around -90 dB and a small peak at the second harmonic of the input signal, that is, at 2 kHz, typically at a level of -88 dB. Record the levels of all observable harmonics. Change the generator frequency to 16 kHz and raise the *oscillator* output about 17 dB to compensate for the RIAA rolloff at the higher frequency. Set the spectrum analyzer to sweep from 10 kHz to 50 kHz with 100 Hz resolution. The voltmeter should show 2 V of output; the noise floor should remain unchanged; there should be a peak of 0 dB at 16 kHz, and you should observe two smaller peaks at the second and third harmonics, i. e., at 32 kHz and at 48 kHz. The level of these peaks is typically between -82 and -87 dB. To determine the total harmonic distortion (THD), add the measured distortion components to obtain their rms sum. In most situations the following rule of thumb will suffice: if other distortion components are more than 5 dB below the highest component, take the highest value as the sum. If there is a component 4 dB or less below the highest value, *add 1 dB* to the highest value when the difference is between 4 and 3 dB, *add 2 dB* when it is 3 to 2 dB, and *add 3 dB* when it is 1 dB or less. Noise or spurious disturbances should not, of course, be added to this sum.

(This rough sum will give a good approximation to IHF Standard A-202, Section 1.15. There are some grounds for believing that a weighting function should be applied before this sum to take auditory masking effects into account. A weighting function may be introduced into this measurement at some future date, as provided in Section 1.16 of IHF A-202.)

If the result is greater than -80 dB, the cause of the distortion should be found and repaired. Typical distortion figures are from -82 to -87 dB.

Measurement with a Distortion Meter. If possible, check the operation of your distortion meter by watching the distortion output on an oscilloscope to see if hum or other spurious signals may be entering the signal path before or after the unit under test. You should be able to correlate the distortion products and the injected signal on a dual trace oscilloscope by looking at the injected signal in one channel and the distortion output in the other. (Note that many dual-trace oscilloscopes will trigger only from the A channel, so you may want to feed the oscillator to the A input.)

Noise. Place the top cover on without screwing it down. Set oscilloscope triggering to "line."

Volume Down Output Noise. Turn Volume control down to zero gain and measure the A-weighted noise at Main Out 1 (if necessary, using the noise meter preamplifier described in Section 13.A). Observe the output from your meter amplifier at 5 msec/div sweep; you should see no hum or hum-related spurious noises on the oscilloscope, nor any of the irregular signals characteristic of RFI. Repeat for other channel.

- 8.10.2** Volume Up Output Noise. Connect the 1 kohm shorting plugs (see Section 13.C) to Tuner R and L. Turn Volume up to maximum gain. Observe the noise level as less than -80 dB. Disable Tape 1, Tape 2, and External Processor one by one, noting that the noise level falls to less than -88 dB (-90 typ.). (The buffers to the external loops typically contribute 2 dB of noise each.) Repeat for other channel.
- 8.10.3** Phono Noise. Connect the cartridge body plug (see Section 13.C) to Phono 1, L. Set selector to Phono 1 and observe voltmeter output on oscilloscope. You will need to find an orientation where hum is nearly absent from the signal observed through the A-weighting filter; this may require turning off transformer-operated equipment positioned near the preamp, or moving the preamp and cartridge to a different part of the test bench. The oscilloscope trace should appear similar to that observed in the earlier noise tests, without spikes or RFI, and with a 60 Hz component considerably smaller in amplitude than the average amplitude of the noise components. (You will be lucky indeed if your cartridge does not pick up *any* noticeable hum.) Subtract 7 dB from the observed full-volume reading and record this number as noise relative to 5 mVrms. (The IHF standard defines phono noise as relative to a 5 mV input to the phono section; subtracting 7 dB from the A-weighted reading in dBV produces an equivalent number.) Repeat for right channel.
- (As in the case of distortion, the A-weighted noise measurement is psychologically not entirely accurate and may be revised at a future date.)
- 8.11** **IHF Standard.** A copy of the current IHF Standard A-202 for specifications and measurements of high fidelity equipment may be obtained for (US) \$7.00 from the Institute of High Fidelity, 489 Fifth Avenue, New York, NY 10017.

9 Options

9.1 Special Termination Impedance. A 10-pin connector for providing special termination impedances is available from the factory (order dummy plug assembly, no. 917-200-01). Upon request, and when feasible, Apt will supply a connector wired for a desired, non-standard input impedance or attenuation. In most cases it will be possible to obtain a desired input impedance by following the instructions below:

1. Find the tinned wire jumpers numbered 1 to 4 between the Phono 2 jack and the 10-connector jack on the main circuit board. (Note that the number 4 is partly hidden by the connector.) Using a small wire-cutter, break each of the wires near its number; then break it again near the other end so that the remaining wire will not make contact anywhere else. Make sure that you have removed the pieces of cut wire from the unit.
2. *Before* installing the dummy plug assembly, add the desired resistances and capacitances, R and C, as follows:

<i>Desired Resistive Load</i>	<i>Use as R</i>
100 ohms	100 ohms
10 kohms	12 kohms
22 kohms	39 kohms
33 kohms	120 kohms

For a desired resistive load above 47 kohms, remove the two 91 kohm resistors, R2 and R102, with wire cutters. These resistors will be found next to the cut jumper wires. Add resistors, R, as follows:

68 kohms	220 kohms
82 kohms	470 kohms

If you desire a capacitive load component greater than 50 pF, add mica capacitors, C, using a value 50 pF lower than the desired capacitive load.

If you desire other impedances or input attenuation, consult the factory.

9.2 Pre-preamplifier. Find the tinned wire jumpers numbered 1 to 4 and cut them according to the directions in Section 9.1 above. Insert the pre-preamp into the 10-pin connector. For further instructions, consult the literature accompanying the model being installed.

9.3 Output Attenuators. When using a power amplifier with unusually high input sensitivity (and no gain control), or when using high efficiency loudspeakers, it may be desirable to reduce the output level of the preamplifier, so that the full range of the Volume control may be exploited. Estimate the desired reduction in preamp output as follows:

Set the Volume control to give the desired, medium-loud output level in the system in which the unit will be used, using the most typical program source (e. g., a phonograph input). Turn the program material off, and advance the Volume control to the 12 o'clock position, counting the number of steps. Multiply this number (the number of steps you have advanced the control) by 2 to obtain the desired attenuation in dB. (The Volume control steps are not exactly equal. They increase in size at the extremes of the control in order to encompass as wide a dynamic range as possible; 2 dB per step represents an average value over the range from the 9 o'clock position to the 12 o'clock position.)

Near the Main Out 2 jack find two 47 kohm resistors (not numbered, next to two tinned wire jumpers) above the words "optional pad." Replace each 47 kohm resistor

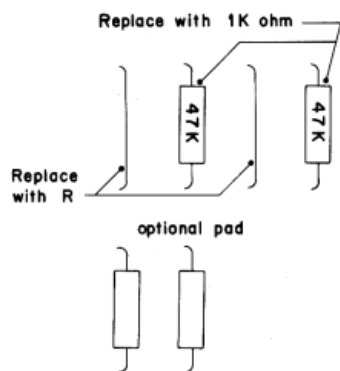


Fig.9-1. Optional Pad.

with 1 kohm. Replace each wire jumper with a resistor of the value R as given by the formula below, where ΔdB is a negative number giving the desired attenuation, 42

$$R = \frac{1000}{\log^{-1}(\Delta dB/20)} - 1330$$

or interpolate from the following table:

ΔdB	R
-6	560 ohms
-10	1.8 kohms
-15	4.3 kohms
-20	8.6 kohms

The attenuated output will now be available from Main Out 2. If for some reason both outputs are to be attenuated, leave the wire jumpers in place and replace R83 and R183, below the words "optional pad," with R + 330 ohms instead. Insert the 1 kohm resistors as before.

9.4 240 Vac Connection.

Remove the top cover and locate the red and black wires from the power transformer which connect to the terminal strip and fuse. Disconnect these from the terminal strip and then solder them together at the remaining, unused terminal strip lug. Check for 50 mA drain on variac.

9.5 Rack-Mount Front Panel Installation.

The optional rack-mount front panel kit consists of a replacement front panel with exterior dimensions to 19 in. American relay rack standards, handles, hardware, and tools.

Installation should be straightforward; however, specific cases may require some extra work on the part of the installer. These will be detailed later.

The original panel is held on by six 4-40 flat head socket screws, and by six 7/16 in. O. D. threaded nuts on the control shafts. The replacement front panel duplicates the dimensions of the preamplifier for hole layout, but its outside dimensions are expanded to 19 x 3-15/32 in. (usually called 3½ in.). The mounting hole punching matches standard American (EIA) dimensions, but not all racks are built to this standard, so you may encounter some difficulty with mounting in foreign-made racks.

9.5.1 Removal of Original Front Panel.

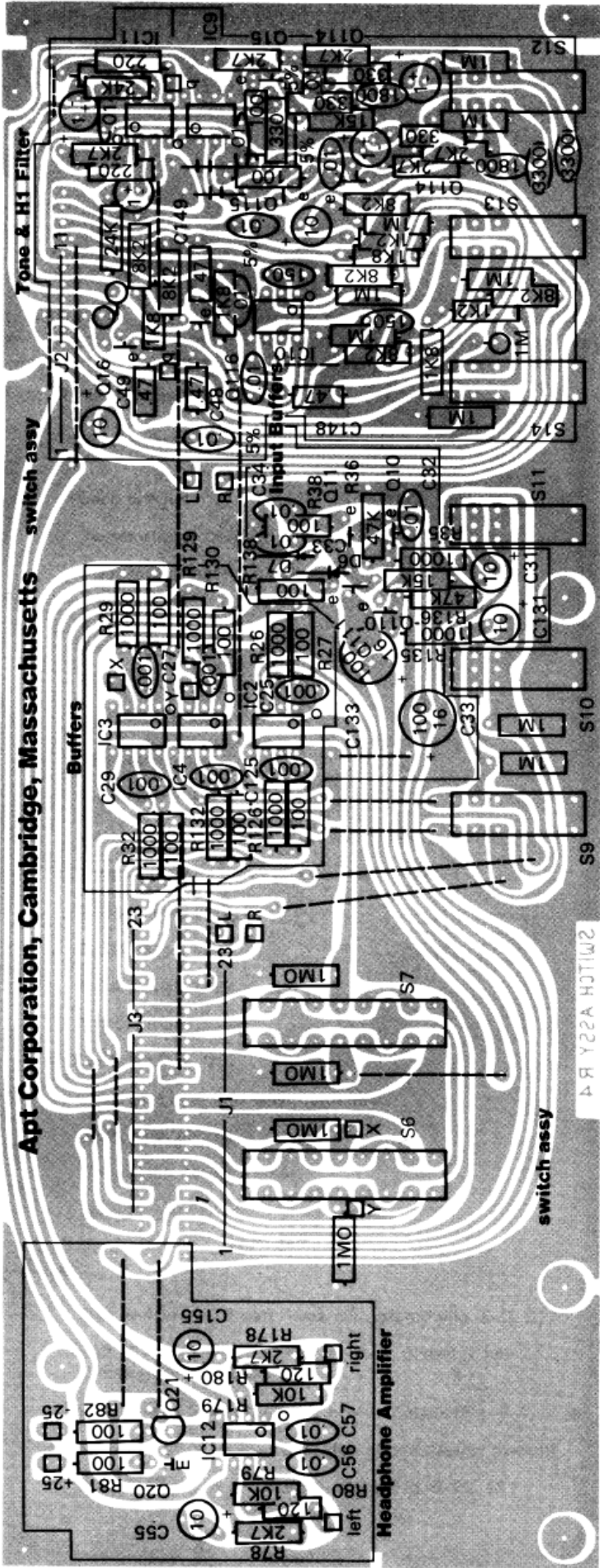
1. Remove all the knobs by loosening the retaining set screws with the 2 mm hex key supplied. A gentle twisting and pulling motion should suffice to get the knobs off their shafts. Do not force a knob off, as you could damage the controls. If you have difficulty removing the knobs, please consult the factory.
2. Remove the retaining nuts and their washers with a 7/16 in. socket or offset box wrench (not supplied). Keep the nut from the Volume control separate, since it has a larger inside diameter than the others.
3. Remove the front panel hex socket screws with the 1/16 in. hex key supplied.
4. Lift off the original front panel.

9.5.2 Installation of Rack-Mount Panel.

1. In most cases installation is simply the reverse procedure from removal. Fit the rack panel over the control shafts and toggles, seating it gently on the toggle switch bushings. If the bushings on the toggle switches do not line up properly with their respective holes, gently maneuver them, using the switch levers, until all of the switches are seated.
2. Start the lower center, 4-40 flat head socket screw into its threads—do not tighten at this time.

3. Replace the control washers and nuts, making certain that the proper nut and washer are returned to the Volume control; only finger tighten at this point.
4. Insert loosely the five other 4-40 flat head socket screws in the following order: top center, top left, top right, bottom left, bottom right. All six of the screws may now be fully tightened. (If the no. 4 holes in the front panel do not line up, it may be necessary to remove the bottom plate in order to bring the top cover into alignment with the front panel.)
5. Tighten the control knob nuts 1/4 turn beyond finger tight. Set the controls as follows: Volume control fully counterclockwise; Balance, Mode, Bass, and Treble at center detents. Mount the knobs with their indicator dots in proper position: 6 o'clock for the Volume control and 12 o'clock for all others. Make certain that the Volume control knob stands far enough off the front panel that it will not scrape the finish. Check the Volume control throughout its range for binding. If the control is tight over part of its range, remove the knob and back off the hex nut slightly. Remount the knob and check it again.
6. Align each of the aluminum backing plates behind the corresponding holes in the front panel. Thread one 6-32 x 1/2 in. screw through a no. 6 internal tooth lockwasher, through the backing plate and the front panel and into one end of the handle. Take care not to allow the handle to swing and mar the finish. (If you do not need the handles, they may be installed in the reverse direction, facing the back.)
7. Use the 10-32 x 1/2 in. machine screws and corresponding fibre washers to mount the preamplifier in the 19 in. rack.
8. Since the rack installation adds an additional ground connection to the preamplifier, ground loops may be formed in the system, injecting hum. If this happens, consult the *Owner's Manual*, p. 35, "Dealing with Hum and Radio Frequency Interference."

- 9.5.3** Parts List: Front panel, 19 x 15/32 in. (1; part no. 220-005-01, gray; or 220-005-02, black). Backing plates, 3-1/2 x 1-1/2 in. (2; part no. 210-004-00). Rack handles (2; part no. 435-001-00). Machine screws, 6-32 x 1/2 in. (4; part no. 400-161-02). #6 internal tooth lockwashers (4; part no. 400-164-02). Rack screws, 10-32 x 1/2 in. (4; part no. 400-201-01). Fibre washers for rack screws (4; part no. 400-203-01). Flat head socket screws, 4-40 x 1/4 (2, spare; part no. 400-141-05). Allen wrench, 1/16 in. (1; part no. 090-001-00). Allen wrench, 2 mm (1; part no. 090-002-00). Knobs for black kit only (2 large; part no. 432-001-02; 4 small; part no. 432-002-02)



Tone & HI Filter

switch assy

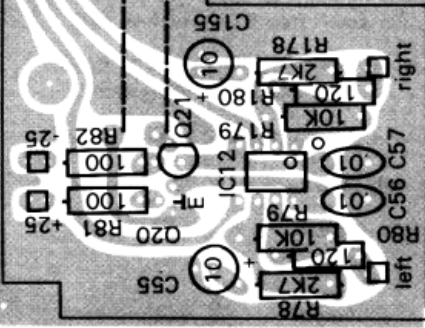
Apt Corporation, Cambridge, Massachusetts

Buffers

Input Buffers

switch assy

Headphone Amplifier



P R Y22A H01102

12 Parts List

12.1 Ordering Parts. For out-of-warranty replacement of parts, send exact description and the part number from the list below to Technical Services, Apt Corporation, Box 512, Cambridge, MA 02139 or phone 617 492-5145.

12.2 Sheet Metal, Mechanical Parts.

<i>Part Number</i>	<i>Description, Location</i>
210-001-00	Jack bracket
210-002-00	Power supply bracket
940-003-00	Sub front panel assembly
220-001-01	Front panel, gray
220-001-02	Front panel, black
220-002-01	Rear panel, gray
220-002-02	Rear panel, black
220-003-01	Top cover, gray
220-003-02	Top cover black
914-004-00	Bottom panel assembly, gray (with feet)
914-004-01	Bottom panel assembly, black (with feet)
432-001-01	Knob, large, gray
432-001-02	Knob, large, black
432-002-01	Knob, small, gray
432-002-02	Knob, small, black
511-001-00	Jack, ac line (US)
511-002-00	Jack, single phono with hardware
511-004-00	Jack, dual phono, p.c.
511-006-00	Connector, 10-pin, male
917-200-01	Dummy plug ass'y (Phono 2 input)

12.3 Hardware.

400-141-01	Bolt, 4-40 × 3/16 pan head, slot, plated (Fig. 6-3, S)
400-141-02	Bolt, 4-40 × 1/4, pan head, phillips, plated (Gold transformer, 620-001-00)
400-141-03	Bolt, 4-40 × 3/8, pan head, phillips, black (Fig. 6-1, J, M)
400-141-04	Bolt, 4-40 × 3/8 pan head, slot, plated (power switch)
400-141-05	Bolt, 4-40 × 1/4 flat, socket, black (Fig. 6-22, C)

400-142-01	Nut, 4-40, plated (regulator IC)
400-144-01	Lockwasher, #4, internal tooth, plated (regulator IC)
400-144-02	Lockwasher, #4, internal tooth, black (Fig. 6-1, J, M)
400-161-01	Bolt, 6-32, pan head, phillips, plated (Ground post)
400-162-01	Nut, 6-32, plated (Ground post)
400-162-02	Nut, 6-32, knurled, brass (Ground post)
400-164-01	Lockwasher, #6, internal tooth, plated (Ground post)
400-181-01	Bolt, 8-32 × 1/4, phillips, plated (Black transformer, no. 620-002-00)
400-322-01	Nut, 3/8, plated (Phones jack)
400-323-01	Flatwasher, 3/8, plated (Phones)
400-324-01	Lockwasher, 3/8, internal tooth, plated (Load: Capacitive switch)
400-632-01	Nut, 8 mm (controls)
400-633-01	Flatwasher, 8mm (controls)
400-642-01	Nut, 9 mm (Volume control)
400-643-01	Flatwasher 9 mm (Volume control)
410-161-01	Sheet mtl screw, #6, pan head, phillips, black (Fig. 6-1, C)

12.4

Semiconductors.

722-001-00	Diode, 1N4148
723-001-00	Diode, power, 1N4002
724-001-00	Diode bridge, EDI-PF-10
731-001-00	Transistor, 2SC1345E
731-002-00	Transistor, 2N3904
732-001-00	Transistor, 2N5087
732-002-00	Transistor, 2N3906
751-001-00	IC, TL072CP
751-002-00	IC, RC4558CP
752-001-00	Regulator, uA7818CKC
752-002-00	Regulator, uA7918CKC
910-002-00	Transistor, 2N5484, noise and voltage tested
930-001-00	L.E.D. assembly

12.5

Passive Components.

554-001-00	Potentiometer, dual, 10 kohm with tap
554-002-00	Potentiometer, dual, step, 10 kohm (Volume)

555-125-00	Trimpot, 250 ohm, vertical (Phono)
555-410-00	Trimpot, 100 kohm, flat (Tone trimmer)
620-001-00	Transformer, power, (before A3400) gold
620-002-00	Transformer, power, (after A3400) black
640-001-00	Fuse, 1/8 amp, slo blo, pigtail (120 V)
640-002-00	Fuse, 1/16 amp, slo blo, pigtail (240V)
673-317-01	Resistor, 17.4 kohm, 1%
673-423-01	Resistor, 232 kohm, 1%
674-182-01	Resistor, 1/2 watt, 820 ohm, 5%
674-215-01	Resistor, 1/2 watt, 1.5 kohm, 5%
<i>Resistors</i>	<i>All other resistors may be ordered by specifying the following:</i> R number from schematic (e. g. R91) Value Axial (flat lying) or p. c. (standup) Tolerance (usually 5%)
684-243-02	Cap., .0043 μ F, 2.5%, polypropylene
684-315-02	Cap., .015 μ F, 2.5%, polypropylene
<i>Capacitors</i>	<i>All other capacitors may be ordered by specifying:</i> C number from schematic (e. g., C21) Value in μ F or in pF Electrolytic, mylar film, or ceramic disc Axial (flat) or p. c. (standup)

12.6

Switches, Relays, Assemblies

551-002-00	Switch, rotary, 3-gang (selector)
551-004-00	Switch, pushbutton, 2PDT
551-006-00	Switch, toggle, 2PDT, p. c. mount
913-003-00	Load: Capacitive switch assembly
913-004-00	Power switch assembly
913-005-00	Window pushbutton, 6PDT, white button
913-006-00	Window pushbutton, 6PDT, green button
712-001-00	Relay, 2PDT, p. c. mount
914-002-00	Rear panel assembly, gray, with ac cord
914-002-01	Rear panel assembly, black, with ac cord
915-001-00	Main circuit board ass'y (without power supply ass'y)
915-004-00	Main circuit board ass'y with power supply
915-002-00	Switchboard assembly
915-003-00	Jackboard assembly
917-200-01	Dummy plug assembly (Phono 2)
918-002-00	Cable, single channel phono, 10 m length
919-102-00	Shorting plugs (package of 2)

13 Appendixes

13.A

Voltmeter Preamplifier Project.

The "Volume down" noise test described in Section 8.10.1 requires reliable measurement of levels as low as -105 dBV ($6 \mu\text{V}$). In addition, psychoacoustically meaningful measurement requires a weighting function that takes into account the varying sensitivity of the ear with frequency and energy distribution. For the purpose of noise measurement, a good approximation to the ear's response is provided by the ASA A-weighting characteristic. While this curve does show some considerable deviations from the curve one would obtain by following closely the low-level sensitivity of the ear, particularly in the area around 3 kHz and at low frequencies, a closer fit would require an electrically more complex filter. And although the A-weighting function overestimates the ear's sensitivity to noise at low frequencies, it does give a useful approximation to its sensitivity to sinusoidal signals, such as power supply hum and its harmonics.

For ac voltmeters which are not capable of reading low voltages accurately, or which do not incorporate an A-weighting filter, the circuit described here will add the facilities needed for accurate noise measurements.

The circuit shown in Fig. 13-1 uses one half of a TL072 operational amplifier to raise the input signal level 38 dB. It drives the weighting filter from the required low impedance (see Howard M. Tremaine, *Audio Cyclopedia* [Indianapolis: Howard W. Sams, 1969], p. 260). The second half of the TL072 presents the high impedance required at the output

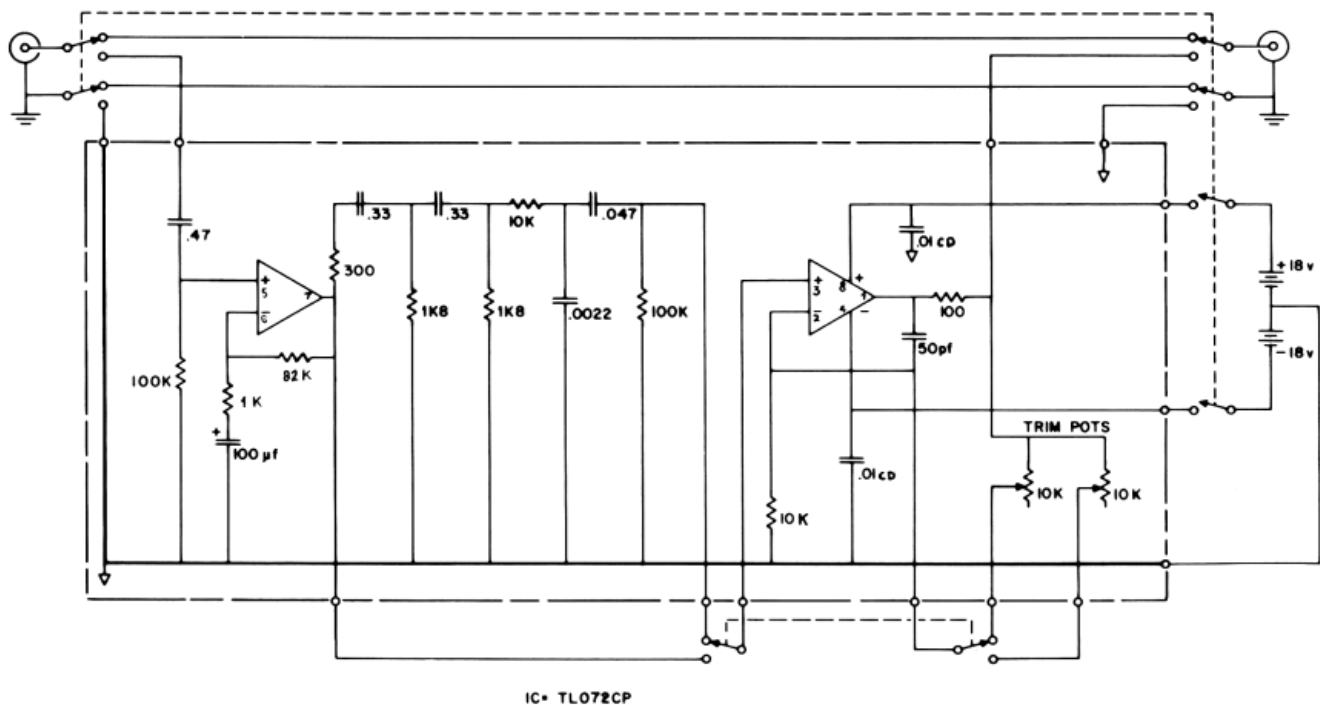


Fig. 13-1. Voltmeter Preamplifier Schematic.

of the filter and allows 0 to 6 dB of gain to overcome the filter's insertion loss. A 4PDT switch permits the input signal to be fed directly through to the meter when additional gain is not needed and at the same time disconnects the battery power supply. Because the TL072 draws only about 3 mA, the preamp can be expected to run more than 100 hours on a set of batteries, and the battery power supply ensures freedom from ground loops, which can be quite troublesome at such low levels. A second switch allows the A-weighting filter to be removed from the circuit and inserts a second feedback resistor to maintain the overall gain of 40 dB.

Construct the filter and amplifier with careful attention to separation of inputs and outputs. The .01 ceramic disc capacitors must be located as close as possible to pins 8 and 4, and must have short leads to ground. The unit must be installed in a grounded, shielded case to prevent hum.

Calibrate the "unweighted" gain by feeding a 10 mV (-40 dBV), 1 kHz signal directly to your voltmeter. Trim the oscillator level to read exactly -40 dBV on the voltmeter. Set the meter to its 1 V range, insert the meter preamp between the voltmeter and the oscillator, and adjust the preamp's "unweighted" gain trimmer to give a reading of exactly 0 dB on the voltmeter. If you are using an rms voltmeter, you may calibrate the "A-weighted" trimmer in the same way. If yours is an *averaging* meter, set the gain of the "A-weighted" trimmer to give a reading of +1.1 dBV. This will correct for the error incurred in measuring noise with an averaging rectifier (see Section 3.2.2); it will mean, of course, that sine waves measured through the A-weighting filter (not that one would have much cause to do this) will read 1.1 dB higher than their true (rms) A-weighted values.

Check the accuracy of the A-weighting filter against the graph in Fig. 13-3. Since the frequency components that lie more than 5 dB below the apogee of the curve will usually not add greatly to the measured noise, the accuracy of the filter below 400 Hz and above 13 kHz is not critical. The standard prescribes a tolerance of ± 1.5 dB from 100 to 10 kHz, ± 2 dB outside this band. (See US ANSI [ASA] S1.4-1969.)

A construction kit for the meter preamp is available from Apt Corporation.

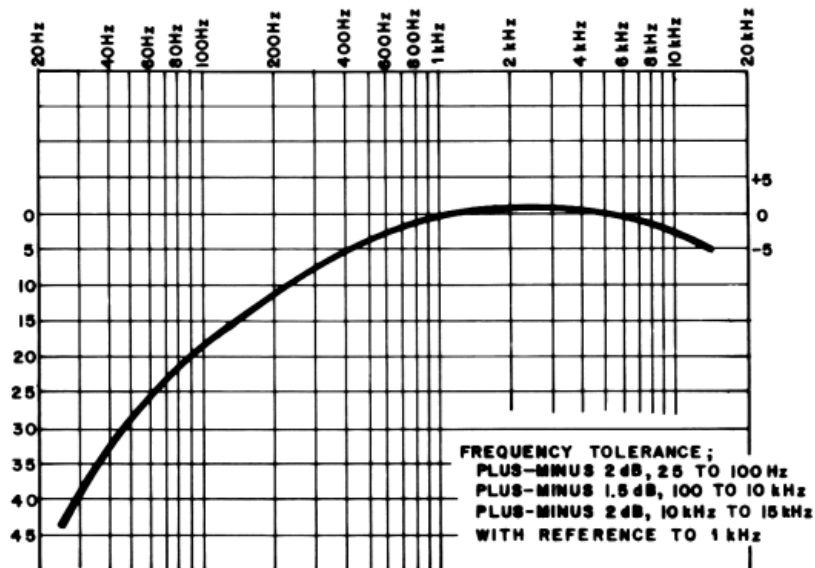


Fig.13-2. ASA A-weighting Function.

13.B Inverse RIAA Project.

For confirming RIAA response accuracy, the following circuit (adapted from a design of Stanley Lipschitz, Waterloo, Ontario) will give accuracy commensurate with that of the best service instrumentation. The circuit uses 1% parts of standard values which give an RIAA accuracy of ± 0.1 dB. (The values given in parentheses are the exact values which give a fit of better than 0.05 dB.) The 1 kHz gain of -36 dB will give nearly the same output level from the phono preamp inputs as will be obtained from a direct input to Tuner. Power has been taken from four 9 V radio batteries; however, any other source of well-regulated ± 15 V will be equally good. Add an additional output buffer if any but a normal phono input is fed from this circuit.

Accuracy may be checked with a calibrated oscillator and ac voltmeter against the table of spot frequencies below (note that the RIAA standard is defined in terms of time constants rather than amplitude vs. frequency; the table has been derived from the time constants).

A kit for the Inverse RIAA amplifier is available from Apt Corporation.

Table: Inverse RIAA Function

Frequency	Amplitude in dB	Frequency	Amplitude in dB
31.5	-18.48	2.00 kHz	+2.59
63.0	-15.85	3.15 kHz	+5.04
160	-9.81	10.0 kHz	+13.73
315	-5.18	16.0 kHz	+17.71
1.00 kHz	0.0	20.0 kHz	+19.62

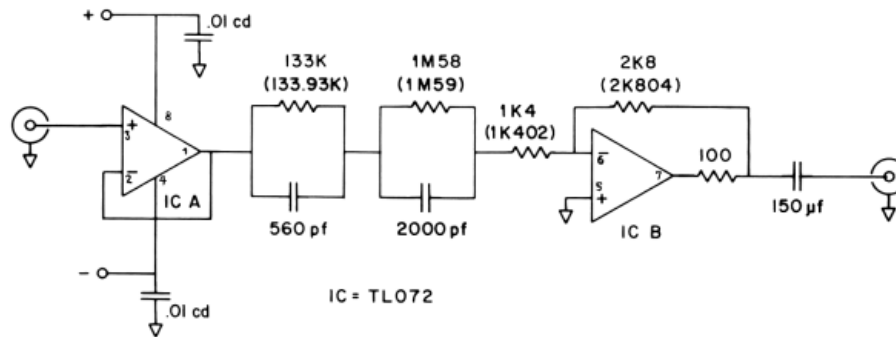


Fig. 13-3. Inverse RIAA Schematic.



Fig. 13-4. Inverse RIAA, Foil Pattern.

13.C Special Test Cables and Plugs.

1. For measurements from line level inputs, IHF Standard A-202 specifies a 1 kohm input termination. To perform measurements of noise and other parameters in compliance with this standard, solder a 1 kohm, 10 percent resistor into a phono plug (one wire to the pin, the other to the shell of the plug).

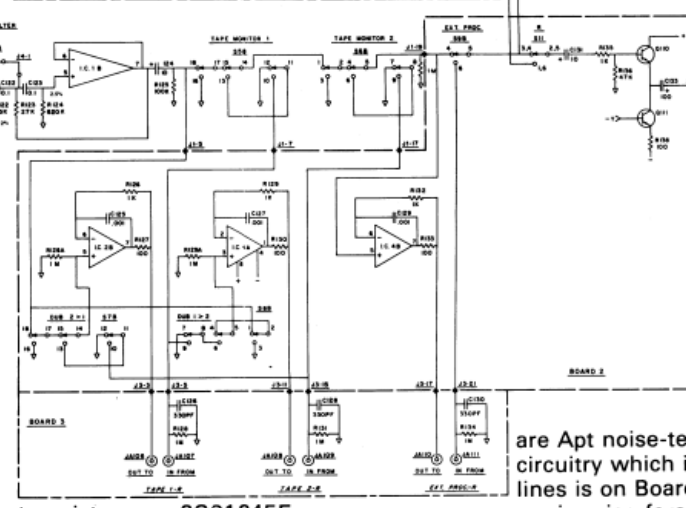
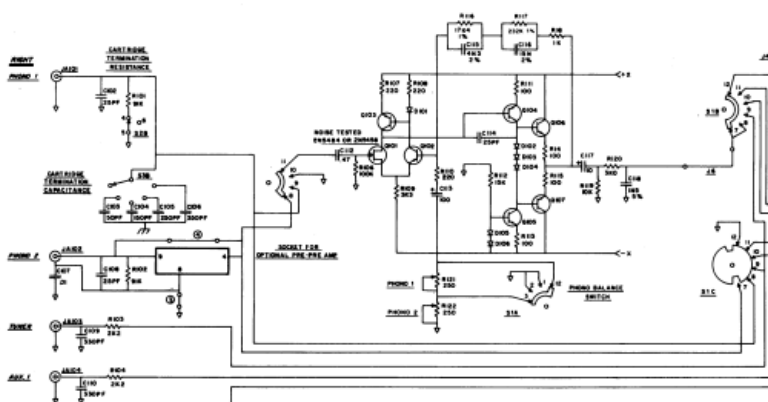
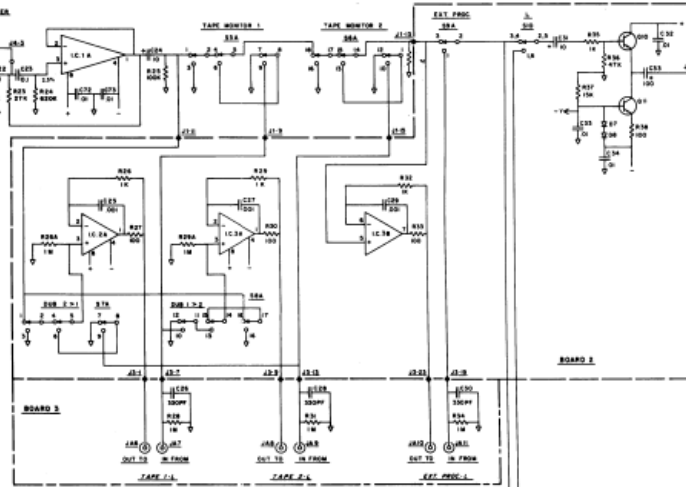
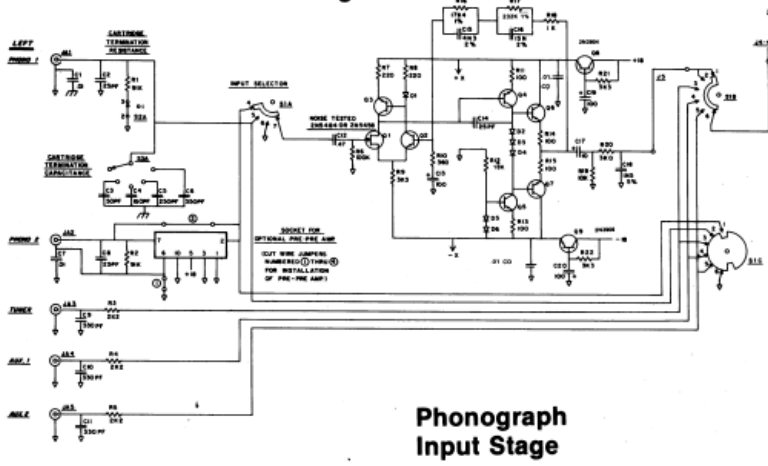
2. For measurements of moving magnet phono input noise, the IHF standard specifies a network of 500 mH in series with 1 kohm, this impedance then in parallel with 250 pF. Because the construction of an adequate magnetic shield for this network is beyond the means of most service shops or audio amateurs, Apt recommends a cartridge body meant for 47 kohm inputs, such as the Shure M-75, which presents an impedance sufficiently close to this value for all service measurements.

Solder a phono plug to one end of a shielded cable 2 in. long. Since the extremely fine wire in the cartridge can be damaged by excessive heat on the pins, prepare the cartridge for soldering as follows: clasp a pin next to the shell with a long-nose pliers; then apply a small dot of solder to the end of the pin with the soldering iron. Repeat for the other pin of the same channel. Liberally tin very short lengths of both the cable's center wire and shield. Tack solder the center wire to the hot pin, the shield to the ground pin. Make sure that the case is firmly grounded to the ground pin. Encase the pins and cable end in epoxy glue. A certain amount of flexible cable must remain between the cartridge and the plug so that during the noise test the cartridge may be moved around to obtain minimum hum.

3. Phono impedance test cable. The phono impedance test described in Section 8.8 requires a high-resistance source at the phono input. The 10 Mohm resistor must be placed as close as possible to the phono jack to avoid hum pickup and additional cable capacitance.

Bare the shield and center wire of a length of phono cable. Insert the bared end of the center wire and one end of a 10 Mohm resistor into the pin of a phono plug and solder into place. Solder the shield and the other end of the 10 Mohm to the shell. Run a little epoxy into the plug for insulation and solder a phono plug to the other end of the cable. Check continuity with an ohmmeter.

10 Schematic Diagram

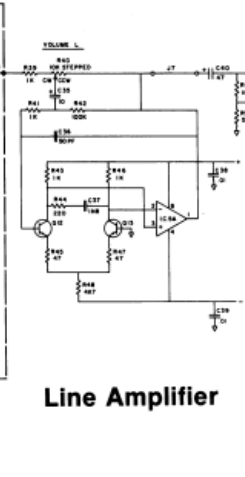


Schematic Notes: Switch S1 shown in Phono 2 position as viewed from rear. Front panel pushbutton switches are shown in *out* position; rear panel pushbuttons are shown in *in* position; toggle switches are shown in *down* position. J1, J2, J3 are multiwire board-to-board jumpers; other jumpers are point-to-point on individual boards. All *nnp*

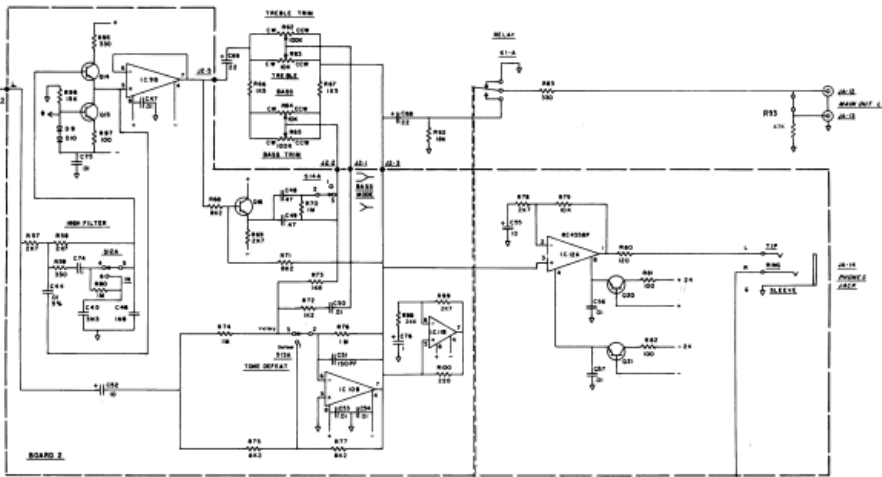
transistors are 2SC1345E or equivalent except as noted; all *pnp* transistors are 2N5087; all integrated circuits are TL072CP except as noted. Pin 1 and pin 7 of all integrated circuits are at 0 Vdc +1 Vdc. Phono preamp input FETs

are Apt noise-test circuitry which is on Board 1. All lines is on Board 1. All lines are in microfarads marked pF (picofarads); marked with "N" (1N5 = 1.5 nanofarads). For resistors, k = kilohms, M = megohms, 3.3 = 3.3 megohms.

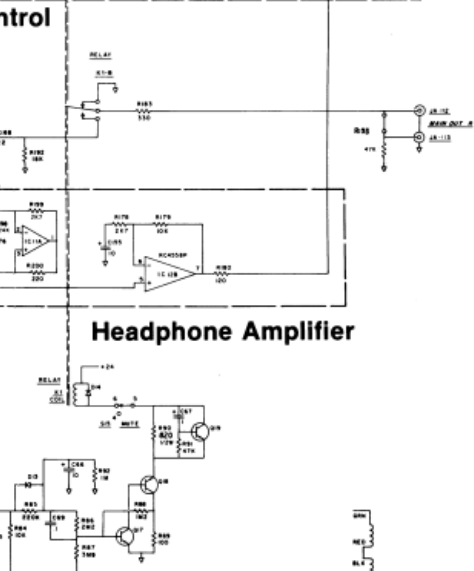
Line Amplifier



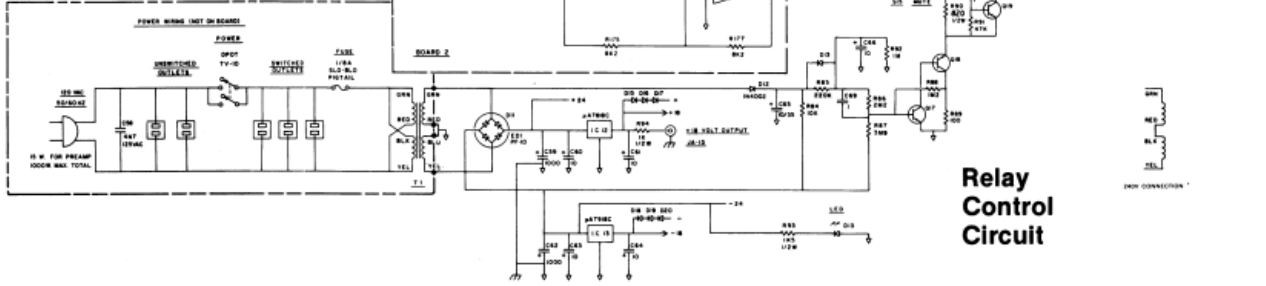
Tone Control



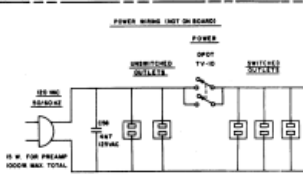
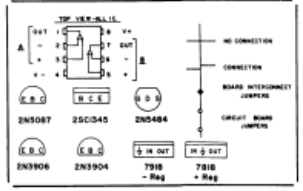
Headphone Amplifier



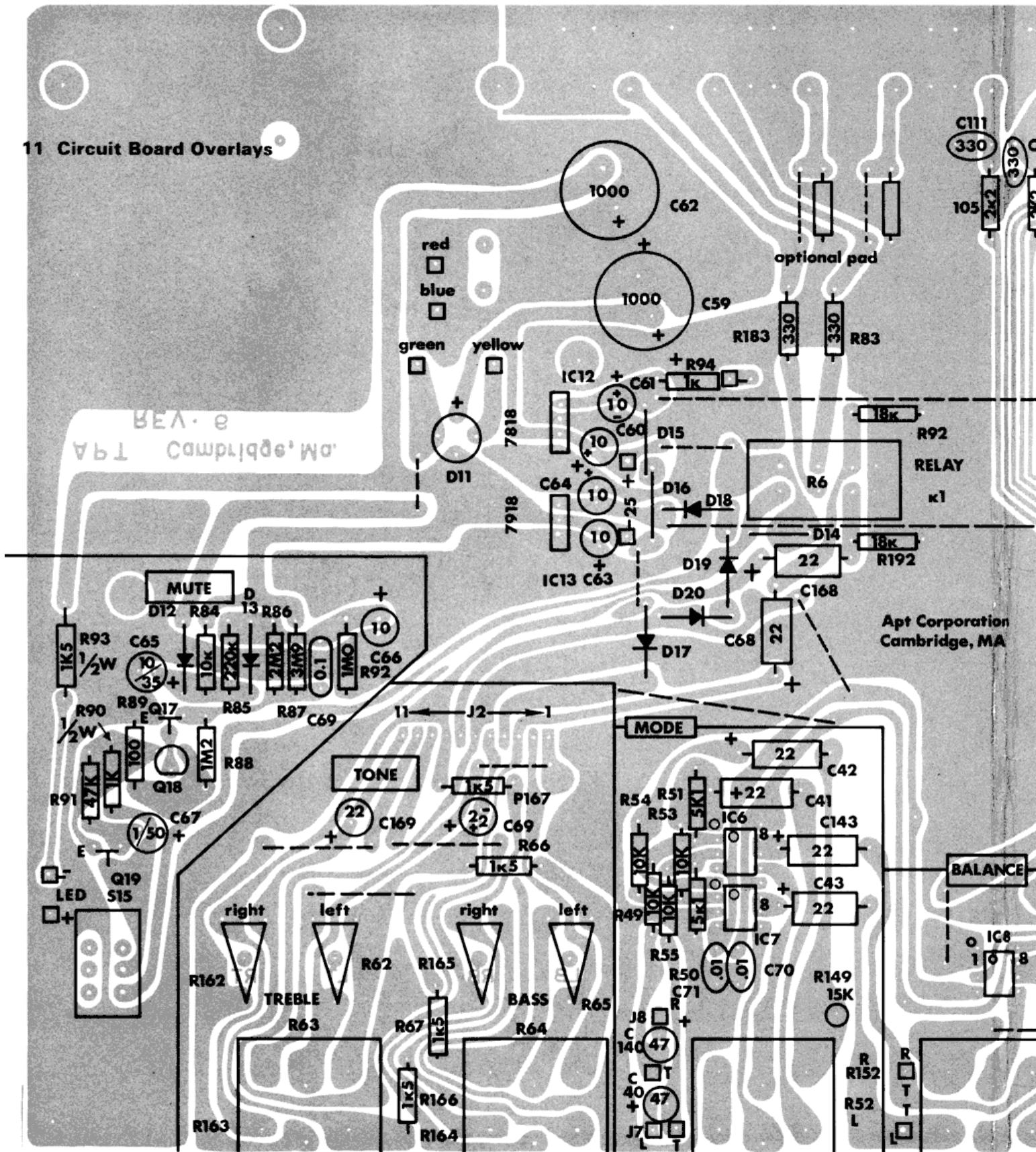
Relay Control Circuit

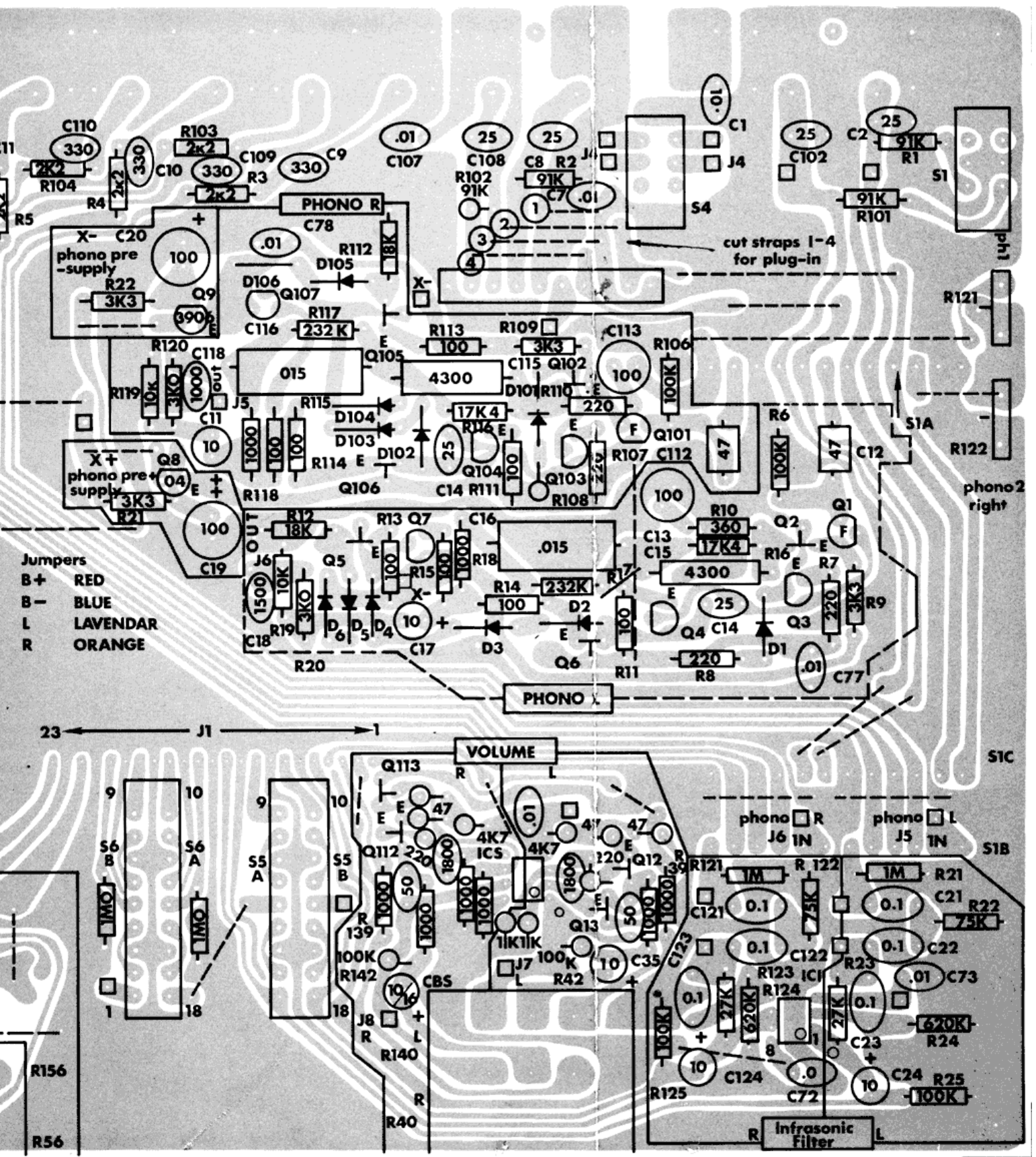


...ted types. All
... not within dotted
... 1. All capacitors
...s except those
... (arads) and those
... as a decimal point
... arads = 1500
... resistors, K =
... ohms. Thus 3M3 =



11 Circuit Board Overlays





Main Board Foil Pattern, Component Side