The Apt/Holman Preamplifier Owner's Manual

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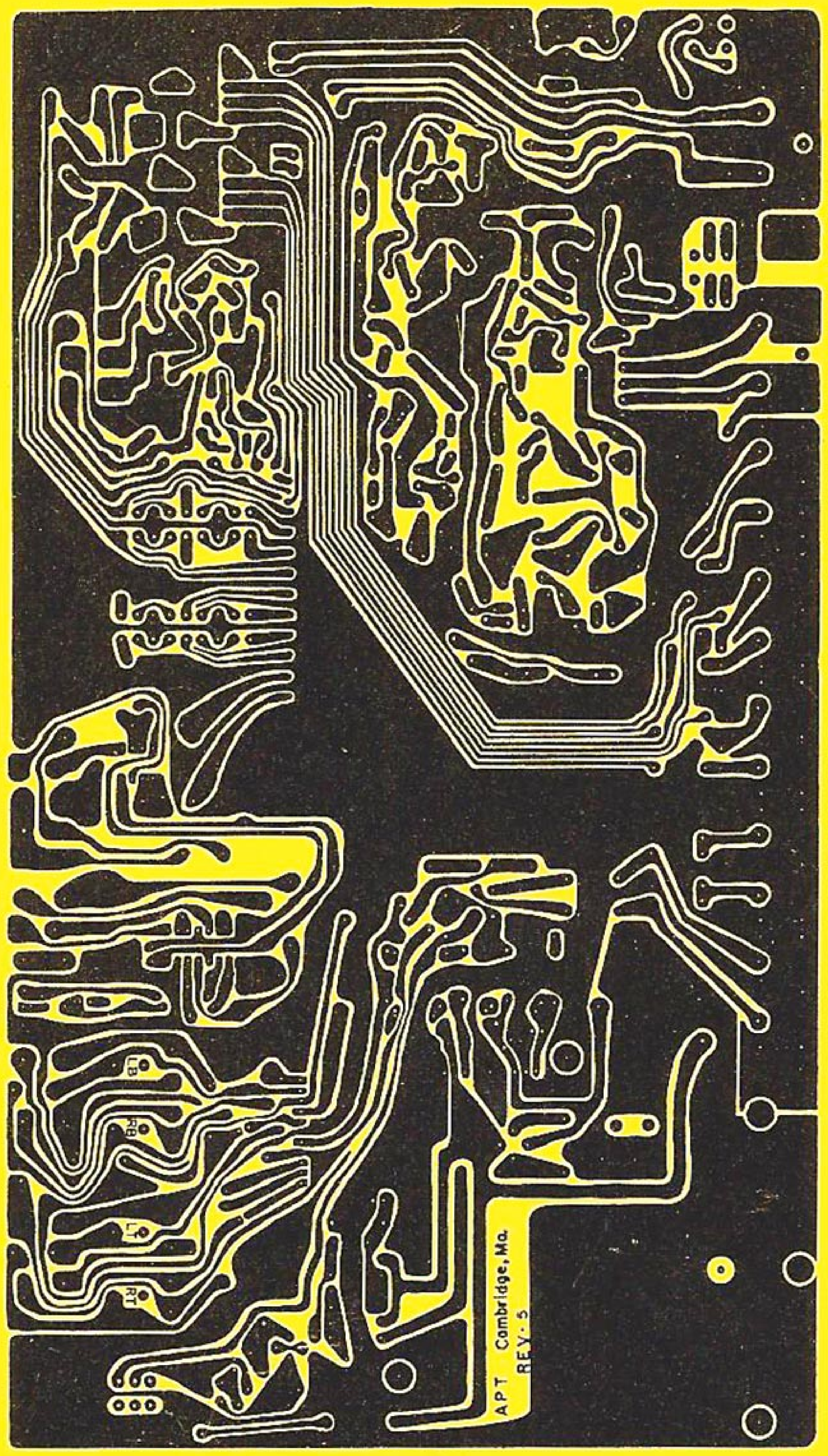

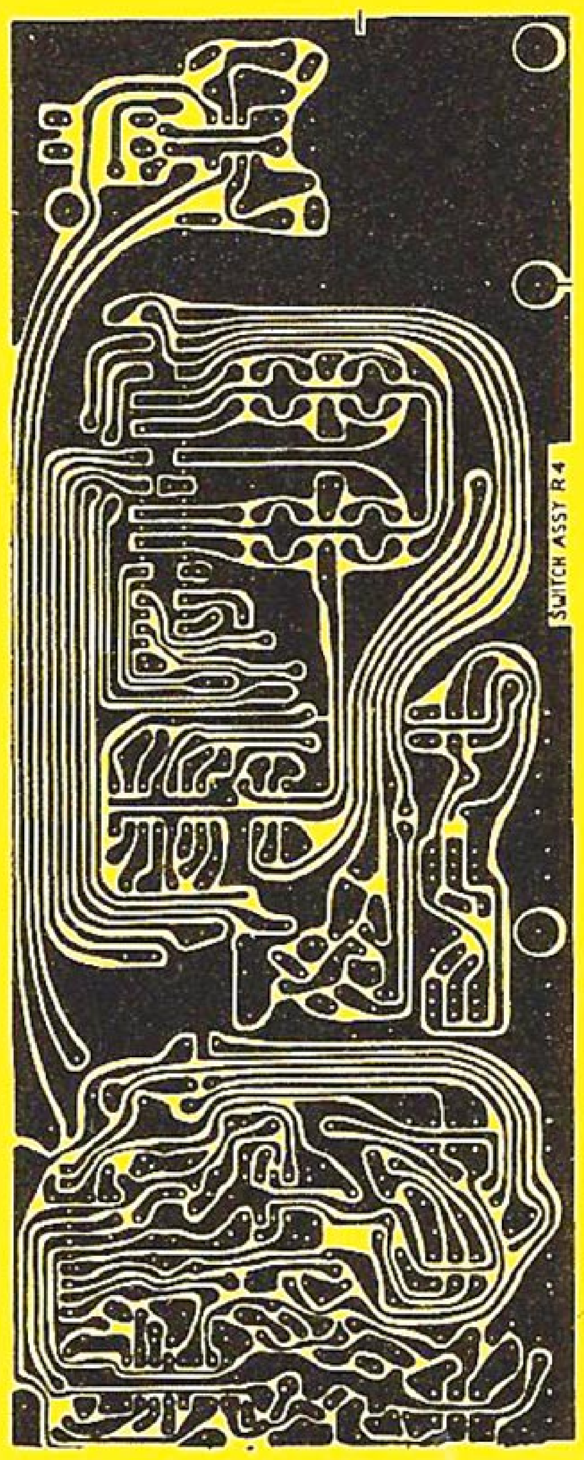

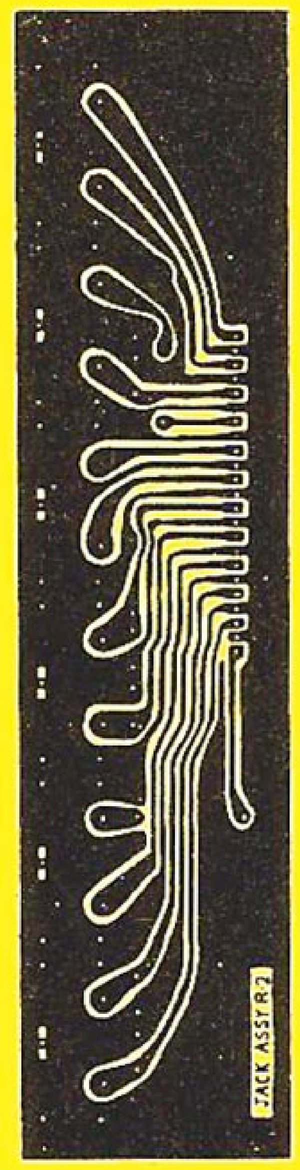

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Underwriting organizations require and common sense dictates the following:

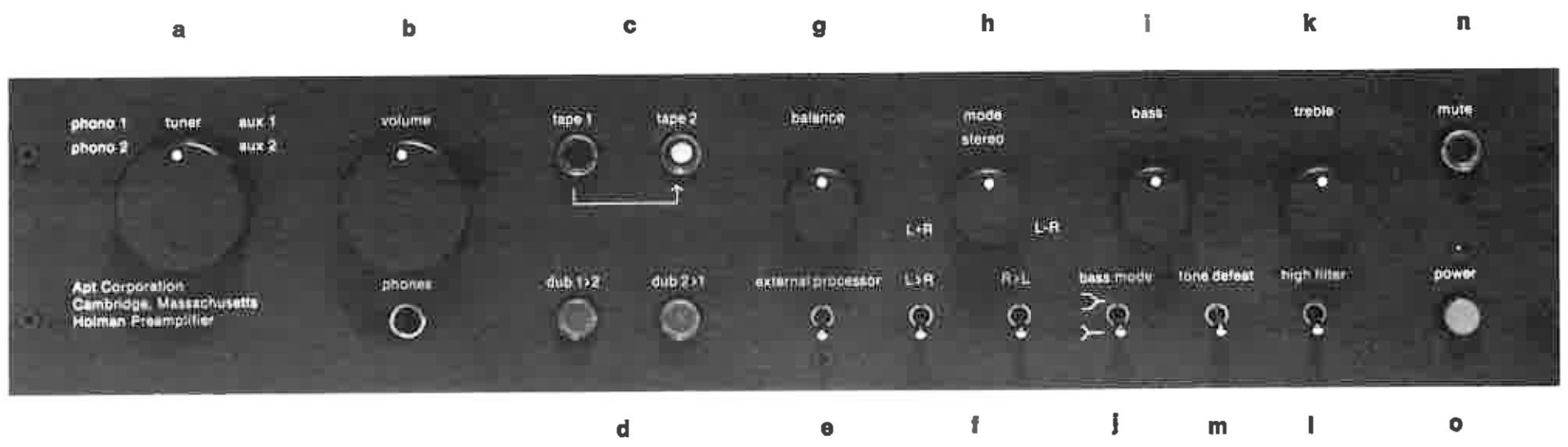
WARNING: To prevent fire or shock hazard, do not expose this appliance to rain or moisture.

Table of Contents

Introduction

Thank you for purchasing the Apt/Holman Preamplifier. Because the design of this product has evolved from a thorough analysis of how a preamplifier should (and should *not*) behave, various aspects of its performance are either subtly or distinctively different from previous preamps. We have provided a detailed Owner's Manual to help you fully exploit its capabilities; we hope that you will find it useful. The Manual has been designed to alternate between material of primary importance, and that required for explicit detail.

Please fill in and mail the Owner's Registration Card.



Front Panel

a program source: Selects input signals, *except* for tape.

b volume: Adjusts level in small increments.

c tape 1 and tape 2: Engaging either button enables the output of the corresponding recorder to be heard, regardless of the settings of the Program Source and Dubbing controls. A white indicator appears in each button when it is engaged. If both buttons are pressed, Tape 2 only is heard.

d dubbing switches: Controls the copying from tape recorder 1 to tape recorder 2 (dub 1>2) and from recorder 2 to recorder 1 (dub 2>1). A green indicator appears in each button when it is engaged. Avoid engaging both switches simultaneously—see text. The dubbing function is unaffected by the tape monitor switches.

e external processor: Permits a signal-processing accessory such as a noise-reduction device or an equalizer to be inserted in the signal path (switch up) or bypassed (switch down).

f channel assignment switches: Enables left or right inputs to be assigned to either or both output channels, as follows:

Switch:	L>R	R>L	
	down	down	normal stereo
	up	down	L source to both outputs
	down	up	R source to both outputs
	up	up	stereo reverse

g balance: Shifts the sound to the left (counter-clockwise) or right (clockwise). Detent at center identifies exactly equal output gains.

h mode: Controls the stereo imaging. Detent identifies normal stereo exactly. Rotation toward left (counterclockwise) progressively blends channels together into mono. Rotation toward right (clockwise) cancels mono components and accentuates stereo difference signals. At full clockwise rotation only difference signals are heard.

i bass: Controls balance of low frequencies.

j bass mode: With switch down (>—), bass control affects mainly deep bass frequencies, with minimal effect on mid-bass. With switch up (⊃), bass control affects entire bass range uniformly.

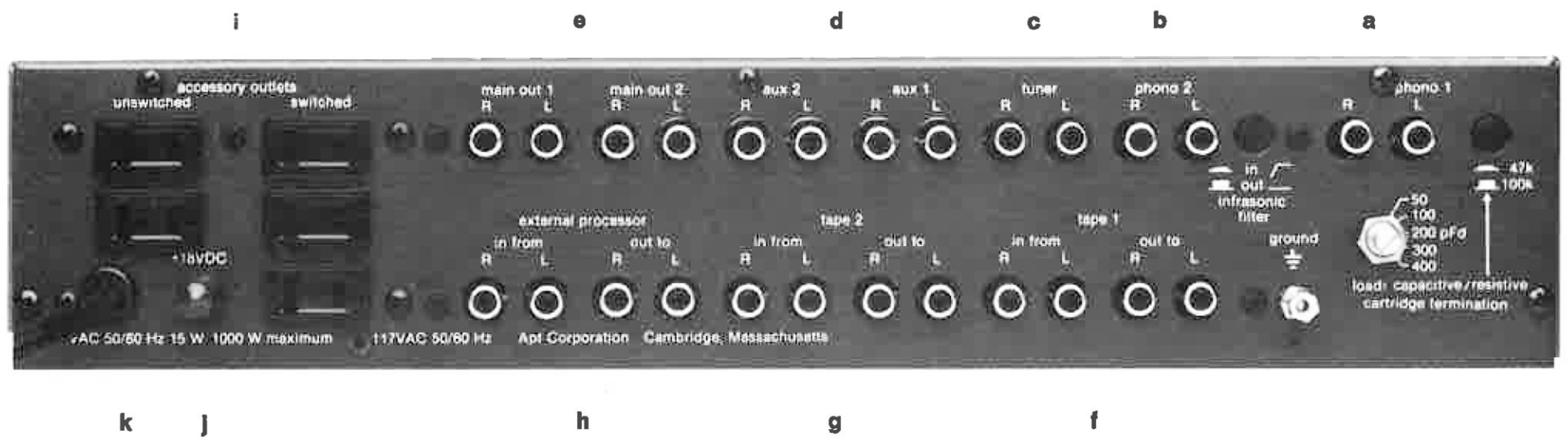
k treble: Controls balance of high frequencies.

l high filter: With switch down, preamp response rolls off above 40 kHz. With switch up, rolloff starts at 8 kHz to remove distortion and noise from worn recordings.

m tone defeat: With switch down, tone controls and high filter circuits operate normally. With switch up, tone and high filter circuits are bypassed.

n mute: Engaging the button silences the output of the preamplifier leaving the headphone jack active; a white indicator appears in the button when engaged.

o power: Switches on power to the preamplifier and to other devices connected to the switched AC outlets on the rear panel. The light emitting diode above the switch lights when power is applied.



Rear Panel

a phono input 1: Connect input from turntable here. As there is no standard color code for turntable cables, consult the owner's manual applicable to your particular situation for proper left/right connection. Set the input resistance and capacitance as required for the particular arm and cartridge combination with the input resistance and capacitance switches. Refer to the accompanying list of cartridge requirements and tone arm cable capacitances.

b phono input 2: Connect input from a second turntable as above. The input resistance is 47 kohms; the input capacitance is 50 picofarads.

c tuner: Connect the output of a tuner here. If you have a choice of fixed and variable output levels from your tuner, the variable output level will allow you to match gain with phono.

d aux 1 and aux 2: Provide additional high-level inputs for connection to the output of an AM radio, TV tuner, tape player, etc.

e main out 1 and main out 2: Provide two sets of output connections. Normally main out 1 will be connected to a power amplifier, while main out 2 will be reserved for a time-delay system, a recorder connection for those times when recording the effect of the tone, mode, and volume controls is desired, or similar uses. Main out 2 has an internal level option which can be used to better match systems with excess sensitivity.

f tape 1 out to/in from: Sends signals to and receives signals from one tape machine. The signal sent to the recorder depends on the setting of the dub 2>1 switch; with the button out the input signal to the recorder is the output of the program selector switch, and with the button engaged, the input signal sent to the recorder is the output of tape recorder 2.

g tape 2 out to/in from: Sends signals to and receives signals from a second tape machine. The signal sent to the recorder depends on the setting of the dub 1>2 switch; with the button out the input signal to the recorder is the output of the program selector switch, and with the button engaged the input to the recorder is the output of recorder 1.

h external processor out to/in from: Allows for a room or speaker equalizer, a compressor-expander, or a similar unit to be inserted in the main signal path. It also may be used for a third recorder.

i AC Outlets: Provide switched and unswitched AC for connection of system components. Connect electronic components to the switched outlets. Connect electro-mechanical components, such as turntables and tape recorders, to the unswitched outlets.

j +18 VDC jack: Connect *only* the power supply cable of accessories designed for +18 VDC power to this jack.

k AC Line Input: Connect the cord to an unswitched source of 120 VAC power.

Connections for a Typical Installation:
Phono 1: Consult the accompanying list of phonograph cartridge termination requirements and tone arm cable capacitances to find the optimum resistance and capacitance values for the cartridge you are using. On the rear panel, set the resistance and capacitance switches as per the instructions on the accompanying sheet.

Install the turntable on a stable, solid surface to prevent vibration, and at a sufficient distance from the power amplifier to prevent interference from stray hum fields. The Holman preamplifier does not itself produce any measurable stray hum field. The turntable's line cord goes to an unswitched ac outlet on the back of the preamp. The ground wire from the turntable (if any) should be connected to the binding posts labeled ground. As there is no standard color code for turntable cables, consult the owner's manual applicable to your particular situation for proper left/right connection. Without a standard, a guideline is: if the pair contains one red plug, it will correspond to the right channel, and among light colored vs. dark colored pairs, the lighter color will represent the left channel.

Tuner: Connect to tuner input. Connect line cord to switched ac outlet.

First tape recorder: Connect the tape recorder's inputs to "tape 1 out to," and connect the tape recorder's outputs to "tape 1 in from." Connect line cord to unswitched ac outlet.

Second tape recorder: Follow the same procedure as for the first tape machine, only connect to tape 2.

Power amplifier or powered loudspeakers: Connect to "main out 1." Connect line cord(s) to switched ac outlets.

Operation
Phonograph: Set selector switch to phono 1.

Since it is connected to unswitched ac, the turntable will need to be switched on separately, ensuring that the turntable will run only when actually in use, and guarding against wear and possible damage to moving parts.

Tuner: Set selector switch to tuner.

Use output level control on tuner to match level with phono.

Tape recorder(s): There is no selector switch position for tape recorder; rather, for the sake of flexibility, the use of a tape recorder is controlled by the sets of switches marked *tape* and *dub.*

The tape switches override the selector switch, and the dub switches connect the two tape recorders together. This feature allows for simple switching with maximum utility:

To listen to first recorder: Press tape 1 so that a white disc appears.

To listen to second recorder: Press tape 2.

To record onto first recorder from phono, tuner, or aux, engage the record mode of recorder 1. The signal supplied by the program selector switch is always present at the tape recorder input, unless one of the dub switches has been actuated. This means that it is possible, for instance, to copy tapes, or to play a tape for friends, while recording a radio program at the same time—without replugging any cables.

To dub (copy): Press the dub button for the appropriate direction.

Power amplifier: Connect main out 1 to inputs of power amplifier. Connect line cord to switched ac outlet. Protect speakers from transients, such as setting down a stylus on the record, by using the mute switch.

Headphones: All dynamic headphones, regardless of impedance, may be used.

Installation

The location of the Holman preamplifier relative to other components is generally not critical. The unit may be mounted in any orientation. Since this preamp does not radiate a significant external hum field, and is designed to be immune to most externally induced hum or other interference, it may safely be stacked on or adjacent to most other audio components.

(Some power amplifiers, however, may radiate such a strong external hum field that their installation near any other audio components is inadvisable.) The preamp has relatively small power dissipation so that, in general, special provisions for ventilation are unnecessary.

In most installations, of course, the preamp and turntable should be located close to each other in order to keep the phono cables short. Long *phono* signal cables increase the likelihood of hum, radio interference, and frequency-response aberrations due to cable capacitance. The best procedure is to begin by locating the turntable on a stable, vibration-free surface, and then to arrange the preamp and other system components in convenient locations nearby.

Because the front-panel controls on the Holman preamp are more genuinely useful and precise than usually found, they invite more frequent use, making it especially desirable, where possible, that the preamp be located within arm's reach of your usual listening chair.

If you wish to rack-mount the preamp, a replacement front panel cut to standard EIA rack dimensions is available from your dealer or direct from the factory. Request the number 100 replacement panel.

AC Convenience Outlets
Five accessory AC power outlets are provided on the rear panel.

Devices which are potentially subject to mechanical wear (e.g. turntables, tape decks) should be plugged into the Unswitched outlets. All-electronic components such as FM tuners, equalizers, and power amplifiers can be powered from the Switched outlets, being turned on and off by the preamp's power switch. The Apt preamp is equipped with a heavy-duty AC line cord and a special 1000 watt power switch, permitting practically all stereo power amplifiers to be powered from the switched accessory outlets.

+18 VDC Outlet
Caution: *Do not plug audio signal cables into this jack.*

This jack, located below the Unswitched AC power outlets, supplies a regulated DC voltage which can be used to power certain accessory devices which are designed for such a supply. The DC current is fed to the jack through a 1000 ohm resistor required by underwriting organizations, so the voltage supplied to an accessory will be reduced by 1 volt for each milliampere of current drawn by the accessory; thus if an accessory draws 3mA, the actual voltage at the jack will be 18 — 3 = 15 volts DC.

Input and Output Connections

Phono 1 and Phono 2
The sound quality of a phono pickup cartridge is strongly influenced by its frequency response, which in most pickups is, in turn, affected by the phono input impedance.

The design of the Holman preamp eliminates the complex impedance interactions which in earlier designs often caused unpredictable response aberrations. The input impedance is a simple combination of resistance and capacitance.

In the Phono 1 input the resistance and capacitance are independently adjustable in order to allow you easily to select the combination which is optimum for your cartridge. The Phono 2 input is provided with the "standard" input impedance: 47,000 ohms in parallel with 50 picofarads. See the sheets labeled "Phono Input Impedance Matching" which accompany this manual for instructions on setting the input resistance and capacitance.

If you are using a single turntable, it should normally be connected to the Phono 1 input. If you use two turntables (or two arms), the one whose cartridge is more load-sensitive should be connected to Phono 1, reserving Phono 2 for cartridges which are optimized for the standard impedance or are insensitive to load impedance.

Connect the left channel of the phono cables to the left input, and the right channel to the right input.

A modification may be made to the Phono 2 input to accommodate unusual cartridge termination requirements. Contact Apt Corporation for details regarding this modification.

If you are using a moving-coil cartridge with a transformer or an external pre-amplifier, the output of the transformer or pre-preamp may be connected to either phono input.

An optional plug-in pre-preamplifier module for moving-coil cartridges is available for installation within the Apt preamp. It replaces the standard Phono 2 input circuit and provides for the differing step-up gain and input-impedance requirements of various moving-coil pickups. It also provides compensation for the undamped high-frequency resonance of many moving coil cartridges.

Installation of the optional pre-preamp module in the Phono 2 circuit of the Apt preamplifier, or custom modification of the Phono 2 input impedance to suit special requirements, has no effect on the Phono 1 input. The two phono inputs have independently determined impedance characteristics. Thus Phono 2 can be optimized for a moving-coil or other non-typical cartridge, while the Phono 1 input retains its flexibility to provide the optimum load impedance for any moving-magnet or induced-magnet cartridge.

Tuner
Plug the signal cables from an FM or AM/FM tuner into these jacks. If your tuner has both "fixed level" and "variable" outlets, in most cases it will be preferable to use the "variable" outputs whose level is controlled by an output-level control on the tuner.

Use that control to adjust the relative volume of the tuner so that you can switch from Phono to Tuner on the preamp without having to substantially alter the setting of the preamp's Volume control.

Aux 1 and Aux 2
Connect to these jacks any "line level" signal sources,

such as an AM tuner, TV audio from an a.c.-line-isolated TV set or TV tuner, the output of a microphone preamplifier, a child's phono with ceramic pickup cartridge, or the output of a play-only tape machine (e.g. an eight-track cartridge player). A normal record/play tape machine is connected to the "tape" inputs and outputs.

Main Out 1
Connect cables from these jacks to the input of your power amplifier.

If you are using an electronic crossover to bi-amplify or tri-amplify your system, connect cables from the Main Out 1 jacks to the input of the electronic crossover unit, then connect cables from the outputs of the crossover unit to the inputs of the power amplifiers. If you are using a time-delay system whose functions include optional processing of the signals going to the primary (front) loudspeakers, then connect cables from the Main Out 1 jacks of the Apt preamplifier to the main input jacks on the time-delay unit, and the front-output jacks of the time-delay unit will then be connected to the power amplifier input (or to an electronic crossover in the case of a bi-amplified system).

If your loudspeakers require the use of a special accessory equalizer (e.g. Bose 901, E-V Interface, etc.), the Main Out 1 signals should be fed to the equalizer, and the output signals from the equalizer should be fed to the power amplifier. But if the Equalizer lacks a bypass switch and you also want to use the power amp to drive electrostatic headphones or speakers not requiring equalization, then the equalizer should be connected to the External Processor jacks, where it can be bypassed at will.

If you are using an octave equalizer, parametric equalizer, or other special purpose equalizer specifically for the purpose of "voicing" the system—i.e., to tailor the speaker/room combination for appropriate response as measured at the listener's chair—this equalizer may be connected to the Main Out 1 jacks, with the equalizer's output feeding the power amplifier input. However, if you are using an equalizer as a highly flexible tone control, to compensate for the varying deficiencies in recordings and broadcasts, then in most systems it is preferable to

preamp main out
l r
↓ ↓
dynamic-range processor (expander, compressor, or noise-reduction device)
↓ ↓
time-delay system → → to other channels
↓ ↓
speaker-equalizer
↓ ↓
electronic crossover
↓ ↓ ↓ ↓
power amplifier(s)

connect it to the External Processor jacks (where the equalizer can easily be bypassed when not in use and can be used to improve signals fed to the Headphones jacks and the Main Out 2 jacks as well).

A dynamic-range processor may be connected to the Main Out 1 jacks, if desired. However, if the dynamic processor is not equipped with a bypass switch, it would be preferable to connect the processor to the preamp's External Processor jacks so that it can be bypassed when not in use. The latter connection also enables the dynamically-expanded signal to be heard at the Headphones jack and the Main Out 2 jacks as well.

If you are connecting more than one of these optional components between the preamp and the power amp(s), the following order of components in the signal path is usually preferable. (Deviations from this order are permissible in special circumstances. If four identical speakers which require electronic equalization are used as the front and rear speakers with a time-delay system, for instance, then a single speaker-equalizer could be connected *ahead* of the time-delay unit to provide proper compensation for all four.)

The main outputs of the Apt preamp will drive any load impedance of 5000 ohms or greater, and any capacitance less than 3000 picofarads to 7 Vrms. Thus, if the input impedance of the power amplifier is 50K ohms, the preamp can drive up to ten such power amps at the same time. A combined load impedance of as little as 2K ohms will limit the maximum output to 3 Vrms, still enough voltage to drive all known power amplifiers. If typical audio connecting cable rated at a capacitance of 50 pF per foot is used, the preamp will drive a total of up to 60 feet of such cable in each channel—either in the form of one 60-foot cable or six 10-foot cables. This means that if you are using powered loudspeakers with built-in power amplifiers, or if you choose to place your power amplifier close to the loudspeakers in order to avoid the adverse effects of long speaker-connecting wires, you may safely place the Apt preamp at the far end of the room from the power amps and speakers. Long connecting cables from the Main Out jacks of the Apt preamp will have no adverse effect on the sound.

Apt Corporation offers a pair of appropriate 30-foot cables for installations requiring such length. Request the number 101 cable set.

Main Out 2

The Main Out 2 jacks are electrically identical to the Main Out 1 jacks and are connected in parallel with them.

They may be used for any of the following purposes:

They enable the connection of a second power amplifier, without requiring the use of Y-connecting adapters.

If you are using a time-delay system whose functions do not include processing of the front-speaker signals, connect it to the Main Out 2 jacks; this permits the delayed rear-channel ambience signals to be derived from the front-channel signals without adversely affecting the primary signals themselves. With the time-delay unit connected at the Main Output of the preamp (rather than at the Tape Out or External Processor jacks), the preamplifier's controls—including the Volume, Tone, and Mode controls—become convenient "master" controls affecting the front-speaker and derived rear-speaker signals equally.

The gain of the Apt preamp has been chosen so that with signal sources having normal output level, power amplifiers of typical sensitivity, and loudspeakers of typical efficiency, the system will produce normal loudness levels when the Volume control is set approximately midway in its range. The Volume control range has been chosen to accommodate wide variations in associated equipment and individual listening preferences. However, it may happen that with a high-output program source, an unusually sensitive power amplifier and a very efficient speaker system, a Volume control setting of only 9 o'clock might produce the highest sound levels you wish to hear. Provision has been made in the design of the Main Output 2 circuit for a simple modification which substantially increases the useful range of the Volume control to accommodate this situation, providing increased operating convenience. The modification involves replacing a jumper in each channel with a pair of precision resistors to form a voltage divider; if you desire this change, see your dealer or contact Apt Corporation for details.

Normally tape recorders are connected to the preamp's Tape jacks in order to make direct recordings of the incoming signals without modifications. Thus all of the preamp's controls may be used to adjust the sound to your taste as you hear it, without having any effect on the character of the signal being recorded from the Tape Out jacks. Similarly, accessories connected to the External Processor jacks (e.g. equalizers, noise filters, dynamic range expanders) may be used to alter and improve the sound which you hear, but will not affect the signals being recorded.

Ground
If your turntable is equipped with a separate grounding wire, it should be connected to this Ground post to minimize hum. Loosen the thumb nut, place the bared wire or spade lug on the post and tighten the thumb nut to fasten it securely.

Tape 1 and Tape 2
These jacks provide for the recording and playback functions of two tape recorders of any type—open reel, cassette, Elcaset, digital, etc. For recording, connect a cable from the "Out To" Tape 1 jacks to the recorder's line-level inputs; these may be labeled "line in," "aux," or "radio," but *not* "microphone." For playback connect to the preamp's Tape 1 "In From" jacks a cable from the recorder's line-level output jacks; these may be labeled "line out," "main out," "monitor," etc. For a second recorder, repeat this process with the Tape 2 "Out To" and "In From" jacks.

However, if you *wish* to use the preamplifier's controls and any accessory connected to the External Processor jacks for the purpose of modifying or improving a signal which is to be recorded, simply connect a cable from the Main Out 2 jacks to the "line" inputs of the recorder, disconnecting the cable which normally would run from the preamp's Tape Out jacks to the tape recorder's line inputs.

The majority of audio systems are satisfactorily hum-free without any need for special grounding. However if you do experience a low-level hum or other interference which is not alleviated by repositioning sensitive components, grounding the preamp chassis to a true earth ground may help. Connect a wire from the preamp's ground post to a true ground point located nearby, such as a cold-water pipe, steam radiator, or the third hole in a correctly grounded AC wall socket, absolutely avoiding any possible connection to either of the parallel *hot* blades.
Do *not* run separate grounding wires from other system components to earth-ground; to do so would create hum-producing "ground loops." In most cases other system components will be satisfactorily grounded to the preamp through the shields of the audio connecting cables, and a single wire from the preamp Ground post to a true earth ground will satisfactorily ground the entire system.

If you are using an outboard tape noise-reduction system (e.g. Dolby B or DBX) with a tape recorder, then the connections from the preamp's Tape jacks will be made to the "amplifier" or "main" input and playback output jacks on the noise-reduction device; the tape recorder will be connected to "tape" input and output jacks on the noise-reduction unit.

Normally a noise-reduction unit is used with only one tape recorder at a time. It is possible to connect two recorders to a single-noise reduction encoder for recording (using Y-connectors at the encoder's "to tape recorder" output jacks, feeding encoded recording signals to the line inputs of both recorders). But for playback it normally is *not* possible to connect two recorders simultaneously to the inputs of the noise reducer for decoding; typical consequences would include severe distortion, response errors, and decoder mistracking. So if a single noise-reduction unit (connected either to the preamp's Tape 1 or Tape 2 jacks but not on both) is to be used with more than one tape recorder, an external stereo selector switch such as the Switchcraft 668 must be used to select one recorder at a time for playback through noise reduction circuitry.

Many tape machines are equipped with a five-pin DIN connector in addition to standard phono jacks for input/output connections. In such cases it is always preferable to use the phono jack connections for recording and playback through the Apt preamp because they provide better matching in terms of impedances and signal levels. If a recorder is equipped *only* with a DIN connector for signal connections, it is possible to use an adapter cable (with a DIN connector on one end and four phono jacks on the other) to make connections to the preamp; the impedances and signal levels will not be ideally matched, but in most cases it will still be possible to record and play tapes.

The signals fed to the tape recorders from the "Out To" tape jacks are selected by the front-panel program-selector and dubbing switches. They are completely unaffected by the settings of any other front-panel controls. If you wish to use the preamplifier controls (e.g. tone, mode, and high-filter), and any signal-modifying device connected to the External Processor jacks, for the purpose of modifying the signals fed to a tape recorder, the recorder should be disconnected from the Tape jacks and connected to the preamplifier's Main Out 2 jacks instead.

However, under the special circumstances that you wish to dub from one tape machine to the other through an external device such as an equalizer, and without effect from the preamplifier's volume and tone controls, you may connect the external device in the Tape 2 loop, and the tape machines in the Tape 1 loop, and the External Processor loop respectively. Then use the dub function to record from recorder 1 through the external device to recorder 2 by actuating Dub 1>2 and the Tape 2 switches.

External Processor

These jacks enable the connection of a broad range of ancillary components which might otherwise use up one of the tape monitor circuits.

Examples include:
A third tape recorder. (The External Processor loop is functionally identical to a tape-monitor loop except that it does not participate in the Dubbing function, and its input source is affected by the tape 1 and tape 2 monitor switches).

A dynamic noise filter, impulse noise supressor, or scratch filter.

A dynamic-range expander, compressor, or limiter.

A graphic equalizer or parametric equalizer.

The special equalizer unit supplied for use with some loudspeakers. Alternatively, such a speaker equalizer may be connected at the Main Out jacks, feeding the power amplifier directly, thus leaving the External Processor loop available for another accessory. But if the speaker equalizer lacks a bypass switch, and if you want to switch the power amplifier to drive electrostatic headphones or a second set of speakers not requiring the equalizer, then the speaker equalizer should be connected to the External Processor jacks where it can be engaged or bypassed at will.

For a 4-channel "matrix" decoder or a time-delay ambience-synthesis system, the "front" outputs from the device could be returned to the External Processor "In From" jacks while the "rear" outputs from the device could be fed to a separate amplifier. Note, however, that in most cases it is preferable to connect a time-delay unit to the preamplifier's Main Outputs instead, so that the preamp's Volume and Tone controls can serve as "master" controls governing the sound to all four speakers.

The signals fed to the External Processor outlets are selected by the Program Selector and Tape Monitor switches but are unaffected by the preamplifier's other controls.

Thus the setting of the "threshold" on a noise filter will not be affected by the preamp's Volume and Tone controls.

Headphones

This is the only connecting socket which is located on the front panel of the preamplifier. The "Phones" jack is wired to a separate headphone amplifier circuit within the preamp. The signals fed to the headphone amp are identical to those fed to the Main Output jacks; i.e. they are equally affected by the Volume, Tone, and other controls.

The gain of the headphone amplifier has been adjusted so that, in most installations, the Volume control setting used for comfortable loudspeaker listening will also provide approximately the right volume for headphones.

Unlike the headphone circuits in many tape recorders, which are adequate to drive only 8-ohm headphones, the headphone amplifier in the preamplifier is designed to be equally effective with the many high-quality headphones whose true impedance is from 100 to 2000 ohms. This category includes many models made by Koss, Sennheiser, AKG, Beyer, Yamaha, etc. However the headphone amplifier is *not* designed to drive electrostatic-type headphones; in most cases these must be connected to a power amplifier through an adapter unit supplied with the phones.

The Phones socket is a standard 3-contact ¼" stereo phone jack, accepting normal tip/ring/sleeve plugs. The wiring to the jack is correct for the majority of models: the tip of the plug carries the left-channel signal, the ring contact on the plug carries the right channel, and the sleeve of the plug is the ground or "common" contact. However, in some models of headphones the tip contact is wired to the right channel and the ring contact to the left channel; with these it will simply be necessary to reverse the phones on the head (or, if you prefer, have a technician re-wire the plug on the headphone cable). If you are unsure of the orientation, you can check it with the balance control: when you turn the balance control to the right (clockwise), the sound should go to the right earcup.

If you wish, you may use headphone extension cables, or headphone Y-connectors to drive two headsets simultaneously, with no adverse effect on the preamp. When two headsets are connected together, they should be identical models; connecting two headphones which differ widely in impedance will usually cause a large loss of loudness in the headset having the higher impedance.

Normally when listening to headphones the preamplifier Mute switch should be engaged in order to silence the loudspeakers. This permits the Volume control to be adjusted freely for headphone listening without overdriving the loudspeakers or disturbing neighbors.

The Headphone output does not participate in the muting function. Therefore, a transient at turn-on and turn-off is normal.

The Phones socket may also be used as a convenience output (instead of Main Output 2 on the rear panel) to feed signals to an extra tape recorder or other device via an adapter cable (Switchcraft 10FK25, Radio Shack 42-2477, or equivalent). The signals at the Phones jack are, of course, affected by the Volume, Tone, and other preamp controls.

The Operating Controls/ Rear Panel

Resistive Load, Capacitive Load

These controls affect the Phono 1 input impedance and are discussed on the accompanying sheets labeled "Phono Input Impedance Matching."

Infrasonic Filter In/Out

This is a push-button switch. Its normal position is In, with the surface of the button almost flush with the rear panel.

For testing purposes the infrasonic filter may be defeated by pressing and releasing the button so that it clearly protrudes through the panel.

The infrasonic filter is an 18 dB/octave filter with a 15Hz turnover frequency. It maintains a flat frequency response within 0.5 dB down to 20 Hz and then rolls off steeply, down more than 30 dB at 4 Hz. Its purpose is to remove the unwanted infrasonic energy content which is present in the output of most turntables due to normal amounts of record warp, tonearm/ cartridge resonance, and direct-drive motor rumble. This energy, if not stripped off the audio signal by filtering, would tend to overload tape recorders, upset the operating points of power amplifier circuits, waste output power, and drive woofers into large cone excursions producing audible amplitude and frequency intermodulation distortion or, even worse, "bottoming" of the woofer. For these reasons it is recommended that the filter always be left in.

The Operating Controls/ Chassis—Left Side

Phono Balance controls

On the left side of the case of the preamplifier are two small access holes. Behind these are controls for trimming the channel balance of the Phono 1 and Phono 2 inputs in order to compensate for the normal 1 to 2 dB errors in channel balance which phono pickups commonly exhibit.

The importance of precise channel balance is often overlooked. Because the subjective limit of perception for differences in loudness is approximately 1 dB, errors in channel balance of that magnitude will not noticeably affect the apparent relative loudness of the two channels, but the stereo imaging of an audio system can be quite noticeably altered by small balance errors. This is particularly true when recordings are played which were made with a minimum number of microphones and which have a natural stereo perspective, with differences in "depth" and hall ambience recorded along with the direct sounds of the instruments and voices. The audibility and character of these differences in depth and ambience depend on precise channel balance and on uniform frequency response in the two channels. The phono balance controls enable you to correct for cartridge balance errors and achieve phono balance to within a very small fraction of a dB without using test equipment. Separate controls are provided for Phono 1 and Phono 2. The procedure is as follows:

Begin by visually examining the installation of the pickup cartridge in the turntable for correct geometric alignment. For example, the body of the pickup should be axially aligned with the tone arm shell (unless it has deliberately been twisted to correct for an offset error in the arm), and, when looked at from the front, the stylus should come straight down from the cartridge body and meet the record surface at a right angle.

Also be sure that any overhang, vertical tracking angle, tracking force, and skating adjustments on the arm have been correctly made.

Then select a monophonic record, or a test record containing monophonic (lateral) cuts. (If neither is available it is possible as a last resort to substitute a stereo recording of a solo vocalist who has been panned into the exact center of the stereo image.) Play the record, adjust the Volume control to a fairly high listening level (use headphones if you prefer), and turn the preamplifier Mode control fully to the right (clockwise), to the L-R position.

Insert a narrow-bladed screwdriver, such as a jeweller's screwdriver or the small screwdriver commonly supplied for installing phono cartridges, through the appropriate access hole on the left side of the case of the preamplifier. For the Phono 1 input, use the access hole located closer to the rear panel of the preamp; for Phono 2, use the access hole closer to the front of the preamp. In either case, gently insert and rotate the screwdriver until you feel the blade go into the slot in the balance control. Then turn the screwdriver to adjust the control for *minimum* sound output.

Ideally a "null" should be obtained, with the sound vanishing. (If using a stereo record, listen only to the sound of the central soloist.) In practice a perfect null will not be obtained, but a substantial reduction in loudness will occur, and the residual sound may be noticeably distorted due to tracing distortions which are largely masked in normal stereo playback.

That completes the balance adjustment. If you have two turntables, place the record on the other turntable, insert the screwdriver into the other access hole, and adjust the other balance control for minimum L-R output. Then re-set the Mode control to normal stereo.

The Operating Controls/ Front Panel

The front panel controls are discussed in the order in which a normal input signal encounters them in its passage through the preamplifier. That order is as follows:

Program Selector
Tape Monitor switches
Tape Dubbing switches
External Processor switch
Channel Assignment switches
Volume control
Mode control (stereo image control)
Balance control
High filter
Bass control
Bass mode switch
Treble control
Tone control defeat switch
Muting relay

Program Selector

This five-position switch selects the input signal to be heard: phono, tuner, or auxiliary source.

The two inputs most frequently used in most systems, Phono 1 and Tuner, are placed adjacent to each other on the switch. The design of the selector eliminates any trace of audible crosstalk (leakage of signals from sources other than the selected one). However, crosstalk may appear at an input which has no device connected to it. If for some reason crosstalk into unused inputs is important to you, you may install shorting plugs in them to cancel crosstalk. Shorting plug pairs are available from Apt Corporation: request one number 103 plug pair for each pair required.

Tape Monitor and Dubbing Switches

All tape-related functions are handled by a set of four pushbuttons. Two are tape monitor switches and two provide for cross-dubbing between tape recorders.

Tape 1 and Tape 2 are "monitor" switches. In each, a white disc is visible when the switch is engaged. Engaging either monitor switch connects the output of the corresponding recorder to the preamplifier's circuitry so that the recorder's monitor output will be heard.

The nature of the output signal from a tape recorder depends on the recorder's own controls. When the recorder is in the playback mode, the signal heard is that of the tape being played. When a 2-head recorder is in the "record" mode, its output signal is usually just the incoming source signal after passing through the tape machine's recording level controls and associated electronics. However, if the recorder is a 3-head machine capable of simultaneous recording and playback, then during recording the machine's output signal may be *either* the incoming source signal or the playback of the just-recorded tape, depending on the setting of the *recorder's own* Monitor switch. Finally, two

To hear a tape being played on recorder 1, engage the Tape 1 button.

To hear a tape being played on recorder 2, engage Tape 2.

If Tape 1 and Tape 2 are *both* engaged, only the monitor output from recorder 2 is heard.

additional situations may occur: (1) If a single-pass noise-reduction system is used when recording, the output signal from the tape machine in the recording mode will either be an encoded version of the input source signal, or, the incoming source signal after passing through the noise-reduction system's recording level controls as associated electronics. (2) If a Dolby-encoded FM broadcast is fed to a recorder having an FM Copy function, then the recorder's output may be a decoded version of the broadcast.

Thus a tape monitor switch may serve several additional functions beyond its basic one of permitting the user to hear recorded tapes. But the rules governing the use of the Tape switches are easily summarized.

With Tape 1 engaged the monitor output signal from recorder is heard, regardless of the settings of the Program Selector or dubbing switches.

With Tape 2 engaged the monitor output from recorder 2 is heard, regardless of the settings of the Program Selector or dubbing switches and regardless of whether Tape 1 is also engaged.

As the arrow between the Tape monitor buttons on the front panel indicates, the Tape 2 monitor circuit is "downstream" from the Tape 1 circuit. The Tape 1 button selects either the Program Selector input or the Tape 1 monitor signal and passes it along to the Tape 2 monitor switch. The Tape 2 button then provides another choice: either the input selected by the Tape 1 button, or the Tape 2 monitor input, is sent to the preamplifier circuits. Thus when the Tape 2 button is engaged, only the Tape 2 monitor signal is heard regardless of the position of the Tape 1 monitor switch. Tape 2 must be disengaged in order for the Tape 1 button to be effective.

Assuming that the Dubbing switches are not engaged, the signal which is fed to the inputs of the recorders is that selected by the Program Selector—typically Tuner or Phono. Since the same input signal is fed to the inputs of both recorders, recordings may be made freely on either recorder 1 or 2, or on both simultaneously.

When making a recording, the Tape monitor buttons permit you to compare the quality of the program source to that of the signal processed through the recorder. If the recorder has an output level control, it should be set so that (with normal settings of the machine's Recording Level controls) the sound is equal in loudness

To copy from recorder 1 to recorder 2, engage the Dub 1>2 switch.

To copy from recorder 2 to recorder 1, engage the Dub 2>1 switch.

to the corresponding Tape monitor button engaged and disengaged. This enables meaningful comparisons.

If the recorder has independent recording and playback functions (with 3 heads), the monitor switch enables the quality of the recording to be evaluated as it is being made. Thus, when taping from Tuner or Phono onto recorder 1, the Tape 1 monitor button may freely be pressed to check the quality of the recording, without disturbing the recording process itself. Similarly, when taping on recorder 2, the Tape 2 monitor button may be used at will to monitor the recording as it is made. The purpose of the Tape monitor buttons is solely to select which recorder's output will be *listened to*.

Control for tape *copying* is provided by the Dubbing switches which permit you to copy from either recorder to the other while simultaneously listening to either recorder or another program source chosen by the Program Selector switch. A green disc appears in each dubbing button when it is engaged.

Now, with both Tape monitor switches *dis*engaged, you may listen to Phono, Tuner, or an auxiliary program source while the tape copying proceeds. To hear the output of recorder 1 (the source machine), press the Tape 1 button. To hear the monitor output of recorder 2 (the copying machine), press Tape 2. In the case of a three-head tape recorder, the Tape 2 button thus becomes a source vs. tape comparision switch, with the output from the recording machine (Tape 2) heard when the button is in. When the copying is completed, disengage the Dub 1>2 switch.

Now, with both Tape monitor switches *dis*engaged, you may listen to Phono, Tuner, or another source while copying proceeds. To check the output of recorder 2 (the source machine), press Tape 2. To hear the output of recorder 1 (the copying machine) engage Tape 1 and disengage Tape 2. The Tape 2 button thus becomes a source vs. tape comparision switch; with the button out, the output is that of tape 1 (the copying machine), and with the button in, he output is from Tape 2 (the source machine). When copying is completed, disengage the Dub 2>1 switch.

Tape recorders always present possibilities for feedback (the output signal returning to the input), resulting in a startling howl. Most such possibilities have been designed out of the tape-processing circuits of the Apt preamplifier, but two remain: If you use microphones to record in the same room with the loudspeakers, activating the corresponding tape monitor switch will cause the recorded signal to be amplified through the speakers and may be picked up by the mikes, producing a howl. So if you ever record with microphones, use the preamplifier's Mute switch to kill the signal to the power amplifier, and use headphones to monitor the recording.

If you activate *both* Dubbing switches simultaneously and have *both* recorders in the recording mode (or have their own monitor switches set to Source), the signal may chase its tail around the loop through both recorders and immediately set up a loud howl. Of course this situation can never arise in the normal use of the dubbing circuits as described above.

Incidentally, although the Volume control is located between the Program Selector switch and the taping switches on the front panel, that does not mean that the Volume control affects the signals fed to the taping or dubbing circuits. Input signals are selected by the Program Selector and are passed directly through the infrasonic filter to the taping circuits; the selected input or tape monitor signal is routed to the External Processor loop, then to the Volume control and remaining preamplifier circuits.

External Processor Switch

If you have connected a signal processing accessory (equalizer, noise filter, dynamic processor, etc.) to the External Processor jacks on the rear panel of the preamplifier, raising this toggle switch will insert the processor into the circuit. Lowering the switch will cause the processor to be bypassed.

Thus, for example, if you have a speaker equalizer connected to these jacks, you may insert it in the circuit for normal loudspeaker listening, and bypass it when listening to headphones or to a second set of speakers which do not require equalization.

If you have no equipment connected to the External Processor jacks, or equipment connected but turned off, then raising the External Processor switch will break the signal path, muting the output.

Channel Assignment Switches

The two switches labeled “L>R” and “R>L” control the channeling of the signals.

In the normal (down) position of the swiches, left-channel input signals go to the left-channel output and right-channel inputs go to the right-channel output.

Raising just the “L>R” switch causes left-channel input signals to appear in both output channels.

Raising just the “R>L” switch causes right-channel input signals to appear in both output channels.

Raising both switches causes the channels to become reversed.

The following table summarizes the assignment of input signals to the output channels of the preamplifier.

Switch	**L>R**	**R>L**	
	down	down	normal stereo
	up	down	L source to both outputs
	down	up	R source to both outputs
	up	up	stereo reverse

Volume

The Volume control provides precise adjustment of the gain of the preamp.

It is a 32-step precision attenuator consisting of thick-film resistors individually trimmed for maximum accuracy. A principal benefit of this design is that the two channels of the Volume control, which are ganged together, track very accurately so that precise channel balance is maintained at all settings of the control. Since the control steps are detented, it is relatively easy to return to a previous setting after a change.

Mode Control

The Mode control on the preamplifier adds either in or out-of-phase left and right channel information at the discretion of the user. It is continuously variable, between mono, stereo, and difference (L-R), but is provided with a detented center position that ensures precise stereo operation. In addition, it also allows a unique degree of control over the stereo field—in particular, over the apparent depth of the stereo image.

As the mode control is rotated slightly counter-clockwise (to the left) from stereo, the two channels are partially blended together and the strength of centrally-imaged L+R information in the signal is increased.

This has several uses:

The variable blend reduces the exaggerated width of the stereo image heard in headphones.

Partial blend also improves the “focus” of solo instruments or voices heard through those loudspeakers which tend to exaggerate spaciousness in some recordings.

Augmentation of L+R information may “bring forward” or strengthen a too-weak or too-distant central soloist.

When the Mode control is rotated fully counter-clockwise, to the L+R position, the stereo channels are fully blended into mono.

One useful application of this mode is for cancelling unwanted vertical rumble and distortion in old monophonic recordings.

As the Mode control is rotated slightly clockwise (to the right) from the detented "stereo" position, the monophonic L+R component of the composite signal is decreased and additional out-of-phase L-R difference information is injected.

This partial L-R injection has several uses:

The added L-R information "opens up" the sound field from loudspeakers which are too constricted. In loudspeakers with normally wide imaging, it can make the stereo image appear to be wider than the spacing between the speakers.

Much of the hall ambience information in true stereo recordings is recorded out-of-phase; strengthening of this signal component adds "depth" and "air" to the stereo field. (Of course, this only works when the right information is present in the recording to begin with.)

Many recordings include a closely-microphoned, centrally-imaged soloist who is over-balanced with respect to the other voices or instruments in the recording. With the mode control you can "push back" the central solo until it blends properly with the ensemble.

In some recordings certain instruments or voices have been recorded out-of-phase, either deliberately (for a spacious effect) or accidentally. Experimentation with the mode control can reveal such interesting effects.

In general, it is useful to experiment with small deviations of the Mode control on either side of the detented stereo position, in order to improve the central focusing of the sound field or to open up the breath and depth of the stereo image, depending on the varying requirements of different recordings. Thus the Mode control can add a new dimension of exploratory interest and long-term listening satisfaction to your enjoyment of recordings.

Examples of the use of this control are: for properly balancing soloists with respect to chorus and orchestra, *Messiah* (RCA LSC 6175) use L-R at 2 to 3 o'clock; for balancing a soloist with respect to orchestra, Renata Scotto, *Verismo Arias* (Columbia M33435); and for cancelling a soloist leaving only reverberation and other interesting effects, Fleetwood Mac, *Rumours* (Warner Bros. BSK 3010).

Full clockwise rotation of the Mode control to the L-R position yields a useful test mode, sometimes called the "null" mode—because when the two channels are identical in level and phase, their L-R difference component is zero (null).

This mode has a variety of uses:

In the making of records, monophonic signals yield lateral groove modulations. Out-of-phase signals and all of the channel-difference information responsible for stereo imaging are cut as vertical groove modulations. The L-R mode provides easy identification of out-of-phase signals, vertical rumble, and other accidents in record production. Incidentally, the L-R signal will usually be found to be bass-shy. This is because record cutting systems commonly employ low-frequency blend to minimize the cutting and playback problems which can arise from large vertical modulations.

To determine the amount of stereo separation in any signal source, compare the loudness of the sound in the stereo mode and in the L-R mode. If the loudness is drastically reduced in the L-R mode, then the stereo separation is slight; for example, classical recordings often have strongly shared information in the two channels in order to present a stable image of the orchestra between the loudspeakers. However, if the L-R mode is nearly as loud as the stereo mode, then the stereo separation is large with little "central mono" L+R energy.

In the full null setting of the control (fully clockwise), the difference information (L-R), which is the out-of-phase component of the sound, is presented to the loudspeakers out-of-phase, maintaining the original relationship. For this reason, the output may become "phasy" sounding at the extreme setting of the control, with sound images appearing outside the normal stereo field. This is a natural consequence of the nature of the stereo signal.

When installing a phono cartridge, play a mono record or the L+R lateral band of a test record, and rotate the Mode control from stereo to L-R. If the loudness remains essentially constant rather than dropping nearly to zero, one of the channels is wired out of phase at the turntable or the cartridge coils are not perpendicular to one another.

The L-R mode makes possible a simple procedure for obtaining exact channel balance on each phono input without test instruments (compensating for balance errors in cartridges) as described on page 21).

If rotating the Mode control from stereo to L-R causes a sound to disappear almost completely, then the input signal is monophonic—regardless of claims to the contrary. FM stations, for example, will sometimes be found to be transmitting in mono, even though the transmitter's stereo pilot generator remains on, activating the stereo indicator in your tuner.

Stylus tracing errors and cartridge mistracking generate distortions in both lateral and vertical directions, but the lateral distortion tends to be masked by the large lateral groove modulation due to the music. Since the vertical modulation of most records is relatively smaller, it tends not to mask the vertical distortion component. So when comparing, installing, or adjusting phono cartridges, the use of the L-R mode makes distortions more clearly audible and thus facilitates cartridge performance evaluation. This may be particularly useful with line-contact styli (Shibata or equivalent), which are especially sensitive to vertical alignment errors.

Stereo FM reception problems—including mistuning, inadequate IF band-width, and multipath (reflected) signals—cause reduced stereo separation and increased distortion in the multiplex decoder. Using the L-R mode you can listen to the multiplex subcarrier output with an increased sensitivity to distortion when it is without the main baseband FM signal. Locate and align your antenna to achieve the loudest and clearest sound in the L-R mode, and you will have the best stereo FM reception which your installation is capable of from that station. This procedure may make a multipath meter or oscilloscope display superfluous. You may have to compromise between the antenna settings which give the loudest (i.e. best-separated) and the least-distorted stereo subcarrier sound—recognizing, of course, that some of the distortion in the L-R mode may be due to phono cartridge mistracking at the FM station. If you can find a station with a clean L-R signal for use as a test source, then the L-R mode may aid in comparing the stereo-decoding and multipath rejection properties of various FM tuners under real-use conditions. If the L-R signal from your favorite FM station is usually noisy, you may need a better antenna.

Finally, since the L-R mode cancels the centrally-located soloists in a recording, you can use it to make recordings to sing or play along with.

Balance

In its normal, detented center position, the Balance control provides precisely equal gain in the two channels of the preamplifier. Rotating the control toward the right shifts the stereo image toward the right-channel speaker by reducing the gain in the left channel of the preamp, and the converse occurs for leftward rotations of the Balance control. The primary use for the Balance control is to compensate for channel-balance errors in a program source.

Slight rotations of the control on either side of the detented center position provide subtle shifts in the stereo image for making useful corrections. Full rotation of the Balance control to either end of its range can be used to fully mute the output of the opposite channel.

To evaluate the imaging of the stereo system, play some music (not a test tone), place the Balance control in its detented center position, rotate the Mode control to L+R (mono), and listen to the image formed by the loudspeakers. Ideally a narrow phantom image should be heard, suspended in the air midway between the loudspeakers. A centrally-located phantom image in mono is desirable because it is an essential pre-condition to obtaining well-balanced stereo images with good resolution of depth and inner detail.

If the phantom image cannot be precisely located at all, it may be that the loudspeakers are wired out-of-phase, are not a well-matched pair, or are designed for spacious qualities rather than precise imaging. If the phantom central image is located off-center closer to one of the loudspeakers, several possible causes may be investigated:

The two power amplifier channels may not be accurately matched in gain, or the loudspeakers may be mismatched in sensitivity.

Check the input level controls (if any) on the power amplifier, and the balance controls (especially any midrange level controls) on the loudspeakers.

The two loudspeakers are very likely to be radiating into radically different acoustical environments. Consider re-locating the speakers to symmetrical room positions, or experiment with the distribution of furniture and wall furnishings near the speakers.

The two loudspeakers may not be located equidistant from your listening chair. Lateral shift of images toward the closer loudspeaker is called the "precedence" effect. For ideal stereo imaging, the loudspeakers should be at equal distances from the listening position. If this is impractical, then you may offset the Balance control to compensate for the image shift due to the precedence effect, or, better yet, if your amplifier has them, offset the relative channel gains with its level controls, reserving the preamplifier balance control for out-of-balance program sources.

High Filter
The High Filter is a maximally-flat, minimum-phase type with a 12 dB/octave rolloff, with a turnover frequency switchable between 8 kHz and 40 kHz. Raising the toggle switch engages the 8 kHz filter, useful for suppressing hiss and the distorted "edge" from old or worn recordings, with minimum intrusion on the musical sound.

In the normal "down" position of the High Filter switch, the frequency response is flat to the limit of human hearing and is rolled off smoothly beyond 40 kHz.

In audio systems there are numerous sources of unwanted ultrasonic energy. These include FM tuners with inadequate SCA filtering, phono cartridge mistracking, the undamped ultrasonic resonance of some phono pickups, bias leakage from tape machines, radio interference of all kinds including nearby CB and commercial AM transmitters, and overload clipping in any program source. Removal of this energy by a linear filter reduces the heating of tweeters (and so improves their musical power-handling), and prevents power amplifiers from producing the intermodulation, slew-limiting, and other distortions which may arise due to non-linear behavior at ultrasonic frequencies. The 40 kHz ultrasonic filter produces no audible time delay, and the two channels are closely identical to preserve lateral directional cues accurately.

The High Filter switch engages the 40 kHz or 8 kHz filters only as long as the Tone Defeat switch is off (down). Raising the Tone Defeat switch bypasses either filter as well as the tone controls, extending the frequency response to 150 kHz for test purposes.

Bass Control and Bass Mode Switch
The Bass control increases and decreases the strength of low frequencies in the sound. There are several different situations in which control of low-frequency levels is valuable, and the flexible and precise design of the Holman preamplifier's bass control circuitry accommodates each of these varying needs.
Speaker Equalization. Many loudspeakers have perfectly adequate mid-bass response but roll off in the lowest octaves below the woofer/cabinet resonance. And some record manufacturers roll off the deepest bass when cutting a disc, in order to increase playing time and minimize tracking difficulties. The Bass control can compensate for these rolloffs; with the Bass Mode switch in the "down" or non-shelving position, the Bass control acts like the lowest-frequency control on an octave equalizer, providing substantial amounts of deep-bass boost with little effect on the upper-bass and midrange tonal balance.
Rumble Filter. In addition to the low-frequency rumble which is present in mediocre turntables, audible rumble is also present in some records;

this may be due to a defective lathe or may be present in the recording studio's master tape due to traffic noise, air conditioners, or floor vibration in the studio. In view of the relatively sharp deep-bass action of the Bass control when the Bass Mode switch is in the "down" position, the Bass control can significantly attenuate such rumble sources *without* noticeably weakening the response in the mid-bass region (around 100 Hz) where much of the real bass energy in music lies.

Loudness Compensation. The human ear's frequency response is not flat, and this disparity worsens as the loudness level is lowered. Thus it is well known that, when music is re-played at lower than its original sound level, some bass boost should be used to prevent the sound from becoming subjectively bass-shy.

However, most "loudness compensation" circuits sound unnatural, in part because they are based on the obsolete Fletcher-Munson curves and produce excessive mid-bass boost.

When the Bass Mode switch is in the "down" position, the Bass control accurately yields computer-derived "differential loudness" curves based on the best modern psychophysical data. You will find that you can easily adjust the Bass control for the most nautral-sounding tonal balance at any desired listening level.

Although it is difficult to give concrete guidelines for setting this control for loudness compensation due to the variations in system gain, loudspeaker efficiency, room acoustics, and individual variations from the composite equal-loudness relations, the bass control shape is such that excess mid-bass boost does not occur when one attempts to balance the lowest octaves appropriately. For most background music listening on good systems, a setting of about 3 o'clock on the bass control in the non-shelving mode should provide approximately correct loudness compensation, while individual variation from this setting can be used to compensate for the variables affecting loudness.

Mid-Bass Boost. Speaker equalization and loudness compensation, as described above, requires a bass contour which provides maximum boost at 20 Hz, less boost at 50 Hz, and little boost at 100 Hz or above, and the Bass control produces this contour when the Bass Mode switch is down. However, in much music the impact of the bass is primarily due to energy in the 60-100 Hz region (drums and bass guitar, for example). In order to enable you to control the mid-bass power without excessively boosting the lowest octaves and risking consequent amplifier overload and woofer damage, the preamplifier is also equipped with a "shelving mode" for the Bass control. When the Bass Mode switch is in the "up" position, the Bass control operates uniformly on the entire bass spectrum below about 150 Hz; in this mode, with most recordings, moderate rotations of the Bass control have a more obvious effect on the tonal balance of the sound, as you may enrich or thin out the entire bass range.

The Bass control is detented at the center of its range. An internal factory-adjusted trimmer ensures that the frequency response is absolutely flat when the control is set to the detented position.

Treble Control
The Treble control increases and decreases the strength of high frequencies in the sound.

Unlike controls which affect only the highest frequencies the Treble control in the Apt preamplifier affects the entire Treble range above 3000 Hz equally with small rotations of the control. This provides effective control over the subjective brightness of the sound, which is due much more to frequencies between 3 and 10 KHz than to the highest overtones above 10KHz.

For example, recordings produced by different record manufacturers tend to have characteristic "house" sounds, with one brand being typically brighter or mellower than another. The Treble control permits you to modify these basic differences in octave-to-octave balance, without producing an exaggerated boosting or dulling of the top end above 10 KHz.

The Treble control is detented at the center of its range. An internal factory-adjusted trimmer ensures that the frequency response is absolutely flat when the control is set to the detented position.

Small rotations of either treble or bass controls away from center produce subtle changes in balance, and the rate of boost or cut increases at extreme settings of the controls.

Tone Defeat
When the Tone Defeat switch is in its "down" position, the 40 KHz ultrasonic filter is engaged and the Bass control, Bass Mode switch, Treble control, and High Filter switch operate normally.

When the Tone Defeat switch is raised, the tone controls and filter are bypassed. By switching the Tone Defeat on and off, you can evaluate the effects of the tone controls and filter on the sound.

Mute
The preamplifier circuitry is connected to the Main Output jacks through a relay. Pressing the Mute switch disconnects the output relay, shutting off the preamp output signal and grounding the output jacks so that no signal can go to the power amplifier. When you activate this muting function, a white disc appears in the Mute button. To restore the normal output connection, press and release the Mute button again.

The principal purpose of the Mute button is to enable you to silence the output to the speakers while listening to headphones via the preamplifier's headphone amplifier. The headphone socket remains "live" when the output muting is engaged. You should also mute the output (or turn off the audio system) whenever changing any input/output signal connections.

The output relay is designed to disconnect automatically in case of "brownout" conditions (AC line voltage below 95 volts), in order to prevent any abnormal pulses or noises from being conducted to the power amplifier. It may be necessary to press and release the mute button to restore operation after a particularly slowly occurring power line voltage change resulting from a brownout.

The output relay provides a delayed connection during turn-on, and it disconnects instantly when the preamplifier is turned off in order to prevent the transmission of transient pulses to the power amplifier.

Dealing with Hum and R.F.I.

The Apt/Holman Preamplifier has been designed with special regard to immunity from hum and radio-frequency interference problems. Tests under exceptionally severe real-use conditions have demonstrated the success of this design approach. So in nearly all cases audible hum or radio interference will be found to have a cause external to the preamp and can be treated or cured without making internal modifications to the preamp itself.

Hum
If audible hum is heard on all inputs, or with the Mute button engaged, then the problem is likely to be in the power amp, in the connecting cables, or in a ground loop. In general, poor connecting cables are the most prevalent cause of hum, either because of a plug making poor contact with its socket or because a poor contact or high-resistance path has developed in the cable. An ohmmeter can be used to check cables to make sure that the signal lead (terminating in the protruding central probe at each phono plug) and the cable shield (connected to the plug's skirt at each end of the cable) each makes a good low-resistance connection from one end of the cable to the other. Intermittent connections can sometimes be revealed by wiggling the cable where it is molded onto the plug.

So-called "ground-loop hum" may be cured by experimenting with the orientation of AC plugs in the sockets; starting with the power amp and working backward toward earlier stages in the system, reverse each AC plug in turn and leave it in the orientation which produces the least hum. Another source of ground-loop hum is the *inappropriate* use of grounding wires to connect stereo components. In most installations the best grounding arrangement involves the *least* grounding. Ground the preamp's chassis to a true earth ground (which, depending on the construction of your home, may be some, all, or none of the following: the *third* (ground) hole in an electrical outlet, the screw that holds the cover plate on an electrical outlet (for these types of ground, be absolutely certain to avoid short circuits with either side of the AC line), a steam radiator, or a cold-water pipe. If the turntable is equipped with a separate grounding wire, connect it to the ground post on the rear of the preamp. All *other* components in the stereo system should be self-grounding through the shields of their signal cables. Adding extra grounding wires from chassis to chassis is likely to make hum increase rather than decrease. Incidentally, if your power amplifier's chassis is already grounded through a three-wire power cord, then connecting the preamp to a true earth ground may create a ground loop as well. In all other cases the usual rule is that the preamp is the only component which should have a ground wire running to a true earth ground.

If you hear hum only from one program source, then the problem is either in the cables from that source, or in the source itself. However, in the case of a phono or tape recorder input, the problem may simply be that the turntable, the cartridge, the playback tape head, or the cables are located in a hum *field.* Such hum fields are created by the large transformers in power amplifiers, by the motor of a refrigerator located nearby (on the opposite side of the wall for example), or occasionally just by the electrical wiring of the house. So in case of hum associated with the phono or tape inputs, after checking the cables for low resistance and secure connections, the best thing is to experiment with the location of the turntable and the layout of the phono signal cables. Do not run the phono signal cables parallel to any AC line cord; for lowest hum, signal cables and AC power cords should be kept separated by at least a few inches to a foot, and if they must intersect they should cross at right angles. When checking the connection of the phono signal cables, do not neglect the way they are usually plugged into sockets underneath the turntable chassis. Another major source of turntable hum is the plug-in connector on the tone arm, as well as the clip-on leads fastened to the terminals at the back of the cartridge. Use a clean pencil eraser to scrub off any visible corrosion from brass or aluminum contacts; all connections must be tight and clean, and all plugs should make a tight fit in their respective sockets.

Radio Frequency Interference (R.F.I.)
Interference from Citizens Band and other radio-frequency transmitters is a widespread problem for which there is no simple prescription. Since the internal circuitry of the Holman preamplifier is unusually resistant to RF interference, we will look at some common external causes and treatments.

If CB or other interference is heard with the preamplifier's Mute button engaged, then the problem is likely to be in the power amplifier—usually because the speaker wires act as antennas and conduct the radio signals into the output ports of the amplifier. Possible treatments include: using heavy-duty shielded cable (with at least 18 gauge conductors, not 20 or 22 gauge) for speaker wiring; placing the power amplifier close to the speakers with extra-long shielded cables running from the preamp to the power amp (available from Apt) so that you may use short speaker wires; and having an RF filter installed at the output terminals of the amplifier (but don't do this without checking with the amplifier's manufacturer first).

Aside from problems arising at the phono cartridge, the other principal cause of RF interference in audio systems is inadequate shielding in the various interconnecting shielded cables. So-called shielded cables are not all equal. They vary widely in the effectiveness of their shielding. Many inexpensive audio cables employ only a spiral-wrapped shield. A cable with a braided shield is usually better, but these vary greatly from brand to brand in the tightness of the braid and in the amount of overlap in the shield wiring. More effective yet, where low capacitance is not a factor, is foil-shielded cable such as that sometimes used for television antenna wiring. Finally, there are special cables marketed with double and triple shielding which, though costly, may be the only cure for really severe cases of interference where powerful transmitters are located nearby.

The various procedures described earlier for eliminating hum may also improve the system's immunity to radio interference. Proper system grounding, for example, improves the general effectiveness of shielding, and cleaning and tightening all contact points in the signal path may eliminate the "diode rectification" which can occur at a slightly oxidized or corroded contact point—for example at a plug socket or at the tone arm headshell.

Tape recorders are especially vulnerable to RF interference, especially in playback, so if you isolate an interference problem occurring in the tape path, you should contact the manufacturer or importer of the recorder for information on available countermeasures.

Sometimes a simple cure for RF interference proves effective—such as substituting better phono signal cables, or shortening the cables, or re-orienting them, or forming a loop of an appropriate size in the cable so as to de-tune the interference. Sometimes a more drastic action is necessary—such as moving your stereo components to the opposite end of your house to get them away from your neighbor's CB antenna which is right outside your window —or convincing your neighbor to move the antenna to the far end of his house. Good luck!

In Case of Difficulty/ Troubleshooting

Many problems people encounter in using good high fidelity equipment are relatively simple in nature and require no more than a little logical detective work to be repaired or at least identified. Troubleshooting usually amounts to following signal paths through the equipment and discovering where a desired signal becomes lost or distorted. By using the accompanying block diagram, a logical series of tests can be constructed in order to track down a failure.

For example, suppose we have no output from the right channel of a system. The first test one would perform would be to ascertain whether the no-right-channel condition applied to all the inputs. This broadly defines the defect, associating it with either a program source, or with the later stages in the preamplifier or subsequent equipment. For our example let us say that all inputs are affected by the no-right-channel condition. The next problem is to assign the problem to the preamplifier, the connecting cable, or the power amplifier. To do this, the power to the system is shut off, and the left and right cable connections are interchanged at the power amplifier input. The power is reapplied and we find that the difficulty has changed channels: the left channel is now dead. This absolves the power amplifier, and leaves the preamplifier and the connecting cable as suspects. We know that the cable now used between the preamplifier's left channel output and the power amplifier's right channel input is good, so the next step is to try that channel of the *cable* between the preamplifier's right channel output and the power amplifier's right channel input. The right channel now functions, so the difficulty must lie in one channel of the cable connecting the preamplifier and the power amplifier, and a new cable or a repair of the old one is required.

There are many features of the Holman Preamplifier which are designed to minimize difficulties in interconnecting components. These include the various buffer amplifiers, the presence and placement of the infrasonic and ultrasonic filters, etc. Switching has been arranged so that in very few cases can output disappear because of absent minded switching, and those conditions which can arise (such as pushing a tape monitor button when no recorder is hooked up) are indicated by tallying pushbuttons.

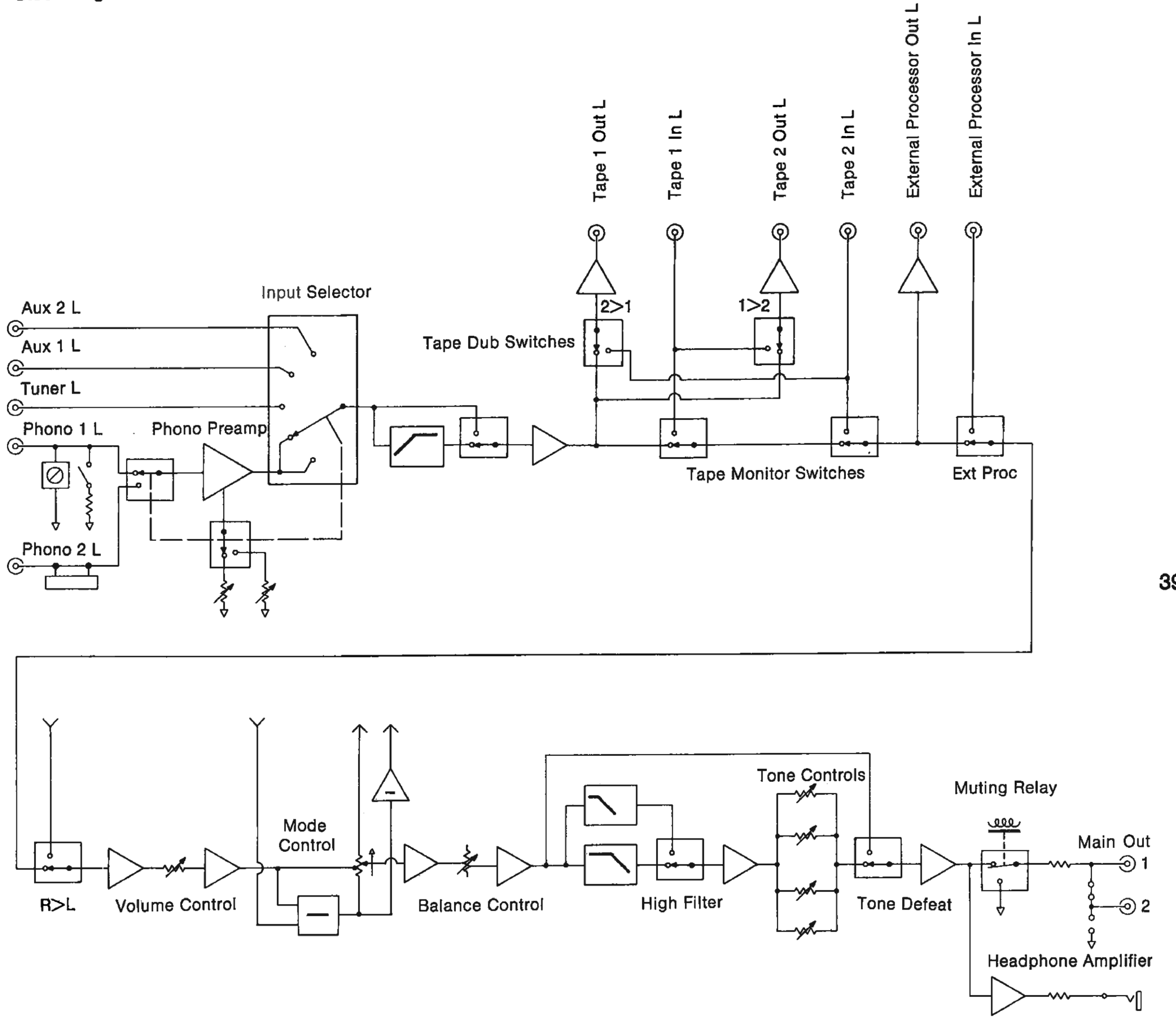
There are two instances remaining where the chosen switching arrangements could result in difficulties. One is the fact that the dubbing switches intentionally do not cancel one another. This allows dubbing in both directions simultaneously, which can be a real convenience when one needs to copy back and forth quickly. However, this arrangement also makes it possible to put both tape recorder 1 and tape recorder 2 into the record mode, so that with both dubbing buttons pushed, a feedback path is formed through both recorders and the preamplifier. Obviously, only one machine at a time should be in the record mode.

Another possibility is that, from mono inputs, there will be no output from the preamplifier when the mode control is set to L-R. Since mono inputs contain no difference between channels by definition, such signals will disappear when the mode control is fully clockwise. And this applies to stereo sources made mono by the action of the channel assignment switches. For example, if on a stereo tuner program, one actuated the L>R switch and turned the mode control to L-R, no output would be heard. This is an inevitable consequence of the nature of mono signals.

Any trouble-shooting procedure should include a check for loose cables and connectors. When checking cable connections, the safest procedure is to turn the power amplifier off in order to prevent speaker damage due to open ground conditions. Alternately, many power amplifiers have speaker on-off switches which may be used, or gain controls which may be operated virtually all the way down.

Many system difficulties turn out to be the result of simple switch mis-settings. Be certain to check for the obvious (amplifier gain controls down, speaker switch off, the Mute switch on the preamplifier engaged, etc.) before proceeding to the sophisticated. It is not a bad idea to return the system to its simplest state in order to check out a problem: switch out all external processors, plug in a phono and headphones and see if music is heard. Problems are often solved simply by disconnecting all components in a system and methodically plugging them all back together. It is also a good idea to operate all the switches and pots through their ranges (with the power amp volume down) a number of times—sometimes this in itself will solve a problem stemming from dirt in the controls.

Block Diagram



The design of the Holman preamplifier grew out of a fresh analysis of why various preamplifier designs do not sound equally good, why typical tests do not correlate well with audible performance differences, what a preamp should do, and what it should *not* do. Some of the specific design goals which evolved are as follows:

The phono preamp must not interact unpredictably with the phono pickup cartridge. Such interactions, not revealed by standard tests, are a primary cause of audible differences among preamps.
The various stages and controls in the preamp must not interact with each other; engaging a circuit or control must not alter the signal level, frequency response, loading, noise, or distortion of other stages. Buffer circuits must be used to isolate succeeding stages of the preamp from each other. Additionally, operation of the Volume control must not alter the channel balance; this requirement for precise tracking rules out the conventional dual-log control potentiometer.

There must be no detrimental interaction between the preamp and other components in the stereo system. All input impedances are exactly as specified, with no hidden inductive or capacitive component. All output impedances are very low and are resistively buffered so that they can drive large cable capacitances or unusual load impedances without distortion, instability, or high-frequency rolloff. Furthermore, the Tape and External Processor outputs are isolated by active buffer stages so that, regardless of how low or non-linear an impedance they may be connected to, no distortion will enter the main signal path through the preamp.

All inputs and outputs of the preamp must be as immune as possible to CB and other radio-frequency interference, but this immunity must not be bought at the expense of degraded audio performance. Where practical, this immunity is obtained by ultrasonic filtering. Where filtering is not a practical solution, the circuit is designed to have very low distortion at radio frequencies as well as in the audio band (because distortion reflects circuit nonlinearity, and a nonlinear circuit will "detect" or demodulate RF signals and produce audio interference products). Thus RF signals which cannot be kept out of the preamp's circuits are passed cleanly on to the filter stages where they are stripped off the audio.

Each stage of the preamp must have flat frequency response and low distortion over a range substantially wider than the audio band, in order to ensure that no infrasonic or ultrasonic intermodulation distortion will produce distortion products falling into the audio range. Each individual stage is internally DC-coupled because such design helps to ensure total stability, even in the presence of severe infrasonic garbage. And every stage has ultra-fast response with small-signal linear operation extending to AM radio frequencies, ensuring that no trace of TIM or slewing-induced distortion will occur in the preamp. But infrasonic and ultrasonic signals can cause severe problems elsewhere in the stereo system if passed on by the preamp. (Examples: tape recorder overload due to infrasonic turntable rumble and record warp signals, TIM in power amplifiers caused by ultrasonic distortion products, non-linear woofer motion and Doppler distortion produced by turntable arm resonance on warped records.) Failure to eliminate such signals is a primary cause of audible differences among stereo systems. So, while individual stages in the preamp can handle signals from DC to a substantial fraction of a megaHertz, linear filters are included to limit the overall input-output preamp response to the 8 Hz to 150 kHz range. Additional linear filters, normally in-circuit but bypassable (with the infrasonic and tone defeat switches) at the user's option, limit the flat response to the 20 Hz to 20 kHz audio range with steep rolloffs below 15 Hz and above 40 kHz. This combination of ultra-fast wideband circuitry with linear filtering yields the virtues of wideband design without its potential detrimental side effects.

The preamp must provide maximum control flexibility with controls optimized for genuine usefulness and effectiveness in real day-to-day music listening. No control should be included out of habit or just to satisfy cosmetic or sales purposes (e.g. High and Low Filters with 6 dB/octave slopes which remove as much music as noise). But sonic purity must not be sacrificed for control flexibility. For example, the volume, balance, and tone controls are in low-impedance circuits so as not to add significant noise, and the tone controls are factory-calibrated with internal trimmers so that when set to their detented positions they are exactly flat in both channels.

The preamp must meet its performance specs, not under unrepresentative test-bench conditions only, but under real-use conditions. For example, the phono signal-to-noise ratio is specified not with a short-circuited input but with a cartridge connected. In production testing of distortion and noise, all of the functions are engaged including tone controls and filter stages, and the Tape and EPL outputs and inputs are connected together so that every active stage and buffer circuit is in the signal path at once.

Dynamic range must be limited only by the signal source. The dynamic range of the high-level inputs must exceed not only all existing media but also exceed that of proposed digital disc and tape systems. Phono input noise should be primarily that of the cartridge's own source impedance, and phono headroom must exceed the highest signals obtainable from any combination of cartridge sensitivity and disc modulation.

The preamp should be totally hum-free in use. This involves not only good grounding and elimination of circuit hum, but also thorough shielding to minimize pickup of hum from nearby sources (such as power amp transformers) and the use of a shielded, hum-bucking power transformer which produces a negligible external hum field (so that the preamp may be placed beneath or adjacent to a turntable without causing induced hum in the phono signal).

Execution
The signal path in the Apt/Holman preamplifier is illustrated in the accompanying block diagram. A full schematic is included in this instruction manual, on the inside back cover.

The Phono 1 input provides the user with two input resistance and five input capacitance choices in order to optimize the loading for various magnetic pickups.

As normally wired the Phono 2 input provides a standard load impedance for the phono cartridge: 47k ohms in parallel with 50 picofarads. This standard input can be replaced by an optional plug-in pre-preamplifier which provides appropriate gain, input impedance (resistance and capacitance), and compensation for the resonance peaking of moving-coil cartridges.

The Phono 1 or Phono 2 input signal is selected by a section of the Program Selector switch, for amplification and RIAA frequency response compensation (equalization). The residual noise of the phono preamp is typically within 2 dB of the theoretical minimum noise attainable (a limit due to the impedance of the cartridge itself). This exceptionally low-noise performance is obtained by careful selection of parts and by designing the circuit to operate at its best with the cartridge connected.

The configuration of the phono preamp circuit is that of an operational amplifier (but in a discrete form, not an IC). Such design assures good performance all the way down to DC; thus the strong infrasonic signals which enter a phono preamp (due to record warps, acoustic feedback, and motor rumble, all amplified by the infrasonic response peak due to the cartridge compliance resonating with the arm and cartridge mass) cannot overload the preamp or cause distortion by upsetting the operating points of the circuit.

Like an op amp, the design employs a "differential" input which isolates the RIAA feedback loop from the cartridge input, preventing interactions which upset the frequency response in other designs. However, the two inputs are not identical; each is optimized for its specific function and for lowest noise. The RIAA feedback is returned to a bipolar input transistor, while the cartridge input is a field-effect transistor (FET). The FET input provides several real advantages:

The characteristically very high input impedance of the FET, combined with the differential pair design, ensures that there will be no unwanted loading of the cartridge impedance. The effective input impedance of the phono circuit is controlled entirely by the user-selected resistance and capacitance; there are no additional reactances or resistances beyond those specified by the input impedance.

The high input-stage current which the FET requires for optimum noise performance (about thirty times higher current than for an bipolar input) also ensures extremely fast charging of circuit capacitances and so yields a very high slew-rate capability for the circuit.

The input FET minimizes the potential for CB or other radio-frequency interference. A FET behaves as nearly an open circuit at radio frequencies, whereas bipolar transistors (including the bipolar transistors used in most op amp ICs) behave as diodes at radio frequencies: the diode is the classic amplitude modulation (AM) detector circuit, and it may even become an FM detector through slope detection; thus, it may demodulate AM or FM RF signals to produce unwanted audio interference.

Separate phono cartridge balance controls, selected by a section of the Program Selector switch, are provided to enable the user to correct for imbalances in cartridge outputs and restore precise stereo imaging. The range of each balance control is approximately ±2 dB.

No adjustment for phono preamp gain is provided, since none is needed. The gain is high enough for good compatibility with low-output cartridges (in some designs the gain is set artificially low in order to make the preamp seem quieter than it really is, at the sacrifice of compatibility). At the same time, the phono input-overload headroom is ample for even the most difficult combination of over-modulated record, disc warp, and high-output moving-coil cartridge used with step up transformer.

The output section of the seven-transistor phono preamp is designed as a miniature power-amplifier, though using small-signal transistors which give much faster ultrasonic response than power transistors could. This design provides ample output current for charging the required RIAA equalization capacitors while simultaneously driving the load well up into the ultrasonic range. This means that ultrasonic distortion products due to cartridge mistracking and ringing will not be "detected" by non-linearities in the pre-amplifier to produce audible intermodulation distortion; instead, such ultrasonic garbage will be cleanly removed by the filter at the output of the phono preamp.

The RIAA equalization is set by a 1% precision film resistor and by 2% precision capacitors in order to ensure that the worst cumulative equalization error is under ±0.2 dB between 30 Hz and 15 kHz. The downward-sloping RIAA curve is followed beyond 20 kHz by an ultrasonic passive filter at the output of the phono preamp (without the consequent leveling off of response of many designs). This approach ensures maximum equalization accuracy between 5 kHz and 20 kHz, and also prevents ultrasonic cartridge distortion products and radio-frequency interference from being passed on to later stage in the preamplifier—or to tape recorders.

The Program Selector is a special three-deck rotary switch. When any input is chosen, all *other* inputs are short-circuited to ground (through a 2200 ohm resistor) so that "crosstalk" or leakage of signals between inputs can not occur in the switch.

The selected input signal passes through a 3-pole, minimum-phase, Butterworth infrasonic filter to remove signals below the audio frequency range. The filter's response is flat ±0.5 dB at 20 Hz and above, is 3 dB down at 15 Hz, and rolls off at 18 dB/octave so that its response is down more than 30 dB at 4 Hz, where the strongest warp signals and tonearm/cartridge resonance commonly occur. This filter also eliminates the powerful infrasonic thump which FM tuners can generate when rapidly tuned off-station. A switch on the rear panel permits the filter to be defeated for special testing purposes.

The outputs to tape recorders and the External Processor Loop are isolated from the main signal path by operational-amplifier buffer stages having low noise and distortion with very fast response; their power bandwidth extends well up into the ultrasonic region so that they will not be overloaded or driven into slew-rate limiting by bias-signal leakage from tape recorders. The buffers provide very low output impedance and can drive long connecting cables and multiple recorders connected in parallel, without difficulty. The reason for the buffers is that some tape recorders and other accessories, when connected but switched off, present a highly non-linear input impedance which—in the absence of buffers—can cause distortion of signals in the main signal path of the preamp.

The tape monitor and dubbing switches are of the six-pole, rather than the usual two-pole, type. The extra poles are used to eliminate crosstalk across the switch, preventing any latent echo from occurring (the playback from a 3-head tape deck bleeding back to the recorder's input through the monitor switch). Dubbing between tape recorders can be done in either direction regardless of the settings of the tape monitor switches. Hence, either recorder may record any program source (including the other tape machine), while the user listens to the program source, the output of the first recorder, or the output of the second recorder. Despite this flexibility, the design eliminates most of the potential feedback paths common in other designs; the user need not be concerned that engaging a tape monitor switch could cause the output of a recorder to feed back to its input and cause a sudden loud howl. Furthermore, since the output signal to both recorders is tapped off ahead of both tape monitor switches, the user can freely engage and disengage *both* tape monitor switches without worrying about accidentally interrupting the signal path to the recorder and thus losing the recording.

The External Processor Loop is identical to the tape monitor circuits except that it does not participate in the dubbing function. It provides a convenient means of connecting an equalizer, dynamic-range processor, or noise suppressor so that it can be engaged or bypassed at will.

Channel assignment switches (labeled L>R and R>L on the front panel) provide a convenient and flexible means of assigning either input channel to either or both output channels. These switches have no effect on the signals fed to the tape recorders or external processor.

A buffer stage located after the channel assignment switches provides a high input impedance for signals entering through the tape monitor or external processor inputs. The input-overload headroom of this buffer is above 10 volts rms (+22 dBm), so that the tape and external processor circuits are even compatible with the high line levels common in recording studio equipment. The buffer provides a low output impedance to drive the volume control, which is a newly-designed low-impedance circuit developed to minimize noise. (High-impedance volume and tone control circuits are significant noise contributors in many designs.) The volume control is a 32-step precision attenuator comprised of individually-trimmed thick-film resistors whose accuracy ensures precise tracking of the two channels and precisely maintained stereo imaging at all settings of the control. To further minimize noise, the volume control is wired in a feedback loop so that as it is turned down it progressively reduces the gain (and thus the noise output) of the associated circuit.

The signal in each channel is fed to the center-tap on a special Mode control which provides complete control over the stereo field. At the counter-clockwise end of the control the two channels are tied together to produce mono (L-plus-R). The opposite end of the Mode control is fed from a difference amplifier which extracts the L-minus-R component of the stereo signal. Thus the user can add into the stereo signal any desired amount of the L+R or L−R component, altering the apparent depth and character of the stereo image. Stereo is defined accurately by a detent on the control which is coincident with a center tap on the potentiometer.

The balance control is another low-impedance type (for minimum noise) fed from an impedance-buffering stage. The taper of the balance control was chosen for best control action, and the detent—for equal gain in both channels—was made accurate by making it coincident with a tap on the control.

The high filter is a minimum-phase type switchable between 8 and 40 kHz. In its normal (40 kHz) position, the filter serves to suppress transient intermodulation distortion in power amplifiers and provides a controlled bandwidth limit for the preamplifier. The "group delay" due to the filter's phase shift is several orders of magnitude below the threshold of audibility, and the two channels are precisely matched in filter phase characteristics in order to prevent any lateral shift in the stereo image. In its 8 kHz position the high filter helps to remove the distortion edge from old or worn recordings, while its 12 dB/octave slope is steep enough to ensure minimal effect on the tonal balance of the music.

The bass and treble controls are low-impedance circuits with 10K ohm potentiometers for minimum noise. Each circuit is individually trimmed in production so that when the controls are set to their detented "flat" positions the frequency response is flat to within ±0.5 dB from 20 Hz to 20 kHz with the infrasonic and ultrasonic filters engaged.

The bass control has two functions selected by a "bass mode" switch. In its "non-shelving" mode the bass control has been designed to provide accurate "loudness" compensation in accordance with modern psychoacoustical data. The control contours approximate the *differential* loudness curves which describe how the ear's response *changes* due to the difference between the original sound level and the chosen playback level. (The fact that the ear's response typically is not flat at the loudness levels of live music should not be compensated for by a "loudness compensation" method.) The design of the bass control circuit also eliminates the unintentional mid-bass cut which accompanies any deep-bass boost in a Baxandall tone control circuit.

In the "shelving" mode of the bass control, the turnover frequency and contour shape of the circuit's action are changed to provide an approximately uniform boost or cut across the entire bass range—so that, for instance, the 100 Hz range can be boosted as desired without excessively overdriving woofers in the 20 Hz range. Similarly, the treble control has a shelving action which provides an approximately uniform boost or cut over the entire range from 5 kHz to 20 kHz at small rotations of the control. Thus the 5-10 kHz range which is primarily responsible for the subjective brightness or dullness of the sound can be adjusted as desired without exaggerated boost or cut in the overtone range beyond 10 kHz.

Both tone controls are de-sensitized in the center of rotation so that small rotations of the control away from the center produce subtle changes in the sound, but similar amounts of rotation near the extremes provides larger change. The ability to make subtle changes about flat contributes substantially to the usefulness of the controls.

For testing purposes the high filter (including the 40 kHz anti-TIM filter) and the tone control circuits may be bypassed using the Tone Defeat switch.

The line amplifier circuit used for the high-level buffer, gain, and output stages is a wide-power-bandwidth operational amplifier using the latest BI-FET technology to provide fast, symmetrical slew rate characteristics. Unlike some instrumentation circuits, it has been specifically optimized for audio use, with very low noise, non-existent crossover distortion, and stable performance down to DC and well up into the ultrasonic range. A special feedback configuration is employed which allows it to drive capacitive loads (such as long connecting cables) with complete stability. Its impedance, gain, frequency response, and distortion performance are simply controlled by input and feedback resistors and capacitors, ensuring extremely uniform performance from channel to channel, unit to unit, and day to day, with no subtle or unpredictable variations in sound quality.

The final line amplifier stage is connected to the output jacks through a muting relay and a 330 ohm cable-termination resistor. When the muting relay is relaxed (i.e. when the power is off, the AC line voltage is low, or the Mute button is pressed), the output line is grounded. When the muting relay pulls in (about five seconds after the preamplifier is switched on, and when the Mute button is released), it connects the line amplifier to the output jacks. The output signal is continuously fed to the headphone amplifier regardless of the action of the muting relay.

The headphone amplifier has appropriate gain and output impedance so that the choice of dynamic headphone type is not critical, and so that, on the average, the loudness in headphones will be approximately the same as loudspeaker listening.

After trying the suggested procedure in the section *In Case of Difficulty—Troubleshooting,* if the difficulty persists, service may be required. Apt Corporation and its authorized service facilities provide service during and after the warranty period. If you wish service from Apt Corporation, write or call for a Return Authorization, in order that your repair be processed promptly.

If, after service, you still encounter difficulty with your system, Apt will attempt to help you solve your system problems. Write or call Apt Corporation to the attention of Customer Service.

The front and rear panel finish may be cleaned with a soft cloth sprayed with a small amount of ammonia/water cleaner *(do not use an alcohol or petroleum based cleaner).*

The Apt/Holman Preamplifier Three-Year Limited Warranty

Apt Corporation warrants that each Apt/Holman Preamplifier is free from defects in workmanship and materials and will perform in accordance with its published specifications for a period of three years from the date of the original purchase. Any necessary adjustments or repairs will be provided at no cost to the purchaser. The warranty covers parts, labor, shipping cost from the service center to the purchaser, and if necessary, packing materials. The warranty is not transferable.

The owner's responsibilities are to use the preamplifier in accordance with its written instructions, to return the enclosed post-paid warranty registration card within fifteen days of purchase, to provide transportation to the Apt factory or to the Apt dealer from whom it was purchased in the event servicing is required, and to provide proof of purchase if requested.

Exclusions: The warranty does not cover malfunction due to electrical, mechanical, or other abuse; nor malfunction due to disassembly and attempted repair by anyone other than Apt Corporation or its authorized service agents; nor shipping damage (which is covered by the carrier: you should ship the unit insured and well-packed); nor to units in non-consumer (i.e., commercial, business, or institutional) use, nor to units whose serial number has been removed or obliterated. The warranty applies only to the Apt/Holman Preamplifier and does not cover consequential damage to other equipment or property arising from a malfunction of the Apt/Holman Preamplifier. (The laws of some states do not permit the exclusion of incidental or consequential damages, so the latter limitation may not apply to you.)

Specifications

1.0. **General**

1.1. The outside dimensions of the case are 3.12 x 15.04 x 8.19" (7.92 x 38.20 x 20.80 cm). The overall dimensions are 3.30 x 15.04 x 9.32" (8.32 x 38.20 x 23.67 cm). The knobs protrude 0.56" (1.42 cm); connectors and cables require up to an additional 1.75" (4.44 cm) depth from the rear surface. The unit weight is 10 lb, (4.5 kg); the packed weight is 12 lb, (5.4 kg.).

1.2. The front and rear panel finish is instrumentation gray with permanent baked-epoxy white markings; the cover finish is neutral gray wrinkle baked enamel.

1.3. All operational modes are indicated visually by means of toggle switches, tallying pushbuttons, indicator dots, or a pilot light.

1.4. Input power line voltage is 120 Vac nominal. Internal rewiring for operation on 240 Vac may be made by a qualified service technician. Either voltage may be supplied at 50 Hz or 60 Hz power line frequency. The range of normal operation for the 120 Vac connection is 95-135 Vac. Low line ("brownout") conditions below 95 Vac activate the muting circuit to prevent transient thumps. Power consumption is less than 15 watts. The switching capacity for the convenience outlets is 1000 watts.

1.5. All data are valid for any combination of input and output, unless otherwise noted.

1.6. Data are valid for 20 Hz to 20 kHz unless otherwise noted.

1.7. Voltages in dBV are referred to 1.0 Vrms.

1.8. Line inputs under test are driven by a generator with source impedance of 1.0 kohm.

1.9. Outputs under test are loaded by 10 kohms in parallel with 1000 pF.

1.10. All inputs and outputs are unconditionally stable under any conditions of source or load impedance.

1.11. 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.

1.12. We reserve the right to make changes as technical progress warrants; but we will not engage in capricious updating. Specifications are subject to change without notice.

1.13. These specifications conform to the requirements of the Institute of High Fidelity Standard IHF-A-202. Specifications to other standards, or without reference to a standard, are, in general, not comparable. In particular, *distortion and noise measurements by other means are not comparable.*

2.0. **Levels, Gains, and Impedances**

2.1. The output level for rated specifications is 2.0 Vrms (+6.0 dBV).

2.2. The maximum undistorted output is at least 7.0 Vrms (+17 dBV).

2.3. The Phono 1 input impedance is switchable between 47 kohms or 100 kohms in parallel with 50 pF, 100 pF, 200 pF, 300 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-preamplifier) is 47 kohms in parallel with 50 pF. Changes to the input impedance to accommodate unusual termination requirements may be made by a qualified service technician.

2.4. Phono 1 or 2 input level for rated output is 5 mVrms (1.25 mVrms for 0.5 Vrms output).

2.5. Phono gain to tape monitor and external processor outputs is 36.5 dB. 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.

2.6. The input impedance for all high level inputs is 50 kohms in parallel with 330 pF. Unselected inputs are terminated by 2.2 kohms.

2.7. The source impedance at the tape and external processor outputs is 330 ohms; the minimum load impedance for those outputs is 10 kohms in parallel with 1000 pF.

2.8. The 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).

2.9. 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.

2.10. A load impedance of 2 kohms on any output will limit the maximum available voltage at that output to 3.0 Vrms.

2.11. 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.

2.12. The input overload for the line inputs is greater than 10 Vrms (+20 dBV, +22 dBm).

2.13. The tracking accuracy of the volume control is within 1 dB over the full range of attenuation.

3.0. **Frequency and Group Delay Response**

3.1. 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.

3.2. 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 4 Hz, with 18 dB/octave slope. Phono and line inputs are all processed through this filter to prevent infrasonic noise from adding distortion to tape recordings or the power amplifier and loudspeakers. The group delay of the filter is 4 milliseconds at 50 Hz, while the limit of perception is over 100 milliseconds at that frequency. With the filter defeated the -3 dB frequency is 8 Hz.

3.3. The ultrasonic filter response is within 0.5 dB at 20 kHz and below, -3 dB 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 and distortion products from being detected by the power amplifier due to inadequate slew rate. The group delay of the filter is 7 microseconds at 20 kHz; the audible limit for a change of timbre is several 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 the -3 dB frequency is 150 kHz.

3.4. The RIAA phono equalization error is within ±0.2 dB from 30 Hz to 15 kHz.

3.5. The phono input impedance interaction is less than 0.2 dB.

3.6. The range of the bass control with the bass mode switch set to non-shelving mode (down) is ±15 dB at 20 Hz, ±7½ dB at 100 Hz, and ±3 dB at 200 Hz. The range of the bass control with the bass mode switch set to shelving mode (up) is ±15 dB at 20 Hz, ±12 dB at 100 Hz, and ±3 dB at 400 Hz.

3.7 The range of the treble control is ±10 dB at 20 kHz, ±5 dB at 6 kHz, and ±3 dB at 4 kHz.

4.0. **Distortion**

4.1. Leinonen and Otala 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 are necessary to reveal various distortion mechanisms. We specify the performance of the Holman Preamplifier on all of those 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.

4.2. All distortion measurements (with the exception of ones which require greater bandwidth) are made with a Hewlett-Packard Model 3580A Spectrum Analyzer. Distortion measurements made with notch-type distortion analyzers will be inaccurate compared to spectrum-analyzer measurements since they null a chosen fundamental frequency and measure the residue in the system under test, whether that residue is distortion or noise. Also, such distortion analyzers use an average-responding voltmeter to measure the residue. Average-responding voltmeters are inaccurate on any but single sinusoid waveforms, therefore they are inaccurate when reading a mixture of sinusoids (the distortion components 2f, 3f,...).

4.3. 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 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 87 dB from the fundamental (0.0045%).

4.4. A 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 sine wave. 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.

4.5. A difference-tone-intermodulation test consists of applying to the input two high-frequency sine waves mixed with an amplitude ratio of 1:1 and examining the output for the difference product at f_2-f_1. 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%.

4.6. A transient-intermodulation-distortion (T.I.M.) test consists of applying to the input a symmetrical square wave 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.

4.7. All stages meet the Jung-Stephens-Todd 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.

4.8. All distortion measurements are made by applying the properly-equalized test signal to the phono input and using each of the tape monitor loop and external processor loop buffers in series. Distortion is then measured at the main outputs with each amplifier in the preamp in cascade.

5.0. **Noise and Crosstalk**

5.1. Unless otherwise indicated, all noise measurements are made with a true rms reading Audio Noisemeter with psychometric weighting for the annoyance value of noise to listeners.

5.2. For the phono inputs the equivalent input noise with a standard cartridge connected and weighted ("A" curve) is less than 74 dB below an input reference level of 5 mVrms at 1 kHz. (The noise is typically 76 dB below 5 mVrms). The noise of a standard cartridge with its load is 78 dB below 5 mVrms. This means the typical noise figure is less than 2 dB: noise figure is a measure of how much noise is being added to a system by an amplifier. In other words, the phono noise is within 2 dB of the lowest noise attainable. Noise measurements made with a shorted input are inaccurate since they do not account for the noise which is generated in the impedance of the phono cartridge and its load.

5.3. For the line level inputs the equivalent input noise is less than -106 dBV "A" weighted.

5.4. The residual output noise with the volume control down is less than -95 dBV "A" weighted.

5.5. 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.

5.6. The 60 Hz hum component and its harmonics are below the weighted noise spectrum.

5.7. 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 3 dB.

5.8. Crosstalk between program sources, including tape monitor, is less than 90 dB to terminated inputs from 20 Hz to 20 kHz.

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

6.0. **Dynamic Range**

6.1. From the data above the line level input dynamic range is shown to be 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.

Schematic Notes: Switch S1 is 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 *npn* 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 within 1 V of 0 Vdc. Phono preamp input FETs are Apt noise-tested types. All circuitry which is not within dotted lines is on Board 1. All capacitors are in microfarads except those marked PF (picofarads) and those marked with "N" as a decimal point (1N5 = 1.5 nanofarads = 1500 picofarads). For resistors, K = kohms, M = megohms. Thus 3M3 = 3.3 megohms.

For conversion to 240 Vac operation, consult the factory.

Phono Input Impedance Matching

Resistance: The pushbutton next to the Phono 1 input jacks provides two options for the Phono 1 input resistance: 47k and 100kohms. The 47k setting (obtained when the button is pressed in, flush with the surface of the rear panel) is correct for most moving-magnet and induced-magnet design cartridges. The 100k setting is preferred with some cartridges which are designed for extended ultrasonic response (particularly those designed for playing discrete four-channel records).

Capacitance: Five selections of input capacitance are available by adjustment of the rear panel rotary switch. In order to make an optimum choice, you should first discover (1) what total load capacitance is recommended by the manufacturer of the cartridge, and (2) how much capacitance is contributed by the wiring between the cartridge and the preamp. The latter consists of the capacitance of the wiring in the tone arm plus the capacitance of the cables connecting the turntable to the preamp. The accompanying tables provide listings of arm and cable capacitance and of the termination requirements for many phono cartridges. In cases of doubt, check the literature supplied with your turntable or arm, or contact the manufacturer. Similarly, the optimum total load capacitance for the cartridge may be listed in the literature supplied with the cartridge, or may be given in published reviews; if in doubt, write to the manufacturer or importer of the cartridge.

Then, from the total load capacitance recommended for the cartridge, *subtract* the capacitance supplied by the tone arm and associated phono signal cables. The difference which remains is the capacitance which should be supplied by the phono input; set the capacitance selector switch to the value nearest the computed difference. If the required capacitance lies midway between two values on the selector switch, either value may be used depending on the response of the other parts of your system (see next paragraph).

If you are unable to obtain correct information, either with regard to the cartridge manufacturer's recommended load capacitance or with regard to the capacitance of the tone arm and phono signal cables, then you may determine the best input impedance setting by experiment. For most cartridges the preferred input resistance is 47kohms. The effect of varying capacitance may be explored while playing recordings with strong high-frequency content. Listen for differences which lie primarily in the top octave (10–20 kHz) which contains overtones and high harmonics which give a sense of airiness or lightness to the sound. Use recordings which contain cymbals or other high-frequency sounds to sensitize the playback system to changes in the upper frequencies. With some cartridges the effect of varying capacitance will be dramatic, and you may find the optimum setting to be fairly obvious. Other cartridges are affected more subtly by the load capacitance. If no obvious effect is heard as the capacitance is varied, then a modest setting is probably best—say, 50 or 100 pF. Combined with the typical arm and cable capacitance of 100 to 150 pF, this will yield a total load of around 200 pF which is optimum for many cartridges.

Should your cartridge be one of the very few which require non-standard load resistance, Apt Corporation can supply a special termination socket for internal installation in the moving-coil pre-preamplifier plug of the Holman Preamplifier. For resistance values of less than 47 kohms, this device plugs in directly without modification to the preamplifier. If you have a need for this service, see the attached accessories order blank.

Tone Arm and Cable Capacitance of Popular Turntables, Arms, and Cables

Make	*Model*	*Capacitance in pF*
ADC	LMF-1, LMF-2	240 (Both models are available with optional 100 pF cables.)
AR	XA, XB, XB-77	135
B&O	All models back to 3000 including 1900, 2400, and 4002 2402, 4004, 3400	180 135
BIC	911, 912, 914, 916, 918 All other models	125 160–180
BSR	All models	240
Decca	Decca	old 300, new 120
Denon	All models	100
Dual	1209 and contemporaries 1225, 1226, 1228, 1237, 1242, 1246, 1249 502, 504, 510, 521, 604, 621, 704, 721	240 100 150
Dynavector	DV505	50 (37 pF/m cable, arm supplied with 1 m.)
Empire	598I, 598II 698	210 100 with 4′ cable 210 with 5′ cable
Fidelity Research	FR54 FR64s, FR66s, FR14, FR12	150 80
GAS	5′	165
Garrard	SL-55, 95 All current models	180 110
Goldens	old, 1 m. new, 2 m.	55 125
Grace	All models	100
Handic	All models	100
Harmon Kardon	ST-6, ST-7, ST-8	115
Infinity	Black Widow	60
JVC	L-A11, L-A55, QL-A2 All other models	140 80
Kenwood	All models	100
Lenco	L133, L236, L830DD L833DD	82 25
Marantz	All models	160
Mayware	Formula 4	120
Micro-Seiki	All tables and arms	150
Mitsubishi	All models	50
Phillips	GA222, GA437 GA212, GA312	100 160
Pioneer	All models	50
Rabco	SL8E	125
Revox	B700 series	200 old, 300 new
Rotel	All models	100
SAEC	WE308	100 old, 37 new

Sansui	All models	85
Sanyo	All models	100
Scott	All models	120
Shure	3009II	125 with 4′ cable
	3009II	75 with CD-4 cable
	3009III	300 with 4′ cable as supplied (75 by deleting the capacitors supplied in the male connector)
	3009IIIS	75
Sony	All models	about 60
Stanton	All models	95
Stax	UA-7	115
Technics	All models	80–100
Thorens	105, 110, 115	230
	Other models back to TD-124	200
Yamaha	YP-800	80
	YP-B4, D6, D8, D10, 211	125

Load Requirements of Popular Moving-Magnet-Type Phonograph Cartridges

Manufacturer	*Model*	*Load Resistance in Ohms*	*Load Capacitance in pF*
Acutex	All models	47k	125
ADC	All models	47k	275
AKG	All models P6, P7, P8	47k	470
Audio Technica	All AT	47k	100–200
B&O	MMC-3000, 4000, MMC-20 series	47k	220
Empire	EDR.9	47k	*
	2000Z, 2000T	47k	300
	2000X	47k	150
	All other 2000 series	47k	500
	4000 series	100k	100
Grado	All F and G series	47k	*
Micro-Acoustics	2002-e, 530-mp, 282-e	47k	*
Nagatronics	All models	47k	*
Ortofon	M20, VMS, FF and Concorde models	47k	400**
Pickering	Models with Q suffix	100k	100
	All XV-15, V-15 series	47k	275
Shure	V15 Type IV	47k	250
	V15 Type III	47k	450
	M24H	100k	100
	M95, M93, M91, M75, M70 series	47k	450
Signet	All moving-magnet models	47k	100
Sonic Research	Sonus models	47k	400
Stanton	Models with Q suffix	100k	100
	500, 600, 680, 681, 881 series	47k	275

*Cartridges thus marked are claimed by their manufacturers to be insensitive to capacitive loading. Therefore, use the 200 pF setting on the preamplifier for these types.

**These cartridges may be shipped with termination capacitors inserted on the back of the cartridge. Be certain to account for this added capacitance if they are present.

This list is compiled of currently-available phono cartridges which are designed for direct connection to conventional moving-magnet phono inputs. For older models, the cartridge manufacturer should be able to supply the most appropriate loading requirement information.

These lists of arm and cable capacitance and of cartridge load requirements have been generally compiled from manufacturer's data, and, as such, Apt has no control over the accuracy of the data.

Input Systems for Moving-Coil and Moving-Ribbon Cartridges

Moving-coil and ribbon cartridges range greatly in sensitivity, internal impedance, required load impedance, and frequency response. For this reason, universal pre-preamplifier designs must be seriously compromised, or must have many difficult-to-make user adjustments.

Therefore, Apt Corporation offers separate models of pre-preamplifiers for installation in an internal socket in the Apt/Holman Preamplifier. Each of the models is matched to the requirements of one or a small range of cartridges. This level of matching allows considerable optimization of such performance parameters as noise and frequency response.

Some moving-coil cartridges are designed for direct connection to conventional phono inputs. Many of these can be terminated by the 47 kohm input load, and are insensitive to the input capacitance setting (for these types, 200 pF capacitive loading is recommended). However, some require other loads, such as 100 ohms. For these types, Apt can supply an input termination board for the phono 2 input. See the accompanying accessories list.

Manufacturer	*Model*	*Recommended Input System*	
		Load Resistance in Ohms	*Load Capacitance in pF*
Adcom	XC series	47k	200
Audio-Technica	AT-30E	Pre-preamplifier Model 201-D	
Coral	777EX	Pre-preamplifier Model 202-B	
Decca	London MkV1 Gold	47k	300
	London MkV1 Plum	47k	300
Denon	DL103 series	Pre-preamplifier Model 201-A	
Dynavector	10A,X; 20AMkII; 20BMkII; 30A,B	47k	270
	20C; 30C; 100D,R	Pre-preamplifier Model 201-A	
EMT	XSD-15	Pre-preamplifier Model 201-C	
Fidelity Research	FR-1 Mk3F, Mk3HE+	Pre-preamplifier Model 202-D	
G A S	Sleeping Beauty	Pre-preamplifier Model 202-B	
Grace	F9-F, F9-U	100k	80
	All other F8, F9 series	47k	250
JVC	MC-1	Pre-preamplifier Model 201-A	
Koetsu	SG-2	Pre-preamplifier Model 202-C	
Linn Sondek	DC2100K	Pre-preamplifier Model 202-C	
N A D	9000	47k	200
Nagatronics	HV9100	Contact Apt Customer Service	
Ortofon	MC-10, MC-20, MC-30	Pre-preamplifier Model 202-A	
Osawa	18 series	47k	200
Satin	M-18E, M-18X, M-18BX, M-117G	100	200
Signet	Mk111E, Mk112E	Pre-preamplifier Model 201-B	
Sony	XL-55	Pre-preamplifier Model 201-A	
Supex	SD-900E+ Super, 900MkII, SDX-1000	Pre-preamplifier Model 202-C	
	SD-901E+ Super	47k	200
Yamaha	MC-1c, MC-1x	Pre-preamplifier Model 201-A	

Pre-preamplifiers

Because the class of cartridges called moving-coil differ greatly in their internal impedance, as well as their required load impedance, sensitivity, and frequency response, Apt Corporation offers a number of different plug-in pre-preamplifiers to satisfy the requirements of the various types.

Each of the pre-preamplifiers consists of a plug-in circuit card for which a mating connector is provided inside the Holman Preamplifier. The Phono 2 input is converted by the installation of the circuit card and by the cutting of four jumper wires to a moving-coil input. A connector to convert back to the moving-magnet function is supplied. Installation is straightforward and may be performed by the end user or by the dealer.

Model	*Cartridge(s)*
201-A	Denon DL103 series, JVC MC-1, Sony XL-55, Yamaha MC-1s, MC-1x
201-B	Signet Mk111E, Mk112E
201-C	EMT XSD-15
201-D	Audio Technica AT-30E
202-A	Ortofon MC-10, MC-20, MC-30
202-B	G.A.S. Sleeping Beauty, Coral 777EX
202-C	Supex SD-900 E+ Super, Linn Sondek DC2100K
202-D	Fidelity Research FR-1 MkIIIF

Specifications

All specifications of the Apt/Holman Preamplifier apply, with the following additions:

2.14. The phono 2 input level for 0.5 Vrms output is adjusted in each model to produce sensitivity equal to the conventional moving-magnet input sensitivity.

2.15. The phono 2 input impedance is adjusted in each model to provide optimum termination.

3.6. The frequency response is the complement of the cartridge for overall flat response.

5.10. The equivalent input noise with the specific design cartridge connected is less than 77 dB below 500 microvolts input at 1 kHz, A-weighted for 201 models, and less than 80 dB below 500 microvolts input at 1 kHz, A-weighted, for 202 models (typ. 82 dB).

Note: The specifications given, like those of the Apt/Holman Preamplifier brochure and Owner's Manual, conform to the requirements of IHF Standard A-202. Specifications written to other standards or where no standard is stated are generally not comparable.

Accessories Order Form

Name ____

Address ____

____

Shipping address (if different) ____

____

Quantity	Model	Description	Price	Weight (kg)	Extension
____	100	Rackmount Kit (*includes replacement front panel*)	$35	1.0 kg	____
____	100-B	Rackmount Kit in Black (*as above but black, with knobs*)	$60	1.0 kg	____
____	101	Pair of 10 meter phono cables	$10	0.5 kg	____
____	102	Special phono termination (*specify required resistance, capacitance*)	$5	0.5 kg	____
____	103	Shorting plugs per pair	$1	0.2 kg	____
____	201-(all suffix)	Pre-preamplifier	$85	0.5 kg	____
____	202-(all suffix)	Pre-preamplifier	$105	0.5 kg	____
____		Six technical papers by Tom Holman	$2	0.2 kg	____
____		Extra Owner's Manual	$4	0.2 kg	____
		Subtotal		$	____
		5% sales tax (Massachusetts only)			____
		Foreign shipping: (*Please specify* air *or* surface; *include sufficient postage for total weight to country of destination. Excess refunded.*)			____
		Total order		$	____