DYNAMIC SOUNDS ASSOCIATES

Phono III

USER MANUAL

120/240 VAC Operation

Naples, Florida

WELCOME

Dynamic Sounds Associates LLC (DSA) welcomes you to our world of the finest possible audio electronic products. We thank you for your purchase and we assure you many years of musical enjoyment. We are always available to answer your questions and we welcome comments regarding our products. Feel free to contact us at any time through our website at www.dynamicsounds-assoc.com or by sending messages directly to info@dynamicsounds-assoc.com. We look forward to your feedback and will gladly respond to all questions and comments.

IMPORTANT INFORMATION

1. All units are set to operate on 120VAC ONLY unless a label is applied to the back panel under the power plug indicating 240VAC operation. Operating a unit set for 120VAC on 240VAC will cause permanent damage and void the warranty.

2. **Dynamic Sounds Associates LLC** reserves the right to make changes or modifications to future units without prior announcement. Any such changes or modifications will be for the purposes of improving the mechanical or sonic performance. **Dynamic Sounds Associates LLC** is under no obligation to incorporate any changes or modifications into prior units; however, it may be possible to provide upgrade packages for prior units—if desired—for a reasonable charge.

3. Registering your component with us by using the form at the conclusion of this manual, will allow us to contact you with potential product upgrade information. **Information regarding upgrades may also be requested by e-mail to info@dynamicsounds-assoc.com.**

INTRODUCTION TO THE DYNAMIC SOUNDS-ASSOCIATES *PHONO III*

Many times over the years since I founded Dynamic Sounds Associates (DSA), I have been asked why did I chose to design a phono preamplifier as the first component of what has become a multiple component line of products? My response has always been, "... because the phono preamplifier is the most critical component in the electronic chain of components in your listening system." Why is it so critical? Because, if the phono preamplifier degrades the audio signal in any way, that information is lost or altered in a way that can never be corrected in any of the subsequent amplifier stages. It is gone forever; and, unfortunately you may never realize that it is gone. I started Dynamic Sounds Associates so that I could design a phono preamplifier that was totally faithful to the input signal in every respect, and honor my "Hippocratic audio oath" of, "First, do no harm to the signal."

Designing a really good phono preamplifier is not easy, it must be able to provide amplification factors as low as 100 (40dB), to as much as 2000 (66dB). It must have a dynamic range of 40-60dB at any gain setting, and a frequency response that exceeds the audio range of 20 Hz – 20 kHz by over an order of magnitude. And, when everything else is done, it must also have very low distortion and noise. This is a tall order for any piece of electronic equipment, but through a process of careful design and analysis, combined with using only the finest component parts in the finished product my goals were achievable. My first phono preamplifier, the DSA Phono ONE, established the basic "architecture" on which the *Phono II* and now the *Phono III* are based. Within this architecture I was able to achieve the competing goals of high gain and high dynamic range, combined with a "flat" bandwidth of >500 kHz. Also, since I dislike using coupling capacitors in the audio chain, I was able to produce a design that is essentially "DC coupled", and remains stable with respect to DC drifts and offsets at the output without resorting to feedback systems for correction. Finally, to eliminate the occurrence of transient harmonic distortion (THD), there are no inter-stage or "global" feedback loops in the preamplifier; and, all of the playback equalization is performed though passive low pass filters between the gain stages.

The end result of this "no-compromise" design approach is the DSA *Phono III* that combines all of the original goals and "lessons learned;" but, combined into a package which offers unmatched operational flexibility. Through the use of a remote control, the listener can now make adjustments in gain, phase, and cartridge loading from the comfort of their favorite listening chair. The DSA *Phono III* also has four playback equalization curves, in addition to the standard RIAA curve, that are user selectable and designed to provide proper playback compensation for many LPs produced prior to the adoption of the RIAA standard.

The ability to adjust cartridge loading from the listening position lets the listener make alterations in the sound at the LP level by "shaping" the response of the phono cartridge in order to enhance the sound from the listener's perspective. Yet, the internal memory will retain the "baseline" value of the loading unchanged so that it is easily recalled when required.

I believe that you will find the DSA *Phono III* listening experience as enjoyable as I found the design process.

Douglas Hurlburt, Ph.D.
Founder and Principle Designer
Dynamic Sounds Associates

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1.0 GETTING STARTED

We know you are eager to get your new *Phono III* into play. This section will provide preliminary information on the features of the *Phono III* and familiarize you with the layout of the controls and the connectors. **Section 2.0** will guide you through the process of the initial setup of the *Phono III* and making the proper connections. **Section 3.0** will provide additional information on the operation and use of the features of the *Phono III*, plus instruction on setting the internal memories for gain and cartridge loading. **Section 4.0** will provide more detailed information on the overall design and capabilities of the *Phono III* and can be reviewed at your leisure. **Section 5.0** will provide information on how to balance the outputs of the *Phono III* to maintain its high degree of performance. The *Phono III* comes pre-balanced and, during normal operation, should not require checking and adjusting the output balance more than once/year.

1.1 Unpacking

The *Phono III* should be carefully removed from the packaging material in the shipping box. The following items should also be found in the *Phono III* shipping box:

- Shielded power cord
- Four Critical Mass Isolators
- 2mm slotted screwdriver
- XLR female connector with three wires
- One pair of RCA (unbalanced) to XLR (balanced) adapters
- Plastic bag containing:
    - 1/16” Hex key wrench
    - 0.050” Hex key wrench
    - Four threaded adapters for mounting the Critical Mass Isolators
    - ¼ watt resistor

1.2 Installing the Critical Mass Isolators

- Remove the *Phono III* from its plastic protective bag.
- Screw the large-diameter end of the threaded adapters (from the above plastic bag) into the Critical Mass isolators. (It is recommended by Critical Mass that the paper disk on the top of the isolators be left in place.) **NOTE:** The threaded adapters are made of titanium and are specifically designed for the Critical Mass Isolators.
- Screw the small diameter end of the threaded adapters into the threaded holes in the bottom of the DSA *Phono III*. (These should be screwed in firmly such that the top surface of the isolators is in direct contact with the bottom of the DSA *Phono III*.)

1.3 First Look at the Phono III

After unpacking your *Phono III*, installing the four isolators, and ensuring that all of the parts are provided, you should take a few moments to familiarize yourself with the features on the front and back panels.

1.3.1 Front Panel

The front panel of the *Phono III* is shown in **Figure 1**.

Figure 1 – *Phono III* Front Panel

The front panel is divided into three sections identified as: right, left, and the center column which has an upper and lower part. The functions located in each section are described below:

- **Right Section (Push buttons)**
  - **Standby** – amplifier is turned on but the output drivers are not operational
  - **Mute** – amplifier and output drivers are powered, but output is muted
  - **Run** – amplifier is fully operational
  - **Mode** – Stereo/Mono selection
  - **Phase** – Non-Inverting/Inverting operation
  - **L-R/R-L** – Channel differencing (only operational in mono mode) (See **Sections 3.2 and 3.5**)
  - **Filter** – High Pass filter for suppression of excess low frequencies from source material

- **Left Section (Push buttons)**
  - Input selection (A, B, C)
  - Gain selection referenced to 1kHz (40dB, 46dB, 50dB, 56dB, 60dB, 66dB)

Note: The input selection and gain selection functions are operational ONLY when the unit is in the Standby or Mute mode.

An internal memory retains the gain setting for each input and, when that input is selected, the gain will be set to the stored value. The *Phono III* comes with the gain for each input set to 40dB and the user must select the desired gain for actual use (see **Section 3.3**). Once selected, that gain will be saved—even if the unit is turned off (for up to a 5 year period of time)—and will be applied the next time that input is selected. **Note:** After turning the *Phono III* power on, and until an input is selected, the unit will always show the "default" gain of 40dB.

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- **Center-Upper Section (See Figure 1A)**
    - The LED display indicates the value of the loading for the selected input.
    - The "Loading" LEDs indicate if the display is showing "Ohms" (for MC) or "pF" (for MM). In addition, the "100K" loading LED indicates if the resistive loading for an MM cartridge is 100Kohms instead of the default 47Kohms. The type of loading (Ohms or pF) is selected via switches on the Center-lower section for each input. (See following section.)
    - The "Equalization Indicator LEDs" indicate what form of playback equalization has been selected for that input using switches on the Center-lower section. (The equalization options are discussed in more detail in **Section 4.4**.) The "EQ Group LED" shows which equalization group is selected (RIAA is the default and will show a blue color.) The LEDs "EQ1" and "EQ2" indicate which of these has been selected within the selected EQ group. These are not illuminated if RIAA is selected. (See following section.)

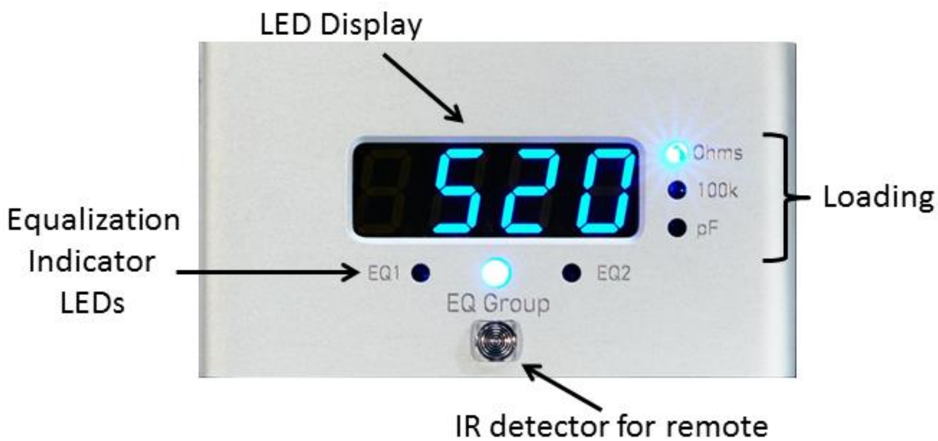

Figure 1A – Center-upper Section

- **Center – Lower Section (See Figure 1B)**

This section is a hinged door (swings to left) that reveals the controls that select the type of cartridge loading, and the desired playback equalization, to be used for each input. Also there are Up/Down buttons that can be used for setting the cartridge loading value for each input. It will be seen that there are three rows of switches, one for each input. Only those switches for the selected input are "active" at any one time.

- Loading type – The first switch selects whether the loading is for a MC (ohms) or MM (pF) cartridge at that input. In the MM position the second switch permits selecting 47Kohms or 100Kohms as the resistive load. If MC is selected, the second switch is not operative. **NOTE:** The first switch for input A has a center "off" position that allows the user to use a custom loading value instead of the internally selected loading. (See **Section 2.2** for details.) If a custom loading is to be used for input A, the LED display in the Center-upper section will show "– – – –".

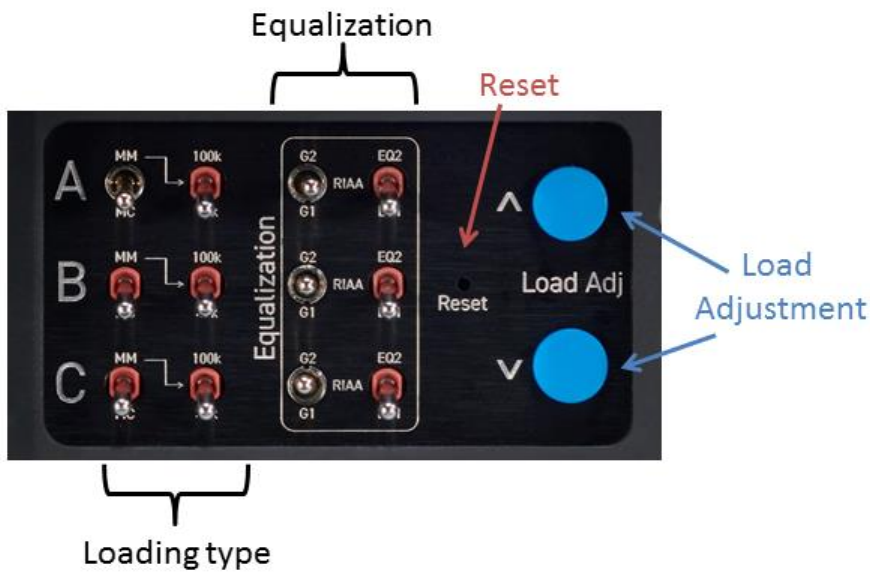

Figure 1B – Center–lower Section

- Equalization – The first switch selects the equalization group (G1 or G2). The center "off" position selects the RIAA playback equalization for that input. The second switch selects either EQ1 or EQ2 within the selected group and is inactive if RIAA is selected. (See **Section 4.4** for details of the other equalization curves)
- The load adjustment buttons may be used instead of using the remote control to adjust the actual cartridge loading; however, the internal memory ***ONLY*** stores the loading value for each input when selected using the buttons shown here in the Center-lower Section. The stored loading value will be applied whenever that input is selected. (See **Section 3.4** for more detail.) When using a MM cartridge, the input capacitance of the *Phono III* (120pF) is automatically added to any selected value. Thus, the minimum capacitance shown on the LED display for a MM cartridge will be 120pF.
- A small opening labelled "Reset" clears the loading memory for the selected input without using the buttons or remote. The LED display will then show either "0 ohm" or "120pF". (Use the provided 0.050" hex key wrench for this purpose instead of a paperclip.)

Notes:

(1) When powered on it takes approximately 30 sec for the main amplifier power supplies to reach operating voltage and stabilize. During this time the Standby LED will show a red-orange color and will blink at a 1Hz rate. **(If the Standby LED shows a red or green color only, turn off the unit and contact DSA.)**
(2) An input selection cannot be made, nor can the *Phono III* be switched to the Mute or Run mode, until after the power supplies have stabilized which will be indicated when the standby LED stops blinking.
(3) When switching between the Standby and the Mute mode, there is a delay of a few seconds for the *Phono III* to come out of the Standby mode, and before the Mute LED will illuminate. This is due to a startup delay in the output stage for each channel.
(4) Once in the Mute mode, switching between the Mute and Run modes is instantaneous, as is the response when switching from Run to Mute.

1.3.2 Back Panel

The back panel of the *Phono III* is shown in **Figure2**. To the right there are three inputs labelled “A, B, C”, and each input has both an RCA (unbalanced) and XLR (balanced) connector for both the Left and Right channels. At each input there is also a ground binding post for use with input cables that have a

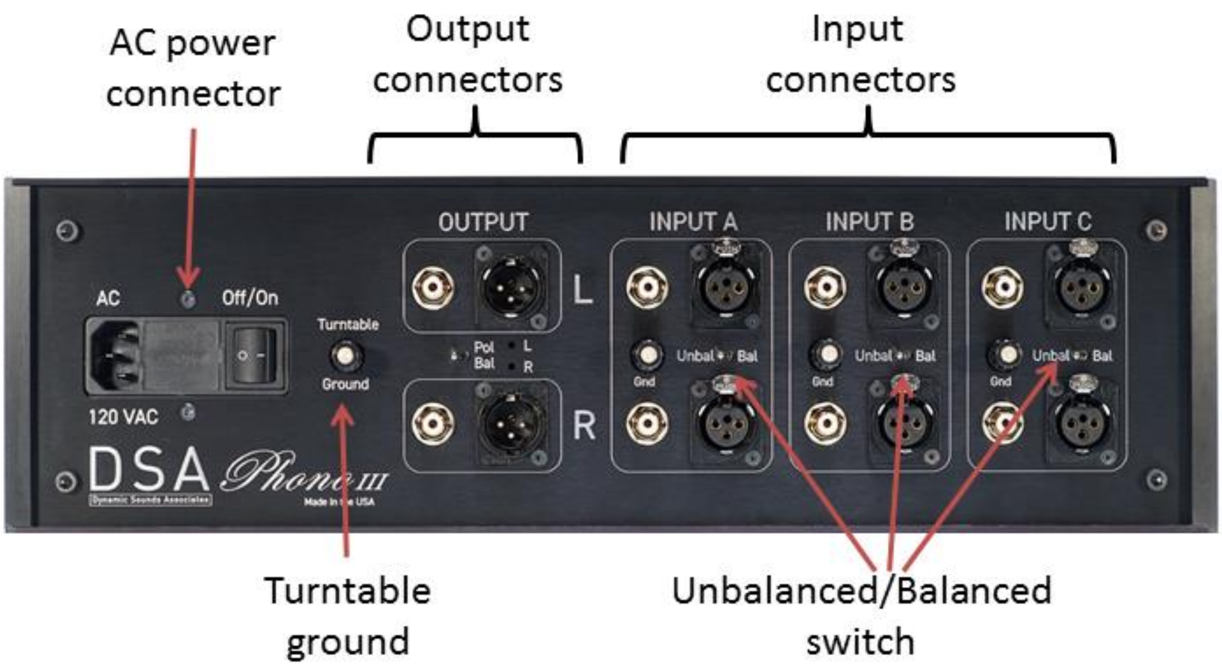

Figure 2 – *Phono III* Back Panel

separate lead for the external shield. At each input there is a switch to select either the unbalanced (RCA) connector or the balanced (XLR) connector. When one of these connectors is selected, the other connector is disconnected from the input.

The output connectors are to the left of the input connectors, and each channel has unbalanced (RCA) and balanced (XLR) output connectors. It is possible to use both output connectors simultaneously, if desired, without damage to the unit. (For example, you can provide audio to a line stage and a separate sub-woofer amp using both output connectors.)

At the far left is the AC power connector with integrated power switch. The AC connector also houses the AC line fuses in a plug-in module located between the AC connector and the power switch. **NOTE: The fuse holder can be removed only when the power cord is detached from the AC receptacle. Two replacement fuses are contained within the fuse holder (see Section 2.1).**

The binding post labeled “Turntable Ground” should **ONLY** be used for attaching a grounding lead from your turntable (if required) to the *Phono III*. Do not connect other grounding wires to this terminal or ground loops and AC hum may occur.

1.4 Remote Control

The *Phono III* remote control is shown in **Figure 3**. The remote control will operate the following functions:

- Mute and Run
- Load adjustment
- Blue LED intensity
- Gain selection
- Phase

The cluster of five buttons in the upper- center of the remote is used for load adjustment, muting the unit, and adjusting the intensity of the blue LEDs.

- The "Mute" button, in the center of the cluster will mute the unit when pressed. Press the "Run" button in the upper left to return to the run mode.
- The vertical buttons with the "up" and "down" arrows will adjust the loading in 10ohm or 10pF increments, respectively; however, the selected values will ***NOT*** be saved in memory. (See **Section 3.4.**)
- The "left" and "right" arrows adjust the Intensity of the blue LEDs. Pressing the left button will dim the LEDs, and pressing the right button will increase their intensity. There are 16-possible LED intensity settings from full brightness to totally off. Holding either button down will scan through these settings in the appropriate direction.
- Pressing one of the gain buttons will change the gain for the chosen input **t**o the selected value and store it in memory—but ***ONLY*** if the unit is in the Mute or Standby mode. (See **Section 3.3**.)
- The phase can be switched between non-inverted and inverted, and back again, using the button in the upper- right marked "Phase."

Figure 3 – Remote Control

There is a small blue LED in the upper-left of the remote which will blink every time a button is pressed, indicating that the remote is transmitting data to the receiver on the front panel of the *Phono III*. If this LED becomes dim, or fails to blink, it is probably necessary to replace the batteries. Using the 1/16" hex key (provided) remove the two screws holding the cover at the bottom end of the remote. Remove the cover and pull out the battery holder. Replace the batteries with two new AAA size batteries and slide the holder back into the remote. Replace the bottom cover and tighten the two retaining screws.

2.0 INITIAL SETUP AND INSTALLATION

This section addresses those setup steps that can be performed with the unit unpowered. Because the initial settings for the gain and internal loading can only be set when the unit is operating, these steps will be addressed in **Sections 3.3 and 3.4**, respectively.

2.1 Connecting the Cables and Selecting Cartridge Type

The cable from each phono cartridge is connected to the respective RCA or XLR input connector for the desired input (A, B, or C) to be used. In accordance with convention, the Right Channel uses connectors with a Red band and the Left Channel uses connectors with either a White or Black band. Note that each input has an associated ground terminal. This should be used only when using input cables that have separate ground leads for the shields. Any ground lead from the turntable should be connected to the ground terminal to the right of the AC power connector.

Since the *Phono III* has a fully balanced input stage, the XLR connector may be used for the cartridge leads; however, the wiring of the cable from the cartridge is critical in this case, or the balanced capabilities of the *Phono III* will not be realized. See **Appendix A** for information on how to properly modify your input cables (or prepare a custom cable) for use with the balanced (XLR) input connector. Failure to use properly wired input cables with the balanced (XLR) input may result in a faulty connection and may also cause an increase in the hum level. Alternatively, a pair of specially wired adapters is provided with each *Phono III* to properly adapt an unbalanced cable for use with the balanced input. These adapters are the **ONLY** adapters that can be used for this purpose. Other commercially available adapters do not properly convert from unbalanced to balanced and will result in the *Phono III* operating in the unbalanced mode, **NOT** the balanced mode. If you are properly using the XLR input connector for your cables, be sure to move the input connector selection switch to the "Bal" position. Otherwise, it should be in the "Unbal" position.

Once the input cables have been connected, the cartridge type for each input should be selected using the switches on the front panel (See **Figure 1B**). The left-most switch for each input is used to select either MC cartridge (resistive loading) or MM cartridge (capacitive loading) for the input in question.[1] This determination should be made before proceeding with the setup. If a MM cartridge is selected, the adjacent switch permits selection of 47Kohms or 100Kohms for the loading resistance. If a cartridge requires a "Custom" load, it should be connected to input "A", because the cartridge type switch for the "A" input has a center "off" position that bypasses the internal loading networks in the *Phono III*. (See **Section 2.2** for instructions on installing a custom load.)

The switches for playback equalization should be checked to ensure that the first of these is in the center "off" position which selects the RIAA equalization curve. Additional information on the use of the equalization switches will be found in **Section 4.4** and **Appendix B**.

The choice of output connector depends on whether balanced or unbalanced output cables are to be used. If unbalanced cables are to be used, they are connected to the Left and Right channel RCA connectors, respectively. Balanced output cables are connected to the respective XLR output connectors. The *Phono III* can drive separate amplifiers using both the balanced and unbalanced outputs, providing that the combined loading impedance does not result in total output currents exceeding 30mA.[2]

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[1] The basis for this selection also applies to moving iron and other types of phono cartridges.

[2] This corresponds to a load of approximately 235 ohms when the *Phono III* is operating at its maximum 7VRMS output level. At lower levels, the allowable loading will decrease correspondingly.

The AC power cord plugs into the connector on the left-hand side of the AC power module on the back panel.[3] This power module also contains the power on-off switch and two line fuses housed in the center section of the module.

NOTE: This module can accommodate either 120VAC or 240VAC supply voltages; however, supply voltages are ***NOT*** interchangeable without internal changes on the power supply board. Unless indicated otherwise, all *Phono III* units are set to operate ***ONLY*** on 120VAC, and connecting to 240VAC will cause failure of the unit. Those units that are set to operate on 240VAC are so indicated by the application of a label indicating use for 240VAC on the back panel.

The fuses are accessible by removing the power cord and then prying the fuse holder out of the center portion of the AC module using a small flat blade screwdriver. (The provided flat-blade screwdriver is too narrow for this purpose and, if used, may be damaged and unable to perform the adjustments in **Section 5.0**). There is a slot to be used for this purpose on the edge of the module adjacent to the location of the power plug. Extra fuses are provided, and are located in small compartments that are integral to the fuse holder. These are accessible once the fuse holder is removed. Be certain to only use replacement fuses that are of the 20 mm size, and 1.5 amp capacities such as the Buss Type GMC fuse. (If you wish, "Audiophile grade" aftermarket fuses may be used provided they are of the same size and amperage rating.)

2.2 Installing a Custom Load

The internal loading networks of the *Phono III* can provide resistive loading for MC cartridges over the range of 10 ohms to 2550 ohms in 10 ohm increments. The capacitance loading for MM cartridges covers the range from 120pF to 1400 pF, in 10pF increments. However, if these ranges are not sufficient, and a custom load is required for a specific cartridge to be installed on input A, proceed as follows.

- Using the provided 2mm slotted screwdriver, undo the two captive screws on the cover plate at the left end of the *Phono III* top panel (see Figure 7) and remove the cover plate. (The two screws are captive and will not come out of the cover plate.)
- Using a small light look down into the unit and what you will see is shown in **Figure 4**. The two circuit boards are the modules for the internally controlled MC cartridge loading. On each board is a pair of pin sockets, shown by the red arrows, for a custom resistor or capacitor to be inserted.
- Having determined the proper value of resistance or capacitance to be used, insert the component leads into the pin sockets on each board.
- Replace cover plate.

The loading thus inserted will be used whenever input A is selected and the cartridge type switch for "A" is in the center "off" position. If this switch is in either the MC or MM position, the custom load will be

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[3] The supplied power cord is a shielded cord; however, users may prefer to use their own power cord having a standard IEC connector.

out of the input loading circuit. Also, when using either input “B” or “C”, the custom load will be ignored.

NOTE: There is a fixed 100Kohm resistance across the input of the *Phono III*.[4] Thus, if the desired custom load resistance is RL, then the actual value of resistance to be inserted, RX, is given by

RX = [RL • 100K] / [100K – RL] ,

(“•” implies multiplication, “K” = 1000) .

Thus, if RL = 10Kohms, RX = [10K • 100K] / 90K = 11.11Kohms .

Based on available resistor values, it may be necessary to use an RX that is not the exact value as calculated above. If a custom capacitive load, CL, is required, the value of capacitance to be inserted, CX, is given by

CX = CL – 120,

where the values of capacitance are given in pF (pico farads = 10^{-12} farads).

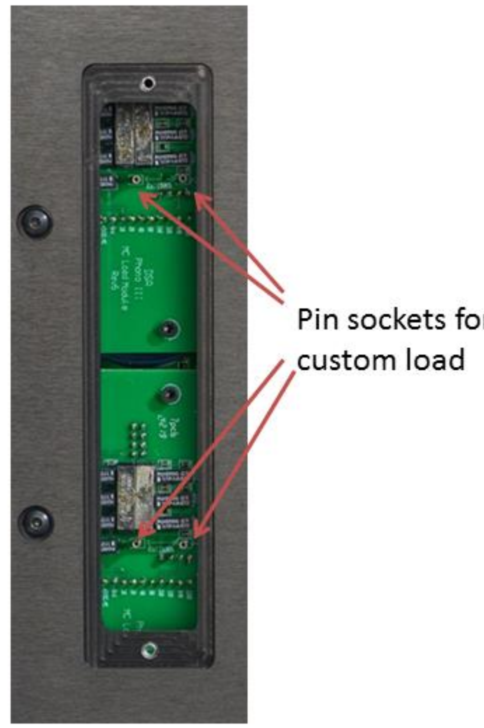

Figure 4 – Layout for Installing Custom Load

3.0 OPERATING THE *PHONO III* AND SETTING THE MEMORIES

You are now ready to finalize some of the initial setup procedures that can be performed only when it is operational, and start enjoying your DSA *Phono III*. The *Phono III* should be placed on a sturdy shelf that will provide adequate support for the unit and permit access to the back panel for cable connections. Clearance at the ends of the unit should be at least 1 inch, and the top panel clearance should be at least 3 inches to permit adequate air circulation for cooling.

3.1 Turning on the Phono III

After connecting the input and output cables, and inserting the power cord into the AC module, you may turn on the *Phono III* using the power switch on the AC module on the back panel. The Standby LED on the front panel will show a red-orange color and will blink at approximately a 1Hz rate. This blinking will continue for about 30 sec during which time the voltages on the internal amplifier boards are ramping up to their operating values. Until this LED has stopped blinking, indicating that the amplifier operating

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[4] The 100K internal load is the same regardless of whether the input is balanced or unbalanced.

voltages have been reached and have stabilized, selecting either Mute or Run will have no immediate effect.

NOTE: If this LED shows either a red or a green color only, it indicates that one of the power supplies is not active. Turn the unit off and contact DSA for instructions.

Once the Standby LED has stopped blinking, the Mute position can be selected by pressing the Mute button on the front panel or remote control. When first moving from Standby to Mute, there will be a 4-6 second delay while the output driver voltages are applied and before the Mute LED will show a light green color. Also, you may observe that the Standby LED will show either a brief green or red color prior to the Mute LED showing a light green color. This is perfectly normal and does not indicate any problems with the *Phono III*. In the Mute mode the *Phono III* is fully powered, but the muting relays at the output are open. This mode represents the mode in which the *Phono III* can be left operating for long periods of time, if desired, and is the mode in which thermal equilibrium will be obtained most readily.

Switching to the Run mode, using the front panel or remote control button, closes the output muting relays and the Run LED will show a blue color. There is no delay between selecting the Run mode and the Run LED turning blue. Switching back to Mute from Run will turn off the Run LED and illuminate the Mute LED again. Switching back to the Standby mode turns off the output stage voltages but not the amplifier board voltages so the Standby LED will glow but will not blink.

NOTE: If either the Mute or Run button is pressed during the initial 30 sec period after AC power on, the *Phono III* will advance directly to the operating mode that was selected once the Standby LED has stopped blinking. However, there will still be the initial 4-6 second delay while the output stage is powered up.

Once operational, the *Phono III* should be left in the Mute or Run mode for about two hours to reach thermal equilibrium. While it can be used and enjoyed during this warm-up period, there is the potential for small DC voltages to appear at the output connectors and some of the switch functions—if activated—may cause unwanted “pops or thumps” in your speaker system. These will be greatly reduced after the *Phono III* has fully warmed up and thermally stabilized, at which point the top panel should be slightly warm to the touch. For those who wish to listen on a regular basis, it is advised that the *Phono III* be left running in the Mute mode since this will maintain the optimal thermal equilibrium condition.

3.2 Front Panel Functions

The functions available on the right-hand-side of the front panel are described in detail below.

Mode: This function toggles between stereo and mono operation. In the Stereo mode, the *Phono III* is a true stereo preamplifier with > 60db channel separation. In the Mono mode, the left and right channels are summed to provide a true mono mode (the identical signal applied to both channels). The mono mode works best with mono LPs, but can also be used when playing stereo LPs if desired. **NOTE: Some mono cartridges will cause excess hum due to internal wiring that creates a ground loop. Selecting the "Mono" mode in this instance will largely cancel this hum.**

Phase: This function selects either normal or inverted absolute polarity of the audio signal. The default condition is non-inverting from the input to either the unbalanced output or the positive side of the balanced output. Repeated pressing of the button on the front panel or remote control toggles between non-inverting and inverting operation. This function can be used in any of the *Phono III*s operating modes.

Filter: This function is either “on” or “off”. The LED is illuminated when the high-pass (HP) filter is “on.” This function is used to reduce low frequency rumble or resonance from the turntable, or other undesirable very low frequency content from the source. When selected, the filter response shown in **Figure 5** is added to the overall response at the selected gain setting and playback equalization curve.

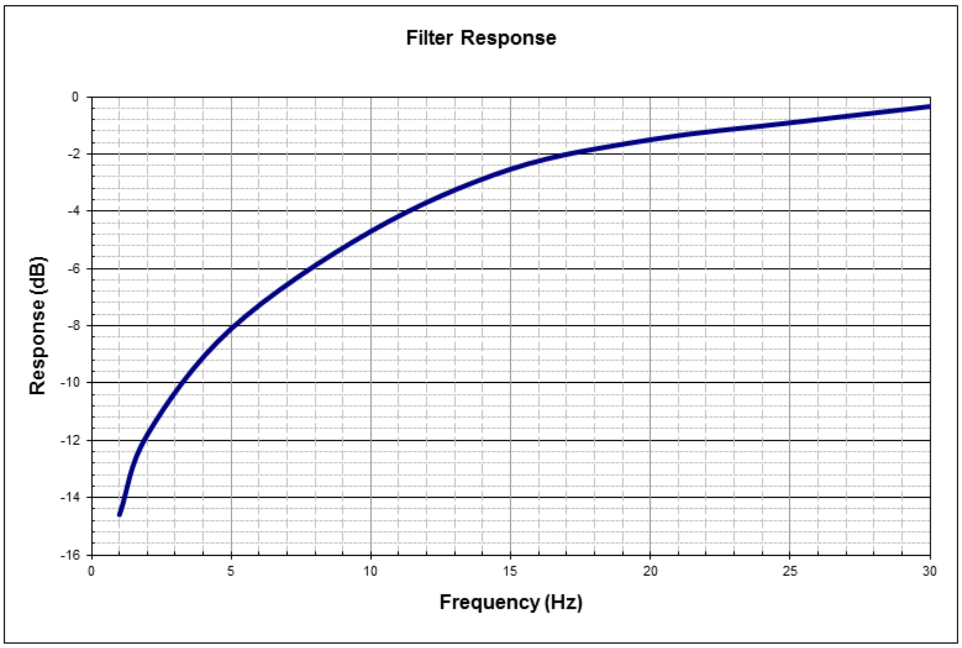

Figure 5 - Relative Filter Response

L-R/R-L: This function operates only when in the Mono mode of operation and it differences the left (L) and right (R) channels as indicated. This function works in conjunction with the phase selection. If the phase is non-inverted, pressing the button will select the L-R function, and pressing it again will revert to mono mode only. If the phase is inverted, pressing the button will result in R-L, and pressing it again reverts to the mono mode. If the unit is operating in the L-R condition, then pressing the phase button repeatedly will “toggle” between L-R and R-L. (See **Section 3.5** for a description of how to adjust the azimuth angle using this function.)

3.3 Setting the Gain Memory

The *Phono III* has a memory that stores the gain setting for each input, and this memory must be “initialized” when the unit is used for the first time. The gain values can ***ONLY*** be set when the unit is in either the Standby or Mute modes of operation. When powered on, the default gain will always show “40dB” until an input is selected. The gain required for each input can be selected, and stored, using the buttons on the front panel, or the remote control. To select and store the gain for each input into memory, proceed as follows:

Note: The Phono III comes with the gain memory set at 40dB for each input. If this is the desired gain value for a specific input, no further action is required.

- Ensure that the unit is operating in the Standby mode (LED not blinking) or the Mute mode. The 40dB gain LED will be illuminated.
- Press the "A" input button and then press the button on the front panel, or remote control, for the desired gain. This stores the desired gain in the memory for the "A" input.
- Repeat the above process until all desired inputs have been assigned a gain value. If the default gain value of 40dB is correct, the 40dB button does not need to be pressed since 40dB is already stored in memory.

Table 1 provides guidance regarding the appropriate gain for various cartridge output levels. If you find that the gain initially selected is too low or too high, place the unit in either Standby or Mute mode and using the front panel buttons, or the remote control, select the new gain for that input. This will store the new value in the gain memory for the selected input. Do not be afraid to experiment with gain settings, but try different ones to get the desired listening level for your system. It is almost impossible to overload the *Phono III*, and even if overloaded, it will not damage the unit.

TABLE 1

	Cartridge output (mV) RMS at 1KHz				
Gain	>3mV	1mV – 3mV	0.5mV – 1mV	0.3mV – 0.5mV	< 0.3mV
40 dB	X	X			
46 dB		X			
50 dB			X		
56 dB			X	X	
60 dB				X	
66 dB					X

3.4 Setting the Cartridge Loading Memory

The internal loading circuits are divided into two sections:

- One section provides the capacitive loading for MM cartridges as well as the 47Kohm/100Kohm selection for resistive load.

- The other is a plug-in module that controls the resistive loading for MC cartridges (this module can be seen in Figure 4).

These modules are controlled by a "master" up/down 8-bit binary counter that is "clocked" either by the "Up/Down" buttons shown in **Figure 1B** or by the "Up/Down" buttons on the remote control shown in **Figure 3**. The loading range and incremental steps are as follows:

- MC cartridge – 0 ohms to 2550 ohms in 10 ohm steps (Using 0 ohms will result in a shorted input and very little, if any, signal.)

- MM cartridge – 120pF to 1400 pF in 10pF steps. Also the resistive load can be 47Kohms or 100Kohms.

For MM cartridges, the minimum 120pF represents the input capacitance of the *Phono III*, and the upper limit is restricted to 7-bits on the counter.

When the *Phono III* is turned on by the user for the first time, the loading memory has all "0s" stored for each input, so it is necessary to provide "baseline" loading data for each input.[5] The baseline data can be set in any of the operating modes (Standby, Mute, Run). To store the "baseline" data, proceed as follows:

- With the unit turned on, and in either the Standby or Mute mode, select an input to be used. (You have already selected the cartridge type in **Section 2.1**).
- The display will show "0"ohms if you selected a MC cartridge, or "120"pF if a MM cartridge was selected. If a MM cartridge was selected, the selection of 47Kohm or 100Kohm resistive loading should have also been made.
- Using either the "Up" button on the front panel, shown in Figure 1B, or the "Up" load button on the remote control, shown in Figure 3, increase the loading to a "baseline" value as recommended by the cartridge manufacturer, or based on previous listening experience. (If you overshoot the desired value, use the "Down" buttons to make corrections.) This will be the "baseline" loading value for that input.
- Press either the "Up" or the "Down" button on the front panel to store the "baseline" loading value. **NOTE:** The "baseline" loading data is ***<u>ONLY</u>*** stored using the buttons on the front panel. Using the up or down buttons on the remote control will **<u>NOT</u>** store any data in the loading memory. (See following paragraph for using the load buttons on the remote.)
- Repeat the above process for the other inputs that will be used.

Having put the "baseline" data into the loading memory, it is now possible to use the remote control to make loading adjustments while listening and not alter the "baseline" data. This permits the user to make minor changes in the cartridge response for the purposes of enhancing the listening experience for a particular recording by using the "Up"/"Down" buttons on the remote without changing the "baseline" data. To return to the stored "baseline" loading value at any time, press the "mute" button and then the input selection button for the input being used, followed by pressing the "run" button. If the listener believes that an improved "baseline" loading was obtained by using the buttons on the remote while listening, it is only necessary to use either the "Up" or "Down" button on the front panel to store the newly obtained loading value for that input (pressing "mute" is not required).

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[5] The "baseline" data represents the load values that will be stored in memory for each input/cartridge combination based on manufacture's recommendations or listening experience. This value can be easily changed if listening experiences offer a better value.

Note: Using the "reset", shown in Figure 1B, will "zero out" the "baseline" loading values in the memory for the selected input.

3.5 Adjusting Cartridge Azimuth Angle

The "L-R/R-L" function can be used to adjust cartridge azimuth alignment; however mono source material is required for this adjustment. The approach relies on the differencing of the signals from the left and right channels. Azimuth alignment is set when there is zero (or equal but very low) signal in both the L-R and R-L settings.[6] The steps to be followed are as follows:

1. Place a monophonic record on the turntable
2. Switch the *Phono III* to the mono mode
3. Push the "L-R/R-L" button and play the record
4. While "toggling" the phase button, listen to the speaker output and adjust the azimuth angle of the cartridge to obtain a minimum sound level in each position (L-R and R-L).
5. When satisfied, switch back to the stereo mode.

4.0 DESIGN PHILOSOPHY AND IMPLEMENTATION

The solitary goal of the Dynamic Sounds Associates (DSA) *Phono III* is to provide the finest possible reproduction from LP recorded media. To achieve this goal, the *Phono III* is based on a "no-compromise" dual channel design using best engineering principles and the finest of components. The *Phono III* does not employ any form of loop feedback to achieve the desired throughput gain or the proper playback compensation. Instead, each gain stage of the *Phono III* has internal feedback to ensure that all forms of distortion are held to very low levels. The playback compensation is provided through the use of passive low-pass networks, with the proper time constants, located between the gain stages of the amplifier chain. This approach provides low distortion plus a very high dynamic range. It also eliminates transient inter-modulation distortion, which is a common byproduct of configurations where the throughput gain and playback compensation are achieved through the use of inter-stage loop feedback.

Furthermore, because even the finest of coupling capacitors can cause minor, but perceptible, degradations in the reproduction of the audio signal, the *Phono III* has no coupling capacitors in the audio chain from input to output. Yet, through the use of an innovative design it remains very stable and resistant to DC drifts. Additional advantages of the design approach are the ability to provide easy polarity inversion of the audio signal and a true monophonic capability for the proper reproduction of monophonic LPs. Combining these two capabilities leads to the L-R and R-L capability in the mono mode which is used, as described, in **Section 3.5**.

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[6] There are other methods for adjusting the azimuth angle that rely on minimizing channel cross talk. The use of these approaches may be beyond the capabilities, or resources, of many audiophiles; and, we believe that the approach described in Section 3.5 will prove satisfactory in most instances.

5.1 Amplifier Chain

The block design of the *Phono III* is shown in **Figure 6**. It consists of four all JFET balanced (differential) amplifier gain stages, two of which can be adjusted to provide the required gain to accommodate virtually any phono cartridge. In addition, each gain stage employs its own precision, temperature compensated, low noise constant current source and voltage regulator for the ultimate in stability, signal control, and isolation.

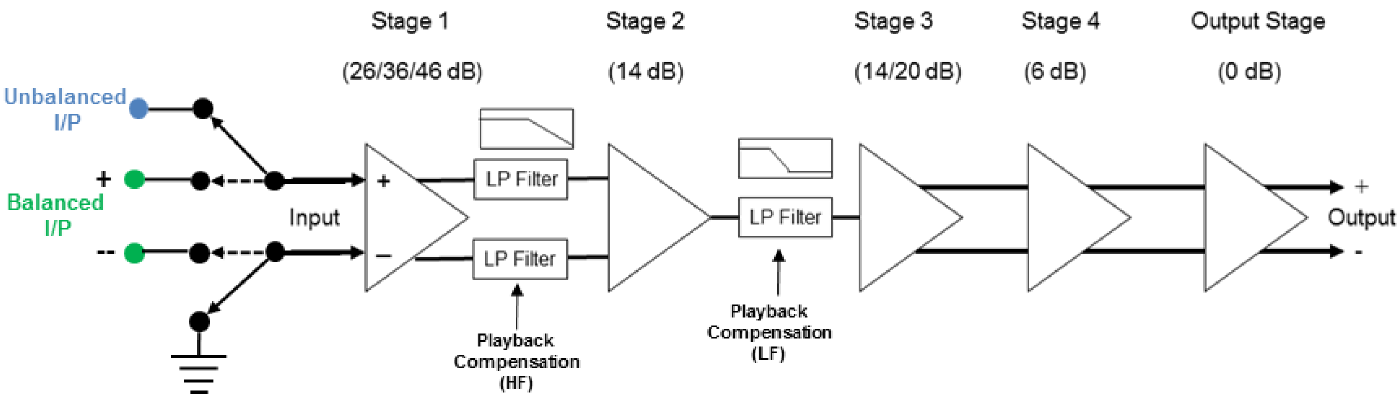

Figure 6 – *Phono III* Block Diagram

As can be seen from **Figure 6**, the first and second gain stages, as well as the third and fourth gain stages are differential pairs. The fourth gain stage is also connected differentially to the unity gain output driver which provides a balanced output through a standard XLR connector and a single-ended output via a standard RCA type phono jack. The two switches shown at the input represent the process that allows the input stage to accept either unbalanced or balanced inputs. The solid lines represent the unbalanced mode configuration and the dashed lines the balanced configuration.

Using two differential pairs of gain stages has significant advantages for noise reduction which will be discussed in more detail in **Section 4.5**. The downside of differentially coupling the first and second gain stages is that two matched LP filters are required for the HF component of the playback equalization curves. However, the filters are of a simple design and can be easily balanced to ensure proper playback equalization. The balancing and calibration of these filters is performed by DSA during the “break-in” period after a new unit is assembled and is not a user accessible adjustment. Once balanced and calibrated, there is no requirement to perform any further adjustments on these filters.

While the *Phono III* has no coupling capacitors in the signal path, it still does not have any significant response at DC which prevents the propagation of DC drifts and offsets. This “DC block” occurs at the input to the third gain stage, the first stage in the second differential pair shown in **Figure 6.** This “DC block” combined with the gain dependent low frequency response of the first stage results in a frequency “roll-off” at very low frequencies as shown in **Figure 7.** This very low frequency roll-off is relative with respect to the absolute gain and any playback equalization being used. If the filter function has been activated, that response will be added to the frequency response shown in **Figure 7**.

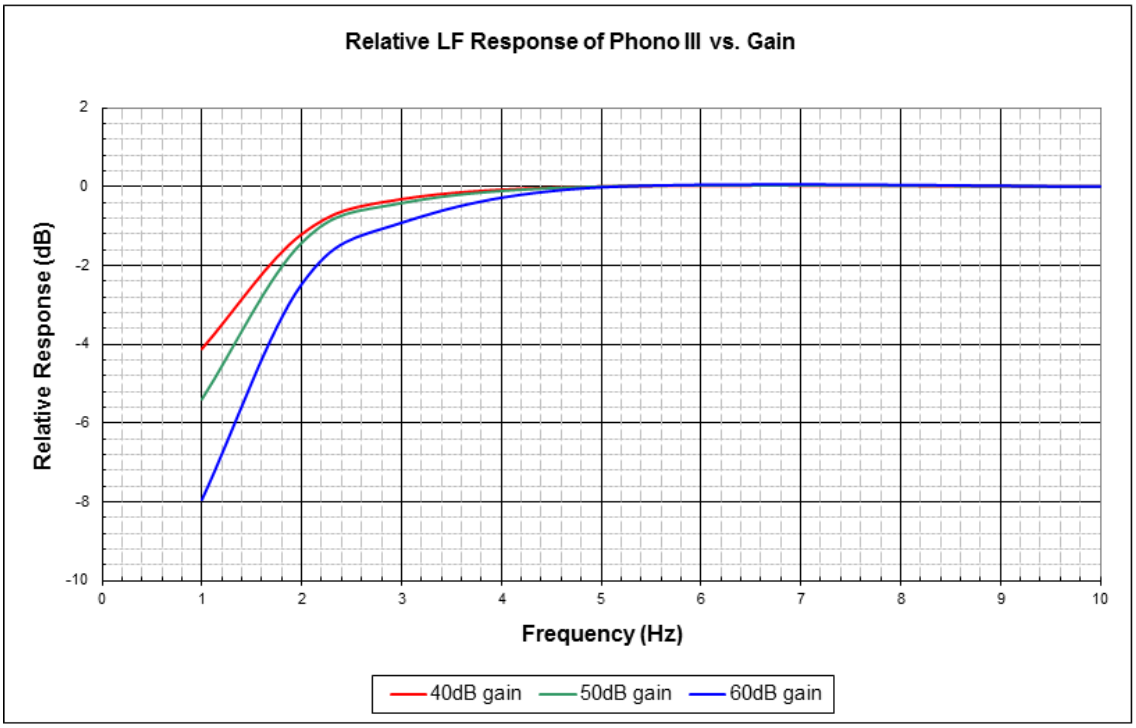

Figure 7 – Relative Low Frequency Response of the *Phono III*

4.2 Output Driver

The *Phono III* uses a separate high bias current Class A output driver for each polarity of the amplified audio signal. The output stage supply voltages are fully regulated and are powered separately from the voltage rails that power the amplifier chain of the *Phono III*. The output impedance for each polarity at the output connectors is 75 ohms.

Because of the high rail voltages, and the inherent linearity of the *Phono III* design, it is capable of providing an output drive voltage of >20 volts peak-to-peak without any clipping of the audio signal. The output stages of the *Phono III* can provide up to 30mA of drive current without distortion, and can drive amplifiers with input impedance as low as 1kOhm, or long cable runs that have a capacitance of greater than 30nF, without any problems[7].

4.3 Power Supply

The *Phono III* employs a dual, fully regulated power supply, and the individual supplies are isolated from each other. The power supply for each channel provides ± 60V rail voltages for the operation of the

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[7] This is based on a 20kHz signal at 30V p-p. At lower frequencies or drive levels, the *Phono III* can drive significantly higher values of capacitance without difficulty.

amplifier section as well as separate rail voltages of ± 47V for the associated output stage. In addition, it also generates regulated ±12VDC for internal use by the regulator, and +5VDC that is used to operate all of the control functions.

To accommodate the fact that the *Phono III* contains no coupling capacitors within the audio path, and to prevent damage to components that could occur if the rail voltages were applied suddenly, the *Phono III* power supply is designed with an approximate 30 sec ramp from about ±5VDC at turn-on to the full ±60VDC. As the voltage slowly increases, the value of both positive and negative rails are monitored by comparison circuitry in the power supply. Only when both rails have achieved the proper final values of ±60VDC, and the regulator section has "clamped" indicating that it is in the fully regulated mode, is a turn-on signal generated that permits the output stage to be turned on when the "Mute" or "Run" button is pressed. In a similar manner, the voltage rails for the output stage are monitored to ensure that they are correct before the "Run" function can be enabled. This prevents operation in the event of a failure within the power supply or amplifier board that could result in applying a large DC bias to the output connectors.

4.4 Playback Equalization

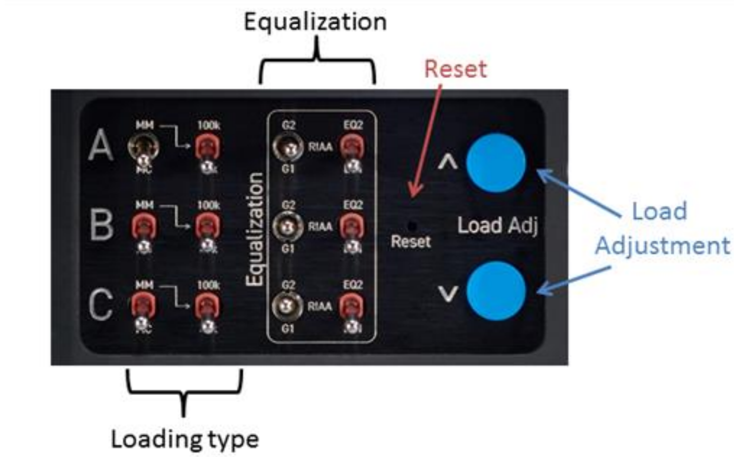
As seen in Figure 5, the playback equalization is divided into two parts. The high frequency (HF) (>1kHz) compensation is located between Stages 1 and 2, and the compensation for frequencies <1kHz is located between Stages 2 and 3. The compensation is designed in this manner to ensure that HF signals from the cartridge which enter the preamplifier at levels greater than those at lower frequencies are correctly attenuated before passing to the other gain stages of the *Phono III*. This approach ensures high dynamic range over the full frequency spectrum as well as significantly reducing the output noise power. Because of the inherent flexibility of this approach, it was decided to incorporate additional playback equalization curves in addition to the RIAA curve which has become the recording industry standard. **Appendix B** provides an in-depth discussion of playback equalization curves and information on their incorporation into the *Phono III* architecture.

As shown in **Appendix B**, the playback equalization curves are determined by a set of three "time constants", where each time constant can be represented by a combination of a resistor and capacitor of specific values. **Table 2** (also **Table B.1**) shows the time constants for the equalization curves that have been incorporated into the *Phono III*, and the recording labels that they are associated with.

TABLE 2

GROUP	EQ Curve	NAME	t1	t2	t3
---	--	RIAA	3180 µsec	318 µsec	75 µsec
G1	**EQ1**	Columbia-Pre RIAA	1590 µsec	318 µsec	100 µsec
G1	**EQ2**	Decca/FFRR - Pre RIAA	1590 µsec	318 µsec	60 µsec
G2	**EQ1**	Columbia 78RPM	---	530 µsec	100 µsec
G2	**EQ2**	Decca/FFRR 78RPM	---	530 µsec	25 µsec

The choice of equalizations based on Columbia and Decca/FFRR was based largely on their early dominance in the production of the LP record, and that information regarding the equalizations they used during the recording process was available in the current literature. The recording equalization used by RCA in the early years of LP development became the RIAA equalization with the addition of the t1 time constant which flattens the LF rise starting at 50Hz.



From **Figure 1B**, shown here for reference, the switches within the "Equalization" section on the center-lower section of the *Phono III* front panel allow selection of the playback equalizations in **Table 2**. The first switch in the section has three options: RIAA (center "off"), G1 (down), or G2 (up). The second switch selects either EQ1 or EQ2 within either group G1 or G2. The "EQ Group" LED shown in Figure 1A will be blue when the RIAA equalization is selected, it will be green with group G1, and red with group G2. (These colors are selected to alert the user that non-RIAA equalization is being used.) Likewise, the EQ1 or EQ2 LEDs will illuminate depending on which is selected within the equalization group G1 or G2. The EQ1 and EQ2 LEDs will not illuminate if the RIAA equalization is selected**.**

Appendix B presents design curves and typical measured responses of the *Phono III* for all of the equalization curves shown in Table 2. It will also show how the different curves were incorporated within the *Phono III* architecture. Precision matched components are used for all of the equalization curves. As seen in **Appendix B,** the measured responses for the RIAA and group G1 curves are accurate to ≤ **±** 0.2 dB from 20Hz to 20KHz and, at many frequencies, the error is <**±** 0.1 dB. **Appendix B** also shows that the curves of group G2 are accurate to **±** 0.2 dB from 100Hz to 20KHz, but are low by < 1dB at 50Hz, 2.5-3dB at 20Hz, and are low by ≈ 8dB at 10Hz.

Note: When the G1 or G2 curves are selected, the overall gain of the *Phono III* will change by ± 1-2dB relative to the RIAA equalization gain, depending on which EQ selection is made. This is to be expected and is not a fault of the unit.

NOTE: Charts with playback equalization measurements made on this particular unit are found at the end of Appendix B.[8]

<u>A General Observation</u>**:** To meet special requirements, it is possible that other equalization curves can be incorporated in lieu of the four that come as standard in the *Phono III*.

4.5 Noise Levels

There are three basic sources for noise at the output of the *Phono III*:

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[8] The measurements are performed using the CLIOwin7 measurement system which is calibrated prior to making each set of measurements. CLIOwin7 is a registered Trademark of Automatica.

- AC power supply noise which has a "noise floor" plus minor noise spikes at harmonics of 60Hz,
- External interference that is picked up and amplified by the *Phono III*,
- Semiconductor and component noise from the internal circuitry.

To suppress the AC power supply noise, the *Phono III* uses two matched shielded toroidal transformers, whose primaries are driven out of phase from each other. These transformers are also encased in a separate magnetically shielded housing to provide virtually total cancellation of residual AC fields within the *Phono III* chassis. The dual power supplies are highly regulated and filtered to ensure that the voltages going to the amplifier boards are free of AC noise and harmonics of 60Hz. The use of individual low noise voltage regulators for each gain stage further reduces the AC noise to levels that are almost unmeasurable.

External noise is suppressed by having the *Phono III* enclosure designed as a "Faraday cage," a fully conducting metal cage that is connected directly to the AC ground. The individual panels of the Phono II chassis have the black anodizing removed on the inside surface where required to ensure that all panels are electrically connected for isolation. In addition, the *Phono III* uses an internal RFI filter on the AC power line to eliminate residual power line interference that might enter via that pathway.

Internally generated noise from the amplifier components is the dominant source of wideband output noise in the *Phono III*. However, due to both the design topology, and the use of ultra-low noise, discrete designs for the voltage regulators and constant current sources for each gain stage, the overall noise level at the output is very low. First, any residual internal noise generated by the voltage regulators or constant current sources for stages 1 and 3 is eliminated by the balanced connection of these stages to stages 2 and 4, respectively. This leaves stage 2 as the only significant noise source since any residual noise generated in stage 4 has no further amplification and will be dominated by any noise from the preceding stages.

All gain stages use ultra-low voltage regulators having ≈ 20nV/Hz $^{1/2}$ of voltage noise at their respective outputs, and the constant current sources for stages 2 and 4 have ≈ 20pA/Hz $^{1/2}$ of current noise at the required bias current levels.[9] However, just as the total bias current is split equally between the two sides of the balanced stage, this noise current is also equally split. Randomly generated, or "uncorrelated", noise generated by the components within the gain stages is not suppressed by the balanced connection to stages 2 and 4; but, through the use of ultra-low noise devices for the gain stages, analysis and measurements have shown that this noise is significantly less than any "common mode" noise.

In addition to designing very low noise gain stages, internally generated noise is further suppressed by the RIAA (or other) playback equalization. **Figure 8** shows the un-weighted noise voltage spectral density (V/Hz $^{1/2}$) for the *Phono III* at 40dB, 50dB and 60dB gain, as measured[10] at the single ended output and with the input shorted. All curves are for the left channel, with identical results for the right channel.

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[9] Analysis and measurements have shown that constant current noise is the major source of noise within each gain stage since it is multiplied by the drain resistance for each amplifier stage.

[10]

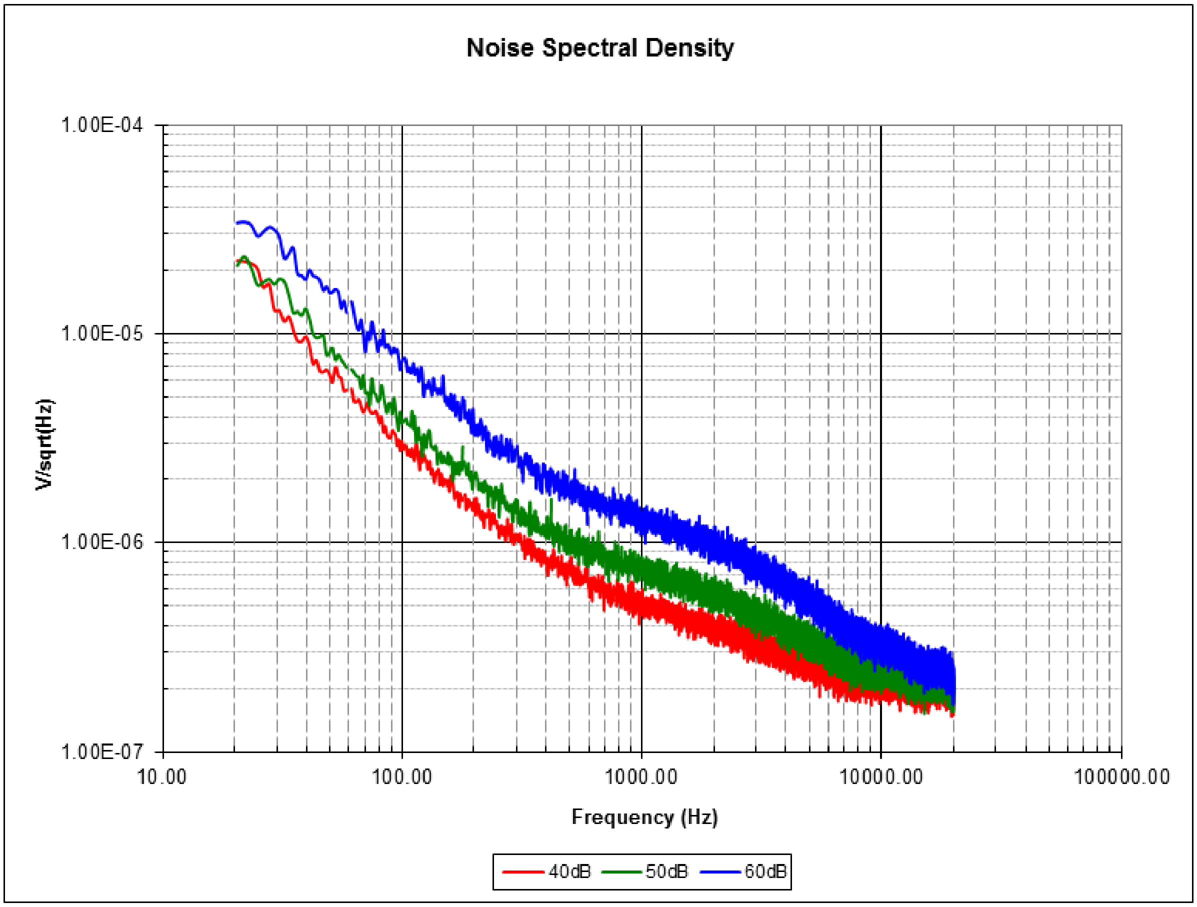

Figure 8 – *Phono III* Noise Level from 20Hz – 20kHz

A complete analysis of the total noise power is used to specify the total noise level of the *Phono III* given in the specifications at the end of this manual.

5.0 BALANCING AND OTHER ADJUSTMENTS

The *Phono III* comes from DSA fully "broken in" and balanced under normal thermal operating conditions. Testing has shown that the circuitry of the *Phono III* is stable for over a year and that additional balancing is generally not required. However, through long periods of use, or use in extreme thermal environments, some rebalancing may be required. The need for such adjustment will become apparent when use of the polarity function, or connecting the *Phono III* output to a line amplifier, results in an excessive "thump." (A small "thump" may occur and is a normal occurrence when selecting the *Phono III* output and does not indicate a problem.)

The location of the balance adjustments is shown in **Figure 9**. These adjustments should be made with the top cover in place because removing the cover will change the thermal equilibrium conditions.

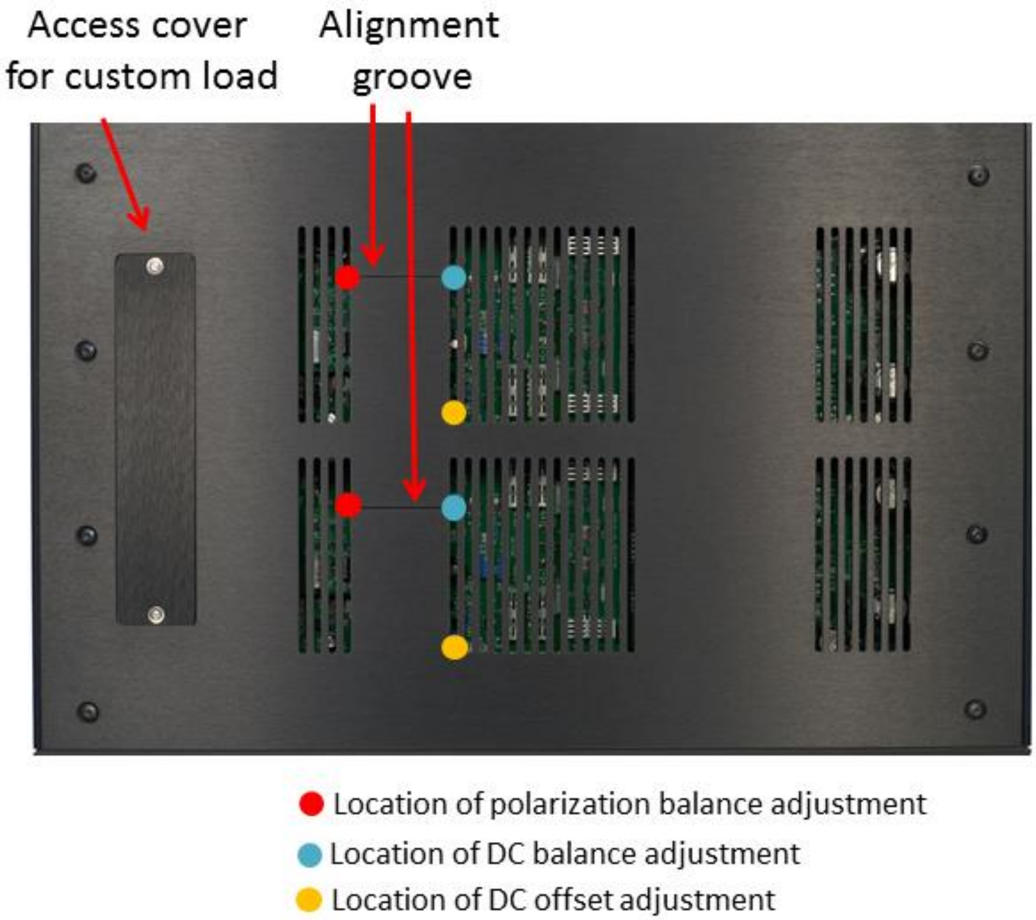

Figure 9 – Location of Balance Adjustments

The adjustments described below should be made with the *Phono III* on and fully operational. To ensure thermal stability, it should be left in "Mute" mode for several hours prior to making these adjustments. If, for any reason you do not wish to make these adjustments yourself, the unit may be returned to DSA for these adjustments. Note the "Alignment groove" which will assist in locating the adjustment potentiometers on the amplifier boards.

NOTE: Making these adjustments can be facilitated by turning on the filter function (button on the right-front panel) because it will eliminate any infrasonic noise that may interfere with a good voltage reading.

5.1 Excessive "Pop Noise" When Selecting the Phono III Output

This situation implies that the DC balance, and/or offset, at the outputs have drifted out of the factory specification. This adjustment uses the #00 Philips head screwdriver and female XLR plug with three wires that are provided with the unit. Also, a DC voltmeter is required, but this is not provided.

After a period of warm-up to ensure thermal equilibrium, insert the female XLR plug into the left channel XLR output socket. Using the DC voltmeter, measure the voltage between the red and blue wires; the voltage should be less than ±10mV. If it is greater than ±10mV, insert the provided 2mm screwdriver through the top panel slot where indicated for the left channel DC balance adjustment. The screwdriver will engage a small adjustment potentiometer that is below the indicated location (a small flashlight may be needed to see the control below). Once engaged, rotate the screwdriver very slightly to obtain “0 volts” (typically ± 10mV) between the two wires.

Once this balance has been achieved, connect the voltmeter between the black wire and the red wire. Insert the screwdriver in the bottom of the same slot where the DC offset adjustment potentiometer is located. Using the screwdriver, adjust the control in a similar manner to obtain a reading of “0 volts” (typically ± 10mV) on the voltmeter. You have now balanced the output of the left channel.

Repeat the above process for the right channel, first moving the female XLR plug to the right channel output socket.

5.2 Excessive “Pop Noise” When Changing Polarity

This adjustment is rarely required; however, if it is the polarization balance function is activated using the small toggle switch on the back panel located between the two sets of output connectors, as shown in **Figure 10**. Moving this switch to the right, toward the “Pol Bal” text will enable the polarization balance to be performed. (In the normal mode of operation this switch should be to the left, which disables the polarization balance function.)

To the right of this switch are two small holes labeled “L” and “R”. There is a socket located behind these holes and the provided ¼ watt resistor will plug into the socket. (When the lead goes into the socket there will be some slight resistance and then the lead will slide in.)

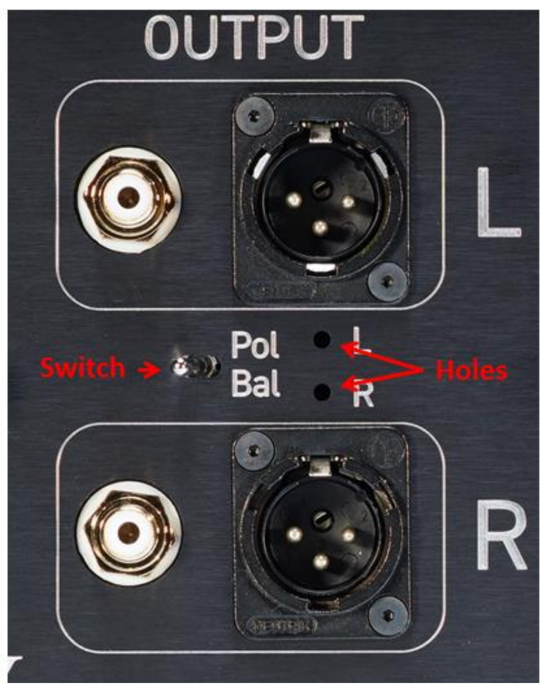

Figure 10 – Location of Polarity Balance Switch

First plug in the XRL female connector used in **Section 5.1** into the right channel XLR output socket. Then, find the ¼ watt resistor provided with the unit and insert the short end into the upper (“L”) of the two holes shown in **Figure 10**. (When properly engaged in the socket, the outer end of the body of the resistor will be almost flush with the outside surface of the back panel.) After a sufficient period of warm-up, connect a DC voltmeter between the protruding resistor lead and the black lead on the female XLR plug. Measure the voltage, press the front panel button to invert the polarity, and measure the voltage again. The two readings should be within ±10mV of each other. If not, insert the provided 2mm screwdriver through the top panel slot in the location indicated in Figure 9 for the left channel polarity balance adjustment. As in **Section 5.1**, adjust the polarity balance potentiometer such that the voltages in the non-inverted and inverted polarity positions are within ±10mV. Then remove the resistor and place it in the lower (“R”) hole, again engaging the socket behind the hole. Repeat the voltage measurement between the protruding resistor lead and the black wire on the female XLR plug and adjust the polarity balance potentiometer for the right channel as before.

When these adjustments are completed, remove the resistor and the XLR connector and return the polarity balance switch to the "off" position (to the left). This completes the balancing adjustments that may be required occasionally.

How often these adjustments may be required will depend on the amount of use that the DSA *Phono III* receives, the operating environment, and the overall age of the unit. It will become obvious that adjustments are required when changing polarity or going between Run and Mute results in excessive "thumps" from the loudspeakers. Within the first year of ownership, if such balancing is required, it is offered as a warranty service through DSA.

6.0 SPECIFICATIONS

Design Topology:	
Gain stages:	**4 - DC coupled, balanced-differential**
Output Stage:	**Class-A, balanced-differential, push-pull**
Playback Compensation:	**Passive, 2-stage LP filter**
Internal Memory	**2 x 16K, 8-bit non-volatile (>5 year storage)**
AC Voltage:	**120 VAC (240VAC option)**
Fuse Type and Rating:	**2 x Buss GMC (20 mm) 1.5 A**
Dimensions:	**17" (W) x 11-1/2" (D) x 5" (H)**
Weight:	**30 lbs**
Cartridge Loading:	
Moving Magnet:	
Load Resistance	**Selectable: 47Kohms or 100Kohms**
Load Capacitance	**Selectable: 120pF to 1400pF in 10 pF steps**
Moving Coil:	**Selectable: 10 ohms to 2550 ohms in 10 ohm steps**
Custom Loading:	**Available on Input A**
Gain referenced to 1kHz (RIAA equalization):	**Selectable: 40dB, 46dB, 50dB, 56dB, 60dB, 66 dB**
Channel Separation:	**≥ 60 dB**
Playback Equalization:	
RIAA Accuracy	**± 0.2dB from 20 Hz – 20KHz**
Group 1 Equalizations (See Section 3.5)	**± 0.2dB from 20 Hz – 20KHz**
Group 2 Equalizations (See Section 3.5)	**± 0.2dB from 50 Hz – 20KHz**
Max Output Voltage:	**20 volts peak-to-peak (7 VRMS)**
Output Impedance (balanced or unbalanced):	**75 ohms**
Output Current:	**Max. 30 mA (for Class A operation)**
Unweighted Signal to Noise ratio measured at the unbalanced output: (Noise Bandwidth = 20Hz – 20kHz)	
40 dB gain (relative to 10mV input at 1kHz)	**80 dB**
50dB gain (relative to 3.16mV input at 1kHz)	**80 dB**
60 dB gain (relative to 1mV input at 1kHz)	**75 dB**
	Note: Using the Balanced output may improve these SNR values by as much as 10dB

WARRANTY

All DSA products carry a three (3) year warranty against defects in material, components, and workmanship. This warranty also includes the balance adjustments described in Section 5.0, if required, during the first year of service. This warranty becomes effective on the date of purchase, or the date of shipping, whichever is later. To ensure proper registration of the product, and to validate the warranty, it is necessary to return the warranty registration card below. (This card may be scanned and e-mailed to info@dynamicsounds-assoc.com if preferred.) Under the terms of the warranty, repairs and/or adjustments will be made at manufacturer's cost, including return shipping to the user during the warranty period. The user is responsible for shipping costs to the manufacturer for warranty repairs. Charges for unauthorized service and shipping are not covered under this warranty. This warranty is null and void where it is apparent that misuse, accident, neglect, and tampering with or modifications by other than DSA have damaged the product. The warrantor assumes no liability for property damage or any other incidental or consequential damage whatsoever which may result from a failure or misuse of this product.

Prior to returning any product for warranty repairs, or adjustments, it is necessary to obtain return authorization from DSA. You must obtain authorization by sending an e-mail to support@dynamicsounds-assoc.com. Identify the product, the serial number and provide a brief description of the problem with the product. You will receive authorization by return e-mail.

DETACH THIS PORTION AND SEND TO DSA TO COMPLETE REGISTRATION

Dynamic Sounds Associates, LLC.
1754 Persimmon Ct.
Naples, FL 34109

MODEL_____________________________ SERIAL No. ___________________________

PURCHASE/SHIPPING DATE ____

NAME OF OWNER ____

ADDRESS ____

CITY, STATE, COUNTRY, ZIP ____

TELEPHONE No. ____

E-MAIL ____

APPENDIX A

This Appendix addresses the proper ways to connect a phono cartridge to the input of the *Phono III* when using the XLR (Balanced) input connectors.

To understand the use of the balanced input (XLR) with a phono cartridge, it is necessary to realize that the cartridge is NOT a balanced generator—if so, it would have three outputs per channel (+, ground, -). Instead, it is a “floating” generator with no fixed reference point for the output voltage. Typically, one terminal for each channel is identified as “ground” but this is totally arbitrary. The cartridge generates a voltage between the two output terminals for each channel independently and without any ground reference. Such a floating generator can be used to drive either a “single-ended” amplifier which has only one input that is referenced to ground, or it can drive a “balanced” (or differential) amplifier where the generator is connected between the two inputs without a ground reference. The input stage of the *Phono III* is designed as a balanced amplifier that can perform as either a single-ended amplifier by connecting the “negative” input to ground (Unbalanced), or in the balanced mode by having the cartridge connected between the two inputs (Balanced). The balanced configuration increases the voltage gain of the signal at the input by a factor of 2 (+6dB); but, since the cartridge voltage is now split between the two inputs, each one seeing only ½ the total voltage (-6dB), the overall gain is the same in either the balanced or unbalanced configuration.

Figure A-1 shows the typical connections for each channel to the phono cartridge and the black lines indicate the typical cable from the tone arm to the phono preamp consisting of a single shielded wire with the shield being the “ground” lead, and terminated with an RCA male connector. Some “higher end” cables may have two conductors and a shield with the shield and the “ground” wire for the channel connected to the shell of the RCA connector.

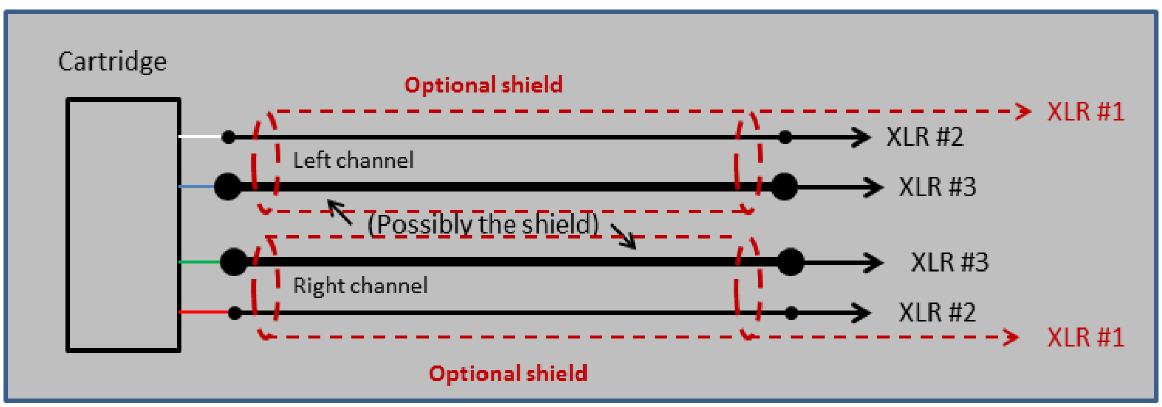

Figure A-1

Before proceeding it is necessary to ensure that the “grounds” for each channel are not interconnected at the turntable end. With the cables disconnected from the phono preamp, use an ohm-meter to check the resistance between the “shells” of the two RCA connectors coming from the tonearm. The meter should show either an open-circuit, or a very large resistance (> 1Mohm). If this is not the case, then using the XLR inputs on the *Phono III* will

result in an unbalanced condition and not a balanced one. The XLR connectors can be used, but the amplifier input will be operating as if the unbalanced input had been selected.

If the ohm-meter check shows an open circuit or high resistance, then you can proceed to install an XLR connector for use with the balanced inputs on the *Phono III*. Proceed as follows:

NOTE: This procedure will require some familiarity with soldering techniques, a soldering station and solder, plus the necessary tools.

1. Remove the RCA connectors at the phono preamp end and strip the outer insulation back to reveal the internal wires and the shield.
2. If only a single wire and shield are present, connect the single wire to pin #2 of the male XLR connector and the shield to pin #3 of the XLR connector. (Pin #2 is the "+" input and pin #3 is the "-" input.)
3. If there are two wires and a shield—or in some cases only two wires and no shield—it is necessary to determine which wire is "+" and which is "-". If possible remove the shell of the detached RCA connector and see which wire is connected to the center pin---this wire is "+", and the other is "-". IF this is not possible, using the ohm-meter see which wire goes to the white (left channel) pin on the phono cartridge and which goes to the red (right channel) pin. These wires will be connected to pin #2 on each of the male XLR connectors. The other wire will be connected to pin #3 on both connectors. If a shield is present, it should be connected to pin #1 (ground) of the XLR connectors.
4. Complete the assembly of the XLR connectors being sure to label them for left and right channel.

The big advantage of using the balanced input is that any external interference that is picked up in the cables from the phono cartridge to the phono preamp is cancelled by the balanced input of the preamp. For this reason an external shield is not required for the cables from the tonearm to the phono preamp when using a balanced input. In cases where the turntable is operating in a high RF noise environment—or even 60Hz—using the balanced input can be very useful.

APPENDIX B

EQUALIZATION CURVES FOR LP PLAYBACK

B.1 BACKGROUND

In the early days of recorded music on 78RPM disks, the recording engineers were confronted by the problems of maximizing the amount of recorded material on the disk, and the need to avoid compressing, or otherwise losing information during the production of the disk. The problem was that, when looking at a constant recorded volume (or level), the low frequencies (LF) required large excursions of the cutter head and high frequencies (HF) required lesser excursions. Those LF excursions cut into the amount of available space between the recorded grooves and reduced the amount of recorded information that could be cut into a single side of the disk. Their desire was to have a uniform spacing of the grooves, not only for maximizing the amount of material on the disk, but it also made the design of the cutting lathe much easier by controlling the extent of its excursions. The solution was to pass the signal to be recorded through a "weighting filter" which reduced the amplitude of the LF signals, and increased the amplitude of the HF signals. This served to equalize the overall amplitude excursions of the cutter head, and permitted an efficient "packing" of recorded information on the disk. The downside of this approach was that the inverse "weighting" had to be applied during the playback process by the listener so that the audio signal matched the original.

The result of this development was the establishment of "equalization curves" (or EQ curves as they became known) to be used for both recording and playback. (The playback curve is the inverse of that used during the recording process.) Different record labels had different EQ curves for their records, and there was no "standard" EQ that was applied. This lack of a standard EQ resulted in the developers of playback equipment adding tone controls and even sets of switches for adjusting portions of the audio spectrum in an attempt to match the EQ curve used in the record fabrication.

This process worked OK during the days of monophonic recordings, since the primary concern was that the frequency responses were reasonably well matched at the midpoints of the LF and HF regions--hence you will see playback curves defined by "+10dB at 50Hz and -12dB at 10kHz." However, with the later introduction of stereo recordings, the matching of the curves would become much more important. This was because frequency weighting, when applied to the recorded signal, introduces non-linear variations in signal phase across the frequency spectrum that were "baked in" to the record. On playback, the frequency weighting had to accurately match the inverse of that used in the recording so that those phase variations could be properly corrected as well. If not, then the sound during playback would lose "focus" and the stereo effect would not be optimal.

To ensure that both recording engineers, and designers of playback electronics, were "working from the same sheet of music," in the late 1950's the Recording Industry Association of America (RIAA) promulgated a pair of "standard" recording and playback equalization curves--known as the RIAA curves. These curves were defined by three "time constants" [11], which defined more

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[11] Identified as t1 =3180μsec, t2 = 318 sec, and t3 =75μsec (μsec = 10^{-6} sec)[3]

rigorously the overall shape of the equalization curves. Around 1955, the RIAA equalization curves became the standard for US recordings, while it became a European standard by 1962. [4] Today, it remains the industry standard world-wide for LP disk recordings. However, recordings that were produced prior to the universal adoption of the RIAA standard have been a subject of much research and study over the years.[12]

B.2 BEFORE RIAA

Before the LP format, recording engineers were using equalization in the recording process, but there were few if any standards and many of the equalization curves used were unique to a particular label. Some of them continued to use the equalization developed for 78 RPM records when recording in the new LP format, first introduced by Columbia in 1948[4], until developing a new equalization that was more appropriate for the new LP format. Fortunately, by 1950 most major record labels had adopted an equalization process for recording in the new format; unfortunately, they were not all the same, and in many instances--not well documented.

B.2.1 Late 1940's to Early 1950's

During this period, record manufacturers were slowly transitioning from what they had been doing for their 78 RPM products to the new "microgroove" LP format. It was largely a time of experimentation as they learned to work with the new format, which meant changing cutter technology as well as new electronics and record stamping processes. They were still producing 78 RPM disks, but the allure of a longer playing format was causing a revolution in disk technology. The part that was slow to change was the equalization needed to optimize the use of the new LP format. An example of this was Decca/FFRR which started producing in the LP format, but used their old 78 RPM EQ curve up to about June 1950.[13] Both Columbia and RCA also had 78 RPM EQ curves that differed from that used by Decca/FFRR, but it appears that they stopped using these prior to 1950. Based on available literature[1], **Figure B.1** shows the 78 RPM playback EQ curves for Columbia, Decca/FFRR, and RCA, but it believed that only DECCA/FFRR may have used their 78 RPM EQ curve for LPs into 1950, or later. The RIAA EQ curve is also shown for comparison. It will be noted that the 78 RPM EQ curves do not have a LF shelf due to the fact that they do not have any specified value for the time constant t1.[3]

B.2.2 The 1950's and Beyond

Three of the largest recording houses at the time [RCA, Columbia, and Decca/FFRR] tended to dominate the recording industry as the LP record format developed, but they each had their own EQ curves. In addition, the National Association of Broadcasters (NAB) had their own EQ curve that was employed by some recording companies. **Figure B.2** shows a graph of the major playback EQ curves that were in use after 1950, but prior to the RIAA standard, and the RIAA EQ

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[12] See for example References [1] and [2] at the end of this Appendix.

[13] [2] however, based on some literature, it appears that Decca continued to produce records using the 78 RPM EQ curve until 1956, but these may have been 78's, and not LPs.[3]

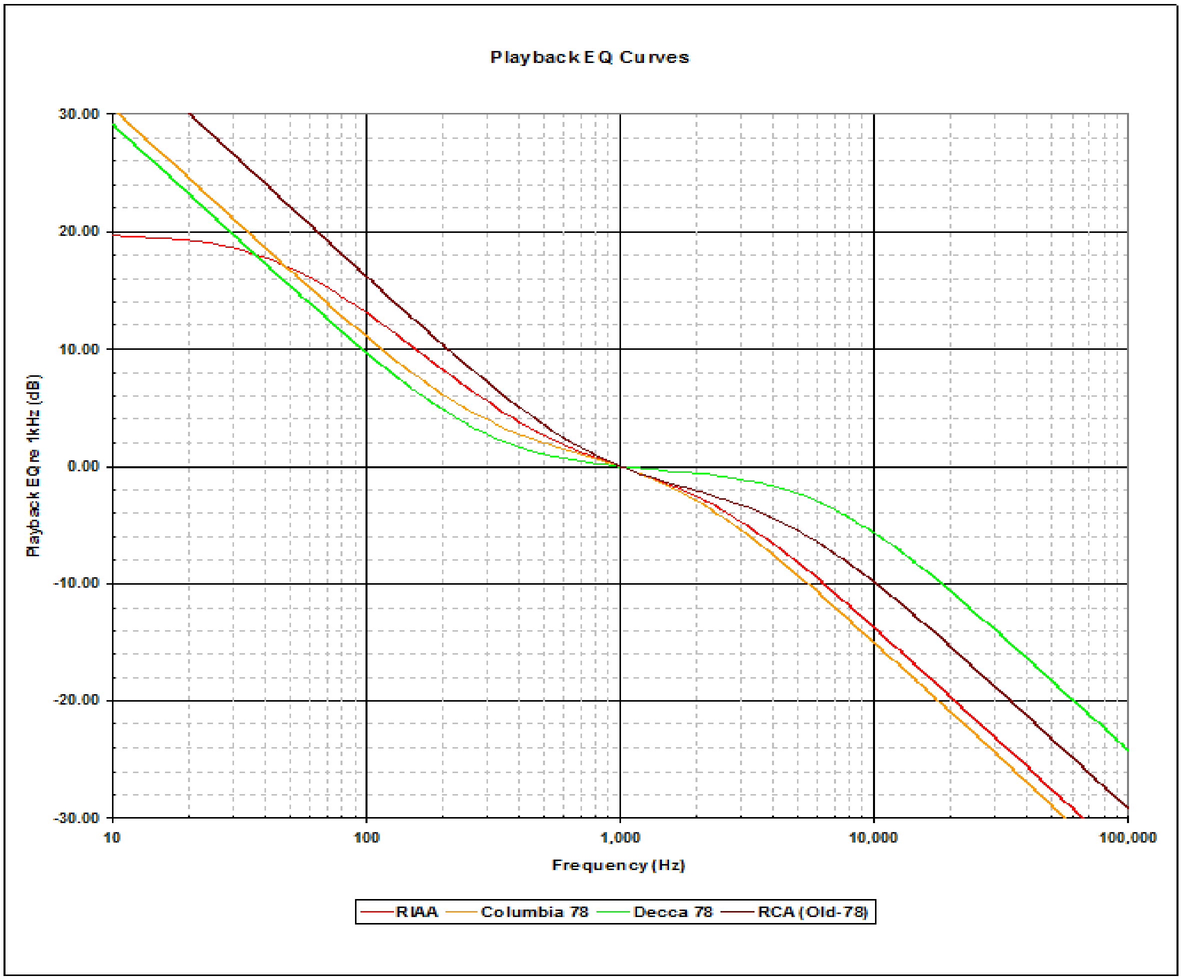

Figure B.1 – Early 78 RPM EQ Curves Compared to the RIAA Curve

curve [1]. Note that these five curves all have the time constant t2 in common (318µsec), which defines the boost (+3dB for playback), or cut (-3dB for record) at 500Hz.

The curve marked "RCA (Old)" was used by RCA from 1947 - 1952[14], after which RCA added the 3180µsec time constant (t1) which gave the "roll-off shelf" of +20dB and which became included in the RIAA curve.

The "Columbia" EQ curve was used by Columbia LPs from 1948 -1955, after which they used the RIAA curve. The last known Columbia LP said to have used the curve shown here was ML4895 [1]; however, the curves shown would be correct for Columbia LPs prior to that date and number.

[14] August 1952 is supposed to have been the change-over date[3].

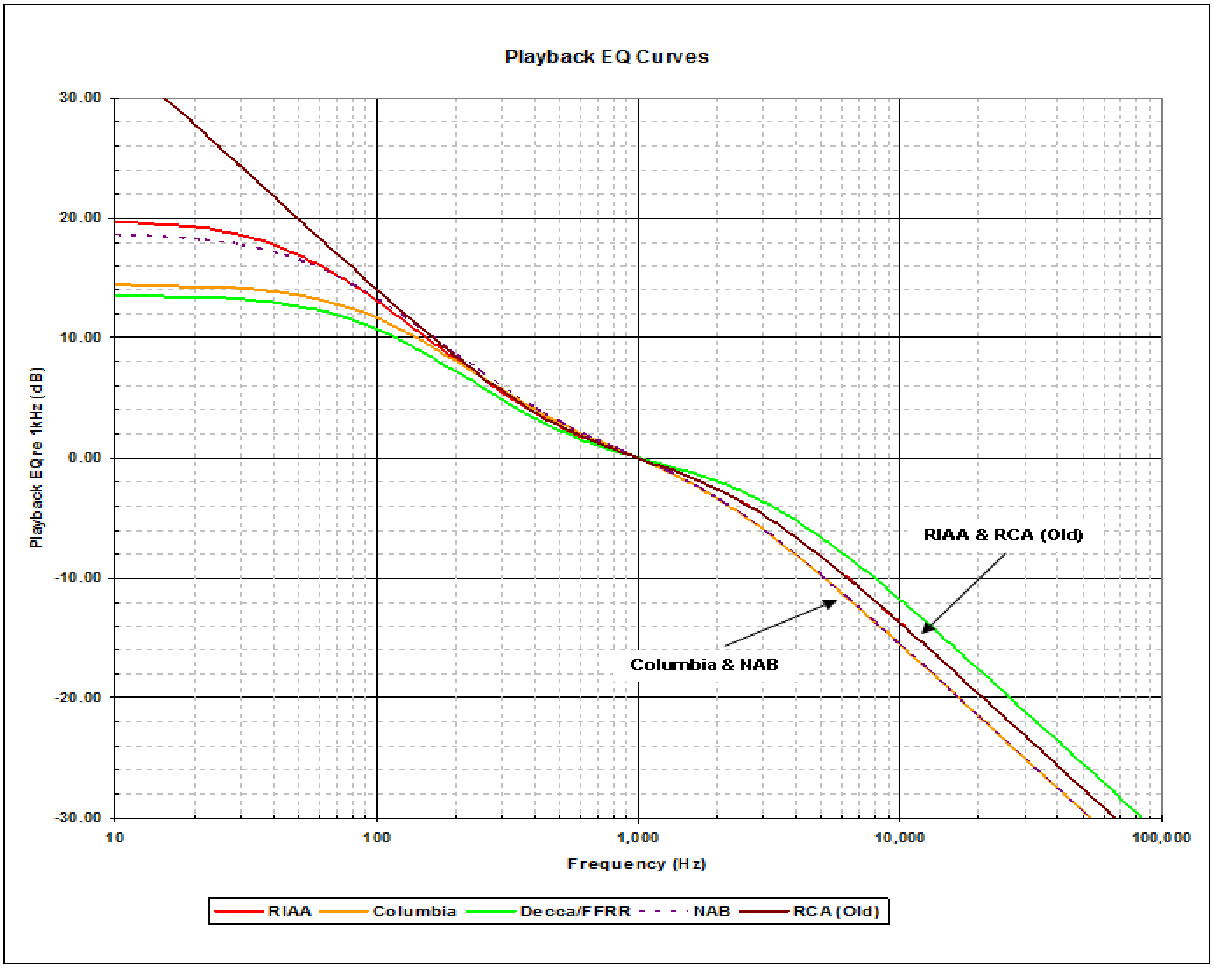

Figure B.2 - Playback EQ Curves Early 1950's

Note that the Columbia EQ curve matches the NAB curve for frequencies >1kHz. [15]

The Decca/FFRR EQ curve shown is supposed to have been used after June 1950 [2] and prior to adopting the RIAA standard. Prior to the June 1950 date, LPs by Decca/FFRR used their 78 RPM EQ curve which is shown in Figure B.1.

Fortunately, the universal adoption of the RIAA curve has rendered these label specific EQ curves obsolete and the issue for playback equipment designers is one of matching the RIAA playback curve as accurately as possible.

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[15] See references at end of Appendix. Due to a lack of good information, many articles have been written about the Columbia EQ curves and what is shown here is believed to be the best representation of what was used at the time.

It will be noted from the curves shown in Figure B.2 that the largest difference between the RIAA EQ and the others is at the LF end of the audio spectrum. At 10 kHz the differences between all of them is on the order of $\pm$ 2dB, whereas the differences at 50Hz are 4-5dB, with the RIAA EQ providing excessive LF boost for some, and not enough for the RCA (Old). With reference to the older EQ curves in Figure B.1, the differences between the RIAA and the other EQs are significantly greater.

The following from reference [4] is a guide to identifying some 1950's recordings that used the RIAA EQ standard.

- "ORTHOphonic" or "New ORTHOphonic" (after August 1952), "NARTB" or "New NARTB" (after June 1953) or "New AES" (after 1954) indicate RIAA equalization.
- The following labels should have used only RIAA: Argo, Bethlehem, Classic Editions, Chess, Clef, Composer Recordings, McIntosh, Montilla, New Jazz, Norgram, Prestige, Romany, Roulette, Savoy, Vocalion and Walden [High Fidelity Magazine, MidiMagic].
- A "HIFi+" sticker on American Columbia's or the catalogue number written in an inverted triangle on German records (like Deutsche Grammophon) is a symbol for RIAA.

B.3 DESIGNING EQ CURVES FOR THE PHONO III

B.3.1 Design Approach and EQ Curves

For the original Phono ONE, and later the Phono II, Dynamic Sounds Associates (DSA) designed a highly accurate RIAA EQ curve for playback. In addition, the Phono ONE also included the option of adding a 3.18µsec time constant to the playback, which would roll-off the decreasing HF response at about 50 kHz. This was supposed to correct for some cutter-head issues; however, this has largely been proven unnecessary, and is not part of the RIAA EQ. This option is not included in the Phono II or the *Phono III*.

In designing the DSA *Phono III*, our desire was to complement the RIAA EQ curve with the addition of additional EQ curves that can be selected by the user at time of playback. However, a decision had to be made regarding the number of alternate EQ curves and which ones were to be included. To allow for the addition of other alternate EQ curves in the future, it was decided to develop a "plug-in" module for this purpose instead of a "hard-wired" solution.

In reviewing the information on the various EQ curves shown previously, it became apparent that there was a good degree of commonality between EQ curves that could be combined into two "groups" identified as those based on 78 RPM EQs, and those that were after that period but prior to the universal adoption of the RIAA EQ curve. In addition, there was considerable commonality between the t2 time constant used by Columbia and Decca/FFRR during these two periods which led to the decision to base the design of the "stock" EQ module for the *Phono III* on the EQ curves from these two vendors. **Table B.1** shows the time constants associated with the two groups of EQ curves, and how they compare within each group and with the RIAA standard.

Based on **Table B.1**, a plug-in module was designed that incorporates all five of the EQ curves shown. The selection of a particular EQ curve is done for each input by using a pair of switches located behind the lower central door on the front panel of the *Phono III*. (This is shown in detail in Section 3.5.) The EQ curve selection follows the selection of MC or MM cartridge loading for each input.

Table B.1

GROUP	EQ Curve	NAME	t1	t2	t3
---	--	RIAA	3180 µsec	318 µsec	75 µsec
G1	**EQ1**	Columbia-Pre RIAA	1590 µsec	318 µsec	100 µsec
G1	**EQ2**	Decca/FFRR - Pre RIAA	1590 µsec	318 µsec	60 µsec
G2	**EQ1**	Columbia 78RPM	---	530 µsec	100 µsec
G2	**EQ2**	Decca/FFRR 78RPM	---	530 µsec	25 µsec

The first switch for EQ selection determines the "group" to be used (G1, or G2), and the second switch selects the appropriate curve (EQ1, or EQ2) from that group. The switch for group selection has a central "off" position which selects the RIAA EQ curve for that input, and the EQ switch for that input is disconnected. For both switches, "down" is the "1" selection, and "up" is the "2" selection.

B.3.2 Designing the EQ Module

The design of the EQ module with the five EQ curves was based on how the RIAA EQ was incorporated into the *Phono III*. As shown in the block diagram for the *Phono III* (Figure 5), the RIAA equalization is divided into two low-pass filter sections. The portion of the equalization that is >1 kHz follows the first gain stage, while the portion <1 kHz follows the second gain stage. The sole purpose of the 2nd gain stage is to add gain such that a low-pass filter can be used for the equalization of the LF portion of the audio spectrum. The EQ module that was designed for the *Phono III* encompasses both of these two low-pass filters, but with the appropriate components selected by miniature relays.

B.3.2.1 EQ above 1 kHz

The EQ for frequencies >1 kHz is accomplished with a series resistor (R) followed by a precision capacitor (C) which shunts the signal to ground such that the value of R x C = desired time constant, t3. With this configuration, only the values of C must change for the different EQ curves, as shown in **Figure B.3**. The RIAA curve (Sw_0 & C_0) is used as "baseline" for the final "trimming" of R to obtain t3 = 75 µsec, after which only the values of capacitance must be selected using relays Sw_1 - Sw_3 to obtain the desired EQ curves. [16]

B.3.2.2 EQ below 1 kHz

The EQ for frequencies <1 kHz is more complicated since there are two time constants to be considered. **Figure B.4** shows the final design concept to meet the requirements as closely as

[16] The trimming of R to obtain an accurate RIAA response is performed by DSA prior to shipping the unit to the user.

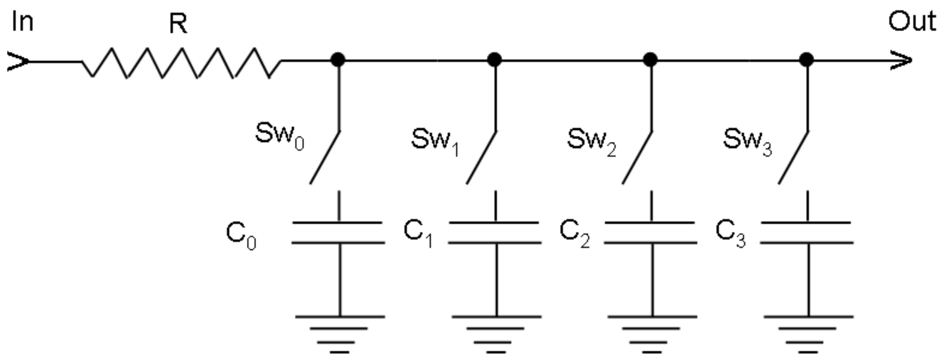

Figure B.3 – EQ Design Above 1 kHz

possible. This design required two switches, with Sw_1 being of the DPDT[17] type, and three resistors in addition to those required for the RIAA EQ.

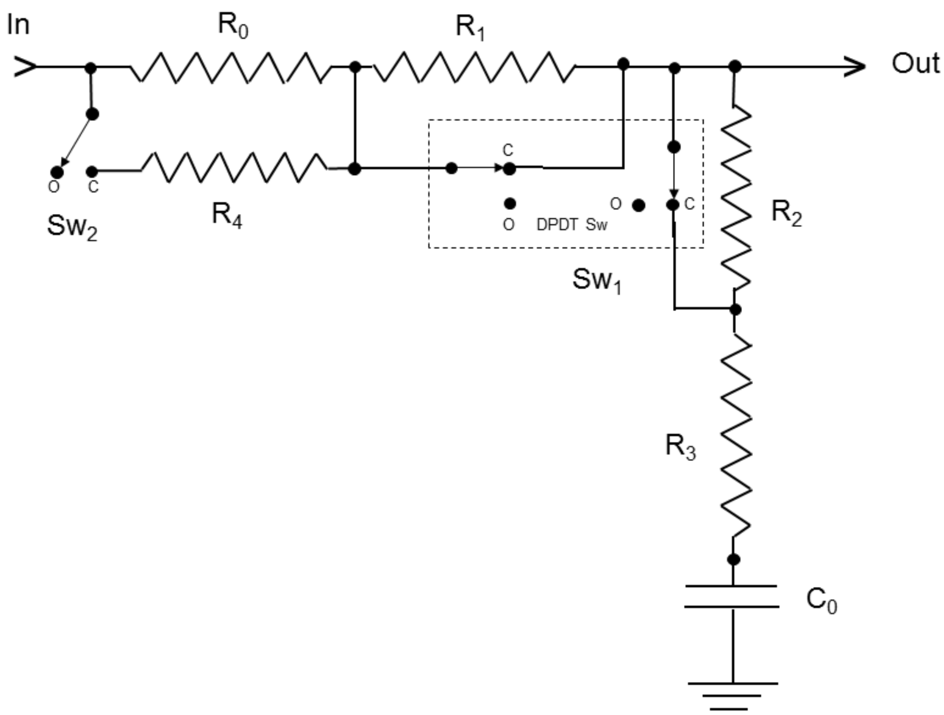

Figure B.4 - LF EQ Curve Selection Design

For the RIAA EQ, Sw_1 is in the "closed" position (as shown) and Sw_2 is "open" (as shown). The pair of R_3 and C_0 (mounted on the amp PCB and not the EQ module) provides the t2 time constant of 318 µsec for the necessary rise in response for frequencies <1 kHz. As the frequency decreases to

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[17] Double-pole & Double throw -- This is the equivalent of a "ganged" pair of double-throw switches that move in unison.

the point where the impedance of C_0 becomes high (≈ 50 Hz), the combination of R_0, R_3 and C_0 provide the t1 time constant of 3180 µsec. This is because the input to this network is fed from a very low impedance source which becomes a "virtual ground" at low frequencies, making R_0 appear to be in parallel with the series network of R_3 and C_0 .

For the EQ curves of Group 2, Sw_1 and Sw_2 are "open" to remove R_4, while adding R_1 and R_2 to the circuit. Then R_2 and R_3 combined with C_0 form the t2 time constant of 530 µsec. The addition of R_1 in series with R_0 increases the value of the t1 time constant, but does not make it so large as to eliminate a LF "shelf" in the response. As will be seen, this creates an error in the very low frequency response as compared to the desired EQ; however, this error can be held to be on the order of ≈ 2dB. The penalty to be paid is a -6dB reduction in the gain of the *Phono III*; or, in other words, each gain setting is reduced by 6dB. In most cases this can be largely corrected by simply selecting a greater value of gain when using these EQ curves.

B.4 RESULTS

The first step was to perform computer simulations on the *Phono III* circuit with the various resistance and capacitance components as shown in the **Figures B.3 and B.4**. These simulations were then compared with the desired design curves to confirm that the proposed designs would perform as desired prior to doing circuit board layouts.

Following assembly of a *Phono III*, the relative[18] equalization responses were measured using the CLIOwin7 system. In all cases, the input signal was adjusted to “normalize” all of the curves for 0dB at 1 kHz to make comparisons easy. As seen in **Figure B.5** which shows both the ideal and typical measured responses for both RIAA and the G1 equalizations, the measured results very accurately correspond to the desired responses in all cases with many errors <±0.1dB.

For the G2 group of equalizations shown in **Figure B.6**, it can be seen that except for the region below 50Hz, the typical measured equalization responses very accurately track the desired EQ curve in both cases. At 20Hz, the measured results are about 3.5-4.0dB lower than the desired boost, and they are about 8-9dB lower than desired at 10Hz. Since 20Hz is probably the lowest frequency that even the best audio system will be expected to deliver, we can probably ignore the errors at 10Hz, and instead focus on the accuracy for frequencies at 20Hz and above where the errors rapidly become insignificant. This is especially true for the frequencies > 1k Hz where the simulations and EQ curves are virtually perfect matches.

See the enclosed copies of the actual measurements of the relative EQ responses for this unit at the end of this Appendix. The error values [Error(f)] are calculated from [Ref(f) – (Meas(f) – Meas(1kHz)]. (The frequency steps are determined by the measurement system.)

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[18] The measurements presented are “relative” because of minor gain changes when selecting either the G1 or G2 group of equalizations relative to the unit gain for the RIAA equalization.

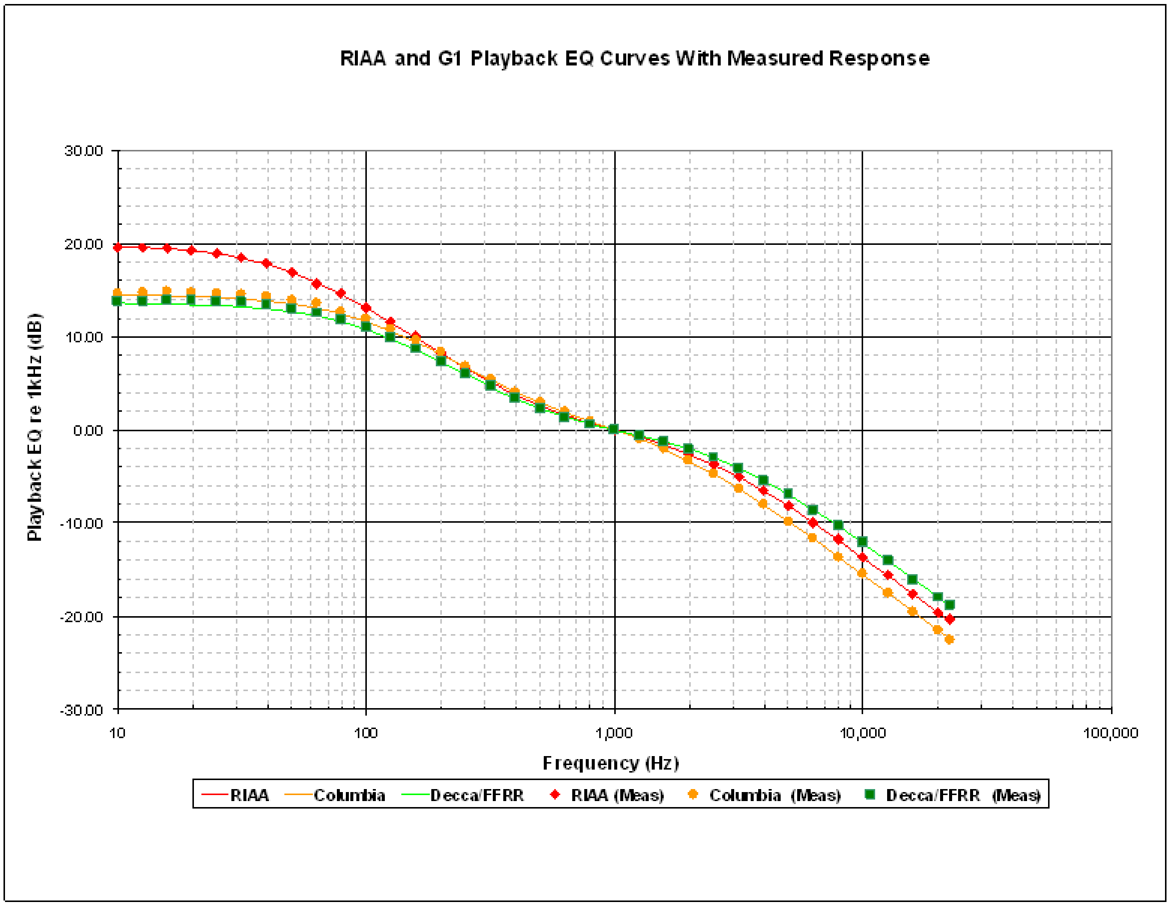

Figure B.5 – G1 Measured EQ Response Compared to Ideal Curves

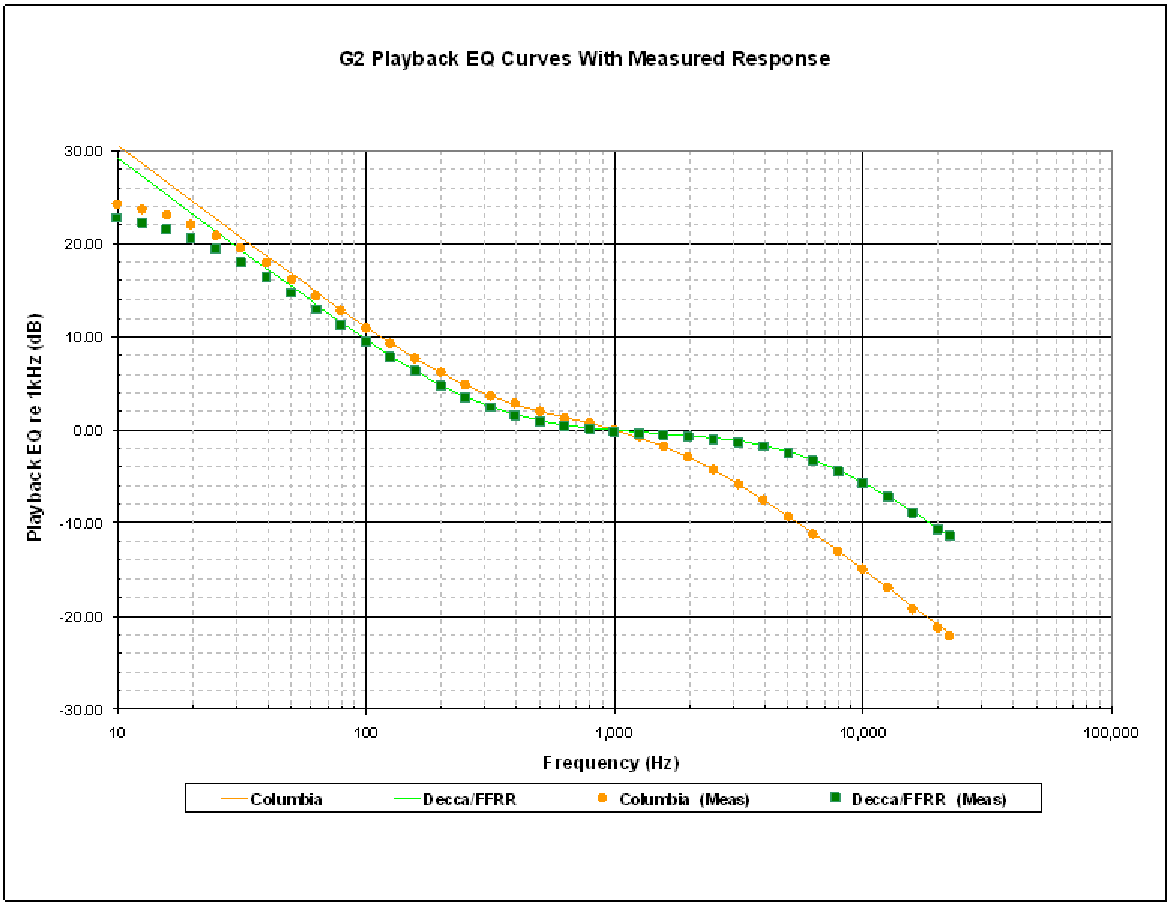

Figure B.6 – G2 Measured EQ Response Compared to Ideal Curves

REFERENCES

[1] Gary A. Galo, "The Columbia Lp Equalization Curve", ARSC Conference Paper, 29 March 2008. This paper can be found on-line at:

https://linearaudio.net/sites/linearaudio.net/files/galocollpcurvefinal.pdf.

See www.arsc-audio.org for further information.

[2] http://pspatialaudio.com/record_characters.htm (This provides a very compendious historical review of EQ curves and examines the Decca/FFRR curves in great detail.)

[3] https://en.wikipedia.org/wiki/RIAA_equalization (This entry has numerous internal links to information and tables of EQ curves used at different times by various record manufacturers, including the link to [4] below.)

[4] https://wiki.audacityteam.org/wiki/78rpm_playback_curves#table_3