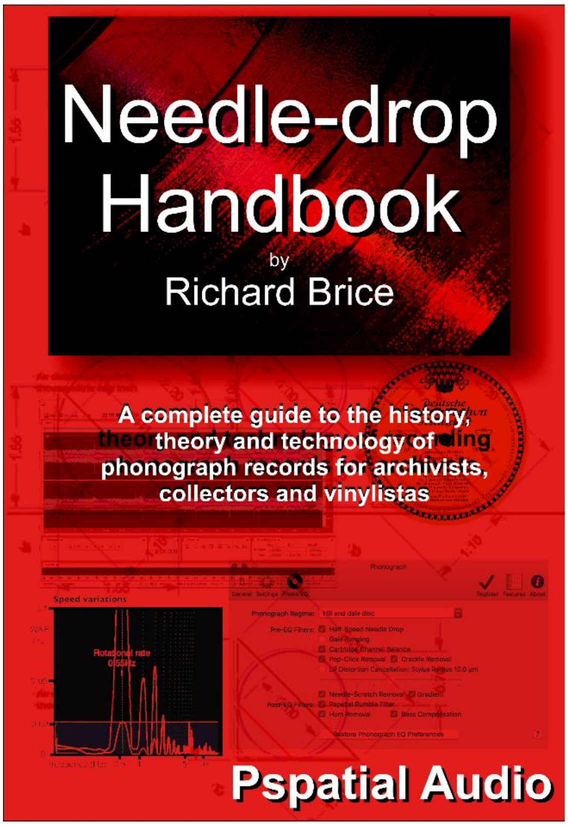
User Manual for

PHLUX-AI

ACTIVE INTERFACE
PHONO PREAMPLIFIER

Version 1.23 (Spring 2025) – **PHLUX-AI**

Preface



The ideal adjunct to the **PHLUX-AI** preamplifier manual is Pspatial Audio's **Needle-drop Handbook** because so much background information is available in this book. [1]

Although the **Needle-drop Handbook** is principally concerned with digital reformatting to preserve the *essence* of audio recordings divorced from their physical carrier, it is as relevant to the audiophile vinylista as it is to the archivist. Both need first-class hardware – where *the rock meets the record*, analogue discs are ultimately a precision mechanical medium and demand precision mechanisms to replay them.

The first five chapters of the **Needle-drop Handbook** are devoted to the mechanical issues of playing records: the mechanics of constant-speed; of tracking and tracing distortion; and of cartridge tracking and groove geometry.

The equipment to cut the master acetates is covered in chapter six of the **Needle-drop Handbook**, as are the subsequent electroforming and compression-moulding processes required to produce commercial records. Only an understanding of how records are made, alerts us to the problems which arise in manufacturing and to the extra information which is encoded in the *dead-wax*.

The care of records – when playing them and storing them - is covered in chapter seven along with information of the care and maintenance of the replay stylus.

The signal sent to the cutter-chisel when cutting a record is equalised, whereby the bass is cut and the treble boosted. The equalisation of modern records is described in chapter eight of the **Needle-drop Handbook** and the equalisation of pre-stereo records has a chapter all to itself in chapter ten.

The revised third edition of the **Needle-drop Handbook** has a series of afterword chapters on: noise mechanisms, both mechanical and electronic; lubrication; filter design; and advice on loudspeakers and amplifiers. Before we get there, there are a series of chapters looking at more unusual forms of records of interest the collector: shellac discs; quadraphonic discs; cylinder records; soundtrack discs for early "talking pictures"; radio transcription discs; and novelty records. We also devote a chapter (thirteen) to looking at popular upgrades and tweaks to turntables and ancillary equipment to see if they work or not.

Richard Brice
France 2025

[1] Go to https://pspatialaudio.com/NDH_book2.htm to purchase a copy of the **Needle-drop Handbook**.

Chapter 1

Introduction

Why buy a Phædrus Audio phono preamplifier?

Phædrus Audio's **PHLUX-AI** preamplifier is designed for all vinylistas – both the hard-line analogue lover and the modernist. The **PHLUX-AI** preamplifier may be used as a no-compromise, high-quality stand-alone phono preamplifier, or in conjunction with Pspatial Audio's **Stereo Lab** app' which runs under OS X on the Apple Mac.

For the analogue purist, the **PHLUX-AI** preamplifier is designed to offer state-of-the -art, low-noise preamplification in a unique, and innovative bridge-based preamplifier configuration. It also employs circuit techniques from the summits of analogue design (for example, the legendary WV2 preamplifier from Neumann).

For the modernist (and the archivist), using the **Stereo Lab** app' isn't just technology for technology's sake. To the purist, it may seem sacrilege to digitise the audio from a record in a, so called, *needle-drop* recording. But digitisation is a necessary stage in applying the most powerful technology of our age – modern computing – to recovering the best possible signal quality from a record. We don't cavil when modern signal processing is used to reconstruct lost architecture or reveal the radiance of a painting. In our view, it's churlish to gainsay the benefits of this technology to phonograph records. For example, phase-linear warp and rumble filtering is only possible in the digital domain and reveals a bottom octave with unprecedented clarity and realism. The accuracy of RIAA equalisation and perfect channel balance ensures an uncoloured sound with superb stereo sound staging. And the flexibility in recording characteristics and intelligent noise-reduction will bring to life records you thought mediocre. Moreover, **Stereo Lab** also includes software decoders for CD-4, *QS* and *SQ* quadraphonic and *Ambisonic* LPs; thereby bringing life to forgotten, analogue, multichannel gems.

For **Stereo Lab** to process needle-drop captures and equalise them, it needs recorded files of the "raw" signal, direct from the groove, unfettered by analogue equalisation. And that is where the **PHLUX-AI** preamplifier comes in. It is a phono preamplifier of the very highest quality but also provides a record output *without* equalisation, so that the signal fed to the computer is the closest version of the signal direct from the groove. The **PHLUX-AI** preamplifier has various options to augment this basic task to aid with monitoring and to ease integrating these units into a high-quality audio system. These are described in the pages that follow.

Features

PHLUX-AI unit

- Bridge-based preamplifier with unique, low noise circuit architecture
    - Supports a wide range of moving-magnet cartridges
        - Also moving-coil, active, and strain-gauge types with added options
- Line level (non-equalised) output available for direct connection to external soundcard for use with **Stereo Lab** software
- USB powered – all internal rails derived from USB V_{bus}
    - Dual, internal high-stability bandgap regulation to eliminate PSU noise

Input/output options

Moving-coil input options

- Support for modern as well as historical MC cartridges (e.g. Neumann DST)
    - Two gain versions available (500µV/ 150µV sensitivity)
        - Third, 0dB version offers a high-Z input (for moving-iron types)
- State of the art low-noise design: -141dBu EIN at highest gain
- Variable load adjustment

PHLUX input option

- Provides phantom power to Phædrus Audio's **PHLUX** active cartridges

Strain-gauge input option

- Semiconductor cartridge support

SYNOPSIS soundcard option

- Stand-alone needle-drop solution
- Driverless, class-compliant USB protocol controller
    - Works with all common operating systems
- USB input-output connection to computer

Neumann WV2 output option

- Raises output to full professional output level (-6dBu = standard level)
- Based on output stage of Neumann WV2 playback equaliser for the AM-32b lathe
    - Considered to be the Holy Grail of RIAA preamplifiers
- Transformer coupled output for massive signal headroom

Processing options

EQ (RIAA) option

- Provides a precision RIAA equalised output

FRANCINSTIEN stereo enhancement option

- Adds stereo enhancement to EQ option

PHOENIX EQ and signal processing

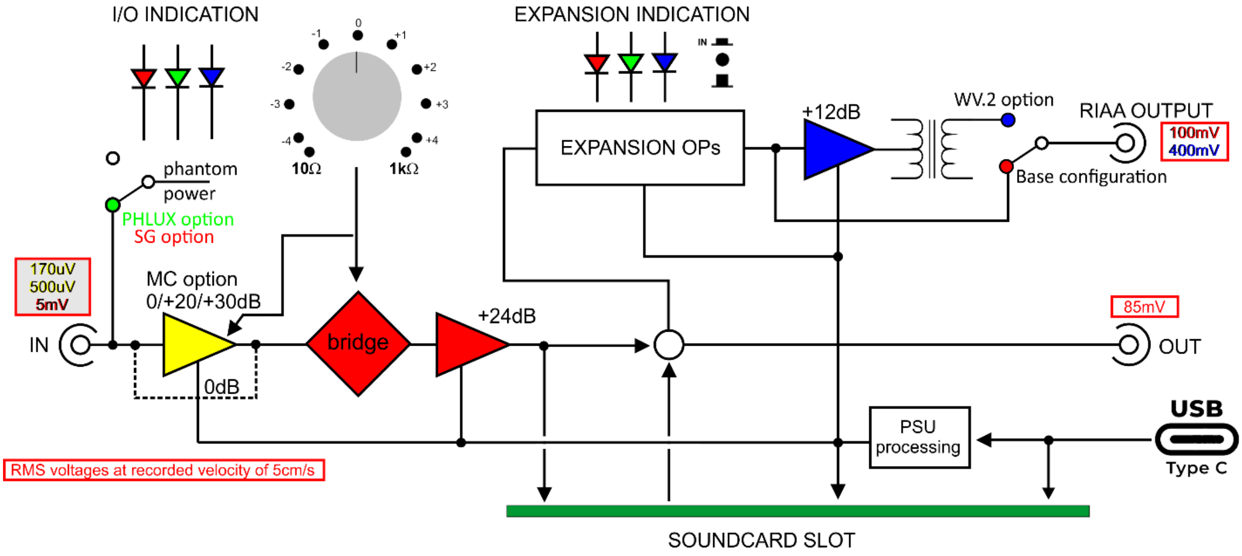
- Adds processing for shellac-era records to EQ option. Selected by push-switch.

Preamplifier option

- Volume control and option bundle to provide a complete preamplifier solution

Facilities diagram

The functionality of equipment like the **PHLUX-AI** preamplifier is often most rapidly grasped with a facilities diagram, like the one given below.



A larger version of this facilities diagram is available in appendix 5.

Multicolour LED configuration indicators

MM
PHLUX
MC
Strain-gauge
MM & WV2
PHLUX & WV2
MC & WV2
Strain-gauge & WV2
USB AUDIO DATA

The left-hand front panel multicolour indicator indicates the input and output (I/O) options fitted in the **PHLUX-AI** preamplifier, according to the table (left).

When a **SYNOPSIS** soundcard is fitted, the LED continues to indicate the input configuration, but when valid audio data is passed over the USB interface, the indicator illuminates WHITE.

No expansion fitted
RIAA EQ
RIAA + PHOENIX
RIAA + FRANCINSTIEN
RIAA + PHOENIX + FRANCI
SIGNALLING

Other expansion options: **PHOENIX** shellac processing; **FRANCINSTIEN** stereo correction; etc. are indicated by the right-hand multicolour LED configuration indicator. These are defined right. Note that the WHITE colour signal has different roles according to the options fitted. See elsewhere in the manual for the meaning of the WHITE signal for specific options.

Chapter 2
Safety

Before using any piece of equipment manufactured by Phædrus Audio, be sure carefully to read the applicable items of these operating instructions and the safety suggestions. Keep them for future reference. Follow the warnings indicated on the unit, as well as in these operating instructions.

THE USER SHOULD NOT ATTEMPT TO SERVICE THE UNIT. ALL SERVICING SHOULD BE REFERRED TO QUALIFIED SERVICE PERSONNEL OR FACTORY ONLY.

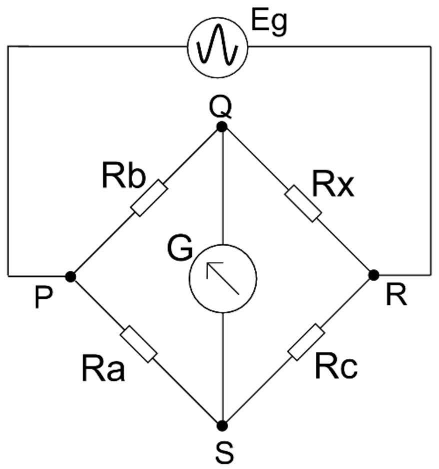
Phædrus Audio products should NEVER be connected to the external power supply or in any other way energised when the case is opened and/or the circuit boards are accessible.

General Safety Instructions

- Do not operate this equipment near any source of water or in excessively moist environments.
- Keep this equipment away from babies, children and pets.
- Do not let objects fall, or liquids be spilled, onto the enclosure.
- Situate this equipment away from heat sources or other equipment that produce heat.
- Ensure this equipment has adequate ventilation.
- When cleaning this equipment, remove all connections to the unit; including power and gently wipe with a clean, lint-free cloth; if necessary, gently moistened with lukewarm or distilled water. Use a dry lint-free cloth to remove any remaining moisture. NEVER use aerosol sprays, solvents, or abrasives on this equipment.
- This equipment should be serviced by qualified service personnel or returned to Phædrus Audio when: an object (or objects) have fallen into the enclosure; or liquid has fallen into, or been spilled into the unit; or the unit has been exposed to rain or high humidity; or the unit does not operate normally or exhibits a marked change in performance; or the unit has been dropped, or the enclosure has been damaged.

Chapter 3
Bridge preamplifier theory

Wheatstone's bridge



In about 1843, Sir Charles Wheatstone designed a circuit called a *bridge* which gave an accurate method for measuring *resistance*. The circuit is given left. In this circuit, resistors ***Ra, Rb, Rc,*** are all known values. ***Rx*** is the unknown resistance.

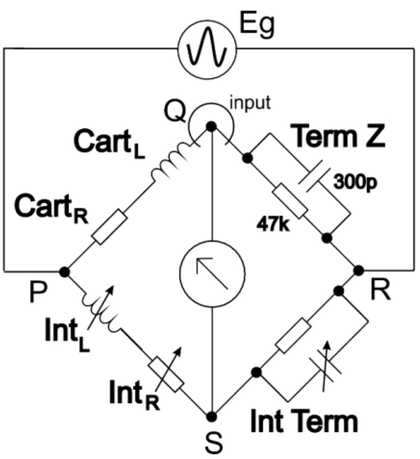
The full explanation of this circuit is in almost every textbook on electricity, so we won't repeat that here. The important feature to appreciate is that when the bridge is said to be *balanced*, current through the galvanometer ***G*** falls to zero, which occurs in the condition,

Ra / Rb = Rc / Rx

In Wheatstone's time, this relation was important because from this it was possible to deduce the resistance of ***Rx*** according to the formula,

Rx = (Rb / Ra) × Rc

Maxwell's bridge



Maxwell's bridge is a development of the Wheatstone bridge, named after James Clerk Maxwell, who first described it in 1873. Maxwell's bridge extends the use of Wheatstone's bridge to measure reactive (not simply resistive) circuit elements.[2] An example is illustrated. This bridge circuit relates directly to the application in which a moving-magnet electrodynamic phono pickup is driving a conventional high-impedance (47kΩ) preamplifier input.

Cart$_R$ and ***Cart***$_L$ represent the resistive and inductive components of the cartridge generator impedance which together we call **Cart**$_Z$. The termination impedance (***Term***$_Z$) is made up of the resistance (typically 47kΩ) in shunt with the cable capacitance – here shown as 300pF, a typical figure. Note that point ***Q*** is the coäxial cable input to the preamplifier. Just like the

[2] It is often said that Maxwell's bridge is for the measurement of *inductance* and that was indeed Maxwell's original intention. However, it may be said to be a bridge of general reactance – capacitive as well as inductive.

Wheatstone bridge, provided the impedances $\boldsymbol{Cart_z}$ and $\boldsymbol{Int_{Term}}$ are balanced with respect to the external impedances, the galvanometer will indicate zero current.

Now, this is of little use practically because point ***P*** doesn't exist in reality. The generator voltage ***Eg*** is developed across $\boldsymbol{Cart_Z}$ and is not independent. And, even if it was, there would be no way to access point ***P***. Nevertheless, the Maxwell bridge indicates the concept that – provided we balance the impedances of the cartridge and the cable, it ought to be possible to recover the "pure" generator EMF (***Eg***).

Bridge preamplifier

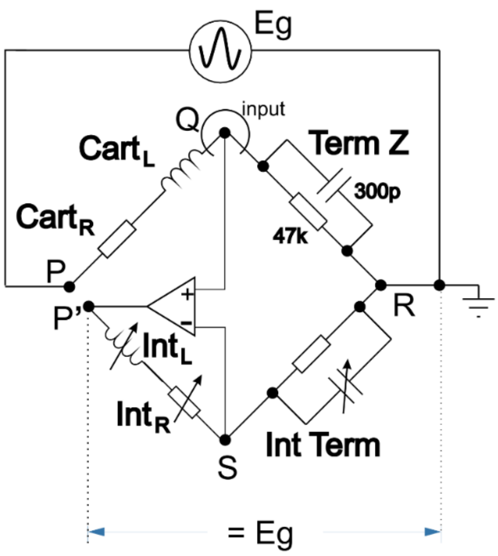
In the case of the **PHLUX** active and strain-gauge cartridges, recovering ***Eg*** is accomplished by the "belt-and-braces" technique of buffering the generator signal, thereby eliminating the effects of the cartridge impedance, the cable, and their mutual interaction. In a similar way, the contributing reactances in the $\boldsymbol{Cart_Z}$ of a moving-coil cartridge are so low that matching the internal impedances is largely unnecessary.

The necessary step in realising a practical preamplifier for moving-magnet cartridges which avoids the need for active electronics in (or very near) the cartridge involves an appreciation that, when the bridge is balanced, point ***S*** and point ***Q*** are at the same potential. (They must be for there to be no current through the galvanometer.)

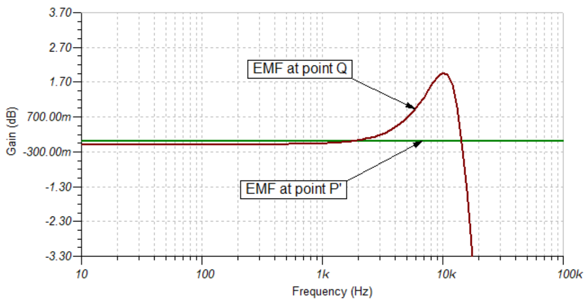
Why we should be interested in moving-magnet cartridges?

See appendix 2.

Consider the following circuit in which a high-gain operational amplifier has its two inputs connected to points ***Q*** and ***S***: the non-inverting input fed from point ***Q***, and the inverting input from point ***S***.



One characteristic of a very high-gain amplifier is that, provided it is not overloaded, the potential between its differential inputs must be very small – because the gain is very high. In this circuit therefore, provided the bridge arms balance – even without a connection to point ***P*** – the operational amplifier will force a new point ***P'*** to be at the same potential as point ***P***. The original ***Eg*** may thereby be recovered between point ***P'*** and point ***R***. This is the principle behind the bridge-balance preamplifier.



If we arrange ***Eg*** to be a frequency sweep, we can monitor the voltages versus frequency at point ***Q*** (the conventional preamplifier input), and at point ***P'***. A graph of the results is given. From which, it is very clear how the effect of the interface impedance on the internal impedance of the cartridge is eliminated. Frequency response variation and restriction are both eradicated.

Other innovations

The bridge arrangement alone does not solve the other issues that the **PHLUX** active cartridge technique addresses. Namely: electronic noise in the high impedance interface between cartridge and preamplifier; and left-right crosstalk in the tiny cables in the tonearm. Likewise, the not inconsiderable problem of calibrating the bridge impedances had to be solved. Various circuit developments were required to address these concerns.

Chapter 4

Using the PHLUX-AI preamplifier

Connecting the equipment

The rear panel of the Phædrus Audio **PHLUX-AI** phono preamplifier is illustrated.

Turntable

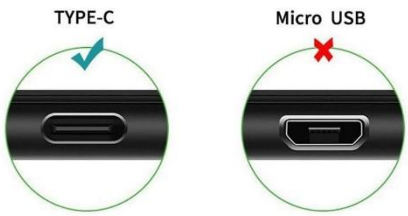
The leads from the turntable should be connected to the sockets labelled IN. Be sure to respect the channel polarity. The convention used in the Phædrus Audio **PHLUX-AI** is the right channel is the upper of the two connectors. In the base configuration, the sensitivity of the IN connectors on the **PHLUX-AI** preamplifier is appropriate for moving-magnet type cartridges. If the moving-coil option is ordered, the sensitivity is raised to be appropriate for this type of transducer. If the PHLUX option (or the strain-gauge option) is added, phantom power is made available on these connectors.

Earth/Ground

The turntable earth (ground) wire should be connected to the green connector labelled GND. You may need to purchase a (4mm) banana plug to mate with this connector to be sure of making a solid and reliable connection. The socket is a 4mm type and has a tin-plated brass insert designed to accept a wide range of 4mm plugs. Whilst audiophiles love gold-plating, gold-tin is not the ideal electrical contact, so we recommend a 4mm plug of nickel-plated brass.

Power Supply (USB)

Finally, connect the USB power. Today, USB power – usually presented via a USB Type-A connector – is ubiquitous. It is available, either via the computer hosting the DAW, or via stand-alone plug top power supplies, as illustrated. **Note that the USB Type-C connector is not the same as the Micro USB connector.**



USB power is not known for its stability or lack of noise. To that end, the **PHLUX-AI** preamplifier incorporates advanced power supply processing which filters any power supply noise and thereby achieves the excellent noise and dynamic range figures it does (see *Specifications*). The current load on the USB supply due to the **PHLUX-AI** preamplifier is very light, less than 100mA.[3] The preamplifier may therefore be powered from USB 1.1, 2.0 ports (white plastic inside), or first or second-generation USB 3.0 & 3.1 ports (blue or red plastic inside).

Some customers may wish to use an external, linear power supply type (especially when not using the internal **SYNOPSIS** soundcard). There are several manufacturers of this type of supply, and they deliver high-quality and low noise 5V DC at very reasonable cost (one is illustrated). If you want to arrange your own PSU, be aware that USB power is specified to be 5V ±5% (4.75V to 5.25V) and these limits must be respected.

RIAA & OUT

Connections to the following soundcard (if used) and preamplifier/ integrated amplifier depend on the configuration of the unit and the options fitted.

Note: there is no iLOOP function as there is on the Groove Sleuth preamplifiers: the middle set of phonos is ALWAYS set to RIAA (as indicated on the rear panel).

User Cases

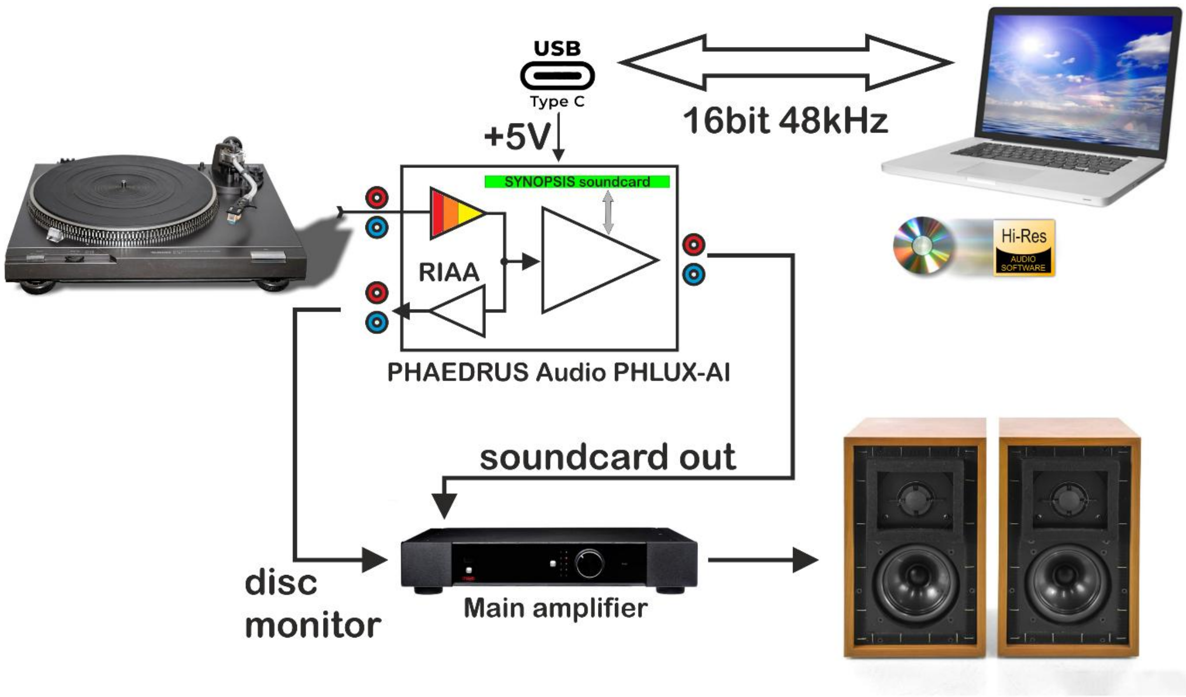
In order to see the multiple ways the **PHLUX-AI** may be employed, we present five user cases: the (**Home) Archivist**; the **Pro Archivist**; the **Audiophile**; the **Record Collector;** and **Mr. Zane Gauge.**

[3] The internal **SYNOPSIS** soundcard option raises the consumption by another 100mA.

User case #1 – the Home Archivist

The **Home Archivist** opted for the most streamlined needle-drop solution, with the internal **SYNOPSIS** soundcard fitted inside the **PHLUX-AI** unit and the RIAA expansion option. She uses the **Hi-Res** precision upscaling algorithm in the **Stereo Lab** app' to double the sample-rate of single-rate files and increase the bit depth to 24-bit with a precision dithering algorithm.[4]

The only user input (apart from setting the monitoring volume) is to select between the RIAA equalised record monitor (on the **RIAA** output connector), and the **SYNOPSIS** soundcard output (on the **OUT** connector). The striped input stage symbol reflects the choice of cartridge options available which include: moving-magnet; moving-coil; strain-gauge; and PHLUX-active. In this case, the **Archivist** opted for a **PHLUX**-active type cartridge. This choice has the added advantage that the Bridge Balance control acts as a fine level control which she uses to "tweak" the record level into the SYNOPSIS soundcard.



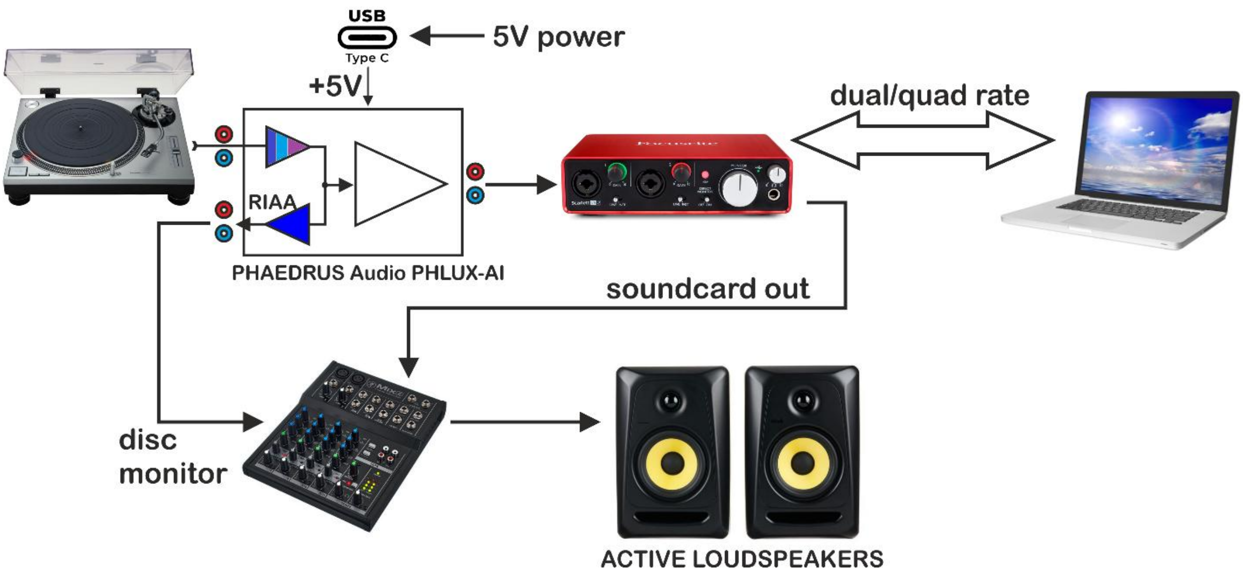
---

4 

Hi-Res may be selected as a specific process in **Stereo Lab** or it may be enabled as a default processing for all files within the Settings menu.

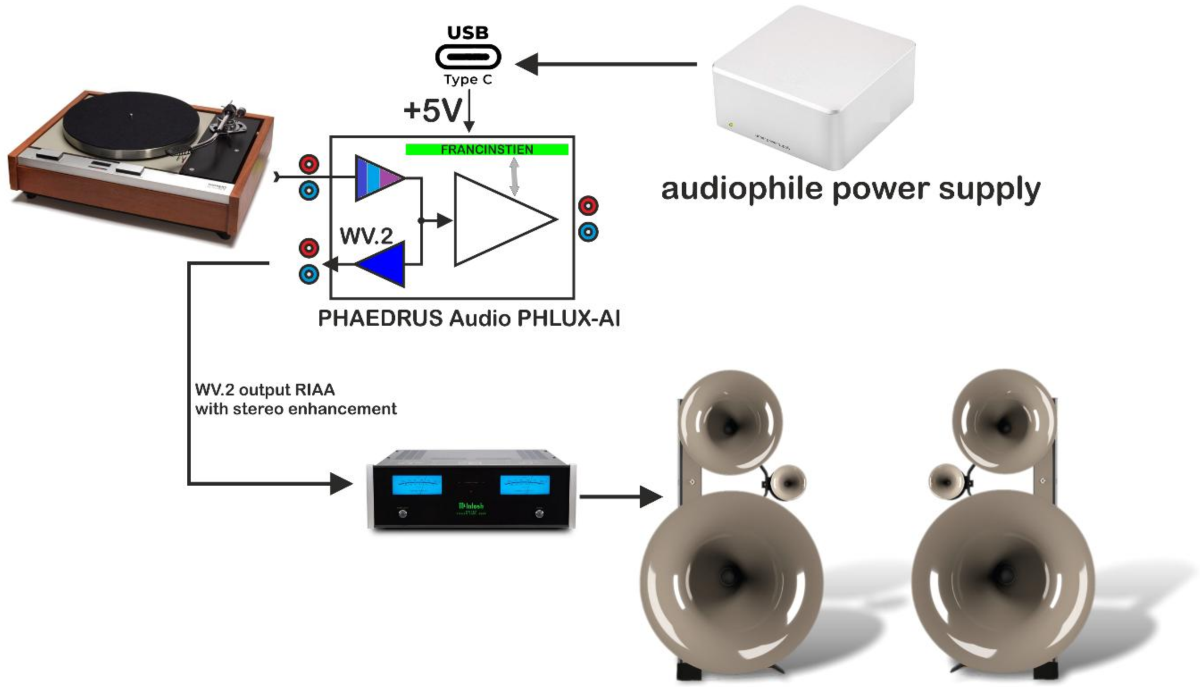
User case #2 – the Professional Archivist

The **Professional Archivist** wanted to be able to needle-drop record at dual (96kHz) or quad rate (192kHz) sampling, and this required an external soundcard. In order better to match the audio levels from a professional soundcard at the mixer, the Pro Archivist has chosen the **WV2 output option** for the RIAA. The striped input stage symbol reflects the choice of cartridge options available (at time of order) which include: moving-magnet; moving-coil; strain-gauge; and PHLUX-active. In line with his organisation's policy on pickups for archiving, the **Pro Archivist** chose a classic (Shure) moving-magnet cartridge.



User case #3 – the Audiophile

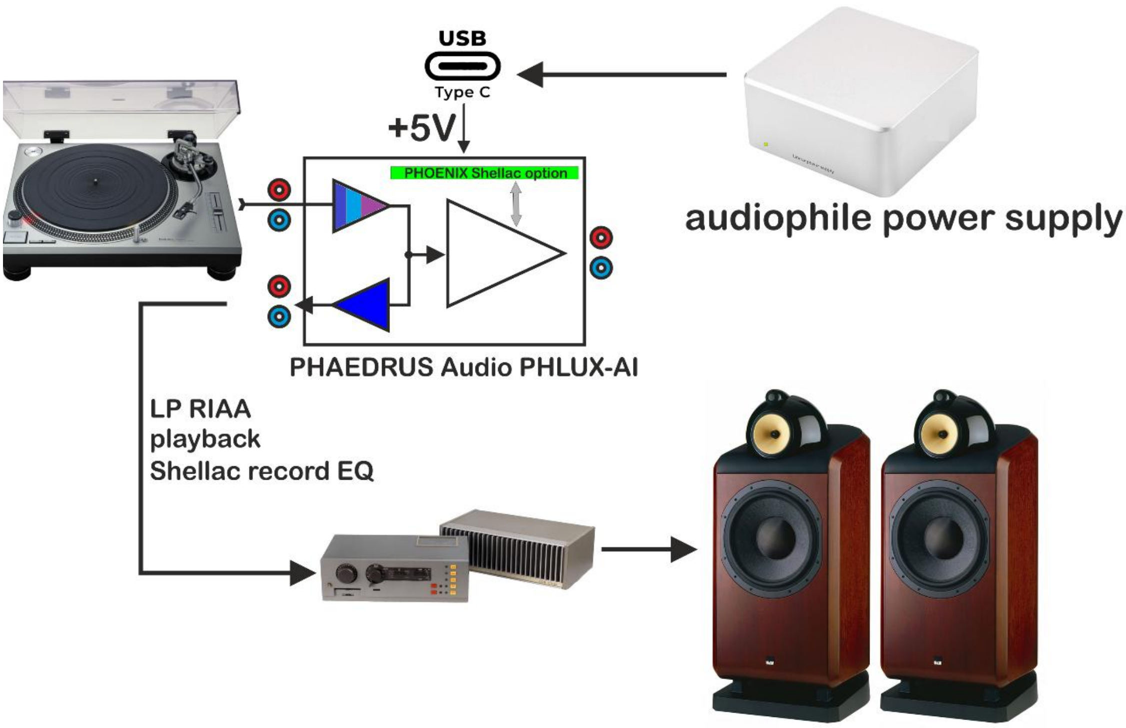
The **Audiophile** has no interest in a computer connection. Instead, the **PHLUX-AI** preamplifier is fed from a quiet, audiophile quality USB power supply. The **RIAA** EQ + **FRANCINSTIEN** option card is fitted, and the **WV2 output option** to maximise stereo-imaging and dynamic-range performance. The striped input stage symbol reflects the choice of cartridge options available (at time of order) which include: moving-magnet; moving-coil; strain-gauge; and **PHLUX**-active. In this case, the **Audiophile** chose an expensive moving-coil type cartridge. The very low moving-coil input stage compliments this cartridge choice perfectly.



PHÆDRUS
phaedrus-audio.com

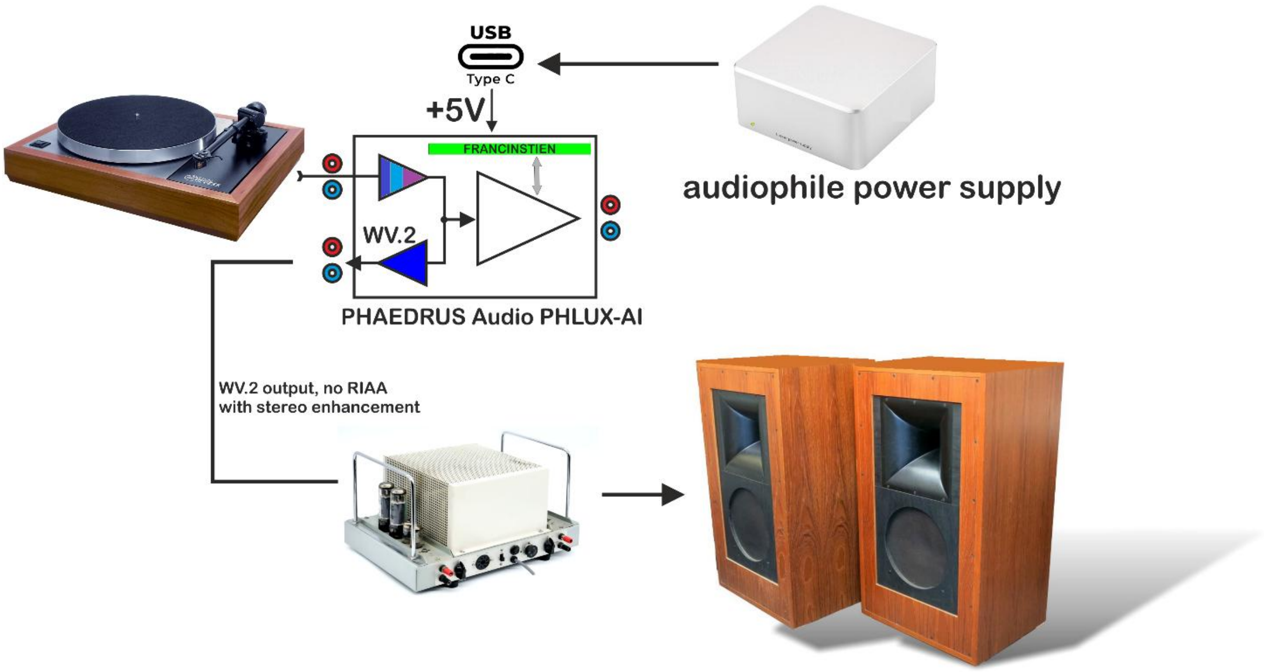
User case #4 – the Record Collector

The **Record Collector** has a collection of valuable LPs and shellac records of classic performances. So, she chose the **RIAA + PHOENIX** expansion option (for shellac records) to maximise the quality possible from this ancient medium. The striped input stage symbol reflects the choice of cartridge options available (at time of order) which include: moving-magnet; moving-coil; strain-gauge; and **PHLUX**-active. Our record collector chose a selection of moving-magnet cartridges, so as to be able to play shellac records, as well as mono and stereo LPs with the appropriate styli.



User case #5 – Strain gauge cartridge user

Our last user case, **(codename: Mr. Zane Gauge)** chose the historical strain-gauge cartridge input stage option. These unusual, semiconductor cartridges were principally manufactured by Panasonic in the 1970s. Some audiophiles consider these cartridges to offer performance that has never been surpassed. (Mr. Zane Gauge is one such***!***) When this option is installed in **PHLUX-AI** preamplifier, no EQ is required at the **RIAA** output – although we recommended the **PHLUX-AI-WV2** output option was purchased. The **OUT** output remains wideband and may be used to decode CD-4 recordings in **Stereo Lab** software (see the **Needle-drop Handbook**).[5]



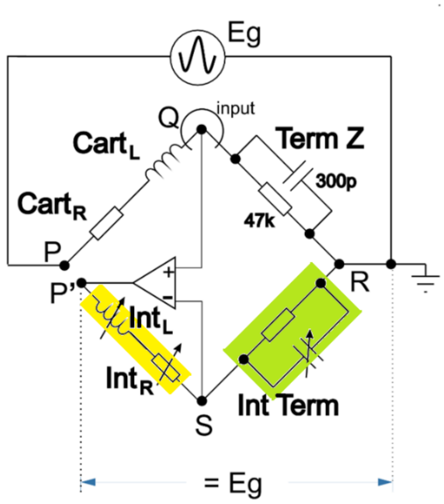
[5] Go to https://pspatialaudio.com/NDH_book2.htm to purchase a copy of the Needle-drop Handbook.

Operating the equipment – the control knob

The function of the single front-panel control knob varies according to the input option fitted. In the case of the moving-magnet input stage, the control knob adjusts the bridge-balance. This is set various ways as explained below. When the moving-coil input stage is fitted, the control knob adjusts the loading presented to the cartridge, which gives subtle voicing shifts. In the case of the **PHLUX** input option, and when the strain-gauge input option is present, the control is a fine level-control. In the case of the preamplifier option, the control acts as a (full range) volume control.

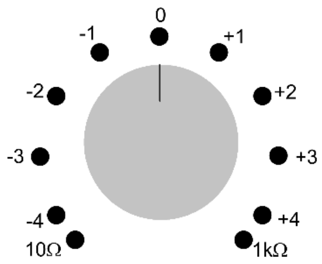
Balancing the bridge – moving-magnet cartridges[6]

Referring to Chapter 3, if we look again at the bridge configuration of the **PHLUX-AI** preamplifier, we can clarify that the preamplifier architecture automatically compensates for ***Term***$_Z$ – the lower right limb of the bridge (marked in green). But it is not possible completely to tune automatically the lower left-hand limb (***Cart***$_Z$, marked in yellow). **Some input is thereby required from the equipment operator.**



There are several (4) ways to set the fine bridge-balance control on the front-panel of the **PHLUX-AI** preamplifier. These are detailed below.

1. It is entirely possible to calibrate the bridge-balance by ear. In fact, this is an excellent approach. It should be noted, this is a *fine balance* adjustment, ***the tonal differences are subtle.*** There is no element of quality expressed in the control calibration: +4 is no better than -4. The numbers simply relate to the degree of adjustment from the nominal balance condition at the zero (12 o'clock) position.



Set up the equipment and play a disc you know well, monitoring via the internal RIAA or iLOOP outputs. Select the control's central, 12 o'clock, position. If the sound is a little "lean" (bass light), rotate the balance control clockwise until you achieve the best tonal balance. If the sound balance is a little "rich" (bass heavy), rotate the control anticlockwise until you achieve the best tonal balance.

If you feel unsure about setting the bridge-balance by ear, method 2 and 3 represent the engineer's approach.

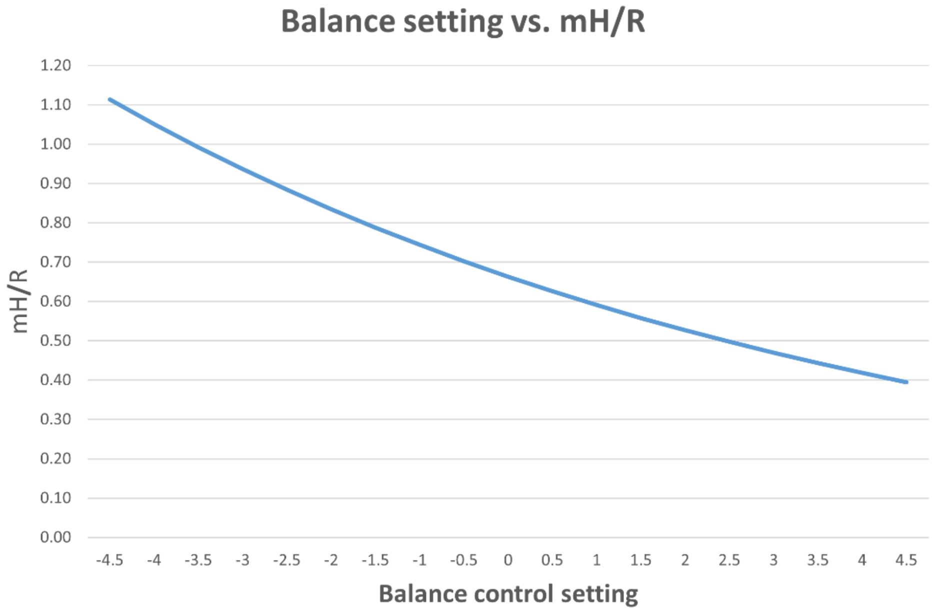
2. Consult the manufacturers' information regarding the cartridge coil resistance and inductance. Once you have these figures, substitute the figures in the following equation to derive a value for the bridge factor, F_b.

---

[6] On early models of the **PHLUX-AI**, the bridge-balance control was simplified to two switches. This restricted the choice of moving-magnet cartridges. The balance control is now a rotary control knob.

F_b = Coil inductance (in mH) / coil resistance (in ohms)

Thus, should a cartridge be specified to have a DC resistance of 800Ω and an inductance of 490mH, F_b will equal 490/800 = 0.61. Select the bridge-balance control position according to the following graph, table, or equation.



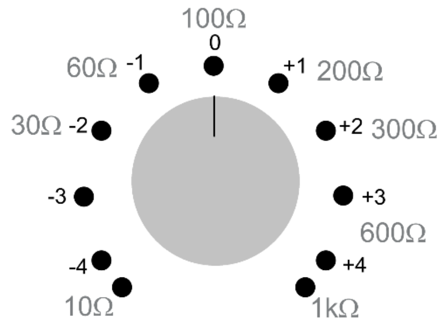
KNOB	mH/R
-4.5	1.11
-4	1.05
-3.5	0.99
-3	0.94
-2.5	0.88
-2	0.83
-1.5	0.79
-1	0.74
-0.5	0.70
0	0.66
0.5	0.63
1	0.59
1.5	0.56
2	0.53
2.5	0.50
3	0.47
3.5	0.44
4	0.42
4.5	0.40

Control position = - [20.log(1.51 × F_b)]

3. We have not always found that cartridge specifications are available, or entirely reliable – due to manufacturing changes and different approaches to measurement. If you, or an engineer colleague or friend, can measure the coil inductance and resistance of the cartridge with a component bridge, then these values should be substituted in the equation for F_b and used to select the fine balance setting accordingly. An example of a modern component bridge is illustrated. These are available from online retailers at a very reasonable price.

4. Finally, ask us at **sales@phaedrus-audio.com**. We have an extensive cartridge database and can advise.

Balancing the bridge – moving-coil cartridges



It is widely believed that input loading adjustment is worthwhile for moving-coil cartridges. To that end, when the **PHLUX-AI** preamplifier has the moving-coil front-end fitted, the bridge-balance knob controls the effect of the input loading on the cartridge by adjusting the mirror internal impedance of the Maxwell bridge.

With the control is in the zero (12 o'clock) position, the cartridge is presented with a 100Ω load. This is the nominal position. At the counterclockwise extreme, the effective load falls to 10Ω, and at the extreme clockwise position, the effective load rises to 1kΩ, as illustrated. The setting is largely subjective, although some manufacturers suggest a load value for their cartridges.

Balancing the bridge – PHLUX active cartridges

When the **PHLUX** front-end is fitted in the **PHLUX-AI** preamplifier, the function of the balance knob alters once again. In this case, there are no reactances to balance within the Maxwell bridge and the control knob acts as a fine level-control (+3.5dB/-6dB). This is especially useful if the internal **SYNOPSIS** soundcard is fitted so that digital dynamic range may be optimised.

Balancing the bridge – Strain-gauge cartridges

When the strain-gauge input option is fitted, there are no reactances to balance within the Maxwell bridge and the control knob acts as a fine level-control (+3.5dB/-6dB).

Chapter 5
Zweikanal Wiedergabeverstärker WV2

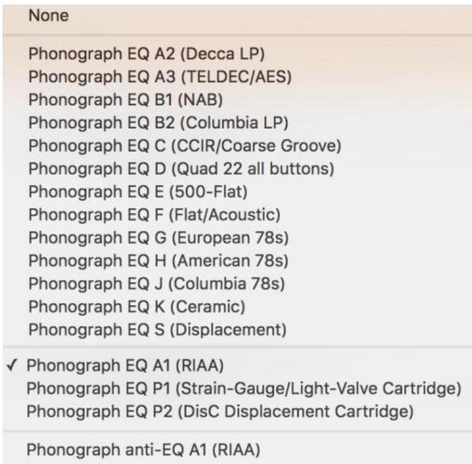
The Neumann WV2 is a vintage Zweikanal-Wiedergabeverstäker (two-channel playback preamplifier). The unit was originally designed for professional vinyl mastering and playback systems. Introduced in 1958, the WV2 was built for demanding, professional use and had a price tag to match (DM1288 - equivalent to about €5000 today). Today, the mono WV1 and stereo WV2 are considered the *Holy grail* of phono preamplifiers, coveted by enthusiasts seeking the best audio fidelity.

The output stage of the WV2 preamplifier is an E88CC triode driven transformer circuit with an anode load formed of the Bv32 choke. The performance of this stage is closely emulated in the **PHLUX-AI WV2 Output** option.

Chapter 6

PHOENIX option for shellac era records

Many recording characteristics were used during the *shellac period*, as described in **Needle-drop Handbook**.[7] For serious archive work, an armoury of different equalisation curves and noise-reduction systems is required for the archiving of shellac era discs. Our high-resolution audio processing program **Stereo Lab** provides the appropriate equalisation curves for shellac discs of all eras – all furnished with a number of intelligent noise reduction systems (see dialogue from the app' below).



But what if you just want to listen in real time to a collection of shellac-era records? That is the rôle of the new **PHOENIX** expansion card for the **PHLUX-AI** preamplifier.

The equalisation curve of the Phædrus Audio **PHOENIX** expansion card was developed with the benefit of Pspatial Audio's years of experience of working with shellac era records. We worked with them and their customers to develop to a compromise equalisation, known as the ***Pspatial Audio Universal Shellac*** (**PAUS**) characteristic which serves for most laterally recorded shellac discs.[8] The **PHOENIX** expansion card also automatically cancels tracing distortion (pinch-effect).

---

[7] Go to https://pspatialaudio.com/NDH_book2.htm to purchase a copy of the **Needle-drop Handbook**.

[8] Use **Stereo Lab** software for the reproduction of hill-and-dale discs. See the **Needle-drop Handbook**. Some manufacturers suggest that a flat-response phono preamplifier is the correct solution to replaying 78s. But this is wrong: shellac era records have all required bass boost (with a turnover of about 300Hz) since the late 1920s. Flat-response preamplifiers have their rôle and are essential when using **Stereo Lab** software equalisation. But they are not the best solution to replaying shellac records in real time.

When this option is fitted, **PAUS** equalisation is selected by pressing the front-panel push-switch to the IN position. The right-hand multicolour LED will indicate WHITE to indicate this EQ is applied.

Chapter 7

FRANCINSTIEN expansion option

The two-channel spatial reproduction system that we know as “stereo” was invented by Alan Blumlein at EMI in suburban London at the turn of the 1930s.[9] Although it took nearly 30 years for Blumlein’s ideas to be recognised and brought to the market as a practical stereo system for LP records,[10] Blumlein’s ideas were remarkably comprehensive. He even identified a weakness in the stereo system, and invented corrective equipment for it. He called this device the *Stereo Shuffler*. But – almost 100 years after he invented it – most stereo systems remain incomplete with his *Shuffler* largely consigned to history.

How stereo works, and what's wrong with it

To understand stereo correction, you must understand a little about three different subjects:

- the physics of sound;
- the psychology of hearing;
- and how recordings are made.

A little physics

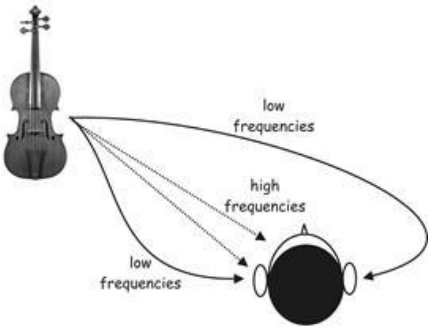
Everyone knows, of course, that sound is waves in air, and that low frequency sounds have long wavelengths and high-frequency sounds have short wavelengths. But most people are surprised by the dimensions of the waves. For example, the lowest note on a bass guitar has a wavelength of 8.5 metres and the wavelength of the highest frequency you can hear is less than 2 cm.

---

[9] Blumlein, A. (1933) British Patent 394,325 June 14th. For a biography of this remarkable man and his untimely death, see https://en.wikipedia.org/wiki/Alan_Blumlein

[10] Clark, Dutton and Vanderlyn (1958) The “Stereosonic” recording and reproducing system: a two-channel system for domestic tape records JAES Vol. 6, number. 8. April 1958

This huge range of wavelengths causes high-frequency sounds to behave differently to low frequency sounds. Essentially, low frequency waves bend around us and aren't absorbed by our bodies. A physicist would say they *diffract*. But high frequency sound waves can't bend around us and our bodies – especially the head – cast an acoustic shadow.



A little psychology

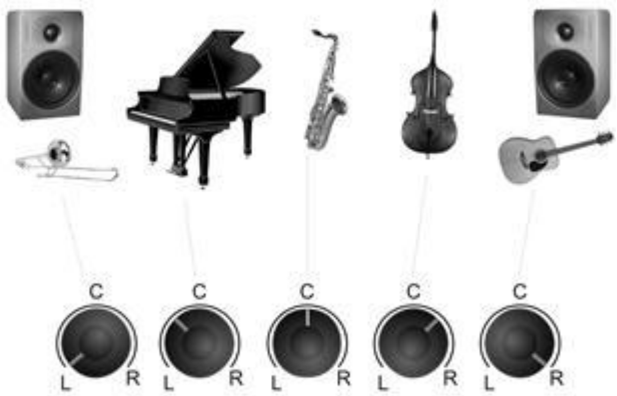
Given this differing behaviour of sounds at low and high frequencies, evolution has provided us with two different processes for determining the direction of sound: one to determine the direction of bass sounds; and another to determine the direction of high-frequency sounds.

A little about stereo recording......

The stereo picture is built up in a recording studio by mixing the signals from the microphones and steering them to a place within the stereo image by means of a panoramic potentiometer or *pan-pot*.

The pan-pot positions the contribution from a particular channel in the stereo picture by the ratio of signal it sends to each channel. So, at the extreme anticlockwise position, the entire signal is sent to the left channel, and the contributing microphone signal will appear to come directly from the left loudspeaker. And at the extreme clockwise position, the opposite will be the case, so that the signal will all be sent to the right channel and the microphone signal will all appear to come from the right-hand loudspeaker.

In between there is a progressive relationship so that, as the pan-pot knob is turned clockwise gradually less signal gets sent to the left, and more to the right and the microphone signal appears to move across the stereo field. In the middle of the pan-pot's travel, equal amounts are sent to right and left and the signal appears to come from the centre of the image.[11]



The problem

Now, you might be a bit surprised, given that we humans possess two hearing systems, one for high frequencies, and one for low frequencies, that such a simple method of constructing a stereo image with a pan-pot wouldn't provide a perfect illusion. And you would be absolutely right.

[11] For the capturing of real sound-fields, Clark et al. (op. cit.) show how the sound-field may be sampled to recreate the appropriate phase-shifts at the listener's ears. There are various techniques. Clark and his team opted for a coincident stereo microphone technique based on crossed figure-of-eight (velocity) microphones.

Essentially a pan-pot steers the low-frequency sounds into one position in the stereo image and steers the high frequencies to another position. The positions are quite close, and that's why stereo has done without correction for 50 years.

The tragedy is that the inventors of the stereo system (Blumlein and the EMI team in the UK) realised this problem when they invented the practical stereo system – nearly a hundred years ago! They even invented a circuit to correct for it. They called it the *Stereo Shuffler*. But their circuit suffered from implementation issues inherent in the technology of the time. It interfered with frequency response and created phase-distortion. So, it was dropped.

And that's meant that Blumlein's stereo system was "broken" – and it has remained that way ever since. Stereophonic sound has never delivered on the promise its inventor had for it nearly a century ago. Rather compromise and "good enough" have ruled the day.

FRANCINSTIEN

Just before the turn of the century, an alternative *Stereo Shuffler* which implemented the same effect by means of inter-channel crosstalk was proposed.[12] Functionally, the technique was identical to the EMI *Stereo Shuffler* but sidestepped the complications and compromises of EMI's implementation. This is the system implemented on the expansion card for the **PHLUX-AI** preamplifier.

What do I hear?

The result of correctly implemented *Stereo Shuffling* is a transformation in the solidity of the stereo image. High and low frequency components of the instruments "lock-in". Voices and instrument take on a precise position in the image and reverberation takes on a 3D character.

Chapter 8

SYNOPSIS internal soundcard

The **SYNOPSIS** internal soundcard consists of a stereo audio CODEC and a USB-compliant full-speed protocol controller. The soundcard is class-compliant and does not require drivers – no matter the computer operating system. The **SYNOPSIS** expansion card employs single-rate, 16-bit Delta-Sigma ADCs and DACs. The analogue to digital conversion equipment is calibrated so that 5cm/s RMS lateral groove velocity modulation at 1kHz indicate -18dBFS peak in both the level meters of each stereo channel. This ensures that the full-scale value

[12] Brice, R. (1997) Multimedia and Virtual Reality Engineering. Newnes and Brice, R. (1998) Music Engineering. Newnes

of the digital system is calibrated to -1dB of absolute maximum record velocity (see chapter eight of the **Needle-drop Handbook**).

When this option is fitted, the rôle of the connectors labelled **OUT** changes. Normally these connectors carry the non-equalised signal to an external soundcard. When the **SYNOPSIS** soundcard is fitted, these connectors carry the soundcard output.

Performance considerations (dynamic range)

In this section, we examine record performance of the **PHLUX-AI** with the internal **SYNOPSIS** soundcard.[13] We say *record performance*, because the only significant long-term performance consideration concerning a soundcard for needle-drop recordings is the performance of the *analogue to digital converter* (ADC). The playback *digital to analogue converter* (DAC) is relevant to what we hear, but DAC performance is usually better than that of the ADC (This is true of the **SYNOPSIS**: the DAC offers a 1-bit of resolution above the ADC performance). Moreover – at a later date – we may even change the DAC to a higher specification device, whereas the recorded signal is fixed forever at the resolution of the ADC.

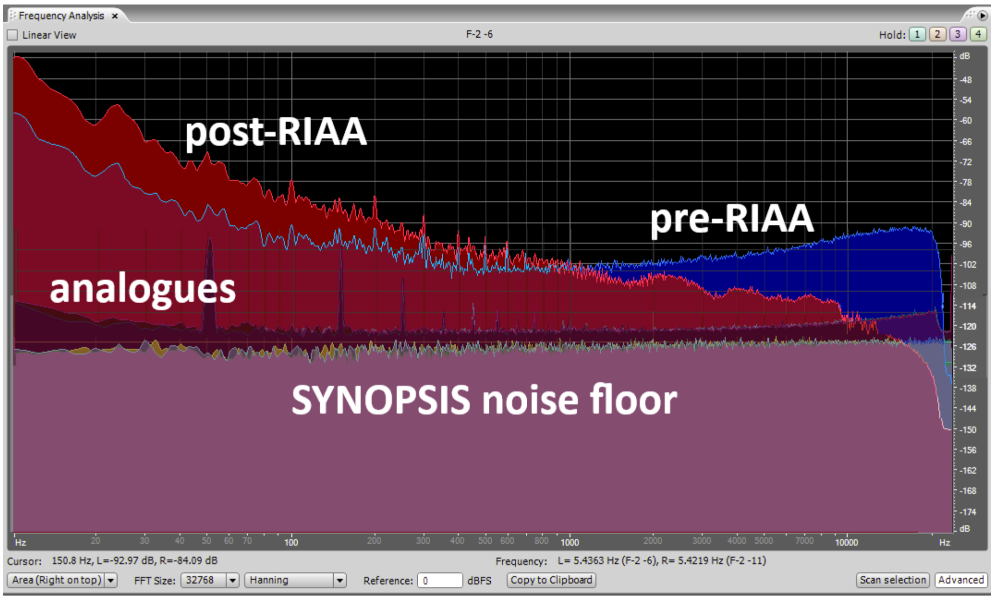
The ADCs in the **SYNOPSIS** soundcard are 16-bit Delta-Sigma (**ΔΣ**), single-rate types, with a theoretical dynamic range of 16 × 6dB = 96dB.[14] You will be unsurprised to know that the noise performance of practical ADC chips does not match their theoretical performance. The 16-bit ADC we chose for the **SYNOPSIS** soundcard offers approximately a 15-bit performance, or ≈ 15 × 6dB = 89dB dynamic range. This is adequate for recordings of LP records provided the analogue to digital conversion is performed prior to RIAA deëmphasis, as it is in the **PHLUX-AI**.

A non-compressed PCM digital signal offers the same resolution at all frequencies. If the resolution is 15-bit, the dynamic range is 90dB at 20Hz and at 20kHz. Therefore, the more spectrally “white” the signal (which is to say, constant energy at all frequencies), the better the resolution of the digital data. Below is a composite spectrogram of the measured **SYNOPSIS** ADC idle noise (purple trace) against a calibrated needle-drop recording of a silent run-in groove from a quiet LP, prior to RIAA equalisation (blue trace) and post RIAA conversion (red trace). The red and the blue trace indicate the noise-floor of the analogue medium.

---

[13] In order to analyse the performance of conversion equipment bridging the analogue and digital worlds, we must establish the relationship between the representations of noise in the time domain – as RMS voltage measured with an AC voltmeter or represented as amplitude values in the DAW waveform display – and the representation of noise in an FFT plot. The reasoning behind the relationship we use is given in an appendix to this manual.

[14] Delta-Sigma (ΔΣ) ADCs are known for their precision and accuracy. At the core of the Delta-Sigma ADC lies the paradigm of oversampling, combined with noise shaping and digital filtering. Oversampling involves sampling the analogue input at a significantly higher rate than the Nyquist rate. This disperses quantisation noise across a broader frequency spectrum – ensuring a substantial portion lies beyond the frequency scope of interest. Following digital filtering (decimation), the sample rate is reduced while out-of-band noise is removed.



If the signal from the record is RIAA equalised prior to conversion, there is a case to be made that 15-bit resolution is inadequate: the red trace dips below the converter noise-floor in the last octave. On the other hand, the unequalised signal remains 20dB or more above the ADC noise-floor throughout the audio band. The resolution of the ADC is an order of magnitude better than the resolution of the medium – *provided it is the raw velocity signal which is digitised*.

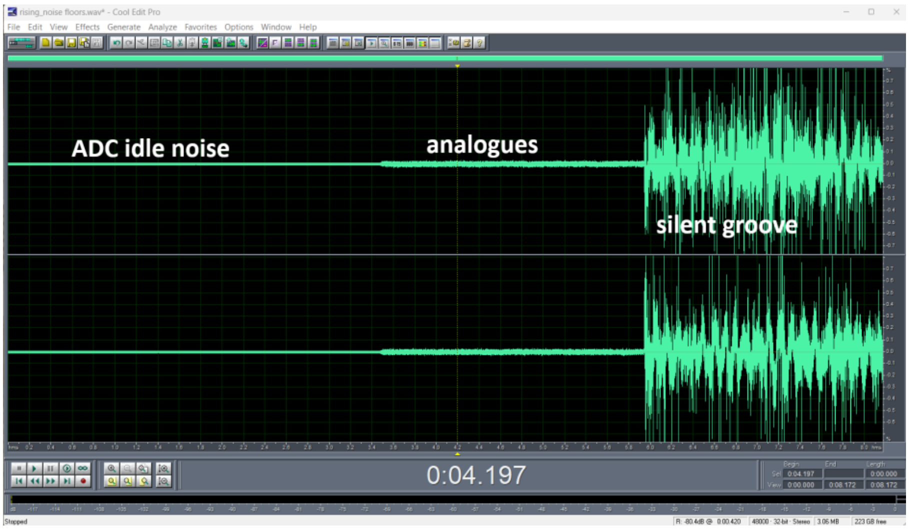
As a further argument, with a 15-bit ADC performance level, the idle noise is considerably lower than the noise performance of the phono cartridge followed by low-noise preamplification. A degree of humility is required in admitting this***!*** As circuit designers, it's normal practice to measure preamplifier circuit noise with the phono input circuit short-circuited and limit the bandwidth of the AC RMS noise measurement from 300Hz to 20kHz (the low frequency limit in place to eliminate mains hum from the measurement). If we do this, we can claim the preamplifier stages in the **PHLUX-AI** have a dynamic range of 93dB - five dB better than the ADC ideal noise. (In fact, we did so this: see *Specifications* chapter).

But measurements made in this way ignore real-world operation. The shadowy trace above the ADC noise-floor in the spectrogram (labelled *analogues*) indicates a real-world analogue noise-floor with tonearm parked, no rotation, and turntable off. Measured in this way, the analogue noise floor is several dB above that of the ADC idle and contains substantial hum components - although these are comfortably below the record surface noise (even pre-RIAA bass boost).[15]

Thus, an ADC with better resolution would lower the idle noise, but this would simply expose more of the noise-floor of the analogue capture system. This phenomenon is most viscerally appreciated when viewed in the time-amplitude domain. The trace below is in three sections: 1) ADC noise; 2) analogue preamp noise; and 3) silent groove-noise.

[15] Being sensitive transducers for the detection of tiny changes in magnetic fields, the Achilles Heel of any electromagnetic cartridge is its susceptibility to mains derived hum.

Note that these findings are all entirely in line with the characterisation of the LP as a medium with a 66dB dynamic range (equivalent to an 11-bit digital system), as derived in afterword nine of the **Needle-drop Handbook**.



Single-rate considerations

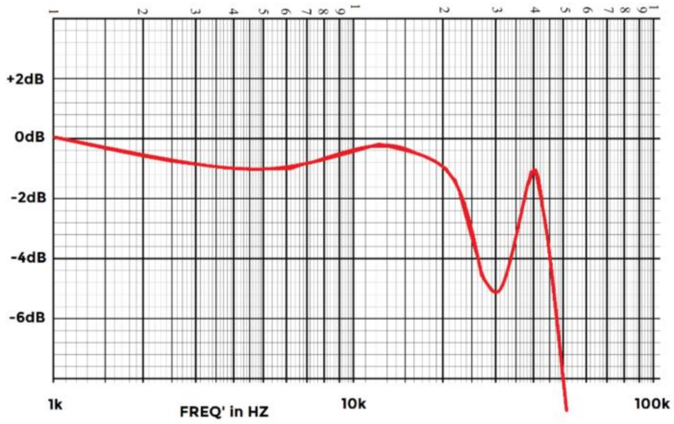
The ADCs in **SYNOPSIS** may be set to run at 32, 44.1, 48 kHz sampling rates. This is an uncontroversial limitation if you want to capture and decode CD-4 supersonic carrier records (for which a workable audio bandwidth of 48kHz is required, see the **Needle-drop Handbook**). For this work, you will need an external soundcard capable of capturing files at 96kHz or above. But how much does single-rate sampling limit the fidelity of normal stereo needle-drops? This question is something of a *cause célèbre* amongst audiophiles. Here, we offer a few thoughts and experimental results.[16]

We generated a spectrogram of a 96kHz needle-drop recording of the outer grooves[17] of a 1975 LP with musical material (two acoustic guitars played with a plectrum) judged to be unusually "bright" and suitable example for testing.[18]

[16] A valuable background reference to this section is *Audibility of a CD-Standard A/D/A Loop Inserted into High-Resolution Audio Playback*. Meyer, E.B. and Moran, D.R. JAES Vol. 55, no 9. September 2007, in which the authors report on a series of double-blind tests comparing the analogue output of high-resolution players playing high-resolution (SACD and DVD-A) recordings with the same signal passed through a 16-bit/44.1-kHz "bottleneck." Their results show that the CD-quality A/D/A loop was undetectable at normal-to-loud listening levels, by any of the subjects, on any of the playback systems. Detectability of the 16/44.1 loop was the same as pure chance: 276 correct answers in 554 trials with 60 subjects; a score of 49.82%.

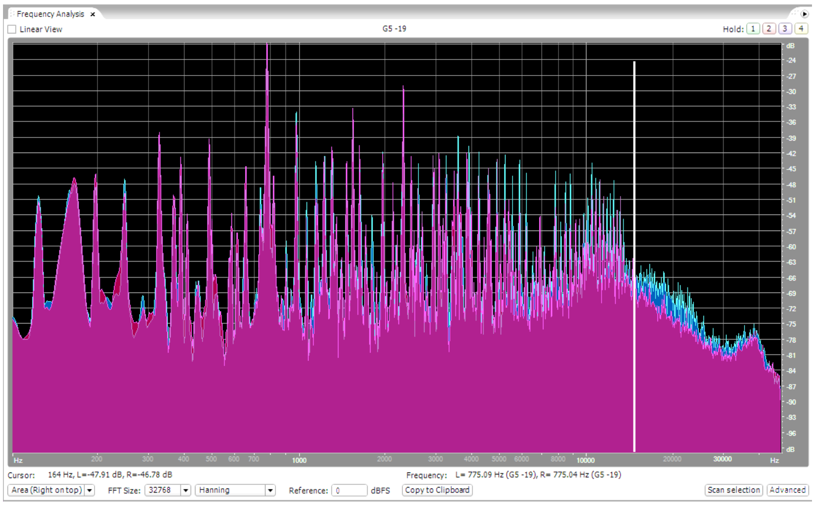
[17] The high frequency loss at inner grooves relative to outer grooves is very substantial in a plastic record – even tracked with a lightweight pickup. The combination of cutting losses (the partial erasure of high frequency information due to the burnishing facets of the cutting stylus), tracing losses (due to the finite size of the stylus, which shunts the energy of the fundamental frequency of the modulation into distortion harmonics), and – above all – deformation losses (due to the yielding of the vinyl material under the force of the stylus) contribute to an inner-groove loss of some 10dB at 20kHz (and infinite attenuation at 25kHz) when replayed with a for a 10µm (0.0035") elliptical stylus tracking at 2.5g. See page 140 of the **Needle-drop Handbook**.

[18] The album was chosen because it was recorded in early 1975 at Studio Sea-Saint Recording Studio in New Orleans – a studio known for state-of the art analogue equipment of the period, and thereby that no digitally



We carefully selected the cartridge too, so that it has an extended ultrasonic amplitude response. We used a **PHLUX-II** cartridge with an elliptical stylus. The measured frequency response of this device is given in the graph. This response was derived from CBS Laboratories STR 120 test record, which was mastered at low speed and has frequency sweeps up to 50kHz.[19]

The results of our needle-drop recording are presented. It's difficult to interpret the



frequency components above 15kHz (right of the vertical white line) as anything much but noise. The recording characteristic is clearly in constant acceleration mode above this frequency,[20] and the spectrum is too uniform and lacking harmonic structure to suggest much musical information. One discernible feature is the "bump" at 38kHz, which we know to be the frequency of the resonance of the effective mass of the cartridge armature of the **PHLUX**-active cartridge bouncing on the compliance of the record material (see Chapter 4 of the **Needle-drop Handbook** and the graph above).

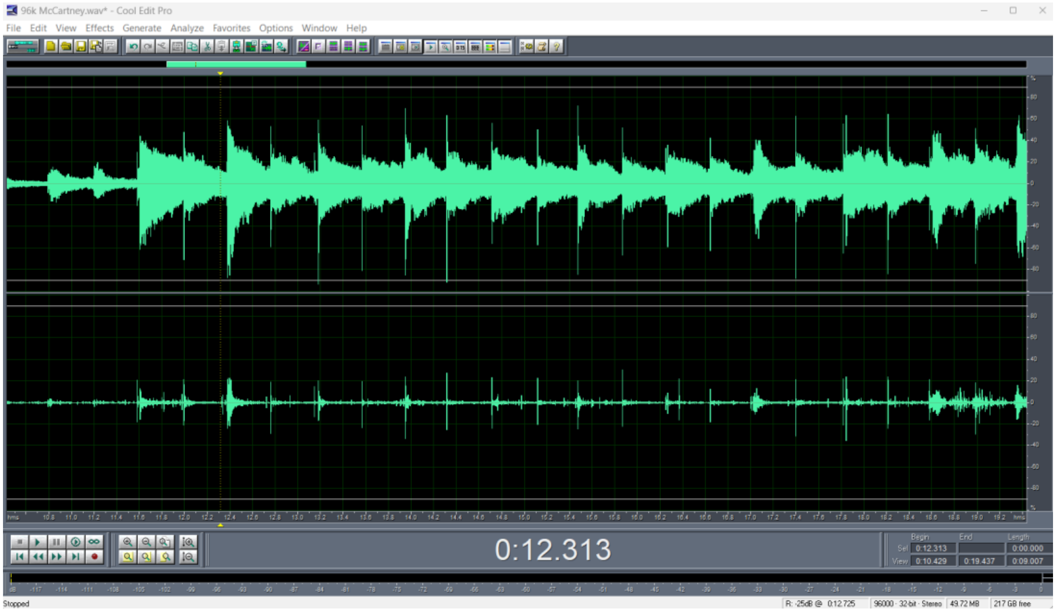
Returning to the time-domain, if we high-pass filtered the recording (with a 16-pole, linear-phase filter with a 20kHz turnover), the amplitude/time waveform indicated information correlated with the original signal. Below we have a comparison of the original waveform (upper, left channel) and the 20kHz high pass filtered version (lower, right channel).

---

imposed Nyquist limit existed in the recording chain. (The first digital recording processor – the SONY PCM-1 – was introduced in 1977.) Our LP pressing has *Porky Prime Cut* inscribed in the dead-wax (of side 1), indicating that the lacquer master was cut by George Peckham, who is widely considered to have been one of the world's best mastering engineers.

[19] The frequency extension of the **PHLUX-II** cartridge is unusual. This is due to its internal amplifier, and the low inertial mass of the moving armature. The design of the moving parts of the PHLUX cartridges derive from work done to recover the 30kHz ±10kHz signals on CD-4 subcarrier records. See the **Needle-drop Handbook** for more information. The frequency response of the **PHLUX-AI** electrical system (including the internal amplifier in the cartridge) was verified to be flat, and only -3dB down at 1MHz.

[20] A falling frequency response from a velocity transducer indicates *constant acceleration*.



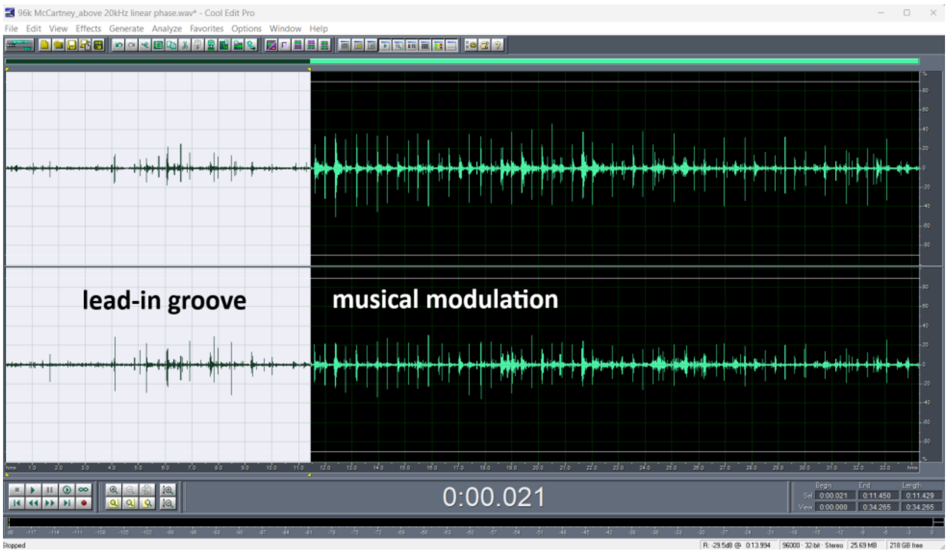
The moot point, of course, is how much of this is real information imparted to the acetate by the cutter, and how much is due to artefacts in the cutting and playback processes?

Given that the high pass filtered version was quite beyond audibility for us,[21] we replayed the high-passed audio at 16kHz sample rate to play it – in effect – at one-sixth speed. The results were interesting, but hardly definitive. The transients (visible in the waveform display above) are audible as noisy "splashes" – like a handclap in a reverberant environment, like a church. It was impossible to judge whether or not these noisy transients were a meaningful addition to the audible experience of listening to a record. It is well known that the upper partials of many musical instruments, include noise-like components, such as key clicks, lip noises, and hissing of air. Was this ultrasonic information due to some percussive effect of the hard plectrum strumming the guitar strings?

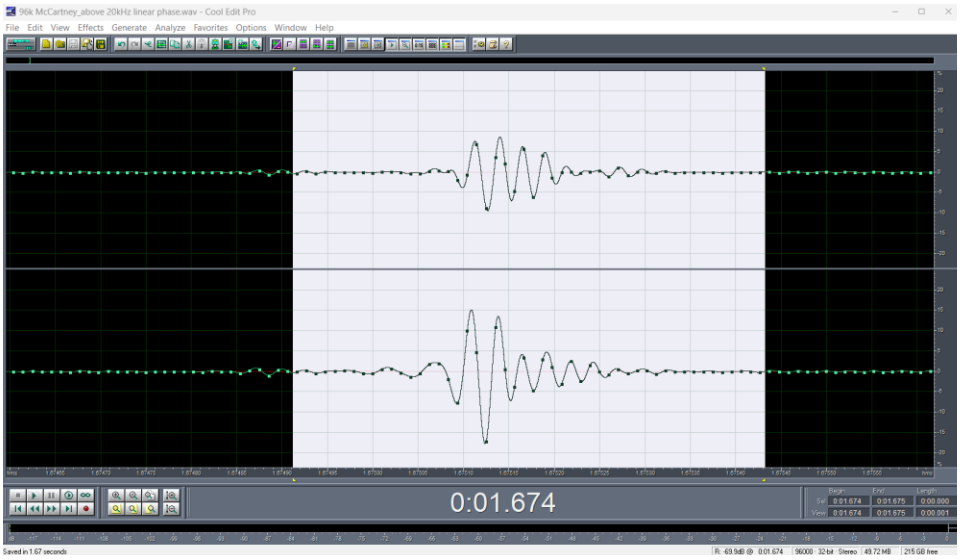
---

[21] The required upper frequency limit of high-quality audio equipment remains controversial, despite substantial and long-standing psychoacoustical evidence for a limit around 15kHz. Nearly 100 years ago, William Snow of Bell Labs determined in a series of careful experiments that the *human audible bandwidth* to be from 40Hz to 15kHz (*Audible Frequency Ranges of Music. Speech, and Music*. Snow, W.B. Acoustical Soc. Am. July 1931). More recent studies with modern audio equipment, have not produced evidence to contradict Snow's early conclusions. Muraoka, T. et al. in *Examination of audio-bandwidth requirements for optimum sound transmission*. Journal Audio Eng. Soc., 29, 2-9. 1981 are explicit on this point and report, "*Extension of frequency range of the audio equipment in recent years, derives an illusion that human audible frequency range has been extended accordingly. However, [our] experiments proved Snow's discovery about 50 years ago.*" Moreover, there is evidence that – any subjective change in audio quality attributed to extended bandwidth audio – is due to intermodulation distortion in the loudspeakers: the superheterodyned distortion products falling in the range below 15kHz. A paper from 2001 (*Detection threshold for tones above 22 kHz* by Kaoru, and Shogo, AES 110th Convention, Amsterdam, May 12–15th) reported, "*All subjects distinguished between sounds with and without ultrasounds* [>22kHz] *only when the stimulus was presented through a single loudspeaker. When the stimulus was divided into six bands of frequencies and presented through 6 loudspeakers in order to reduce intermodulation distortions, no subject could detect any ultrasounds*." An uncharitable, though legitimate, interpretation of Kaoru and Shogo's results would be that the reproduction of frequencies above 20kHz may have a harmful effect on sound quality***!***

For some time, we were stumped on this point – we couldn't see how we could determine if these "splashes" were a legitimate part of the music, or not. Then, we had a stroke of good luck. By choosing an outside track for the needle-drop, we had recorded – quite by chance – plenty of silent lead-in groove before the track started, and we noticed that these "splashes" were provoked as effectively by dust and damage in the silent groove as they were by the musical information in the track.



Clearly, these transients in the lead-in section were not part of the signal sent to the cutter head. One such lead-in "click" is illustrated below, with a magnification of the time axis sufficient to see the individual sampling points.



In the trace, we can see three cycles of oscillation in eight sample periods – provoked by a single "click". At 96kHz, the sample period is 10.41µs. Eight periods = 83µs, leading us to estimate a frequency of 36kHz. Too close to the known surface-armature resonance frequency of the cartridge to be coincidental.

We determined, by inspection, that all the transients in the lead-in grooves and the modulated grooves were of the same oscillatory nature. A screen grab of a train of these transients in the musical track is given below. The presence of the fundamental resonance frequency is evident throughout the example.

It seems therefore that, the “information” above 20kHz in the 96kHz needle-drop capture is largely a result of lower-frequency groove modulation provoking the inevitable resonance of the effective mass of the cartridge armature as it bounces on the compliance of the record material.

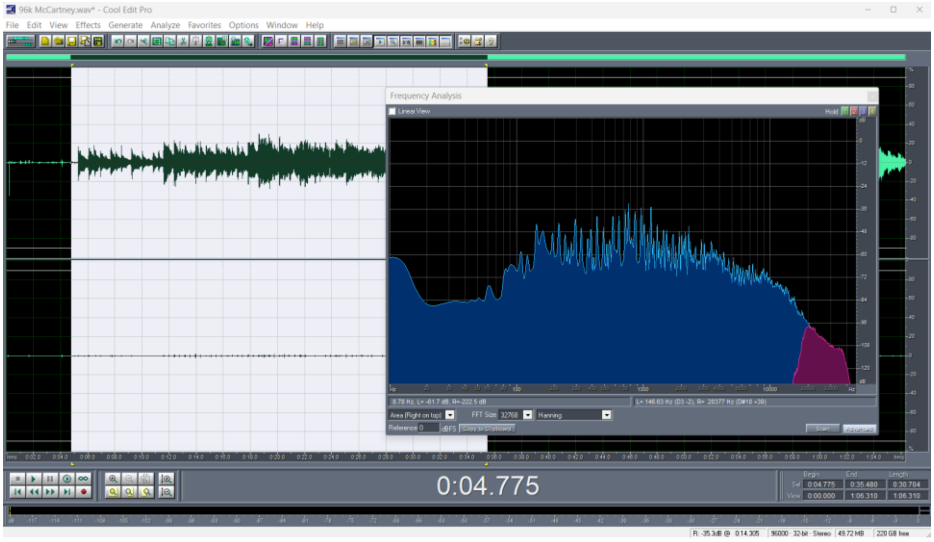
Filtered with a high-Q filter tuned to 38kHz and played back at 1/16th speed, the provoked resonance is clearly audible as a faltering 2kHz tone. It’s quite something being able to listen to the ultrasonic stylus resonance like the tinkling of a little bell***!*** And it demonstrates palpably that the ultrasonic information has nothing whatsoever to do with the recorded signal – except insofar as it is provoked by frequency components within the normal audio bandwidth.

RIAA 75µs deëmphasis

All our efforts to isolate and make audible the frequency range above 20kHz from the pick-up have ignored the effects of replay deëmphasis which is a fundamental, practical part of the replay from an LP record. In so doing, we have greatly exaggerated the magnitude of the effects by concentrating on the velocity signal direct from the cartridge. This was deliberate, in order to study them. But, if we want to understand the audibly of the end-result of these effects, we must apply high frequency deëmphasis.

The artefacts we have discovered around 38kHz are some two octaves above 10kHz which we know to be attenuated by 13.7dB relative to 1kHz in the RIAA replay characteristic. We may therefore add nearly another 12dB of attenuation to allow for a further two octaves of attenuation at 6dB/octave. (In fact, the exact attenuation figure at 38kHz is -25dB).

If we apply 75µs deëmphasis to the recorded data, the results (in the time and frequency domain) look like this. The spectrogram of the high pass filtered track is the purple trace in the spectrogram.



Listening (once again, at 1/6th speed), the attenuated information >20kHz is very quiet indeed. The RMS level of the high pass filtered version is -76dBFS, relative to -27dBFS for full frequency version. In tangible terms, the power of the signal above 20kHz is 0.001% of the total power of the signal.

We created the high-pass (>20kHz) and low-pass (<20kHz) versions of the track as two separate (stereo) tracks in a multitrack session in the DAW and solo-ed the tracks to audition them separately and combined. Even slowed down by a factor of six, the high frequency components are at such a low level compared with the programme below 20kHz that they are masked by the low-passed signal and their effect is inaudible.

We can make a couple of observations with certainty.

1) The information that exists above 20kHz in a stereo needle-drop is of a noise-like character. It appears consist largely of oscillatory bursts of the resonance of the effective mass of the cartridge armature with the compliance of the record material. It's only correlation with the wanted information from the record appears to be that this oscillation is provoked by groove modulation frequencies within the conventionally defined audio band.

2) Attenuation due to playback deëmphasis is so significant in this frequency range that we have no evidence that this ultrasonic information is audible when auditioned in the normal way.

Performance considerations – conclusions

We don't aim to be prescriptive. We are sure that many users will continue to insist that 96kHz (perhaps even 192kHz) sampling and 24-bit resolution are required to capture the "analogue magic" of the LP format. They will want to use an external soundcard rather than adopt the internal **SYNOPSIS** solution. Our conclusion, gained from the experimental data above, is that there is no evidence that 16-bit, single-rate sampled needle-drops has insufficient resolution or frequency extension to capture all the significant information stored in the grooves of a LP record.[22]

USB power and standby operation

The **SYNOPSIS** soundcard – like the rest of the **PHLUX-AI** preamplifier is powered from the five volts present on the USB interface (known as USB V_{BUS}). When an attached computer goes into sleep mode, the behaviour of the USB V_{BUS} depends on a host of design decisions and system settings. These include BIOS/UEFI configuration, and the specific hardware implementation. Refer to your computer manufacturer for information.

The standby power consumption of the **PHLUX-AI** preamplifier is compatible with European Commission's Regulation (EU) 2023/826 with the new rules from 2025, in which power consumption must be less than ½ Watt in all standby configurations. The only circuit option which causes power dissipation to exceed ½ watt is the internal **SYNOPSIS** soundcard. When a valid USB audio computer audio session has terminated and there exists a constant idle state on the USB data bus, the **SYNOPSIS** soundcard enters a suspend state, which reduces overall power consumption of the **PHLUX-AI** to below ½ watt. It remains that way whilst there exists an idle state on the USB data bus. When a new, valid audio session is asserted, the CODEC and protocol controller wake up almost immediately.

---

[22] A little historical perspective reinforces this point. In September 1972, the audio circuits from Broadcasting House, the BBC's home in London, to the Wrotham FM radio transmitter in Kent were switched to the new BBC Pulse Code Modulation system which was based on sampling the input analogue audio waveform at 32k samples/sec and generating 13-bit LPCM (Linear PCM) sample values. (See: BBC Engineering Design Information. *BBC Designs Department 10155(2)* September 1974. The complete PCM system supported 13 mono channels. The resultant 6336kbit/s multiplex was conveyed on analogue "vision" circuits provided by the Post Office, and https://www.audiomisc.co.uk/BBC/PCMandNICAM/History.html). The sampling was dithered to suppress quantisation effects, and the system was measured to have better than 80dB dynamic range. The input to each analogue-to-digital convertor (ADC) was low-pass filtered to reject audio frequencies above 15kHz. (See: *Idle-channel noise in p.c.m. sound-signal systems*. Ely, S.R. Research Department, Engineering Division THE BRITISH BROADCASTING CORPORATION Report: BBC RD 1978/4 February 1978. Ely measured 80dB end to end dynamic range. A very good result. This was accomplished with the use of 50µs pre- and deëmphasis. Note how this has direct parallel with the digitisation of records prior to deëmphasis.) Provided the listener had a good FM receiver and a decent signal, the system was widely considered to offer audio performance better than any domestic audio medium of the time – both LP disc and tape. In the November 1972 issue of Hi-Fi News & Record Review magazine, the journalist Angus McKenzie commented on the impact of the new PCM system and declared, "*Those capable of receiving the BBC PCM [fed] stereo radio are almost certainly hearing the finest sound quality available in the world*."

Preamplifier option

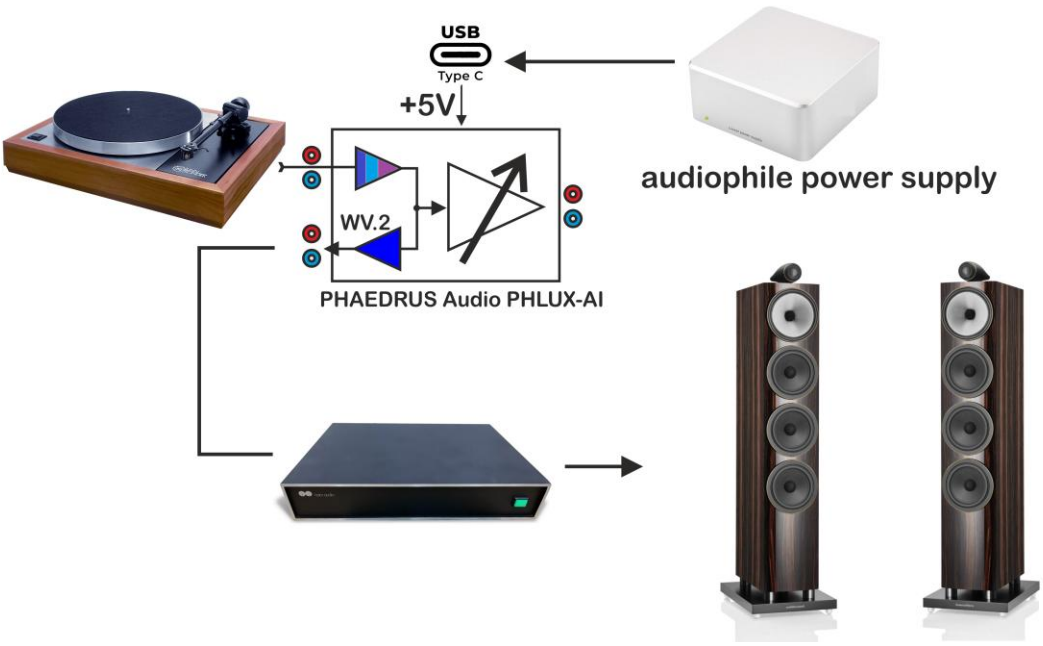
The preamplifier option for the **PHLUX-AI** adapts the control knob to operate as a high-precision volume control (of advanced design) and includes a bundle of options to provide a complete control preamplifier. The options included are:

PHLUX-AI-WV2 - Neumann WV2 output stage (5cm/s velocity → -6dBu)
EXP-RIAA – Expansion card to add precision RIAA equalised output
EXP-FRANC – Stereo “shuffling” enhancement

And either:

PHLUX-AI-MC0 - Optional hi-Z support (5mV sensitivity)
PHLUX-AI-MC20 - Optional MC support for PHLUX-AI (500µV sensitivity)
PHLUX-AI-MC30 - Optional MC support for PHLUX-AI (150µV sensitivity)
PHLUX-AI-PHLUX - Optional PHLUX support
PHLUX-AI-STRAIN - Optional strain-gauge support

A system drawing of how the preamplifier option is used is given. Note, the **PHLUX-AI** feeds the power amplifier directly.



Note: The SYNOPSIS soundcard is not available in the preamplifier configuration.

Specifications

PHLUX-AI Preamplifier

Size: PHLUX-AI: 112 × 50 × 225 mm (W-H-L)
Preamplifier Supply: 5V ±5% via USB Type-C connector.
Power: Compatible with European Commission's Regulation (EU) 2023/826 with the new rules from 2025. Standby power consumption is less than ½ Watt in all configurations.
Function: Line output to external computer soundcard
Sensitivity @ 5cm/s recorded velocity: Nominal 5mV RMS (-44dBu)
Input load: Active (Bridge-node)
Gain: Nominal 24dB to OUT
Signal to noise (rel. 5cm/s velocity, ENBW RIAA): 68dB*
Headroom (rel. 5cm/s velocity): 23dB
*Input must be dummy MM load: *do not short input* for this measurement.

RIAA output

Function: Adds RIAA-equalised output at middle-pair RIAA connectors
Gain (to RIAA output): 26dB at 1kHz
Noise figure (all configurations, unweighted in 20kHz): 2dB, this being $\mathbf{SNR_{out}/SNR_{in}}$
Headroom (rel. 5cm/s velocity): 23dB

WV2 output

Function: Adds last-stage of Neumann WV2 playback equaliser: raises the level of the RIAA-equalised to full operating line-level output (standard level → -6dBu)
Gain (to RIAA output): 38dB at 1kHz
Noise figure (all configurations, unweighted in 20kHz): 2dB, this being $\mathbf{SNR_{out}/SNR_{in}}$
Headroom (rel. 5cm/s velocity): 23dB
Rumble filter: -1dB@30Hz; -3dB@22Hz; -15dB@10Hz; -40dB@3Hz
Maximum output signal level at RIAA output: +18dBu.

MC20 Moving coil cartridge option

Function: Increases input sensitivity by +20dB
Input load: Nominal 100Ω (adjustable with control-knob)
Sensitivity: Nominal 500µV RMS (-64dBu)
Signal to noise (rel. 5cm/s velocity, unweighted in 20kHz): 70dB to RIAA and MAIN output

MC30 Moving coil cartridge option

Function: Increases sensitivity of input by 30dB
Input load: Nominal 100Ω (adjustable with control-knob)
Sensitivity: Nominal 150µV RMS (-74dBu)
Signal to noise (rel. 5cm/s velocity, unweighted in 20kHz): 68dB RIAA and MAIN output†
† This represents a class-leading unweighted equivalent input noise of 70nV or -141dBu

MC0 Moving coil cartridge option

Function: Provides high-Z input for MI type cartridges: no increase in sensitivity
Input load: 47kΩ nominal
Sensitivity: Nominal 5mV RMS (-44dBu)
Signal to noise (rel. 5cm/s velocity, unweighted in 20kHz): 74dB RIAA and MAIN output

PHLUX active cartridge option

Function: Phantom supply provided at input connector
Input load: >47kΩ nominal
Sensitivity: Nominal 5mV RMS (-44dBu)
Signal to noise (rel. 5cm/s velocity, unweighted in 20kHz): 74dB RIAA and MAIN output

Strain gauge semiconductor cartridge option

Function: Phantom supply & LH signal inversion
Input load: >47kΩ nominal
Sensitivity: Nominal 5mV RMS (-44dBu)
Signal to noise (rel. 5cm/s velocity, unweighted in 20kHz): Depends on cartridge model

SYNOPSIS soundcard option

Function: Stereo audio CODEC and USB-compliant full-speed protocol controller
Drivers: Class-compliant, irrespective of the computer operating system. The **SYNOPSIS** USB
ADC/DAC technology: Single-rate, 16-bit Delta-Sigma (**ΔΣ**) ADCs and DACs
Calibration: 5cm/s RMS lateral groove velocity indicate -18dBFS peak (stereo)
DAC resolution: 93dB
ADC resolution: 89dB

Phædrus Audio reserves the right to change specifications without notice.

Chapter 11

Warranty and service

PLEASE RETAIN YOUR SALES RECEIPT. IT IS YOUR PROOF OF PURCHASE COVERING YOUR LIMITED WARRANTY. LIMITED WARRANTY IS VOID WITHOUT SUCH PROOF OF PURCHASE.

Phædrus Audio's Limited Warranty

Warranty service conditions are subject to change without notice. For the latest warranty terms and conditions and additional information regarding Phædrus Audio limited warranty, please see complete details online at www.phaedrus-audio.com.

Appendix 1 - Model Codes/SKUs

The model/order codes for **PHLUX-AI** preamplifier are:

PHLUX-AI - Active interface phono preamp for moving-magnet cartridges
PHLUX-AI-MC0 - Optional hi-Z support (5mV sensitivity)
PHLUX-AI-MC20 - Optional MC support for PHLUX-AI (500µV sensitivity)
PHLUX-AI-MC30 - Optional MC support for PHLUX-AI (150µV sensitivity)
PHLUX-AI-PHLUX - Optional PHLUX support
PHLUX-AI-STRAIN - Optional strain-gauge support
EXP-SYNOPSIS – Optional internal soundcard
EXP-RIAA – Optional expansion card to add precision RIAA equalised output
PHLUX-AI-WV2 – Optional Neumann WV2 output stage (5cm/s velocity → -6dBu)
EXP-PHOENIX – Expansion EQ replay of shellac era records (requires **EXP-RIAA** option)
EXP-FRANC – Optional stereo "shuffling" enhancement (requires **EXP-RIAA** option)
PHLUX-AI-PRE – Volume control and option bundle for complete preamplifier solution

Appendix 2 – Why moving-magnet cartridges continue to be relevant

Most vinyl-record enthusiasts share a belief – enthusiastically endorsed by cartridge manufacturers – that the more expensive, and more profitable, moving-coil type cartridges outperform the moving-magnet type. This is a simplification bordering on misrepresentation. Market competition was so fierce during the period when records were a mass-market commodity that all the margin was squeezed out of the more popular moving-magnet type cartridge leaving the moving-coil models as the only product in which manufacturers could invest and hope for a reasonable return. History forced the moving-magnet to dominate the lower-end of the cartridge market leaving moving-coils to remain unchallenged at the top-end.

It was not always so. In the 1950s, ceramic (crystal) cartridges were the norm in the home and moving-coil cartridges were standard in broadcasting and in other high-quality applications. It was considered a very great step forward when the American cartridge manufacturer Shure Brothers introduced the first, modern moving-magnet cartridge.[23] As Ben Bauer explains in a paper[24] describing the development of the phono cartridge,

> *...By devising a moving-magnet pickup, in which the traditional roles of armature and magnet were reversed, it became possible to lower the ..... stylus bearing force.. [to]*

---

[23] The originators of the moving magnet principle were ELECTROACUSTIC GmbH in Kiel, Germany. They patented the idea in 1957 and licensed the concept to Shure.

[24] The High-Fidelity Phonograph Transducer. Bauer, B. JAES October/November 1977, Volume 25, Number 10/11

one-gram. [The first time this was] achieved in a practical, commercial device... Its tracking capability set a new standard which has not been exceeded to this very day.

But all was not perfect in the land of moving magnets. It is easy to demonstrate that the electrical characteristics of the moving-magnet cartridge set a lower bound on noise and variability of frequency and phase response due to the internal impedance of the electrical generator interacting with the reactance of the cables and preamplifier (see chapter four of the **Needle-drop Handbook**). **The moving-coil cartridge is unarguably the better electrical generator**. This type is less affected by interface impedances.

But, as explained in chapter four of the **Needle-drop Handbook**,[25] the performance of any phonograph cartridge ultimately relies on the stylus' ability to remain in contact with the walls of the groove, which is referred to as the cartridge's ability to *track the groove*. The greater part of this accomplishment lies in the design of the moving part of the cartridge which must be made as light as possible – especially its inertial effect at the stylus point. Good tracking performance is a prerequisite, not only for good reproduction, but for the responsible care of records too, because the result of poor tracking is, not only information loss, but quite possibly, permanent damage to the medium.

Vinylistas' dilemma

The vinylista is faced with a dilemma because the moving-coil cartridge type, with its better electrical performance, does not generally offer superior tracking performance due to the typically greater mass of the moving armature. To support this assertion, we gave the results of our own tests and the study of the data from *Hi-Fi World* magazine technical reviews in the **Needle-drop Handbook**, which confirmed that moving-coil cartridges do not (generally) outperform moving-magnet types in terms of tracking. Serious archivists have known for many years the moving-magnet cartridge type is superior in terms of its tracking ability. For example, the sound archivists for the American Library of Congress considered the tracking performance of Shure's V15 moving-magnet cartridge so highly that they purchased all remaining stock of the V15 line from Shure in 2009 when the manufacturer stopped production of this cartridge.

Pspatial Audio and our partners Phædrus Audio developed the active **PHLUX** cartridges to address the electrical limitations of the moving-magnet cartridge. It's legitimate to see the **PHLUX** cartridge as a development of the moving magnet type in which the various electrical compromises of this pickup type are addressed. But some customers asked us for a solution so that they could use existing (non-active) moving-magnet cartridges – especially those historical models known and admired for archive work. It is for those customers that we developed the original bridge-based preamplifier. The technique secures all the advantages of the **PHLUX** cartridges without active electronics in the cartridge or headshell.

Moving-magnet cartridge support

The bridge adjustment-range for variable-field cartridge types (moving-magnet, variable reluctance, or moving-iron) is wide, but not infinite. All historical and current production types due to Audio Technica, Ortofon, Shure, Nagaoka, Goldring, Clearaudio, Rega, the

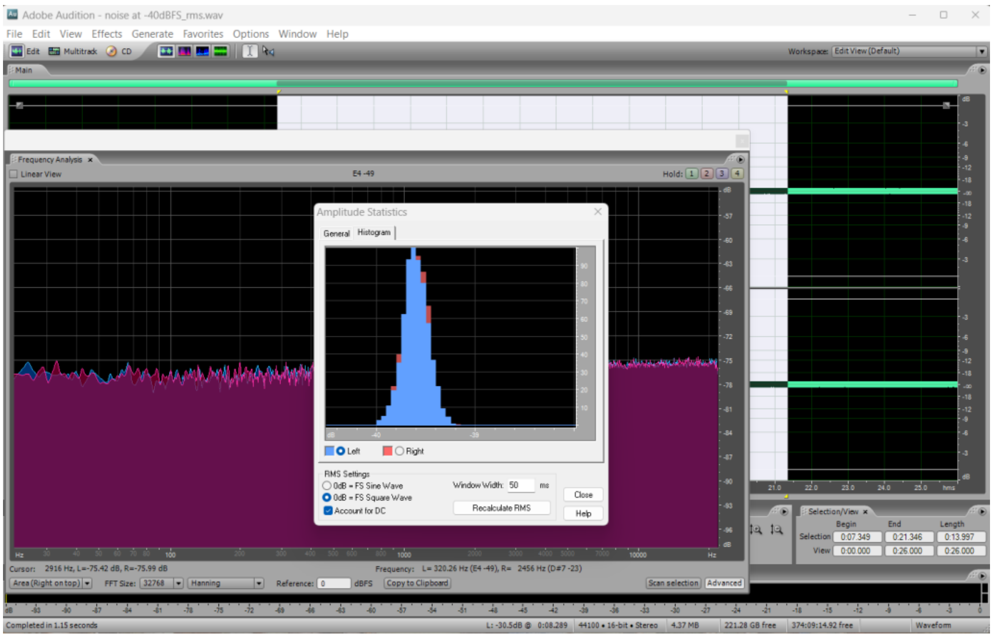
[25] Go to https://pspatialaudio.com/NDH_book2.htm to purchase a copy of the Needle-drop Handbook.

many OEM models based on Excel Sound Corp. products are all supported, as are many others. However, as an example, the moving-iron types due to Grado are not supported – the ratio of inductance to resistance in these models is too extreme for the range of the balancing bridge in the **PHLUX-AI** preamplifier. These cartridges require the **PHLUX-AI-MC0** option to be fitted. Contact us at **sales@phaedrus-audio.com** to discuss.

Appendix 3 - Relationship between the amplitude noise and its representation in a Fourier Transform plot

In order meaningfully to analyse the performance of conversion equipment bridging the analogue and digital worlds, we need to understand the relationship between the representations of noise in the amplitude domain – as RMS voltages measured with an AC voltmeter or represented as amplitude values in the DAW waveform display – and the representation of noise in a spectrum analysis as an FFT plot.[26]

Our first step in investigating the relationship was to generate a file of stereo white noise with a peak reading (on the DAW meters) of -30.5dB peak. The file was digitally generated at 44.1kHz sampling rate (***fs*** = 44100). We know that the RMS value of white noise is 9.5dB below measured peak, so this is RMS noise voltage at -40dBFS level (see afterword one of the **Needle-drop Handbook**). This was confirmed using the statistical analysis in the DAW, which indicated the RMS value of the noise was indeed at -40dBFS level (below).



This signal was then FFT analysed at various FFT *sizes* (***n***). The following results were taken from the screen metric. See illustration of the DAW screen (which also illustrates the statistical amplitude analysis). The results are given in the following table.

[26] In considering noise in this section, we only consider noise that is spectrally "white", which is to say that it has equal energy at all frequencies. The Fourier transform of noise of this type produces a horizontal noise-floor in a transform plot.

FFT size (n)	FFT result
65536	-78dB
32768	-75dB
16384	-72dB
8192	-70dB (ish)
4096	-67dB
2048	-64dB
1024	-61dB
512	-59dB
256	-56dB
128	-53dB

Two observations:

1) The FFT bin noise level is always considerably lower than the total noise RMS level. This is to be expected because the FFT divides our total noise up into frequency *bins* [27] – but what is the mathematical relationship?

2) Increasing FFT size (***n***) of the noise signal lowers the noise floor, with a *3dB drop in noise-floor for each doubling of FFT size*.

In fact, there is a known mathematical relationship between noise-floor (*per Hertz*) and RMS level in a given bandwidth (***B***) which is,

Noise-floor (dBFS/Hz)= RMS noise in dBFS - 10 log(***B***) dBFS/Hz

Applying this to our noise file,

10 log (22050) = 43,

So, our noise-floor is,

= -40 - 43 = -83.4 dBFS/Hz   This is the figure "***per Hz***".

But, in a practical FFT analysis, the noise is divided into *bins*, where each bin has a width of: ***δf = fs/n***, where ***fs*** is the sampling rate and ***n*** is the FFT length. We must allow for this "binning" of the noise. Let's take an example of a FFT size n = 32768. We know the bandwidth of each bin

= 44.1kHz/32768 = 1.35Hz/ bin.

[27] The discrete form of the Fourier Transform is used to transform a series of discrete, sampled, amplitude points in the time domain into a series of *discrete frequencies* in the frequency domain. The exact frequencies of the different analysis points in the DFT output depend upon the sampling rate of the time-based signal (***fs***) and the number of samples included in the analysis (***n***). The result of trying to force the real world into the straitjacket of the Discrete Fourier Transform is termed, *leakage*: a signal which falls anywhere but exactly upon the analysis frequencies, will "leak" into all the other, adjacent frequency analysis points. In DFT terminology, the frequency analysis points are termed, *frequency bins*.

Our bins are wider than 1Hz. And, because there's a bit more than a "Hertz-worth" in each bin, we can expect the recorded noise level to be a bit higher in each bin. Expressed as a power ratio relative to 1Hz, 1.35/1 = 1.3dB. This figure must be *added* to our dBFS/Hz figure to allow for bin size This, the complete calculation becomes,

= -40 - 43 = -83.4 dBFS/Hz + 1.3dB, to give = -82.1 dBFS/bin.

In short, for a 44.1 kHz sample rate and 32768-point FFT, the expected noise floor in the FFT should be around -82.1 dBFS per FFT bin. How do we reconcile this with the -75dB figure we see in the practical FFT analysis in the table? There's an "error" of ≈ 7dB.

Let's take another example: the 2048 FFT. Here, the noise per Hertz figure doesn't change. So,

= -40 - 43 = -83.4 dBFS/Hz

But each frequency bin is 21.5Hz wide. That will add 10 log (21.5) = 13.33 to each bin to give a theoretical noise-floor per bin of,

= -40 - 43 = -83.4 dBFS/Hz + 13.33dB, to give = -70.1 dBFS/bin.

Once again, an offset exists between the theoretical value and the recorded result in the FFT window. We might attribute some of this to the rounding of calculations and to experimental error in reading the (very cramped) FFT ordinate axis. But there appears to be a consistent offset of around 6dB. Let's work through all the results we measured against the theoretical calculation. We used a spreadsheet to do this, and the results are given below; the most significant being the columns labelled **Error,** which indicates the error between the mathematical prediction (**Theory**) and the measured value (**FFT result**). The workings illustrate that the error (last column) is always 6dB ±1dB.

Relationship between RMS white noise figure and FFT representation

FFT size (n)	FFT result	RMS	Fs	Level/Hz	Hz/bin	Bin adj	Theory	Error
65536	-78dB	-40	44100	-83.43	0.67	-1.72	-85.15	7.15
32768	-75dB	-40	44100	-83.43	1.35	1.29	-82.14	7.14
16384	-72dB	-40	44100	-83.43	2.69	4.30	-79.13	7.13
8192	-70dB (ish)	-40	44100	-83.43	5.38	7.31	-76.12	6.12
4096	-67dB	-40	44100	-83.43	10.77	10.32	-73.11	6.11
2048	-64dB	-40	44100	-83.43	21.53	13.33	-70.10	6.10
1024	-61dB	-40	44100	-83.43	43.07	16.34	-67.09	6.09
512	-59dB	-40	44100	-83.43	86.13	19.35	-64.08	5.08
256	-56dB	-40	44100	-83.43	172.27	22.36	-61.07	5.07
128	-53dB	-40	44100	-83.43	344.53	25.37	-58.06	5.06

We don't know exactly how the FFT calculation is made in the software we use (Adobe *Audition*), but, very likely, the FFT is taking a *direct magnitude value* rather than a *power value*. This would lead to values 6 dB higher than the theoretical figures given above.

Importantly, further experiments established that altering the magnitude of the noise in the amplitude domain always results in an identical level shift in the amplitude vs. frequency domain. So, if we raise the level of the noise by 12dB, the transform noise-floor level rises by 12dB too. Thus, all we have to do as we convert between RMS noise and the FFT representation is to allow for the offset which causes the FFT result to read 6dB higher than the theoretical figure. The take-away information is the following:

To get a reliable link between the noise-floor value recorded in the (*Adobe Audition*) FFT and amplitude in the time-domain, it is most convenient to use an FFT size which is approximately equal to the sample rate/second so as to get close to a true noise per Hertz (dBFS/Hz) figure. To simplify even further, provided we use an FFT size of 32k (***n*** = 32768) at a sampling rate of 44.1kHz (or 48kHz, see footnote)[28], we can say that the RMS noise voltage (relative to 0dBFS) is 35dB higher than the measured noise-floor in the FFT plot in Adobe *Audition*. This is the rule we applied in our description of the **SYNOPSIS** soundcard in chapter eight.

---

[28] It's worth working through this calculation with a sampling frequency of 48kHz. The change results in an almost identical value – the *noise per Hz* figure is a bit lower, but the *Hz per bin* figure is higher. This is convenient because it means that we may the same rules-of-thumb for both single-rate sampling frequencies.

Appendix 4 - Declaration of Conformity

The Manufacturer of the Products covered by this Declaration is

Phædrus Audio
Maidstone
Kent
UK

The directives covered by this declaration are:
2014/30/EU Electromagnetic Compatibility directive
2014/35/EU Low Voltage Equipment directive

The products covered by this declaration are:
PHLUX-AI active interface phono preamplifier

The basis on which conformity is being declared: The manufacturer hereby declares that the products identified above comply with the protection requirements of the EMC directive and with the principal elements of the safety objectives of the Low Voltage Equipment directive, and that the following standards have been applied:

IEC INTERNATIONAL STANDARD 60065: 2005. - Audio, video and similar electronic apparatus – Safety requirements

The technical documentation required to demonstrate that the products meet the requirements of the Low Voltage Equipment directive has been compiled and is available for inspection by the relevant enforcement authorities. The CE mark was first applied in 2025.

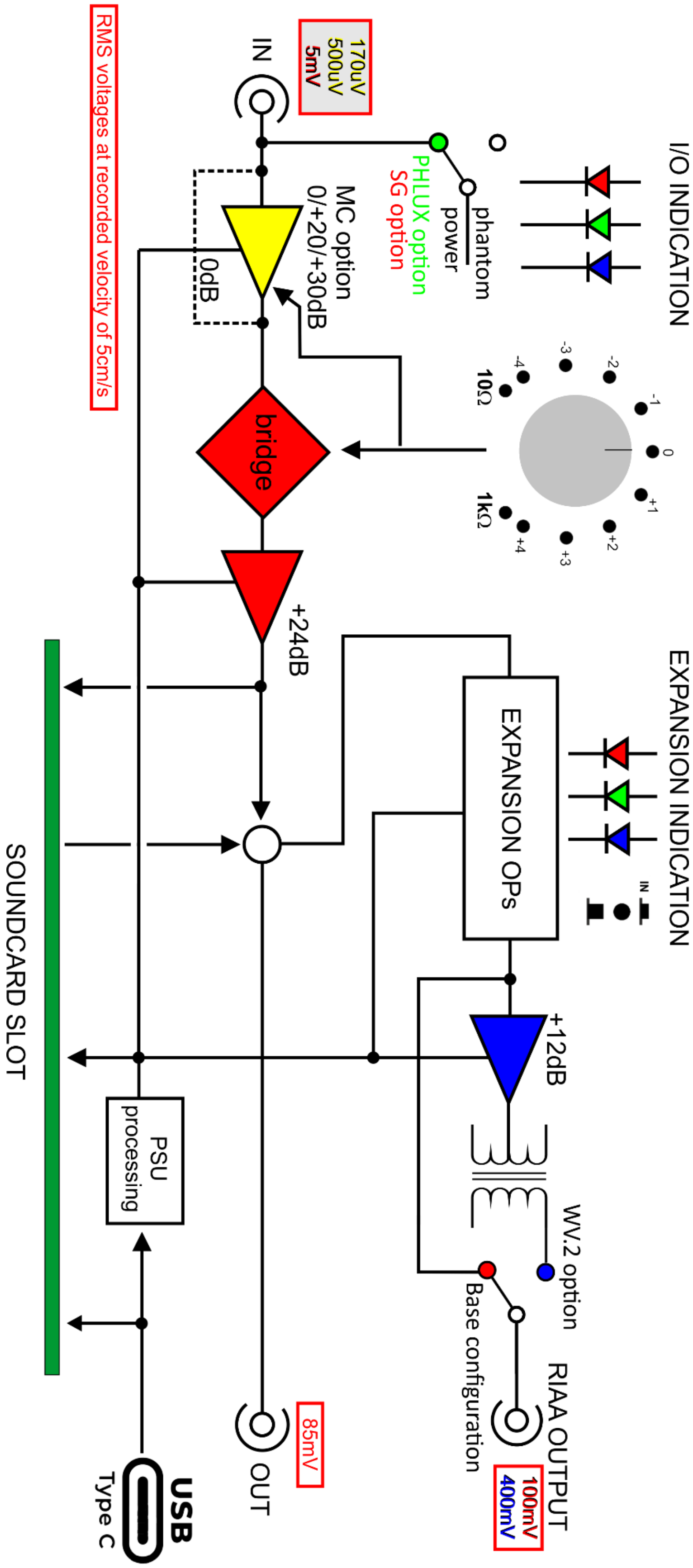
Signed:

Richard Brice, Technical Director Date: January 2025

Appendix 5 – End-of-life: the dustbin symbol

Protect our environment: do not dispose of electrical equipment in domestic waste. Please return any electrical equipment that you will no longer use to the collection points provided for their disposal. This helps the potential effects of incorrect disposal on the environment and human health. This will contribute to the recycling and reutilisation of electrical and electronic equipment. Information where the equipment can be disposed of may be obtained from your local authority.

Appendix 6 – Facilities diagram



PHÆDRUS
phaedrus-audio.com