User Manual for

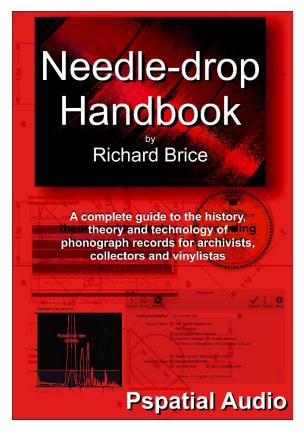
PHLUX-AI ACTIVE INTERFACE PHONO PREAMPLIFIER



Version 3.5 – PHLUX-AI







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 fourth 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 <u>https://pspatialaudio.com/NDH_book2.htm</u> 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, moving-coil, active, and straingauge cartridges
- 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
 - Optional *Sigē* PSU supercapacitor upgrade

Input/output options

Moving-coil input options

- Support for modern as well as historical MC cartridges (e.g. Neumann DST)
 - $_{\odot}$ $\,$ Two gain versions available (500 μ V/ 150 μ V sensitivity)
- 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

- Internal soundcard for 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
 - \circ $\,$ 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.

Sigē PSU option

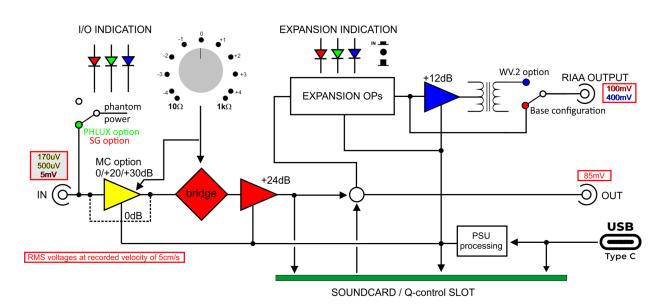
- Power supply upgrade
 - Supercapacitor smoothing

Preamplifier option

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

Facilities diagram

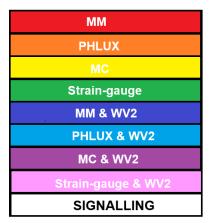
The functionality of equipment like the **PHLUX-AI** preamplifier is often most rapidly grasped with a facilities diagram, like the one given below.



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



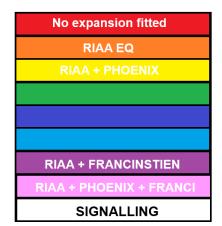
Multicolour LED configuration indicators



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).

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.

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



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 in any other way energised when the case is opened and/or the circuit boards are accessible.

General Safety Instructions

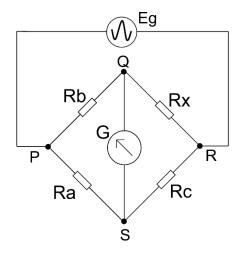
- Do not operate this equipment near any source of water or in excessively moist environments.
- Keep this equipment away from babies, children and pets.
- Do not let objects fall, or liquids be spilled, onto the enclosure.
- Situate this equipment away from heat sources or other equipment that produce 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



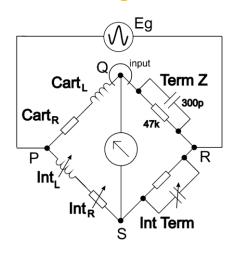
In about 1843, Sir Charles Wheatstone designed a circuit called a *bridge* which gave an accurate method for measuring *resistance*. The circuit is given left. In this circuit, resistors *Ra*, *Rb*, *Rc*, are all known values. *Rx* is the unknown resistance.

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,

 $Rx = (Rb / Ra) \times Rc$



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 (47k Ω) 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 $47k\Omega$) 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.



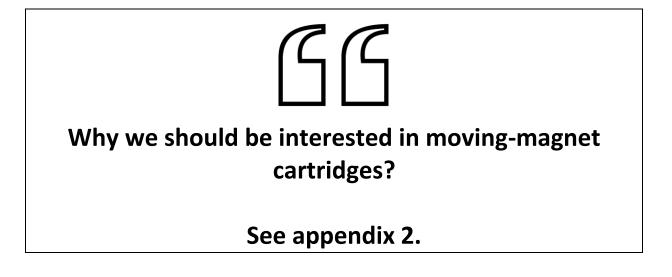
Wheatstone bridge, provided the impedances $Cart_z$ and Int_{Term} are balanced with respect to the external impedances, the galvanometer will indicate zero current.

Now, this is of little use practically because point P doesn't exist in reality. The generator voltage Eg is developed across $Cart_z$ 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 **Cart**_Z of a moving-coil cartridge are so low that matching the internal impedances is largely unnecessary.

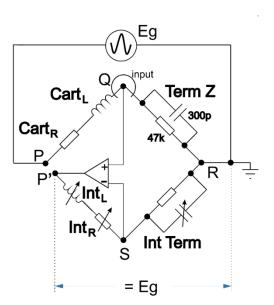
The necessary step in realising a practical preamplifier for moving-magnet cartridges which avoids the need for active electronics in (or very near) the cartridge involves an appreciation that, when the bridge is balanced, point S and point Q are at the same potential. (They must be for there to be no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





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 bridgebalance preamplifier.

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

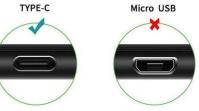
The turntable earth (ground) wire should be connected to the connector labelled GND. You may need to purchase a (4mm) banana plug to mate with this connector to be sure of making a solid and reliable connection. The socket is a 4mm type and has a tin-plated brass insert designed to accept a wide range of 4mm plugs.

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). You may also like to consider the **Sige** upgrade (see below). 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). An alternative – to keep away from mains power completely – is to use a USB Power bank. 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).

³ The internal **SYNOPSIS** soundcard option raises the consumption.



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.**





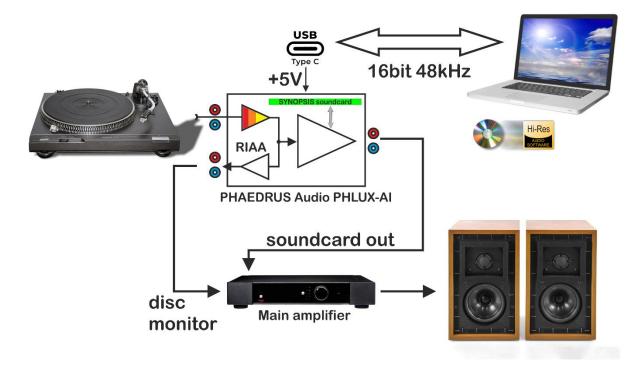
User case #1 – the Home Archivist



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; movingcoil; 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.





See chapter 8.



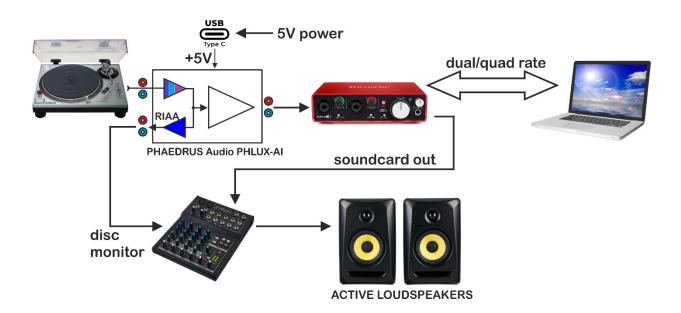
User case #2 - the Professional Archivist



The Professional Archivist

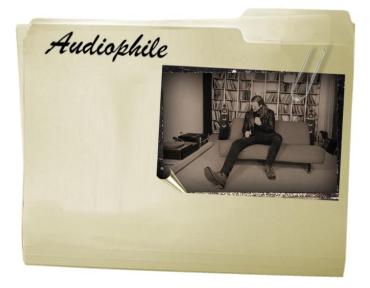
wanted to be able to needledrop record at dual (96kHz) or quad rate (192kHz) sampling, and this required an external soundcard. In order better to match the audio levels from a professional soundcard at the mixer, the Pro Archivist has chosen the **WV2 output option** for the RIAA. The striped input 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.



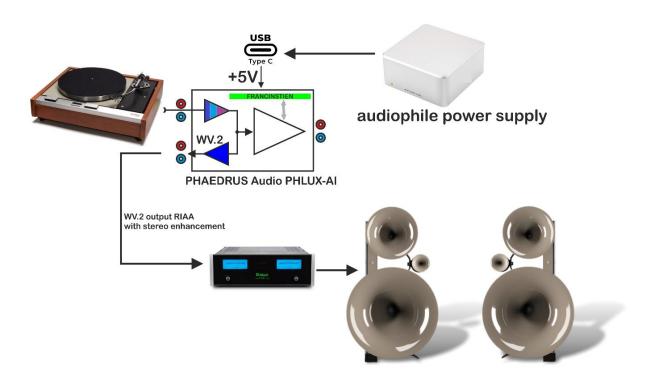


User case #3 - the Audiophile



The Audiophile has no interest in a computer connection. Instead, the PHLUX-AI preamplifier is fed from a quiet, audiophile quality USB power supply. He also chose the *Sigē* PSU-upgrade option (see below). 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.



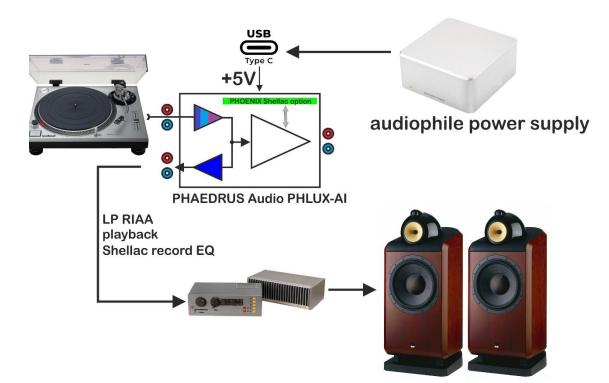


User case #4 – the Record Collector



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.



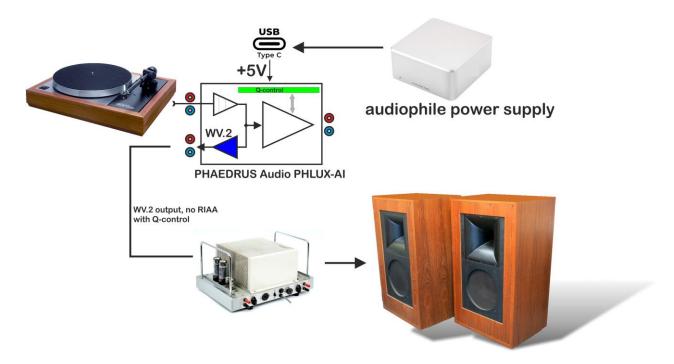


User case #5 – Strain gauge cartridge user



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 is 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 <u>https://pspatialaudio.com/NDH_book2.htm</u> to purchase a copy of the Needle-drop Handbook.



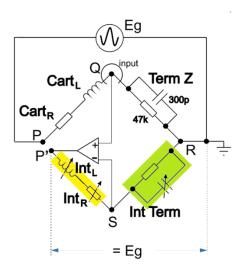
Operating the equipment – the control knob

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 fine bridge-balance. This is set 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

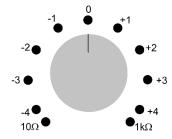
Referring to Chapter 3, if we look again at the bridge configuration of the **PHLUX-AI** preamplifier, the internal bridge automatically compensates for *Term_z* and *Cart_z*. But this automatic process, though leading to the technically correct outcome, may not lead to the best subjective results. **Some optional input is thereby offered to the equipment operator** who may further "tweak" the bridge balance for the most satisfactory subjective results.

It is possible to use test records to make this optional adjustment. However, it is entirely possible to calibrate the bridge-balance by ear. It



should be noted, this is a fine balance adjustment, the tonal differences are subtle, and that 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.

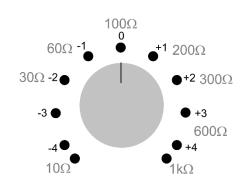
Adjustment



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.



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 $1k\Omega$, as illustrated. The setting is largely subjective, although some manufacturers suggest a load value for their cartridges.

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). This is especially useful if the internal **SYNOPSIS** soundcard is fitted so that digital dynamic range may be optimised.

Balancing the bridge – PHLUX active cartridges

When the **PHLUX** front-end is fitted in the **PHLUX-AI** preamplifier, 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.

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**.⁶ 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 characteristic which serves for most laterally recorded shellac discs.⁷ The **PHOENIX** expansion card also "monos"

	Phonograph EQ A2 (Decca LP)
	Phonograph EQ A3 (TELDEC/AES)
	Phonograph EQ B1 (NAB)
	Phonograph EQ B2 (Columbia LP)
	Phonograph EQ C (CCIR/Coarse Groove)
	Phonograph EQ D (Quad 22 all buttons)
	Phonograph EQ E (500-Flat)
	Phonograph EQ F (Flat/Acoustic)
	Phonograph EQ G (European 78s)
	Phonograph EQ H (American 78s)
	Phonograph EQ J (Columbia 78s)
	Phonograph EQ K (Ceramic)
	Phonograph EQ S (Displacement)
1	Phonograph EQ A1 (RIAA)
	Phonograph EQ P1 (Strain-Gauge/Light-Valve Cartridge)
	Phonograph EQ P2 (DisC Displacement Cartridge)
	Phonograph anti-EQ A1 (RIAA)

⁶ Go to <u>https://pspatialaudio.com/NDH_book2.htm</u> to purchase a copy of the **Needle-drop Handbook**.

⁷ 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.



[(L+R)/2] the signal path which cancels: vertical groove damage; tracing distortion (pinch-effect); and rumble.

When this option is fitted, historic equalisation is selected by pressing the front-panel pushswitch 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.⁸ 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,⁹ 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

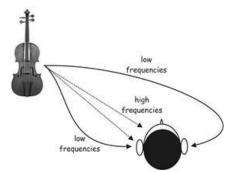
Everyone knows, of course, that sound is waves in air, and that low frequency sounds have long wavelengths and high-frequency sounds have short wavelengths. But most people are surprised by the dimensions of the waves. For example, the lowest note on a bass guitar has a wavelength of 8.5 metres and the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

⁹ 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



⁸ 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

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

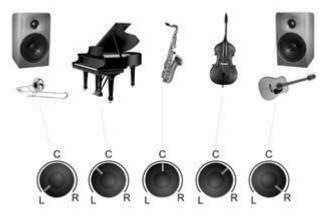
Given this differing behaviour of sounds at low and high frequencies, evolution has provided us with two different processes for determining the direction of sound: one to determine the direction of bass sounds; and another to determine the direction of high-frequency sounds.

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.¹⁰



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.

¹⁰ 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.¹¹ 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 classcompliant and does not require drivers for any OS. The emphasis is on a simple-to-use single-unit solution for needle-drop recording especially when post-processing with Pspatial Audio's **Stereo Lab** app'.¹² Version 4 of **Stereo Lab** is downloadable from Apple's App Store.



¹¹ Brice, R. (1997) Multimedia and Virtual Reality Engineering. Newnes and Brice, R. (1998) Music Engineering. Newnes

¹² https://pspatialaudio.com/



The goal of the combination of **PHLUX-AI** and **Stereo Lab** is that needle-drop recordings are best captured unequalised and are deëmphasised in software (see **Needle-drop Handbook**.¹³) 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 DAC output. When a **SYNOPSIS** soundcard is fitted, the left-hand front-panel LED continues to indicate the input configuration, but when valid data is passed over the USB interface, the indicator illuminates WHITE.

Level control

The analogue to digital conversion equipment on the **SYNOPSIS** expansion card 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 of the digital system is calibrated to -1dB of absolute maximum record velocity (see chapter eight of the **Needle-drop Handbook**). Provided the correct input option has been chosen for the cartridge employed, little or audio level control is required of the user. See **Operating the equipment – the control knob**.

Performance considerations with Stereo Lab

The ADCs in the **SYNOPSIS** soundcard are 16-bit Delta-Sigma ($\Delta\Sigma$), single-rate types, with a measured dynamic range of 90dB and a frequency response from DC to 20kHz of ±1dB. The converters were chosen to be used in conjunction with the **Stereo Lab** app' from Pspatial Audio. The performance figures reported here are for the **PHLUX-AI** equipped with the **MC20** moving-coil option, connected directly to a MacBook Pro laptop via the computer's USB interface, this being a reasonably representative (yet demanding) application.

Stereo Lab Gain Ranging & Hi-Res

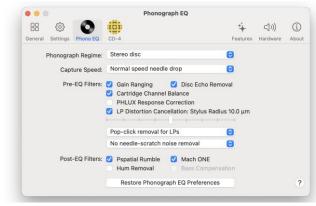
Hi-Res is a precision upscaling algorithm in **Stereo Lab** which doubles the sample-rate of single-rate files (44.1/48kHz) and increases the bit depth to 24-bit with a precision dithering algorithm. **Hi-Res** may be selected as a specific process or it may be enabled as a default processing for all files within the Settings menu. We suggest the latter method is chosen when needle-drop capture is via the internal **SYNOPSIS** soundcard.



¹³ Go to <u>https://pspatialaudio.com/NDH_book2.htm</u> to purchase a copy of the **Needle-drop Handbook**.



Gain-Ranging ensures that residual front-end and ADC noise is over 40dB below surface noise on a brand-new vinyl record. This feature is selected in the Phono EQ dialogue as illustrated.



A recording of system noise (INPUTS terminated) captured with the **SYNOPSIS** card and analysed using *Adobe Audition's* RMS calculator, reveals that the RMS noise level of the file data is -80dB below maximum modulation (-80dBFS). Once up-sampled with the Hi-Res algorithm and RIAA deëmphasised in the **Stereo Lab** app', the resulting 24-bit, 96kHz file has a noise level of -101dBFS. Reduced to an effective bandwidth of 20kHz and A-weighted, the noise level falls to -111dBFS, an exceptional figure: 45dB below the vinyl noise-floor.

Single rate capture considerations

The ADCs in **SYNOPSIS** may be set to run at 32, 44.1, 48 kHz sampling rates. 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**), you will need an external soundcard capable of capturing files at 96kHz or above.

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 <u>new rules from 2025</u>, 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 USB protocol controller wakes up almost immediately.



Chapter 9 PREVIEW - Preamplifier option*

* This option is not available at launch

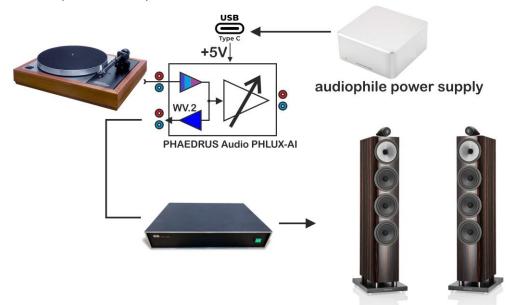
The preamplifier option for the **PHLUX-AI** adapts the control knob to operate as a highprecision volume control (of advanced design) and includes a bundle of options to provide a complete control preamplifier. The options included are:

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
EXP-Sigē-PSU – PSU upgrade

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.



Chapter 10 Sigē PSU upgrade option

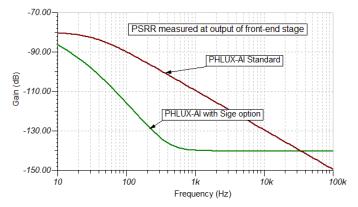
Hence hearing is of sound and silence (sigē), for true hearing must grasp both noise and lack of noise — Aristotle, De Anima III.8, 422a23–26 c. 350 BCE 14

Sigē (pronounced *see-gay*) was the personification of silence in the traditions of Ancient Greece. The *Sigē* option for the **PHLUX-AI** is a supercapacitor-based upgrade to the **PHLUX-AI** front-end, low-noise power supply. *Sigē* employs an electric double layer (EDLC) supercapacitor which removes any noise or interference on the incoming power supply, thereby further lowering the noise floor of the **PHLUX-AI** preamplifier. The optional *Sigē* PSU upgrade occupies same slot as the **SYNOPSIS** soundcard, so these two options are incompatible.

Supercapacitor

An EDLC is a type of supercapacitor that stores energy electrostatically at the interface between two electrodes and an electrolyte. The supercapacitor employs porous carbon electrodes with enormous effective surface area. At each electrode/electrolyte interface, an electric double layer forms, similar to a capacitor but with nanometre-scale separation (\approx 0.3 to 1 nm). This, and the huge effective surface area of the porous carbon electrodes (thousands of square metres), results in a capacitance measured in farads (F) instead of microfarads (μ F).

The exceptional power supply rejection ratio (PSRR) of the *Sigē* circuitry is illustrated, where it is compared with the (already good) PSRR of the standard **PHLUX-AI**.



Data blocking

The *Sigē* upgrade also acts as a USB data blocker. This is especially useful if the best performance is required when feeding the **PHLUX-AI** from (for example) a computer host. The *Sigē* implements carefully designed terminations to the USB D+ and D– signalling lines.

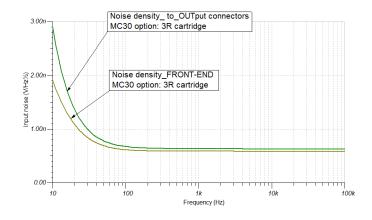
¹⁴ Aristotle's conceptually recognised that that perception is not merely about presence, but also about limits and absences. It would take twenty-two centuries for Gustav Fechner, at the University of Leipzig in the mid nineteenth century to quantify these ideas into measurable psychological laws, transforming perceptual thresholds into the cornerstone of experimental psychology.



This damping is sufficient to stop EMC interference but is benign from the host's perspective: it does not violate signalling thresholds and will not cause the host to attempt enumeration.

Operation

When the unit is first energised, left-hand front panel multicolour indicator will light white, indicating the presence of the *Sigē* option. After about 10 seconds, the white LED will extinguish, and the LED colour will indicate the input option installed. When this change occurs, it indicates that the *Sigē* circuits are active. The noise density of the **PHLUX-AI** (with the **MC30** input option and the *Sigē* is illustrated in the graph below. Note, over the audio band, noise density is 625pV/VHz with a very low flicker noise corner frequency.



Sigē and low-output moving-coil cartridges

The **PHLUX-AI** was originally conceived as a special type of preamplifier for moving magnet cartridges, but it's important to remember that it represents an exceptional partner for moving-coil and other types of cartridges. To illustrate this, we introduce another user case.

User case #6 – Trace Signal



Trace is the proud owner of an Audio Technica's state-of-the-art ART-1000 moving coil cartridge. This component is remarkable in several ways. It was invented in the teens of this century by Mitsuo Miyata of Audio Technica and the design principles patented.¹⁵

Inspired by the DST cartridges of Neumann, the coils and magnet system of the ART-1000 cartridge are placed directly above the stylus instead of inside the cartridge's

¹⁵ Moving-coil type Stereo Pickup. US Patent No.: US 2016/0192080 A1 Myata. Jun. 30, 2016.



body, behind the cantilever's fulcrum – the advantage being that the twisting and bending distortions of the cantilever are eliminated from the movements of the pickup coils themselves.

However, this location of the coils at the tip of the cantilever means their effective mass (as seen at the stylus point) is at a maximum – there's no leverage effect at all. In order that the physical tracking is not compromised, the coils themselves must be very tiny and light. To this end, the two coils of the ART-100 are <1mm in diameter, and each wound of just eight turns of 20µm diameter (52AWG) wire (right).



These conditions impose real constraints on the preamplifier. Despite a powerful magnet, the electrical output from the tiny coils is a meagre 0.2mV at 5cm/s velocity. Only the best preamplifiers can deal with a signal of this magnitude without adding unacceptable amounts of noise. Trace chose the **PHLUX-AI** with the **MC30** moving-coil input, and the **Sigē** PSU options fitted. The unweighted noise output at the OUTput connectors is very low: -68dB below cart output at standard level. This translates to an unweighted equivalent input noise (EIN) figure of -140dBu. This is only 7dB above the thermal noise generated in the tiny (3 Ω) coils, and right on the edge of what is possible from a practical amplifier.

Chapter 11

PREVIEW - Displacement cartridge support

During our work with Pspatial Audio on quadraphonic CD-4 decoding (see chapter twelve and afterword eight of the **Needle-drop Handbook**), we became interested in displacementsensitive phono cartridges based upon strain-gauge transducers which were developed in the 1970s to play wide bandwidth CD-4 records. Intrigued by the performance of these cartridges, we developed a modern preamplifier to support them.¹⁶ This support is continued in the latest **PHLUX-AI** preamplifier. Our own subsequent engineering work has led to the development of an elegant displacement-sensitive phono cartridge, which operates on an entirely different principle to that of the strain-gauge type: one which combines excellent electrical performance with a greatly simplified mechanism. This type of cartridge is also supported by the **PHLUX-AI** preamplifier.

¹⁶ https://www.phaedrus-audio.com/Groove_sleuth.htm#strain

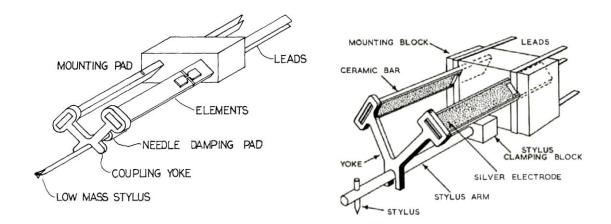


Electrical advantages

Like the old ceramic pickups of the past, strain-gauge cartridges are not electrodynamic generators and thus do not require *movement* of an armature to derive a signal. The transduced quantity is thereby not stylus *velocity*, as it is in a moving-coil or moving-magnet cartridge. Instead, the output voltage is a direct measure of the amplitude or *displacement* of the stylus movement within the groove. Because of this direct translation of stylus position (or stylus *displacement*) to electrical signal, only a relatively modest degree of equalisation is required with a concomitant simplification of the replay electronics. Essentially, *we can do away with the RIAA equaliser*.

Mechanical drawbacks

In the strain-gauge phono cartridge, the movement of the stylus directly controls the resistance of a conducting channel by applying pressure to a semiconductor strain sensor – a heavily doped silicon fibre. In developing such a transducer, there is the significant engineering hurdle to overcome in that the mechanical impedance of the transducing element is several hundred times greater than the permissible stylus-tip impedance.



As with ceramic cartridges, the solution is a lever system as illustrated above (left – *strain-gauge*: right – *ceramic*).^{17,18} But it's a major mechanical engineering challenge to design a phono cartridge in which the stylus ultimately pushes and pulls upon a mass reactance.¹⁹

Phædrus Audio's displacement sensitive transducer

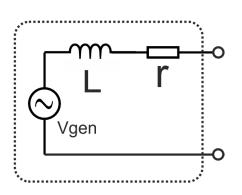
We realised that a complicated mechanical solution was not required. Instead, an electrical engineering solution was available involving the use the mechanics of a standard *electrodynamic* (moving-magnet or moving coil) cartridge: our work being limited to modifications to the electrical generator and recovery preamplifier.

¹⁹ Application of the Silicon Semiconductor Phonograph Cartridge, op. cit.



¹⁷ From: Application of the Silicon Semiconductor Phonograph Cartridge. Wood, J. and Grover, G. Presented at the 17th Annual Meeting of the Audio Engineering Society, Oct 1965.

¹⁸ Inside The Ceramic Cartridge. *Radiotronics* Vol. 30, No. 8. August 1965.



The simplified equivalent circuit of the electrodynamic pickup is given, in which a voltage generator feeds a complex internal impedance comprising a resistive component (*r*) and an inductive component (*L*). Of course, these components are not discrete and, in a stereo pickup, two such circuits exist.

Normal operation of the electrodynamic pickup involves the open-circuit voltage of the pickup being amplified by a low-noise amplifier.

An electrodynamic phono cartridge may be made displacement-sensitive if the short-circuit output-current is sensed, rather than the open-circuit output-voltage. Faraday's law states that the induced EMF in a coil of thin wire is,

where *E* is the electromotive force, ϕ is the magnetic flux, and *k* is a constant which also incorporates the number of turns in the coil.

In an electrodynamic phono cartridge, the magnetic field is dominated by the proximity of the permanent magnet relative to the coil. So, we can take $\phi = P$, where P is the position of the magnet relative to the coil. If we recast Faraday's law in terms of **P** (and ignore the direction convention for convenience) we can say,

$$E=\frac{dP}{dt}$$

This equation is the heart of a velocity-sensitive pickup: it is according to this relationship that all MM or MC cartridges operate.

To see what happens when this EMF is forced to circulate in the coil of the cartridge, we can substitute this expression into the current-voltage relationship in an inductor which is,

$$i=\frac{1}{L}\int v dt$$

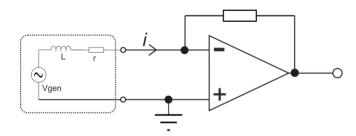
Like this,

$$Is = \frac{1}{L} \int \frac{dP}{dt} dt$$

Which simplifies to demonstrate that the short-circuit current in the circuit (*Is*) is a direct measure of position of the magnet relative to the coil.

$$Is = \frac{P}{L}$$
PHAEDRUS
phaedrus-audio.com

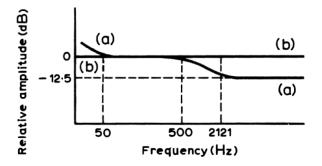
Now, at this point, it would be reasonable to ask: what's the use of a transducer with shortcircuited terminals? How do you get the signal out? The circuit below illustrates how this current may be detected and amplified. Here, the cartridge feeds a *trans-resistance* or *trans-impedance* preamplifier.



With careful electronic design, the result is a line-level voltage generated at the output of the amplifier stage which is an extremely accurate representation of the groove displacement.

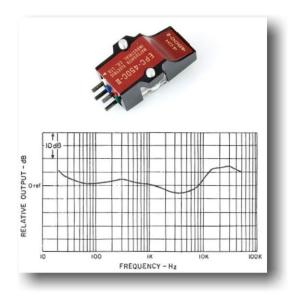
Two complications

In a practical application of this invention there are two complications. Firstly, the information recorded onto the gramophone record does not have an entirely uniform amplitude characteristic vs. frequency. The overall amplitude vs. frequency of the signal engraved on the gramophone disc is given in the diagram below (curve *b* is a pure constant-amplitude characteristic, curve *a* is the characteristic of a modern RIAA standard record).



The equalisation of the signal amplitudes above 500Hz may be secured by means of a wide, damped mechanical resonance at higher frequencies – this is how strain-gauge cartridges work. Or the equalisation may be accomplished by electronic means - this is the solution adopted in the **PHLUX-AI** preamplifier. It is worth noting that the maximum range of equalisation required to return recorded frequencies above 500Hz to the correct level is only *four percent of the degree of equalisation required in the normal RIAA equaliser* circuit.





The second complication arises due to the RIAA recording characteristic which reverts to constant-velocity in the portion of the frequency-range below 50Hz. The pickup must therefore be returned to velocity operation in the lowest octaves of the frequency-range to compensate. This was not realised in the classic strain-gauge cartridges of the 1970s – see response curve of the Panasonic EPC-450C-II cartridge which illustrates this very clearly (left). The result is an exaggerated bass which, despite the frequency response inaccuracy, some listeners appreciate and associate with strain-gauge pickups.

When the **PHLUX-AI** preamplifier is configured for **PHLUX-III-***i* displacement cartridge, the

front-panel control offers a choice. With the front panel control-knob in the 12 o'clock position, correction is not applied, and the bass is boosted – as it was in the Panasonic strain-gauge devices. Rotating the control counterclockwise gradually "tames" this response. The flat position is when the control is in -4 position.

Note that, when the **PHLUX-III-i** cartridge is used with the **PHLUX-AI** preamplifier, the signal on the OUTput connectors (or available over the USB interface if the **SYNOPSIS** soundcard is fitted) is converted to a velocity signal.

Chapter 12

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 <u>new</u> <u>rules from 2025</u>. Standby power consumption is less than ½ Watt. 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): 70dB to RIAA and MAIN output Headroom (rel. 5cm/s velocity): 23dB Crosstalk: Lower than -60dB (1kHz)



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 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 RIAAequalised to full operating line-level output (standard level \rightarrow -6dBu) Gain (to RIAA output): 38dB at 1kHz Noise figure (all configurations, unweighted in 20kHz): 2dB, this being 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)
Crosstalk: Lower than -60dB (1kHz)
Signal to noise (rel. 5cm/s velocity, A-weighted 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)
Crosstalk: Lower than -60dB (1kHz)
Signal to noise (rel. 5cm/s velocity, A-weighted in 20kHz): 68dB 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, A-weighted in 20kHz): 70dB 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, A-weighted in 20kHz): Depends on cartridge model

PHLUX-III-I displacement cartridge option

Function: Support for Phaedrus Audio's displacement cartridge **Input load:** 0Ω nominal (current input) **Sensitivity:** Nominal 50µA RMS at standard recording level



Signal to noise (rel. 5cm/s velocity, ENBW RIAA): 70dB

SYNOPSIS soundcard option

Function: Stereo audio CODEC and USB-compliant full-speed protocol controller Drivers: Class-compliant, irrespective of the computer operating system. 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.



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 <u>www.phaedrus-audio.com</u>.

Appendices

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-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 PHLUX-AI-DISPLACE – Optional PHLUX-III-i displacement cartridge support EXP-SYNOPSIS – Optional internal soundcard (occupies same slot as Sigē PSU upgrade) EXP-Sigē-PSU – Optional PSU upgrade (occupies same slot as SyNOPSIS) 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.²⁰ As Ben Bauer explains in a paper²¹ 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] 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**,²² 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.

²² Go to <u>https://pspatialaudio.com/NDH_book2.htm</u> to purchase a copy of the Needle-drop Handbook.



²⁰ 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.

²¹ The High-Fidelity Phonograph Transducer. Bauer, B. JAES October/November 1977, Volume 25, Number 10/11

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.



Appendix 3 - 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:

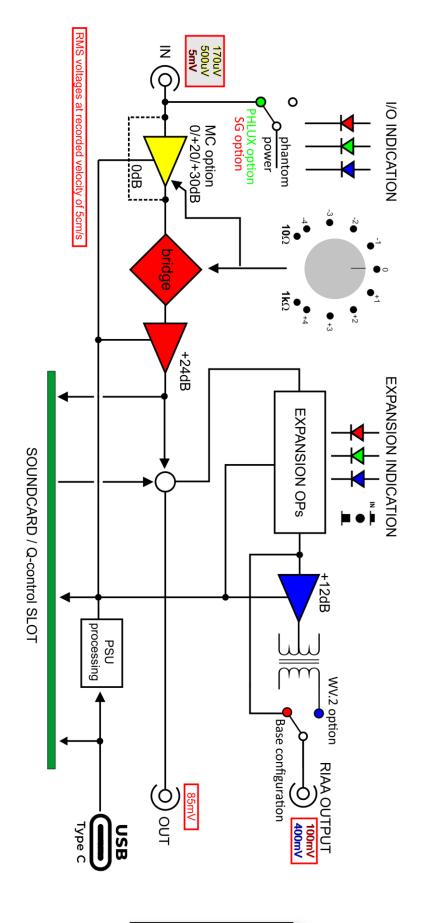
Richard Brice, Technical Director Date: January 2025

Appendix 4 – 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 5 – Facilities diagram

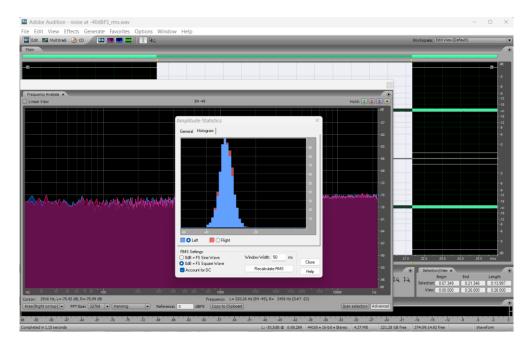


PHÆDRUS phaedrus-audio.com

Appendix 6 - 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.²³

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.

²³ 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* ²⁴ – 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: $\delta f = fs/n$, 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.

²⁴ 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 \approx 7dB.

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

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 represent									
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),²⁵ we can say that the RMS noise voltage (relative to 0dBFS) is 35dB higher than the "eyeballed" noise-floor in the FFT plot in Adobe *Audition*.

²⁵ 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.

