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Orpheus - the ultimate professional FireWire audio interface by Prism Sound

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OperationManual

PrismSound

Orpheus

Operation Manual

by Ian Dennis

This manual is also available as 'on-line help' from the Orpheus Control Panel applet. You can access the on-line help from the 'Help' button in the top right-hand corner of the applet.

Table of Contents

Part 1

General information

1 General information

Manual revision history

Rev	Date	Author	Notes
1.00	8th October 2007	I.G.Dennis	Product Launch
1.02	22nd April 2008	I.G.Dennis	Firmware Revision 1.02
1.03	11th August 2008	I.G.Dennis	Firmware Revision 1.03
1.04	4th September 2008	I.G.Dennis	Change of addresses
1.05	20th October 2008	I.G.Dennis	Correction to fuse type

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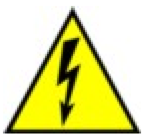
Or contact your local Prism Sound distributor as detailed on the website.

WARNING!



TO PREVENT FIRE OR SHOCK HAZARD DO NOT EXPOSE THIS EQUIPMENT TO RAIN OR MOISTURE. DO NOT REMOVE THE COVER. NO USER-SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.

Statements of conformity

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against interference in a residential area. This device generates and uses radio frequency energy and, if not installed and used in accordance with the instructions, may cause interference to radio or TV reception. If this unit does cause interference to radio or TV reception, please try to correct the interference by one or more of the following measures:
a) Reorient or relocate the receiving antenna.
b) Increase the separation between the equipment and the receiving antenna.
c) Plug the equipment into an outlet on a different circuit from the receiver.
d) If necessary, consult your dealer or an experienced radio or TV technician.
CAUTION: Changes or modifications to this equipment not expressly approved by the manufacturer could void the user's authority to operate this equipment.

THIS DIGITAL APPARATUS MEETS ALL CLASS B LIMITS FOR RADIO NOISE EMISSIONS AS LAID DOWN IN THE RADIO INTERFERENCE REGULATIONS OF THE CANADIAN DEPARTMENT OF COMMUNICATIONS.
CET APPAREIL NUMÉRIQUE RESPECTE TOUTES LES EXIGIENCES APPLICABLES AUX APPAREILS NUMÉRIQUES DE CLASSE B SUR LE BROUILLAGE RADIOELECTRIQUE EDICTE PAR LE MINISTERE DES COMMUNICATIONS DU CANADA.

Prism Media Products Ltd hereby declares that this equipment conforms to the following standards:
EN55103-1, environment category E4
EN55103-2, environment category E4

NOTE: The use of this equipment with non-shielded interface cabling is not recommended by the manufacturer and may result in non-compliance with one or more of the above directives. All coaxial connections should be made using a properly screened 75R cable with the screen connected to the outer of the connector at both ends. All analogue XLR and jack connections should use screened cable with the screen connected to pin 1 of the XLR connector, or the jack outer, at both ends.

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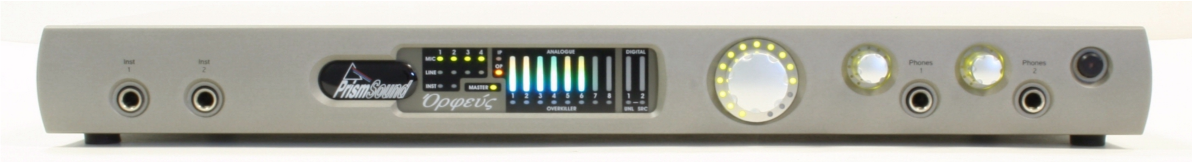
In accordance with our policy of continual development, features and specifications are subject to change without notice.

Part 2

Introduction to Orpheus

2 Introduction to Orpheus



Orpheus is a FireWire multi-channel audio interface for Windows PC and Mac. As well as eight analogue line inputs and outputs, Orpheus provides four high-quality microphone preamplifiers, two high-impedance instrument inputs, a MIDI interface as well as a host of advanced synchronization and monitoring facilities.

In short, Orpheus is intended to provide 'studio in a box' functionality for the digital audio workstation (DAW) user. That's not so unusual - there are many other interfaces on the market which provide similar functionality. However, Orpheus is unique among them in providing Prism Sound's unique pedigree of conversion, analogue, clocking and signal processing to the DAW user for the first time in an integrated package, with the plug-and-play convenience of an ordinary sound card.

For a more detailed summary of the features and capabilities of Orpheus, continue to the Features section.

For directions for getting started quickly, see the Quick start guides.

For detailed hardware and software installation procedures, see the Installation procedures.

For full details of the hardware controls and connections, as well as a block diagram, look at the Hardware section.

To learn about the control software, see the Orpheus Control Panel section.

2.1 Features

Orpheus provides eight line input channels and eight line output channels, which can be operated in balanced or unbalanced mode, and each of which can be used with professional ('+4dBu') or consumer ('-10dBV') signal levels. Four of the input channels have selectable microphone preamplifiers, and two of these also have high-impedance, unbalanced instrument input jacks. 24-bit conversion is used throughout, and sample rates up to 192kHz are supported.

Stereo digital I/O is provided in both S/PDIF and optical (TOSLINK) formats. The S/PDIF input can also accept professional AES3 signals, and the S/PDIF output can be switched to AES3 mode if required. The optical connectors can alternatively carry eight channels of ADAT I/O at 44.1k or 48kHz sample rates, or four channels at 88.2kHz or 96kHz.

Orpheus has a dedicated stereo analogue headphone output, with two stereo headphone jacks each with its own level control. MIDI input and output are also provided.

Under Windows, Orpheus can be accessed by any software applications with WDM (Windows) or ASIO capability. Under Apple OS X, Orpheus appears as a Core Audio device. Multiple Orpheus units can be cascaded in both PC and Mac setups by daisy-chaining the FireWire connections.

To deal with low-latency requirements in live sound and over-dubbing, Orpheus has its own fully-featured mixer for each output channel pair (including digital and headphones). When selected, each mixer allows a low-latency mix of any input channels (as well as the output's associated workstation feeds) to be sent to the required outputs.

A front-panel volume control can be assigned to any desired analogue or digital outputs, primarily for use as a stereo or surround monitor level control.

2.2 System requirements

Orpheus will work with any modern host PC or Mac with a suitable operating system and FireWire 400 port. Macs (both Intel and PowerPC platforms are supported) must be running OS X 10.4 Tiger, 10.5 Leopard or later; PCs must be running Windows XP, Vista (32-bit) or later.

This is not to say that the computing power of the host is unimportant, but it is more a requirement of the application than of Orpheus. If you need to record or playback large numbers of channel, perhaps at high sample rates or with a lot of processing or plug-ins, you will need a host computer with a fast processor and bus, plenty of RAM, and probably a fast hard disk too. On the other hand, playback of moderate channel counts at lower sample rates can be accomplished with even a modest computer.

A good way to gauge this is to be guided by the system requirements of the audio software which you are intending to use. For more information about this, see the Stability and latency section.

2.3 About this manual

The Orpheus Operation Manual is provided in two different formats: as a conventional printed manual, and also as HTML-based 'on-line help' which can be viewed on your computer. The on-line help version can be launched directly from the Orpheus Control Panel applet by clicking the help [?] button in the upper right-hand corner of the panel.

The on-line version of the manual is displayed with a navigation panel to the left of the topic pages. Links within each topic page can be clicked to navigate to pages of interest. The heading bar of each topic page contains buttons for navigating to the previous or next topics, and for moving back to the start of the current chapter. Alternatively, the navigation area on the left contains a "Contents" section which shows a hierarchical map of the entire document from which desired pages can be selected directly. The navigation area can also be used to select topics using a full keyword index, or by searching the document for a particular word or phrase. The selection mode is switched using the controls at the top of the navigation area.

The printed version of the manual is also provided in 'electronic' format, as a 'pdf' file, on the Orpheus software disc. The pdf file can be viewed and printed using the Adobe Acrobat Reader, which can be downloaded free at www.adobe.com.

Updates of both the on-line and pdf forms of the manual are available from the Prism Sound website at www.prismsound.com.

Part 3

Quick start guides

3 Quick start guides

This section contains brief instructions to get up and running quickly. If you need more detailed instructions, see the Installation procedures section.

3.1 Quick start for Mac

The Orpheus software can be installed on Macs with either PowerPC or Intel processors. You need a spare FireWire 400 (IEEE1394A) port. You must have OS X 10.4 Tiger, or 10.5 Leopard, or later.

If you have Tiger, a patch must be installed before installing Orpheus - see the instructions at the end of the full Mac installation section.

- Connect your Orpheus to the mains supply with the power-cord provided. DO NOT connect it to the Mac yet.
- Insert the installation disc into a DVD-ROM drive on your Mac.
- Double-click on the "Install_Orpheus" package.
- Follow the on-screen instructions.
- When software installation is complete, connect the Orpheus unit to the FireWire port of the Mac.

The Orpheus device's ports should now be visible in Audio/MIDI Setup as Core Audio ports.

NOTE: Do not connect the Orpheus unit to the FireWire port of the Mac until software installation is complete.

3.2 Quick start for Windows

To install the Orpheus software, your PC must be running Windows XP or Vista (32-bit), or later. You need a spare FireWire 400 (IEEE1394A) port.

- Connect your Orpheus to the mains supply with the power-cord provided. DO NOT connect it to the PC yet.
- Insert the installation disc into a DVD-ROM drive on your PC.
- If the PC is set to 'Autoplay' , installation will begin automatically, otherwise double-click on the "setup.exe" icon in the root folder of the disc.
- Follow the on-screen instructions.

The Orpheus device's ports should now be visible to Windows and any applications as both ASIO and a WDM audio ports, and a bidirectional MIDI port. Note that only the first two ADAT ports (if enabled by the Orpheus Control Panel applet) are visible in the WDM case.

NOTE: Do not connect the Orpheus unit to the FireWire port of the PC until instructed to do so during the installation process.

NOTE: Orpheus' drivers are have not been submitted for Windows Logo testing, and so warnings of this will be issued during installation. Select 'Continue anyway' when these occur.

Part 4

Installation procedures

4 Installation procedures

This section contains detailed installation instructions for your Orpheus. If you are keen to get going quickly, you could use the Quick start guides section.

4.1 Mac installation

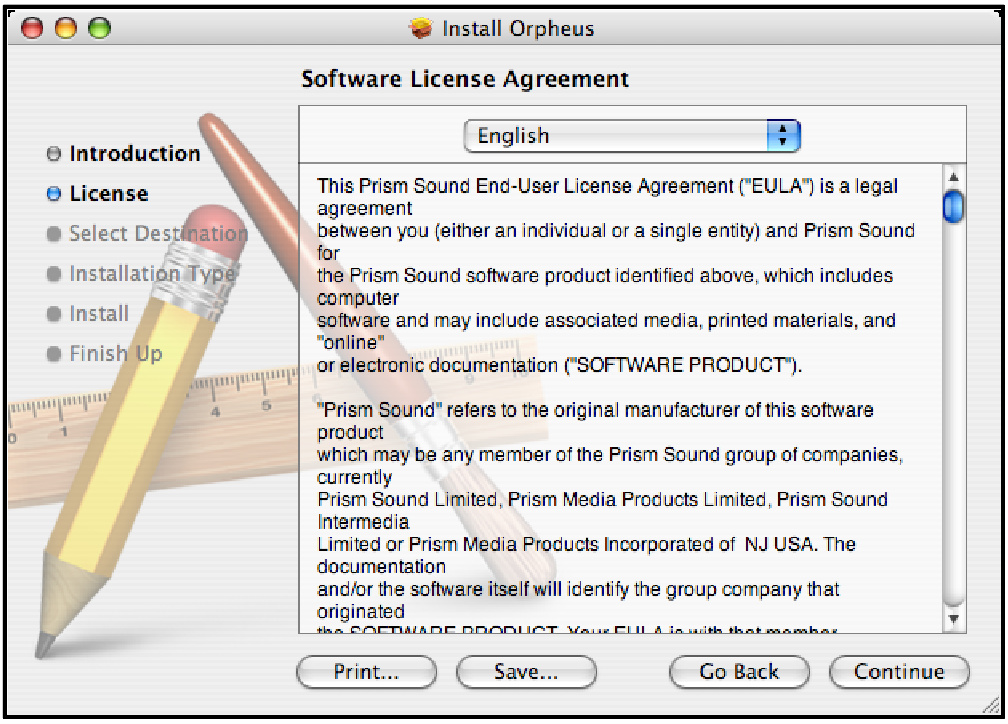
The Orpheus Control Panel applet can be installed on Macs with either PowerPC or Intel processors. You need a spare FireWire 400 (IEEE1394A) port. You must have OS X 10.4 Tiger, or 10.5 Leopard, or later. No driver is required, since OS X can interface to Orpheus directly.

If you have Tiger, a patch must be installed before installing Orpheus - see the instructions at the end of this section.

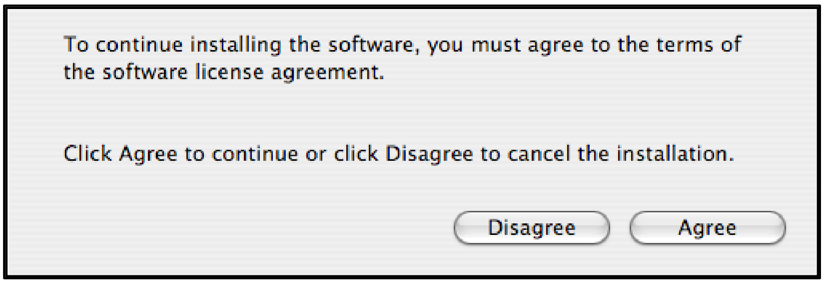
The full installation procedure is as follows:

- Connect your Orpheus to the mains supply with the power-cord provided. DO NOT connect it to the Mac yet.
- Insert the installation disc into a DVD-ROM drive on your Mac.
- If 'Autorun' is not enabled, double-click on the DVD-ROM device
- Double-click on the "Install_Orpheus" package:
- A dialogue box will guide you through the installation process; click 'Continue':

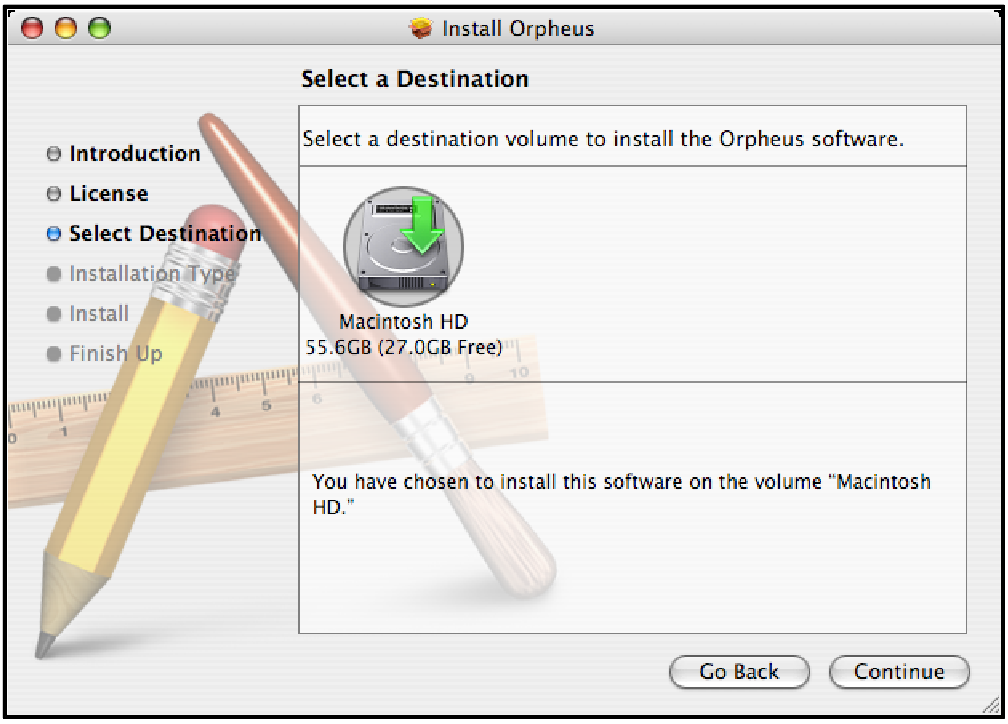
- A copy of the Orpheus EULA will appear, which you should read:



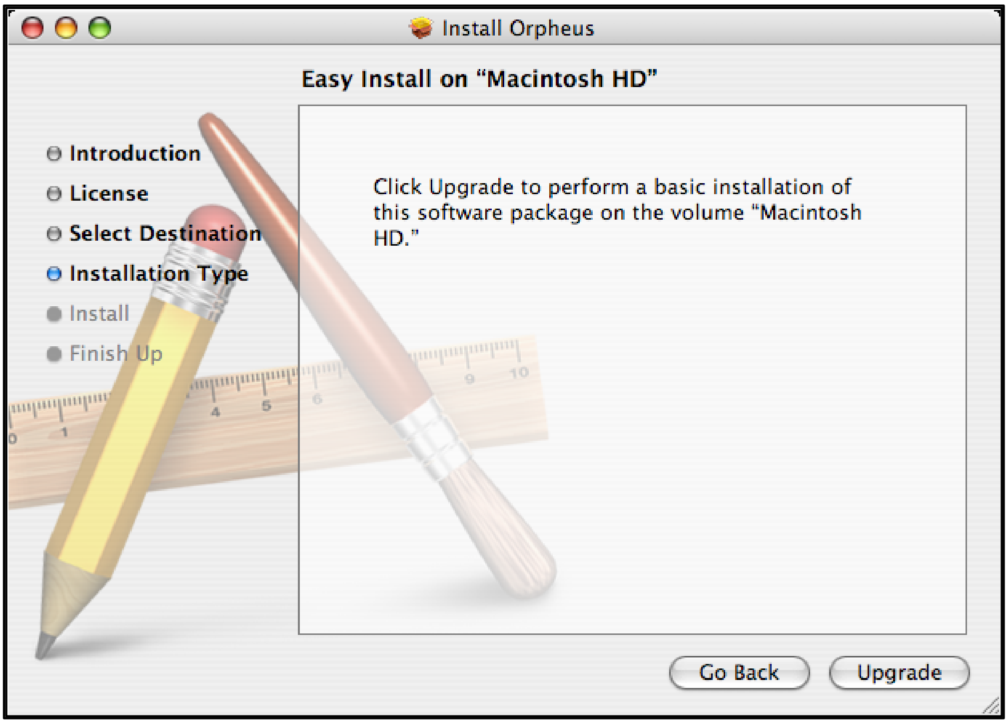
- Agree to the EULA by clicking 'Agree':



- Select a destination volume for the installation, if necessary:

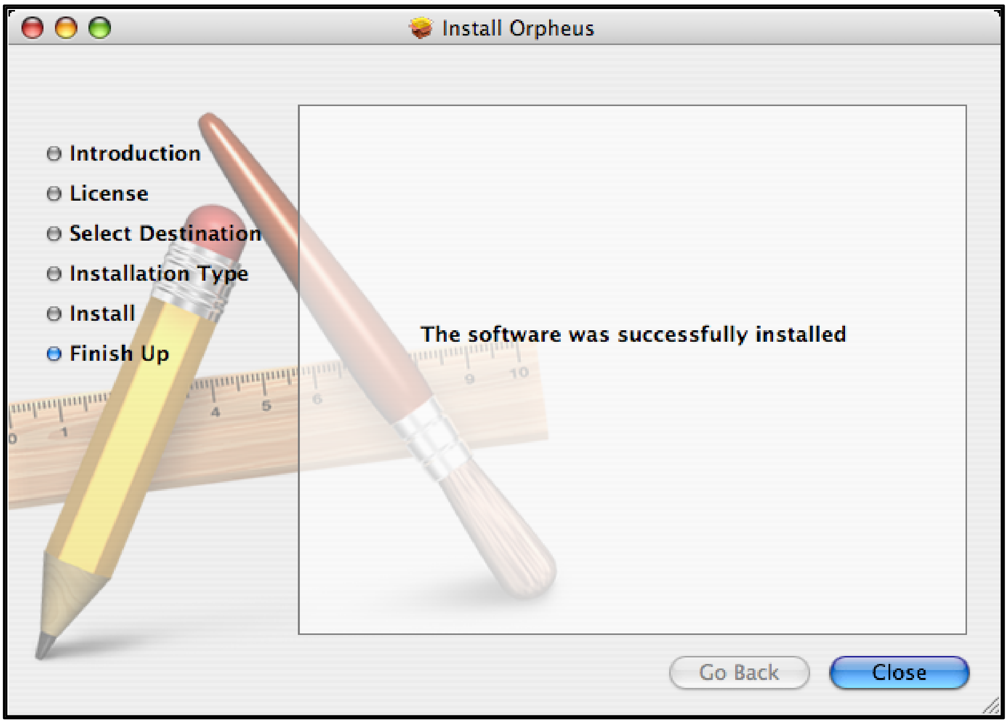


- Click 'Upgrade' to perform a basic installation::



- You may need to enter an administrator's name and password. Click 'OK':

- The installation process will then complete:



- When software installation is complete, close the box and connect the Orpheus unit to the FireWire port of the Mac.

The Orpheus device's ports should now be visible in Audio/MIDI Setup as Core Audio ports.

NOTE: Do not connect the Orpheus unit to the FireWire port of the Mac until software installation is complete.

- Remember to register your Orpheus at http://www.prismsound.com.

OS X 10.4 Tiger patch

To ensure correct operation of Orpheus, users of OS X 10.4 Tiger must install a 'patch' before following the Orpheus installation procedure. The patch is provided by Apple, but is not part of the routine Tiger online update process. A copy of the patch can be found on the Orpheus installation disc.

Before installing the patch, ensure that your Mac has been updated to at least version 10.4.11. This should happen automatically if online updates are enabled.

Locate and double-click the following patch package on the installation disc:

AppleFWAudio-2.2.0f c9-e1.pkg

The patch will overwrite files in he directory:

/System/Library/Frameworks/FWAUserLib.framework

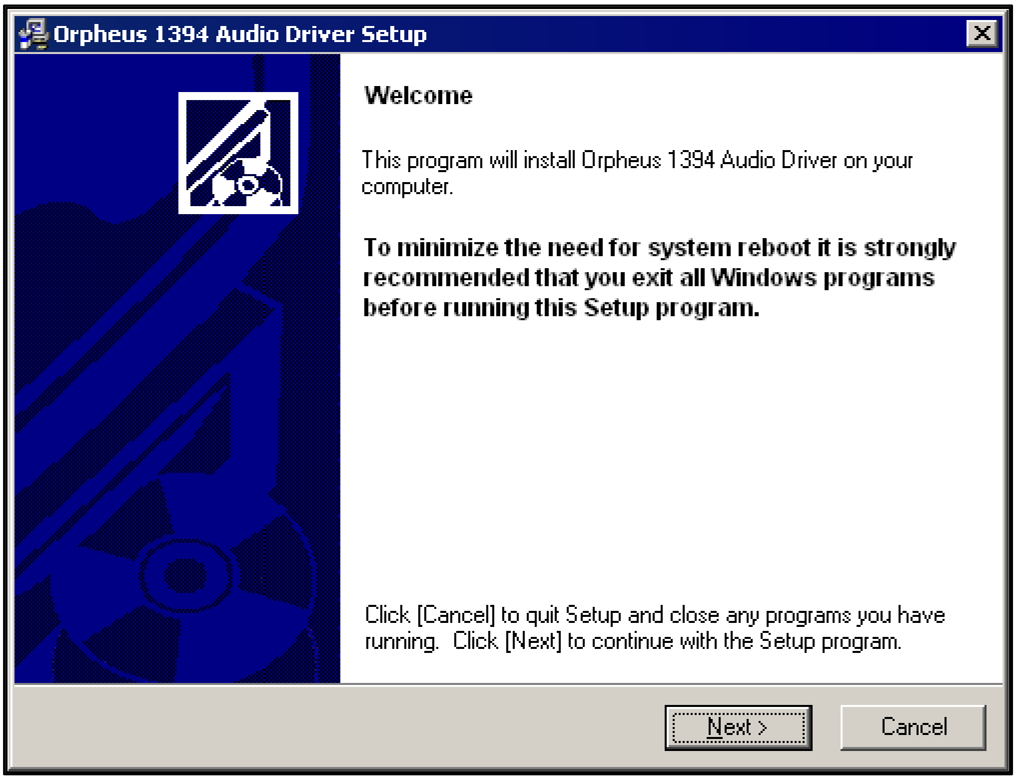
So be sure to back up this directory before applying the patch. In the worst case, the original files can be restored from the OS X installation disc.

You are now ready to proceed with Orpheus installation as described above.

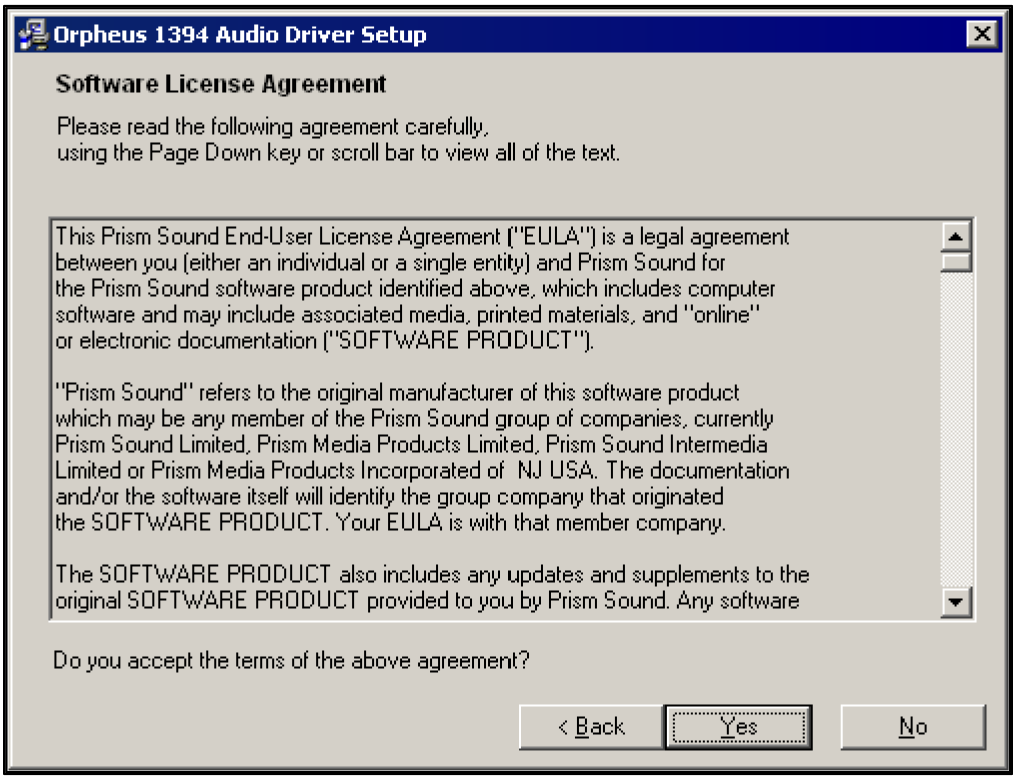
4.2 Windows installation

The following procedure installs the Orpheus ASIO and WDM drivers, and the Orpheus Control Panel applet, on your Windows PC. You must have Windows XP, Vista (32-bit) or later, and a spare FireWire 400 (IEEE1394A) port.

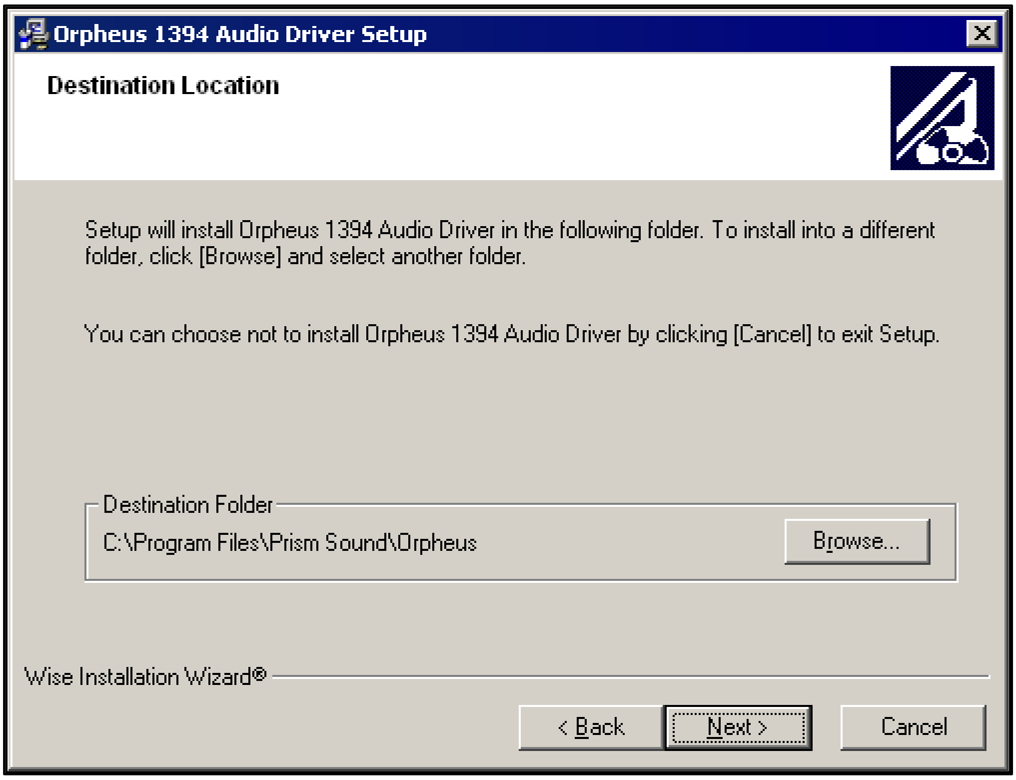
- Connect your Orpheus to the mains supply with the power-cord provided. DO NOT connect it to the PC yet.
- Insert the installation disc into a DVD-ROM drive on your PC.
- If the PC is set to 'Autoplay' , installation will begin automatically, otherwise double-click on the "setup.exe" icon in the root folder of the disc.
- A welcome screen warns you to shut down other applications prior to installation. Do so and click 'Next':



- You will be asked to agree to the Orpheus EULA - if you agree, click 'Yes':

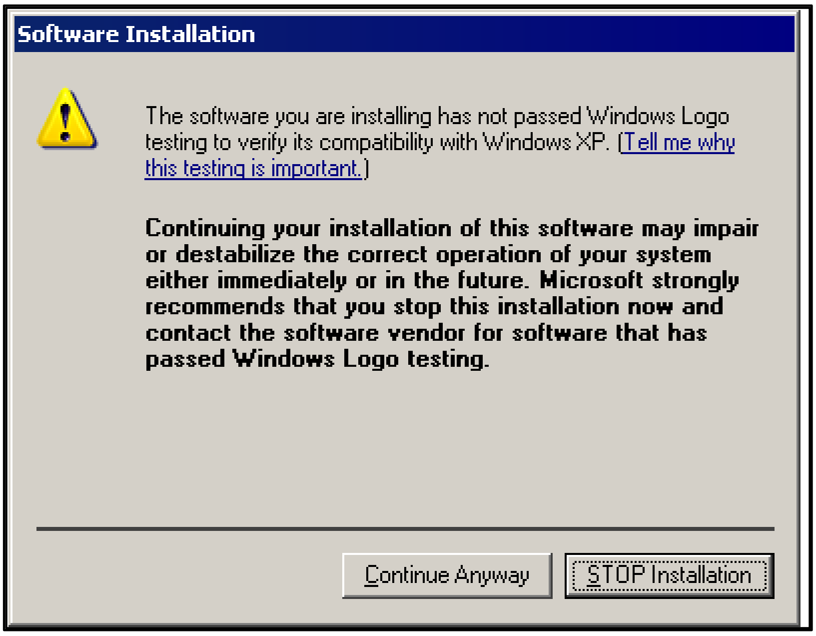


- Choose the installation folder location by clicking 'Browse', if required, or else use the default (recommended); click 'Next':

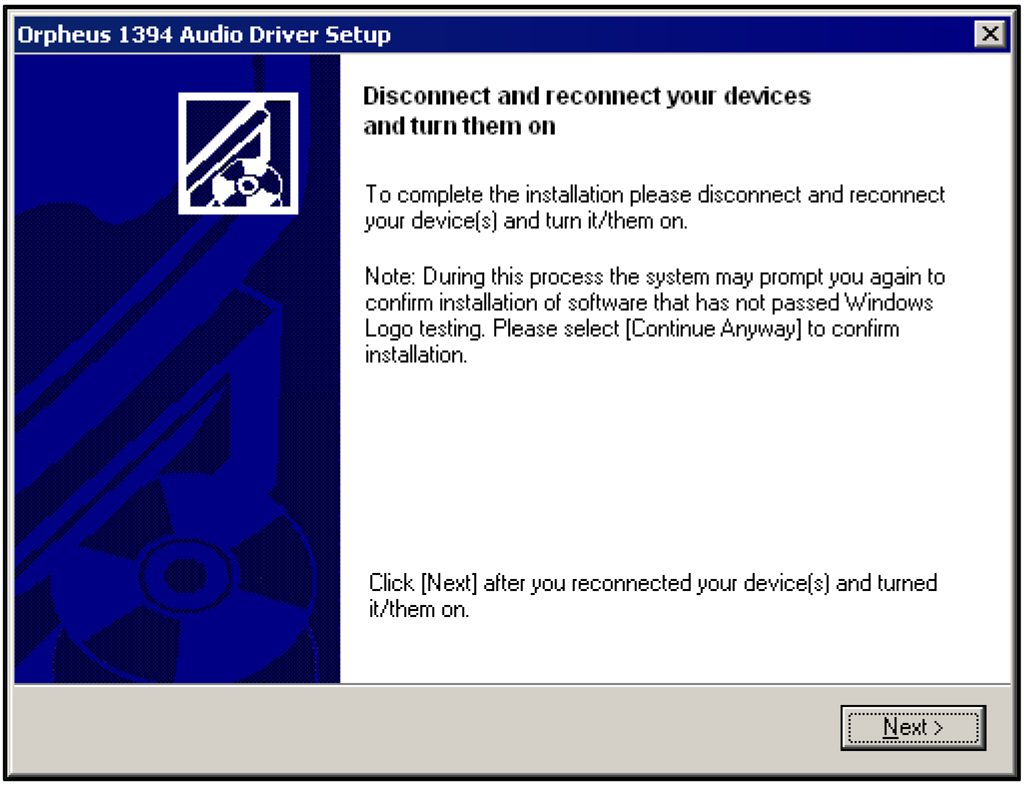


- Installation will then proceed:

- Orpheus' drivers are have not been submitted for Windows Logo testing, and so warnings of this will be issued during installation. Select 'Continue anyway' when these occur:



- During installation, you will be prompted to connect the Orpheus to the FireWire port of the PC. Do so, then click 'Next':



The Orpheus device's ports should now be visible to Windows and any applications as both ASIO and WDM audio ports, and a bidirectional MIDI port. Note that only the first two ADAT ports (if enabled by the Orpheus Control Panel applet) are visible in the WDM case.

- Remember to register your Orpheus at http://www.prismsound.com.

4.3 Cascading multiple units

Multiple Orpheus units can be used on the same PC or Mac by simply 'daisy-chaining' their FireWire cables, i.e. connecting the second unit to the spare FireWire connector on the unit already connected to the host.

The number of units which can be used successfully depends on the bandwidth of the FireWire bus, and also on the processing power of the host computer and its audio software. The latter depends mostly on the host's CPU and RAM fitment; for more information see the Stability and latency section.

Bus-bandwidth limitations are absolute, and apply no matter how powerful the computer is. They depend on the sample rate:

Sample rate	Maximum number of units
44.1kHz, 48kHz	6
88.2kHz, 96kHz	3
176.4kHz, 192kHz	1

It is not recommended to mix Orpheus and other FireWire devices on the same FireWire interface. This can lead to loss of audio channel capacity, or even loss of audio.

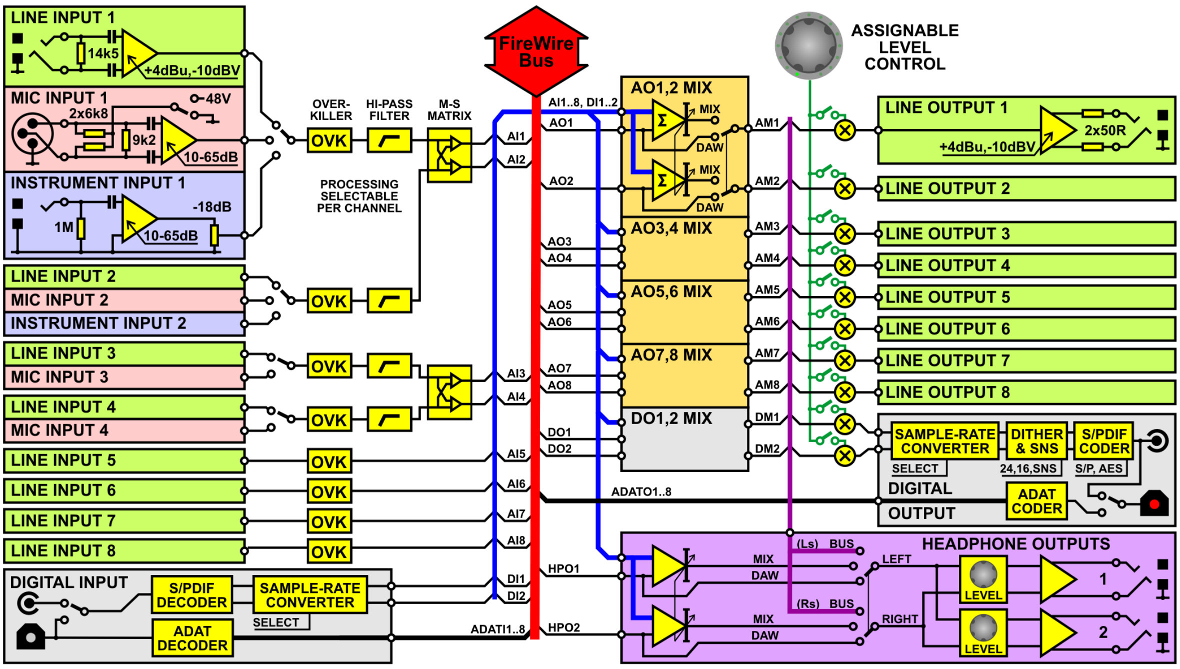
Part 5

Orpheus hardware

5 Orpheus hardware

This section describes in detail the capabilities of the Orpheus hardware.

5.1 Signal path architecture



The figure above is a simplified block diagram of the Orpheus audio signal paths.

Orpheus is basically a sound card, with all inputs made available to the host computer via the FireWire bus, and all outputs likewise driven from the FireWire bus. However, Orpheus' signal paths contain a range of enhanced processing and mixing functions, which are described in the following sections.

5.1.1 Analogue inputs

All eight analogue input channels feature balanced line inputs on TRS jacks, with dual switchable sensitivity to allow connection to professional or consumer line-level sources. The '+4dBu' setting accommodates professional signals with a nominal level of +4dBu and allows a maximum level of +18dBu (0dBFS). The '-10dBV' setting accommodates consumer signals with a nominal level of -10dBV and allows a maximum level of +6dBu (0dBFS).Unbalanced sources are automatically accommodated.

Input channels 1-4 also have balanced XLR microphone inputs, with gains variable from 10dB to 65dB in accurate 1dB steps, and with individually switchable phantom power.

Input channels 1 and 2 also have high-impedance front-panel unbalanced instrument jacks, also with fine and accurate gain control.

Selection of input modes is automatic: line input mode is automatically selected on channels 1-4 whenever a TRS jack is inserted into the rear-panel combo connector, otherwise mic input mode is selected. On channels 1 and 2, mic input mode is automatically over-ridden whenever a mono jack is inserted in the front-panel instrument jack.

The input mode and phantom power state of channels 1-4 is indicated on the front panel of the unit, and also in the Input Setup tab of the Orpheus Control Panel applet. Line, mic and instrument gains are also adjusted in the Input Setup tab, as is selection of the Overkiller, high-pass filter and MS matrix functions described in the following sections.

5.1.1.1 Overkiller

All eight input channels include a switchable Prism Sound 'Overkiller' circuit. The Overkiller is an instantaneous progressive limiter which protects against converter overload by a margin of up to 10dB, gently absorbing transients and allowing recording levels to be raised without risk. The Overkillers in Orpheus operate identically to those in other Prism Sound converters; they can be used with any input mode (line, mic or instrument), and their operating thresholds are automatically adjusted for any gain setting.

The Overkillers are switched on and off in the Input Setup tab of the Orpheus Control Panel applet. A per-channel indication of Overkiller activity is provided both on the front panel of the unit (below each meter, if the meters are in 'Input' mode), and also in the Input Setup tab. Note that these indicators are dynamic, and show when the Overkiller is actually limiting.

5.1.1.2 High-pass filter

Analogue input channels 1-4 have switchable high-pass 'impact filters', which roll off below 80Hz. These are most useful in mic input mode, in removing unwanted low-frequency content. The filters are also available in instrument and line modes, which can be useful if, for example, external microphone pre-amplifiers without filters are used.

The filters are switched on and off in the Input Setup tab of the Orpheus Control Panel applet.

Note that inputs 1&2 have an RIAA de-emphasis filter selectable as an alternative to the high-pass filter, allowing them to be used with vinyl decks. The RIAA filter is aligned to have a gain of 16dB at 1kHz, so that phono cartridges fall within the sensitivity range of the instrument inputs, as described in the vinyl decks section.

5.1.1.3 MS matrix

Analogue input channels 1-4 have switchable MS matrices. These are intended for use with 'mid-side' stereo microphones, where sum and difference signals are derived from the two input channels creating left and right output channels. The MS matrices are also available in instrument and line modes, allowing external microphone preamplifiers without matrixing to be used.

Note that there is no explicit stereo width control provided in the matrix; however width can be adjusted by balancing the gains of the mid and side inputs - the gain steps of the Orpheus microphone preamplifiers are fine and precise. In line input mode, no fine gain adjustment is available, so if the Orpheus MS matrices are used with external preamplifiers, these must have fine and accurate gain control if width adjustment is required.

The MS matrices are switched on and off in the Input Setup tab of the Orpheus Control Panel applet.

5.1.2 Digital inputs

RCA and TOSLINK connectors accept two-channel digital audio signals in the S/PDIF format at any standard sample rate between 44.1kHz and 192kHz. The RCA input can also automatically accept digital audio in the AES3 (AES/EBU) format using the XLR-RCA adapter supplied.

The Input Setup tab of the Orpheus Control Panel applet contains the control for selecting the RCA or TOSLINK connector for S/PDIF input, and also an indicator to show that the S/PDIF input is UNLocked (i.e. no S/PDIF carrier is recognized). The UNLock indicator is also shown beneath the digital meter on the unit's front panel (providing that the unit's meters are in Input mode).

The TOSLINK connector can also accept 8-channel digital input in ADAT format (at 44.1kHz or 48kHz sample rates) or 4-channel input in ADAT S/MUX format (at 88.2kHz or 96kHz sample rates).

Note that ADAT input and output is only possible when operating Orpheus in one of its specific ADAT-capable modes. This is to reduce overhead on the host PC or Mac when ADAT input and output is not required. For more information see the Unit Settings tab of the Orpheus Control Panel applet.

It is possible to configure a sample-rate converter in the S/PDIF input, as described in the following section.

5.1.2.1 Sample-rate converter

A two-channel sample-rate converter (SRC) can be activated in the S/PDIF input if desired. This provides very high-quality conversion of any incoming digital audio signal to Orpheus' current sample rate. The SRC is selected in the Unit Settings tab of the Orpheus Control Panel applet. Note that the SRC can be configured in the S/PDIF input, or in the S/PDIF output, but not in both simultaneously.

Note that presence of an SRC in the S/PDIF input is shown by an indicator beneath the digital meter on the unit's front panel (providing that the unit's meters are in Input mode).

5.1.2.2 DI synchronization

Note that it is necessary to ensure that the sample clock of any digital audio input is synchronous with Orpheus' sample clock (unless the SRC is active in the digital input path). This can be achieved either by synchronizing Orpheus to the source (by using DI or ADAT sync source), or by synchronizing the source to Orpheus' S/PDIF, ADAT or Wordclock output. Orpheus also has a Wordclock sync input for synchronization to Wordclock-equipped sources or house syncs.

The Input Settings tab of the Orpheus Control Panel applet contains an indicator to show that the S/PDIF input is ASYNChronous (i.e. there is an S/PDIF signal present but it is not synchronous with Orpheus' sample clock.

For more information about synchronization settings, see the Synchronization section and also the section describing the Unit Settings tab of the Orpheus Control Panel applet.

5.1.3 Analogue outputs

Orpheus provides eight analogue output channels on TRS jacks, with dual switchable output level to allow connection to professional or consumer line-level equipment. Connection to unbalanced equipment is automatically accommodated by a level-compensation 'bootstrapping' circuit.

In normal operation, the eight analogue outputs are fed directly with individual signals from the host PC or Mac; however, it is possible to feed the outputs from local digital mixers within the Orpheus hardware if desired - this is described in the Output mixers section below.

It is also possible to assign a level control to any desired outputs, primarily for use as a monitor volume control - this is described in the Assignable level control section below.

Setting of analogue output levels, as well as activation of output mixers and assignment of the level control are all managed in the Outputs Setup tab of the Orpheus Control Panel applet.

5.1.4 Digital outputs

RCA and TOSLINK connectors output two-channel digital audio in the S/PDIF format at any standard sample rate between 44.1kHz and 192kHz. The RCA output can also output digital audio in the AES3 (AES/EBU) format using the RCA-XLR adapter supplied. To do this, select 'AES3' instead of 'S/PDIF' in the DO1/2 strip of the Outputs Setup tab of the Orpheus Control Panel applet. This causes the carrier voltage to be increased to the AES3 level, and the Channel Status to adopt the professional AES3 format instead of the consumer format of S/PDIF.

The TOSLINK connector can alternatively output 8-channel digital audio in ADAT format (at 44.1kHz or 48kHz sample rates) or 4-channel audio in ADAT S/MUX format (at 88.2kHz or 96kHz sample rates).

Note that ADAT input and output is only possible when operating Orpheus in one of its specific ADAT-capable modes. This is to reduce overhead on the host PC or Mac when ADAT input and output is not required. For more information see the Unit Settings tab of the Orpheus Control Panel applet.

It is possible to perform sample-rate conversion and word-length reduction (dithering or noise-shaping to 16-bits) as described in the following sections.

In normal operation, the S/PDIF digital outputs are fed directly with individual signals from the host PC or Mac; however, it is possible to feed the outputs from local digital mixers within the Orpheus hardware if desired - this is described in the Output mixers section.

It is also possible to assign a level control to the S/PDIF outputs, primarily for use as a monitor volume control - this is described in the Assignable level control section below.

Activation of output mixers and assignment of the level control are all managed in the Outputs Setup tab of the Orpheus Control Panel applet.

5.1.4.1 Sample-rate converter

A two-channel sample-rate converter (SRC) can be activated in the S/PDIF output if desired. This provides very high-quality conversion of the output signal from Orpheus' current sample rate to any standard rate between 44.1kHz and 192kHz. The SRC is selected in the Unit Settings tab of the Orpheus Control Panel applet. Note that the SRC can be configured in the S/PDIF input, or in the S/PDIF output, but not in both simultaneously.

When using an SRC in the S/PDIF output, it is necessary to select a synchronization source and sample rate for the converted output. The sync source can be local, DI (the S/PDIF input) or the Wordclock input. These settings are made in the Output Setup tab of the Orpheus Control Panel applet.

Note that presence of an SRC in the S/PDIF output is shown by an indicator beneath the digital meter on the unit's front panel (providing that the unit's meters are in Output mode).

5.1.4.2 Word-length

It is possible to control the word-length of the S/PDIF output using the word-length control, uppermost in the DO1/2 strip in the Output Setup tab of the Orpheus Control Panel applet. The control operates as follows:

Setting	Action
24 bit	All 24 bits sent to the digital output are transmitted from the S/PDIF or AES3 output. Channel Status is set to indicate 24 bit output. This setting can be used to pass Dolby or DTS data to an external decoder since it leaves the audio data from the host unchanged. Note, however, that the use of Orpheus' local DO mixer, or the assignable level control, or the SRC in the digital output will prevent bit-identical data from being transmitted.
16 bit	Audio data sent to the digital output is re-dithered using flat TPDF dither to produce a 16 bit output at the S/PDIF or AES3 output. Channel Status is set to indicate 16 bit output. This setting is not generally preferable to the SNS settings, since the noise level is not psycho-acoustically optimized.
SNS1 SNS2 SNS3 SNS4	Audio data sent to the digital output is noise-shaped using Prism Sound's proprietary SNS (Super Noise Shaping) process to produce a 16 bit output at the S/PDIF or AES3 output. Channel Status is set to indicate 16 bit output. These settings are generally preferable to the 16 bit setting, since the noise level is psycho-acoustically optimized. SNS1 offers the least optimization, but with the flattest residual noise spectrum, with optimization increasing up to SNS4, which offers the lowest subjective noise floor, but with significant colouration

Note that if the audio data has already been word-length-processed for 16 bit output by the DAW software, Orpheus' word-length control should be set to 24 bits to prevent unwanted additional dithering.

For further discussion of dithering and noise shaping, and details of the SNS process, see the Dither and noise-shaping section.

5.1.4.3 DO synchronization

Note that it is necessary to ensure that the sample clock of any digital audio device to which Orpheus' digital outputs are connected is synchronous with Orpheus' own sample clock. This is usually achieved by synchronizing the receiving equipment to Orpheus' S/PDIF, ADAT or Wordclock output, but can be achieved by synchronizing Orpheus to the receiving device's clock or house sync (by setting DI or Wordclock as Orpheus' sync source).

For more information about synchronization, see the main Synchronization section , the Clocking and jitter section, and the section describing the Unit Settings tab of the Orpheus Control Panel applet.

5.1.5 Output mixers

Orpheus' output mixers are high-quality, versatile stereo digital mixers available at all of Orpheus' outputs (excluding ADAT outputs). The signal processing in Orpheus' mixers is as precise and sophisticated as in a professional digital console. All coefficients are filtered at sample-rate to minimise unwanted quantization effects such as zipper noise.

In normal operation, Orpheus' output pairs are set in 'DAW' mode, i.e. they output their respective feeds from the DAW software directly - in this case, Orpheus' output mixers are disabled. By selecting 'MIX' mode for an output pair, the mixer is enabled: its inputs comprise all eight of Orpheus' analogue inputs, the two-channel digital input, plus the respective stereo feed from the DAW software. Each input has a dedicated fader and pan pot, plus mute and solo buttons and high-resolution level meter. Input pairs can be designated as 'stereo', wherein a single ganged fader and balance pot are provided, plus a single mute and solo button. The stereo output also has fader, mute button and high-resolution level meters.

The output mixers are primarily intended to provide low-latency foldback or monitor mixes incorporating Orpheus' audio inputs in conjunction with feeds from the DAW software - since the mix is performed locally, the delay involved in passing live audio up to the host computer and back is removed. However, it is also possible to configure the output mixers for general purpose use, where inputs can be mixed to outputs without involving the Host's audio at all. Having set up such mixes using the Orpheus Control Panel applet, it is possible to use stand-alone mode to retain the mix features with no computer connected.

For more information, see the Mixer tabs section of the Orpheus Control Panel applet, and the Stability and latency section.

5.1.6 Assignable level control

Orpheus' front panel has a large assignable level control. This control can be assigned individually to any of Orpheus' analogue of digital outputs (excluding the ADAT outputs, and the headphone outputs which have their own dedicated level controls). Operation of the front panel control fades all assigned outputs. The setting of the control is indicated by a halo of LEDs around the knob. The assignable level control is primarily intended to implement stereo or surround control-room monitoring systems.

Assignment is via the Output Setup tab of the Orpheus Control Panel applet. In this tab, there is also an indication of the position of the control, which can also be operated on the screen using the mouse.

5.1.7 Headphone outputs

The headphone outputs signal path differs from Orpheus' other output signal paths in a couple of respects.

Firstly, the 'DAW'/'MIX' switch has a third 'BUS' position in the case of the headphones. This means that as well as outputting the DAW headphone feeds directly ('DAW'), or outputting a mix of inputs plus the DAW headphone feeds ('MIX'), the headphone outputs can be switched across any of the other output pairs (except for the ADAT outputs). In the Output Setup tab of the Orpheus Control Panel applet, just above the 'DAW'/'MIX/'BUS' selectors is a row of radio buttons for selecting which output pair is monitored by the headphone outputs in 'BUS' mode. Note that the headphone monitoring point of the outputs is before the application of the assignable gain control (if assigned).

Second, the headphone outputs cannot be assigned to the assignable gain control. This is because they already have dedicated volume controls on the front panel.

5.1.8 Metering system

Orpheus' front panel meters can meter the level of either inputs or outputs as selected in the Unit Settings tab of the Orpheus Control Panel applet. A 'Follow Global' setting is also available which allows the input/output mode of all front panel meters in a multi-unit system to be switched simultaneously. The left-most eight meters show the levels of the eight analogue inputs (or outputs); the right-most two meters show the level of the S/PDIF inputs (or outputs). The bar-graphs change colour progressively from blue, through green to orange as signal level increases. A red 'overload' LED is lit if the signal reaches -0.05dBFS. Each of the eight analogue meters has an indicator beneath which shows when the Overkiller (progressive limiter) is active in the case of an analogue input.

Within the Orpheus Control Panel applet, the Input Setup tab shows the levels of the eight analogue and two S/PDIF inputs (as per the front panel, including Overkiller indicators), and the Output Setup tab shows the levels of the eight analogue and two S/PDIF outputs (as per the front panel) plus the headphones level. The Mixer tabs show levels of all inputs and the stereo output of each mixer. The ADAT tab shows the levels of all eight ADAT send and return channels.

5.2 Synchronization

This section seeks to clarify some potentially confusing issues to do with synchronization.

Sync sources, masters and slaves

A single Orpheus connected to a host computer supports a range of reference sync options:

Sync Source	Master/Slave	Description
Local	Master	System is sync'ed to Orpheus' local clock
Wordclock	Master	System is sync'ed to external wordclock applied to Orpheus
DI	Master	System is sync'ed to external S/PDIF applied to Orpheus
ADAT	Master	System is sync'ed to external ADAT (ADAT input must be enabled)
PC DAW	Slave	Orpheus is slaved to system clock from host PC
CSP	Master	System is sync'ed to Orpheus' 1394 clock (not recommended)

When multiple Orpheus units are connected to a single host, only one unit (or the host) can be master. The remainder are slaves to the master's reference sync which is transmitted on the FireWire bus.

Thus every port of every Orpheus in the system must operate at a common sample rate.

An exception to this is when a sample rate converter (SRC) is configured in Orpheus' digital input or output. In the former case, the SRC simply converts any incoming digital signal of whatever sample rate to the sample rate of the Orpheus system. But if the SRC is in the digital output, it is necessary to specify what the output sample rate must be. Furthermore, it may be necessary to lock the output rate to an arbitrary external reference. Orpheus allows for this; as described in the Output Setup tab section of the Orpheus Control Panel chapter.

'Master' in the table reflects the state of the unit's front panel "Master' indicator. If the device's is providing the sample clock to the FireWire for the host (and any other Orpheus devices), the master indicator is lit. If the device is receiving its sample clock from the bus, the indicator is not lit. Note that if an Orpheus is being clocked from its wordclock or DI input, and thus is providing the clock via the FireWire to other interfaces, it is considered to be a master.

Wordclock output

As well as outputting a clock at the selected sample rate, Orpheus' wordclock output can be configured to produce a '256x clock' (a clock at 256x the selected sample rate, e.g. a 'superclock') or a 'base clock' (44.1kHz if the sample rate is a 44.1kHz multiple, or 48kHz if the sample rate is a 48kHz multiple)

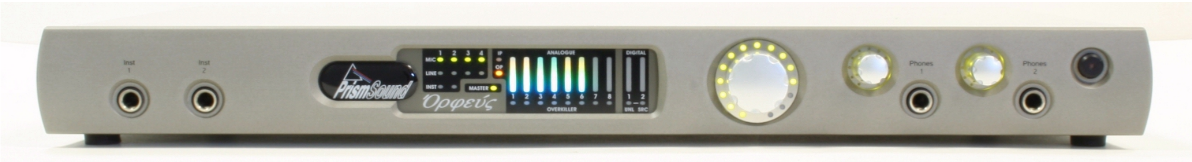
TAKE NOTE

It is not possible to change the sample rate in a Windows system whilst any of Orpheus' ports are in use in WDM mode. In this case it is necessary to disconnect the ports from the application before changing the sample rate.

In Mac OS X, the Audio MIDI Setup panel contains a drop control for selection of the desired Sync Source. However, in versions to date, the names of the selectable Sync Sources may not exactly match those in the Sync Source control within the Orpheus Control Panel applet. For this reason, it is recommended that the Orpheus Control Panel applet be used for setting the desired Sync Source.

If an external sync source such as DI or wordclock is selected, but is either absent or at a different rate from Orpheus' selected sample rate, all audio is muted. If the reference is later applied at the appropriate rate, audio is re-enabled.

5.3 Front panel



Orpheus' front panel contains a limited number of physical controls and indicators. A greater degree of control is available using the Orpheus Control Panel applet software provided. The front panel also contains the instrument input and headphone output jacks.

From left to right:

- Instrument input jacks 1&2: mono unbalanced jacks, high impedance, with finely adjustable gain control. See Analogue inputs.
- Meter panel: see below.
- Assignable level control: Volume knob which can be assigned to any of Orpheus' analogue or S/PDIF outputs, as required. This is primarily intended as a monitor volume control for stereo or surround monitoring.
- Headphone jacks 1&2: each with its own volume control.
- Standby button: puts the unit into a low-power standby state. Note that the FireWire interface is still active in standby mode, so the Orpheus unit can still be recognised by the host, although its inputs and outputs are inactive. The LED in the standby button flashes to identify the unit in multi-unit setups when the 'Identify' button in the Control Panel applet is clicked. Entering standby mode causes Orpheus to retain its current software control settings in flash, for example for use in stand-alone mode.

Meter panel

The meter panel contains metering for the eight analogue input and output channels and the two S/PDIF input and output channels. It also contains input selection indicators for inputs 1-4, Overkiller activity indicators for all the analogue inputs, unlock and SRC indicators for the S/PDIF input, and an SRC indicator for the S/PDIF output.

From left to right:

- Input selection indicators: inputs 1-4 auto-select whichever type of input device is plugged in; these indicators show the state of the selection; mic, instrument or line for inputs 1&2, mic or line for inputs 3&4. The mic selection indicators change from green to orange to indicate that phantom power is switched on. See Analogue inputs.

- Master indicator: this is lit when the device is sample clock master (from the FireWire bus point of view). If a device is configured to be clock master, but is itself slaved to a wordclock or DI reference which is absent or at a different frequency from the selected sample rate, the master indicator flashes. For more information see the Synchronization section.

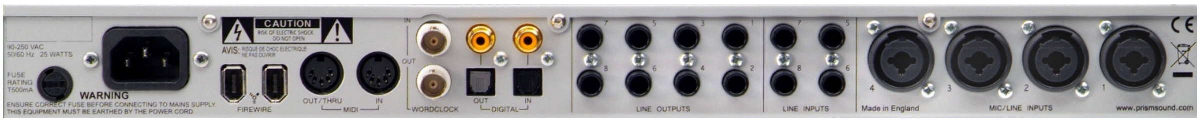
- Meter input/output indicator: shows whether the ten bar-graph meters are assigned to the analogue and S/PDIF inputs, or to the analogue and S/PDIF outputs. This is selected in the Orpheus Control Panel applet. See the Metering system section for more details.

- Overkiller indicators: indicate that the Overkiller progressive limiter is operating in that channel. Note that the indication is dynamic, and shows when the Overkiller is actually limiting, and not simply that it is enabled. Note that the Overkiller indicators are only active when the meters are in input mode.

- DI unlock indicator: indicates that the S/PDIF input is unlocked, i.e. no S/PDIF carrier is recognised. Only active when the meters are in input mode.

- SRC indicator: shows that the SRC (sample-rate converter) is configured in the S/PDIF input (if lit when the meters are in input mode) or is configured in the S/PDIF output (if lit when the meters are in output mode.

5.4 Rear panel



Orpheus' rear panel contains all Orpheus' connections, except for the instrument inputs and headphone outputs, which are on the front panel.

From left to right (viewed from rear):

- 6A IEC inlet (regional power cord supplied): adjacent is the mains fuse holder.

- Two FireWire 400 (IEEE1394A) connectors (4-pin and 6-pin FireWire cables supplied). These have identical function; either can be used to connect to the host computer, the other can be used to daisy-chain additional Orpheus units if required. See Cascading multiple units.

- MIDI in and out/thru DIN sockets.

- Wordclock input and output BNC sockets: the wordclock output can supply base-clock or 256x clock if required. See the Synchronization section.

- S/PDIF input and output RCA sockets: these can also be operated as AES3 interfaces (RCA-XLR and XLR-RCA adapter cables supplied). See Digital inputs and Digital outputs.

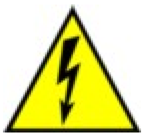
- TOSLINK input and output: can be used for S/PDIF (up to 192kHz sample rate) or ADAT (44.1kHz/48kHz or 88.2kHz/96kHz in SMUX mode). See Digital inputs and Digital outputs.

- Line output TRS jacks 1-8: switchable +4dBu/-10dBV level, can operate in balanced or unbalanced mode. See Analogue outputs.

- Line input TRS jacks 5-8: switchable +4dBu/-10dBV level, can operate in balanced or unbalanced mode. See Analogue inputs.

- Mic/line input combos 1-4: XLR for microphones, with 10dB to 65dB finely-adjustable gain and switchable phantom power; TRS jack for line inputs, with switchable +4dBu/-10dBV level, can operate in balanced or unbalanced mode. See Analogue inputs.

5.4.1 Fuses and ratings



TO PREVENT SHOCK HAZARD, THE ORPHEUS HARDWARE SHOULD ONLY BE OPENED BY QUALIFIED PERSONNEL. REMOVE THE POWER LEAD FROM THE UNIT BEFORE REMOVING THE TOP COVER.

Fuse locations and ratings are as follows:

FUNCTION	LOCATION	TYPE
Mains	Rear panel	500mA(T), 20mm, glass

Note that no fuses or any other user-serviceable parts or options are located inside the Orpheus unit.

5.5 Stand-alone operation

It is possible to operate Orpheus without a FireWire connection to a host computer. This is done by setting up the unit as required using the Orpheus Control Panel applet whilst it is connected to a host computer via its FireWire interface, then placing the unit in standby by pressing the standby button, before disconnecting the unit from the host and power source. When the unit is re-powered, and detects that no FireWire connection is active, it reloads the settings which were previously stored.

Since all of the outputs have optional mixers within the Orpheus hardware whose inputs include all of Orpheus' input connectors, it is possible, for example, to connect the digital input pair to one or more analogue output pairs, and to mix one or more analogue input pairs to the digital output pair. The synchronization source and sample rate can be set in the usual way. Thus stand-alone mode can be used to configure a stand-alone D/A converter and A/D converter, with static mixing if required.

ADAT Direct mode

ADAT Direct mode allows Orpheus to operate as an eight channel analogue-to-ADAT and ADAT-to-analogue converter. ADAT Direct mode is activated by setting the **ADAT** control in the Unit Settings panel to 'Direct'.

In ADAT Direct mode it is possible to control Orpheus using the Control Panel applet in the usual way while Orpheus is connected to a host computer via FireWire. If Orpheus is not connected to a host computer, ADAT Direct mode can still be used as a stand-alone mode, by saving pre-selected settings using the standby button as described above.

Signal routing in ADAT Direct mode

The eight ADAT output (send) channels are fed directly from the eight analogue input channels. Any signals sent from the host computer over FireWire (if connected) destined for the ADAT outputs are ignored by Orpheus.

The eight channels sent from the host computer over FireWire (if connected) destined for the Orpheus analogue outputs are ignored by Orpheus, and replaced by the eight ADAT input (return) channels. This means that the analogue output **DAW/Mix** switch can still be used: in the DAW position, the analogue outputs are fed directly from the ADAT inputs; in the Mix position, the analogue output mixer can be used to mix the ADAT inputs with any of the other Orpheus inputs.

The ADAT inputs and analogue inputs are available to the host computer (if connected) for recording or monitoring if required. The ADAT outputs and analogue outputs appear to be available to the host computer (if connected) but are ignored by Orpheus. Orpheus' S/PDIF inputs and outputs, and headphone outputs, remain available to a connected host computer.

In common with all ADAT modes, ADAT Direct mode can only operate with all eight channels if the sample rate is set to 44.1kHz or 48kHz. At sample rates of 88.2kHz or 96kHz, ADAT Direct mode operates in SMUX mode, supporting only the first four analogue input and output channels. ADAT Direct mode cannot be selected at sample rates of 176.4kHz or 192kHz.

NOTE: Since ADAT Direct mode modifies Orpheus' normal signal routing, it is very important to ensure that ADAT Direct mode is not selected accidentally.

5.6 Rack mounting

Orpheus is supplied configured for table-top operation, with rubber feet attached and no rack-mount ears fitted.

To convert for rack-mounting: First fit the rack-mount ears by removing the front four screws from each side of the unit, using the hex key provided; then replace the same screws to retain the rack ears provided. If necessary, the rubber feet can be removed by withdrawing the plastic centre-hub of each foot a little way prior to pulling the whole foot out. The hub can be initially raised with a small flat-bladed screwdriver. Retain the feet for later use.

If Orpheus units are rack mounted, an empty 1U gap should be left above each Orpheus to ensure effective cooling.

Part 6

Orpheus software

6 Orpheus software

This section describes the software supplied with Orpheus.

6.1 Orpheus Control Panel

The Orpheus Control Panel applet is a program which can be used to adjust Orpheus' settings, and view its metering and status indicators, from the screen of the Mac or PC.

Whilst Orpheus can be controlled in a limited way from within the Mac or Windows operating system, or by some audio application programs, most of its detailed features can only be accessed using the Orpheus Control Panel.

Accessing the Orpheus Control Panel

The Orpheus Control Panel can be run directly like any other Mac or Windows program, or it can be accessed from the operating system itself, from the "Sound and Audio devices" dialogue in the Windows' Control Panel (or from the Device Manager) or from the Audio MIDI Setup in Mac OS X. Some audio applications, such as Nuendo, also allow guided access to the Orpheus Control Panel.

Operating the Orpheus Control Panel

The Orpheus Control Panel user-interface is a fixed-size dialogue box which can be activated and 'put away' like any other Mac or Windows application. When active, it cannot be resized. The upper part of the Control Panel contains Unit and Global settings, and beneath are a stack of 'tabs' allowing context-switching of the bulk of the user-interface area.

The Input Setup tab contains everything to do with setting up analogue and digital inputs, the Output Setup tab similarly for outputs. A row of Mixer tabs provide access to the low-latency built-in mixers, whilst the ADAT tab displays meters for the ADAT inputs and outputs.

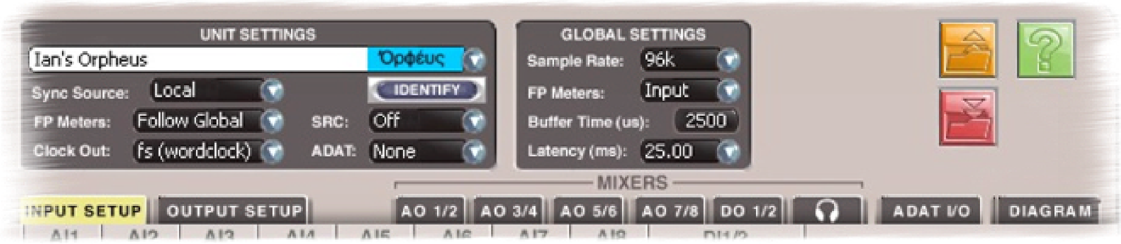
Where more than one Orpheus unit is connected, the Orpheus Control Panel controls all the connected units. The selector at the top of the "Unit Settings" panel is used to assign the Control Panel to whichever unit is to be controlled. If no Orpheus units are connected, the Control Panel remains blank.


orpheus
UNIT SETTINGS
Ian's Orpheus
Sync Source: Local
IDENTIFY
FP Meters: Follow Global
SRC: Off
Clock Out: fs (wordclock)
ADAT: None
GLOBAL SETTINGS
Sample Rate: 96k
FP Meters: Input
Buffer Time (us): 2500
Latency (ms): 25.00
MIXERS
INPUT SETUP
OUTPUT SETUP
AO 1/2
AO 3/4
AO 5/6
AO 7/8
DO 1/2
ADAT I/O
DIAGRAM
AI1 AI2 AI3 AI4 AI5 AI6 AI7 AI8 DI1 DI2 DAW1 DAW2 AO5/6
Mixer
SOLO
MUTE
STEREO
-18.00 -14.50 -21.00 -17.25 -7.25 -4.75 -11.25 -13.00 -19.00 0.00
firewirecontroller
prismsoundrecording

6.1.1 Unit and Global settings

The upper area of the Orpheus Control Panel contains Unit Settings and Global Settings. Unit Settings are applied only to the Orpheus unit which the Control Panel is currently assigned to control, whereas Global Settings apply to all connected Orpheus units as a whole.

The upper area also contains buttons for loading and saving Orpheus configurations, and for accessing the on-line help.



Unit Settings

The topmost control in the Unit Settings panel is the **Unit Selector**. This selects which Orpheus unit is controlled by the rest of the Unit Settings panel, and by the tabbed panels in the lower part of the Control Panel (the Global Settings panel controls all connected units, and is the only part of the Control Panel not affected by the Unit Selector). The Unit Selector is a drop-list which can be dropped by clicking the down-arrow at the right-hand-side, presenting a list of all the connected devices for selection. The name of the selected device is then displayed, along with the device type (e.g. Orpheus, in the high-lit area on the right).

By default, the name of each device is its GUID (Globally Unique Identification number), but this can be changed to a more friendly name (Windows only) by clicking in the left-hand side of the selector and typing in a new name. This doesn't change the unit's GUID, of course - it only associates a friendly name with the GUID whenever the OS or Control Panel applet sees the unit.

Below the Unit Selector is the **Identify** button. This can be latched on or off by clicking it. When it's on (high-lit red) the LED in the standby switch of the currently-selected unit flashes. This helps to identify each unit in a multi-unit system.

The **Sync Source** control allows each unit's reference synchronization source to be selected. For more details, see the Synchronization section.

NOTE: If an external sync source such as DI or wordclock is selected, but is either absent or at a different rate from the selected sample rate, all audio is muted. If the reference is later applied at the appropriate rate, audio is re-enabled.

The **FP Meters** control allows the front panel meters to be switched between the analogue and S/PDIF inputs, and the analogue and S/PDIF outputs. An additional selection, 'Follow Global', allows multiple units' meters to be switched using a single control in the Global Settings. For more information, see the Metering system description. Note that right-clicking on the FP Meters control (control-clicking on one-button Macs) brings up a brightness control for the Orpheus front panel - four different brightness settings can be selected to suit ambient lighting conditions.

The **Clock Out** control can be used to cause the unit's wordclock output to produce base clock or 256x clock instead of wordclock if required. For more details, see the Synchronization section.

A two-channel sample-rate converter (SRC) can be configured in the S/PDIF input path, or in the S/PDIF output path, or can be disabled, using the **SRC** control.

ADAT inputs, or ADAT outputs, or both, can be enabled using the **ADAT** control. By default, ADAT ports are disabled to ease load on the host computer. ADAT Direct mode is a special mode which allows Orpheus to operate as an eight channel analogue-to-ADAT and ADAT-to-analogue converter, as described in the Stand-alone operation section.

NOTE: Changing the ADAT mode causes changes in the number of input and output channels reported to the host computer by Orpheus. It is therefore advisable to close your DAW application before changing the ADAT mode, and to restart it again afterwards, in order to ensure reliable operation.

Global Settings

The Global Settings panel contains a small number of controls which apply to all units in a multi-unit system.

Most important is the **Sample Rate** control. All units in a multi-unit system must operate at a common rate. For further information about sample rates, see the Synchronization section.

NOTE: It is not possible to change the sample rate in a Windows system whilst any of Orpheus' ports are in use in WDM mode. In this case it is necessary to disconnect the ports from the application before changing the sample rate.

The **FP Meters** control allows I/O switching of front panel meters across multiple boxes as described above.

In Windows systems, the audio delay through the input and output buffers can be controlled with the **Buffer Time** and **Latency** controls. The Buffer Time control sets the duration of a buffer, and the Latency control sets the overall path delay by applying a number of such buffers to the audio path. In general, it is better to keep the Latency in general, and the Buffer Time in particular, quite long. This reduces the risk of audio glitches, as described in the Stability and latency section. Orpheus' low-latency on-board foldback mixing facility reduces the need for the Latency controls to be short. In Mac systems, latency control is handled by OS X.

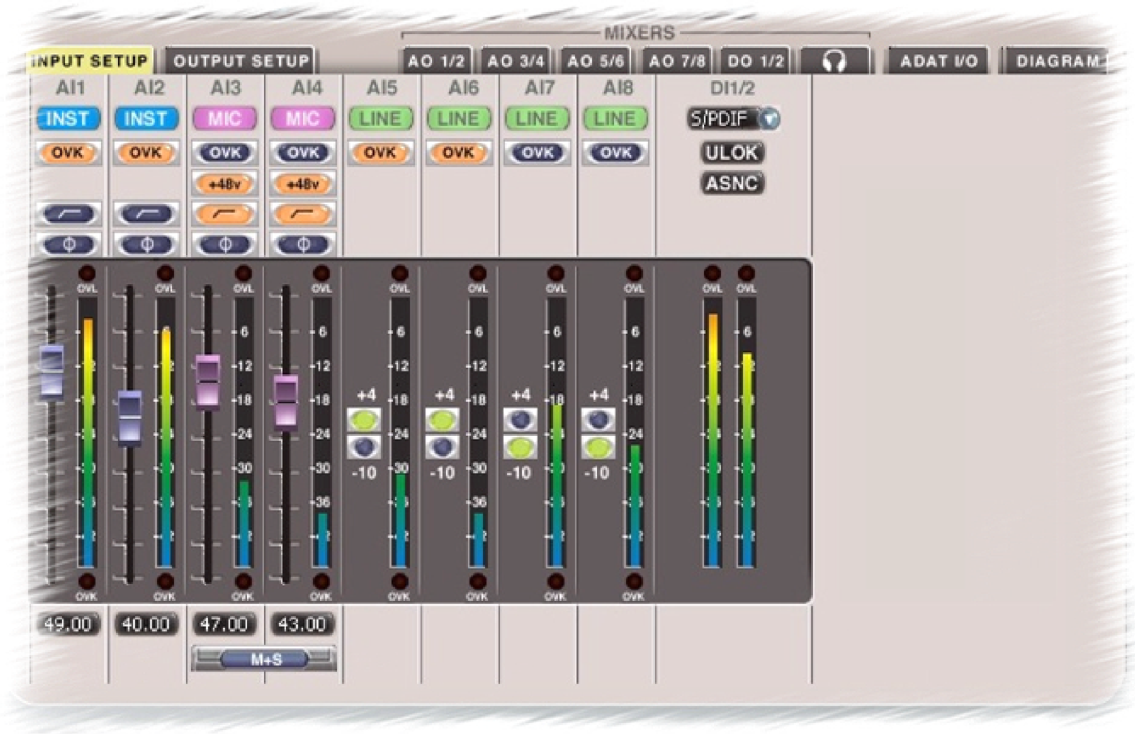
Load, Save and Help buttons

The green **Help** button ('?') opens the online version of this manual in a browser window.

The red **Save** and the orange **Load** buttons save and load Orpheus settings to and from disk. Note that only settings of the currently-selected unit are affected, so it is necessary to save and load settings of each unit individually in multi-unit systems.

6.1.2 Input Setup tab

The Input Setup tab contains the controls and status indicators for all functions of the analogue inputs and S/PDIF inputs as described in the hardware section. Inputs 5-8 handle line-level inputs, whereas inputs 3&4 can operate as mic or line inputs, and inputs 1&2 as instrument, mic or line. Mode selection is handled automatically depending on which connectors are used.



Instrument input mode is indicated by the blue 'INST' legend at the top of the strip, microphone mode by the pink 'MIC' legend and line mode by the green 'LINE' legend. Line input sensitivity is switched between +4dBu and -10dBV nominal by the '+4/-10' radio buttons, whereas mic and instrument input gains are adjusted in 1dB steps by the slider controls, and indicated by the number beneath, which can also be directly entered if required.

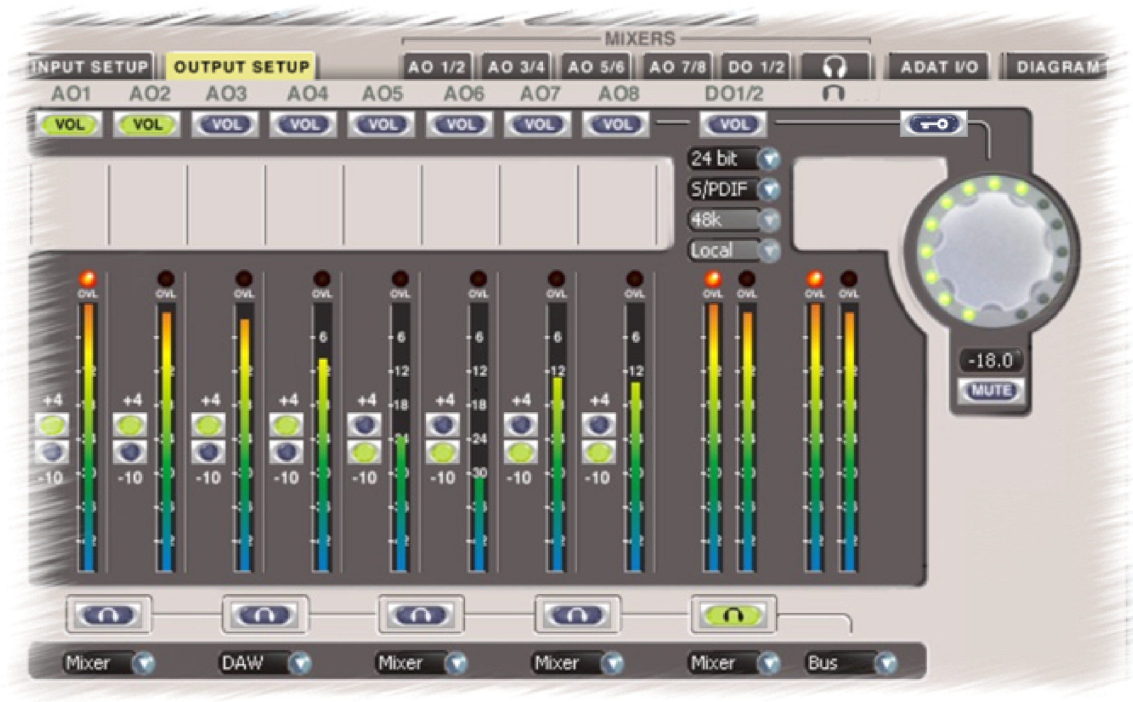
Overkiller progressive limiters are selectable for each analogue input, with phase-reverse and high-pass filters being available for mic and instrument inputs. Mic inputs have +48V phantom power switchable per-channel. Inputs 1&2 have an RIAA de-emphasis filter selectable as an alternative to the high-pass filter, allowing them to be used with vinyl decks. MS matrixing is available for mic or line inputs 1-4, allowing use of mid-side microphone configurations.

The DI strip can be switched between RCA and TOSLINK S/PDIF inputs, and UNLock and ASNC indicators are provided. UNLock is lit when no S/PDIF carrier is detected at the selected input; ASNC (asynchronous) is lit when the incoming carrier is not locked to Orpheus' selected sync source.

All inputs have high-resolution peak metering, with overload indication 0.05dB below clipping; the analogue input meters also have Overkiller-active indicators which light dynamically when the Overkillers are limiting.

6.1.3 Output Setup tab

The Input Setup tab contains the controls and status indicators for all functions of the analogue outputs and S/PDIF outputs as described in the hardware section.



Line output level is switched between +4dBu and -10dBV nominal by the '+4/-10' radio buttons.

The stereo digital output has a versatile word-length control using TPDF dither or Prism Sound SNS (Super Noise Shaping), and the RCA connector can be switched to operate as either S/PDIF or AES3 as required. If the sample rate converter (SRC) is configured in the digital output, a separate sync source and sample rate can be selected.

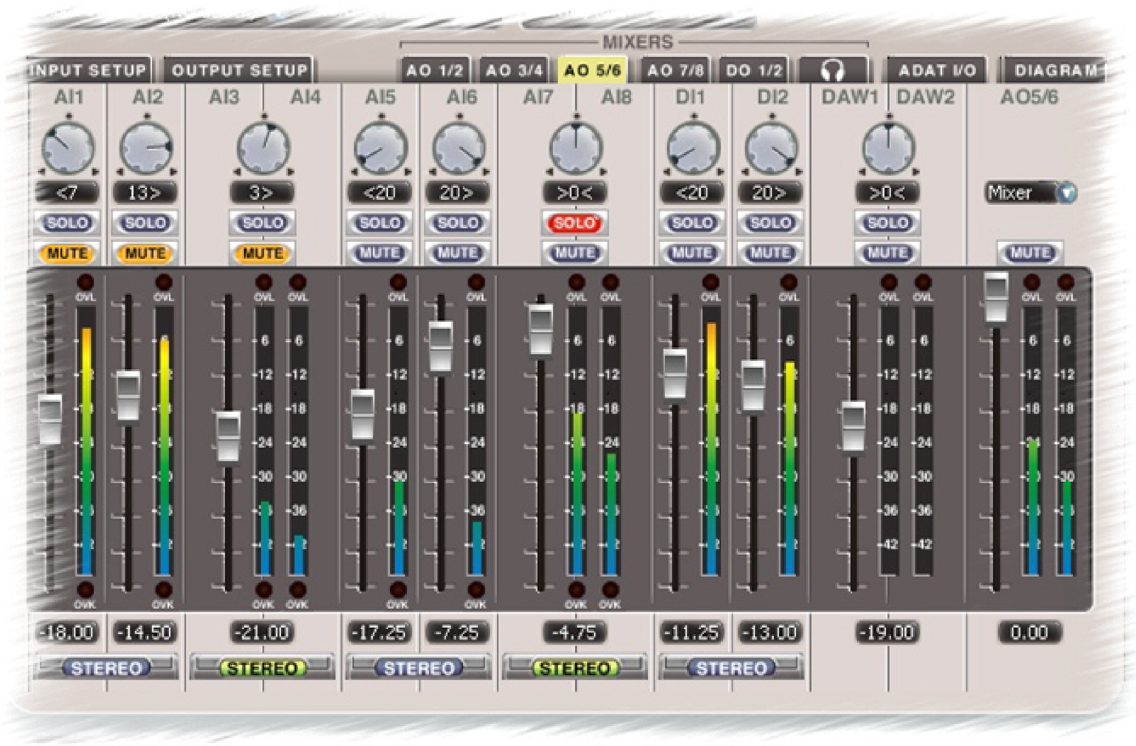
An assignable level control is available, for use as a stereo or surround monitor volume control, which can be assigned to any desired outputs using the row of 'VOL' buttons. The volume control can be adjusted using the mouse, as well as with the front-panel knob. There is also a mute button and a numerical readout/setting box. Note that changes to the assignment of the level control can be prevented by engaging the lock button (marked with a key symbol) just above the level control. When engaged (red) the lock button prevents changes to the level control assignment in order to avoid accidental full-level output.

Below each stereo output strip is a drop-list control allowing each output pair to be fed either directly from the workstation ('DAW') or from a dedicated low-latency foldback mixer ('Mixer'). Each stereo mixer can mix any of Orpheus' analogue or digital inputs to the output pair, along with the DAW feed, as described in the following section. The headphone output pair has a third 'Bus' setting which allows it to monitor any of the other output pairs using the row of headphone bus radio buttons.

All outputs have high-resolution peak metering, with overload indication 0.05dB below clipping.

6.1.4 Mixer tabs

The Mixer tabs control the low-latency foldback mixers which are available for each analogue output pair, plus the stereo digital and headphone outputs. The mixers are enabled using the drop-list controls on the Output Setup tab, which are duplicated in the output strip of each respective mixer.



Each input channel has a fader, a high-resolution peak meter, with overload indication 0.05dB below clipping, plus mute and solo buttons and a pan-pot. By engaging the 'Stereo' button beneath an input pair, the pair is controlled by a single stereo fader, mute button and solo button, and the pan-pots are replaced by a single balance control. The DAW contribution and output strip are always in stereo mode.

Note that if the mixer tab is not active (because the output pair is in 'DAW' mode), the mixer controls are still available but the output mode control in th output strip is highlighted in red.

For more information, see Output mixers in the hardware section.

6.1.5 ADAT tab

The ADAT tab contains eight high-resolution peak meters for the ADAT inputs and outputs. The meter bank is assigned to the ADAT outputs by selecting 'Send' or to the ADAT inputs by selecting 'Return'

6.2 Orpheus drivers

For Windows systems, Orpheus is supplied with a driver which provides ASIO and WDM connectivity to the device. This driver is installed when the initial installation is performed. Thereafter, this connectivity is permanently available. It is not necessary to run the Orpheus Control Panel applet for applications to be able to use Orpheus; however, the applet is needed if any but the most basic control of Orpheus' functions is required.

The same is true in Mac systems, except that no driver is installed because OS X is able to operate Orpheus directly to obtain basic Core Audio functionality. However, whilst some degree of control is possible from the Mac's Audio MIDI Setup, the Orpheus Control Panel applet must still be run to control the majority of the unit's functions.

Part 7

Technical topics

7 Technical topics

The following sections contain detailed discussions of various relevant technical issues. The content of these sections is not required to operate Orpheus, but is provided merely as background information.

7.1 Stability and latency

Since audio production has found its way inside the computer, new problems concerning issues of stability and latency have arisen.

Pre-computer digital audio gear introduced the concept of delays through devices, which hadn't usually been the case with analogue equipment. This was an inevitable consequence of sampling the audio, and passing the samples through multiple layers of buffering during conversion, processing and interfacing operations. However, the 'latency' (buffer delay) was generally quite short and didn't usually cause problems even in delay-sensitive applications such as live sound or over-dubbing. Reliable operation was generally guaranteed, since the digital devices were essentially 'sausage machines' performing nothing but the same limited series of operations repeatedly.

When general-purpose computers began to be used for audio production, problems with latency and stability suddenly had to be addressed. The reason is that computers are always busy doing other things than processing audio, even in situations where the operator is only interested in performing that dedicated task. Because of this, the computer generally accumulates a large buffer of incoming audio samples, which are then processed whilst a new buffer is being collected. Even though the required processing can (hopefully) be accomplished faster than real-time (i.e. the sample processing rate is faster than the sample rate), there is always the possibility that the computer may be called upon to interrupt its processing of the audio in order to deal with some other essential routine task, such as maintaining screen graphics, moving data on and off disc, servicing other programs etc. In non-optimized systems, tasks such as collecting emails, virus-checking and countless low-importance system operations can interrupt audio processing. Without the accumulation of sample buffers, any interruption taking longer than about one sample period (1/fs) would cause incoming audio samples to be missed, resulting in disruption of the audio signal. Nearly every kind of interruption is long enough to do this. However, with a large enough buffer, the interruptions don't cause audio to be disrupted so long as the computer has enough time available during the buffer period to process the entire buffer. This problem doesn't only happen for incoming samples: audio outputs from the computer must likewise be buffered so that a continuous output stream can be maintained even when the processor is called away for a while.

Why is this a problem? First of all, the amount of latency required in order for a particular computer with a particular audio processing and non-audio workload not to suffer audio disruptions can be problematically large. This is particularly the case in live sound and over-dubbing situations where the delay between the computer's input and output has to be essentially imperceptible. This is often difficult or impossible to achieve, unless the computer has a powerful processor, a lot of memory, a heavily audio-optimized operating system workload, an efficiently written audio processing program, and not too many audio channels, not too much audio processing complexity, and not too high a sample rate. The operator merely has to make sure that all these conditions are met, and all will be well!

But how do you do that? Even if we worry only about the computer and operating system themselves, the duration and frequency of interruptions is very non-deterministic: something can happen very infrequently which causes a huge interruption. This might not be a problem: you can always run that track again (assuming you noticed the glitch) - but what if you're recording an important one-off live event? Even worse, the onset of trouble is greatly affected by audio factors such as number of tracks, sample rate, how many EQs are in use, etc. This makes the onset of instability even harder to predict reliably.

On the other hand, situations where latency is critical are relatively few, so it is normally OK to operate generous buffers - such as in the live recording example.

In the case of Orpheus, problems of latency and stability are improved by a couple of useful features:

First of all, the operator can control the buffer delays within the Mac and Windows drivers directly, irrespective of what buffering is employed by the user's particular audio software. It is generally recommended that these buffer delays are set long, in order to provide best stability. However, for the user with a powerful and tightly-optimized setup, who has contained audio processing task and needs low latency, the buffer delay can be minimized. For more information, see the Unit and Global settings section..

For foldback and over-dubbing situations, all of Orpheus' outputs (analogue 1-8, S/PDIF DO, and headphone outputs) have a comprehensive mixer capability which can mix any of the unit's inputs with each output's computer feed in order to build a dedicated monitor mix with extremely low latency. Incoming audio to the mix doesn't have to go in and out of the computer at all - the mix is handled within the Orpheus hardware itself. For more information, see the Output Setup tab and Mixer tabs sections.

7.2 Clocking and jitter

Good clock stability is probably the single most important issue separating good-quality analogue interfaces from the rest. With the linearity of modern A/D and D/A converter chips beginning to rival and exceed the performance of the best analogue circuits, digital recordings would already be ‘ beyond reproach’ if clock stability did not so often degrade their potential quality.

Why is good clock stability so rare? Probably because most conversion equipment has to compromise between clock stability, operational requirements and cost. The ideal clock system in an A/D or D/A converter would be ultimately stable, i.e. would exhibit no jitter (frequency variations) at the point of conversion, whether operating from an internal clock or from an external synchronization reference of any format and at any sample rate. But this is a very tall order for circuit designers, especially if they are on a budget.

Why are good clocks so rare?

Most analogue interfaces can provide workmanlike performance when internally clocked, since this is only a matter of providing a stable clock oscillator (or range of oscillators) at a fixed frequency (or frequencies) – although even this is not always well-executed. The real problem is that in many installations the analogue interfaces can almost never operate from their own internal clocks since they must be slaved to an external reference sync, or maybe to a clock from a host computer.

The externally-clocked design challenge has traditionally been a trade-off. since the more stable a clock oscillator is, the less is its range of frequency adjustment: but we would ideally like an oscillator which can operate over a wide range of sample rates, perhaps from <44.1kHz to >48kHz, plus multiples thereof. But such an oscillator would inevitably have poor stability – at least in terms of the stringent requirements for high-quality audio conversion. On the other hand, if we limit the range of rates at which the oscillator needs to operate to small ‘islands’ around the standard sample rates we could use a bank of oscillators, selecting the appropriate oscillator according to our desired sample rate. But this is expensive and, in any case, the 'pull-range' of an ordinary quartz crystal oscillator is still generally insufficient to meet the tolerance demands of the digital audio interfacing standards.

As well as a very stable clock oscillator, a good sounding converter must have a PLL (phase-locked loop) with a loop-filter which steeply attenuates incoming reference jitter towards higher frequencies. Unfortunately, even if sourcing equipment provides a reference clock with low jitter, cabling always adds unacceptable amounts, especially poor quality or high-capacitance cable, which results directly in sampling jitter in the analogue interface if jitter-filtering is inadequate.

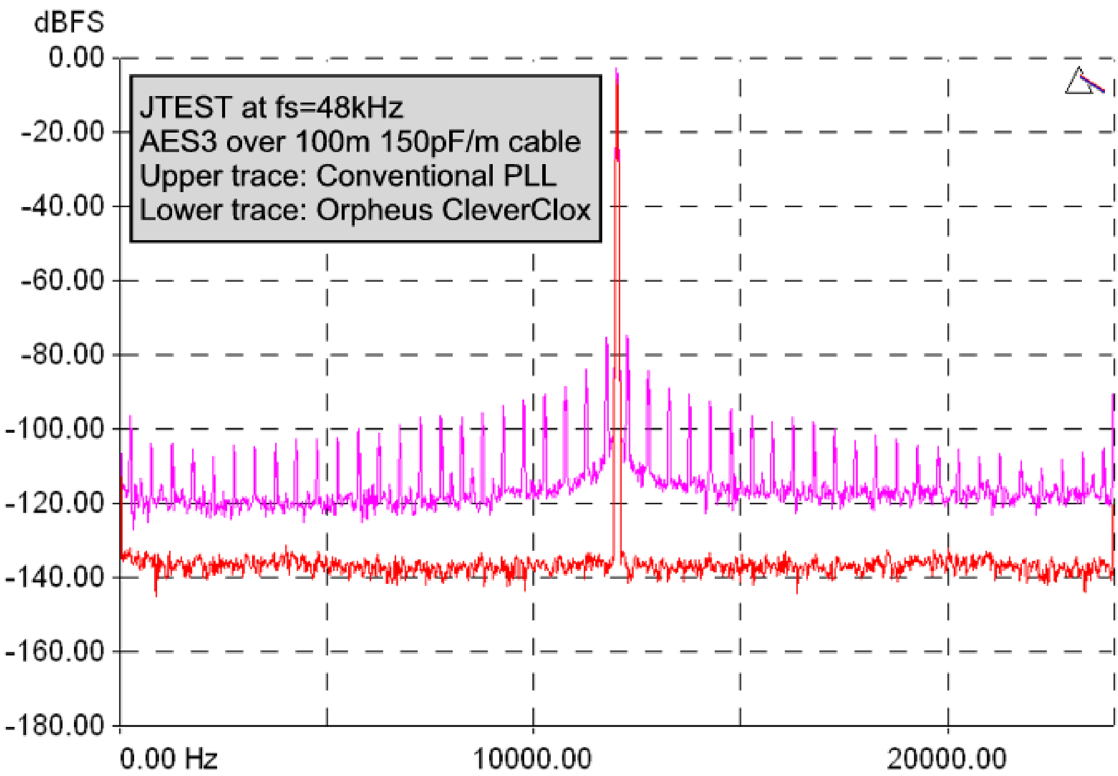
Prism Sound's unique CleverClox technology breaks these traditional constraints, allowing a low jitter clock to be re-created from any reference sync, no matter how much jitter it has and no matter what its frequency.

But why is clock jitter so important?

Analysis of sampling jitter

Analysis of sampling jitter (small variations in the sampling intervals of an A/D or D/A converter) shows that it produces a similar effect to phase modulation, where distortion components appear as ' sidebands' spaced away from the frequency of a converted tone by the frequency of the jitter itself. These components get louder as the amount of jitter increases, but also as the frequency of the converted tone increases. So sampling jitter produces distortions which should sound much worse than conventional analogue harmonic distortions, since the spurious components appear at aharmonic frequencies. High audio frequencies should suffer worse distortion than low frequencies. For low-frequency jitter, the resulting distortion sidebands appear close in frequency to the audio signals which produce them – this should mean that they are 'masked' from our hearing by the same psycho-acoustic phenomenon upon which are based sub-band (perceptual) coding schemes such as MPEG. This is fortunate, since it is quite difficult for a PLL to remove jitter to a good degree even at moderate frequencies, but for very low frequencies it would be very difficult indeed.

The graph below shows the effects of 'JTEST', a special test stimulus to expose jitter susceptibility of D/A converters. JTEST is basically an fs/4 tone (12kHz at fs=48kHz) which is specially coded to cause an AES3 or S/PDIF carrier transmitted over a lossy cable to become very jittery by the time it reaches the receiving D/A converter. The jitter produced has regular frequency components fs/96 apart (500Hz at fs=48kHz). The quality of the D/A converter's jitter rejection is shown by the degree to which it suppresses the resulting 500Hz-spaced side-tones. In the example below, the upper trace shows the poor jitter rejection of 'conventional' D/A converter design, where the conversion clock is derived directly from the AES3 or S/PDIF receiving chip, without any further jitter filtering. Remember that none of these side-tones is present in the digital audio signal - they are caused only by jitter. The lower trace shows almost complete jitter rejection across the band by the CleverClox process in Orpheus.



Listening experience

In practice, it seems that the benefits of careful clock design are very apparent in listening tests. On the other hand, it can sometimes be difficult to expose the shortcomings of converters with poor clocks, because these units often have other analogue problems whose severity might obscure jitter-related effects.

In general, some of the widely-noted effects of sampling jitter are not surprising – for example the muddying of brass, strings and high-frequency percussion and the loss of stereo (or multi-channel) imaging. These are well explained by the worse distortions which result in the lab at loud, high frequencies, and the way that sampling jitter produces quiet, aharmonic components, perhaps only subliminally perceptible, which blur our impression of the ambience which creates a soundstage.

Other effects are harder to explain – for example there is wide observation that large amounts of sampling jitter can take the edge off extreme bass rendition. Such reports are probably too widespread to be ignored, but defy explanation within current theory.

Orpheus and CleverClox

Orpheus is designed to source clocks which are as stable and accurate as possible, and also with the aim of being insensitive to the quality of incoming clocks. It is designed to remove jitter from any selected reference sync source before it is used as a conversion timebase, so as to eliminate any audible effects of sampling jitter, whatever sync source is used.

Orpheus does this with the help of Prism Sound's unique CleverClox clock technology, which removes the jitter from any selected clock source down to sub-sonic frequencies, without the need for a narrow-band quartz VCO. CleverClox can adapt to any reference, irrespective of frequency, and regardless of how much jitter it has, derives an ultra-stable conversion timebase.

7.3 Dither and noise-shaping

Orpheus can dither or noise-shape its digital output to produce high-quality 16 bit output (for, say, a CD master) from 20 bit or 24 bit recordings. This section discusses the principles and choices involved in word-length reduction.

Truncation and dithering

There are many points in a digital audio signal path where precision can be lost. For example, in a digital transfer from 24-bits to 16-bits, or in an analogue to digital conversion. In this situation it is not sufficient just to discard low-order bits – this causes truncation distortion, characterised by aharmonic frequency components and unnatural, harsh decays.

Instead, it is preferable to use some sort of ‘dithering’ process, whereby the truncation process is linearized by modulating the signal prior to the truncation, usually by the addition of a small amount of noise. By adding a random element to the truncation decision, small components as far as 30dB below the noise floor can be accurately represented, and an analogue-like low-signal performance can be realised. This is achieved at the expense of slightly raising of the noise floor, although with some dithering schemes such as noise-shaping, linearization can be achieved with no noticeable increase in noise.

How can dithering allow information to be preserved below the least-significant bit? It seems impossible. Consider a simple example where the audio samples are numbers between one and six, and we are going to ‘truncate’ them (i.e. reduce their resolution) so that numbers from one to three become zero, and those from four to six become one. Clearly much information will be lost, and all excursions of the signal between one and three and between four and six will not affect the output at all. But if we throw a die for each sample, add the number of spots to that sample, and translate totals of six and below to zero and totals of seven and above to one, we have a simple dithering scheme. Input samples of three will be more likely to result in outputs of one than will inputs of one. The throw of the die is our dither noise. Since all the faces of the die have an equal chance of occurring, this is known as ‘rectangular probability distribution function’ (RPDF) dither, which in fact does not produce perfect linearization. We actually use ‘triangular probability distribution function’ (TPDF) dither, which is like throwing two dice with a resultant increase in the probability of medium sized numbers – totals of two and twelve occur much less often than seven.

Noise shaping

It is possible to reduce the subjective effect of the added dither noise by either using spectrally weighted (''blue'') dither noise, which is quieter in the more sensitive registers of the ear, or by an even more effective technique called ‘noise shaping’.

Noise shaping is just like conventional dithering, except that the error signal generated when the unwanted low-order bits are discarded is filtered and subtracted from the input signal. You can’t get something for nothing – the error cannot be simply cancelled out, because we already know that the output hasn’t got enough bits to precisely represent the input. But by choosing an appropriate shape for the error filter, we can force the dither noise / error signal to adopt the desired shape in the frequency domain – we usually choose a shape which tracks the low-field perception threshold of the human ear against frequency. As can be seen from the plots below, this has the effect of actually lowering the noise floor in the more sensitive frequency bands when compared to the flat dither case.

The theory of noise shaping has been around for a long time – certainly since well before DSP in real-time was feasible for audio signals. It has applications in many signal processing and data conversion applications outside audio. It has been well researched, and is not in the least bit mysterious. ‘Proprietary’ word-length reduction algorithms are generally conventional noise shapers. Assuming that the basic implementation and dither levels are correct, the only significant freedoms available to the designer are to choose the actual shape of the noise floor, and to decide how to adapt this (if at all) to different sample rates.

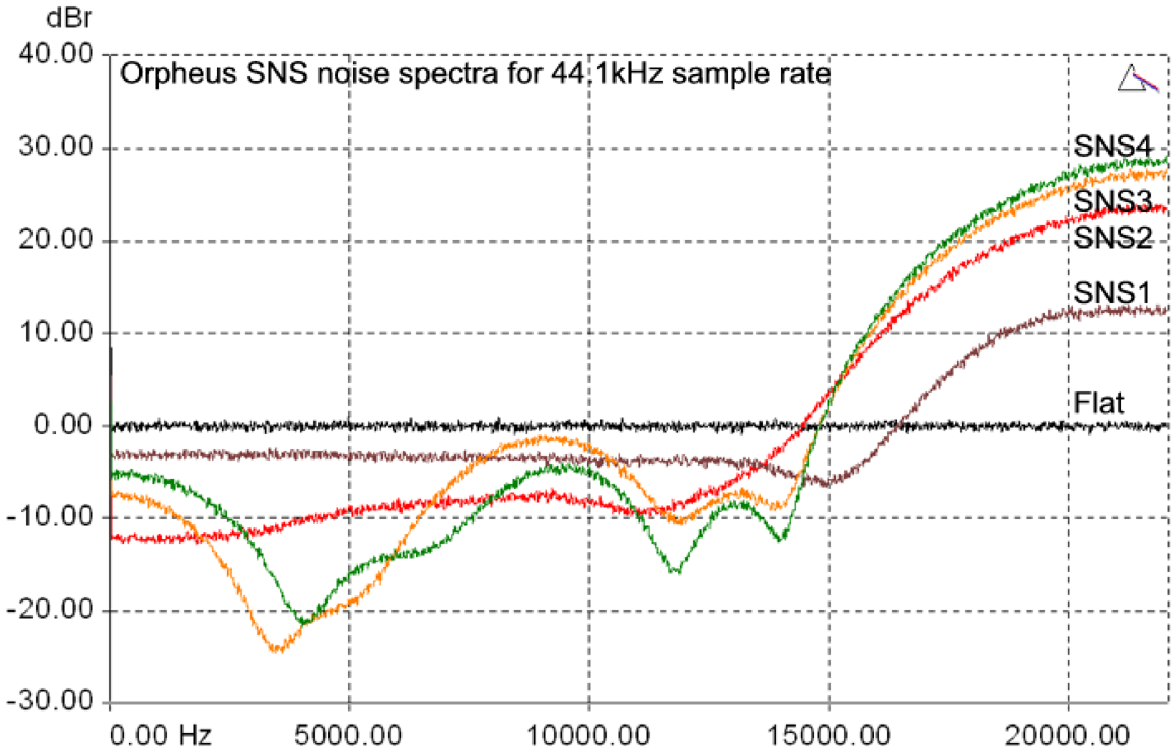
Prism Sound SNS (Super Noise Shaping)

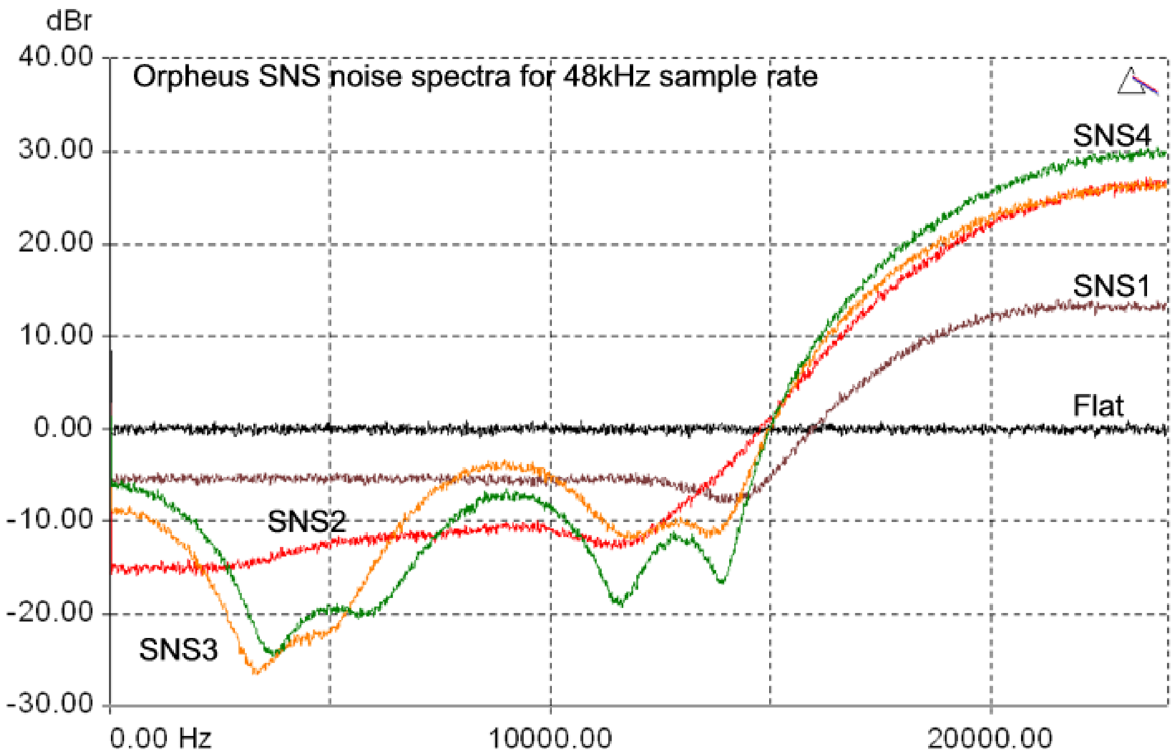
Orpheus provides a comprehensive choice of dithering and noise-shaping processes. These comprise ‘flat’ dithering, plus a selection of four Prism Sound ‘SNS’ (‘Super Noise Shaping’) algorithms. All produce high-quality 16 bit output: the choice of which one to use is purely subjective. The four SNS algorithms are designated SNS1 to SNS4, in increasing order of the degree of shaping. The spectra of the four SNS algorithms are shown below. Note that, unlike some noise shaping algorithms, SNS spectra are adjusted automatically to provide optimum subjective advantage at each different sample rate. The spectra are shown below for 16-bit output, at 44.1kHz, 48kHz and 96kHz sample rates.

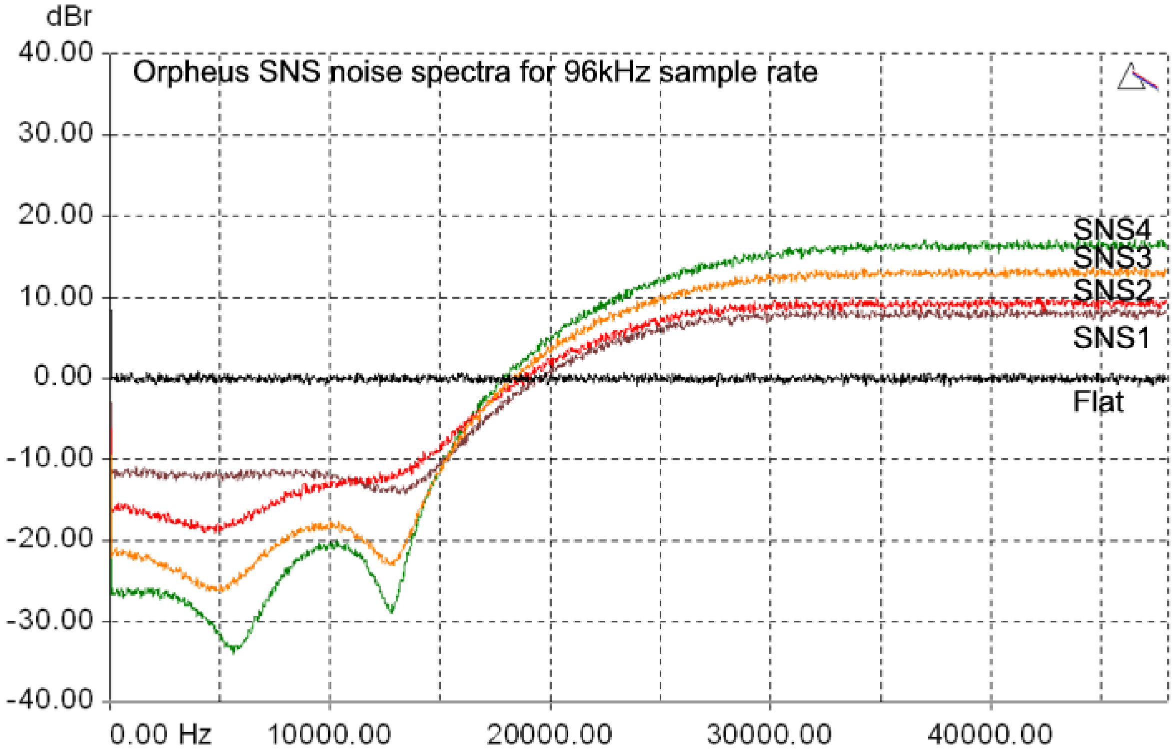
	SNS1 provides the smallest subjective noise advantage, but only applies limited noise-lift at quite high frequencies. In many applications, particularly those where the program material is already quite noisy, this type of shaper is preferred.
sns	SNS2 is a happy medium. It provides a good amount of subjective lowering of the noise floor, but with addition of only moderate amounts of high-frequency noise. It also has the advantage that the noise floor remains subjectively white, even when artificially amplified. In the fifteen years since Prism Sound first developed the four SNS curves, SNS2 has been the most widely preferred.
	SNS3 and SNS4 are ‘optimal’ shaper designs – their shaping is quite extreme in order to get the maximum theoretical subjective improvement in noise performance based on an average human low-field sensitivity curve. This results in the addition of larger amounts of high-frequency noise. These shapers are only really useful if the original recording has a very low noise floor.

It is difficult to assess the difference in sound between different noise shapers for any given program material, since their effects are at very low amplitudes (the 0dB line on the plots below represents flat dither with an rms noise amplitude of about –93.4dBFS). It is tempting to audition noise shapers by using a low signal level and boosting the shaper output by tens of dBs in the digital domain prior to monitoring. Using this method it is easy to hear that the noise floor of more extreme shapers is clearly not white – switching, say, from SNS1 to SNS4 sounds like shhhhh..ssssss as the noise is shifted towards the higher frequencies. However, this is not really a meaningful test since the sensitivity of the ear at different frequencies is very dependent on level, and the design of the more extreme shapers is in any case intended to render the noise floor completely inaudible at normal listening levels. Ultimately, the only 'right' choice of noise shaper is the one which sounds best for the material. SNS2 is a good starting point for most situations.

The Prism Sound SNS logo shown above is found on many of the world's finest CDs, and is recognised as a standard of technical excellence. The logo, and accompanying sleeve note, is available by contacting sales@prismsound.com.







7.4 Analogue interconnections

To maintain the high sound quality of Orpheus, it is important to follow some basic guidelines when making analogue connections to the unit. This section discusses some things to watch out for.

Cable quality

Use of good-quality, heavy duty audio cables is recommended. For microphone use, quad-twisted cables may give best results. Cables with heavy screens are recommended, especially for unbalanced use. Owing to mechanical differences between connectors from different manufacturers, it is advised to use cables with identifiable connectors from reputable manufacturers. This is especially true for jacks, where unreliable tip connection can owing to the slightly non-conforming shape of some manufacturers' parts. Neutrik connectors are used in Orpheus, and these are recommended to ensure reliably-mating cables.

Balanced versus unbalanced connections

Where possible, balanced interconnections should be used, since the audio signal is represented as a voltage difference between two dedicated conductors (neither of which is ground-coupled), which are usually closely-twisted to ensure that any interference pickup is cancelled out. In unbalanced connections, the signal is represented as a voltage difference between a single signal conductor and an accompanying ground conductor. Where dynamic ground-potential differences exist between the source equipment and the receiving equipment, this difference is effectively added to the unbalanced audio signal.

This effect has long been familiar in audio systems as 'hum loops', where the variation in ground potential occurred at line-frequency, and was developed by the flow of line-frequency currents to linear power supplies. Hum loops were usually resolved by either steering the currents along non-critical routes by re-arranging the topology of the system ground interconnections, or by mass-interconnection the system grounds using heavy gauge cable so as to minimize the hum voltage resulting from the current.

Obviously many items of analogue audio equipment only have unbalanced connections; this is especially true of consumer equipment, which is often used for monitoring even in professional studios. If you must use unbalanced connections, keep them as short as possible and use good-quality cables with substantial screens. If you have a choice, keep the signal level as high as possible on the interconnection, since this will make any interference proportionally less noticeable.

Instrument connections are often particularly vulnerable to hum and other interference, since they are usually unbalanced and low-level, and frequently employ a long cable not selected for its interference-immunity qualities. Also, the source impedance is usually high, making the connection particularly vulnerable to interference.

Some digital audio and computer equipment with switched-mode power supplies can cause particularly troublesome interference problems, especially for low-level, unbalanced signals. This is discussed in the following section.

Interference

The increasing use of low-cost digital equipment and computers in the audio production process results in various potential problems for the remaining analogue devices. It is well-known that the hostile power and EMC environment inside a typical computer is likely to be the limiting factor governing the audio quality of an internal analogue sound card. A solution to this is the use of external 'sound cards', such as Orpheus, with their own enclosures and power supplies allowing adequate space, power and electromagnetic peace and quiet for the well-being of studio-quality analogue circuits.

However, even the sound quality of external devices can be compromised by the proximity of some types of digital equipment. Many low-cost switched-mode power supplies emit interference which can compromise system audio quality even at a distance. The hostile mechanism is usually 'conducted interference', wherein the high-speed switching action of the power converter results in voltage and current transients being conducted back down their power cords. If the equipment is connected to mains safety-ground, transients can also be conducted down the ground connection. Radiated emissions (airborne radio interference) can also be a problem, but it is less common that this will have such a serious effect on audio quality.

Conducted power-line interference can cause problems in analogue equipment within the installation if its own power supply allows the transients to pass through to the audio circuits. However, conducted ground interference can be even worse since, if the ground connection of the analogue equipment is modulated by switching interference, there is nothing that the designer of the equipment can do to combat it.

How much any conducted ground interference affects audio quality depends on many factors, mostly to do with how the various analogue boxes in the system are interconnected and grounded. Where possible, high-level balanced connections should be used, just as in the case of hum-loops as discussed in the previous section.

Where ground-potential variations are caused by switching power supplies, the effect can be more difficult to resolve, since the signals can occur at more noticeable frequencies: although the supplies usually switch at frequencies too high to hear, the frequency is often modulated by variations in the load current over time, resulting in a continuous modem-like chirping in which can be heard particular events such as computer screen updates, disk activity etc.). Another problem is that even heavy ground cabling may not reduce the effect of the interference, since high-frequency currents may not see much less resistance in a thick conductor than a thin one.

How do the equipment manufacturers get away with this? Surely there are stringent regulations covering conducted and radiated emissions? Well that's true, but the level of emissions which can result in audible degradation of low-level, unbalanced audio interconnections are well below legislation levels. Unfortunately, computer power supplies (and especially the switching wall-warts and line-warts which power notebook computers and other small items) are amongst the worst offenders.

Vinyl decks

Orpheus is equipped with an RIAA de-emphasis filter to allow direct connection of a vinyl deck, as described in the Analogue inputs section. Since vinyl decks usually have a low-level, unbalanced output it is important to minimise interference as discussed above when connection a vinyl deck.

Since most magnetic cartridges require a higher input impedance than that of the Orpheus microphone preamplifier input, it is usually best to connect a vinyl deck to the instrument inputs using a pair of phono-to-mono-jack cables. The instrument gain controls can then be set to an appropriate level for the particular cartridge. The 1MR input impedance of the instrument inputs will work satisfactorily with most magnetic phono cartridges (which are 'moving magnet' types), but with some cartridges, improved frequency response and noise levels can be achieved by fitting the cartridge's required load resistance (usually 22kR or 47kR) across the instrument input terminals; this is best achieved by soldering it inside the jack. Moving coil cartridges have a lower output level and require a lower preamplifier input impedance. These are best connected to Orpheus' mic inputs, or may require a dedicated preamplifier.

Most vinyl decks have a ground wire separate from the audio connectors. Connection of this wire for lowest hum is often a matter of trial and error. Ideally this should be connected to Orpheus' analogue signal ground (the outer of the instrument input jacks, or pin 1 of the mic input XLRs). Since no dedicated terminal exists on Orpheus, it is usually easiest to connect the wire to the outer of one of the deck's unbalanced output connectors. In some situations, a direct connection to local mains ground may work better.

In summary

- Use good-quality cables with reputable connectors;
- Use balanced connections where possible; if you must use unbalanced connections, keep them short;

- Ensure that signals passing between equipment do so at as high a level as is practical;
- If switching interference is heard, try to identify the source equipment by unplugging things one by one. When you find the culprit, either re-plug it a long way from the audio equipment, or use a power filter, or both.

Part 8

Specifications

8 Specifications

Front Panel

Instrument inputs 1&2:	Two 6.3mm mono jack sockets, auto-select to analog inputs 1&2 when plugged
Headphone outputs 1&2:	Two 6.3mm stereo jack sockets, each with illuminated volume control
Master volume control:	Assignable to any selection of outputs, with LED halo indication
Level meters:	Multi-segment, multi-color bargraphs with overload indication; eight for analog, two for digital, assignable to inputs or outputs
Input selector indicators:	Indicate modes of analog inputs 1-4 as mic/line/instrument; plus phantom power indicator for mic mode
Overkiller-active indicators:	For every analog input, lit when Overkiller limiters are acting
Digital input indicators:	Input unlocked, and SRC (sample-rate converter) selected
Digital output indicator:	SRC (sample-rate converter) selected
Sync indicator:	Master, lit when interface is providing system sync
Standby button:	With standby indicator (also flashes when unit is in 'identify' mode)

Rear Panel

Mic/Line inputs 1-4:	Four combo connectors with XLR sockets for mic input and 6.3mm TRS jack sockets for line input (balanced or unbalanced)
Line inputs 5-8:	Four 6.3mm TRS jack sockets (balanced or unbalanced)
Line outputs 1-8:	Eight 6.3mm TRS jack sockets (balanced or unbalanced)
Digital inputs:	RCA socket for S/PDIF in, TOSlink for S/PDIF or ADAT in; (RCA can operate as AES3 input using XLR-RCA adapter supplied)
Digital outputs:	RCA socket for S/PDIF out, TOSlink for S/PDIFor ADAT out; (RCA can operate as AES3 output using RCA-XLR adapter supplied)
MIDI I/O:	Two 5-pin DIN sockets, in and out
Wordclock I/O:	Two BNC sockets, output and input (75R)
FireWire ports:	Two IEEE1394A ports (allowing cascading of multiple units)
Mains power:	3-pin 6A IEC inlet

Software Support

Mac OS support:	OS X 10.3 or later, PPC or Intel platform
Windows OS support:	Windows XP, Vista or later
Mac audio driver:	Core Audio device
Windows audio driver:	WDM and ASIO devices
Control Panel applet:	Graphical user interface for control of single or multiple Orpheus units under Mac OS X or Windows

Analog Line Inputs

Configuration:	Electronically balanced, with fully-balanced analog signal path
Input sensitivity:	Switchable '+4dBu' (0dBFS=+18dBu) or '-10dBV' (0dBFS=+6dBu)
Input impedance:	14.5kR
Unbalanced mode:	Automatic
Total harmonic distortion:	-117dB (0.00014%, -0.1dBFS)
THD+n:	-111dB (0.00028%, -0.1dBFS)
Dynamic range:	116dB (-60dBFS)
Gain accuracy:	±0.05dB
LF roll-off:	-0.05dB at 8Hz; -3dB at <1Hz
HF roll-off:	fs=44.1kHz: -0.05dB at 21.1kHz; -3dB at 22.0kHz fs=48kHz: -0.05dB at 23.0kHz; -3dB at 23.9kHz fs=96kHz: -0.05dB at 32.0kHz, -3dB at 47.9kHz fs=192kHz: -0.05dB at 32.0kHz, -3dB at 78.0kHz
CMRR:	20Hz..20kHz: >70dB
Inter-channel cross-talk:	1kHz: <-140dB; 20Hz..20kHz: <-120dB
Inter-channel phase:	10Hz..5kHz: ±0.25º, 5kHz..20kHz: ±1.0º, 20kHz..50kHz: ±2.0º
Overkiller:	Progressive limiter, auto-aligning, selectable per channel
Impact filter:	High-pass filter, -3dB at 80Hz, 40dB/decade (selectable for channels 1-4; whether in line, mic or instrument mode)
RIAA de-emphasis filter:	Response accuracy: ±0.22dB at fs=44.1kHz; ±0.14dB at fs=48kHz;± 0.015dB at all other sample rates
M-S matrix:	Allows direct connection of mid-side mics, or non-matrix mic-preamps (selectable for channels 1-4; whether in line or mic mode)

Microphone Preamplifiers

Configuration:	Electronically balanced, with fully-balanced analog signal path
Gain:	10dB to 65dB in 1dB steps (0dBFS = -56dBu to -1dBu)
Gain accuracy:	±0.05dB
Input impedance:	5.5kR
THD	+10dB gain: -116dB at -0.1dBFS (0.00016%) +40dB gain: -110dB at -0.1dBFS (0.00032%)
THD+n:	+10dB gain: -108dB at -0.1dBFS (0.00040%)
Equivalent input noise (EIN):	+30dB gain: -128.5dBu (0R source); -126.3dBu (150R source) +40dB gain: -130.9dBu (0R source); -127.6dBu (150R source) +50dB gain: -131.2dBu (0R source); -127.7dBu (150R source) +60dB gain: -131.4dBu (0R source); -127.8dBu (150R source)
LF roll-off:	-0.05dB at 20Hz; -3dB at 5Hz
HF roll-off:	As per Analog Line Input data (dependent on fs)
CMRR:	50Hz/60Hz: >110dB at all gains 1kHz: >100dB at all gains 20kHz: >90dB at all gains
Phantom power:	+48V, switchable per channel

Instrument Preamplifiers

Configuration:	Unbalanced, high impedance buffer
Gain:	10dB to 65dB in 1dB steps, 18dB pad (0dBFS = -38dBu to 17dBu)
Gain accuracy:	±0.05dB
Input impedance:	1MR

Analog Line Outputs

Configuration:	Electronically balanced, with fully-balanced analog signal path
Output amplitude:	Switchable '+4dBu' (0dBFS=+18dBu) or '-10dBV' (0dBFS=+6dBu)
Output impedance:	100R balanced, 50R unbalanced
Unbalanced mode:	Automatic, with bootstrapping level compensation
Total harmonic distortion:	-107dB (0.00045%, -0.1dBFS)
THD+n:	-106dB (0.00050%, -0.1dBFS)
Dynamic range:	115dB (-60dBFS)
Gain accuracy:	±0.05dB
LF roll-off:	-0.05dB at 8Hz; -3dB at <1Hz
HF roll-off:	fs=44.1kHz: -0.05dB at 21.4kHz; -3dB at 22.0kHz fs=48kHz: -0.05dB at 23.2kHz; -3dB at 23.9kHz fs=96kHz: -0.05dB at 32.0kHz, -3dB at 47.8kHz fs=192kHz: -0.05dB at 32.0kHz, -3dB at 76.0kHz
Output balance:	>50dB
Inter-channel cross-talk:	1kHz: <-135dB; 20Hz..20kHz: <-120dB.
Inter-channel phase:	10Hz..5kHz: ±0.4°, 5kHz..20kHz: ±0.25°, 20kHz..50kHz: ±0.5°.

Digital Input

Formats supported:	S/PDIF (RCA or TOSlink), ADAT, ADAT SMUX (TOSlink)
Sample rates supported:	S/PDIF (2 channel): 44k1, 48k, 88k2, 96k, 176k4, 192k ADAT (8 channel): 44k1, 48k; ADAT SMUX (4 channel): 88k2, 96k
Channel Status:	Ignored
Word-length:	24 bits
AES3 operation:	Via S/PDIF RCA, using XLR-RCA adapter (supplied)
Sample-rate converter (SRC):	Selectable at S/PDIF input, allowing input at any sample rate
Bit-transparency:	Maintained (allows recording of Dolby or DTS streams)

Digital Output

Formats supported:	S/PDIF (RCA or TOSlink), ADAT, ADAT S/MUX (TOSlink)
Sample rates supported:	S/PDIF (2 channel): 44k1, 48k, 88k2, 96k, 176k4, 192k ADAT (8 channel): 44k1, 48k; ADAT S/MUX (4 channel): 88k2, 96k
Channel Status:	Full implementation, Consumer (S/PDIF) or Professional (AES3)
Word-length:	24 bits, or reduction to 16 bits using flat TPDF dither or Prism Sound SNS (Super Noise-Shaping - four alternative shapes available)
AES3 operation:	Via S/PDIF RCA, using RCA-XLR adapter (supplied)
Sample-rate converter (SRC):	Selectable at S/PDIF output; output rate can be referenced to Wordclock, DI or local clock at 44k1, 48k, 88k2, 96k, 176k4 or 192k
Bit-transparency:	Maintained in 24 bit mode (allows playback of Dolby or DTS streams to external decoder)

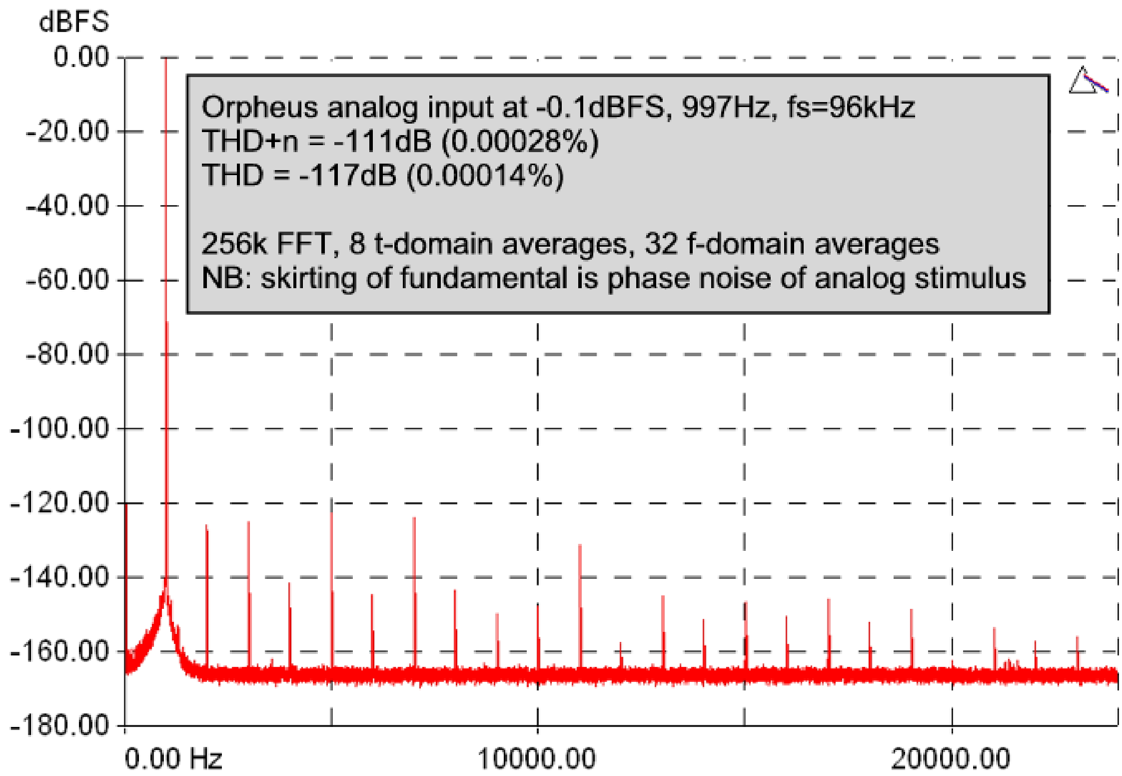
Synchronization

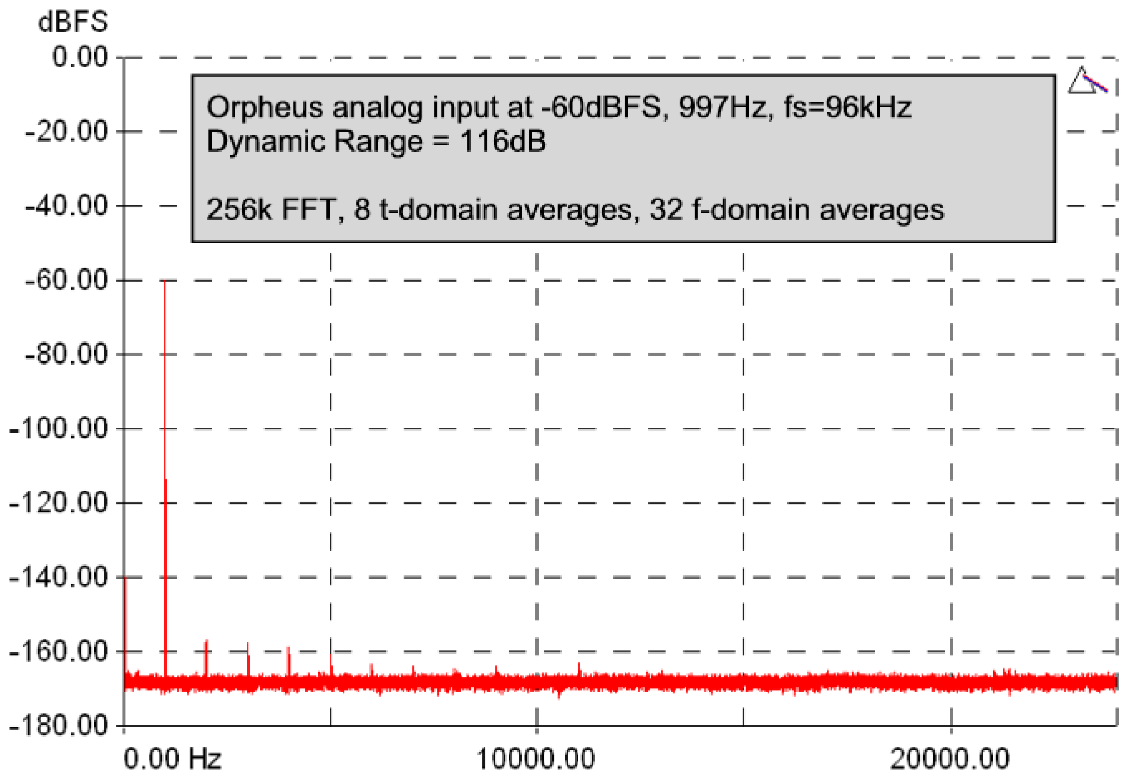
System sample rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz
Synchronization sources:	Master: Local, Wordclock, DI (S/PDIF input) or ADAT Slave: 1394 bus master (another FireWire interface or host computer)
Local clock accuracy:	+/-10ppm
Jitter rejection:	60dB/decade above 100Hz
Internal mixer delay: (total analog in to out)	44.1k: 0.57ms; 48k: 0.52ms; 88k2: 0.20ms; 96k: 0.18ms; 176.4k: 0.09ms; 192k: 0.08ms

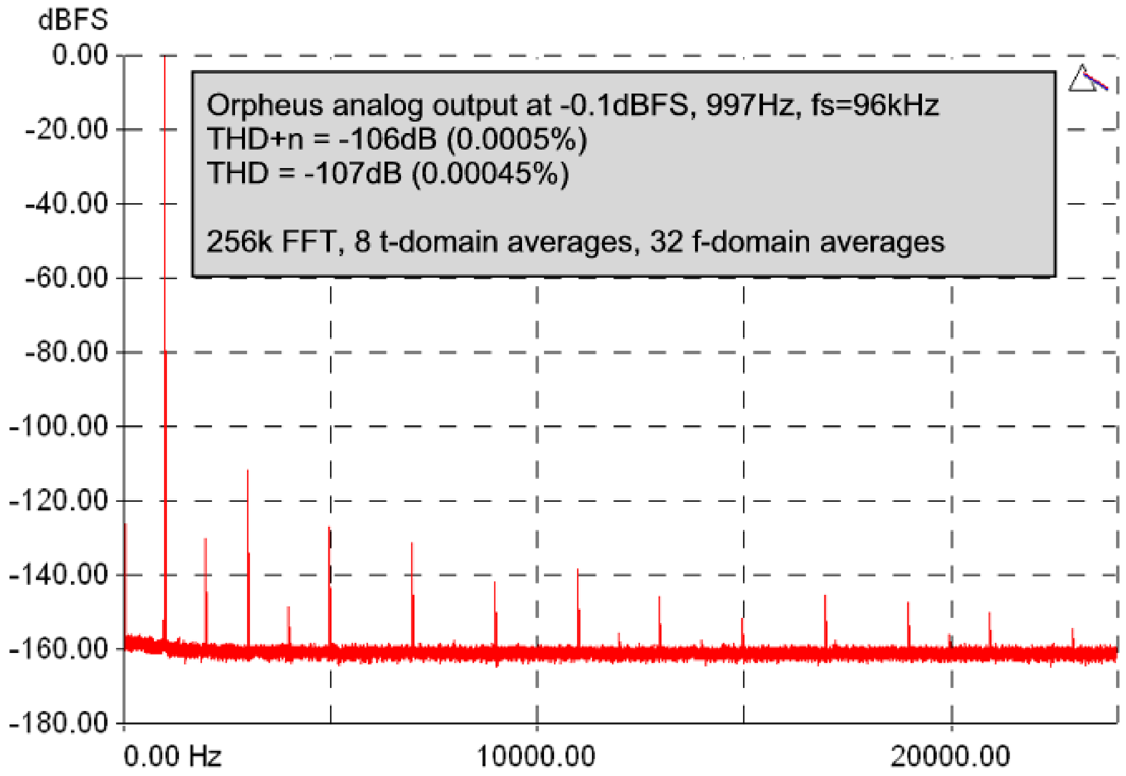
Physical

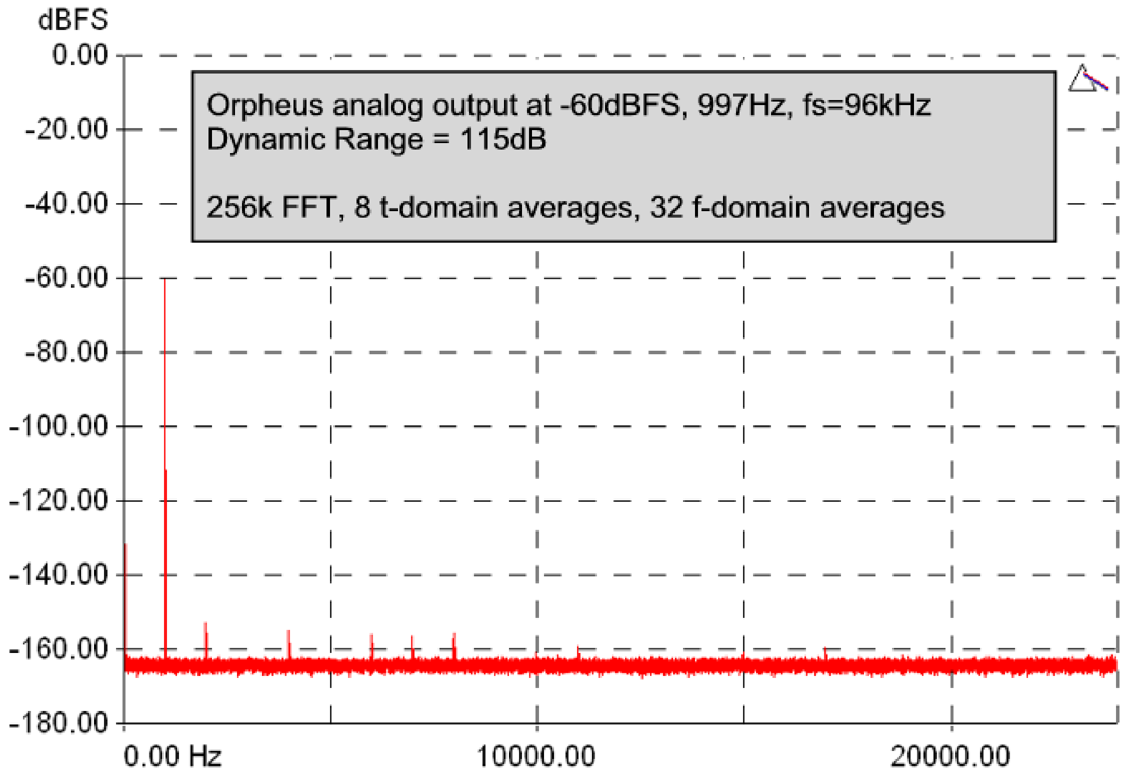
Dimensions:	Table-top (including feet): 440x290x50mm 1U rack-mount (including ears): 483x290x44.5mm
Weight:	3.7kg.
Mains voltage:	90VAC-250VAC, 50-60Hz
Power consumption:	35W
Fuse rating:	0.5A(T), 20mm, glass
Operating ambient:	0 to 35ºC, 85% maximum relative humidity

Performance Plots









Except where otherwise stated, audio performance data are typical, RMS, unweighted, 20Hz..20kHz figures, measured at 997Hz, using fs=96kHz and '+4dBu' sensitivity settings.

In keeping with our policy of continual development, specifications are subject to amendment without notice.

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prismsoundrecording

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