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CALLIA

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Operation Manual

Callia

Operation Manual

by Ian Dennis

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Part 1

General information

1 General information

Manual revision history

Rev	Date	Author	Notes
1.00	8th March 2016	I.G.Dennis	Product Launch
1.01	12th August 2016	I.G.Dennis	Added Linux support details

The latest version of this Operation Manual can be downloaded from the Prism Sound website at www.prismsound.com. It is provided in PDF format which can be viewed and printed using the Adobe Acrobat Reader, available free at www.adobe.com.

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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.

TO PREVENT POSSIBLE HEARING DAMAGE, DO NOT LISTEN AT HIGH VOLUME LEVELS FOR LONG PERIODS.

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 Callia

2 Introduction to Callia

Callia is a USB and S/PDIF D/A converter and pre-amplifier for domestic use. Prism Sound A/D and D/A conversion has been the technology of choice in the world's top recording and mastering studios for over 20 years; Callia brings the conversion quality of these renowned facilities into your home for the first time.

For a summary of Callia's capabilities, continue to the Features section.

For directions for getting started quickly, see the Getting started section below.

For detailed software installation procedures, see the Installation procedures.

For full details of Callia's controls and connections, look at the Hardware section.

2.1 Features

Callia provides two channels of studio-quality D/A conversion with the following premium features:

Analogue and conversion

- Balanced (XLR, +14dBu) and unbalanced (RCA, 2.00Vrms) output amplifiers
- Wide dynamic range (>115dB rms unweighted at -60dBFS)
- Low distortion (<-107dB rms unweighted at FS)
- Low maximum aharmonic spuriae (<-158dB rms unweighted at FS)
- Fully-balanced signal path minimises distortion artifacts and interference
- Low impedance, low distortion headphone amplifier, optimised for headphone impedance
- Prism Sound CleverClox hybrid PLL for ultimately low sampling jitter
- Analogue circuits galvanically isolated from PSU and digital circuits to eliminate interference
- High-precision digital interpolation filter
- Multi-bit delta-Sigma modulator
- Precise absolute / inter-channel gain matching, +/-0.05dB, includes line volume control tracking

Digital inputs

- USB input: class compliant asynchronous USB Audio Device Class 2.0 (UAC2) device; compatible with Windows and Mac computers as well as Linux-based PC, player, streamer and server devices
- S/PDIF inputs: optical (TOSLINK) and RCA formats
- Supported audio data formats:

Input	PCM sample rates	PCM max wordlength	DSD bit rate
USB (UAC2)	44.1,48,88.2,96,176.4,192,352.8,384 kHz	32 bits	DSD64, DSD128
Optical S/PDIF	44.1,48,88.2,96,176.4,192 kHz	24 bits	DSD64
RCA S/PDIF	44.1,48,88.2,96,176.4,192 kHz	24 bits	DSD64

Controls and indicators

- Automatic or manual source selection
- Source format and wordlength indication
- Line volume control, can be disabled for use with external pre-amplifiers (line outputs may be optionally muted by headphones)
- Dedicated headphone volume control
- Adjustable front panel illumination to match ambient lighting

PSU

- High reliability, low noise, universal input (90-260Vrms) mains PSU
- Low-power standby mode

2.2 Getting started

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

First, install the software onto your Windows PC or Mac if necessary:

A USB drive is shipped with your Callia, which contains software to be installed on your Windows PC or Mac. Note that no software installation is required for use with Linux-based PC, player, streamer and server devices.

If you intend to play audio from a Windows PC via USB, you MUST install this software since it contains a UAC2 driver to allow Windows to send audio to Callia. If you don't intend to play from a PC, but only want to play audio from a Mac via USB, or via one of Callia's S/PDIF inputs, you don't need to install any software and you can skip to the next section...

... however: we still recommend that you install the software on your Windows PC or Mac since this will allow you to install future firmware updates, but you can install the software later if you like.

- DO NOT connect the Callia unit to the computer via USB until AFTER software installation is complete.

- To install the software on a Windows PC, allow the USB drive to auto-run, or else double click on setup.exe in the root of the drive.

- To install the software on a Mac, double click on the .pkg file in the root of the USB drive.

- In either case, a series of dialogue boxes will guide you through the process.

Next, connect up your Callia:

- Connect an audio source: either an S/PDIF source via Callia's optical (TOSLINK) or RCA S/PDIF input, or else a Windows PC or Mac host computer (USB cable provided). If you're using a Windows PC, make sure you remembered to install the software in the previous section.

- Connect Callia's analogue outputs to a monitoring amplifier or active speakers, using the XLR or RCA analogue output connectors. Make sure that the volume of the amplifier or speakers is turned down until we're sure everything is OK. On the other hand, you could just plug in some headphones - make sure the headphone volume control is backed off to start with.

- Connect Callia to the mains using the power lead provided. Some front panel LEDs should light up. If you've connected a computer, Callia should show up as a WDM device (and some ASIO ports) on a Windows PC, or a Core Audio device on a Mac.

Now you should be able to play some audio:

- Press the standby/source button at the right hand end of Callia's front panel until the desired source is selected, with the 'AUTO' LED off.

- Start playback on the S/PDIF source, or by running a player application on PC or Mac.

- By advancing the line volume control (having turned up the volume control on the amplifier or active speakers, if you turned it down earlier) you should have audio. If you're using headphones, you may have to turn up the headphone volume control.

Once everything is working, register your Callia at http://www.prismsound.com/audiophileregister. You can familiarise yourself with the front panel controls by looking at the Front panel section. You might want to customise some aspects of Callia's operation by referring to the DIP switch functions section.

2.3 System requirements

For S/PDIF operation, Callia will work with any optical (TOSLINK) or RCA/phono PCM source at sample rates up to 192kHz.

For USB operation, Callia will work with any modern host PC or Mac with a suitable operating system and a USB 2.0 or 3.0 port, and with Linux-based PC, player, streamer and server devices.

Macs must be Intel platform and must be running OS X 10.5 Leopard or later; PCs must be running Windows Vista, 7 , 8, 10 or later (32-bit or 64-bit); Linux devices must incorporate the ALSA UAC2 driver. This is not to say that the computing power of the host is unimportant, but it is more a requirement of the audio applications than of Callia. If you need to play out audio files 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, play out 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.

Part 3

Installation procedures

3 Installation procedures

This section contains detailed installation instructions for your Callia. If you are keen to get going quickly, you could use the Getting started section above. The following sections describe installation for Windows and Mac systems. Note that Callia is 'plug and play' with Linux hosts, and no software installation is required.

3.1 Windows installation

To install the Callia software, your PC must be running Windows Vista, 7, 8, 10, or later (32-bit or 64-bit). You need a spare USB 2.0 or USB 3.0 port.

- DO NOT connect the Callia unit to the PC via USB until AFTER software installation is complete.

- If you are installing from the supplied USB drive:
  - Plug the USB drive into a spare USB socket on your PC
  - If the PC is set to 'auto-run', installation will begin automatically, otherwise double-click on the "setup.exe" icon in the root folder of the drive

- If you are installing from latest software version on www.prismsound.com:
  Download the latest software from the Support\Downloads page
  Double-click on the downloaded .exe file

- Follow the on-screen instructions. When software installation is complete, your Callia should be visible to Windows as a WDM audio device, and to suitable applications as an ASIO audio device.

- Connect your Callia to the mains supply and to a USB port on your PC with the cables provided

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

3.2 Mac installation

Callia will work with all Macs with Intel processors and OS X 10.5 Leopard, or later. You need a spare USB 2.0 or USB 3.0 port.

- Connect your Callia to the mains supply and to a USB port on your Mac with the cables provided.

- If you are installing from the supplied USB drive:
  - Plug the USB drive into a spare USB socket on your Mac
  - Double-click on the .pkg file in the root folder of the drive

- If you are installing from latest software version on www.prismsound.com:
  - Download the latest software from the Support\Downloads page
  - Double-click on the downloaded .pkg file

- A dialogue box will guide you through the installation process; after installation, your Callia should be visible in Audio/MIDI Setup as a Core Audio device.

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

NOTE: The Prism Sound Mac software has been digitally signed, which means it should install without additional intervention on most Macs. Under OS X 10.7.5 and onwards, if your Mac's Gatekeeper policy (Apple menu > System Preferences… > Security & Privacy > General tab under the header "Allow applications downloaded from:") has been set to "Mac App Store", you will need to temporarily change it to "Mac App Store and Identified Developers". For further information on Gatekeeper, see https://support.apple.com/en-gb/HT202491.

Part 4

Callia hardware

4 Callia hardware

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

4.1 Front panel

Callia's front panel contains a number of controls and indicators; from left to right:

- Source selection indicators to show which digital audio source is currently selected, and whether source selection is in automatic or manual mode;
- Format indicators to show the current audio format of the selected source;
- Line volume control which adjusts the volume of the analogue line outputs;
- Headphone jack, with its own volume control;
- Standby/source button which is used for manual source selection and to switch the unit in and out of standby mode.

Source selection

In automatic source selection mode (indicated by the 'Auto' LED being lit) sources are selected automatically according to sensing of valid audio at the inputs. In automatic mode, the current source remains selected until it is silent (or otherwise absent) for more than four seconds whilst another input is playing - in that case the other input is selected. The four second delay is to prevent spurious switching during silences between tracks.

Alternatively, in manual source selection mode (indicated by the 'Auto' LED being unlit) sources are manually selected by repeatedly pressing the standby/source button (on the right of the front panel), which causes the selection to rotate from left to right around the three sources. Note that presses must be short - holding the button in for more than one second will switch the unit into standby mode (see below). Actually, the source can be manually overridden in automatic mode too (but only from those sources which are present and playing) by pressing the standby/source button.

Toggling between automatic and manual mode is achieved by pressing the standby/source button in quick succession so as to bring the selection back to its original position.

Audio format indication

The sample rate of a PCM input is shown by a combination of the '44k1' and '48k' LEDs and the 'x2' and 'x4' LEDs. For example, a source sampled at 44.1kHz is indicated by the '44k1' LED alone, whereas a 96kHz source is indicated by the '48k' and 'x2' LEDs in combination, and a 352.8kHz source would cause the '44k1', 'x2' and 'x4' LEDs all to be lit. If a PCM source has 16 active bits (or fewer), the '24b' LED is unlit, whereas it is lit blue to indicate a wordlength between 16 and 24 bits, or pink to indicate a wordlength greater than 24 bits (the latter is only possible in USB source mode).

A DSD64 source (a 2.8224MHz bitstream) is indicated by the 'DSD' LED on its own, whereas a DSD128 source (a 5.6448MHz bitstream) is indicated by the 'DSD' and 'x2' LEDs together. Note that DSD sources are detected by the presence of DoP (DSD-over-PCM) flags, and that DSD128 is only possible in USB source mode. Callia can accept DSD sources which are at multiples of 48kHz as well as conventional DSD streams at multiples of 44.1kHz. These are indicated by the '48k' LED in combination with the 'DSD' (and possibly 'x2') LEDs.

Line volume control

The line volume control allows Callia to drive a power amplifier directly, without the need for an external volume control. The halo of LEDs around the knob shows the volume setting. Note that the line volume control can be disabled, as described in the DIP switch functions section, in case an external volume control is to be used.

Note that in this mode, the analogue line outputs are always driven at full volume, so care must be taken to ensure that damage does not occur to equipment or listeners.

Headphone jack and volume control

Callia contains a high-quality low-impedance headphone amplifier with a dedicated volume control independent of the line volume control. By default, the detection of a plug in the headphone jack causes the analogue line outputs to be muted. For details of how to optimise the headphone amplifier's output for different headphone impedances, as well as how to defeat this muting of the analogue line outputs, see the DIP switch functions section.

TO PREVENT POSSIBLE HEARING DAMAGE, DO NOT LISTEN AT HIGH VOLUME LEVELS FOR LONG PERIODS.

Standby mode

Holding in the standby/source button for one second causes Callia to enter standby mode. In this mode, the conversion and analogue functions are turned off to save power, although the USB functionality is retained, so the unit remains visible to a host computer. Normal operation is resumed by pressing the standby/source button. Standby mode is indicated by the lighting of the LED in the standby/source button.

Front panel LED brightness

The brightness of the front panel LEDs can be adjusted by turning the line volume control whilst holding in the standby/source button. Turn the control clockwise to increase the brightness and anticlockwise to decrease.

Retention of control settings

The line output volume, auto/manual source selection mode and LED brightness are remembered when the unit is in standby or unpowered. However, note that recent changes may be forgotten if power is removed from the unit without going into standby mode first.

4.2 Rear panel

Callia's rear panel contains all of the input and output connections, except for the headphone jack which is on the front panel.

From left to right (viewed from rear):

- 6A IEC inlet (regional power cord supplied): adjacent is the mains fuse holder.
- USB 2.0 device port for connection to a host computer (cable supplied).
- DIP switches: to customise some aspects of Callia's operation. See the DIP switch functions section below.
- RCA S/PDIF input: can accept PCM streams up to 192kHz sample rate with up to 24 bit wordlength, or DSD64 streams in DoP format. See the Inputs and outputs section below.
- Optical (TOSLINK) S/PDIF input: can accept PCM streams up to 192kHz sample rate with up to 24 bit wordlength, or DSD64 streams in DoP format. See the Inputs and outputs section below.
- XLR analogue line outputs: aligned for 0dBFS = +14dBu (3.88Vrms). See the Inputs and outputs section below.
- RCA analogue line outputs: aligned for 0dBFS = 2.0Vrms (+8.24dBu). See the Inputs and outputs section below.

4.2.1 DIP switch functions

A bank of four DIP switches are accessible on Callia's rear panel which can be used to customise operation of the the unit. From left to right, the functions of the switches are:

DIP switch 1 determines whether the line volume control is enabled:
When ON (default): Line volume control is enabled;
When OFF: Line volume control is defeated, for use with downstream volume control - **CAUTION: in this position the analogue line outputs are driven at full volume!**

DIP switch 2 sets the DSD headroom:
When ON (default): DSD headroom is +3.1 dB SA-CD so 'hot' DSD streams (permitted by Annex D of Scarlet Book 1.3 after March 2003) can be accommodated without clipping;
When OFF: DSD headroom is 0 dB SA-CD, so excessive DSD transients may be clipped, but noise floor is 3.1dB lower.

DIP switches 3, 4 optimise the headphone amplifier for different headphone impedances:
When ON,ON (default): Headphone amplifier is optimised for medium-impedance headphones (32-50Ω); line outputs are muted when headphones are plugged in;
When OFF,ON: Headphone amplifier is optimised for high-impedance headphones (>50Ω); line outputs are muted when headphones are plugged in;
When ON, OFF: Headphone amplifier is optimised for low-impedance headphones (<32Ω); line outputs are muted when headphones are plugged in;
When OFF, OFF: Headphone amplifier is optimised for medium-impedance headphones, and headphone-sense line output muting is DISABLED.

4.2.2 Fuses and ratings

TO PREVENT SHOCK HAZARD, THE CALLIA 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 Callia unit.

4.3 Inputs and outputs

Callia accepts digital audio inputs via USB as well as RCA and optical (TOSLINK) S/PDIF connections. Stereo analogue outputs are via RCA and XLR connections, as well as on a front-panel headphone jack.

In common with Prism Sound professional converters, the analogue parts are of Callia are galvanically isolated from the digital parts and the PSU in order to eliminate interference from the analogue line and headphone outputs.

USB input

When connected via USB, Callia appears as an asynchronous USB Audio Device Class 2.0 (UAC2) device, and supports PCM at standard sample rates between 44.1kHz and 384kHz, and wordlengths up to 32 bits. DSD, formatted at DSD-over-PCM (DoP) is supported at bit-rates of 2.8224MHz (DSD64) and 5.6448MHz (DSD128), plus 48kHz-multiple DSD64 and DSD128.

As a UAC2 device, Callia is supported natively by Mac hosts running OS X - so no driver software needs to be installed, but software installation under OS X is nonetheless recommended in order that future firmware update is supported (see Software and firmware updates). Windows does not natively support UAC2, so installation of the Prism Sound UAC2 Windows driver is essential. See the Installation procedures section for details.

As an 'asynchronous' UAC2 device, Callia operates from a high-stability low-jitter local clock reference when sourced from USB.

Note that the Prism Sound UAC2 Windows driver does not currently support simultaneous operation of multiple Callia (or other UAC2) units connected via USB to a single host PC.

S/PDIF inputs

Callia can also be connected to both optical (TOSLINK) and RCA S/PDIF sources, and supports PCM at standard sample rates between 44.1kHz and 192kHz, and wordlengths up to 24 bits. DSD, formatted as DSD-over-PCM (DoP) is supported at a bit-rates of 2.8224MHz (DSD64), plus 48kHz-multiple DSD64. Note that professional AES3 and AES3-id digital carrier formats will also work using an appropriate adapter at the RCA input.

In S/PDIF mode, Callia uses Prism Sound's acclaimed CleverClox hybrid PLL technology to lock its clock to the selected S/PDIF source with ultra-low jitter.

Note that no software installation is required for users who only intend to use Callia with its S/PDIF inputs (and not USB), although it will still be necessary to install the software (on either Mac or Windows PC) in order to be able to update the unit's firmware (see Software and firmware updates).

Analogue line outputs

Callia provides stereo analogue line outputs in both balanced (XLR) and unbalanced (RCA) formats. These are subject to the front-panel line volume control by default, but this can be disabled: see the DIP switch functions section.

The balanced outputs are precisely level-aligned at 0dBFS = +14dBu (3.88Vrms) within +/-0.05dB. They are transformerless, electronically bootstrapping, outputs which will maintain their correct output level and behavior even if connected to unbalanced inputs of following equipment. Industry-standard XLR pin-out is observed: 1 = signal-ground, 2 = hot (non-inverted), 3 = cold (inverted), so connection using standard pre-wired XLR cables is recommended.

The unbalanced outputs are precisely level-aligned at 0dBFS = +8.24dBu (2.00Vrms) within +/-0.05dB.

Both the balanced (XLR) and unbalanced (RCA) analogue line outputs may be used simultaneously if required.

Headphone jack

Callia's 6.3mm front-panel headphone jack is driven from a professional quality headphone amplifier, with low noise and distortion, and very low output impedance, giving unparalleled performance with any type of headphones. It has a dedicated volume control. By default, plugging headphones into the jack will mute the analogue line outputs, but this can be disabled: see the DIP switch functions section.

The performance of the headphone amplifier and the range of its volume control can be optimised according to the impedance of your headphones: this is also described in DIP switch functions section.

Part 5

Callia software and firmware

5 Callia software and firmware

Callia's software content comprises two separate parts. In the terminology of this manual, the 'firmware' is the part which resides, stored in flash memory, within the Callia unit itself and controls all of the analogue, digital, conversion and DSP functionality at a low level. The 'software' resides on any Windows PC or Mac which will stream audio to Callia.

Callia is a class-compliant USB Audio Device Class 2.0 (UAC2) device; as such, it will work with any UAC2-compliant host. Mac hosts are natively UAC2-compliant, and so no driver software needs to be installed under OS X for Callia to operate as a Core Audio device. Windows is not natively UAC2-compliant, so Callia is supplied with a driver (part of the software install) which provides ASIO and WDM audio connectivity under Windows. This driver is installed when the initial installation is performed. Obviously no PC or Mac driver is needed for Callia to operate from an S/PDIF input. Likewise, no driver is required for use with Linux hosts.

So, in summary, it is only necessary to install the software if you need to stream from a Windows PC. However, the software install also contains an application to allow Callia's firmware to be updated (flashed) from the PC or Mac in the case of new firmware becoming available. For that reason, it is recommended that you always install the software on a PC or Mac, even if you will only stream audio from a Mac or through the S/PDIF inputs. Note that no software application is currently available to allow firmware to be updated from a Linux host.

5.1 Software and firmware updates

From time to time, new versions of the Windows or Mac software, or the Callia firmware itself, may become available. To view the latest versions, please visit www.prismsound.com/callia.

Windows and Mac software updates can be installed by downloading the relevant files from the website and following the procedures in the Installation procedures section.

Callia's firmware is updated by downloading the relevant firmware update package from the website and unpacking it. If the Callia unit is re-powered whilst holding in the standby/source button, it enters 'bootloader' mode and prepares to receive new firmware from the PC or Mac. Running the firmware downloader application (which must be installed on the connected PC or Mac - see Installation procedures) causes a dialogue to pop up when the unit in bootloader mode is detected, inviting you to download the unpacked .firmware file. Full instructions for updating the firmware are included with the firmware download. Note that no software application is currently available to allow firmware to be updated from a Linux host.

Part 6

Technical topics

6 Technical topics

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

6.1 Stability and latency

Ever since audio found its way inside the computer, 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 mixing 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 to record, process and play audio, 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 has to operate quite large audio 'buffers' so that attending to non-audio tasks doesn't cause interruptions in the audio.

When a computer is playing back an audio file or stream, say via USB, small chunks of audio (normally just a few samples per channel) are loaded onto the USB every time the USB hardware alerts the operating system that it has finished sending the last chunk and now needs to send the next one. This is OK, because this 'low level' shovelling activity is done within the inner 'kernel' of the operating system and so is guaranteed to be able to happen without undue delay: its priority is guaranteed by the design of the hardware and operating system, and the drivers which run there. It can't be unexpectedly held off by high level 'application layer' activities like collecting emails, virus checking, moving data on and off disk etc.

But unfortunately the player application runs up in the application layer, and so IS prone to be held off for whenever the need to perform other tasks arises. So 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 away from its processing of the audio in order to deal with some other routine task. In non-optimized systems, many tasks which the user (listener!) would probably regard as non-urgent can interrupt audio processing. The player deals with this by filing up a buffer of audio samples which the low level process then empties down the USB. Without the use of sizeable sample buffers, any interruption taking longer than a few sample periods would cause gaps or jumps in the playback sample stream. Nearly every kind of application layer interruption is long enough to do this. However, with a large enough buffer, the interruptions don't cause audio to be disrupted.

Of course this is a much more serious problem in professional audio recording and production, or live sound mixing, say, than it is in domestic playback applications. Not only are the consequences of a disruption potentially greater, but also the time delay involved in the buffering can be unacceptable unless the buffers are very short, so it's a vicious circle. In general, computer playback systems should be easy to manage because we normally don't mind having a comparatively large buffer and hence a comparatively long delay. Having said this, computers used for professional audio work are usually well-optimised and aren't running a plethora of non-audio tasks, whereas the opposite is usually true of domestic computers used for audio playback as well as many other tasks.

In both Mac and Windows systems, it is usual for the playback buffer length to be adjustable within the player application. The Prism Sound Control Panel software supplied with our professional audio USB interfaces allowed adjustment of buffering within the Prism Sound USB Audio Device Class 2.0 driver for Windows in order to be able to adjust the system for minimum recording and playback latency. That feature is not available or necessary for use with Callia.

Callia should be able to play audio files and streams reliably even at high bit rates (such as DSD128 or 384kHz 32 bit PCM) simply by ensuring that adequate buffering is chosen within the player application. However, there is a possibility - particularly in older or badly-configured Windows systems - that glitches or interruptions may occur in playback of higher bit rates despite long player buffers, as a consequence of a problem known as 'DPC latency'. This is a result of driver contentions which can cause the low level audio shovelling to be inadequately serviced. There many discussions of how to diagnose and cure this problem on the web, including a troubleshooting guide among the Tech Notes in the support section at www.prismsound.com/optimizePCaudio.

6.2 Clocking and jitter

Good clock stability is probably the single most important issue separating good-quality audio converters 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 situations an analogue interface can't operate from its own internal clocks since it must be locked to an external reference sync. In the case of Callia, and 'asynchronous' USB interfaces in general, the problem is alleviated in the USB input case by the interface hardware generating a local clock to which the host computer has to synchronise. But when the unit has to lock to an external S/PDIF input the problem returns.

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.

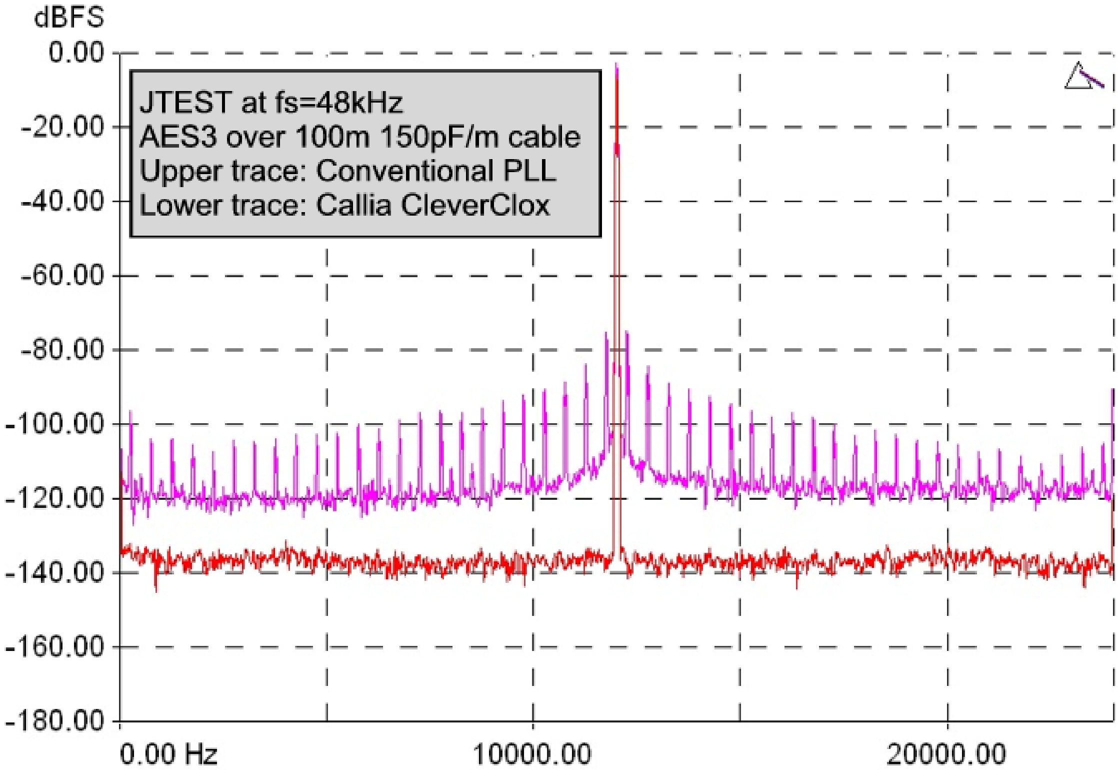
Prism Sound's unique CleverClox technology breaks these traditional constraints, allowing a low jitter clock to be re-created from any reference sync, no matter how much jitter it has and no matter what its frequency.

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 plot 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 Callia.



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.

Callia and CleverClox

Callia 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 S/PDIF 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.

Callia 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.

6.3 Analogue interconnections

To maintain the high sound quality of Callia, 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. 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 headphone jacks, where unreliable tip connection can occur owing to the slightly non-conforming shape of some manufacturers' parts.

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.

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 Callia, 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 during 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.

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.

6.4 Digital interconnections

It is understandable that little attention is usually paid to the quality of digital audio cabling. We are used to interconnecting our computer equipment with low-cost cables without mishap, and with digital audio it's rather logical to assume that no sound quality issues exist since we are simply moving digital data around. But the choice of digital audio cabling can be important, because the problems of transmitting digital audio data aren't really the same as for computer data at all.

It should be noted that the comments below apply to S/PDIF cabling only. There is no special cable quality requirement when passing audio over a USB cable, since in that case the audio IS just data, and Callia does not attempt to derive synchronisation or clocking information from the USB data stream. There is no particular risk of audio data degradation, since the audio data is passed in exactly the same way as any computer data over USB, and all USB cables must pass strict compliance testing in order to bear the USB logo.

Data integrity issues

In general, digital audio interfacing problems are usually (but not always) the result of inadequate interface bandwidth, which is most often due to the choice of cabling. In extreme cases this can result in loss of data (and resulting dropouts in the audio) because (unlike computer interconnection protocols) simple digital audio interfaces such as S/PDIF transmit the data only once, and without the possibility of error correction. Although there is a possibility that an error can be detected, this is of little use since no correction or retransmission is possible. So, unlike a computer interconnection, a mission-critical digital audio connection must ensure that no bit errors can occur in the data stream EVER! This can be hard to guarantee in the real world, especially when the system sample rate is high.

This was not really much of a problem when these interfaces were first standardised, since the bandwidth requirement was quite modest when the maximum sample rate was only 48kHz. Unfortunately, back then, the use of analogue audio cables for digital audio transmission was actively encouraged by the choice of XLR and RCA connectors for AES3 and S/PDIF respectively, even though they typically have poor bandwidth. But for AES3 and S/PDIF, the bandwidth requirement is directly proportional to the sample rate, since a fixed number of audio and status bits are transmitted per stereo sample.

Many S/PDIF-connected digital audio devices can operate at sample rates as high as 192kHz, and (sad to say) many digital audio cabling setups don't have the bandwidth to support this reliably. Actually, it's worse than that - much of the 192kHz-capable equipment has digital audio ports which (either admittedly or otherwise) don't support reliable operation at 192kHz whatever cable is used. This is particularly true of TOSLINK ports (the optical variant of S/PDIF).

Conversion quality issues

But surely the sound quality of a digital audio setup can't depend on the choice of digital audio cabling, so long as all the data bits get through? Sadly, and familiarly, though - it can. Because in many cases the audio data stream is used to pass the sampling clock as well as the audio data between equipment. If the receiving equipment gets a clock which has been degraded by a low-bandwidth interface, and if it uses this clock for A/D or D/A conversion, then the sound quality of that box will be degraded. This effect is known as 'sampling jitter'. Unfortunately the biphase coding scheme used in AES3 and S/PDIF is very effective at converting low cable bandwidth into clock jitter. It should be pointed out that this is an entirely avoidable problem, since any box which relies on deriving a jitter-free clock for A/D or D/A conversion (or for sample-rate conversion) can take steps to eliminate incoming jitter - but many don't. The Prism Sound CleverClox technology in Callia does exactly this, as explained in the Clocking and jitter section. This problem isn't really a cabling issue, but an equipment design issue. However, in most cases we can't change the design of poor-quality converters, but we can cover up their problems to some extent with good cabling!

Interference issues

A properly designed copper AES3 or S/PDIF interface will not cause audio-frequency ground continuity between the connected equipments, so hum loops should not occur. However, high-frequency ground continuity is essential if EMC legislation is to be met. This means that high-frequency interference such as from poor-quality switch-mode power supplies (see the Analogue interconnections section) can equally well be passed through copper digital audio interconnections. If this is a problem in your system, consider using a TOSLINK connection instead.

Maximising cable performance

In general, the best copper cable for digital audio is the cable with the lowest capacitance, since that will cause the least loss of bandwidth. For that reason, prefer cables specifically designed for digital audio, or for analogue video; don't use analogue audio cables - they don't have the bandwidth for digital audio use, especially at high sample rates. Prefer also the shortest cable, since (all other things being equal) loss of bandwidth is proportional to length.

Maximising cable bandwidth is important in optimising AES3 and S/PDIF data integrity at high sample rates such as 192kHz, and in optimising conversion quality in systems which include poor-quality converters. It is of little importance in protecting the data integrity of low sample rate systems, unless cable lengths are very long.

We are taught to choose cables of the correct impedance for the job. Whilst this doesn't have a direct impact on bandwidth, it can have a significant effect on data integrity at high sample rates and where cabling is short (and let's face it: at 192kHz cables had better be short...) because the reflections resulting from an impedance mismatch can affect the eye pattern at the receiver horribly. This can be much worse where non-matched connectors (such as XLRs) are present part way along the cable, such as in the case of 'breakout' cabled systems. For this reason, it may be better at high sample rates to use a continuous cable suitably terminated at each end rather than a 'breakout' arrangement.

In summary

- Use good-quality high-bandwidth cables - this means cables specifically designed for digital audio, or perhaps for analogue video - analogue audio cables are not suitable;
- Don't use cables that are longer than you need;
- At high AES3 or S/PDIF sample rates, consider eliminating 'breakout' connectors in the line by using a single length of high-bandwidth cable suitably terminated at each end;
- Consider using TOSLINK interconnections in systems where switch-mode interference is a problem, but remember that poor-quality TOSLINK cables can have very low bandwidth.

6.5 DSD (Direct Stream Digital) conversion

DSD was conceived in an age when monolithic sigma-delta converters were relatively new. The quantisers in the converter chips of the day were one-bit and ran at 64x the eventual PCM rate ('64fs'); their digital decimation and interpolation filters were real-estate-limited and so had very non-ideal behavior in terms of flatness and stop-band attenuation - as well as being confined to PCM rates below 50kHz. The low sample rate meant that brick-wall filters were needed, with their attendant time-domain dispersion problems.

Although the idea of transacting and storing pulse-density-modulation (PDM) signals (like DSD) directly, without down-conversion to PCM ('decimation') was not new, it had not been used for audio until the mid 1990s. Experiments were made by connecting the 64fs one-bit product of the front end of a sigma-delta ADC directly to the one-bit back end of a DAC, bypassing the decimation and interpolation filter stages. The result, unsurprisingly, sounded better than with the filters in circuit (no processing, be it analogue or digital, or even up-sampling, can make a signal more like the original - but the filters of the day were also very far from perfect). So it was that DSD and the SACD were born.

By the time the SACD format came to fruition, higher quality monolithic PCM converters had become available with longer wordlengths and higher PCM sample rates. These formats were specified for DVD Audio (DVD-A), and a format war ensued which killed them both (or perhaps it was the lack of enthusiasm amongst punters for ANY new higher-cost format whose main selling point was better sound quality than the CD, which most people found quite acceptable). Although SACD and DVD-A never caught on, DSD and high-resolution PCM audio are now enjoying a renaissance amongst quality-conscious listeners in download and streaming form.

So which is better? Although Prism Sound is a manufacturer of no-compromise ADCs for the music recording industry, it is not necessarily for us to say: whilst we may have an opinion on the best way to convert analogue audio to digital, and maybe even an opinion on the best format in which to store it, we must give our customers the means to produce whatever format they want: both! And as a manufacturer of no-compromise DACs we must strive to give them the best conversion to analogue from either format - after all, there are some excellent recordings in each format! In the case of Callia this involves converting incoming DSD streams to something with a longer wordlength and a lower sample rate prior to conversion. But surely that sacrifices the wide bandwidth which is the whole point in DSD? Let's look at the pros and cons:

Since the days when DSD was conceived, converter technology has moved on - dynamic performance and linearity have greatly improved, largely owing to the use of multi-bit front and back end circuits running much faster than 64fs - the limitations of one-bit quantisers at 64fs were surpassed not long after the initial DSD experiments. This presents us with an awkward choice in converting a DSD stream to analogue: do we present the DSD directly to a lower-quality 64fs one-bit back end, or do we decimate the signal to produce a longer wordlength which we can then re-process for a faster state-of-the-art multi-bit back end? We might prefer the latter.

All this is borne out by studying the datasheets of today's highest-performing audio DAC chips: they nearly all support both DSD and PCM inputs, so surely we will see that the dynamic performance and linearity is worse for DSD than for PCM? Actually, no! Most of them seem to have exactly the same performance for both! OK, then surely we can use them in DSD mode without losing all those DSD advantages like super-wide bandwidth and inherently linear one-bit quantisation? Sadly though, further study reveals that these devices are actually decimating the DSD to a longer wordlength and lower sample rate internally - that's why the performance is identical, since the decimation process can be done with minuscule degradation compared to that of the D/A conversion process itself. In many cases, the fact that the chip's digital volume control works in DSD mode gives the game away! So I'm afraid that there are few DACs which really do convert DSD to analogue without PCM in between, although many designers who adopt them will not realise it. Like many things lurking in the audio woodshed, this reality is anathema to the purist. But we should remember that nearly all DSD releases have been processed somewhat in the recording and mastering studio, even if it's just a tad of levelling or a smidgeon of EQ - and that NONE of this can be done without going via PCM or

(worse) back to analogue, so maybe we shouldn't be prissy about the DAC chip. Having said this, there are a few rigorously-produced minimalist DSD recordings which haven't been processed at all, but one wonders whether the ADC had a one-bit 64fs front end: if not, we're back to DSD via PCM, if so, performance was presumably limited.
Actually, there are a few DAC chips out there which DO allow the DSD stream to be presented directly to the back end DAC, without prior conversion to some form of PCM - but their datasheets tend not to quote the direct-mode performance, only the via-PCM figures - which are usually (you guessed it) the same as for PCM. The rest tend either to admit to worse performance in direct DSD mode, or else their performance may not be such that we care much either way.

So, having decided that we may prefer to sacrifice some bandwidth for linearity by going via PCM, what will be the price? Even a basic DSD64 signal, which is sampled at 2.8224MHz, could represent frequencies up to nearly the Nyquist frequency of 1.4112MHz, whereas the current state-of-the-art PCM signals are only sampled at 192kHz (with a bandwidth up to only about 90kHz), or perhaps 384kHz (with a bandwidth up to maybe 180kHz). We'd better hope that what we're sending to the back end DAC isn't sampled anything like that slowly!

In reality, though, there is no evidence that anybody can hear much above a few tens of kHz so perhaps the higher PCM rates might not sound too awful? Also, it turns out that we really don't want super-high frequencies entering our other items of analogue audio equipment - which are so rarely capable of either rejecting them thoroughly or transmitting them intact, so the consequence is often that they are 'demodulated into the baseband', which is audio-speak for making odd noises which weren't there in the first place. This problem was addressed in later versions of the SACD specification (the so-called Scarlet Book), which added the requirement for DSD DACs to incorporate a 50kHz analogue low-pass post-filter in order to protect subsequent equipment. So maybe we don't need to worry about limiting ourselves to higher PCM rates.

So, to sum up, it seems that since those days, reduction in silicon geometry and resulting increases in speed and complexity have meant that DACs' interpolation filters have become far closer to the ideal, and can work at much higher PCM sample rates, meaning that brick-wall responses are no longer necessary; whilst at the same time, faster multi-bit front and back ends have pushed back the limits of linearity and dynamic range - all of which are moving PCM quality ever onwards and upwards. For DSD, those same digital filters can convert to PCM on the way to the back end DAC, whilst perhaps providing the stipulated 50kHz low-pass filter at the same time - surely this is the ideal solution?

Actually, not quite. We think that it's nice to do the conversion process ourselves so that we aren't limited to the arbitrary response provided by a particular DAC device (which is often inscrutable for obvious reasons); thus we can achieve exactly the response we want. It also means that we can devote a suitable silicon budget to doing a really nice job of it. I might just add that we haven't adopted this approach lightly: Prism Sound and SADiE have been researching such processing and conversion technologies since the earliest days of DSD. Our Prism Sound ADA-8XR A/D D/A converter, which has long been the DSD conversion tool of choice in many of the world's top SACD mastering studios, uses exactly the same architecture as Callia.

I hope that this discussion has made it clear why Prism Sound approach DSD conversion in the way that we do. When all is said and done, the proof is hopefully in the listening. If you are minded to look further, you could Google Prof. Stan Lipshitz, and find out why he thinks that "1-bit Sigma-Delta Conversion is Unsuitable for High Quality Applications". Or you could search "DSD PCM myths" - and be sure to stop by Eelco Grimm's excellent analysis. I've just been scanning the "Direct Stream Digital" page on Wikipedia, which seems to address some of the same issues I've been discussing above, and is a good starting point for researching this interminable debate.

By the way, on the subject of the SACD specification: there was also at one point an amendment to exactly what constituted the maximum permissible DSD signal level. Early SACDs were prepared with a maximum peak signal level of 0dB SACD, whereas a later annex permitted maximum peak levels up to 3.1 dB SACD. So in designing a DSD DAC, we need to accommodate the higher level without clipping, but that means that for DSD recorded in observance of the lower maximum level we are needlessly squandering 3dB of our hard-won dynamic range. So we have provided a setting (see the Callia hardware section above) to allow you to choose how best to deploy the DAC's available dynamic range.

Part 7

About Prism Sound

7 About Prism Sound

Prism Sound was founded in 1987, the brainchild of two engineers: Graham Boswell and Ian Dennis. They had first met when working on the ground-breaking DSP program at world-renowned audio mixing desk manufacturer Rupert Neve & Company in Cambridge, England. In its early years, Prism Sound was a specialist Research & Development consultancy, designing digital audio equipment for a number of leading hi-fi and pro-audio manufacturers. Later, in the mid-1990s, Prism Sound began to manufacture its own range of no-compromise professional audio products for the recording and broadcast industries.

Prism Sound has worked behind the scenes of the international music industry, unseen, for nearly 30 years and has had considerable influence on recorded music. The world's top recording studios, independent recording producers and engineers are equipped with Prism Sound A/D-D/A conversion; further along the production chain, top mastering studios use Prism Sound converters and signal processors as their tools of choice to create the final masters for CD or download release. Now, younger music producers are learning why Prism Sound is the professional's choice, with the company's new range of desktop recording interfaces.

The giants of consumer electronics, top pro-audio manufacturers, hi-fi companies and many others use Prism Sound measurement technology to evaluate their products in the R&D lab and on the production line. Prism Sound has been particularly successful in providing audio measurement solutions for the automotive industry. Even if it wasn't recorded, produced or played back on Prism Sound equipment, the music you are listening to might well be playing on a device that has been tested using Prism Sound measurement technology.

Today, in addition to its comprehensive catalogue of professional products, Prism Sound still continues to provide R&D consultancy to industry clients, whilst investing heavily in new audio technologies and products. This continued investment ensures that the company will remain at the forefront of audio technology.

In Callia, for the first time, Prism Sound's acclaimed audio transparency has made the jump from the recording studio into the home.

You can read more about Prism Sound's rich audio heritage at http://www.prismsound.com/hifi/heritage.php.

Part 8

Specifications

8 Specifications

Front Panel

Input selection LEDs:	Indicate auto/manual select mode and current input: optical S/PDIF, RCA S/PDIF or USB
Sample rate / format LEDs:	Indicate sample rate: 44.1, 48, 88.2, 96, 176.4, 192, 352.8*, 384* kHz, DSD64 DSD128* (also indicates whether PCM word-length is > 16 bits or > 24 bits)
Line volume control:	Controls the volume at the analogue line outputs when enabled, halo indication
Headphone output:	6.3mm stereo jack socket, with illuminated volume control
Standby/source button:	Switches between operation and standby modes; also used for manual input selection
LED brightness:	Adjustable

Rear Panel

Analogue line outputs:	Two XLR male for balanced connection; two RCA sockets for unbalanced connection
S/PDIF inputs:	RCA and TOSLINK formats
DIP switches:	Line volume control enable/disable, DSD headroom select, headphone impedance select
USB port:	USB 2.0 B type device socket
Mains power:	3-pin 6A IEC inlet

Software Support

Mac OS support:	OS X 10.5 or later, Intel platform No driver required (Core Audio compliant UAC2 device)
Windows OS support:	Windows Vista, 7, 8.x, 10 or later (32 or 64 bit) ASIO and WDM drivers supplied
Linux OS support:	Distributions incorporating ALSA UAC2 driver No driver required for PC, player, streamer and server devices

Analogue Line Outputs

Configuration:	Fully-balanced analogue signal path to electronically-balanced XLRs, RCAs pseudo-balanced
Output amplitude:	XLR: 0dBFS=14dBu; RCA: 0dBFS=2Vrms
Output impedance:	50R
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 6Hz; -3dB at 1.5Hz
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, including DSD64, DSD128: -0.05dB at 32.0kHz, -3dB at 76.0kHz
Output balance:	XLR: >50dB
Inter-channel cross-talk:	1kHz: <-135dB; 20Hz..20kHz: <-120dB.
Inter-channel phase:	10Hz..5kHz: ±0.4º, 5kHz..20kHz: ±0.25º, 20kHz..50kHz: ±0.5º.

Headphone Output

Output amplitude:	0dBFS=18.0dBu (HiZ), 14.45dBu (MedZ), 8.45dBu (LoZ) at top of potentiometer
Output impedance:	4R
Total harmonic distortion:	-104dB (0.00045%, -0.1dBFS)
THD+n:	-103dB (0.00050%, -0.1dBFS)
Dynamic range:	113dB (-60dBFS)
Gain accuracy:	±0.05dB at top of potentiometer
LF roll-off:	-0.05dB at 6Hz; -3dB at 1.45Hz
HF roll-off:	fs=44.1kHz: -0.05dB at 20.3kHz; -3dB at 22.0kHz fs=48kHz: -0.05dB at 21.1kHz; -3dB at 23.9kHz fs=96kHz: -0.05dB at 27.0kHz, -3dB at 47.8kHz fs=>=192kHz, including DSD64, DSD128: -0.05dB at 27.0kHz, -3dB at 71.0kHz
Inter-channel cross-talk:	1kHz: <-120dB; 20Hz..20kHz: <-105dB.
Inter-channel phase:	10Hz..5kHz: ±0.1°, 5kHz..20kHz: ±0.25°, 20kHz..50kHz: ±0.5°.

USB Input

Interface protocol:	USB Audio Device Class 2.0 (UAC2)
Audio formats supported:	44k1, 48k, 88k2, 96k, 176k4, 192k, 352k8, 384k PCM at word-lengths up to 32 bits DSD64 and DSD128 in DoP frame, also 64x48kHz and 128x48kHz DSD variants * Note that 352kHz, 384kHz and DSD128 are via a precision digital decimation filter

S/PDIF Inputs

Interface protocols:	S/PDIF (RCA and TOSLINK); RCA also accepts AES3-ID
Audio formats supported:	44k1, 48k, 88k2, 96k, 176k4, 192k PCM at word-lengths up to 24 bits DSD64 in DoP frame, also 64x48kHz DSD variant
Channel Status:	Ignored

Synchronization

Clock recovery technology:	Prism Sound CleverClox hybrid PLL
Synchronization sources:	Local or S/PDIF input (automatic selection)
Local clock accuracy:	+/-50ppm
Jitter rejection:	60dB/decade above 100Hz

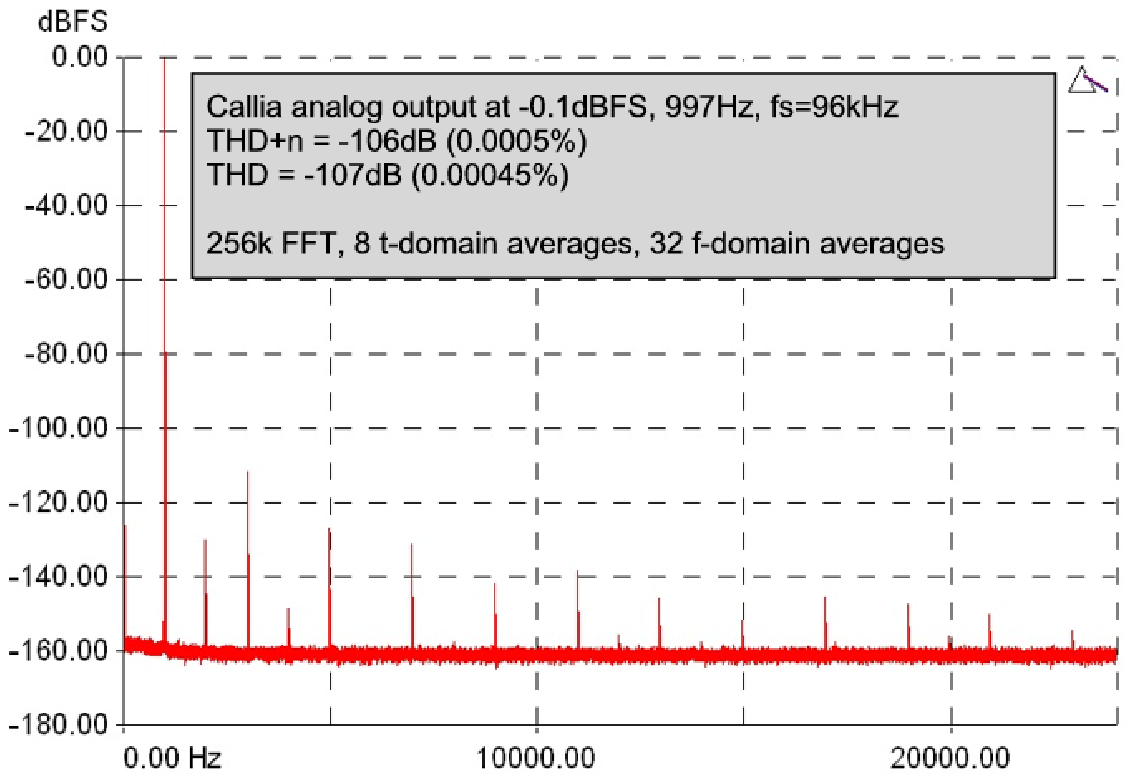
Physical

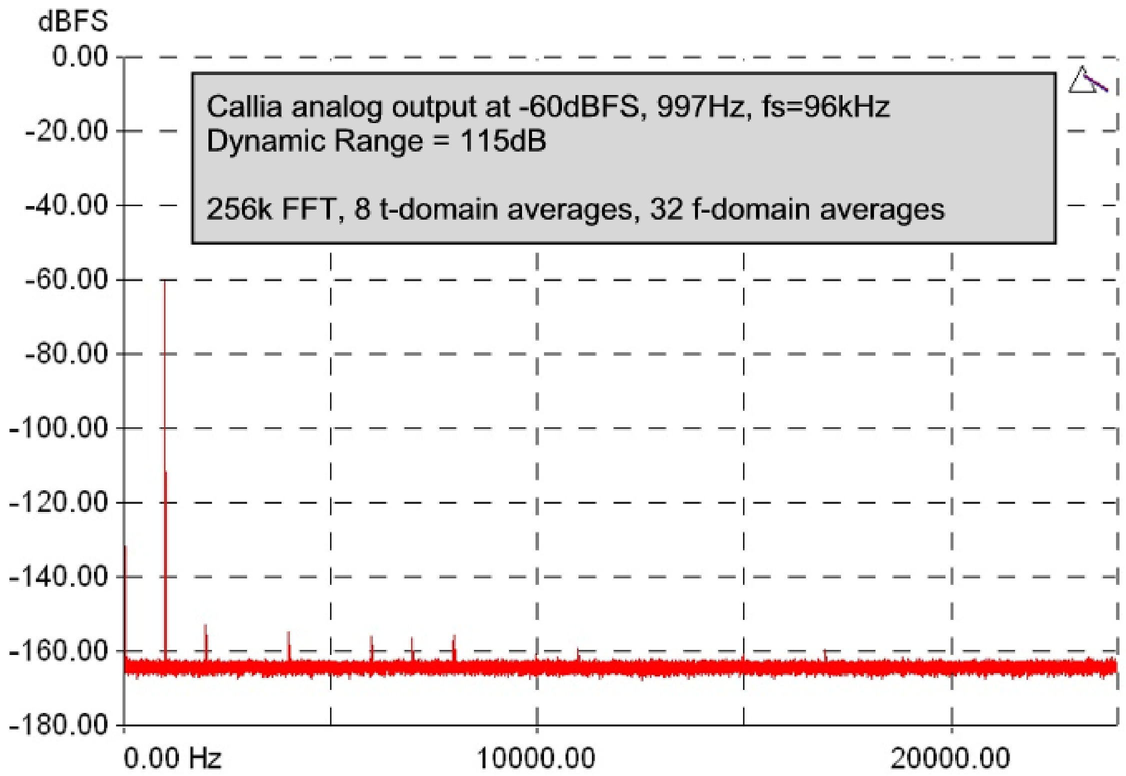
Dimensions:	Table-top (including feet): 285x242x50mm including feet
Weight:	2.1kg.
Mains voltage:	90VAC-250VAC, 50-60Hz
Power consumption:	15W
Fuse rating:	0.5A(T), 20mm, glass
Operating ambient:	0 to 35ºC, 85% maximum relative humidity

Supplied Accessories

USB flash drive:	Contains firmware loader software, Windows drivers, PDF manual etc.
Regional IEC mains lead:	x1
USB lead:	A-B x1

Performance Plots





Except where otherwise stated, audio performance data are typical, RMS, unweighted, 20Hz..20kHz figures, measured at 997Hz, using fs=96kHz at the XLR outputs.

In keeping with our policy of continual development, specifications are subject to amendment without notice. E&OE. All Trade Marks acknowledged.

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