



Redbox User Handbook No 1

Digital Audio Converters, Synchronisers, Delays & Silence Detectors

- | | |
|-----------------|---|
| RB-ADDA | Combined A/D and D/A unit (24 bit, 96kHz capable) |
| RB-ADDA2 | Combined A/D and D/A unit (24 bit, 192kHz capable) |
| RB-SC1 | Sample rate converter (24 bit, 96kHz capable) |
| RB-SC2 | Dual sample rate converter (24 bit, 192kHz capable) |
| RB-DAC1 | Digital to analogue converter |
| RB-SP1 | Digital AES/EBU splitter & combiner unit |
| RB-DS2 | Stereo audio delay synchroniser |
| RB-PD2 | Stereo profanity delay |
| RB-SD1 | Silence detection unit |
| RB-DSD1 | Digital silence detection unit |

SONIFEX



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Warranty

Warranty and Liability - important the purchaser is advised to read this clause

- (a) The Company agrees to repair or (at its discretion) replace Goods which are found to be defective (fair wear and tear excepted) and which are returned to the Company within 12 months of the date of despatch provided that each of the following are satisfied:
 - (i) Notification of any defect is given to the Company immediately upon its becoming apparent to the Purchaser;
 - (ii) The Goods have only been operated under normal operating conditions and have only been subject to normal use (and in particular the Goods must have been correctly connected and must not have been subject to high voltage or to ionising radiation and must not have been used contrary to the Company's technical recommendations);
 - (iii) The Goods are returned to the Company's premises at the Purchaser's expense;
 - (iv) Any Goods or parts of Goods replaced shall become the property of the Company;
 - (v) No work whatsoever (other than normal and proper maintenance) has been carried out to the Goods or any part of the Goods without the Company's prior written consent;
 - (vi) The defect has not arisen from a design made, furnished or specified by the Purchaser;
 - (vii) The Goods have been assembled or incorporated into other goods only in accordance with any instructions issued by the Company;
 - (viii) The defect has not arisen from a design modified by the Purchaser;
 - (ix) The defect has not arisen from an item manufactured by a person other than the Company. In respect of any item manufactured by a person other than the Company, the Purchaser shall only be entitled to the benefit of any warranty or guarantee provided by such manufacturer to the Company.
- (b) In respect of computer software supplied by the Company the Company does not warrant that the use of the software will be uninterrupted or error free.
- (c) The Company accepts liability:
 - (i) For death or personal injury to the extent that it results from the negligence of the Company, its employees (whilst in the course of their employment) or its agents (in the course of the agency);



WARRANTY

WARRANTY



WARRANTY

- (ii) For any breach by the Company of any statutory undertaking as to title, quiet possession and freedom from encumbrance.
- (d) Subject to conditions (a) and (c) from the time of despatch of the Goods from the Company's premises the Purchaser shall be responsible for any defect in the Goods or loss, damage, nuisance or interference whatsoever consequential economic or otherwise or wastage of material resulting from or caused by or to the Goods. In particular the Company shall not be liable for any loss of profits or other economic losses. The Company accordingly excludes all liability for the same.
- (e) At the request and expense of the Purchaser the Company will test the Goods to ascertain performance levels and provide a report of the results of that test. The report will be accurate at the time of the test, to the best of the belief and Knowledge of the Company, and the Company accepts no liability in respect of its accuracy beyond that set out in Condition (a).
- (f) Subject to Condition (e) no representation, condition, warranty or other term, express or implied (by statute or otherwise) is given by the Company that the Goods are of any particular quality or standard or will enable the Purchaser to attain any particular performance or result, or will be suitable for any particular purpose or use under specific conditions or will provide any particular capacity, notwithstanding that the requirement for such performance, result or capacity or that such particular purpose or conditions may have been known (or ought to have been known) to the Company, its employees or agents.
- (g)
 - (i) To the extent that the Company is held legally liable to the Purchaser for any single breach of contract, tort, representation or other act or default, the Company's liability for the same shall not exceed the Price of the Goods.
 - (ii) The restriction of liability in Condition (g)(i) shall not apply to any liability accepted by the Seller in Condition (c).
- (h) Where the Goods are sold under a consumer transaction (as defined by the Consumer Transactions (Restrictions on Statements) Order 1976) the statutory rights of the Purchaser are not affected by these Conditions of Sale.

Unpacking the Redbox

The Redbox is shipped with the following equipment. Please check your packaging to ensure that you have all of the items below. If anything is missing, please contact the supplier of your equipment immediately.

Item	Quantity
Redbox audio interface	1
IEC Mains lead fitted with moulded mains plug	1
Handbook and warranty card	1

Fig A: Packing List.



Each Redbox is shipped in protective packaging and should be inspected for damage before use. Where an item is found to have transit damage, notify the carrier immediately with all the relevant details of the shipment. Packing materials should be kept for inspection and also for if the product needs to be returned.

Warranty Registration

In order to register the date of purchase so that we can keep you informed of any design improvements or modifications, it is important to register online at: www.sonifex.co.uk/register or complete the warranty registration document that is enclosed and return it to Sonifex Ltd in the UK. For your own records you should write down the serial number (which can be found on the rear of the Redbox).

Serial Number
---------------	-------

Safety Information

Safety of Mains Operated Equipment

CE This equipment has been designed to meet the safety regulations currently advised in the country of purchase and it conforms to the safety regulations specified by use of the CE Mark.

Warning : There are no user serviceable parts inside the equipment. If you should ever need to look inside the unit, always disconnect the mains supply before removing the equipment covers.

Voltage Setting Checks

Ensure that the machine operating voltage is correct for your mains power supply by checking the box in which your Redbox was supplied. The voltage is shown on the box label. The available voltage settings are 115V, or 230V. Please note that all Redboxes are either switchable between 115V and 230V, or have a universal power supply.

Fuse Rating

The Redboxes are supplied with a single fuse in the live conducting path of the power infeed at the power supply. For reasons of safety it is important that the correct rating and type of fuse is used. Incorrectly rated fuses could present a possible fire hazard, under equipment fault conditions. See the Technical Specifications of each product for the fuse rating. The active fuse is fitted on the outside rear panel of the unit.



SAFETY & INSTALLATION INFORMATION

SAFETY & INSTALLATION INFORMATION



Power Cable and Connection

An IEC power connector is supplied with the Redbox, which has a moulded plug, attached – this is a legal requirement. If no moulded plug has been supplied with your Redbox, please contact your supplier, because an IEC connector is always supplied from the Sonifex factory. If for any reason, you need to use the Redbox with a different power cable, you should use the following wiring guidelines:

Wire Colour	Connection
Green, or green and yellow	Earth (E)
Blue, or Black	Neutral (N)
Brown, or Red	Live (L)

Fig B: Mains Cable Wire Colours

Ordering the Correct Mains Lead

When ordering a Redbox from Sonifex, it is helpful if you can specify your required operating voltage and mains lead. After the product code add:

UK, for 230V, UK 3 pin to IEC lead	
EC, for 230V, European Schuko 2 pin to IEC lead	
US, for 115V, 3 pin to IEC lead	
AU for 230V, Australasian 3 pin to IEC lead	

Fig C: Mains Lead Table.

E.g. order RB-SC1 UK for a UK IEC lead to be supplied.

Installation Information

Atmosphere

The units should be installed in an area that is not subject to excessive temperature variation (<0°C, >50°C), moisture, dust or vibration.

Electromagnetic Radiation

The cover is connected to earth by means of the fixing screws. It is essential to maintain this earth ground connection to ensure a safe operating environment and provide electromagnetic shielding.

Fitting Redboxes

Redboxes can be fixed to the underside of a mixing desk, or other surfaces using 4.2mm holes in the sides and fixed with 2 x M4 screws or 2 x No. 6 countersink wood screws.

They can also be rack-mounted, with either the front, or rear of the Redbox positioned at the front of the rack:

Front Mounting Redboxes: For rack mounting smaller (28cm) units the optional **RB-RK1** (Red) or **RB-RK1B** (Black) kit can be used (which include 4 off M6 panel fixing screws).



Fig D: RB-RK1 Small Redbox Front Rack-mount Kit.

Wherever you see this symbol an RB-RK1 front panel rack kit can be used :



Rear Mounting Redboxes: For rear panel mounting you can use either the RB-RK2, or RB-RK3, depending on the size of your Redbox.

RB-RK2 1U rear panel rack kit for small Redbox range, e.g., RB-BL2



Fig E: RB-RK2 Small Redbox Rear Rack-mount Kit.

e.g. for fitting an RB-BL2:



Wherever you see this symbol an RB-RK2 small Redbox rear panel 19" rack kit can be used :



RB-RK3 1U rear panel rack kit for large Redbox range, e.g., RB-DA6



Fig F: RB-RK3 Large Redbox Rear Rack-mount Kit.

e.g. for fitting an RB-DA6:



Wherever you see this symbol an RB-RK3 large Redbox rear panel 19" rack kit can be used :



Note: When fitting the rear-mounting rack-kits, a notch has been left on the inside of the right-hand rack-piece for the mains cable to pass through. Make sure that the mains cable has been put through the notch before attaching the right hand rack-piece.



SAFETY & INSTALLATION INFORMATION



Installing the Optional Video Sync Boards in the RB-DSD1

There are 2 optional video sync boards which can be used to synchronise the outputs of the RB-DSD1 to a 48kHz sample rate from an analogue or digital video signal:

RB-SYA - The Analogue video sync board will accept a composite signal of NTSC (525), PAL (625) & SECAM (625) signals covered by SMPTE-170-M (NTSC) and ITU-R BT.470-6 (PAL & SECAM).

RB-SYD - The Digital video sync board will accept 270Mbps SD-SDI and HD-SDI signals covered by SMPTE-259-M-C (SD) and SMPTE-292M (HD).

Opening the RB-DSD1

Warning: The power must be switched off at the supply or the power lead must be disconnected before attempting to open the unit. Removal of the cover can expose dangerous voltages.

1. Remove the 4 screws in the corners of the rear panel.
2. Remove the 4 screws on the top and bottom panels which hold the rear panel in place (2 on the top and 2 on the bottom).
3. Remove the screw in the centre of the front panel.
4. Slide the rear panel and main PCB backwards out of the metal chassis giving you internal access.
5. Remove the rubber grommet/bung on the rear panel which covers the hole for the video sync connector.
6. Remove the 2 screws from the bottom of the sync card pillars and, making sure to keep the plastic washers in place at the bottom of the pillars, fit the 20 way pin header into the 20 way connector on the RB-DSD1 motherboard.
7. Underneath the board, insert the 2 screws to fix the board in place.

To put the unit back together, slide the PCB back into the chassis and refit the screws in reverse order.

The video synchronisation is chosen using the front panel DIPswitches (see page 6).

Follow these instructions to fit either of the sync boards.

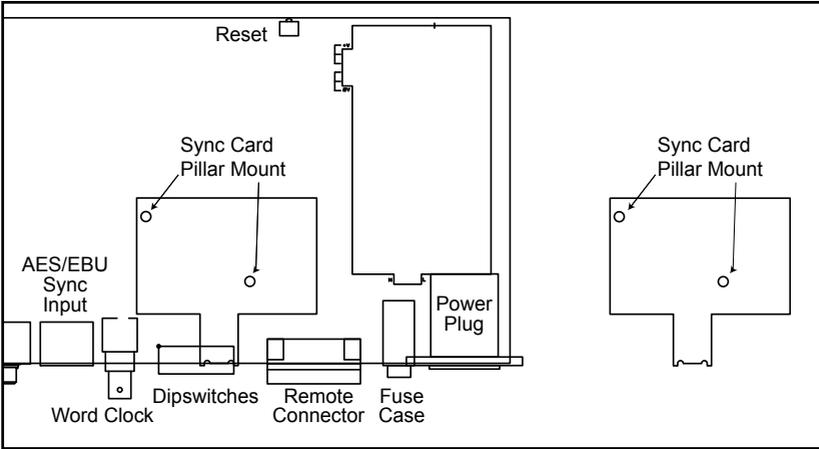


Fig G: RB-DSD1 Optional Video Sync Installation



SAFETY & INSTALLATION INFORMATION



WEEE & RoHS Directives - Sonifex Statement



The Waste Electrical and Electronic Equipment (WEEE) Directive was agreed on 13 February 2003, along with the related Directive 2002/95/EC on Restrictions of the use of certain Hazardous Substances in electrical and electronic equipment (RoHS).

The **Waste Electrical and Electronic Equipment Directive (WEEE)** aims to minimise the impacts of electrical and electronic equipment on the environment during their life times and when they become waste. It applies to a huge spectrum of products. It encourages and sets criteria for the collection, treatment, recycling and recovery of waste electrical and electronic equipment. All products manufactured by Sonifex Ltd have the WEEE directive label placed on the case. It gives a contact for individuals who are unsure about the correct procedure when the product has reached its “end of use”.

Sonifex Ltd will be happy to give you information about local organisations that can reprocess the products, or alternatively all products that have reached “end of use” can be returned to Sonifex and will be reprocessed correctly free of charge.

Sonifex Ltd has phased out the use of certain hazardous substances identified in the European Union’s **Restriction of Hazardous Substances (RoHS)** directive. The RoHS directive limits the use of certain hazardous substances currently used in EEE manufacture, including lead, mercury, cadmium, hexavalent chromium, and halide-containing compounds PBB (polybrominated biphenyl) and PBDE (polybrominated diphenyl ether). Elimination of these substances will result in more environmentally friendly recycling of electronic equipment. For the products which Sonifex manufacture, the main area where products were affected was in the use of lead for manufacturing and assembling electronics circuit boards.

Sonifex Ltd practices lead-free (LF) manufacturing processes. LF solder is used on the surface-mount PCB manufacturing processes and for hand soldering. The printed circuit boards (PCBs) used are either gold plated, or immersion tin plated, both of which use no lead. Historically the PCBs were hot air solder levelled (HASL) PCBs which used tin/lead based solder.

The manufacturing processes include the assembly of purchased components from various sources. Product is offered as RoHS compliant, or LF, only after sufficient evidence is received from the component manufacturers that their components are RoHS compliant. Sonifex Ltd relies solely on the distributor, or manufacturer, of the components for identification of RoHS compliance. Thus whilst every effort is made to ensure compliance, Sonifex Ltd makes no warranty, or certification, or declaration of compliance concerning said components.

Sonifex Ltd defines “Lead Free” as pertaining to any product, which has been manufactured by Sonifex Ltd using components which have been declared by the manufacturers as “Lead Free”. All statements by Sonifex Ltd of RoHS compliance are based on component manufacturer documentation.

1 RB-ADDA Combined A/D and D/A Converter

Introduction



Fig 1-1: RB-ADDA Front Panel

Using 24 bit, 96kHz capable devices, the RB-ADDA A/D and D/A Converter is a 1U rack-mount which produces an AES/EBU or S/PDIF level digital audio output from a balanced XLR or unbalanced phono stereo audio input. The unit also produces a stereo balanced XLR or unbalanced phono output from an incoming AES/EBU or S/PDIF digital input signal.

The unit operates in four modes:

**24^{RIT}
96^{KS}**

Master Mode - In this mode the unit receives an analogue audio signal, which is digitised and formatted for digital serial transmission (IEC958). The necessary clock signals are generated internally from an on board master clock at a selectable rate (32kHz, 44.1kHz, 48kHz, 64kHz, 88.1kHz or 96kHz).

Slave Mode - In this mode the unit automatically detects the presence of a digital audio sync signal, if present at the digital input, and synchronises the digital output to it. If no sync is present, no output will be generated.

Auto Mode - Here the unit synchronises to the digital audio sync signal if present at the digital input and uses the internal master clock only if no sync input signal is detected. In this case, the internal master clock is used at the selected sample rate.

Auto Lock Mode - This operates like the auto mode. The difference is that if the sync input signal is lost, the unit will revert to an internal master clock rate as near as possible to the last sync signal received.

When operating in sync modes, the front panel power LED flashes whenever the unit is not synchronised to the incoming digital signal, or when the unit is being calibrated. The RB-ADDA should be calibrated once it has been powered up for more than 10 minutes.

The analogue inputs have left and right level controls using pre-set potentiometers and DIP switches allowing a signal range from +9dBu to +27dBu. The analogue outputs have an output level control, allowing full-scale settings selectable from +12dBu, +18dBu or +24dBu. There are factory-set internal level controls for the analogue outputs allowing gain adjustment of ± 1 dB.

There are buttons to select either the AES/EBU or S/PDIF input or output for the D/A and A/D sections respectively. The output bit depth can be selected from 16, 20 or 24 bits. Inputs of a different bit depth to the output are dithered using a psychoacoustic noise filter.



For the digital output, there is a switch available to define the content of the channel status bits embedded within the digital audio stream. The channel status bits can be set to either Professional or Consumer Mode.

System Block Diagram

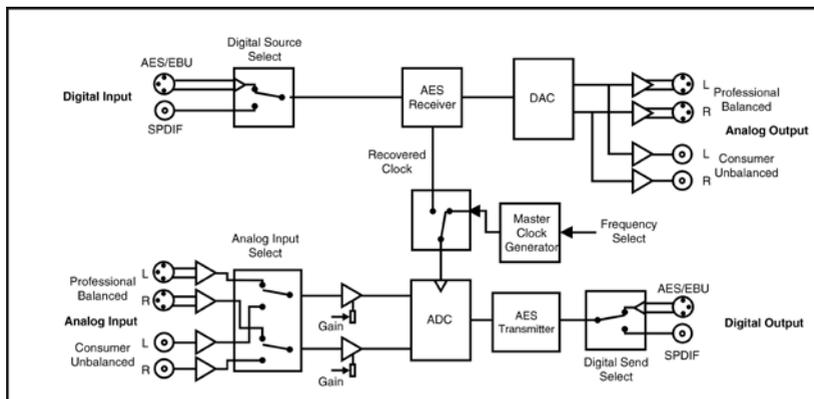


Fig 1-2: RB-ADDA System Block Diagram

Front Panel Indicators

The LED on the front panel is normally red to indicate that power is present on the unit. However, it also has a secondary role to indicate the status of the digital inputs

Fast flashing between red and amber – indicates a loss of digital input signal or that the unit is being calibrated.

Rear Panel Connections and Operation

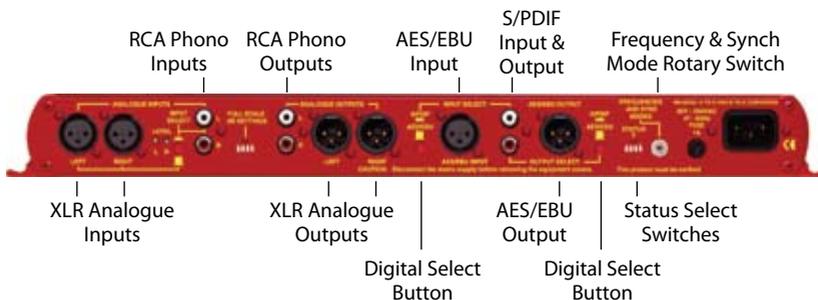


Fig 1-3: RB-ADDA Rear Panel



RB-ADDA Inputs

XLR Analogue Inputs (Left and Right)

The XLR 3 pin sockets used for the left and right channel inputs are electronically balanced and have an impedance of greater than 10k Ω bridging. Each XLR has the following connections:

Pin 1: Screen.

Pin 2: Phase.

Pin 3: Non-phase.

RCA Phono Inputs (Left and Right)

The two left and right RCA inputs are unbalanced and have an impedance of greater than 20k Ω .

Input Level Adjustment

The input gain can be individually adjusted for left and right channels by dipswitches and through pre-set potentiometers accessible on the rear panel. For full scale dB settings refer to Fig 1-4.

Individual preset pots give a further \pm 3dBu to give a total gain range of +9dBu to +27dBu for full-scale digits. The consumer input on the phono connector has a further 10dBu gain incorporated to give a total gain range of -1dBu to +17dBu for full-scale digits.

AES/EBU Inputs

The digital input XLR 3 pin socket has an impedance of 110 Ω . It has the following connections:

Pin 1: Screen

Pin 2: Phase

Pin 3: Non-phase

The signals on this connector should meet the IEC 60968 specification

S/PDIF Inputs

The S/PDIF digital phono inputs have an impedance of 75 Ω .

RB-ADDA Outputs

Analogue Outputs (Left and Right)

The XLR 3 pin output plug connectors are electronically balanced with an output impedance of less than 50 Ω . They have the following connections:

Pin 1: Screen.

Pin 2: Phase.

Pin 3: Non-phase.

RCA Phono Outputs (Left and Right)

These RCA (phono) outputs are unbalanced and have an output impedance of less than 75 Ω .



Output Level Adjustment

The output gain can be individually adjusted for left and right channels through the rear panel by dipswitches. Each output gain can be set for a signal of full-scale digits in the digital domain to give +12, +18 or +24dBu output on the XLR connectors (see Fig 1-4). The consumer output on the phono connector has a further 10dBu attenuation incorporated.

AES/EBU Output

The digital output XLR 3 pin socket has an impedance of 110Ω. It has the following connections:

Pin 1: Screen

Pin 2: Phase

Pin 3: Non-phase

The signals on this connector will comply with the IEC 60968 specification

S/PDIF Output

The digital output S/PDIF phono output has an impedance of 75Ω.

Rear Panel Controls

Full Scale dB Settings

The full-scale dB settings can be set for signals of +12, +18, +24 dBu to give full-scale digits in the digital domain.

Analogue (A/D) Input Full Scale Setting (dBu)	Switch 1	Switch 2
+12	OFF	OFF
+18	ON	OFF
+24	ON	ON

Analogue (D/A) Output Full Scale Setting (dBu)	Switch 3	Switch 4
+12	OFF	OFF
+18	ON	OFF
+24	ON	ON

Fig 1-4: RB-ADDA Full Scale dB Settings



Status Select Switches

These switches are used to determine the status or content of the digital signals. The type of information encoded in the channel status bits of a digital audio signal can be professional or consumer and is determined by switch 1.

If de-emphasis is selected (switch 2) the RB-ADDA will decode 50/15µs emphasis when indicated by certain channel status bits in the incoming digital audio data.

The sample size for the analog to digital conversion can be set to 24, 20 or 16 bits (switch 3 & 4). When the signal is truncated from 24 bits, a psycho-acoustic filter is applied to maintain optimum signal quality. These settings are summarised by the table in Fig 1-5, which is also shown on the top panel of the unit.

Status		
1	ON	Professional
1	OFF	Consumer
2	ON	De-emphasis On
2	OFF	De-emphasis Off

Bits			
	16	20	24
3	OFF	OFF	ON
4	OFF	ON	ON

Fig 1-5: RB-ADDA Status Select Switches

Digital Select Buttons

These buttons are used to switch the digital connection between the AES/EBU XLR connector (button out) and the S/PDIF phono connector (button in) for the digital input and the digital output.

Analogue Select Button

This button is used to switch the Analogue input between the balanced XLR connector (button out) and the unbalanced phono connector (button in).

Frequency and Sync Mode Rotary Switch

This rotary switch is used to select the Synchronisation Mode and to select the frequency of the digital output when using the on-board clock generator. There are 4 modes of operation: - Master Mode, Auto Sync Mode, Auto Lock Sync Mode & Slave Mode.

In **Master Sync Mode**, switch positions 0 – 5, the digital output sample rate is simply set by, and locked to, the internal on-board clock generator. No sync signal is used or required.

In **Auto Sync Mode**, switch positions 6– B, the digital output sample rate follows the digital input. When the digital input signal is not present the output sample rate will be set by, and locked to, the internal on-board clock generator at a frequency determined by the switch position.

In **Auto-Lock Sync Mode**, switch position C, No output will be generated until lock is achieved with a digital input signal. The digital output sample rate now follows the digital input. If the digital input signal is removed then the output sample rate will be set by, and locked to, the internal on-board clock generator at the closest frequency available to the previous digital input.

In **Slave Sync Mode**, switch position D, the digital output sample rate follows the digital input. When the digital input signal is not present the digital output is turned off.

The following table, also printed on the top of the unit, summarises the above settings and shows the sample rate generated by the internal clock generator in master and auto sync modes.

Frequencies and Sync Modes		
Rotary Switch Number	Mode	Sample Frequency (kHz)
0	Master	32
1	Master	44.1
2	Master	48
3	Master	64
4	Master	88.2
5	Master	96
6	Auto	32
7	Auto	44.1
8	Auto	48
9	Auto	64
A	Auto	88.2
B	Auto	96
C	Auto Lock	-
D	Slave	-

Fig 1-6: RB-ADDA Frequency and Sync Rotary Switch

Test/Calibration Mode

For optimum performance of the RB-ADDA, the unit should be calibrated when it has been powered up for approximately 10-15 minutes. The circuitry and chipsets contained in the unit will warm up during this time and the performance will deteriorate unless calibrated (the noise floor and dynamic range will be 1-2dB down on their best). The calibration cycle calibrates the gain and the zero reference of the A/D converter.

To calibrate the RB-ADDA, set the rotary FREQUENCIES AND SYNC MODES switch to position "F". The power LED on the front panel will flash quickly for 2 – 3 seconds and will illuminate fully when the unit is calibrated. Once calibration is complete, reset the rotary switch to the position that you require.





Technical Specifications RB-ADDA

A/D Connections

Analogue Inputs:	2 x XLR 3 pin (balanced) (L & R)
	2 x RCA phono (unbalanced) (L & R)
Digital Outputs:	1 x AES/EBU XLR 3 pin plug
	1 x S/PDIF RCA phono socket

A/D Audio Specification

Maximum Input Level:	+27dBu (balanced inputs)
Maximum Input Level:	+17dBu (unbalanced inputs)
Input Impedance:	>10kΩ bridging (balanced inputs)
Input Impedance:	>20kΩ (unbalanced inputs)
Input Levels:	Switchable +24dBu/+18dBu/+12dBu for FSD
Gain Range:	Adjustable 3dB loss to 3dB gain (L and R adjust)
Signal to Noise:	Better than -109dbFS (RMS A-weighted at 24bit)
Dynamic Range:	>110dB
Distortion and Noise:	>96dB THD + N at 1kHz

D/A Connections

Digital Inputs:	1 x AES/EBU XLR 3 pin female
	1 x S/PDIF RCA phono
Analogue Outputs:	2 x XLR 3 pin male (balanced) (L & R)
	2 x RCA phono (unbalanced) (L & R)

D/A Audio Specification

Max Output Level:	+24dBu (balanced outputs)
Max Output Level:	+14dBu (unbalanced outputs)
Output Impedance:	<50Ω (balanced outputs)
Output Impedance:	<75Ω (unbalanced outputs)
Dynamic Range:	>100dB
Gain Range:	Selectable 12dBu, 18dBu or 24dBu output level, ref FSD

Other Connections

Mains Input:	Filtered IEC, 110-120V, or 220-240V switchable, 10W max
Fuse Rating:	Anti-surge fuse 100mA 20 x 5mm (230VAC)
	Anti-surge fuse 250mA 20 x 5mm (115VAC)

Operational Controls

Analogue Input Select:	XLR or phono, via push-switch
Bit Depth:	16, 20 or 24 bits via DIP switch
Digital Output Select:	AES/EBU or S/PDIF, via push-switch
Modes & Frequencies:	16 way rotary DIP switch
Digital Input Select:	AES/EBU or S/PDIF, via push-switch
Channel Status Bits:	Set to consumer or professional mode via DIP switch

Equipment Type

RB-ADDA RK3	Combined A/D and D/A converter
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Physical Specifications

Dimensions (Raw)	48cm (W) x 10.8cm (D) x 4.2cm (H) (1U)
	19" (W) x 4.3" (D) x 1.7" (H) (1U)
Dimensions (Boxed)	58.5cm (W) x 22.5cm (D) x 7cm (H)
	23" (W) x 8.9" (D) x 2.8" (H)
Weight	Nett: 1.6kg Gross: 2.2kg Nett: 3.5lbs Gross: 4.8lbs

2 RB-ADDA2 Combined A/D and D/A Converter 24bit 192kHz

Introduction

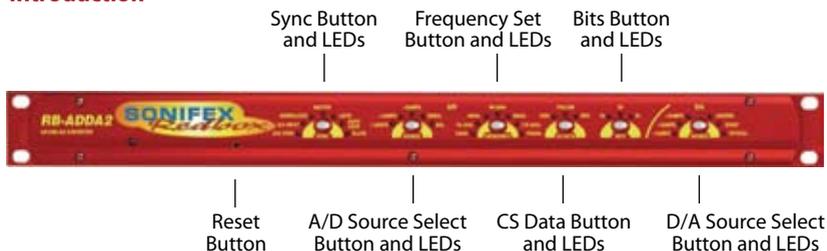


Fig 2-1: RB-ADDA2 Front Panel

24 BIT
192 kS/s

The RB-ADDA2 A/D and D/A converter is a 1U rack-mount which produces an AES/EBU, S/PDIF or TOSlink optical level digital audio output from a balanced XLR or unbalanced phono stereo audio input. It also produces a stereo balanced XLR or unbalanced phono output from an incoming AES/EBU, S/PDIF or TOSlink optical digital input signal.

The RB-ADDA2 is a high performance, enhanced version of the RB-ADDA providing the following additional features:

- It supports higher sample frequency rates up to and including 176.4kHz and 192kHz.
- It has additional independent AES/EBU and Word Clock synchronising inputs, so that the A/D and D/A sections can operate independently, with the digital outputs synchronised to an external master reference clock.
- It has TOSlink optical digital audio input and output.
- It has front panel push-button switches for all the main settings. The buttons are arranged in sets, where pressing the button advances the current selection and LED indicator.
- A serial RS232 port is included so that the RB-ADDA2 settings can be controlled remotely. The front panel LED indicators alter automatically when using RS232 commands.

The A/D SOURCE push-button is used to select from either the balanced or unbalanced stereo analogue inputs and this push-button also defines the input level for full scale digits at one of +12dBFS, +18dBFS or +24dBFS. These values can then be fine-tuned by using rear-panel pre-set potentiometers which give another ± 3 dB of gain adjustment, allowing a signal range from +9dBu to +27dBu. The RCA phono inputs have a further 10dB nominal gain incorporated to give a total signal range of -1dBu to +17dBu for full-scale digits.

For the digital output, there are three push-button switches to select the sample frequency, bit depth and status bit modes. The FREQUENCY button allows selection of the master sample frequency from one of 32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz.



The BITS button sets the output bit depth as one of 16, 20 or 24 bits, and the CS DATA button defines the content of the channel status bits embedded within the digital audio stream. The channel status can be forced to either Professional Mode (PRO), Consumer Mode (CON) or to follow the mode of the input (FOLLOW).

The SYNC button is used to select the synchronisation input, from Word Clock, AES/EBU or the D/A input, and also the synchronisation mode of the digital output. The A/D section of the RB-ADDA2 operates in four selectable modes:

Master Mode

In this mode the unit receives an analogue audio signal, which is digitised and formatted for digital serial transmission (IEC958). The necessary clock signals are generated internally from an on board master clock at a selectable rate (32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz).

Slave Mode

In this mode the unit is synchronised to an external source, using the digital audio sync or D/A input signal from which the clock signals are stripped, or to the TTL level Word Clock. The FREQUENCY LED will indicate the synchronised sample frequency and if no sync is present, no output will be generated.

Auto Mode

Here, the unit is synchronised to an external source, using the digital audio sync or D/A input signal from which the clock signals are stripped, or to the TTL level Word Clock. If no sync signal is present the unit runs from the onboard master clock at a rate selected by the front panel control (32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz).

Auto Lock Mode

This operates like the auto mode except that if no sync signal is present the unit will run at the closest master clock rate to the last locked incoming signal. The FREQUENCY LED will indicate the synchronised sample frequency.

When operating in sync modes, the SYNC button flashes whenever the unit is not synchronised to the incoming digital signal.

The D/A section has one SOURCE push-button which is used to select the digital input source from AES/EBU, S/PDIF or Toslink optical and which also sets the analogue output level to be generated for full scale digits, from either +12dBFS, +18dBFS or +24dBFS. The RCA phono outputs have a further 10dB nominal attenuation to reduce the signal to that expected by consumer equipment. There are factory-set internal level controls for fine tuning the analogue output gain adjustment. If no digital audio source is present, the D/A SOURCE button flashes.

The RB-ADDA2 automatically decodes 50/15µs emphasis if this is indicated by certain channel status bits in the incoming digital audio data. In both A/D and D/A sections, audio is sent to all of the outputs simultaneously. A red LED indicates when power to the RB-ADDA2 is on.

System Block Diagram

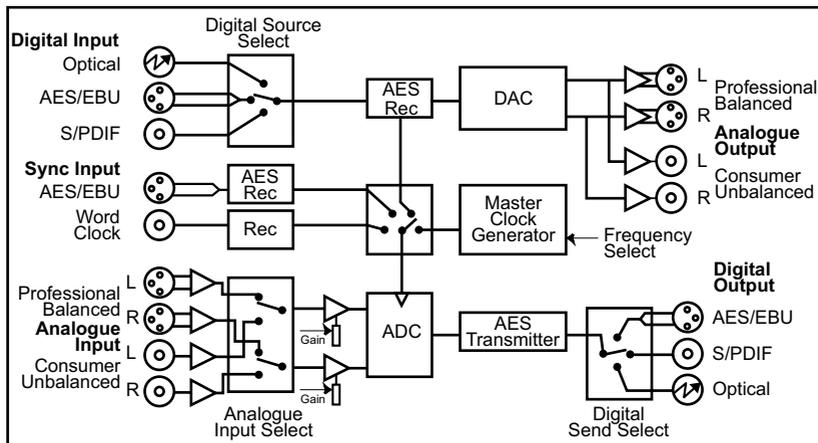


Fig 2-2: RB-ADDA2 System Block Diagram

Front Panel Controls & Indicators

The LED on the front panel is normally red to indicate that power is present on the unit.

SYNC Button & LEDs

This button will select one of the 4 operating modes for the RB-ADDA2 Analogue to Digital conversion. 3 modes use an external sync and this button is also used to select the external sync source that generates the sampling frequency used. The Mode button will flash when the external sync signal selected is not present.

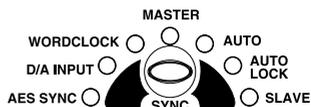


Fig 2-3: SYNC Button & LEDs

A/D SOURCE Select Button & LEDs

This button will select between the balanced and unbalanced analogue audio inputs for the RB-ADDA2 Analogue to Digital conversion. The signal level corresponding to full-scale digits can be set to +12, +18 or +24dBu.

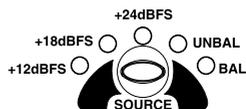


Fig 2-4: A/D SOURCE Select Button & LEDs

FREQUENCY Set Button & LEDs

When the button is not illuminated it will select the internal master frequency that is used in Master Mode and in Auto Mode when no sync signal is present. At other times, when the button is illuminated, the LEDs will show the actual sampling frequency of the incoming sync source. To view the Master Frequency press the button once to enter the setting mode. Any further button presses will advance the set frequency.

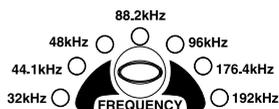


Fig 2-5: FREQUENCY Set Button & LEDs

CS DATA Button & LEDs

This button will select the type of channel status bits to be embedded in the digital audio output – either professional or consumer. The channel status type can be set directly or can follow the digital audio input source.

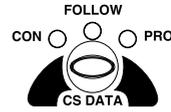


Fig 2-6: CS DATA Button & LEDs

BITS Button & LEDs

This button sets the bit depth of the RB-ADDA2 Analogue to Digital conversion to 16, 20 or 14 bits. The bit depth is reflected in the appropriate channel status bits in the digital output stream.

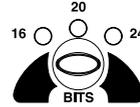


Fig 2-7: BITS Button & LEDs

D/A SOURCE Select Button & LEDs

This button will select the digital audio input source for the RB-ADDA2 Digital to Analogue conversion and will also set the level of the output signal corresponding to full-scale digits to +12, +18 or +24dBu. The D/A SOURCE select button will flash if the digital audio source selected is not present.

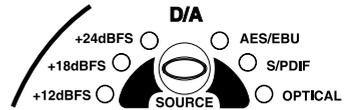


Fig 2-8: D/A SOURCE Select Button & LEDs

Reset Button

In the unlikely event that the RB-ADDA2 unit fails to respond, press the reset button to reboot the unit (see Fig 2-1 for location).

Rear Panel Connections and Operation

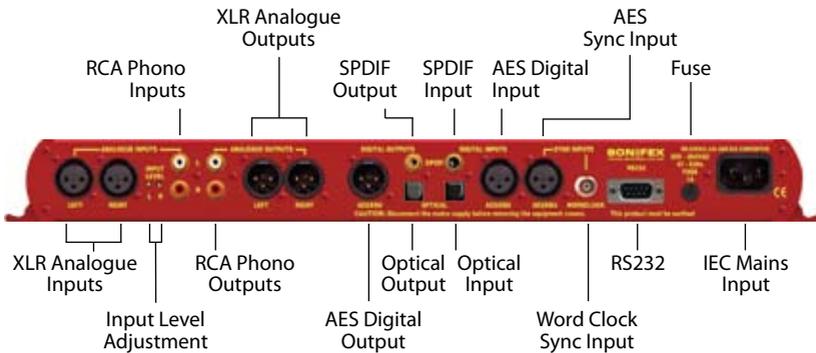


Fig 2-9: RB-ADDA2 Rear Panel





RB-ADDA2 Inputs

XLR Analogue Inputs (Left and Right)

The XLR 3 pin sockets used for the left and right channel inputs are electronically balanced and have an impedance of greater than 10k Ω bridging. Each XLR has the following connections:

- Pin 1: Screen.
- Pin 2: Phase.
- Pin 3: Non-phase.

RCA Phono Inputs (Left and Right)

The two left and right RCA inputs are unbalanced and have an impedance of greater than 20k Ω .

Input Level Adjustment

The input gain is set for both left and right channels by using the A/D SOURCE Select Button (Fig 2-4). Pre-set potentiometers, accessible on the rear panel, give a further \pm 3dBu to give a total signal range of +9dBu to +27dBu for full-scale digits.

The RCA Phono Inputs have a further 10dBu nominal gain incorporated to give a total signal range of -1dBu to +17dBu for full-scale digits.

AES/EBU Input

The digital input XLR 3 pin socket has an impedance of 110 Ω . It has the following connections:

- Pin 1: Screen
- Pin 2: Phase
- Pin 3: Non-phase

The signals on this connector should meet the IEC 60968 specification

S/PDIF Input

The S/PDIF digital phono input have an impedance of 75 Ω .

Optical Input

The digital audio optical input meets the TOSLink specification used by most professional & consumer equipment.

AES/EBU Sync Input

The digital input XLR 3 pin socket has an impedance of 110 Ω . It has the following connections:

- Pin 1: Screen
- Pin 2: Phase
- Pin 3: Non-phase

The signals on this connector should meet the IEC 60968 specification.

Word Clock Input

The Word Clock input is designed to receive a distributed clock running at the actual sampling frequency. The signal can be differential, or single ended, TTL level.

RB-ADDA2 Outputs

Analogue Outputs (Left and Right)

The XLR 3 pin output plug connectors are electronically balanced with an output impedance of less than 50Ω. They have the following connections:

Pin 1: Screen.

Pin 2: Phase.

Pin 3: Non-phase.

RCA Phono Outputs (Left and Right)

These RCA (phono) outputs are unbalanced and have an output impedance of less than 75Ω.

Output Level Adjustment

The output gain can be individually adjusted for left and right channels through the front panel by the D/A SOURCE Select button (Fig 2-8). Each output gain can be set for a signal of full-scale digits in the digital domain to give +12, +18 or +24dBu output on the XLR connectors. The RCA phono outputs have a further 10dBu nominal attenuation incorporated.

AES/EBU Output

The digital output XLR 3 pin socket has an impedance of 110Ω. It has the following connections:

Pin 1: Screen

Pin 2: Phase

Pin 3: Non-phase

The signals on this connector will comply with the IEC 60968 specification

S/PDIF Output

The digital output S/PDIF phono output has an impedance of 75Ω.

Optical Output

The digital audio optical output meets the TOSLink specification used by most professional & consumer equipment.





Serial Port Control

The Serial Port allows the RB-ADDA2 to be controlled and updated from a PC via a pin-to-pin serial cable, using the Sonifex Serial Control Interface (SCI) software. This software is available as a free download from the Sonifex website at www.sonifex.co.uk/sci.

Default Settings for the Serial Port

Baud Rate: 19200

Data Bits: 8

Stop Bits: 1

Parity: Even

Handshaking: XON/XOFF

Serial Interface Commands and Responses

Most of the commands follow the same structure: a 3 letter command followed by a colon, followed by a parameter (if any) and terminated by Carriage Return with optional Line Feed. A Line Feed character may be sent but it will be ignored by the RB-ADDA2. Commands are not case sensitive and all parameters are in hex.

Responses are CR & LF terminated.

After the RB-ADDA2 has been powered-up, an initialisation string is sent "Initialising ADDA2".

Following are the commands and the expected responses:

Command	Description	Response
ADB:nn	- Set ADC Bit Size nn=00 - 16 Bits nn=01 - 20 Bits nn=02 - 24 Bits	- ACK:
ADC:nn	- Set ADC Channel Status Type nn=00 - Send Consumer CS Data nn=01 - Follow Sync Input (or use last set) nn=02 - Send Professional CS Data	- ACK:
ADF:nn	- Set ADC Channel Status Type nn=00 - Select 32kHz Master Clock Sample Rate nn=01 - Select 44kHz Master Clock Sample Rate nn=02 - Select 48kHz Master Clock Sample Rate nn=03 - Select 88kHz Master Clock Sample Rate nn=04 - Select 96kHz Master Clock Sample Rate	- ACK:



Command	Description	Response
	nn=05 - Select 176kHz Master Clock Sample Rate	
	nn=06 - Select 192kHz Master Clock Sample Rate	
ADM:nn	- Set ADC Mode & Select Sync Source	- ACK:
	nn=00 - Master Mode	
	nn=04 - Auto Mode	
	nn=08 - Auto Lock Mode	
	nn=10 - Slave Mode	
For last three modes add		
	nn=01 - Select External AES Sync input as sync source	
	nn=02 - Select DA Stage input as sync source	
	nn=03 - Select Word Clock input as sync source	
ADS:nn	- Set ADC Source & Full Scale Input Level	- ACK:
	nn=00 - Select Balanced XLR Inputs	
	nn=04 - Select Unbalanced Phono Inputs	
And add		
	nn=01 - +12dBu represents Full Scale Digits (+2 for unbalanced)	
	nn=02 - +18dBu represents Full Scale Digits (+8 for unbalanced)	
	nn=03 - +24dBu represents Full Scale Digits (+14 for unbalanced)	
Bnn:	- Baud Rate	- ACK: (at old rate)
	nn=11 (115200)	
	nn=57 (57600)	
	nn=38 (38400)	
	nn=19 (19200)	
	nn=96 (9600)	
DAS:nn	- Set DAC Source & Full Scale Output Level	- ACK:
	nn=04 - Select AES Digital Audio Input	
	nn=08 - Select SPDIF Digital Audio Input	
	nn=10 - Select Optical Digital Audio Input	
And add-		
	nn=01 - Full Scale Digits Sends +12dBu Out (+2 for unbalanced)	
	nn=02 - Full Scale Digits Sends +18dBu Out (+8 for unbalanced)	
	nn=03 - Full Scale Digits Sends +24dBu Out (+14 for unbalanced)	
DWN:	- Download new code	-OK: then erasing message
	Then send S-records followed by S-record terminator to force programming of flash memory and reboot	
FPS:	- Request Front Panel Settings	- FPS:aa_bb_cc_dd_ee_ff
	Where aa = value from ADC Mode as ADM: above	
	bb = value from ADC Source as ADS: above	

2

DIGITAL AUDIO CONVERTERS



DIGITAL AUDIO CONVERTERS - RB-ADDA2

Command	Description	Response
	cc = value from ADC Frequency as ADF: above dd = value from ADC Bits as ADB: above ee = value from ADC CS Data Select as ADC: above ff = value from DAC Source as DAS: above	
LOC:	- Lock Front Panel	- ACK:
MEM:	- Memory Dump Where add is memory address starting at 060(hex) and data is 16 bytes of data, each byte shown as 2 hex digits	- Memory, then 10 lines Madd:data
MFn:	- Limit Front Panel Frequency Selection Where n=0 for No Limit n=1 for 48kHz Limit & n=2 for 96kHz Limit	- ACK:
REG:	- Register Dump - Registers, then 4 lines of 8 x Rrr:dd Where rr is register number and dd is value shown as 2 hex digits	
SER:	- Serial Number request	- SER:012345
SRQ:	- Status Request Where aa = ADC Status dd = DAC Status ss = Internal Status	- STA:aa_dd_ss
STK:	- Stack Dump Where aaaa= current Stack Pointer, add is memory address starting at 0800(hex) and data is 16 bytes of data, each byte shown as 2 hex digits	- STK: aaaa, then 6 lines Sadd:data
UNL:	- Unlock Front Panel	- ACK:
UID:	- Unit ID Request	- UID:ADDA2
VER:	- Firmware Version Request	- VER:1.23 or BOOT:1.23

Error messages

The following error messages can be returned for illegal commands

Err:01	- return if Command Not Found
Err:02	- return if Missing Parameter
Err:04	- return if Parameter out of range



Upgrading Firmware

Occasionally, it may be necessary to upgrade the firmware on the RB-ADDA2 to add new functionality and fix software bugs. New firmware updates will be made available from time to time on the Sonifex website. Visit www.sonifex.co.uk for details.

The firmware is upgraded using the Serial Control Interface - SCI. This PC software downloads the new firmware and initiates the upgrade process. To upgrade the firmware, select the required firmware file in the SCI upgrade firmware section. The RB-ADDA2 will automatically reboot and enter the bootstrap mode prior to the transfer of the firmware file. The SCI software downloads the firmware to internal memory, and then erases the current firmware before programming the update. When the firmware is complete, the unit reboots with the new firmware.

Please note: Firmware files can take several minutes to transfer to the RB-ADDA2 at lower baud rates. To speed up the process, select a higher baud rate prior to transferring the new firmware.

Technical Specifications RB-ADDA2

A/D Connections

Analogue Inputs:	2 x XLR 3 pin (balanced) (L & R) 2 x RCA phono (unbalanced) (L & R)
Digital Outputs:	1 x AES/EBU XLR 3 pin plug 1 x S/PDIF RCA phono socket 1 x TOSLink optical output
Sync Inputs:	1 x AES/EBU XLR 3 pin female 1 x Word Clock BNC

A/D Audio Specification

Maximum Input Level:	+27dBu (balanced inputs)
Maximum Input Level:	+17dBu (unbalanced inputs)
Input Impedance:	>10kΩ bridging (balanced inputs)
Input Impedance:	>20kΩ (unbalanced inputs)
Input Levels:	Switchable +24dBu/+18dBu/+12dBu for FSD (on bal inputs)
Gain Range:	Adjustable 3dB loss to 3dB gain (L and R adjust)
Signal to Noise:	Better than -113dbFS (RMS A-weighted at 24bit)*
Dynamic Range:	Better than -110dB*
Distortion and Noise:	Better than -100dB THD + N at 1kHz*
Cross-talk:	Better than -112dB (20Hz to 20kHz)*

* Measured using balanced inputs



A/D Operational Controls

Analogue Input Source:	Balanced XLRs or unbalanced phonos, via A/D SOURCE push-button
Analogue Input Level for FSD:	+12dBFS, +18dBFS or +24dBFS, via A/D SOURCE push-button
Analogue Input Level:	+9dBu to +27dBu via rear-panel
Adjust:	+3dB via pre-set pots
Sample Frequency Rates:	32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz, via FREQUENCY push-button
Bit Depth:	16, 20 or 24 bits, via BITS push-button
Channel Status Bits:	Consumer mode, professional mode or follow input, via CS DATA push-button
Sync Input Select:	AES/EBU, Word Clock or D/A input, via SYNC push-button
Sync Mode Select:	Master, slave, auto, auto lock, via SYNC push-button

D/A Connections

Digital Inputs:	1 x AES/EBU XLR 3 pin female 1 x S/PDIF RCA phono 1 x TOSLink optical input
Analogue Outputs:	2 x XLR 3 pin male (balanced) (L & R) 2 x RCA phono (unbalanced) (L & R)

D/A Audio Specification

Max Output Level:	+24dBu (balanced outputs)
Max Output Level:	+14dBu (unbalanced outputs)
Output Impedance:	<50Ω (balanced outputs)
Output Impedance:	<75Ω (unbalanced outputs)
Gain Range:	Selectable 12dBu, 18dBu or 24dBu output level, ref FSD (on balanced inputs)
Signal to Noise:	Better than -106dB (RMS A-weighted at 24 bit)*
Dynamic Range:	Better than -100dB*
Distortion and Noise:	Better than -85dB THD +N at 1kHz*
Cross-talk:	Better than -112dB (20Hz to 20kHz)*

* Measured at balanced outputs



Other Connections

Mains Input:	Universal filtered IEC, continuously rated 85-264VAC @47-63Hz, max 10W
Fuse Rating:	Anti-surge fuse 1A 20 x 5mm
Serial Port:	RS232 9 Pin D-type socket

D/A Operational Controls

Digital Input Select:	AES/EBU, S/PDIF or TOSlink optical, via D/A SOURCE push-button
Analogue Output Level for FSD:	Selectable +12dBu, +18dBu or +24dBu output level, ref FSD, via D/A SOURCE push-button

Equipment Type

RB-ADDA2 Combined A/D and D/A converter, 24 bit 192kHz



Physical Specifications

Dimensions (Raw)	48cm (W) x 15.8cm (D*) x 4.2cm (H) (1U) 19" (W) x 6.2" (D*) x 1.7" (H) (1U)
Dimensions (Boxed)	59cm (W) x 27.4cm (D*) x 10.8cm (H) 23.2" (W) x 10.8" (D*) x 4.3" (H)
Weight	Nett: 1.6kg Gross: 2.3kg Nett: 3.5lbs Gross: 5lbs

* Note that this product is deeper than standard Redboxes.

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DIGITAL AUDIO CONVERTERS



3 RB-SC1 Sample Rate Converter

Introduction



Fig 3-1: RB-SC1 Front Panel

The RB-SC1 Sample Rate Converter standardises the sample rate of a digital audio signal to one of 32kHz, 44.1kHz, 48kHz or 96kHz, or to a synchronising input, selectable from AES/EBU, S/PDIF or TTL Word Clock.

If synchronising to an external signal there are several modes causing different actions in case of loss of the synchronising signal.

There are also switches available to define the content of the channel status bits embedded within the digital audio stream.

System Block Diagram

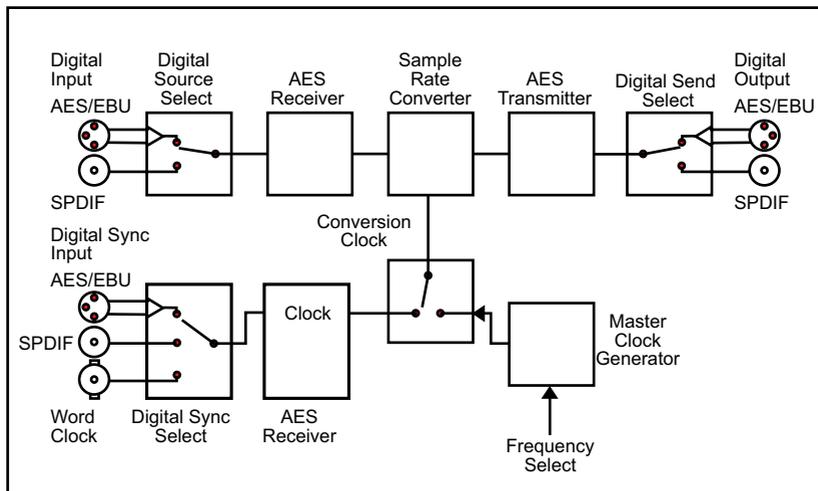


Fig 3-2: RB-SC1 System Block Diagram



Front Panel Indicators

Front Panel LED

The LED on the front panel is normally red to indicate that power is present on the unit. However, it also has a secondary role to indicate the status of the digital inputs

Fast flashing between red and amber – indicates a loss of digital input signal.

Slow flashing between red and amber - when not in master mode this indicates the absence of a synchronising input.

Rear Panel Connections and Operation

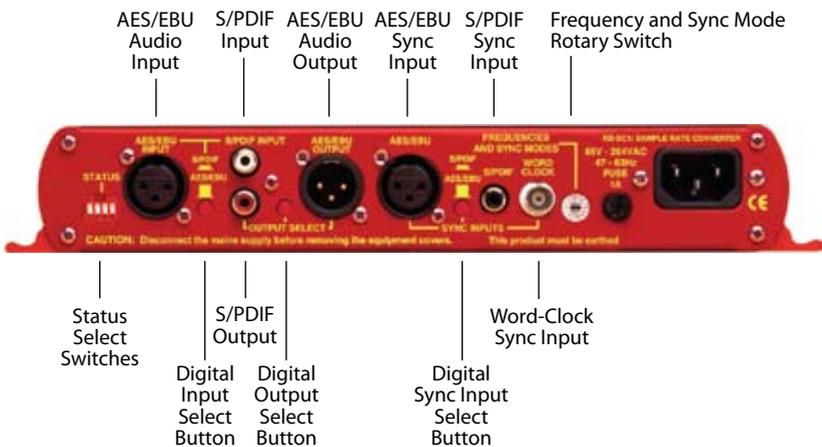


Fig 3-3: RB-SC1 Rear Panel

Inputs and Outputs

AES/EBU Inputs

The digital source and digital sync XLR 3 pin sockets both have an impedance of 110 ohms. They have the following connections:

- Pin 1: Screen
- Pin 2: Phase
- Pin 3: Non-phase

The signals on these connectors should meet the IEC 60968 specification

S/PDIF Inputs

The digital source and digital sync S/PDIF phono inputs both have an impedance of 75Ω.

Word Clock Input

The BNC TTL word clock input has an impedance of 50Ω.

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DIGITAL AUDIO CONVERTERS

AES/EBU Output

The digital output XLR 3 pin socket has an impedance of 110 ohms. It has the following connections:

Pin 1: Screen

Pin 2: Phase

Pin 3: Non-phase

The signals on this connector will comply with the IEC 60968 specification

S/PDIF Output

The digital output S/PDIF phono output has an impedance of 75 ohms.

Rear Panel Controls

Status Select Switches

These switches are used to determine the content of the channel status bits embedded within the digital audio stream (switches 1 and 2) and to select the source for the digital sync signal from either digital audio input or TTL word clock (switch 3).

The channel status bits will be forced to Professional Mode for the highest 3 sample rates as they are not supported by consumer mode. For the lowest three rate these status bits can be either set to follow the input signal type (switch 1 off) or can be forced to either professional or consumer mode (switch 1 on and switch 2 either off or on). These settings are summarised in Fig 3-4 and are also on top of the unit.

Status Select Switch Settings		
1	On	Force channel status type to...
1	Off	Follow input
2	On	Professional output
2	Off	Consumer output
3	On	Digital sync
3	Off	Word-clock sync



Fig 3-4: RB-SC1 Status Switches

Digital Input Select Buttons

These buttons are used to switch the digital connection between the AES/EBU XLR connector (button out) and the S/PDIF phono connector (button in) for the digital source, the digital sync input and the digital output

Frequency and Sync Mode Rotary Switch

This rotary switch is used to select the synchronisation mode and to select the frequency of the digital output when using the on-board clock generator. There are 4 modes of operation :- Master mode, Auto Sync Mode, Auto Lock Sync Mode & Slave Mode.

In Master sync mode, switch positions 0 – 5, the digital output sample rate is simply set by, and locked to, the internal on-board clock generator. No sync signal is used or required.

In Auto sync mode, switch positions 6– B, the digital output sample rate follows the sync input. When the sync signal is not present the output sample rate will be set by, and locked to, the internal on-board clock generator at a frequency determined by the switch position.

In Auto-Lock sync mode, switch position C, no output will be generated until lock is achieved with a sync signal. The digital output sample rate now follows the sync input. If the sync signal is removed then the output sample rate will be set by, and locked to, the internal on-board clock generator at the closest frequency available to the previous sync input.

In Slave sync mode, switch position D, the digital output sample rate follows the sync input. When the sync signal is not present the digital output is turned off.

Fig 3-5, also printed on the top of the unit, summarises the rotary switch settings and shows the sample rate generated by the internal clock generator in master and auto sync modes.

Frequencies and Sync Modes		
Rotary Switch Number	Mode	Sample Frequency (kHz)
0	Master	32
1	Master	44.1
2	Master	48
3	Master	96
4	Master	96
5	Master	96
6	Auto	32
7	Auto	44.1
8	Auto	48
9	Auto	96
A	Auto	96
B	Auto	96
C	Auto Lock	-
D	Slave	-

Fig 3-5: RB-SC1 Frequency and Sync Rotary Switch Settings



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DIGITAL AUDIO CONVERTERS



DIGITAL AUDIO CONVERTERS - RB-SC1

Technical Specifications RB-SC1

Audio Specification

Dynamic Range:	120dB
Distortion & Noise:	-114dB THD + N at 1kHz, ref 0dB FS
Sample Freq Range:	30kHz – 100kHz
Bit Depth:	Up to and including 24 bits.

Connections and Controls

Audio Inputs:	1 x AES/EBU XLR 3 pin female 1 x S/PDIF RCA phono female (Input button select between AES/EBU and S/PDIF)
Sync Inputs:	1 x AES/EBU XLR 3 pin female 1 x S/PDIF RCA phono female 1 x TTL BNC female (Input button select between AES/EBU and S/PDIF, and DIP switch select between TTL and either of the other two)
Outputs:	1 x AES/EBU XLR 3 pin male 1 x S/PDIF RCA phono female (Output button select between AES/EBU and S/PDIF);
Mains Input:	Filtered IEC, continuously rated 85-264VAC @47-63Hz, max 10W
Fuse Rating:	Anti-surge fuse 100mA 20 x 5mm (230VAC) Anti-surge fuse 250mA 20 x 5mm (115VAC)
Operational Modes:	Master mode, auto sync mode, auto lock mode and slave mode, set via rotary switch
Status bits:	Forced to consumer mode, professional mode, or set to follow input

Equipment Type

RB-SC1	Sample rate converter
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Physical Specifications

Dimensions (Raw)	28cm (W) x 10.8cm (D) x 4.2cm (H) (1U) 11" (W) x 4.3" (D) x 1.7" (H) (1U)
Dimensions (Boxed)	36cm (W) x 20.5cm (D) x 6cm (H) 14.2" (W) x 8" (D) x 2.4" (H)
Weight	Nett: 1.0kg Gross: 1.4kg Nett: 2.2lbs Gross: 3.1lbs

RB-SC2 Dual Sample Rate Converter, 24bit 192kHz

Introduction

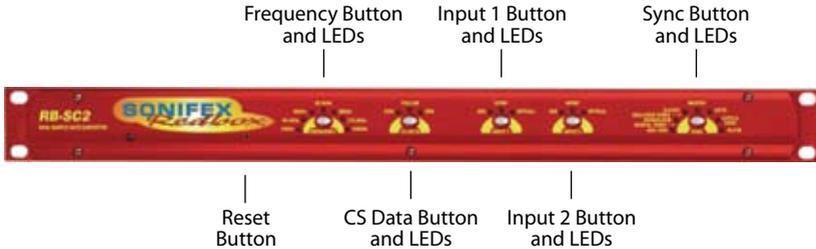


Fig 1: RB-SC2 Front Panel

24^{BIT}
192^{KS/s}

The RB-SC2 sample rate converter is a 1U rack-mount which produces AES/EBU, S/PDIF and TOSlink optical level digital audio outputs from a balanced AES/EBU, S/PDIF and TOSlink optical level digital audio inputs. The sample rate of the outputs can be set by an internal clock or from various external synchronizer sources.

The RB-SC2 is a high performance, enhanced version of the RB-SC1 providing the following additional features:

- It supports higher sample frequency rates up to and including 176.4kHz and 192kHz.
- It has 2 independent sample rate converter circuits that use a common clock source to set the output sample rate.
- It has 2 optional video synchronising boards. These set the output sample rate to 48kHz that is synchronised to either an analogue video signal or SDI digital video signal (HD or SD).
- A special X-Lock mode allows the unit to function as a full bi-directional sample rate converter.
- It has TOSlink optical digital audio inputs and outputs.
- It has front panel push-button switches for all the main settings. The buttons are arranged in sets, where pressing the button advances the current selection and LED indicator.
- A serial RS232 port is included so that the RB-SC2 settings can be controlled remotely. The front panel LED indicators alter automatically when using RS232 commands.

For the digital outputs, there are three push-button switches to select the sample frequency (FREQUENCY), channel status bit type (CSDATA), and sync source and mode of operation (SYNC).

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The FREQUENCY button allows selection of the master sample frequency from one of 32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz. The CS DATA button defines the content of the channel status bits embedded within the digital audio stream, and can be forced to either Professional Mode (PRO), Consumer Mode (CON) or to follow the mode of the input (FOLLOW).

The SYNC button is used to select the synchronisation input, from the AES/EBU sync input, the Wordclock input or, for X-Lock, the other digital input. The X-Lock synchronisation allows the unit to act as a bi-directional sample rate converter with the output of sample rate converter 1 syncing the input of sample rate converter 2 and vice versa so that they follow each other. The application for the X-Lock mode is so that the RB-SC2 can be inserted between 2 digital devices which run at different sample rates, such as a PC recorder and a digital player. Using the RB-SC2 in X-Lock mode ensures that the 2 devices remain synchronised at all times regardless of the sample rate of the 2 devices.

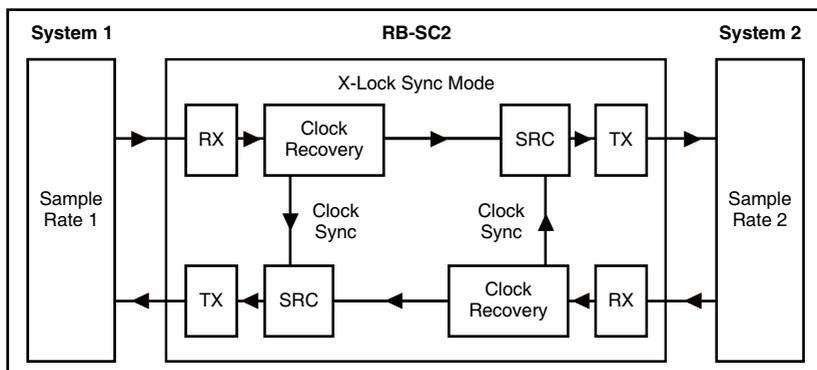


Fig 2: RB-SC2 X-Lock Sync Mode

The SYNC button will also select the operating mode of the unit as described below. If an optional video sync board is fitted then 2 sync LEDs light together to show the active video sync.

Master Mode

In this mode the unit receives a digital audio signal, which is passed to the sample rate converter and then re-formatted for the digital serial transmitter (IEC958). The sample rate converter clock signal is generated internally from an on board master clock at a selectable rate (32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz).

Slave Mode

In this mode the unit is synchronised to an external source, using the digital audio sync, or to the TTL level Word Clock. The FREQUENCY LED will indicate the synchronised sample frequency and if no sync is present, no output will be generated.

Auto Mode

Here, the unit is synchronised to an external source, using the digital audio sync, or to the TTL level Word Clock. If no sync signal is present the unit runs from the onboard master clock at a rate selected by the front panel control (32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz).

Auto Lock Mode

This operates like the auto mode except that if no sync signal is present the unit will run at the closest master clock rate to the last locked incoming signal. The FREQUENCY LED will indicate the synchronised sample frequency.

When operating in sync modes, the SYNC button flashes whenever the unit is not synchronised to the incoming sync signal.

There are 2 further push-button switches (INPUT1 & INPUT2) that are used to select the input connector used for each of the 2 sample rate converter circuits. These switches select between AES/EBU, S/PDIF and TosLink optical connectors.

A red LED indicates when power to the RB-SC2 is on.

Installing the Optional Video Sync Boards

There are 2 optional video sync boards which can be used to synchronise the outputs of the RB-SC2 to a 48kHz sample rate:

RB-SYA - The Analogue video sync board will accept a composite signal of NTSC (525), PAL (625) & SECAM (625) signals covered by SMPTE-170-M (NTSC) and ITU-R BT.470-6 (PAL & SECAM).

RB-SYD - The Digital video sync board will accept 270Mbps SD-SDI and HD-SDI signals covered by SMPTE-259-M-C (SD) and SMPTE-292M (HD).

Opening the RB-SC2

Warning : The power must be switched off at the supply or the power lead must be disconnected before attempting to open the unit. Removal of the cover can expose dangerous voltages.

1. Remove the 4 screws in the corners of the rear panel.
2. Remove the 4 screws on the top and bottom panels which hold the rear panel in place (2 on the top and 2 on the bottom).
3. Remove the screw on the front panel underneath the CS DATA button.
4. Slide the rear panel and main PCB backwards out of the metal chassis giving you internal access.
5. Remove the rubber grommet/bung on the rear panel which covers the hole for the video sync connector.
6. Remove the 2 screws from the bottom of the sync card pillars and, making sure to keep the plastic washers in place at the bottom of the pillars, fit the 20 way pin header into the 20 way connector on the RB-SC2 motherboard.
7. Underneath the board, insert the 2 screws to fix the board in place.



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DIGITAL AUDIO CONVERTERS - RB-SC2

To put the unit back together, slide the PCB back into the chassis and refit the screws in reverse order.

The RB-SC2 will auto-sense the video cards on powering up and the relevant option will be available using the SYNC button (see page 5).

Follow these instructions to fit either of the sync boards.

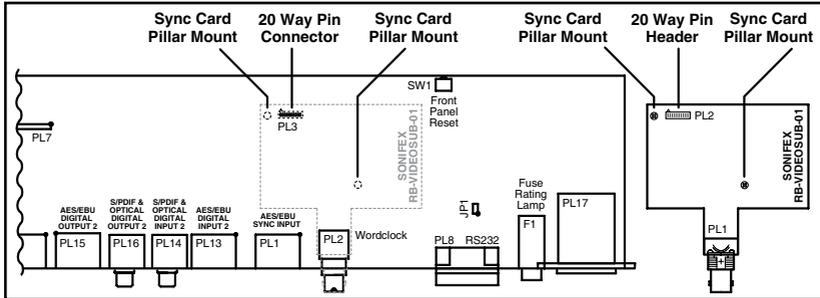


Fig 3: RB-SC2 Optional Video Sync Installation

System Block Diagram

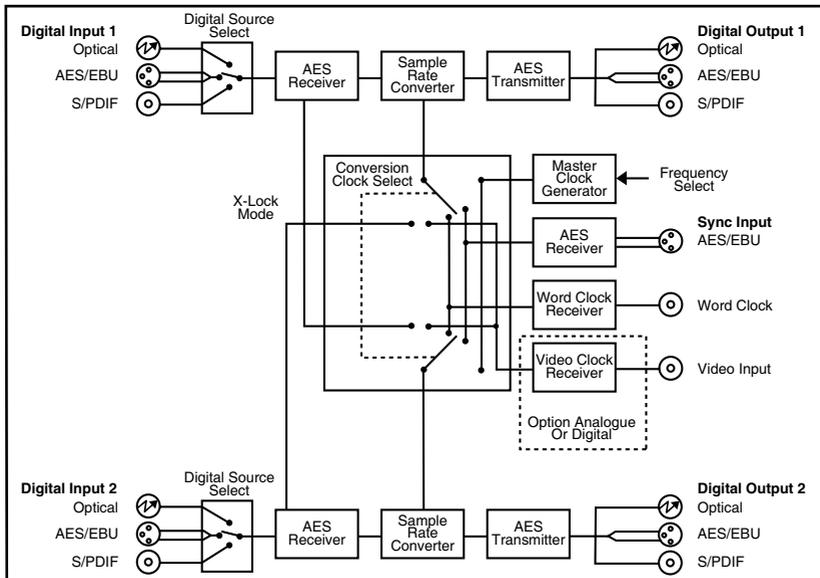


Fig 4: RB-SC2 System Block Diagram

Front Panel Controls & Indicators

The LED on the front panel is normally red to indicate that power is present on the unit.

FREQUENCY Set Button & LEDs

When the button is not illuminated it will select the internal master frequency that is used in Master Mode and in Auto Mode when no sync signal is present. At other times, when the button is illuminated, the LEDs will show the actual sampling frequency of the incoming sync source. To view the Master Frequency press the button once to enter the setting mode. Any further button presses will advance the set frequency.

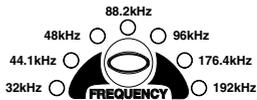


Fig 5: FREQUENCY Set Button & LEDs

CS DATA Button & LEDs

This button will select the type of channel status bits to be embedded in the digital audio output – either professional or consumer. The channel status type can be set directly or can follow the digital audio input source.



Fig 6: CS DATA Button & LEDs

INPUT1 & INPUT2 Source Select Button & LEDs

This button will select between the balanced AES/EBU, unbalanced S/PDIF and optical digital audio inputs for the RB-SC2 sample rate conversion.

The input buttons will flash when no suitable source is available on the selected connector.

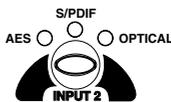


Fig 7: INPUT 1 & INPUT2 Source Select Button & LEDs

SYNC Button & LEDs

This button will select one of the 4 operating modes for the RB-SC2 sample rate conversion (master, auto, auto & lock or slave). 3 modes use an external sync source that generates the sampling frequency used, from one of AES/EBU, wordclock, x-lock or optionally analogue video or digital video (if sync card is fitted). The Sync button will flash when the external sync signal selected is not present.

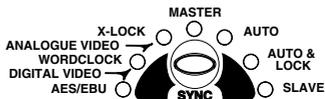


Fig 8: SYNC Button & LEDs

Reset Button

In the unlikely event that the RB-SC2 unit fails to respond, press the reset button to reboot the unit (see Fig 1 for location).



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DIGITAL AUDIO CONVERTERS



Rear Panel Connections and Operation

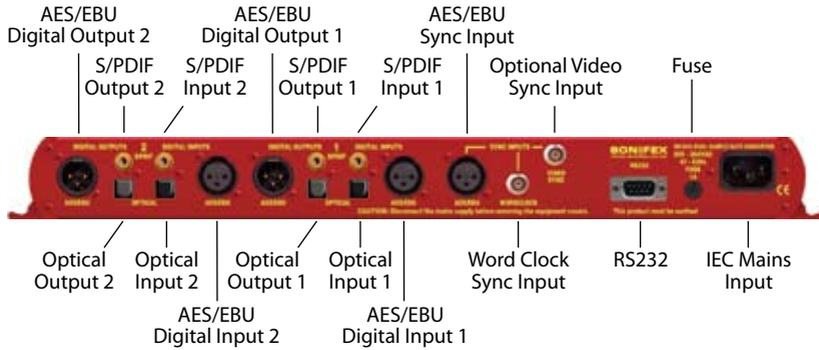


Fig 9: RB-SC2 Rear Panel

RB-SC2 Inputs

AES/EBU Inputs

The digital input XLR 3 pin socket has an impedance of 110Ω. It has the following connections:

- Pin 1: Screen
- Pin 2: Phase
- Pin 3: Non-phase

The signals on this connector should meet the IEC 60968 specification

S/PDIF Inputs

The S/PDIF digital phono input have an impedance of 75Ω.

Optical Inputs

The digital audio optical input meets the TOSLink specification used by most professional & consumer equipment.

AES/EBU Sync Input

The digital input XLR 3 pin socket has an impedance of 110Ω. It has the following connections:

- Pin 1: Screen
- Pin 2: Phase
- Pin 3: Non-phase

The signals on this connector should meet the IEC 60968 specification

Word Clock Input

The S/PDIF digital phono input have an impedance of 75Ω.



RB-SC2 Outputs

AES/EBU Outputs

The digital output XLR 3 pin socket has an impedance of 110Ω. It has the following connections:

- Pin 1: Screen.
- Pin 2: Phase.
- Pin 3: Non-phase.

The signals on this connector will comply with the IEC 60968 specification

S/PDIF Outputs

The digital output S/PDIF phono output has an impedance of 75Ω.

Optical Outputs

The digital audio optical output meets the TOSLink specification used by most professional & consumer equipment.

The outputs will have a unweighted dynamic range of at least 138dB and a THD+N noise of or better than -137dB. The data at the outputs will be presented as 24 bit wide.

Serial Port Control

The Serial Port allows the RB-SC2 to be controlled and updated from a PC via a pin-to-pin serial cable, using the Sonifex Serial Control Interface (SCI) software. This software is available as a free download from the Sonifex website at www.sonifex.co.uk/sci.

Default Settings for the Serial Port	
Baud Rate:	19200
Data Bits:	8
Stop Bits:	1
Parity:	Even
Handshaking:	XON/XOFF

Fig 10: Serial Port Default Settings

Serial Interface Commands and Responses

Most of the commands follow the same structure: a 3 letter command followed by a colon, followed by a parameter (if any) and terminated by Carriage Return with optional Line Feed. A Line Feed character may be sent but it will be ignored by the RB-SC2. Commands are not case sensitive and all parameters are in hex.

Responses are CR & LF terminated.

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DIGITAL AUDIO CONVERTERS - RB-SC2

After the delay synchroniser has been powered-up, an initialisation string is sent "Initialising SC2".

Following are the commands and the expected responses:

Command	Description	Response
Bnn:	- Baud Rate nn=11 (115200) nn=57 (57600) nn=38 (38400) nn=19 (19200) nn=96 (9600)	- ACK: (at old rate)
CSD:nn	- Set Channel Status Type nn=00 nn=01 nn=02	- ACK: - Send Consumer CS Data - Follow Input (or use last set) - Send Professional CS Data
DWN:	- Download new code Then send S-records followed by S-record terminator to force programming of flash memory and reboot	-OK: then erasing message
FPL:	- Request Front Panel Limits Where a = Lock Status (0=unlocked, 1=locked) b = Master Frequency Limit (0=no limit, 1=48kHz, 2= 96kHz)	- FPL:a_b
FPS:	- Request Front Panel Settings Where aa = value of Sync Mode as MOD: below bb = value of Master Frequency as FRQ: below cc = value of Sync Frequency as FRQ: below dd = value of CS Data as CSD: above ee = value of Input 1 Source Select as SS1: below ff = value of Input 2 Source Select as SS2: below	- FPS:aa_bb_cc_dd_ee_ff
FRQ:nn	- Set Master Clock Frequency nn=00 nn=01 nn=02 nn=03 nn=04 nn=05 nn=06	- ACK: - Select 32kHz Master Clock Sample Rate - Select 44kHz Master Clock Sample Rate - Select 48kHz Master Clock Sample Rate - Select 88kHz Master Clock Sample Rate - Select 96kHz Master Clock Sample Rate - Select 176kHz Master Clock Sample Rate - Select 192kHz Master Clock Sample Rate
LOC:	- Lock Front Panel	- ACK:
MEM:	- Memory Dump Where add is memory address starting at 060(hex) and data is 16 bytes of data, each byte shown as 2 hex digits	- Memory, then 10 lines Madd:data



MFQ:nn	- Limit Front Panel Frequency Selection Where nn=0 for No Limit nn=1 for 48kHz Limit & nn=2 for 96kHz Limit	- ACK:
MOD:nn	- Set Mode & Select Sync Source nn=00 - Master Mode nn=08 - Auto Mode nn=10 - Auto Lock Mode nn=20 - Slave Mode For last three modes add nn=01 - Select External AES Sync input as sync source nn=02 - Select Word Clock input as sync source nn=03 - Select Other Digital input as sync source (X-Lock) nn=04 - Select Optional Video input as sync source	- ACK:
REG:	- Register Dump - Registers, then 4 lines of 8 x Rrr:dd Where rr is register number and dd is value shown as 2 hex digits	
SER:	- Serial Number request	- SER:012345
SRQ:	- Status Request Where aa = SRC 1 Status bb = SRC 2 Status ss = Sync Status	- STA:aa_bb_ss
SS1:nn	- Select Channel 1 Digital Audio Source nn=00 for AES/EBU nn=01 for S/PDIF & nn=02 for Optical	-ACK:
SS2:nn	- Select Channel 2 Digital Audio Source nn=00 for AES/EBU nn=01 for S/PDIF & nn=02 for Optical	-ACK:
STK:	- Stack Dump - STK: aaaa, then 6 lines Sadd:data Where aaaa= current Stack Pointer, add is memory address starting at 0800(hex) and data is 16 bytes of data, each byte shown as 2 hex digits	
UNL:	- Unlock Front Panel	- ACK:
UID:	- Unit ID Request	- UID:SC2
VER:	- Firmware Version Request	- VER:1.23 or BOOT:1.23

Fig 11: Serial Interface Commands and Responses

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Error messages

The following error messages can be returned for illegal commands

Err:01	- return if Command Not Found
Err:02	- return if Missing Parameter
Err:04	- return if Parameter out of range

Upgrading Firmware

Occasionally, it may be necessary to upgrade the firmware on the RB-SC2 to add new functionality and fix software bugs. New firmware updates will be made available from time to time on the Sonifex website. Visit www.sonifex.co.uk for details.

The firmware is upgraded using the Serial Control Interface - SCI. This PC software downloads the new firmware and initiates the upgrade process. To upgrade the firmware, select the required firmware file in the SCI upgrade firmware section. The RB-SC2 will automatically reboot and enter the bootstrap mode prior to the transfer of the firmware file. The SCI software downloads the firmware to internal memory, and then erases the current firmware before programming the update. When the firmware is complete, the unit reboots with the new firmware.

Please note: Firmware files can take several minutes to transfer to the RB-SC2 at lower baud rates. To speed up the process, select a higher baud rate prior to transferring the new firmware.

Technical Specification RB-SC2

Connections

Digital Inputs: 2 x AES/EBU XLR 3 pin female
 2 x S/PDIF RCA phono
 2 x TosLink optical input

Digital Outputs: 2 x AES/EBU XLR 3 pin plug
 2 x S/PDIF RCA phono socket
 2 x TosLink optical output

Sync Inputs: 1 x AES/EBU XLR 3 pin female
 1 x Word Clock BNC
 1 x Video Input (optional)

Video Sync Specs: The RB-SYD Digital video sync board will accept 270Mbps SD-SDI and HD-SDI signals covered by SMPTE-259-M-C (SD) and SMPTE-292M (HD).
 The RB-SYA Analogue video sync board will accept a composite signal of NTSC (525), PAL (625) & SECAM (625) signals covered by SMPTE-170-M (NTSC) and ITU-R BT.470-6 (PAL & SECAM).



Operational Controls	
Master Frequency Select:	32, 44.1, 48, 88.2, 96, 176.4 or 192kHz via FREQUENCY push-button
Channel Status Bits:	Consumer mode, professional mode or follow input, via CS DATA push-button
Digital Input Select:	AES/EBU, S/PDIF or TOSlink optical, via INPUT1 or INPUT2 push-buttons
Sync Input Select:	AES/EBU, Word Clock, X-Lock or Video, via SYNC push-button
Sync Mode Select:	Master, slave, auto or auto lock, via SYNC push-button
Other Connections	
Mains Input:	Universal filtered IEC, continuously rated 85-264VAC @47-63Hz, max 10W
Serial Port:	RS232 9 Pin D-type socket
Equipment Type	
RB-SC2	Dual Stereo Sample Rate Converter, 24 bit 192kHz
Physical Specifications	
Dimensions (Raw)	48cm (W) x 10.8cm (D*) x 4.2cm (H) (1U) 19" (W) x 4.3" (D*) x 1.7" (H) (1U)
Dimensions (Boxed)	59cm (W) x 27.5cm (D) x 11cm (H) 23.2" (W) x 10.8" (D) x 4.3" (H)
Weight	Nett: 1.4kg Gross: 2.0kg Nett: 3.1lbs Gross: 4.4lbs
Accessories	
RB-SYA	Analogue video sync board (NTSC, PAL & SECAM)
RB-SYD	Digital video sync board (SD-SDI & HD-SDI)
RB-RK3	1U Rear panel rack kit for large Redboxes

Fig 12: Technical Specifications

5 DIGITAL AUDIO CONVERTERS

5 RB-DAC1 Digital to Analogue Converter

Introduction



Fig 5-1: RB-DAC1 Front Panel

Using 24 bit, 96kHz capable devices, the RB-DAC1 Digital to Analogue Converter is a 1U rack-mount which produces a stereo balanced XLR or unbalanced phono output from an incoming AES/EBU or S/PDIF digital input signal. There is also a headphone output for monitoring purposes.

The analogue outputs have an output level control, allowing full-scale settings selectable from +12dBu, +18dBu or +24dBu. De-emphasis on the output can be enabled via dipswitch.

There is a button to select either the AES/EBU or S/PDIF input for the D/A converter, which is located on the rear panel.

When operating, the front panel power LED flashes whenever the unit is not synchronised to the incoming digital signal.

System Block Diagram

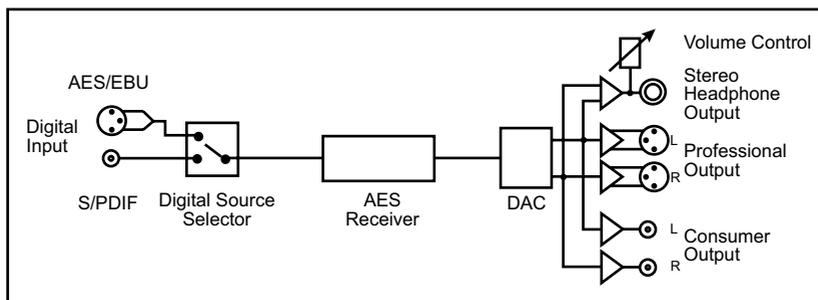


Fig 5-2: RB-DAC1 System Block Diagram

Front Panel Indicators & Controls

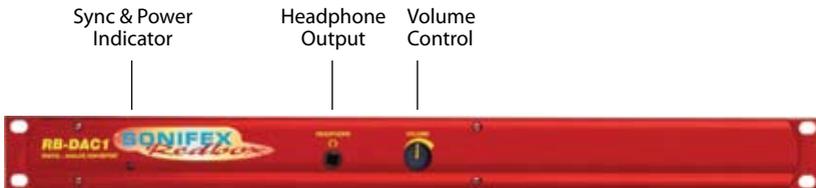


Fig 5-3: RB-DAC1 Front Panel

Sync & Power Indicator

The LED on the front panel is normally red to indicate that power is present on the unit. However, it also has a secondary role to indicate the status of the digital inputs :

Flashing between red and amber – indicates a loss of a valid digital input signal.

Headphone Output

The output available on the front panel through a 1/4" stereo jack socket, is designed to drive 150 mW into 32Ω to 600Ω professional headphones.

Volume Control

The headphone output has its own volume control, which is independent of the level adjustment for the main outputs, and has a maximum output level of +12dBu.

Rear Panel Connections and Operation

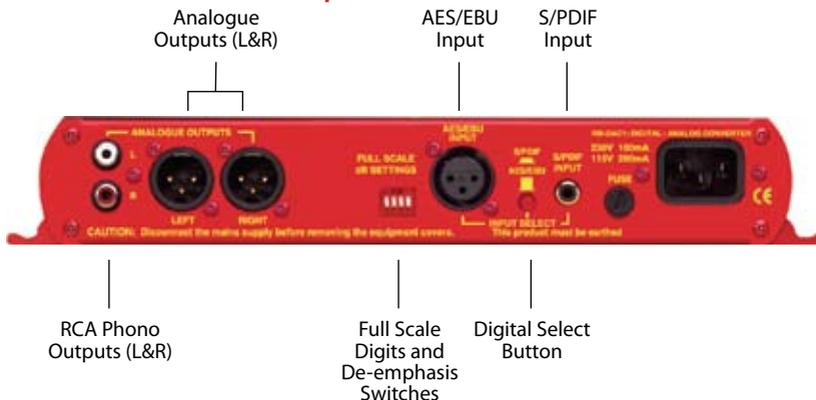


Fig 5-4: RB-DAC1 Rear Panel



5 DIGITAL AUDIO CONVERTERS



RB-DAC1 Inputs

AES/EBU Input

The digital input XLR 3 pin socket has an impedance of 110Ω. It has the following connections:

Pin 1: Screen

Pin 2: Phase

Pin 3: Non-phase

The signals on this connector should meet the IEC 60968 specification

S/PDIF Input

The S/PDIF digital phono input has an impedance of 75Ω.

RB-DAC1 Outputs

Analogue Outputs (Left and Right)

The XLR 3 pin output plug connectors are electronically balanced with an output impedance of less than 50Ω. They have the following connections:

Pin 1: Screen.

Pin 2: Phase.

Pin 3: Non-phase.

RCA Phono Outputs (Left and Right)

These RCA (phono) outputs are unbalanced and have an output impedance of less than 75Ω.

Rear Panel Controls

Full Scale dB Settings			
Switch Number	+12	+18	+24
1	OFF	ON	ON
2	OFF	OFF	ON
Emphasis Settings			
3	ON	De-emphasis On	
3	OFF	De-emphasis Off	
4		Reserved	

Fig 5-5: RB-DAC1 Full Scale Digits and De-emphasis Settings Switches

Output Level Adjustment

The output gain can be adjusted by dipswitches on the rear panel. The output gain can be set for a signal of full-scale digits in the digital domain to give +12, +18 or +24dBu output on the XLR connectors. The consumer output on the phono connector has a further 8½dBu attenuation incorporated.

De-emphasis Switch

If de-emphasis is selected (switch 3) the RB-DAC1 will decode 50/15µs emphasis when indicated by certain channel status bits in the incoming digital audio data.

Digital Select Button

This button is used to switch the digital input from the AES/EBU XLR connector (button out) to the S/PDIF phono connector (button in).



Technical Specifications RB-DAC1

Connections

Digital Inputs:	1 x AES/EBU XLR 3 pin female 1 x S/PDIF RCA phono
Analogue Outputs:	2 x XLR 3 pin male (balanced) (L & R) 2 x RCA phono (unbalanced) (L & R)
Headphone Output:	1 x ¼" (6.35mm) A/B gauge 3-pole stereo jack socket
Mains Input:	Filtered IEC, 110-120V, or 220-240V switchable, fused 10W max
Fuse Rating:	Anti-surge fuse 100mA 20 x 5mm (230VAC) Anti-surge fuse 250mA 20 x 5mm (115VAC)

Audio Specification

Max Output Level:	+24dBu (balanced outputs) +14dBu (unbalanced outputs) +12dBu (headphone outputs)
Output Impedance:	<50Ω (balanced outputs) <75Ω (unbalanced outputs)
Dynamic Range:	>100dB
Noise & Distortion:	<0.01% THD + N @1kHz
Sample Freq Range:	30kHz – 100kHz
Gain Range:	Selectable 12dBu, 18dBu or 24dBu output level, ref. FSD
Headphones:	Drives 150 mW into 32Ω to 600Ω headphones
Max Output Level:	+12dBu

5 DIGITAL AUDIO CONVERTERS



DIGITAL AUDIO CONVERTERS - RB-DAC1

Operational Controls

Digital Input Select:	AES/EBU or S/PDIF, via push-switch
Gain Select:	Selectable 12dBu, 18dBu or 24dBu output level, ref. FSD
De-emphasis On/Off:	Dipswitch

Equipment Type

RB-DAC1 Digital to Analogue Converter



Physical Specifications

Dimensions (Raw)	48cm (W) x 10.8cm (D) x 4.2cm (H) (1U)
	19" (W) x 4.3" (D) x 1.7" (H) (1U)
Dimensions (Boxed)	58.5cm (W) x 22.5cm (D) x 7cm (H)
	23" (W) x 8.9" (D) x 2.8" (H)
Weight	Nett: 1.4kg Gross: 2.0kg
	Nett: 3.1lbs Gross: 4.4lbs

6 RB-SP1 Digital Splitter & Combiner

Introduction



Fig 6-1: RB-SP1 Front Panel

The RB-SP1 Digital Splitter & Combiner is used to interface various double sampling pieces of equipment. Some older equipment uses 2 AES/EBU connectors for double sampling with each connector carrying an audio signal at a normal frame rate, whilst other equipment has a single connector using twice the frame rate. The RB-SP1 can interface between them, either combining the signals from 2 XLR's into 1, or splitting the signal from 1 XLR into 2.

The RB-SP1 can also be used for interfacing stereo and mono signals to digital mixing desks by splitting the left and right signals of a stereo XLR to two separate XLR's, and vice versa by combining them.

Additionally, a sample rate converter on the second digital input can be used to convert the sample rate of the secondary input to that of the primary input. The RB-SP1 can handle sample rates up to 96kHz and sample sizes of 16, 20 and 24 bit.

There are two types of operation : Split 96, and Stereo/Mono. These each have three different switch modes : Split, Bypass and Combine.

Both inputs and outputs can be selected as either AES/EBU or S/PDIF with the resultant digital level following the switch selection.

System Block Diagram

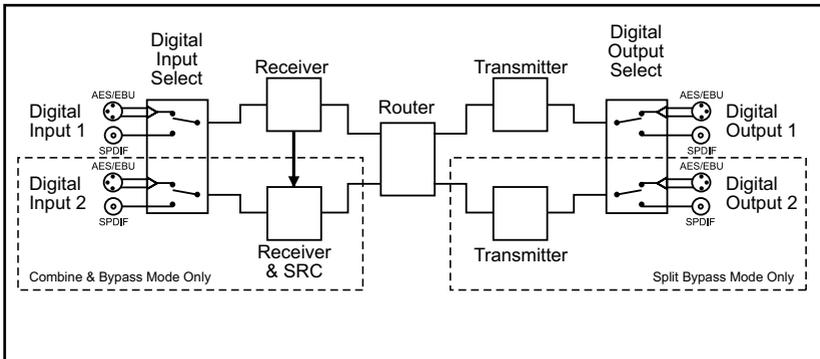


Fig 6-2: RB-SP1 System Block Diagram





Front Panel Indicators & Controls

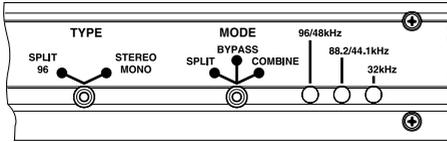


Fig 6-3: RB-SP1 Front Panel

Front Panel LED's

There are four LED indicators situated on the front of the unit. The red LED on the far left of the front panel is to indicate that power is present on the unit.

The three LED's grouped together on the right hand side have two roles, see Fig 6-3. The first is to indicate the synchronisation frequencies of the incoming digital signals, and the second is to flash when a signal has been lost. These indicators are labelled individually to show the current sync frequencies.

Type & Mode Switches

The Type and Mode switches are on the front panel – see Fig 6-3. The Type switch sets the unit into either the Split 96 or Stereo/Mono style of operation. The Mode switch sets the unit into Split, Bypass or Combine mode.

See Figure 6-4 for diagrammatic explanations of the different types and modes available.

Split 96 – This is a method that allows older equipment to handle 48kHz double sampled (96kHz) digital signals. To do this, Split 96 uses two digital signals running at 48kHz, where both sub-frames of a single 48kHz stream are used to carry information about a mono signal, with the resultant signal of both 48kHz streams being equivalent to a stereo 96kHz signal. The unit will also perform the same function for 88.2kHz stereo and 44.1kHz double sampled signals.

- In **Split Mode** a single stereo 96kHz signal is received into input 1 which is then output as two separate 48kHz signals. Output 1 will contain just left channel data and output 2 will contain just right channel data. If the signal from the input disappears then both Outputs will be muted.
- In **Combine Mode** two 48kHz double sampled input signals are combined to create a single 96kHz signal on output one. If either input 1 or 2 are lost then the output will be muted. In this mode output 2 will always be muted.

Stereo/Mono – This allows a digital stereo signal to be separated into two mono digital signals and vice-versa.

- In **Split Mode** a single stereo digital signal is routed to two digital outputs. Output 1 will contain the original left channel data on its left output and a muted signal on

its right output. Output 2 will contain the original right data on its left output and a muted signal on its right output. This mode can be altered by the dipswitch settings, to a channel swap mode or a dual mono mode. These are described in more detail in the Rear Panel Controls section below.

- In **Combine Mode** two mono signals are combined to create a single stereo signal. Output 1 will contain the input 1 left channel data on its left channel and input 2 left channel data on its right channel. This mode can be altered by a dipswitch setting, as with the split mode. Where either the left or right channel data of input 2 is output on the right channel data of the output. If the two signals are of different sample rates, a sample rate converter can be switched in place to convert the sample rate of input channel 2 to that of input channel 1. In the Stereo/Mono Combine mode, output 2 is always muted.

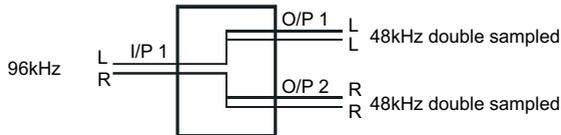
Bypass Mode – In Bypass mode, input 1 is routed to output 1 and input 2 is routed to output 2. Input 1 is used as the master clock input. If input 2 is at a different sample rate to input 1 then input 2 is sample rate converted to match input 1. If the signal from input 2 is lost then output 2 will be muted. However, if the signal from input 1 is lost then both outputs will be muted and will only return once a signal is present on input 1.

Note that when the unit is operating in **Bypass Mode** the Type switch is ignored.

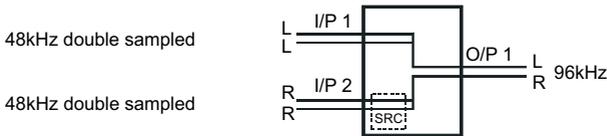


Split 96

Split



Combine



6

DIGITAL AUDIO CONVERTERS

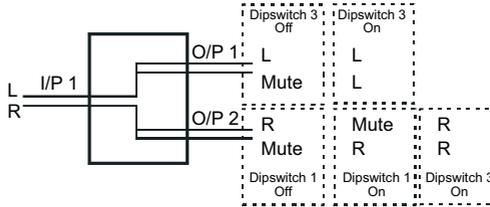


DIGITAL AUDIO CONVERTERS - RB-SP1

Stereo/Mono

Split

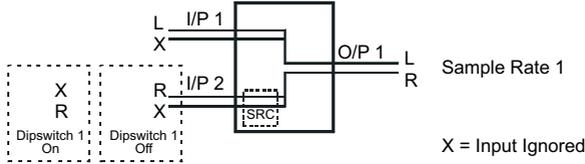
Any Sample Rate



Combine

Sample Rate 1

Sample Rate 2



Bypass

Sample Rate 1

Sample Rate 2

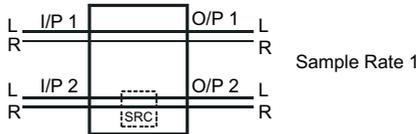


Fig 6-4: RB-SP1 Type and Mode Flow Diagrams

Rear Panel Connections and Operation

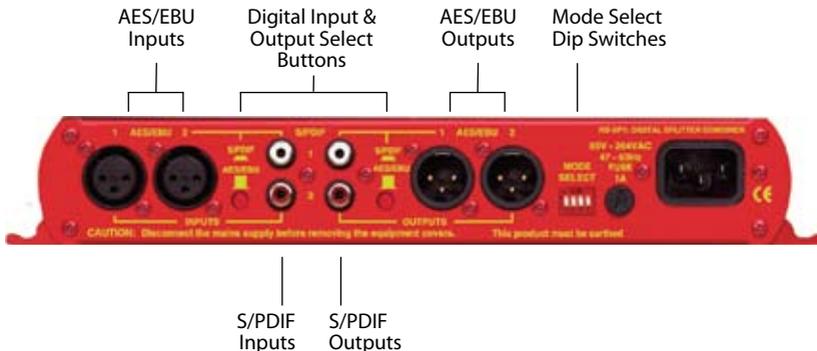


Fig 6-5: RB-SP1 Rear Panel



Inputs and Outputs

AES/EBU Inputs

The digital source XLR 3 pin sockets have an impedance of 110Ω. They have the following connections and meet the IEC 60968 specification:

- Pin 1: Screen
- Pin 2: Phase
- Pin 3: Non-phase

S/PDIF Inputs

The digital source RCA phono inputs both have an impedance of 75Ω.

AES/EBU Outputs

The digital output XLR 3 pin plugs have an impedance of 110Ω. They have the following connections and meet the IEC 60968 specification:

- Pin 1: Screen
- Pin 2: Phase
- Pin 3: Non-phase

S/PDIF Outputs

The digital output S/PDIF phono outputs have an impedance of 75Ω.

Rear Panel Controls

Digital Input & Output Select Buttons

These buttons are used to switch the digital connection between the AES/EBU XLR connector (button out) and the S/PDIF phono connector (button in) for the digital source and digital output.

Mode Select Dip Switches

RB-SP1 Mode Select Settings		
1	ON	Input 1 Channel B = Output 2 Channel B in Stereo/Mono Split Mode
1	OFF	Input 1 Channel B = Output 2 Channel A in Stereo/Mono Split Mode
2	ON	SRC Enabled
2	OFF	SRC Disabled
3	ON	Dual Mono in Stereo/Mono Split Mode
3	OFF	Single Mono in Stereo/Mono Split Mode
4		Reserved

Fig 6-6: RB-SP1 Mode Select Dip Switches

Altering the Stereo/Mono Split/Combine Mode – In Split mode With switch 1 off, channel A (Left) and B (Right) on the input are transferred to channel A (Left) of outputs 1 and 2 respectively. With switch 1 on, channel A (Left) on the input is transferred to channel A (Left) on output 1 and channel B (Right) on the input is transferred to channel B (Right) of output

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DIGITAL AUDIO CONVERTERS



DIGITAL AUDIO CONVERTERS - RB-SP1

2. With switch 3 on (switch 1 is ignored), channel A (Left) is transferred to both channels on output 1 and channel B (Right) on the input is transferred to both channels on output 2.

In Combine mode with switch 1 off, channel A (Left) on input 1 is transferred to channel A (Left) of output 1, and channel A (Left) on input 2 is transferred to channel B (Right) of output 1. With switch 1 on channel A (Left) on input 1 is transferred to channel A (Left) of output 1, and channel B (Right) on input 2 is transferred to channel B (Right) of output 1.

Switching on the Sample Rate Converter - Switch 2 is used to turn the sample rate converter on, or off and can be used in all modes. When there are two inputs connected to the unit which are at different sample rates or which need to be synchronised, then the sample rate converter should be switched on. If it is not, then you may suffer from missed samples and bit errors affecting the signal output. If the sample rates of the incoming signals are always going to be the same and are synchronised, then switch the sample rate converter off, as leaving it on will worsen the output signal (signal jitter will increase).

Technical Specifications RB-SP1

Audio Specifications

Input Impedance:	110Ω ± 20% balanced (AES/EBU)
Input Impedance:	75Ω ± 5% unbalanced (S/PDIF)
Output Impedance:	110Ω ± 20% balanced (AES/EBU)
Output Impedance:	75Ω ± 5% unbalanced (S/PDIF)
Signal Level	3V/10V peak to peak min/max (AES/EBU)
	0.5V ± 20% peak to peak (S/PDIF)
Sample Freq Range:	30-100kHz (i.e. including 32kHz, 44.1kHz, 48kHz, 64kHz, 88.2kHz and 96kHz)

Connections

Audio Inputs:	2 x AES/EBU XLR 3 pin female
	2 x S/PDIF RCA phono female
	(Input button select between AES/EBU and S/PDIF)
Audio Outputs:	2 x AES/EBU XLR 3 pin male
	2 x S/PDIF RCA phono female
	(Output button selects between AES/EBU and S/PDIF)

Mains Input: Filtered IEC, continuously rated
85-264VAC @ 47-63Hz max 10W

Fuse Rating: Anti-surge fuse 1A 20 x 5mm

Equipment Type

RB-SP1 Digital splitter & combiner



Physical Specifications

Dimensions (Raw) 48cm (W) x 10.8cm (D) x 4.2cm (H) (1U)
19" (W) x 4.3" (D) x 1.7" (H) (1U)

Dimensions (Boxed) 58.5cm (W) x 22.5cm (D) x 7cm (H)
23" (W) x 8.9" (D) x 2.8" (H)

Weight Nett: 1.6kg Gross: 2.2kg
Nett: 3.5lbs Gross: 4.8lbs



7 SYNCHRONISERS, DELAYS & SILENCE DETECTORS

7 RB-DS2 Audio Delay Synchroniser

Introduction



Fig 7-1: RB-DS2 Front Panel

The RB-DS2 is a stereo audio delay synchroniser used for resynchronising audio to video following delay processes such as standards conversion, transmission delay, logo insertion, video aspect ratio conversion and network delays. It can be used for fixed installations to correct a permanent audio delay, or on an intermittent basis to provide occasional correction, for example for live links. Accepting digital audio signals up to 96kHz, 24 bit, the sonic quality of the RB-DS2 is superb and silent switching is used to provide the smoothest, cleanest audio delay available.

The RB-DS2 has both balanced analogue and AES/EBU digital audio inputs and outputs on 3 pin XLR connectors. It can act as a combined A/D and D/A unit meaning that analogue inputs can be delayed and output as AES/EBU and vice-versa. It is a stereo delay, but can also be used as a dual mono delay to process each audio path separately, or as a mono delay using the left inputs and outputs.

As standard the RB-DS2 can provide up to 10.5 seconds of delay at 96kHz sampling, 24 bit (42 secs at 48kHz, 16 bit). An internal Compact Flash™ expansion allows up to 2GB of memory to be accessed providing delay times of over 4 hours, for example, to delay a programme output across different time-zones, or to shift a broadcast programme by 1 hour for a satellite rebroadcast. Delay times can be selected in samples, fields, frames, milliseconds, metres and, with the Compact Flash™ expansion, in hh:mm:ss. Frame and field definitions can be for PAL (25 frame) or NTSC (30 frame) signals.

A front panel blue vacuum fluorescent display with rotary controller is used for selecting the various settings of the delay, which include the source (analogue or digital), channels, sample rate, sample bit width, format (PAL or NTSC), delay units and the delay itself. Additionally, input peak digits can be selected from +12dBu, +18dBu and +24dBu for FSD and two left and right pre-set potentiometers on the rear panel allow the input gain range to be altered by ± 3 dB around the selected peak digits.

The analogue output gain range can be altered from -6dBu to +24dBu output level, ref FSD.

Both analogue and digital outputs can be separately muted and a front panel Bypass button disengages electro-mechanical relays to divert both analogue and digital inputs to their outputs. This is also disengaged automatically when a power-fail occurs.

All of the settings in the unit can be saved to one of 8 configuration settings. These Configs can be viewed, edited, saved and loaded, and also remotely loaded by using one of the 8 GPI contacts, meaning that any setting, such as delay time or Bypass, can be altered instantaneously using a GPI signal. The RB-DS2 also has an RS232 serial port for remotely

controlling the unit and there are 4 remote outputs which can be used for signalling. The front panel controls can be locked-out for situations where remote control is being used to run the unit, or where physical security is required.

Extended Mode using Compact Flash™

Extended mode allows the delay synchroniser to provide audio delays in excess of 4 hours so that the unit can act as a broadcast time-zone delay. The memory card slot is internally mounted and currently supports Compact Flash™ cards up to 2Gbytes - see page 67 for information on opening the RB-DS2 to insert a CF card. To operate correctly, the flashcard must support PIO Mode 4, or higher, such as the SanDisk Standard CF cards. The delay synchroniser uses the DOS format with a 16bit FAT. If a Compact Flash™ memory card is installed in the internal card slot, the status of the card will be indicated in the lower right side of the display. If the card is not formatted, or the format is corrupted, the display will show "NOFMT" indicating that the card cannot be written to until it has been formatted (see System Menu: Memory Card for details on formatting the card). When the card is correctly formatted, the capacity is shown in Mbytes.

The delay synchroniser generates a file on the Compact Flash™ memory card, prior to starting the delay process, which effectively reserves all the space available and provides for the largest possible delay. If the Compact Flash™ memory card was removed and placed in a PC card reader, the file, named "RB-DS2.WAV", would play using any utility that's supports Broadcast Wave Format - PCM files.

Note: Extended Mode can only be used for mono and stereo CHANNELS settings, not for dual mono signals.

System Block Diagram

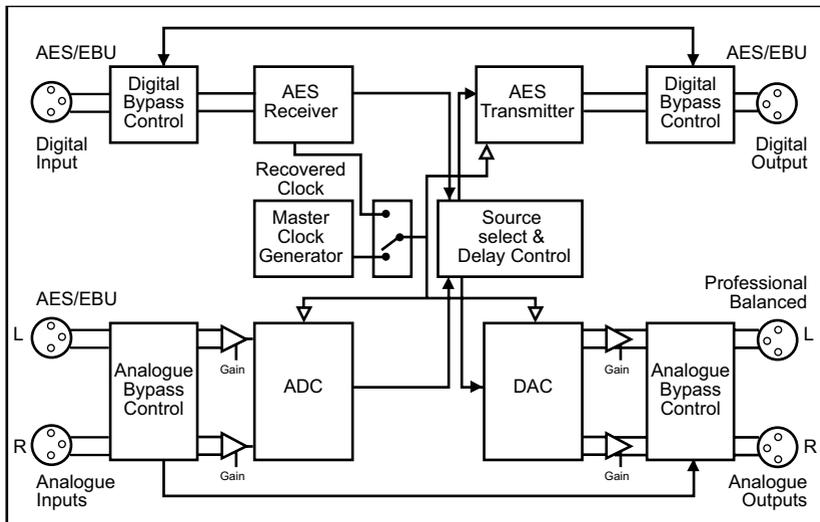


Fig 7-2: RB-DS2 System Block Diagram

7 SYNCHRONISERS, DELAYS & SILENCE DETECTORS

7 SYNCHRONISERS, DELAYS & SILENCE DETECTORS

Front Panel Controls

All of the RB-DS2's configurations can be accessed with the single front panel mounted rotary control in conjunction with the vacuum fluorescent display.

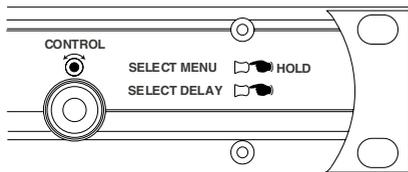


Fig 7-3: Front Panel CONTROL

Setting the Delay

To adjust the delay, **lightly press the rotary CONTROL** to enable the delay select mode. A solid line will appear under the selected digit. To change the selected digit, turn the rotary control. When configured for dual mono operation, the selection highlight will move between left and right channel delay values. Once the required digit is highlighted, press the rotary control to switch to delay edit mode. The highlight under the selected digit will begin to flash indicating that delay edit mode is enabled. The value of the selected digit can be changed, by turning the rotary control. When the selected digit is set to the required value, press the rotary control again to return to the delay select mode. A different digit can now be selected. If no further digit is selected, delay select mode will be disabled and the selection highlight will disappear after 2.5 seconds.

Changes to the delay are implemented dynamically. If the selected delay value exceeds the amount of audio data that has been stored, the analogue and digital outputs will be muted until the required delay has been acquired. The audio will then be presented at the outputs. In extended mode, where the memory card installed in the internal Compact Flash™ Storage Card Slot is used, delay periods can be in excess of 1 hour. During acquisition of long delays in extended mode, a progress bar is displayed to give an indication of when the audio output will start. The size of the progress bar decreases as the delay is acquired.

The delay value that can be set cannot exceed the maximum delay for the selected mode. For example, for an analogue input sampled at 96kHz, 24bit, using 25fps, the maximum delay in frames for normal mode is 268 (see figures 4, 5 and 6 for maximum delay values).

Selecting The Main Menu

The RB-DS2's configuration options are accessed from the main system menu. To access the main system menu, **press and hold the rotary CONTROL for at least 1.5 seconds**. The menu structure is arranged in a multi-tiered format with several menu items providing access to sub-menus. To select a menu item, turn the rotary control until the required item is highlighted, then press the rotary control to select the item. Each configuration is shown as a list of available options. The currently selected option will be shown with a tick along side it. The last item shown in each menu is ►CLOSE. Selecting close exits the system menu and returns to the main display. In each sub-menu, the penultimate item shown is ◀BACK. Selecting back displays the previous menu.

Bypassing The Delay

The Bypass button enables the delay bypass mode, which internally connects the inputs to outputs in both the analogue and digital domains. The Bypass button is illuminated when the delay bypass mode is enabled. When power is removed from the unit, the bypass mode is automatically enabled.

System Reset

A single LED on the left side of the front panel indicates that power is applied to the unit. The system reset button is accessible through the small aperture to the right of the power LED.

Rear Panel Connections and Operation



Fig 7-4: RB-DS2 Rear Panel

XLR Analogue Inputs (Left and Right)

The XLR 3 pin sockets used for the left and right channel analogue inputs are electronically balanced and have an impedance greater than 10k Ω bridging. Each XLR has the following connections:

- Pin 1: Screen
- Pin 2: Phase
- Pin 3: Non-phase

Analogue Input Adjustment

The Input Peak Digits setting (see System Menu: Volume) sets the analogue input gain for full-scale digits in the digital domain. The preset potentiometers next to the analogue inputs allow an additional adjustment of this value by ± 3 dB.

AES/EBU Input

The digital input XLR 3 pin socket has an impedance of 110 Ω . It has the following connections:

- Pin 1: Screen
- Pin 2: Phase
- Pin 3: Non-phase

The signals on this connector should meet the IEC 60968 specification.

XLR Analogue Outputs (Left and Right)

The XLR 3 pin output plugs are electronically balanced with an output impedance of less than 50 Ω . They have the following connections:

- Pin 1: Screen
- Pin 2: Phase
- Pin 3: Non-phase

7 SYNCHRONISERS, DELAYS & SILENCE DETECTORS



AES/EBU Output

The digital output XLR 3 pin plug has an impedance of 110Ω. It has the following connections:

Pin 1: Screen

Pin 2: Phase

Pin 3: Non-phase

The signals on this connector comply with the IEC 60968 specification.

Remote Inputs and Outputs

The 15-way 'D' type socket connector presents 8 active low inputs and 6 open collector outputs. The inputs are used to remotely select one of 8 configuration setups (see System Menu: Setups) when the Remote Loading function is enabled. The outputs are not currently used. It has the following connections:

Pin 1: Active low input 1

Pin 2: Active low input 2

Pin 3: Active low input 3

Pin 4: Active low input 4

Pin 5: Active low input 5

Pin 6: Active low input 6

Pin 7: Active low input 7

Pin 8: Active low input 8

Pin 9: Signal ground

Pin 10: Open collector output 1

Pin 11: Open collector output 2

Pin 12: Open collector output 3

Pin 13: Open collector output 4

Pin 14: Open collector output 5

Pin 15: Open collector output 6

RS232

The 9-way 'D' type socket connector allows direct connection to a serial port on a PC via a pin-to-pin cable. See the section on Serial Port Control for details of this interface. It has the following connections:

Pin 1: n/c

Pin 2: Transmit data (TxD)

Pin 3: Receive data (RxD)

Pin 4: n/c

Pin 5: Ground

Pin 6: n/c

Pin 7: Clear To Send (CTS)

Pin 8: Request To Send (RTS)

Pin 9: n/c

The Default Screen

The default screen (Fig 7-5) shows the current selections for audio source, sample rate, delay units and delay. In stereo mode, the delay is common to both left and right channels. In dual mono mode, the delay for the left and right channels is shown separately as each channel is delayed individually. Also shown on the main screen is the mute status of the analogue and digital outputs and the currently selected frame rate and sample size. All of the Audio Delay Synchronisers' settings are configured from the main system menu with the exception of the actual delay. The delay is adjusted directly from the main screen following the instructions on page 50.



Fig 7-5: The Default Screen

The Main Menu

To access the main menu, press and hold the rotary CONTROL knob and follow the instructions on page 50.

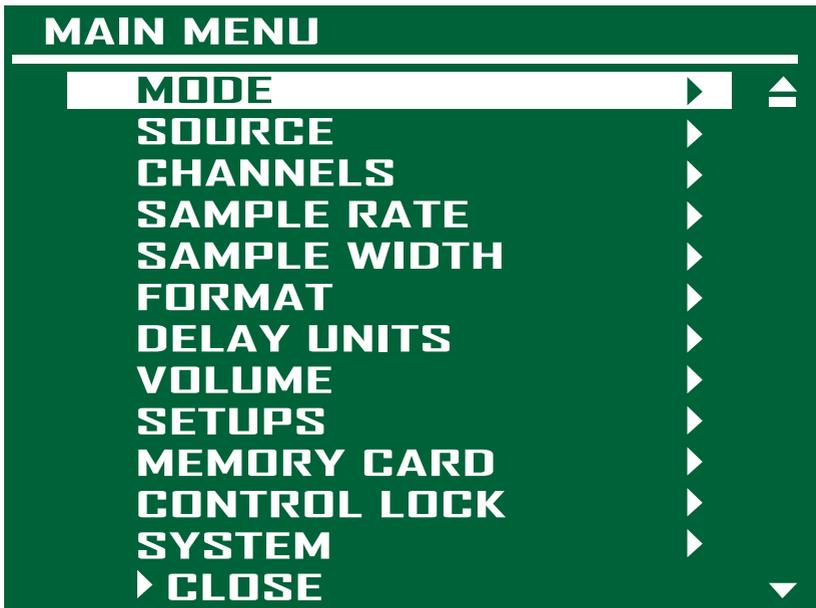


Fig 7-6: The Main Menu

7 SYNCHRONISERS, DELAYS & SILENCE DETECTORS



Mode

This option selects which memory is used for the audio delay process. From the main system menu select MODE. The available options are NORMAL and EXTENDED USING CF. Changing the mode will force the analogue and digital audio outputs to be muted as the current delay duration is reset and the delay is acquired on the new media. See figures 3, 4 and 5 for the maximum delays in each mode.

- Normal – This mode selects the internal RAM to be used for the delay process. The size of the internal memory used is 8232960 bytes.
- Extended Using CF – This mode selects the Compact Flash™ memory card loaded in the internal memory card slot. The internal memory card slot currently supports Compact Flash™ cards up to 2Gbytes. If the Extended mode is selected with no Compact Flash™ memory installed, a message box will be display and the audio output will not be interrupted.

The default mode is Normal.

Source

This option selects which input is used for the audio delay process. From the main system menu select SOURCE. The available options are ANALOGUE, ANALOGUE SYNC and DIGITAL. The delayed audio appears at both the analogue and digital outputs regardless of the input source selected.

- Analogue – This selects the audio at the analogue inputs as the source for the delay. The input is sampled and delayed using the selected sample rate generated from an internal master clock. The digital input is not used.
- Analogue Sync – This selects the audio at the analogue inputs as the source for the delay. The input is synchronised to the digital audio sync signal at the digital input and the analogue and digital audio outputs will be muted until a valid sync signal is detected. Once a valid sync signal has been received and the delay process starts, any subsequent loss of sync signal will force the unit to revert to an internally generated master clock rate as near as possible to the last sync signal received.
- Digital – This selects the audio at the digital input as the source for the delay. With no digital signal applied, the analogue and digital audio outputs will be muted. Once a valid digital input is received, the unit locks to this signal and the delay process starts. Any change of sample rate on the digital input signal will force the unit to stop the delay and mute the analogue and digital outputs. The delay will then restart once the new sample rate has been validated. The analogue inputs are not used.

The default source is analogue.

Channels

This option selects how the left and right channels are used. From the main system menu select CHANNELS. The available options are MONO, DUAL MONO and STEREO. Changing the channels selection will mute the analogue and digital outputs while the delay is acquired with the new setting.

- Mono – The left channels only are used for the mono signal and the right channels are muted. This mode doubles the delay durations available, compared to stereo and dual mono settings.
- Dual Mono – This allows the delay on the left and right channels to be independently set. All other configuration options are common to both the left and right channels.
- Stereo – The selected delay applies to both the left and right channels.

The default channel configuration is stereo.

Sample Rate

This option selects the sample rate used when the input source is set to Analogue. From the main system menu select SAMPLE RATE. The available options are 32KHZ, 44.1KHZ, 48KHZ, 64KHZ, 88.2KHZ and 96KHZ. Changing the sample rate will mute the analogue and digital outputs while the delay is acquired with the new setting. The default sample rate is 48kHz.

Sample Width

This option selects the sample size used in the analogue to digital and digital to analogue conversion. From the main system menu select SAMPLE WIDTH. The available options are 16BIT and 24BIT. Changing the sample width will mute the analogue and digital outputs while the delay is acquired with the new setting. The default sample width is 24bits.

Format

This option is used to set the number of frames per second that apply when setting the delay in units of either Fields or Frames. From the main system menu select FORMAT. The available options are 25 FRAMES PER SEC, 24 FRAMES PER SEC, 30 FRAMES PER SEC and 30 DROP FRAME.

- 25, 24 and 30 Frames Per Second – Selects the required frame rate. When using Fields as the selected delay units, there are 2 Fields to 1 Frame, so the Fields per second equivalents of these settings are 50, 48 and 60 respectively.
- 30 Drop Frame – This refers to an actual frame rate of 29.97 frames per second.

The default format is 25 frames per second.

Delay Units

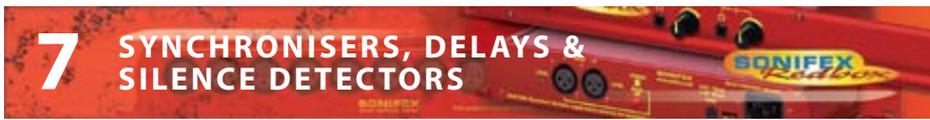
This option selects the units for the delay. From the main system menu select DELAY UNITS. The available options are SAMPLES, FIELDS, FRAMES, TIME and METRES.

- Samples – This refers to the number of 16bit or 24bit samples.
- Fields – This uses the Format option to determine the actual number of Fields/second.
- Frames – This uses the Format option to determine the number of Frames/second.
- Time – In Normal mode, the time units are milliseconds. In Extended Using CF mode, the time units are hours, minutes and seconds.
- Metres – This allows for the delay of sound travelling the selected number of metres (calculated as the speed of sound at 20 degrees C, 343.371 m/sec). The delay is set in units of 10 metres and is only available in Normal mode.

The default delay units are samples.



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Volume

This selection shows the Volume sub-menu. The sub-menu items are INPUT PK DIGITS, OUTPUT GAIN, ANALOGUE MUTE and DIGITAL MUTE.

Input Peak Digits

This option sets the full-scale dB input amplitude that will give full-scale digits in the digital domain i.e the absolute maximum analogue input level that can be put into the unit before digital limiting will occur.

From the volume sub-menu select INPUT PK DIGITS. The available options are +12 dB, +18 dB and +24 dB. The default value for input peak digits is +12dB.

Output Gain

This option is directly related to the Input Peak Digits; it sets the full scale digits on the output of the unit. It can be used to apply gain to the input signal provided that the input peak digits are set lower than 24 dBu or attenuation if required.

For an illustration of how the values relate please see the table below:

Input Peak Digits (dBu)	Output Gain (dBu)	Effective Gain (dBu)
12	6	-6
12	12	0 (unity gain)
12	18	6

From the volume sub-menu select OUTPUT GAIN. The available gain settings are from -6 dB to +24 dB in 1 dB increments. The default value for output gain is +12dB.

Analogue Mute

This option mutes the analogue audio output. From the volume sub-menu select ANALOGUE MUTE. The available options are OFF and ON. By default, the analogue mute is off.

Digital Mute

This option mutes the digital audio output. From the volume sub-menu select DIGITAL MUTE. The available options are OFF and ON. By default, the digital mute is off.

Setups

This selection shows the Setups sub-menu. The sub-menu items are VIEW, LOAD, SAVE and REMOTE LOADING. Setups provide an easy way of saving and recalling configurations settings. There are 8 available setups and each one includes the following parameters:

- Mode (normal or extended),
- Source (analogue, analogue sync or digital),
- Channels (mono, dual mono or stereo),
- Sample rate (32kHz, 44.1kHz, 48kHz, 64 kHz, 88.2kHz or 96kHz),
- Sample width (16bit or 24bit),
- Format (24fps, 25fps, 30fps or 30 drop frame),
- Delay units (samples, fields, frames, time or metres),
- Delay,
- Input peak digits (+12dB, +18dB or +24dB),
- Output gain (-6dB to +24dB),

Analogue mute,
Digital mute,
Bypass.

View

This selection lists the configuration settings of the selected setup. From the setups sub-menu select VIEW followed by the required setup number. The details of the selected setup are shown as a list.

Load

This selection loads the selected setup. From the setups sub-menu select LOAD followed by the required setup number. The analogue and digital audio outputs will be muted whilst the new configuration is loaded.

Save

This selection allows the current configuration settings to be saved. From the setups sub-menu select SAVE followed by the required setup number. All of the current settings will be saved to the selected setup, which can then be loaded at a later time.

Remote Loading

When remote loading is enabled, each setup can be loaded automatically by connecting the ground pin (pin 9) to pins 1 to 8 on the 15-way D-Type remote socket on the rear panel. Connecting pin 1 to ground selects setup 1, pin 2 selects setup 2, etc. From the setups sub-menu select REMOTE LOADING. The available options are DISABLED and ENABLED. By default, remote loading is disabled.

Memory Card

This selection shows the Memory Card sub-menu. The sub-menu items are DETAILS and FORMAT CARD. An internally mounted Compact Flash™ memory card slot accepts memory cards up to 2Gbytes. A memory card is only used in Extended Mode.

Details

This selection shows details of the installed memory. From the memory card sub-menu select DETAILS. The information shown is the model number, the capacity and the amount of space available for use by the delay process.

Format Card

This selection formats the memory card using the DOS format with a 16-bit FAT. From the memory card sub-menu select FORMAT. When Format Card is selected, the system menu is exited and the main screen is shown. A message box is then displayed indicating that the memory card is being formatted.

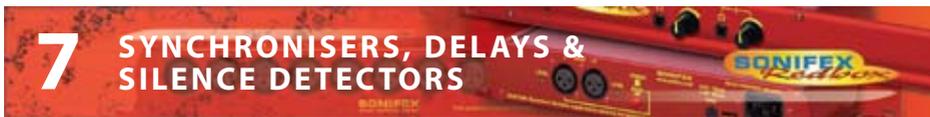
Please note: Formatting will delete all files on the memory card.

Control Lock

This option allows the front panel bypass and rotary controls to be disabled. From the main system menu select CONTROL LOCK. The available options are OFF and ON. Access to the system menu is still enabled when the control lock is on to allow the lock to be removed. When the control lock is enabled, a small key icon is displayed in the upper right side of the main screen display. By default, the control lock is off.



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System

This selection shows the System sub-menu. The sub-menu items are BAUD RATE, SET TIME, SET DATE and INFORMATION.

Baud Rate

This selection allows the baud rate of the serial port to be set. From the system sub-menu select BAUD RATE. The available options are 9600, 19200, 38400, 57600 and 115200. The default baud rate is 19200.

Set Time

This selection allows the time stored in the system real time clock to be set. From the system sub-menu select SET TIME. The current time, if set, will be displayed. To set the time, turn the rotary control to move the selection highlight between hours, minutes and seconds. Once over the required selection, press the rotary control to enable editing, which is performed by turning the rotary control. Pressing the rotary control again allows the selection highlight to be moved. Selecting OK saves the time, whilst CANCEL discards any changes made.

Set Date

This selection allows the date stored in the system real time clock to be set. From the system sub-menu select SET DATE. The current date, if set, will be displayed. To set the date, turn the rotary control to move the selection highlight between day, month and year. Once over the required selection, press the rotary control to enable editing, which is performed by turning the rotary control. Pressing the rotary control again allows the selection highlight to be moved. Selecting OK saves the date, whilst CANCEL discards any changes made.

Information

This selection shows product and software version information. From the system sub-menu select INFORMATION.

The Product Id is set to RB-DS2. This is used to identify the unit when using a PC and the Redbox control software.

The Serial Number is factory set. Always quote this number in any correspondence with Sonifex technical support.

The Bootstrap Version Number is the revision of the software that boots the system and controls the firmware upgrade process. This software is installed at the factory and can only be upgraded by Sonifex.

The Version Number is the revision of the main software. This software can be upgraded. For the latest software updates, check the Sonifex website at www.sonifex.co.uk

The Display Version Number is the revision of the software that is running on the display PCB.

Serial Port Control

The Serial Port allows the RB-DS2 to be controlled and updated from a PC via a pin-to-pin serial cable, using the Sonifex Serial Control Interface (SCI) software. This software is available as a free download from the Sonifex website at www.sonifex.co.uk/sci.

Default Settings for the Serial Port

Baud Rate: 19200

Data Bits: 8

Stop Bits: 1

Parity: Even

Handshaking: None

Serial Interface Commands and Responses

Most of the commands follow the same structure: a 3 letter command followed by a colon, followed by a parameter (if any) and terminated by Carriage Return with optional Line Feed. A Line Feed character may be sent but it will be ignored by the RB-DS2. Commands are not case sensitive.

Responses will be CR & LF terminated.

After the delay synchroniser has been powered-up, an initialisation string is sent "Initialising DS2".

Following are the commands and the expected responses:

Command	Description	Response
Bnn:	**Set baud rate where nn = 11 (115200 baud) nn = 57 (57600 baud) nn = 38 (38400 baud) nn = 19 (19200 baud) nn = 96 (9600 baud)	ACK: or NAK: (at old rate)
BPD:	Bypass disable	ACK: or NAK:
BPE:	Bypass enable	ACK: or NAK:
BSV:	**Bootstrap version request where n.n = bootstrap version	BSV:Vn.n

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SYNCHRONISERS, DELAYS & SILENCE DETECTORS - RB-DS2

Command	Description	Response
DLY:c,n	Set delay where c = 0 (left/stereo channel) c = 1 (right channel) n = delay	ACK: or NAK:
DWN:	**Download firmware new firmware installed when 'S-Record' terminator received	ACK: or NAK: (ACK: indicates download can start)
FMC:	Format memory card	ACK: or NAK:
LCS:n	Load configuration setup where n = setup number (1-8)	ACK: or NAK:
LOC:	Enable control lock	ACK: or NAK:
MCD:	Request memory card details where x = model number y = capacity z = size for delay	MCD:x,y,z or NAK:
SAM:n	Set analogue mute where n = 0 (off) n = 1 (on)	ACK: or NAK:
SCH:n	Set channels where n = 0 (mono) n = 1 (dual mono) n = 2 (stereo)	ACK: or NAK:
SDM:n	Set digital mute where n = 0 (off) n = 1 (on)	ACK: or NAK:
SDU:n	Set delay units where n = 0 (samples) n = 1 (fields) n = 2 (frames) n = 3 (time)	ACK: or NAK:
SER:	**Serial number request where nnnnnn = serial number	SER:nnnnnn

Command	Description	Response
SFM:n	Set format where n = 0 (25 frames per second) n = 1 (24 frames per second) n = 2 (30 frames per second) n = 3 (30 drop frame)	ACK: or NAK:
SIP:n	Set input peak digits where n = 0 (+12dB) n = 1 (+18dB) n = 2 (+24dB)	ACK: or NAK:
SMD:n	Set mode where n = 0 (normal mode) n = 1 (extended mode)	ACK: or NAK:
SOG:n	Set output gain where n = 0 (-6dB) n = 1 (-5dB) n = 2 (-4dB) n = 3 (-3dB) n = 4 (-2dB) n = 5 (-1dB) n = 6 (-0dB) n = 7 (+1dB) n = 8 (+2dB) n = 9 (+3dB) n = 10 (+4dB) n = 11 (+5dB) n = 12 (+6dB) n = 13 (+7dB) n = 14 (+8dB) n = 15 (+9dB) n = 16 (+10dB) n = 17 (+11dB) n = 18 (+12dB) n = 19 (+13dB) n = 20 (+14dB) n = 21 (+15dB) n = 22 (+16dB) n = 23 (+17dB) n = 24 (+18dB) n = 25 (+19dB) n = 26 (+20dB)	ACK: or NAK:



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SYNCHRONISERS, DELAYS & SILENCE DETECTORS - RB-DS2

Command	Description	Response
	n = 27 (+21dB) n = 28 (+22dB) n = 29 (+23dB) n = 30 (+24dB)	
SRL:n	Set remote setup loading where n = 0 (off) n = 1 (on)	ACK: or NAK:
SRQ:	Status request where i = bypass status J = left/stereo delay k = right delay l = control lock status m = analogue mute status n = channel setting o = digital mute status p = delay units setting q = format setting r = input volume setting s = mode setting t = output volume setting u = remote loading setting v = source setting w = system date ddmmyy x = sample rate setting y = system time hhmmss z = sample width setting	ACK: or NAK:
SSC:n	Set source where n = 0 (analogue) n = 1 (analogue sync) n = 2 (digital)	ACK: or NAK:
SSD:ddmmyy	Set system date where dd = day (1-31) mm = month (1-12) yy = year (00-99)	ACK: or NAK:

Command	Description	Response
SSR:n	Set sample rate where n = 0 (32kHz) n = 1 (44.1kHz) n = 2 (48kHz) n = 3 (64kHz) n = 4 (88.2kHz) n = 5 (96kHz)	ACK: or NAK:
SST:hmmss	Set system time where hh = hours (0-23) mm = minutes (0-59) ss = seconds (0-59)	ACK: or NAK:
SSW:n	Set sample width where n = 0 (16bit) n = 1 (24bit)	ACK: or NAK:
UID:	**Unit ID request	UID:DS2
UNL:	Disable control lock	ACK: or NAK:
VER:	Version request where y.yyy = firmware version zzzz = display version	VER:Vy.yyy,Vzzzz

** Supported in Bootstrap mode

Upgrading Firmware

Occasionally, it may be necessary to upgrade the firmware on the delay synchroniser to add new functionality and fix software bugs. New firmware updates will be made available from time to time on the Sonifex website. Visit www.sonifex.co.uk for details.

The firmware is upgraded using the Serial Control Interface - SCI. This PC software downloads the new firmware and initiates the upgrade process. To upgrade the firmware, select the required firmware file in the SCI upgrade firmware section. The delay synchroniser will automatically reboot and enter the bootstrap mode prior to the transfer of the firmware file. The SCI software downloads the firmware to internal memory, and then erases the current firmware before programming the update. When the firmware is complete, the unit reboots with the new firmware.

Please note: Firmware files can take several minutes to transfer to the delay synchroniser at lower baud rates. To speed up the process, select a higher baud rate prior to transferring the new firmware.



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Technical Specifications RB-DS2

Audio Specification

Maximum Input Level:	+27dBu
Input Impedance:	> 10k Ω bridging
Input Levels:	+24dBu / +18dBu / +12dBu for FSD
Gain Range:	Adjustable 3dB loss to 3dB gain (L & R adjust)
Signal to Noise:	Better than -101dBFS (RMS A-weighted at 24bit)
Dynamic Range:	> 110dB
Distortion & Noise:	> 96dB THD + N at 1kHz
Maximum Output Level:	+24dBu
Output Impedance:	< 50 Ω
Dynamic Range:	> 100dB
Gain Range:	-6dBu to +24dBu output level, ref FSD
Sampling Frequency:	32kHz, 44.1kHz, 48kHz, 64kHz, 88.2kHz or 96kHz
Sample Width:	16bit or 24bit
Channels:	Stereo, Dual Mono or Mono
Format (Fields & Frames):	25, 24 or 30 frames per second or 30 drop frame
Delay Units:	Samples, Fields, Frames or Time (msecs or hh:mm:ss in ext. mode) & Metres (normal mode only)
Minimum Delays	30 samples at 32kHz, 44.1kHz and 48kHz 20 samples at 64kHz, 88.2kHz, 96kHz
Maximum Delays:	See Figures 7-7, 7-8 and 7-9.

Analogue/Digital Connections

Analogue Inputs:	2 x XLR 3 pin female (balanced) (L & R)
Analogue Outputs:	2 x XLR 3 pin male (balanced) (L & R)
Digital Inputs:	1 x AES/EBU XLR 3 pin female
Digital Outputs:	1 x AES/EBU XLR 3 pin male

Additional Connections

Mains Input:	IEC Power Plug, 100-240 VAC, 47-63Hz, fused 60W peak, 30W average
Fuse Rating:	Anti-surge fuse 2A 20 x 5mm
Serial Communications Port:	9-way D-type socket
Remote Input/Output Port:	15-way D-type socket
Memory Expansion:	Internal Compact Flash™ storage card slot

Operational Controls

Display:	Vacuum fluorescent display
System Navigation:	Rotary selector with integral push-switch
Audio Bypass:	Via push-switch

Physical Specifications

Dimensions (Raw)	48cm (W) x 15.8cm (D *) x 4.2cm (H) (1U) 19" (W) x 6.2" (D*) x 1.7" (H) (1U)
Dimensions (Boxed)	59cm (W) x 27.4cm (D*) x 10.8cm (H) 23.2" (W) x 10.8" (D*) x 4.3" (H)
Weight	Nett: 1.6kg Gross: 2.2kg Nett: 3.5lbs Gross: 4.8lbs

* Note that this product is deeper than standard Redboxes.



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ON-BOARD		32 kHz		44.1 kHz		48 kHz	
		16 bit	24 bit	16 bit	24 bit	16 bit	24 bit
Samples		2058240	1029120	2058240	1029120	2058240	1029120
Fields	25fps	3216	1608	2333	1166	2144	1072
	24fps	3087	1543	2240	1120	2058	1029
	30fps	3859	1929	2800	1400	2572	1286
	30df	3855	1927	2797	1398	2570	1285
Frames	25fps	1608	804	1166	583	1072	536
	24fps	1543	771	1120	560	1029	514
	30fps	1929	964	1400	700	1286	643
	30df	1927	963	1398	699	1285	642
Time (ms)		64320	32160	46672	23336	42880	21440
Metres		22080	11040	16020	8010	14720	7360

ON-BOARD		64 kHz		88.2 kHz		96 kHz	
		16 bit	24 bit	16 bit	24 bit	16 bit	24 bit
Samples		2058240	1029120	2058240	1029120	2058240	1029120
Fields	25fps	1608	804	1166	583	1072	536
	24fps	1543	771	1120	560	1029	514
	30fps	1929	964	1400	700	1286	643
	30df	1927	963	1398	699	1285	642
Frames	25fps	804	402	583	291	536	268
	24fps	771	385	560	280	514	257
	30fps	964	482	700	350	643	321
	30df	963	481	699	349	642	321
Time (ms)		32160	16080	23336	11668	21440	10720
Metres		11040	5520	8010	4000	7360	3680

Fig 7-7: Maximum Delays in Normal Mode (Using On-Board Memory Only)

Note: 1. The delays shown are for a stereo, or dual mono, signal.
 2. The delay times shown can be doubled for a mono signal.



128MB		32 kHz		44.1 kHz		48 kHz	
		16 bit	24 bit	16 bit	24 bit	16 bit	24 bit
Samples		32878592	16439296	32878592	16439296	32878592	16439296
Fields	25fps	51372	25686	37277	18638	34248	17124
	24fps	49317	24658	35786	17893	32878	16439
	30fps	61647	30823	44732	22366	41098	20549
	30df	61585	30792	44688	22344	41057	20528
Frames	25fps	25686	12843	18638	9319	17124	8562
	24fps	24658	12329	17893	8946	16439	8219
	30fps	30823	15411	22366	11183	20549	10274
	30df	30792	15396	22344	11172	20528	10264
Time (hh:mm:ss)		00:17:07	00:08:33	00:12:25	00:06:12	00:11:24	00:05:42

128MB		64 kHz		88.2 kHz		96 kHz	
		16 bit	24 bit	16 bit	24 bit	16 bit	24 bit
Samples		32878592	16439296	32878592	16439296	32878592	16439296
Fields	25fps	25686	12843	18638	9319	17124	8562
	24fps	24658	12329	17893	8946	16439	8219
	30fps	30823	15411	22366	11183	20549	10274
	30df	30792	15396	22344	11172	20528	10264
Frames	25fps	12843	6421	9319	4659	8562	4281
	24fps	12329	6164	8946	4473	8219	4109
	30fps	15411	7705	11183	5591	10274	5137
	30df	15396	7698	11172	5586	10264	5132
Time (hh:mm:ss)		00:08:33	00:04:16	00:06:12	00:03:06	00:05:42	00:02:51

Fig 7-8: Maximum Delays in Extended Mode with 128 MB Compact Flash™ Memory Card Installed

Note:

1. A maximum of 2GB can be installed.
2. In Extended Mode, the Metres selection is disabled.
3. The delays shown are for a stereo signal; in Extended Mode dual mono operation is not available.
4. The delay times shown can be doubled for a mono signal.
5. To operate correctly, the flashcard must support PIO Mode 4, or higher, such as the SanDisk Standard CF cards.

Opening the RB-DS2 to Add a Compact Flash Card

Warning: The power must be switched off at the supply and the power lead disconnected before opening the unit. Removal of the cover can expose dangerous voltages.

1. Remove the 4 screws in the corners of the rear panel.
2. Remove the 4 screws on the top and bottom panels which hold the rear panel in place (2 on the top and 2 on the bottom).
3. Remove the screw on the front panel immediately to the left of the Bypass button.

The rear panel and main PCB will slide backwards out of the metal chassis giving you access to the Compact Flash slot. To put the unit back together, slide the PCB back into the chassis and refit the screws in reverse order.

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1GB		32 kHz		44.1 kHz		48 kHz	
		16 bit	24 bit	16 bit	24 bit	16 bit	24 bit
Samples		261468160	130734080	261468160	130734080	261468160	130734080
Fields	25fps	408544	204272	296449	148224	272362	136181
	24fps	392202	196101	284591	142295	261468	130734
	30fps	490252	245126	355738	177869	326835	163417
	30df	489762	244881	355383	177691	326508	163254
Frames	25fps	204272	102136	148224	74112	136181	68090
	24fps	196101	98050	142295	71147	130734	65367
	30fps	245126	122563	177869	88934	163417	81708
	30df	244881	122440	177691	88845	163254	81627
Time (hh:mm:ss)		02:16:10	01:08:05	01:38:48	00:49:24	01:30:47	00:45:23

1GB		64 kHz		88.2 kHz		96 kHz	
		16 bit	24 bit	16 bit	24 bit	16 bit	24 bit
Samples		261468160	130734080	261468160	130734080	261468160	130734080
Fields	25fps	204272	102136	148224	74112	136181	68090
	24fps	196101	98050	142295	71147	130734	65367
	30fps	245126	122563	177869	88932	163417	81708
	30df	244881	122440	177691	88845	163254	81627
Frames		102136	51068	74112	37056	68090	34045
	25fps	98050	49025	71147	35573	65367	32683
	24fps	122563	61281	88934	44467	81708	40854
	30df	122440	61220	88845	44422	81627	40813
Time (hh:mm:ss)		01:08:05	00:34:02	00:49:24	00:24:42	00:45:23	00:22:41

Fig 7-9: Maximum Delays in Extended Mode with 1 GB Compact Flash™ Memory Card Installed.

Note:

1. A maximum of 2GB can be installed, doubling the above delay times.
2. In Extended Mode, the Metres selection is disabled.
3. The delays shown are for a stereo signal; in Extended Mode dual mono operation is not available.
4. The delay times shown can be doubled for a mono signal.
5. To operate correctly, the flashcard must support PIO Mode 4, or higher, such as the SanDisk Standard CF cards.

RB-PD2 Stereo Profanity Delay

Introduction



Fig 8-1: RB-PD2 Front Panel

The RB-PD2 is a stereo audio profanity delay used for live broadcast programs to prevent unwanted or obscene material from being transmitted. It features an automatic audio stretch algorithm that allows between 2 and 55 (*) seconds of delay to be built up live whilst "on air", whilst maintaining the correct pitch. Higher delay durations are available at lower sample rates and the speed at which the algorithm builds and exits the delay is also adjustable. The delay can also be acquired whilst playing a pre-selected audio file on a Compact Flash™ memory card. When the program is complete, the audio stretch algorithm seamlessly reduces the delay to zero.

The RB-PD2 has both balanced analogue and AES/EBU digital audio inputs and outputs on 3 pin XLR connectors and provides sample rates up to 48kHz at 24 bit. It can act as a combined A/D and D/A unit meaning that analogue inputs can be output as AES/EBU or vice-versa.

The delay can be initiated by pressing the BUILD DELAY button on the front panel. A front panel display shows the delay amount being built-up, up to the amount initially selected.

There are several ways to make sure that any unwanted material is removed from the audio at the outputs. A COUGH function, activated from a dedicated front panel button, allows locally generated sounds being presented at the inputs, such as the presenter coughing or equipment switching noises, to be discarded.

The DUMP function, which is also activated from a front panel button, has 2 different modes. The first DUMP mode removes a section of audio that has already been buffered, by a pre-selected amount. The second DUMP mode plays a pre-selected audio file on the Compact Flash™ memory card. When the file has finished playing, the delay is then equal to the duration of the file. The DUMP button can be used multiple times to use up the built-up delay and once used, the unit automatically starts to rebuild the original delay time. As a last resort, all the buffered audio can be discarded by pressing and holding the DUMP button which activates the DROP function.

At the end of a radio show when you want to broadcast live, the delay can be ramped down by pressing the front panel EXIT DELAY button.

A dedicated record mode allows audio presented at either the analogue or digital inputs to be recorded to a linear WAV file on a Compact Flash™ memory card. Additionally, the card format used is PC readable, allowing pre-recorded linear WAV files to be transferred easily from a PC or other such device.

(* at 32kHz 16bit)



A front panel blue vacuum fluorescent display with rotary controller is used for selecting the various settings of the profanity delay, which include the start delay and dump modes, safe delay build/exit speed, period, source (analogue or digital), sample rate and sample bit width as well as the required delay time. The current delay value, in seconds, is permanently displayed as is the current status of the unit. Additionally, input peak digits can be selected from +12dBu, +18dBu and +24dBu for FSD and two left and right pre-set potentiometers on the rear panel allow the input gain range to be altered by ± 3 dB around the selected peak digits.

The analogue output gain range can be altered in software from -6dBu to +24dBu output level, ref FSD. Both analogue and digital outputs can be separately muted and a front panel Bypass button disengages electro-mechanical relays to divert both analogue and digital inputs to their outputs. This is also disengaged automatically when a power-fail occurs.

The RB-PD2 features a remote port supplying 8 inputs and 6 outputs, all of which are freely assignable. The inputs can be used to trigger any of the unit's functions such as build delay, activate cough or enter record mode and start a new recording. The outputs can provide external signaling to indicate when certain events have occurred such as the delay reaching the required value or the outputs being muted. From V1.06 firmware onwards, the RB-PD2 can delay remote inputs 1-6 to the remote outputs 1-6, so that remote control commands can also be synchronised with the same delay as the RB-PD2.

Because playback from a Compact Flash™ card can be triggered remotely, the RB-PD2 can also be used at transmitter sites to play an emergency audio file via GPI in the event of silence detection.

The front panel controls can be locked out for situations where remote control is being used to run the unit, or where physical security is required.

System Block Diagram

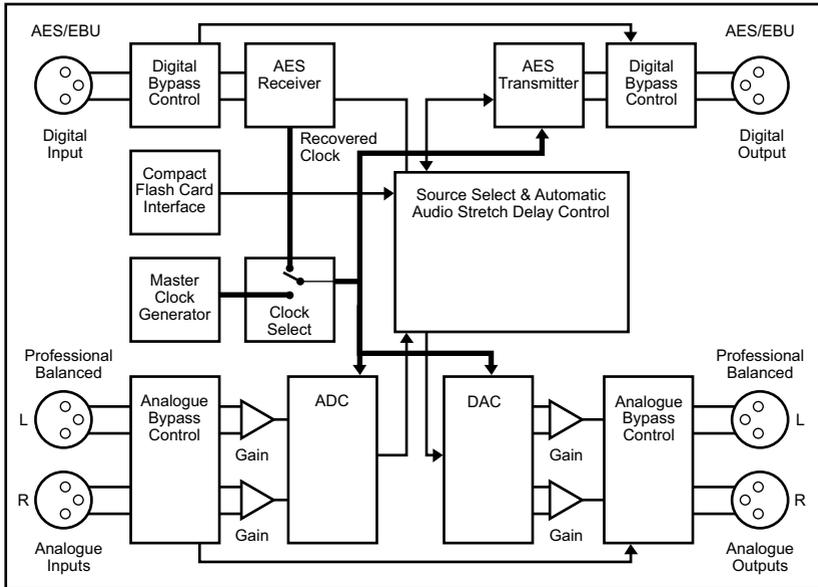


Fig 8-22: RB-PD2 System Block Diagram

Quickstart Guide

This quick start guide briefly explains the steps required to get the RB-PD2 up and running if you just want to get the unit on-air quickly.

There are 2 different methods of operation for using the RB-PD2. The first is to use the **Audio Stretch** algorithms to build up a delay time which can then be 'dumped' in audio segment lengths that you define. The second is to **Play a File** from a Compact Flash card which can be inserted into the slot in the front panel.

For both methods, there are 2 menu options that need to be set prior to operating the delay: **Start Mode** and **Dump Mode**. Start Mode defines how the RB-PD2 initially builds up the delay period and the Dump Mode defines what happens when the DUMP/DROP button is pressed. Other options such as audio source and sample rate are not discussed here as they do not affect the operation of the unit.

To change options in the main menu, use the SELECT MENU rotary control to the right of the front panel; press and hold it for 1.5 seconds to enter the menu, turn the control to highlight the required menu item and press to select it.

The last item shown in each menu is >CLOSE. Selecting close exits the system menu and returns to the main display. In each sub-menu, the penultimate item shown is <BACK. Selecting <BACK displays the previous menu.



Using the Audio Stretch Method of Operation

1. [MAIN MENU > DELAY OPTIONS > START MODE > BUILD DELAY]. Select the delay START MODE as BUILD DELAY. BUILD DELAY start mode uses an audio stretch algorithm to gradually increase the delay from zero to the selected duration.
2. [MAIN MENU > DELAY OPTIONS > BUILD/EXIT SPEED]. The BUILD/EXIT SPEED sets the rate at which the audio is altered by the audio stretch algorithm. There are 16 speeds available and each setting shows the approximate time required to build 10 seconds of delay.
3. [MAIN MENU > DELAY OPTIONS > SAFE PERIOD]. The SAFE PERIOD is a delay threshold that you can set to define when it is safe for you to use the DUMP/DROP button. When the current delay is less than the SAFE PERIOD, the "SECS" label on the main screen flashes. The Safe Period can be set from 1 to 10 seconds in units of 0.1 seconds.
4. [MAIN MENU > DUMP MODE > DUMP BUFFER]. Select the DUMP MODE which defines what happens when you press the DUMP/DROP button. The DUMP BUFFER mode discards the most recently buffered section of audio when the dump button is pressed so select this.
5. [MAIN MENU > DUMP MODE > BUFFER LENGTH]. Further down the same DUMP MODE menu, select the BUFFER LENGTH. This option selects the length of audio dumped from the delay when the DUMP/DROP button is pressed. It can be set from 1 to 10 seconds in units of 0.1 seconds.
6. If you now select CLOSE and exit out of the menu, you can select the delay for the Build Delay Start Mode. The Delay Time is set from the main screen. Briefly press and then turn the SELECT DELAY rotary control to select a total delay value. Press again to set the value.

The RB-PD2 is now setup for use.

7. To initiate the delay at the start of a programme, press the BUILD DELAY button. The delay between audio at the inputs and outputs will increase at the selected build speed until the selected delay is acquired. Whilst acquiring the delay, the delay time will flash when it is less than the SAFE PERIOD value. Once the delay has been acquired, it will remain constant.
8. To discard a section of unwanted audio (of a duration defined by the BUFFER LENGTH) that has already been internally buffered (e.g. a telephone caller swearing), press the DUMP/DROP button. The presenter should continue with the programme regardless of the dump mode currently selected to avoid periods of silence being output.
9. To discard a section of unwanted audio that is about to be presented to the audio inputs (i.e. the presenter coughing), press and hold the COUGH button for the duration of the unwanted noise. The current delay will count down whilst the button is held.
10. To discard ALL of the buffered audio and mute the audio outputs, press and hold the DUMP/DROP button to initiate the DROP function. The audio outputs will remain muted until the button is released. At this point, if the unit was exiting the delay prior to the drop

function, the delay will remain at zero. Otherwise BUILD DELAY mode will be initiated.

11. At the end of a programme, press the EXIT DELAY button to begin ramping the delay to zero.

Using the Play File Method of Operation

1. [MAIN MENU > DELAY OPTIONS > START MODE> PLAY FILE]. Select the delay START MODE as PLAY FILE. This start mode plays an audio file (linear PCM) stored on a Compact Flash™ memory card inserted in the front panel card slot.
2. [MAIN MENU > DELAY OPTIONS > START MODE > SELECT PLAY FILE]. The SELECT PLAY FILE option is used to choose the pre-recorded .wav file (stored on the CF card) which will be played out when the delay is first being built up. Note that the sample rate and bit depth of the RB-PD2 output must be the same as the recording on the CF card (see page 83: Sample Rate & Sample Width). Also, you can make a recording directly on the RB-PD2 (see page 83: Record).
3. [MAIN MENU > DUMP MODE > PLAY FILE]. Select the DUMP MODE which defines what happens when you press the DUMP/DROP button. The PLAY FILE mode plays the audio file selected by the SELECT PLAY FILE option below.
4. [MAIN MENU > DUMP MODE > SELECT PLAY FILE]. The SELECT PLAY FILE option selects the audio file to be played when the Play File dump mode is selected. Select the required dump Play File from the list of available files.
5. Select CLOSE and exit out of the menu

The RB-PD2 is now setup for use.

6. To initiate the delay at the start of a programme, press the BUILD DELAY button and begin your radio programme immediately. When the BUILD DELAY button is pressed, the selected audio file is played and the unit immediately begins to buffer the audio presented at the inputs. When the file has finished playing, the output of the RB-PD2 reverts to the buffered audio and the length of the delay is equal to the duration of the audio file.
7. To discard a section of unwanted audio that has already been internally buffered (e.g. a telephone caller swearing), press the DUMP/DROP button. When the DUMP/DROP button is pressed, all the internally buffered audio is discarded, the selected audio file is played and the unit immediately begins to buffer the audio presented at the inputs. When the file has finished playing, the output of the RB-PD2 reverts to the buffered audio and the length of the delay is equal to the duration of the audio file. The presenter should continue with the programme regardless of the dump mode currently selected to avoid periods of silence being output.
8. To discard a section of unwanted audio that is about to be presented to the audio inputs (i.e. the presenter coughing), press and hold the COUGH button for the duration of the unwanted noise. The current delay will count down whilst the button is held. And once released, the delay will be built up again using the audio stretch method.



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SYNCHRONISERS, DELAYS & SILENCE DETECTORS



9. To discard ALL of the buffered audio and mute the audio outputs, press and hold the DUMP/DROP button to initiate the DROP function. The audio outputs will remain muted until the button is released. At this point, if the unit was exiting the delay prior to the drop function, the delay will remain at zero. Otherwise the delay will be built up again using the audio stretch method.

10. At the end of a programme, press the EXIT DELAY button to begin ramping the delay to zero.

Front Panel Controls

All of the RB-PD2's configurations can be accessed with the single front panel mounted rotary control in conjunction with the vacuum fluorescent display.



Fig 8-3: RB-PD2 Rotary Control Operation

Setting the Delay

The target delay of the RB-PD2 is dependent on the selected Start Mode. There are 2 start modes; Build Delay and Play File.

Build Delay Mode

In Build Delay mode, the delay will be acquired using the automatic audio stretch algorithm until the target delay has been built up. The rate at which the delay is built up depends on the current audio stretch algorithm speed setting.

To adjust the delay, lightly press the rotary control to enable the delay edit mode. The decimal point on the selected delay value on the left hand side of the main screen will flash to indicate that delay edit mode has been entered. The value of the delay can now be changed by turning the rotary control. Once the required delay value has been set, press the rotary control again to return to normal operation. The delay edit mode is automatically cancelled if no change is made to the delay within a period of 2.5 seconds.

If the current delay is not zero, i.e. a certain amount of delay has already built up, changes to the delay are implemented dynamically. For example, if the current delay is 7.0 seconds and a new delay of 5.5 seconds is selected, the delay will automatically decrease to the new value and the status line will show "DECREASING DELAY".

Play File Mode

In Play File mode, an audio file stored on a Compact Flash™ memory card is played whilst simultaneously buffering the audio at the selected audio input source. Once the audio file has finished playing, the buffered audio is output and the delay is set to the duration of the file just played.

In this start mode, the selected delay value on the left hand side of the main screen shows the length of the selected start play file. The selected delay cannot be edited until the start file has finished playing.





The BUILD DELAY Button

The BUILD DELAY button activates the build delay mode when the audio stretch algorithm starts to silently build up audio delay. This mode can only be entered if the unit is at zero delay or in exit delay mode. The audio stretch algorithm builds the delay based on the current build/exit speed (the Delay Build/Exit Speed – see page 81).

The actual operation will depend on the current delay and the currently selected start mode. If the current delay is zero, the delay will build using the selected start mode. However, if the current delay is greater than zero i.e. some delay has already been acquired, the automatic audio stretch algorithm is activated to build the delay to the selected value.

The BUILD DELAY button is illuminated red whilst the delay is being acquired and the status line on the display will show either “PLAYING DELAY START FILE” or “BUILDING DELAY” depending on which build mode is active. Once the delay reaches the pre-selected value, the BUILD DELAY button is illuminated green and the status line shows “DELAY OK”.



Fig 8-4: RB-PD2 Front Panel Buttons

This mode can only be activated if the unit is currently in build delay mode or is at the currently selected delay (shown via the status line on the display).

The EXIT DELAY button is illuminated red whilst the delay is being reduced and the status line will show “EXITING DELAY”. Once the delay has been reduced to zero, the EXIT DELAY button will illuminate green for 1.5 seconds and the status line on the display will show “ZERO DELAY”.

The COUGH Button

The COUGH button activates the cough function which allows unwanted audio that is about to be presented to the audio inputs to be discarded. The cough function is only active whilst the COUGH button is pressed.

The COUGH button is illuminated red and the status line on the display will show “COUGH ACTIVE”. The current delay shown on the display will count down whilst the button is pressed. If the COUGH button continues to be pressed and the current delay counts down to 0, the audio outputs will be muted until the COUGH button is released.

When the unwanted noise has passed, the COUGH button can be released and the status line will indicate the current state. If the exit delay mode was active prior to the cough function, the status line will show “EXITING DELAY” unless the delay has reached 0, in which case the status line will show “ZERO DELAY”. Otherwise the build delay mode will re-engage and status line will show “BUILDING DELAY”.

The EXIT DELAY Button

The EXIT DELAY button activates the exit delay mode when the audio delay is reduced to zero. The delay ramps down at the rate specified by the current build/exit speed (the Delay Build/Exit Speed – see page 81).

The Audio BYPASS Button

The BYPASS button enables the delay bypass mode, which internally connects the inputs to outputs in both the analogue and digital domains. The BYPASS button is illuminated red when the delay bypass mode is enabled and the analogue and digital icons on the display will change to . When power is removed from the unit, the bypass mode is automatically enabled.

The DUMP/DROP Button

Pressing the DUMP/DROP button activates the currently selected dump mode. The dump mode can be initiated at any time provided the current delay is not at 0. The button is illuminated red and the status line displays "DUMP ACTIVE". With the dump buffer mode selected, a pre-selected length (the Dump Buffer Length - see page 82) of most recent audio data is discarded from the internal audio buffer each time the dump button is pressed. If the current delay is less than the dump buffer length, the delay is set to 0. When the dump buffer function has completed, the delay will automatically start to build unless the exit delay mode was active prior to the dump being activated.

When the dump mode is set to play file, all the buffered audio is discarded and the selected audio file on the Compact Flash™ memory card is played. Audio is immediately buffered to create a new delay which will be equal to the duration of the audio file being played. When the file has finished playing the status display will show "DELAY OK".

Pressing and holding the DUMP/DROP button for 0.8 seconds, activates the DROP function. All of the currently acquired delay is discarded, the audio outputs are muted and the status display will show "DROP ACTIVE". When the DUMP/DROP button is released, its illumination is turned off, the audio outputs are un-muted and the status line indicates the current mode. If the exit delay mode was active prior to the drop function, the status line will show "ZERO DELAY". Otherwise the build delay mode will re-engage and status line will show "BUILDING DELAY".

System Reset

A single LED on the left side of the front panel indicates that power is applied. The system reset button is accessible through the small aperture to the right of the power LED.

Selecting the Main Menu

The RB-PD2's configuration options are accessed from the main system menu. To access the main system menu, press and hold the rotary control for at least 1.5 seconds. The menu structure is arranged in a multi-tiered format with several menu items providing access to sub-menus. To select a menu item, turn the rotary control until the required item is highlighted, then press the rotary control to select the item. Each configuration is shown as a list of available options. The currently selected option will be shown with a tick along side it. The last item shown in each menu is >CLOSE. Selecting close exits the system menu and returns to the main display. In each sub-menu, the penultimate item shown is <BACK. Selecting back displays the previous menu.





Rear Panel Connections



Fig 8-5: RB-PD2 Rear Panel

XLR Analogue Inputs (Left and Right)

The XLR 3 pin sockets used for the left and right channel analogue inputs are electronically balanced and have an impedance greater than 10k Ω bridging. Each XLR has the following connections:

Pin 1: Screen

Pin 2: Phase

Pin 3: Non-phase

Analogue Input Adjustment

The Input Peak Digits setting (see System Menu: Volume) sets the analogue input gain for full-scale digits in the digital domain. The preset potentiometers next to the analogue inputs allow an additional adjustment of this value by $\pm 3\text{dB}$.

AES/EBU Input

The digital input XLR 3 pin socket has an impedance of 110 Ω . It has the following connections:

Pin 1: Screen

Pin 2: Phase

Pin 3: Non-phase

The signals on this connector should meet the IEC 60968 specification.

XLR Analogue Outputs (Left and Right)

The XLR 3 pin output plugs are electronically balanced with an output impedance of less than 50 Ω . They have the following connections:

Pin 1: Screen

Pin 2: Phase

Pin 3: Non-phase

AES/EBU Output

The digital output XLR 3 pin plug has an impedance of 110 Ω . It has the following connections:

Pin 1: Screen

Pin 2: Phase

Pin 3: Non-phase

The signals on this connector comply with the IEC 60968 specification.

Remote Inputs and Outputs

The 15-way 'D' type socket connector presents 8 active low inputs and 6 open collector outputs which can be programmed to your requirements. The inputs are used to remotely activate a pre-assigned function and the outputs provide an indication that a pre-assigned state is currently active (see System Menu: Remotes).

The remote outputs 1-6 can also be optionally be made to follow the inputs 1-6 with the current programme delay inserted between actuation on the input and actuation on the output. This can be useful for timed events that need to account for the delay built up by the RB-PD2.

It has the following connections:

- Pin 1: Active low input 1
- Pin 2: Active low input 2
- Pin 3: Active low input 3
- Pin 4: Active low input 4
- Pin 5: Active low input 5
- Pin 6: Active low input 6
- Pin 7: Active low input 7
- Pin 8: Active low input 8
- Pin 9: Signal ground
- Pin 10: Open collector output 1
- Pin 11: Open collector output 2
- Pin 12: Open collector output 3
- Pin 13: Open collector output 4
- Pin 14: Open collector output 5
- Pin 15: Open collector output 6

RS232

The 9-way 'D' type socket connector allows direct connection to a serial port on a PC via a pin-to-pin cable. See the section on Serial Port Control for details of this interface. It has the following connections:

- Pin 1: N/C
- Pin 2: Transmit data (TxD)
- Pin 3: Receive data (RxD)
- Pin 4: N/C
- Pin 5: Ground
- Pin 6: N/C
- Pin 7: Clear To Send (CTS)
- Pin 8: Request To Send (RTS)
- Pin 9: N/C





The Default Screen

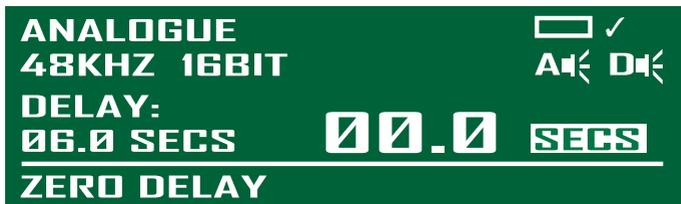


Fig 8-6: The Default Screen

The default screen shows the current selections for audio source, sample rate and sample bit width in the top left of the screen. The selected delay value is shown on the left while the current delay is shown in larger characters on the right. The "SECS" label shown next to the current delay will flash if the current delay is below the SAFE PERIOD.

The current status of the unit is displayed on the status line at the bottom of the screen. Also shown on the main screen is the mute status of the analogue and digital outputs, and the status of the Compact Flash™ memory card, if fitted. When a memory card is fitted, a small memory card symbol will appear above the analogue and digital audio output status icons. A tick next to the memory card symbol indicates the card is ready for use, whereas a cross indicates that there is an error.

All of the RB-PD2's settings are configured from the main system menu with the exception of the selected delay. The delay is adjusted directly from the main screen following the instructions on page 75.

The Main Menu

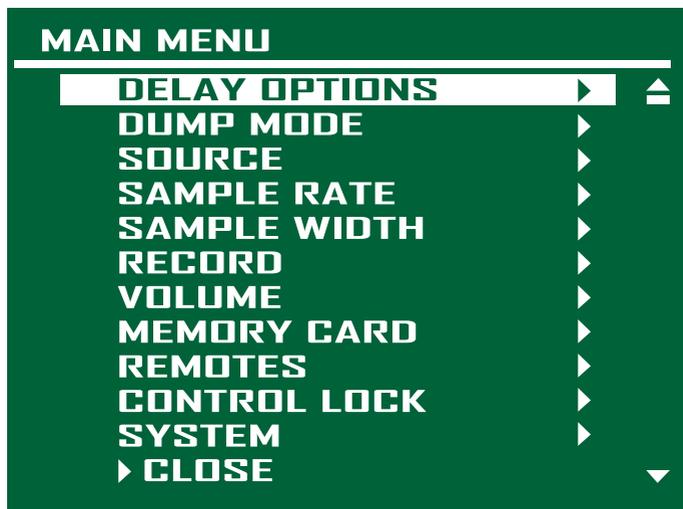


Fig 8-7: The Main Menu Screen

To access the main menu, press and hold the rotary control knob for 1.5 seconds. Instructions for navigating the menu structure are on page 77.

The available menu and sub menu items are described below together with the associated configuration options.

DELAY OPTIONS

This selection shows the DELAY OPTIONS sub-menu. The sub-menu items are START MODE, BUILD/EXIT SPEED and SAFE PERIOD.

START MODE

This option defines how the delay is built-up. From the DELAY OPTIONS sub-menu select START MODE. The available options are BUILD DELAY, PLAY FILE and SELECT PLAY FILE.

- Build Delay – With this option selected, the delay is built up using the automatic audio stretch algorithm. The length of the delay is selected via the main screen.
- Play File – With this option selected, the delay is built up whilst playing a pre-selected audio file on a Compact Flash™ memory card. The length of the delay will be equal to the duration of the selected audio file.
- Select Play File – This option selects the audio file to be played when the Play File start mode is selected. It shows all available audio files on the Compact Flash™ memory card which match the current audio configuration settings (sample rate and sample width). If the card has sub-directories defined, these are also shown. To select a file, turn the rotary control and then press when the required file is highlighted. To move to a sub-directory, press the rotary control with the highlight over the required directory name. Whilst viewing a sub-directory, the top item in the list is UP which, when selected, returns to the parent directory.

The default start mode is Build Delay.

BUILD/EXIT SPEED

This option selects the rate at which the audio stretch algorithm builds and exits the delay. There are 16 speed settings and each is shown with the approximate time required to build/exit a 10 second delay. These settings are 5mins 30secs, 5mins 00secs, 4mins 30secs, 4mins 00secs, 3mins 30secs, 3mins 00secs, 2mins 30secs, 2mins 20secs, 2mins 10secs, 2mins 00secs, 1min 50secs, 1min 40secs, 1min 30secs, 1min 20secs, 1min 10secs and 1min 00secs. When building from zero delay, the audio stretch algorithm is designed to start at a slow speed and increase to the configured speed. This helps to avoid sudden tempo changes. Likewise, when exiting the delay and as the current delay approaches zero, the speed is decreased. Consequently, the times shown for each speed setting will be slightly less for subsequent 10 seconds periods. For example, if the delay is set to 20 seconds, and the speed selected is 3mins 00secs, the first 10 seconds of delay will be acquired in the time specified, while the remaining 10 seconds will take slightly less time. The build/exit speed can only be changed when the current delay is zero.

The default build/exit speed is 2mins 30secs.





SAFE PERIOD

This option selects the safe period duration, i.e., amount of delay that needs to be built up before the DUMP button can be used without muting the outputs. To edit the safe period, turn the rotary control until the highlight is over the safe period value. Press the rotary control to enter edit mode. The highlight changes to a flashing line under the current value. Now turn the rotary control to select a new safe period between 1 and 10 seconds in units of 0.1 seconds. When the required value is shown, press the rotary control again to leave edit mode. Now turn the rotary control to select either OK which saves the new value, or CANCEL which discards the changes.

The "SECS" label on the right hand side of the main screen, next to the current delay value, flashes when the current delay is less than the safe period. If the safe period is set to the same duration as the dump length, it provides a quick visual indication of when the dump can be used without the outputs muting, as would be the case if the dump period is longer than the current delay.

The default safe period is 2.0 seconds.

DUMP MODE

This option selects the mode of operation when the DUMP button is pressed. From the main system menu select DUMP MODE. The available options are DUMP BUFFER, PLAY FILE, BUFFER LENGTH and SELECT PLAY FILE.

- **Dump Buffer** – This option discards a pre-selected amount of audio from the internal delay buffer. For example, if the current delay is 10 seconds and the dump buffer length is 4 seconds, when the dump is activated, the position at which the audio at the selected input is stored in the internal memory buffer is moved back by 4 seconds. This effectively overwrites the last four seconds of audio and the delay will be reduced to 6 seconds. The audio stretch algorithm will automatically activate after the dump to return the delay to the selected duration.
- **Play File** – This option plays a pre-selected audio file on a Compact Flash™ memory card and discards all of the currently buffered audio. Delay acquisition begins again immediately and the new delay length is equal to the duration of the file played.
- **Buffer Length** – This option selects the dump buffer length when the Dump Mode is set to Dump Buffer. To edit the buffer length, turn the rotary control until the highlight is over the dump buffer value. Press the rotary control to enter edit mode. The highlight changes to a flashing line under the current value. Now turn the rotary control to select a new dump buffer length between 1 and 10 seconds in units of 0.1 seconds. When the required value is shown, press the rotary control again to leave edit mode. Now turn the rotary control to select either OK which saves the new value, or CANCEL which discards the changes.
- **Select Play File** – This option selects the audio file to be played when the Play File dump mode is selected. It shows all available audio files on the Compact Flash™ memory card which match the current audio configuration settings (sample rate and sample width). If the card has sub-directories defined, these are also shown.

To select a file, turn the rotary control and then press when the required file is highlighted. To move to a sub-directory, press the rotary control with the highlight over the required directory name. Whilst viewing a sub-directory, the top item in the list is UP which, when selected, returns to the parent directory.

The default dump mode is dump buffer with a buffer length of 2.0 seconds.

SOURCE

This option selects which input is used for the audio delay process. From the main system menu select SOURCE. The available options are ANALOGUE, ANALOGUE SYNC and DIGITAL. The delayed audio appears at both the analogue and digital outputs regardless of the input source selected.

- Analogue – This selects the audio at the analogue inputs as the source for the delay. The input is sampled and delayed using the selected sample rate generated from an internal master clock. The digital input is not used.
- Analogue Sync – This selects the audio at the analogue inputs as the source for the delay. The input is synchronized to the digital audio sync signal at the digital input and the analogue and digital audio outputs will be muted until a valid sync signal is detected. Once a valid sync signal has been received, audio will appear at the outputs and the delay can then be started. Any subsequent loss of sync signal will force the unit to revert to an internally generated master clock rate as near as possible to the last sync signal received.
- Digital – This selects the audio at the digital input as the source for the delay. With no digital signal applied, the analogue and digital audio outputs will be muted. Once a valid digital input is received, the unit locks to this signal and audio will appear at the outputs. The analogue inputs are not used.

The default source is analogue.

SAMPLE RATE

This option selects the sample rate used when the input source is set to Analogue. From the main system menu select SAMPLE RATE. The available options are 32kHz, 44.1kHz and 48kHz. The sample rate can only be changed when the current delay is zero. The default sample rate is 48kHz.

SAMPLE WIDTH

This option selects the sample size used in the analogue to digital and digital to analogue conversion. From the main system menu select SAMPLE WIDTH. The available options are 16BIT and 24BIT. The sample width can only be changed when the current delay is zero. The default sample width is 24bits.

RECORD

This selection shows the Record sub-menu which allows you to make a recording onto a CF card. Note that pre-recorded .wav files can also be played by selecting DUMP MODE>Select Play File.

The sub-menu items are NEW RECORDING, SOURCE, SAMPLE RATE, SAMPLE WIDTH and INITIALS.





New Recording

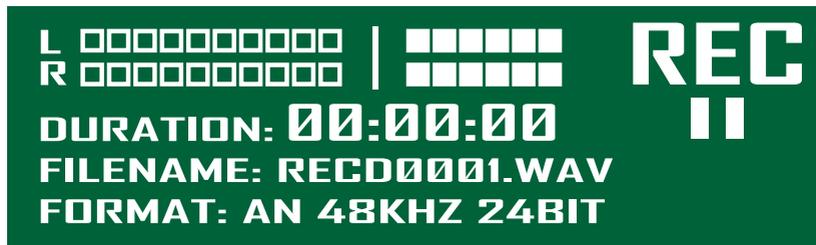


Fig 8-8: The Record Screen

This option enters the record screen provided a valid Compact Flash™ memory card is inserted in the front panel Compact Flash™ memory card slot. An error message will be displayed if no memory card is present, or a new record audio file cannot be initialized, or the maximum number of recordings already exists in the current directory, or insufficient space exists.

The record screen shows the filename that the audio will be recorded to as well as the record parameters that are currently set. The PPM display at the top of the record screen gives a visual indication of the audio levels for the selected source with the line on the scale indicating the 0dB point. In record mode, the DUMP/DROP button doubles as a control for the record process. To start recording, press the button once. The button illuminates, the record pause icon is cleared and the recording starts with the on-screen duration indicating the length of the recording. To stop the recording, press the DUMP/DROP button again. The new audio file is saved, the record mode is exited and the main default screen is displayed.

To exit the record screen prior to starting record, press and hold the rotary control for 1.5 seconds. Position the highlight over EXIT RECORD and press the rotary control.

Source

This option selects which input is used for the record process. From the record sub-menu select SOURCE. The available options are ANALOGUE, ANALOGUE SYNC and DIGITAL.

The analogue and digital outputs are muted whilst in record mode.

- Analogue – This selects the audio at the analogue inputs as the source for recording. The input is sampled using the selected record sample rate generated from an internal master clock. The digital input is not used.
- Analogue Sync – This selects the audio at the analogue inputs as the source for recording. The input is synchronized to the digital audio sync signal at the digital input.
- Digital – This selects the audio at the digital input as the source for recording. The analogue inputs are not used. The default record source is analogue.

Sample Rate

This option selects the sample rate used for recording when the input source is set to Analogue. From the record sub-menu select SAMPLE RATE. The available options are 32kHz, 44.1kHz and 48kHz. The default record sample rate is 48kHz.

Sample Width

This option selects the sample size used for recording in the analogue to digital conversion. From the record sub-menu select SAMPLE WIDTH. The available options are 16BIT and 24BIT. The default record sample width is 24bits.

Initials

This option sets the first 4 characters of the record filename that is automatically generated when entering record. The filename consists of these 4 characters followed by a 4 digit number which automatically increments as new recordings are made. The file extension for all recordings is "WAV". For example, with the initials set to "RECD", the first record filename will be "RECD0001.WAV".

To set the initials, turn the rotary control to highlight a character and then press the rotary control to enter edit mode. The highlight changes to a flashing line under the selected character. Now turn the rotary control to select a new character in that position. When the required character is shown, press and turn the rotary control to select a new edit position. When the required changes have been made, select either OK to save the new initials, or CANCEL to discard the changes. The default initials are "RECD".

VOLUME

This selection shows the Volume sub-menu where you can set input and output gains and also mute the outputs. The sub-menu items are INPUT PK DIGITS, OUTPUT GAIN, ANALOGUE MUTE and DIGITAL MUTE.

Input Peak Digits

This option sets the full-scale dB input amplitude that will give full-scale digits in the digital domain. From the VOLUME sub-menu select INPUT PK DIGITS. The available options are +12 dB, +18 dB and +24 dB. The default value for input peak digits is +12dB.

Output Gain

This option sets the gain of the analogue output stage. From the VOLUME sub-menu select OUTPUT GAIN. The available gain settings are from -6 dB to +24 dB in 1 dB increments. The default value for output gain is +12dB.

Analogue Mute

This option mutes the analogue audio output. From the VOLUME sub-menu select ANALOGUE MUTE. The available options are OFF and ON and by default, the analogue mute is off. This option can be remotely controlled.

Digital Mute

This option mutes the digital audio output. From the VOLUME sub-menu select DIGITAL MUTE. The available options are OFF and ON and by default, the digital mute is off. This option can be remotely controlled.





MEMORY CARD

This selection shows the Memory Card sub-menu. The sub-menu items are DETAILS, FILE DELETE and FORMAT CARD. A Compact Flash™ memory card slot, accessible from the front panel, accepts memory cards up to 2Gbytes.

Details

This selection shows details of the memory card currently inserted into the Compact Flash™ memory card slot. From the MEMORY CARD sub-menu select DETAILS. The information shown is the model number and the size of the card in Mbytes.

File Delete

This option allows any of the files on the Compact Flash™ memory card to be deleted. From the memory card sub-menu select FILE DELETE. The files and sub-directories on the Compact Flash™ memory card are shown in a list with the sub-directories at the top. Turn and press the rotary control to highlight and select the file to delete. A confirmation message box will be displayed; select OK to delete the file or select CANCEL to return to the list. To move to a sub-directory, press the rotary control with the highlight over the required directory name. Whilst viewing a sub-directory, the top item in the list is UP which, when selected, returns to the parent directory.

Format Card

This formats the CF card using a 16-bit FAT DOS format. From the MEMORY CARD sub-menu select FORMAT. When Format Card is selected, the system menu is exited, the main screen is shown & a message box indicates that the memory card is being formatted.

Please note: Formatting will delete all files and directories on the memory card.

REMOTES

This selection shows the Remotes sub-menu and allows you to configure the remote inputs and outputs to control the RB-PD2 externally. The sub-menu items are VIEW SETTINGS, CONFIGURE INPUTS and CONFIGURE OUTPUTS.

View Settings

This selection lists the current settings for the 8 remote inputs and the 6 remote outputs. From the REMOTES sub-menu select VIEW SETTINGS.

Configure Inputs

This selection sets the function that will be activated by the corresponding remote input. From the REMOTES sub-menu select CONFIGURE INPUTS followed by the required input number. A list is shown displaying all the possible functions that can be activated by the remote input with the default being NO ACTION:

NO ACTION

BUILD DELAY

EXIT DELAY

BUILD/EXIT DELAY (toggles the mode between build and exit delay)

ACTIVATE DUMP

ACTIVATE DROP

ACTIVATE COUGH

START RECORDING

MUTE OUTPUTS (both analogue and digital) and

ACTIVATE BYPASS

DELAY TO OUTPUT.

The options for remote input 7 and 8 remain unchanged.

If a remote input is set to "Delay To Output", the corresponding remote output in the remote output configuration options (Main Menu->Remotes->Configure Outputs->Output 1-6) is inaccessible and a message box is displayed stating that this remote output is controlled by the corresponding input.

For remote inputs that are configured as "Delay To Output", an active remote input (shorted to 0V) produces an active open collector output on the corresponding remote output after the current delay time period has elapsed. The open collector output is de-activated after the current delay period when the remote input is de-activated.

Configure Outputs

This selection sets the remote output signaling options for the corresponding remote output. From the REMOTES sub-menu select CONFIGURE OUTPUTS followed by the required output number. Each remote output can signal when a certain function or state has been activated. A list is shown displaying all the possible states that can be signaled by the remote output, with the default being NO ACTION:

NO ACTION

ZERO DELAY

BUILDING DELAY

DELAY SAFE (current delay is greater than safe period)

DELAY OK

EXITING DELAY

DUMP ACTIVE

DROP ACTIVE

COUGH ACTIVE

RECORD ACTIVE

OUTPUTS MUTED and

BYPASS ACTIVE.

CONTROL LOCK

This option allows the front panel bypass and rotary controls to be disabled. From the main system menu select CONTROL LOCK. The available options are OFF and ON. Access to the system menu is still enabled when the control lock is on to allow the lock to be removed. When the control lock is enabled, a small key icon is displayed in the upper right side of the main screen display. By default, the control lock is off.



**SYSTEM**

This selection shows the System sub-menu. The sub-menu items are BAUD RATE and INFORMATION.

Baud Rate

This selection allows the baud rate of the serial port to be set. From the SYSTEM sub-menu select BAUD RATE. The available options are 9600, 19200, 38400, 57600 and 115200. The default baud rate is 19200.

Information

This selection shows product and software version information. From the SYSTEM sub-menu select INFORMATION.

The Product Id is set to RB-PD2. This is used to identify the unit when using a PC and the Sonifex SCi control software.

The Serial Number is factory set. Always quote this number in any correspondence with Sonifex technical support.

The Bootstrap Version Number is the revision of the software that boots the system and controls the firmware upgrade process. This software is installed at the factory and can only be upgraded by Sonifex.

The Version Number is the revision of the main software. This software can be upgraded. For the latest software updates, check the Sonifex website at www.sonifex.co.uk

The Display Version Number is the software revision running on the display PCB.

Serial Port Control

The Serial Port allows the RB-PD2 to be controlled and updated from a PC using the Sonifex Serial Control Interface (SCI) software. This software is available as a free download from the Sonifex website at www.sonifex.co.uk/sci.

Default Settings for the Serial Port

Baud Rate:	19200
Data Bits:	8
Stop Bits:	1
Parity:	Even
Handshaking:	None

Serial Interface Commands and Responses

Most of the commands follow the same structure: a 3 letter command followed by a colon, followed by a parameter (if any) and terminated by Carriage Return with optional Line Feed. A Line Feed character may be sent but it will be ignored by the RB-PD2. Commands are not case sensitive.

Responses will be CR & LF terminated.

After the profanity delay has been powered-up, an initialisation string is sent "Initialising PD2".

Following are the commands and the expected responses:

Command	Description	Response
Bnn:	**Set baud rate where: nn = 11 (115200 baud) nn = 57 (57600 baud) nn = 38 (38400 baud) nn = 19 (19200 baud) nn = 96 (9600 baud)	ACK: or NAK: (at old rate)
BPD:	Bypass disable	ACK: or NAK:
BPE:	Bypass enable	ACK: or NAK:
BSV:	**Bootstrap version request where n.n = bootstrap version	BSV:Vn.n
CST:	Cough start	ACK: or NAK:
CSP:	Cough stop	ACK: or NAK:



Command	Description	Response
DLB:	Build delay	ACK: or NAK:
DLC:	Clear delay immediately to zero	ACK:
DLE:	Exit delay	ACK: or NAK:
DLF:c	Delete file on Compact Flash™ memory card where: c = "\path\filename" (c = "\filename" for files in root directory)	ACK: or NAK:
DLY:n	Set required delay where: n = delay in milliseconds (100ms resolution)	ACK: or NAK:
DRO:	Drop start	ACK: or NAK:
DRS:	Drop stop	ACK: or NAK:
DSP:n	Set delay build/exit speed where: n = 0 (5mins 30secs) n = 1 (5mins 00secs) n = 2 (4mins 30secs) n = 3 (4mins 00secs) n = 4 (3mins 30secs) n = 5 (3mins 00secs) n = 6 (2mins 30secs) n = 7 (2mins 20secs) n = 8 (2mins 10secs) n = 9 (2mins 00secs) n = 10 (1min 50secs) n = 11 (1min 40secs) n = 12 (1min 30secs) n = 13 (1min 20secs) n = 14 (1min 10secs) n = 15 (1min 00secs)	ACK: or NAK:
DST:	Dump start	ACK: or NAK:

Command	Description	Response
DWN:	**Download firmware. New firmware installed when 'S-Record' terminator received	ACK: or NAK: (ACK: indicates download can start)
FMC:	Format memory card	ACK: or NAK:
GDL:c	Get directory list where: c = "\path" for base directory of list (c="" for root directory) z = comma separated list of directories	GDL:z
GFL:n_c	Get file list where: n = 1 (list wav files) n = 2 (list wav files with same settings as current configuration) n = 3 (list all files) c = "\path" for base directory of list (c="" for root directory) z = comma separated list of files	GFL:z
LOC:	Enable control lock	ACK: or NAK:
MCD:	Request memory card details where: y = model number z = capacity in MBytes	MCD:y,z or NAK:
NRC:	Initialise new recording	ACK: or NAK:
REC:n	Record control where: n = 1 (start) n = 0 (stop)	ACK: or NAK:
RIN:c	Set record initials where: c = 4 character initials	ACK: or NAK:
RSC:n	Set record source where: n = 0 (analogue) n = 1 (analogue sync) n = 2 (digital)	ACK: or NAK:





Command	Description	Response
RSR:n	Set record sample rate where: n = 0 (32 kHz) n = 1 (44.1 kHz) n = 2 (48 kHz)	ACK: or NAK:
RSW:n	Set record sample width where: n = 0 (16 bit) n = 1 (24 bit)	ACK: or NAK:
SAM:n	Set analogue mute where: n = 0 (off) n = 1 (on)	ACK: or NAK:
SDB:n	Set dump buffer length where: n = dump length in milliseconds (100ms resolution)	ACK: or NAK:
SDF:c	Set dump play file where: c = "\path\filename" (c="\filename" for files in root directory) z = error message	ACK: or NAK:z
SER:	**Serial number request where: z = serial number (6 digits)	SER:z
SDM:n	Set digital mute where: n = 0 (off) n = 1 (on)	ACK: or NAK:
SIP:n	Set input peak digits where: n = 0 (+12dB) n = 1 (+18dB) n = 2 (+24dB)	ACK: or NAK:
SMD:n	Set dump mode where: n = 0 (dump buffer) n = 1 (play file)	ACK: or NAK:
SMS:n	Set start delay mode where: n = 0 (build using audio stretch algorithm) n = 1 (play file) z = error message	ACK: or NAK:z

Command	Description	Response
SOG:n	Set output gain where: n = 0 (-6dB) n = 1 (-5dB) n = 2 (-4dB) n = 3 (-3dB) n = 4 (-2dB) n = 5 (-1dB) n = 6 (-0dB) n = 7 (+1dB) n = 8 (+2dB) n = 9 (+3dB) n = 10 (+4dB) n = 11 (+5dB) n = 12 (+6dB) n = 13 (+7dB) n = 14 (+8dB) n = 15 (+9dB) n = 16 (+10dB) n = 17 (+11dB) n = 18 (+12dB) n = 19 (+13dB) n = 20 (+14dB) n = 21 (+15dB) n = 22 (+16dB) n = 23 (+17dB) n = 24 (+18dB) n = 25 (+19dB) n = 26 (+20dB) n = 27 (+21dB) n = 28 (+22dB) n = 29 (+23dB) n = 30 (+24dB)	ACK: or NAK:





Command	Description	Response
SRI:i_n	Set function for remote input where: i = remote input id (0-7) n = 0 (no action) n = 1 (build delay) n = 2 (exit delay) n = 3 (activate dump) n = 4 (activate drop) n = 5 (activate cough) n = 6 (start new recording) n = 7 (mute all audio outputs) n = 8 (activate audio bypass)	ACK: or NAK:
SRO:o_n	Set function signal for remote output where: o = remote output id (0-5) n = 0 (no action) n = 1 (delay is at zero) n = 2 (delay building) n = 3 (delay is greater than safe period) n = 4 (delay is at selected value) n = 5 (delay exiting) n = 6 (dump active) n = 7 (drop active) n = 8 (cough active) n = 9 (record active) n = 10 (audio outputs muted) n = 11 (audio bypass active)	ACK: or NAK:
SRQ:n	Status request where: e = unit status (8 hex digits) f = current delay (8 hex digits) g = selected delay (8 hex digits) h = start mode setting i = start play filename j = safe period setting k = dump mode setting l = dump buffer length setting m = dump play filename n = source setting o = sample rate setting p = sample width setting	STA:e_f_g_h_i_j_k_l_m_n _o_p_q_r_s_t_u_v_w_x_y_z

Command	Description	Response
	q = record source setting r = record sample rate setting s = record sample width setting t = input peak digits setting u = output gain setting v = analogue mute status w = digital mute status x = bypass status y = control lock status z = delay build/exit speed setting	
SSC:n	Set source where: n = 0 (analogue) n = 1 (analogue sync) n = 2 (digital)	ACK: or NAK:
SSF:c	Set delay start play file where: c = "\path\filename" (c="filename" for files in root directory) z = error message	ACK: or NAK:z
SSP:n	Set safe period length where: n = dump length in milliseconds (100ms resolution)	ACK: or NAK:
SSR:n	Set sample rate where: n = 0 (32 kHz) n = 1 (44.1 kHz) n = 2 (48 kHz)	ACK: or NAK:
SSW:n	Set sample width where: n = 0 (16bit) n = 1 (24bit)	ACK: or NAK:
UID:	**Unit ID request	UID:PD2
UNL:	Disable control lock	ACK: or NAK:
VER:	Version request where y.yyy = firmware version zzzz = display version	VER:Vy.yyy,Vzzzz





Command	Description	Response
VRS:	Request remote settings where: m = remote input 1 setting n = remote input 2 setting o = remote input 3 setting p = remote input 4 setting q = remote input 5 setting r = remote input 6 setting s = remote input 7 setting t = remote input 8 setting u = remote output 1 setting v = remote output 2 setting w = remote output 3 setting x = remote output 4 setting y = remote output 5 setting z = remote output 6 setting	VRS:m_n_o_p_q_r_s_t_u_v _w_x_y_z

** Supported in Bootstrap mode

Upgrading Firmware

Occasionally, it may be necessary to upgrade the firmware on the RB-PD2 to add new functionality and fix software bugs. New firmware updates will be made available from time to time on the Sonifex website. Visit www.sonifex.co.uk for details.

The firmware is upgraded using the Serial Control Interface - SCI. This PC software downloads the new firmware and initiates the upgrade process. To upgrade the firmware, select the required firmware file in the SCI upgrade firmware section. The RB-PD2 will automatically reboot and enter the bootstrap mode prior to the transfer of the firmware file. The SCI software downloads the firmware to internal memory, and then erases the current firmware before programming the update. When the firmware is complete, the unit reboots with the new firmware.

Please note: Firmware files can take several minutes to transfer to the RB-PD2 at lower baud rates. To speed up the process, select a higher baud rate prior to transferring the new firmware.

Technical Specification RB-PD2

Audio Specification

Maximum Input Level:	+27dBu	
Input Impedance:	> 10kΩ bridging	
Analogue & Digital Input Levels:	Selectable +12dBu, +18dBu, +24dBu for FSD	
Analogue Pre-set Input Gain Range:	Adjustable 3dB loss to 3dB gain (L & R adjust)	
Signal to Noise:	Better than -101dBFS (RMS A-weighted at 24bit)	
Dynamic Range:	> 110dB	
Distortion & Noise:	> 96dB THD + N at 1kHz	
Maximum Output Level:	+24dBu	
Output Impedance:	< 50Ω	
Dynamic Range:	> 100dB	
Analogue Output Gain Range:	Selectable -6dBu to +24dBu output level, ref FSD	
Sampling Frequency:	Selectable 32kHz, 44.1kHz, 48kHz	
Sample Width:	Selectable 16bit or 24bit	
Channels:	Stereo	
Minimum Delay Duration:	2 seconds	
Maximum Delay Duration:	16 bit	24 bit
32kHz:	55.1 secs	27.5 secs
44.1kHz:	40.0 secs	20.0 secs
48kHz:	36.8 secs	18.4 secs



Rear Panel Connections

Analogue Inputs:	2 x XLR 3 pin female (balanced) (L & R)
Analogue Outputs:	2 x XLR 3 pin male (balanced) (L & R)
Digital Inputs:	1 x AES/EBU XLR 3 pin female
Digital Outputs:	1 x AES/EBU XLR 3 pin male
Remote I/O Port:	15-way 'D'-type plug, 8 GPI inputs, 6 GPI outputs
Serial Comms Port:	9-way 'D'-type plug
Mains Input:	Filtered IEC, continuously rated 100-240 VAC, 47-63Hz, fused, 60W peak, 30W average
Fuse Rating:	Anti-surge fuse 2A 20 x 5mm

Front Panel Controls

Display:	Vacuum fluorescent display
Direct Control Push-Buttons:	Build Delay, Exit Delay, Cough, Bypass & Dump
System Navigation:	Rotary selector with integral push-switch
Removable Audio Storage Device:	Compact Flash™ memory card port (supporting up to 2GB)

Physical Specification

Dimensions (Raw):	48cm (W) x 15.8cm (D*) x 4.3cm (H) (1U) 19" (W) x 6.2" (D*) x 1.7" (H) (1U)
Dimensions (Boxed):	59cm (W) x 27.4cm (D*) x 10.8cm (H) 23.2" (W) x 10.8" (D*) x 4.3" (H)
Weight:	Nett: 1.7kg Gross: 2.3kg Nett: 3.7lbs Gross: 5lbs

* Note that this product is deeper than standard Redboxes

9 RB-SD1 Silence Detection Unit

Introduction



Fig 9-1: RB-SD1 Front Panel

The RB-SD1 Silence Detect Unit is a 1U rack mount device used to monitor an unattended stereo studio feed and in the event of the signal going “quiet” after a given period the unit will switch through an alternative stereo audio signal. This signal could be a recorded message (e.g. “normal service will be resumed”, etc), a feed from a CD player or minidisk machine, or an alternative recorded program. Controls are provided to start external equipment and to provide remote status indication.

It has 2 balanced stereo audio inputs with the maximum input level being +28dBu. Each input is user-defined as either the main source or auxiliary source and both sources are monitored for failure, each having a remote failure alarm. In the event of the main source dropping below a pre-set level for a pre-determined amount of time, the unit will automatically switch through to the auxiliary signal. The silence detect level is adjustable between -60dBu and -15dBu in 3dB steps via a 16 position rotary switch on the rear panel. The silence interval can be adjusted between 2 seconds to 30 seconds in 2 second steps, or, alternatively, set to 2 minutes 5 seconds also via a 16 position rotary switch on the rear panel. The audio outputs use stereo professional balanced XLR-3 male connectors.

The unit can operate in 2 modes - automatic or manual. In both modes it will automatically switch over to the auxiliary source on detecting silence. When the main signal is again detected it will either return to the main signal automatically or manually depending on the mode chosen.

The RB-SD1 has a number of remote operational features. Remote outputs provide separate relay contact closures for failure of the main and auxiliary inputs. You can also control remotely all of the front panel switches for source selection, mode selection and signal Restore. You can remotely start and stop another piece of equipment on alarm failure and main signal return respectively. Also, the longest silence time (2min 5sec) can be set remotely, which is useful if you are expecting to broadcast a long silence.

The unit can be configured to alarm when either the left or right channel of the main input source fails, or if the whole stereo signal fails. There are also options to set the remote start output as momentary or latched, to disable switching to the auxiliary input on alarming and to increase the gain on the auxiliary input so that an unbalanced input can be used, for example, from a domestic minidisk player.

Front panel LED indicators show individually left and right programme and alarm conditions for both the main and auxiliary inputs. The status of the source, mode and alarm state are also shown on the front panel with LED indicators.



Additionally, the RB-SD1 can be programmed for specific applications, which can be defined on power-up of the unit, e.g. for extended silence detect times. See page 105 for current additional power-up modes, or contact Sonifex for further information if you have a particular requirement.

The RB-SD1 has been designed to have a passive signal path through the main input, so if power to the unit fails, the signal input will still be routed through to the output. This is essential for applications such as installation at transmitter sites, where a power failure to the unit should not prevent the audio input signal from being output to the transmitter.

System Block Diagram

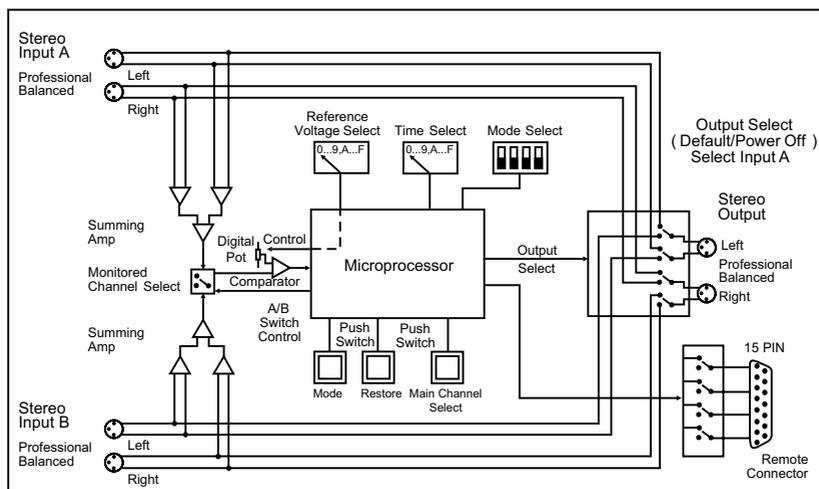


Fig 9-2: RB-SD1 System Block Diagram

Rear Panel Connections and Operation

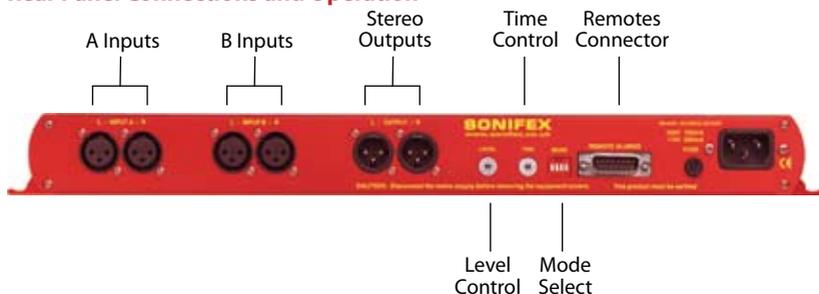


Fig 9-3: RB-SD1 Rear Panel

A/B Inputs (Left and Right)

There are four XLR-3 inputs, two for channel A (Left & Right) and another two for channel B (Left & Right). The XLR 3 pin sockets are used for the input channels and are electronically balanced. They have the following connections: -

- Pin 1: Screen.
- Pin 2: Phase.
- Pin 3: Non-phase.

Outputs

The stereo input consists of two XLR male connectors professionally balanced with following connections: -

- Pin 1: Screen.
- Pin 2: Phase.
- Pin 3: Non-phase.

Silence Detect Trigger Level

The trigger level switch (Level) adjusts the level below which silence detection occurs. This level may be varied from -15dB to -60db in 3db steps by adjusting the switch, which is accessible on the rear panel.

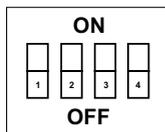
Switch	0	1	2	3	4	5	6	7	8	9	A	B	C	D	E	F
Level dBu	-60	-57	-54	-51	-48	-45	-42	-39	-36	-33	-30	-27	-24	-21	-18	-15

Silence Detect Interval Control

The silence detect interval rotary switch (Time) adjusts the duration over which a silence is detected before alarming and ranges from 2-30 seconds (0 – E in 2 second intervals) with F on the switch being a 2 min 5 second silence. This maximum time can also be activated or de-activated remotely via the remote connector.

Switch	0	1	2	3	4	5	6	7	8	9	A	B	C	D	E	F
Seconds	2	4	6	8	10	12	14	16	18	20	22	24	26	28	30	125

Mode Dip Switch Settings



1. Stereo/Mono.
2. Remote Start Mode Switch.
3. Professional levels/Consumer levels (Input B).
4. Switch/No switching when alarmed.

1. Stereo/Mono Switch – The configuration of this defines whether you want to switch sources when left and/or right channel of the incoming source go silent.





Switch	Description
On	When on, the unit operates in stereo mode, whereby if one channel goes quiet the unit will switch, and requires both channels to be present before it switches back.
Off	When off, the unit operates in mono mode. In this mode the unit will only switch when both channels go quiet, and requires only one channel to be present before the unit switches back.

2. Remote Start Mode Switch – This defines whether the remote start switch is momentary or latched. Used for starting external equipment when silence is detected.

Switch	Description
On	When on, the remote start pin (pin 15) on the remote connector is pulled low for half a second when the unit switches to the auxiliary input. (Momentary contact).
Off	When off, the remote start pin on the remote connector is pulled low when the unit switches over to the auxiliary input and remains low until the unit switches back to the main source or, if in manual mode, is restored by the user locally or remotely. (Latched contact).

3. Professional/Consumer Switch – This allows you to use an unbalanced piece of equipment as the auxiliary input, by raising the input gain.

Switch	Description
On	When on, Input B accepts professional balanced signal level.
Off	When off, Input B accepts consumer unbalanced signal level and raises the input gain received by 12dB.

4. Switch/No Switching in alarm state – This defines whether the unit switches to the auxiliary input on silence detection.

Switch	Description
On	When on, if the unit goes into the alarm state the unit switches to the auxiliary input.
Off	When off, if the unit goes into the alarm state the unit does not switch to the auxiliary input.

Remotes Connector

Displayed below are the pin connections and descriptions for the remote connector:

Pin No.	Signal	I/O	Description
Pin 1	Master Alarm Normally Open	O	Relay 1 N/O to Pin 9 in alarm state
Pin 2	Master Alarm Normally Closed	O	Relay 1 N/C to Pin 9 in alarm state
Pin 3	Aux. Alarm Normally Open	O	Relay 2 N/O to Pin 11 in alarm state
Pin 4	Aux. Alarm Normally Closed	O	Relay 2 N/C to Pin 11 in alarm state
Pin 5	Mode Switch	I	Momentary make to Pin 8
Pin 6	Mode Indicator	O	Internal Open Collector to Digital Ground
Pin 7	Restore Switch	I	Momentary make to Pin 8
Pin 8	Digital Ground	-	-
Pin 9	Master Alarm Common	O	N/O to Pin 1, N/C to Pin 2 in alarm state
Pin 10	Max Time Whilst Latched	I	Latched make to Pin 8
Pin 11	Aux. Alarm Common	O	N/O to Pin 3, N/C to Pin 4 in alarm state
Pin 12	+5V	O	To power up to a maximum 100mA
Pin 13	Source Select Indicator	O	Internal Open Collector to Digital Ground
Pin 14	Source Select Switch	I	Momentary make to Pin 8
Pin 15	Remote Start	O	Internal Open Collector to Digital Ground

Pins 1 - 4 are for external use to replicate the alarm conditions for the Main and Auxiliary inputs.

Pins 5 - 7, 13 and 14 are to replicate the switches and indicators for the source select, mode and restore functions.

Pins 8, 9, 11 and 12 are common or voltage pins.

Pin 10 is to select remotely the maximum silence time (2min 5sec). This may be useful for the broadcast of Remembrance Day services, or where you expect a silence of up to 2 minutes to be broadcast. The maximum silence time is set whilst the contact is latched.

Pin 15 is used to remotely start an external piece of equipment and operates on audio fail.

Front Panel Selectors and Indicators



Fig 9-4: RB-SD1 Front Panel



Source Select and Indicator

The normal Main input source is selectable via a push switch accessed via a hole located on the front panel, or it can be controlled remotely (pins 13 and 14). This allows you to define whether input A, or input B is going to be your Main audio input. There is an LED to indicate which state the source select is set:

LED	Description
LED On	Main source is input A, Aux source is input B
LED Off	Main source is input B, Aux source is input A

Note : If the unit is powered off, for example during a black-out, input A routes through to the output. Therefore if the unit is subject to a power fail while the main source is set to input B, the unit will output source A.

Mode Selector and Indicator

The Mode Switch defines how the unit should operate during an alarm condition, when the main audio source returns. There is an option to allow the device to switch back Automatically or Manually. The mode is selected by a push switch accessed through a hole on the front panel with a corresponding LED to represent its state, or it can be controlled remotely (pins 5 and 6).

LED	Description
LED On	Automatic Mode – During an alarm condition when the main source returns, it is switched back automatically, although there must be a continuous signal present for two seconds before it switches.
LED Off	Manual Mode – When the alarm condition is reached, the LED begins to flash.
LED Flashing	Manual Mode – The main audio has returned after an alarm condition. To switch to the main source, push the Restore button (or control it remotely).

Main and Aux Indicators

On the front panel there are four Main indicators and four Aux indicators. Each left/right channel has a Program Content and Alarm Status indicator. The Program Content Indicator represents the input signal level for that channel and the Alarm Status LED indicates when the channel has dropped below the threshold for longer than the time selected.

Both the Main and Aux inputs are continuously monitored so that you can check that your backup signal is operating correctly, as well as your main input source.

Note: Although one channel of the stereo input may have alarmed, the main alarm may not be set, due to the setting of the Stereo/Mono Dipswitch.

Alarm Indicator

The Alarm indicator situated on the front panel is used to display the alarm status of the Main input. Its operation is dependant on the selected mode. When in stereo mode the unit alarms on a single main channel timeout, and in mono, alarms on both main channels timing out. The unit exits the alarm state on the return of the main source signal, depending on the setting of the Stereo/Mono Dipswitch. The alarm indicator is remotely indicated on pins 1 and 2 of the remotes connector.

Restore Button

The restore button is used for restoring the main source signal when the unit is operating in Manual Mode. When the main source signal returns after it has timed out, the Mode LED flashes, indicating that the source signal can be restored. When the Restore button is pressed the main source returns. This can be remotely controlled using pin 7 of the remotes connector.

Additional Modes

An option to set the unit in different modes of operation is available and is selectable when the unit is powered on. The current available modes of operation are as follows:

Normal Mode is as described previously for normal machine power-up.

Remote Stop Mode operates as follows. When the main source returns from an alarmed state, pin 4 on the remote connector (Aux Alarm) closes to Pin 11 (Aux Alarm Common) for half a second. (Note: this will only occur when the Remote Start Mode Switch is set on (page 102) and whilst in Remote Stop Mode the Aux alarm is not available to indicate the presence of audio on the auxiliary input).

Remote Alarm Set Mode operates as follows. When the RESTORE pin on the remote connector (pin 7) closes to the DIGITAL GROUND pin on the remote connector (pin 8) the unit will immediately enter an alarmed state and switch to the auxiliary input. This alarm state is continued until the RESTORE pin opens to the DIGITAL GROUND pin. After this the unit will switch back to the main input when audio is present.

NOTE: When operating in manual mode, restoration of output from the main input is only available via the front panel restore switch. All other functions operate as normal.

Signal Detection Mode In this mode the three controls, source, remote source, mode, remote mode, restore and remote restore have no effect.

The count in time for the signal detection is zero, i.e. as soon as signal is detected on an input the relevant relay will switch. The relays operate the same as in normal mode, i.e. when audio is detected on the main input, pin 1 on the remote connector closes to pin 9. When audio is lost on the main input, pin 2 on the remote connector closes to pin 9. All other functions operate as with normal mode.

Timeout x 2 Mode. In this mode, the timeout selection is multiplied by two (see table on the following page), except for the max time selection (position F on the time rotary encoder). All other functions operate as normal.





Switch	0	1	2	3	4	5	6	7	8	9	A	B	C	D	E	F
Seconds	4	8	12	16	20	24	28	32	36	40	44	48	52	56	60	125

Force Main Mode. In this mode only the remote mode control input has any effect. When this control input is active (low) the output is forced to input B (the master input) regardless of whether audio is present. The unit operates in auto mode and with input B as the master input. All other functions operate as with normal mode.

Emergency Program Override Mode. In this mode all alarm LEDs are illuminated whenever the main input is active. This is to indicate that the emergency paging program is active on the main input. When the paging program goes silent, the unit will revert back to the aux. input and all the alarm LEDs will switch off. The front panel source button is disabled. All other functions operate as normal.

Detection Enable/Disable Mode. In this mode the silence detection can be disabled and enabled remotely. The front panel and remote source select controls are used to enable or disable the Silence Detection feature.

Front panel Source indicator on	=	Silence Detection enabled
Front panel Source indicator off	=	Silence Detection disabled

The front panel source select button toggles this state, while the remote Source control (Pin 14 on the remote connector) can only ENABLE Silence Detection. Pin 10 on the remote connector is used to DISABLE Silence Detection. When the unit powers up Silence Detection is always enabled. The remote source select indicator (Pin 13 on the remote connector) is enabled when Silence Detection is disabled. In this mode input A is always the Main input and input B is always the Aux input. Since the front panel and remote source selection controls are disabled, source selection is not allowed in this mode. The Aux alarms are masked for both front panel indicators and remote alarm relay, for as long as the main input is present.

Return Time Set Mode. In this mode, the unit operates as normal, except that during power up the return time delay can be programmed. The return time delay is the amount of time, during an alarm condition, that the unit takes to switch from the Aux to Main input once audio is presented to the Main input. To set the return time, immediately after the unit is switched on, press and hold the MODE switch. While the MODE switch is pressed set the TIME rotary switch to required setting (see below). When finished, release the MODE switch, and the unit will start operating within a few seconds. Remember to return the TIME rotary switch to the required setting.

Rotary Setting	0	1	2	3	4	5	6	7	8	9	A	B	C	D	E	F
Seconds	0	2	4	6	8	10	12	14	16	18	20	22	24	26	28	120

To Set the Additional Modes:

Apply power to the unit, and while the front panel ALARM LED is flashing, press and hold the RESTORE button. The MAIN and AUX PROG and ALARM LEDs will flash alternately - these are used to indicate the selected mode. The SOURCE LED will also alternate to show the mode bank that is currently selected.

Note: For RB-SD1 units with serial number greater than RB031084, the SOURCE LED is used to show the bank of the selected mode. For serial numbers before this, the SOURCE LED is not used. Contact Sonifex Ltd if you want to upgrade your RB-SD1 unit to the latest version to take advantage of a new mode.

To select the particular mode, release the RESTORE button when the corresponding LED below is on. After the button is released the unit will start working after five seconds. You only have to do this once as the mode is stored in non-volatile memory and you will only need to repeat this procedure if you wish to select a different mode. Each time the unit is powered on, the selected mode is visible by checking which LED is on:

Bank 0 (SOURCE LED off)		Mode Selected
MAIN LEFT PROG LED on	=	Normal Mode
MAIN LEFT ALARM LED on	=	Remote Stop Mode
MAIN RIGHT PROG LED on	=	Remote Alarm Set Mode
MAIN RIGHT ALARM LED on	=	Signal Detection Mode
AUX. LEFT PROG LED on	=	Timeout Times 2 Mode
AUX. LEFT ALARM LED on	=	Force Main Mode
AUX. RIGHT PROG LED on	=	Emergency Program Override Mode
AUX. RIGHT ALARM LED on	=	Detection Enable/Disable Mode

Bank 1 (SOURCE LED on)		
MAIN LEFT PROG LED on	=	Return Time Set Mode
MAIN LEFT ALARM LED on	=	Reserved
MAIN RIGHT PROG LED on	=	Reserved
MAIN RIGHT ALARM LED on	=	Reserved
AUX. LEFT PROG LED on	=	Reserved
AUX. LEFT ALARM LED on	=	Reserved
AUX. RIGHT PROG LED on	=	Reserved
AUX. RIGHT ALARM LED on	=	Reserved

Technical Specifications RB-SD1

Audio Specifications

Maximum Input Level:	+28dBu
Input Impedance:	> 100kΩ balanced
Maximum Output Level:	+28dBu
Output Impedance:	As input, except when using unbalanced auxiliary input where output impedance < 50Ω
Frequency Response:	20Hz to 20kHz ±0.1dB
Gain:	+12dB (for unbalanced input B – optional)
Noise:	<-87dB, unity gain, ref +8dBu output for unbalanced input.





Distortion: As input for balanced input, <0.05% ref +8dBu output for unbalanced input.

Connections

Inputs (Main & Auxiliary): 4 x XLR 3 pin female (balanced, auxiliary can be unbalanced)

Output: 2 x XLR 3 pin male (balanced)

Remotes: 15 way D-type plug

Power: Filtered IEC, 110-120V, or 220-240V switchable, fused, 6W maximum

Fuse Rating: Anti-surge fuse 100mA 20 x 5mm (230VAC)
Anti-surge fuse 250mA 20 x 5mm (115VAC)

Rear Panel Controls

Alarm Threshold: -15dBu to -60dBu in 3dB steps via rotary switch

Silence Detect Duration: 2 sec to 30 sec in 2 second intervals and 125 second option via rotary switch

Detection Type: Mono or stereo, via dipswitch

Silence Switch Defeat: Disable/enable silence switching, via dipswitch

Remote Start Mode: Latched or momentary, via dipswitch

Front Panel Controls and Indicators

Controls: Source select, mode select and restore

Indicators: Program and alarm indicators for left and right source for both main and auxiliary channels
Source, mode and restore LEDs

Equipment Type

RB-SD1:  Silence detection unit

Physical Specifications

Dimensions (Raw): 48cm (W) x 10.8cm (D) x 4.2cm (H) (1U)
19" (W) x 4.3" (D) x 1.7" (H) (1U)

Dimensions (Boxed): 53cm (W) x 20.5cm (D) x 6cm (H)
21" (W) x 8" (D) x 2.4" (H)

Weight: Nett: 1.4kg Gross: 2.0kg
Nett: 3.1lbs Gross: 4.4lbs

10 RB-DSD1 Digital Silence Detection Unit

Introduction



Fig 10-1: RB-DSD1 Front Panel

The RB-DSD1 digital silence detection unit works in a similar way to the Sonifex RB-SD1 analogue silence detection unit, but has AES/EBU, S/PDIF and Toslink inputs and outputs instead of analogue inputs and output respectively. Designed to switch from one input to another in the event of loss of audio, the unit is ideal at transmitter sites, or after the master output of a studio, to switch in another audio source, or simultaneous broadcast, should a master source fail.

The unit can switch:

- On loss of level of the main input.
- On loss of level on one channel of the main input.
- On loss of synchronisation lock of the main input.

The RB-DSD1 has 2 x digital stereo audio inputs, each one selectable via front panel MAIN and AUX push buttons, from either AES/EBU balanced XLRs, S/PDIF unbalanced phonos or Toslink unbalanced optical inputs. Sample rate converters on each input mean that sources of different sample rates can be used with the output sample rate being defined independently. Each input is user-defined as either the main source or auxiliary source and both sources are monitored for failure, each having a remote failure alarm. The colour of the MAIN and AUX push-buttons indicate which input is the current output channel (green) and which input is the waiting channel (red) input, with a flashing LED indicating loss of synchronisation.

In the event of the main source dropping below a pre-set level for a pre-determined amount of time, the unit will automatically switch through to the auxiliary signal. The silence detect level is adjustable between -39dBfs and -84dBfs in 3dBfs steps via front panel DIPswitches. The silence interval can be adjusted between 0 seconds and 252 seconds in 2 second steps via another front panel DIPswitch block. A small cover panel can be screwed in place to obscure the DIPswitches to prevent tampering of the settings.

There are 2 stereo outputs to allow for distribution of the selected input to multiple outputs. Each output is available as simultaneous AES/EBU balanced XLRs, S/PDIF unbalanced phonos or Toslink unbalanced optical outputs. The output sample rates are selectable via rear panel DIPswitches from one of 32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz.

The unit has TTL wordclock BNC and AES/EBU XLR synchronising inputs as standard and optionally, the RB-SYA and RB-SYD synchronisation boards can be fitted to synchronise

10 SYNCHRONISERS, DELAYS & SILENCE DETECTORS



the unit to analogue or digital video signals. A front panel DIPSwitch block is used to decide whether the unit is synchronised to Input1, Input2, the AES/EBU sync input, the wordclock sync input or an optional video sync board. A front panel SYNC button selects the synchronisation mode of the unit and the button flashes whenever the unit is not synchronised to an incoming sync signal. Selectable sync modes are as follows:

Master Mode

In this mode the digital output sample rate is simply set by, and locked to, the internal on-board clock generator. No sync signal is used or required.

Auto Sync Mode

In this mode the digital output sample rate follows the selected sync input. When the sync signal is not present the output sample rate will be set by, and locked to, the internal on-board clock generator at the selected output frequency.

Auto Lock Mode

In this mode no output will be generated until lock is achieved with a sync signal. The digital output sample rate now follows the sync input. If the sync signal is removed then the output sample rate will be set by, and locked to, the internal on-board clock generator at the closest frequency available to the previous sync input.

Slave Mode

In this mode the digital output sample rate follows the sync input. When the sync signal is not present the digital output is turned off.

The unit can operate in 2 modes - automatic or manual, selectable using a rear panel DIPSwitch. In both modes it will automatically switch over to the auxiliary source on detecting silence. When the main signal is again detected it will either return to the main signal automatically or manually depending on the mode chosen. In manual mode, the front panel RESTORE button is used to return to the main signal.

The RB-DSD1 has a number of remote operational features. Remote outputs provide separate relay contact closures for failure of the main and auxiliary inputs. You can also remotely select between auto and manual mode (with tally output), action the signal RESTORE, set the silence detection delay to be 2mins 5 seconds and define which input is the main input (with tally output). You can remotely start and stop another piece of equipment on alarm failure and there is an option to set the remote start output as either momentary or latched.

The unit can be configured to alarm when either the left or right channel of the main input source fails, or if the whole stereo signal fails. Additionally, if one channel of a stereo signal is lost, you can define whether to mute the lost channel, or whether to mix the remaining channel to the lost side, effectively creating a mono signal. If the main source synchronisation is lost, you can define whether the unit switches to the auxiliary input in the minimum time (2 seconds), or whether to treat the signal as silence to be detected and then switched based on the unit's silence detection settings.

Front panel LED indicators by the MAIN and AUX buttons show individually left and right programme and alarm conditions for both the main and auxiliary inputs.

A powerful feature of the RB-DSD1 is that by using the Sonifex SCI serial software, the unit can be programmed for different delay durations, levels and switching functions so that you can programme the unit for your specific application. A front panel DIPSwitch configures the unit to be controlled serially and a front panel LED indicates serial operation. Contact Sonifex for further information if you have a particular requirement that isn't catered for by the RB-DSD1 as standard.

The RB-DSD1 has been designed to have a passive signal path through the main input, so if power to the unit fails, signal input 1 is routed to output 1 and signal input 2 is routed to output 2. This is essential for applications such as installation at transmitter sites, where a power failure to the unit should not prevent the audio input signal from being output to the transmitter. Note: This applies to the AES/EBU & S/PDIF I/O, but not the TosLink optical I/O.

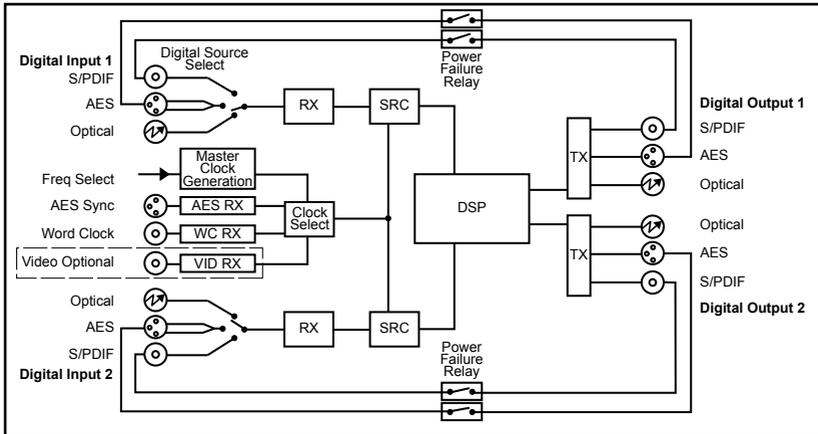


Fig 10-2: RB-DSD1 System Block Diagram

Front Panel Controls and Indicators

The LED in the front panel is normally red to indicate power to the unit.

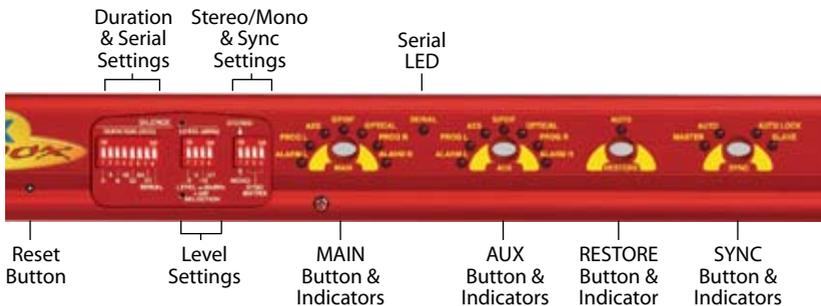


Fig 10-3: RB-DSD1 Front Panel Controls and Indicators

10 SYNCHRONISERS, DELAYS & SILENCE DETECTORS



MAIN Source Select and Indicators

This button allows you to select which input source you would like to use, from one of the AES/EBU, S/PDIF or TOSlink inputs. The three LEDs above the button illustrate which source is selected. The button itself will also be illuminated.

Button Colour	Indication
Solid green	Indicating that this is the input that is currently being output
Solid red	Indicating that this input is not selected
Flashing between green and red	Indicating a loss of lock to that input

There are also green and red LEDs, one of each for both left and right channels to show the programme and alarm state of the input. The programme LEDs will be illuminated if the level is above the current silence threshold. If the input is in an alarmed state, the red alarm LEDs will be illuminated.

AUX Source Select and Indicators

The operation of this button is identical to the above.

RESTORE Button

The Restore button is used for restoring the main source signal when the unit is operating in Manual Mode. When the main source signal returns after it has timed out, the AUTO LED flashes, indicating that the source signal can be restored. When the Restore button is pressed the main source returns. This can be remotely controlled using pin 7 of the remotes connector.

SYNC Mode button

This button allows you to cycle through the synchronisation modes. The LEDs around the button display which mode has been selected. If the selected synchronisation source (see front panel DIPSwitch section for settings) is not locked, the button will flash green and red.

SERIAL Mode indicator

If serial mode is selected (see front panel DIPSwitch section for settings) this led will be illuminated.

Reset Button

In the unlikely event that the RB-DSD1 unit fails to respond, press the reset button to reboot the unit (see Fig 10-3 for location).

Front Panel DIPSwitches

The front panel DIPSwitches are concealed behind the screw panel located on the front of the unit.

Silence Detect Duration Control

Switch	1	2	3	4	5	6	7
Seconds	2	4	8	16	32	64	X2

The silence detect interval DIPSwitches adjust the duration over which a silence is detected before alarming which ranges from 2-252 seconds (by adding together the raised DIPSwitches, in 2 second intervals). If all the DIPSwitches are down, (off) the duration defaults to 30 seconds. This can be changed using the SCi serial software.



Fig 10-4: SILENCE Detect Duration DIPSwitch

Serial Mode Setting

To activate serial mode, set DIPSwitch 8 on (up). When activated, the serial mode LED illuminates on the front panel (see Fig 10-3) and the unit can be controlled serially instead of from the front panel controls, using the SCi software.

Silence Detect Audio Trigger Level

Switch	1	2	3	4
Seconds	-3	-6	-12	-24

The trigger level DIPSwitches adjust the level below which silence detection occurs. This level may be varied from -39dBFS to -84dBFS in 3db steps by summing DIPSwitches that are on and adding -39dBFS. 0 (all off) represents -39 dBFS.

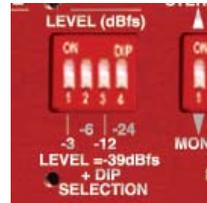


Fig 10-5: SILENCE Detect Audio Trigger Level DIPSwitch

Stereo/Mono Selection

The configuration of this defines whether you want to switch sources when the left and/or right channel of the incoming source go silent.

Switch	Description
On	When on, the unit operates in stereo mode, whereby if one channel goes quiet the unit will switch, and requires both channels to be present before it switches back.
Off	When off, the unit operates in mono mode. In this mode the unit will only switch when both channels go quiet, and requires only one channel to be present before the unit switches back.



Fig 10-6: SILENCE Detect Stereo/Mono & Sync Source DIPSwitch

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Synchronisation Source Selection

Select which synchronisation source you would like to use by setting DIPSwitches 2 - 4, marked SYNC MATRIX.

Synchronise from	DIPSwitch 2	DIPSwitch 3	DIPSwitch 4
Main input	Off	Off	Off
Aux input	On	Off	Off
AES synchronisation input	Off	On	Off
Wordclock input	On	On	Off
Video sub board	Off	Off	On

Rear Panel Controls



Fig 10-7: Rear Panel
DIPSwitches

Output Sample Rate Selection

DIPSwitches 1 - 3 allow you to select what sample rate the output will be when the unit is in master mode. Set the DIPSwitches where:

Sample Rate (kHz)	DIPSwitch 1	DIPSwitch 2	DIPSwitch 3
32	Off	Off	Off
44.1	On	Off	Off
48	Off	On	Off
88.2	On	On	Off
96	Off	Off	On
176.4	On	Off	On
192	Off	On	On

Main/Aux Input Selection

DIPSwitch 4 defines whether the Main input is input 1 or 2.

DIPSwitch 4	Description
On	When on, the Main input is input 2. The Aux input is input 1.
Off	When off, the Main input is input 1. The Aux input is input 2.

Manual/Auto Mode Selection

DIPSwitch 5 defines whether the unit is in manual or automatic switch mode.

DIPSwitch 5	Description
On	When on, the unit is automatic mode. The restore LED will be lit in this mode. The unit will switch back to the main input automatically if the control has been switched to aux and the main input level has recovered for 2 seconds.
Off	When off, the unit is in manual mode. The restore LED will begin to flash if the control has been switched to aux and the MAIN level has recovered. The restore button will then switch the unit back to the main input.

Remote Start Operation

DIPSwitch 6 defines whether the remote start switch is momentary or latched. Used for starting external equipment when silence is detected.

DIPSwitch 6	Description
On	When on, the remote start pin (pin 15) on the remote connector is pulled low for half a second when the unit switches to the auxiliary input. (Momentary contact).
Off	When off, the remote start pin on the remote connector is pulled low when the unit switches over to the auxiliary input and remains low until the unit switches back to the main source or, if in manual mode, is restored by the user locally or remotely. (Latched contact).



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Operation On Loss Of One Channel

DIPSwitch 7 defines how the unit reacts to the loss of one channel.

DIPSwitch 7	Description
On	When on, the unit will output the remaining channel to the lost channel to create a dual mono output.
Off	When off, the unit will simply mute the lost channel and continue.

Operation On Loss Of Lock

DIPSwitch 8 defines whether the unit treats a loss of lock as a severe event or as a simple loss of level.

DIPSwitch 8	Description
On	When on, the unit will treat the loss of lock condition as severe and the unit will switch to the Aux input after 2 seconds.
Off	When off, the unit will treat the loss of lock no differently to a loss of level.

Remote Stop Operation

DIPSwitch 9 defines whether the remote stop is operational. Used for stopping external equipment when the unit recovers from a switched state.

DIPSwitch 9	Description
On	When on and Remote Start is set to momentary, pin 4 will close to pin 11 on the remote connector for half a second. Remote start operation is also affected. If no level is detected on the channel which has been switched to, the remote start pulse will repeat every ten seconds. Please note that the aux alarm will not work in this mode.
Off	When off, the aux alarm operates normally.

Switching Operation

DIPSwitch 10 defines whether the unit switches when it loses lock or the level drops, i.e. any failure condition. This is so that the unit can signal a failure without actually switching.

DIPSwitch 10	Description
On	When on, the unit will not switch on any fail condition.
Off	When off, the unit will switch on any fail condition.

Firmware Mode

DIPSwitch 12 defines the boot up mode.

DIPSwitch 12	Description
On	When on, the unit boots in bootstrap mode.
Off	When off, the unit boots in 'normal' configuration.

Remotes Connector

Displayed below are the pin connections and descriptions for the remote connector:

Pin No.	Signal	I/O	Description
Pin 1	Master Alarm Normally Open	O	Relay 1 N/O to Pin 9 in alarm state
Pin 2	Master Alarm Normally Closed	O	Relay 1 N/C to Pin 9 in alarm state
Pin 3	Aux. Alarm Normally Open	O	Relay 2 N/O to Pin 11 in alarm state
Pin 4	Aux. Alarm Normally Closed	O	Relay 2 N/C to Pin 11 in alarm state
Pin 5	Mode Switch	I	Momentary make to Pin 8
Pin 6	Mode Indicator	O	Internal Open Collector to Digital
Pin 7	Restore Switch	I	Momentary make to Pin 8
Pin 8	Digital Ground	-	-
Pin 9	Master Alarm Common	O	N/O to Pin 1, N/C to Pin 2 in alarm state
Pin 10	Max Time Whilst Latched	I	Latched make to Pin 8
Pin 11	Aux. Alarm Common	O	N/O to Pin 3, N/C to Pin 4 in alarm state
Pin 12	+5V	O	To power up to a maximum 100mA
Pin 13	Source Select Indicator	O	Internal Open Collector to Digital Ground
Pin 14	Source Select Switch	I	Momentary make to Pin 8
Pin 15	Remote Start	O	Internal Open Collector to Digital Ground

Pins 1 - 4 are for external use to replicate the alarm conditions for the Main and Auxiliary inputs.

Pins 5 - 7, 13 and 14 are to replicate the switches and indicators for the source select, mode and restore functions.

Pins 8, 9, 11 and 12 are common or voltage pins.

Pin 10 is to select remotely the 2min 5secs silence time. This may be useful for the broadcast of Remembrance Day services, or where you expect a silence of up to 2 minutes to be broadcast. The maximum silence time is set whilst the contact is latched.

Pin 15 is used to remotely start an external piece of equipment and operates on audio fail.



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RS232

The 9-way 'D' type socket connector carries a standard RS232 interface and allows direct connection to a serial port on a PC via a pin-to-pin cable. The pin assignments are as follows:

Pin 2: Transmit data

Pin 3: Receive data

Pin 5: Ground

All other pins are unused.

RB-DSD1 Inputs & Outputs

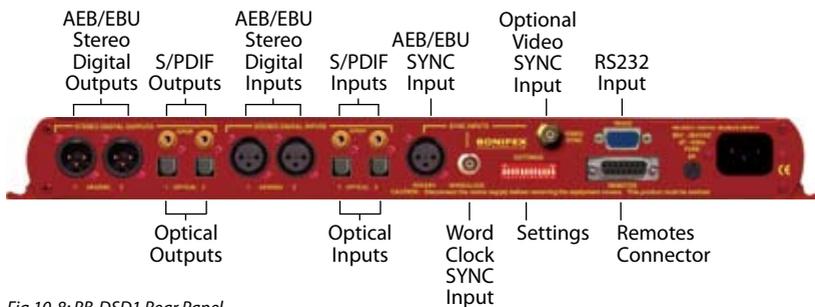


Fig 10-8: RB-DSD1 Rear Panel

AES/EBU Inputs

The digital input XLR 3 pin socket has an impedance of 110Ω. It has the following connections:

Pin 1: Screen

Pin 2: Phase

Pin 3: Non-phase

The signals on this connector should meet the IEC 60968 specification

S/PDIF Inputs

The S/PDIF digital phono input have an impedance of 75Ω.

Optical Inputs

The digital audio optical input meets the TOSLink specification used by most professional & consumer equipment.

AES/EBU Sync Input

The digital input XLR 3 pin socket has an impedance of 110Ω. It has the following connections:

Pin 1: Screen

Pin 2: Phase

Pin 3: Non-phase

The signals on this connector should meet the IEC 60968 specification

Word Clock Input

The S/PDIF digital phono input has an impedance of 75Ω.

AES/EBU Outputs

The digital output XLR 3 pin socket has an impedance of 110Ω. It has the following connections:

Pin 1: Screen.

Pin 2: Phase.

Pin 3: Non-phase.

The signals on this connector comply with the IEC 60968 specification

S/PDIF Outputs

The digital output S/PDIF phono output has an impedance of 75Ω.

Optical Outputs

The digital audio optical output meets the TOSLink specification used by most professional & consumer equipment.

The outputs have an unweighted dynamic range of at least 138dB and a THD+N noise of or better than -137dB. The data at the outputs are presented as 24 bit wide.



10 SYNCHRONISERS, DELAYS & SILENCE DETECTORS



Serial Port Control

The Serial Port allows the RB-DSD1 to be controlled and updated from a PC via a pin-to-pin serial cable, using the Sonifex Serial Control Interface (SCI) software. This software is available as a free download from the Sonifex website at www.sonifex.co.uk/sci.

Default Settings for the Serial Port

Baud Rate:	19200
Data Bits:	8
Stop Bits:	1
Parity:	Even
Handshaking:	XON/XOFF

Fig 10-9: Serial Port Default Settings

Serial Interface Commands and Responses

Most of the commands follow the same structure: a 3 letter command followed by a colon, followed by a parameter (if any) and terminated by Carriage Return with optional Line Feed. A Line Feed character may be sent but it will be ignored by the RB-DSD1. Commands are not case sensitive.

Responses are CR & LF terminated.

After the RB-DSD1 has been powered-up, an initialisation string is sent

"Initialising DSD1".

Following are the commands and the expected responses:

Serial Protocol RB-DSD1

Command	Description	Response
AOM:nn	nn represents which channel has been chosen 00 = Manual return 01 = Auto return	-ACK:
Bnn:	Baudrate change where nn is the new baudrate value where: 11 – 115200 57 – 57600 38 – 38400 19 – 19200 96 – 9600	-ACK:

Command	Description	Response
DEL:nnn	Delay length in seconds where nnn is the number of seconds between 2 and 252	-ACK:
DFD:nnn	Default delay length in seconds where nnn is the number of seconds between 2 and 252	-ACK:
DWN:	Initiates a firmware upgrade	-ACK:
FPS:	<p>Front panel and unit status where response is: -FPS:aa_bb_cc_dd_ee_ff_gg_hh_iii_jjj_kk_llllll_mmn_nnn aa = Input1 source selection bb = Input2 source selection where 00 = AES, 01 = SPDIF, 02 = TOSLINK cc = Sync mode selection where 00 = Master, 01 = Auto, 02 = Auto lock, 03 = Slave mode dd = Serial flag indication where 00: Serial mode off, 01 = Serial mode on ee = Frequency where 00 = 32k, 01 = 44.1k, 02 = 48k, 03 = 88.2k, 04 = 96k, 05 = 176.4k, 06 = 192k ff = Mono/stereo where 00 = mono, 01 = stereo gg = Sync from where 00 = Input 1, 01 = Input 2, 02 = AES sync, 03 = Word clock, 04 = Video sync hh = Rear DIPSwitch settings where hh is a hex value built from the sum of all applicable from: 01 = Remote stop selected 02 = No switching 08 = Input 2 as main source 10 = Auto mode selected 20 = Remote start is momentary 40 = Dual mono on lost channel in stereo mode 80 = Fail immediately on loss of lock</p>	



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Command	Description	Response
	iii: Default delay jjj: Delay time settings where iii and jjj are a value in seconds between 2 and 252 kk: Remote settings where 80 = Pin 10 is actioned on remote port llllll: Level settings where llllll is a 24 bit hex value defining the current switch level. mmm: Recovery time 1 where mmm is the current recovery time set for MAIN between 2 and 252 in seconds nnn: Recovery time 2 where nnn is the current recovery time set for AUX between 2 and 252 in seconds	
FRQ:nn	Output framerate selection where nn selects which framerate is selected for the output where: 00 = 32k 01 = 44.1k 02 = 48k 03 = 88.2k 04 = 96k 05 = 176.4k 06 = 192k	-ACK:
IOD:nn	Fail immediately or delay on loss of lock where nn selects between the two modes 00 = Use standard delay on loss of lock 01 = Fail immediately	-ACK:
LEV:nn	Level settings in dBFS nn sets the level of silence between -39 and -84	-ACK:
MIS:nn	Switch which input is represented on the MAIN side where nn represents which channel has been chosen 00 = Main is input 1 01 = Main is input 2	-ACK:
MOD:nn	Sync mode selection where nn selects the synchronization mode where: 00 = Master mode 01 = Auto mode 02 = Auto lock mode 03 = Slave	-ACK:

Command	Description	Response
MOS:nn	Mono or stereo selection where nn selects between the two options where: 00 = Mono mode 01 = Stereo mode	-ACK:
NOS:nn	No switch on failure where nn selects between the two options where: 00 = Switch on failure 01 = No switch on failure	-ACK:
OCL:nn	one channel (left or right) lost mode where nn selects between two options where: 00 = Mute the lost channel 01 = Send remaining channel to the other (dual mono)	-ACK:
REM:nm	Remote Control options where n selects which option is being set where: 1 = Remote start mode 2 = Remote stop mode 3 = Remote Restore select m selects the state of the option chosen where: 0 = Option deselected or off 1 = Option selected	-ACK:
RTx:nnn	x sets the channel which is being changed where: 1 = MAIN 2 = AUX nnn sets the amount of seconds it takes for the channel to recover in seconds	-ACK:
SPM:nm	Special mode selection where n selects which modes is being selected where: 1 = signal detection mode 2 = Force main mode 3 = Emergency program override 4 = Detection enabled/disabled mode m selects the state of the option chosen where: 0 = Mode turned off 1 = Mode selected	-ACK:



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Command	Description	Response
SRQ:	<p>Status Request where</p> <p>aa = Input1 lock status</p> <p>bb = Input2 lock status</p> <p>where 00 = unlocked and 01 = locked</p> <p>cc = Restore flash</p> <p>dd = Sync flash</p> <p>where 00 = Not flashing and 01 = flashing</p> <p>ee = Output switched</p> <p>where 00 = Switched and 01 = Not switched</p> <p>fg:</p> <p>f = alarm status and g = program level status</p> <p>where f and g are hex values built from the sum of:</p> <p>1 = Input 1 L</p> <p>2 = Input 1 R</p> <p>4 = Input 2 L</p> <p>8 = Input 2 R</p>	-SRQ:aa_bb_cc_dd_ee_fg_hhhh_ii
	<p>For example if input 1 was in a good state but input 2 was alarmed the value would be C3</p> <p>Alarm status = Input 2L + Input 2R = 4+8 = C</p> <p>Program level status = Input 1L + Input 1R = 1+2=3</p> <p>hhhh: Remote port status</p> <p>where hhhh represents hex values built from the sum of:</p> <p>01: Main alarm</p> <p>02: Aux alarm\remote stop indicator</p> <p>04: Auto indicator</p> <p>08: Remote start indicator</p> <p>10: Source indicator</p>	
	<p>Follow above example for further clarification</p> <p>ii: Current special mode</p> <p>where ii is a hex value based on which mode is selected:</p> <p>00: No special mode</p> <p>01: Signal detection mode</p> <p>02: Force main mode</p> <p>04: Emergency override mode</p> <p>08: Detection enabled/disabled mode</p>	

Command	Description	Response
SSx:nn	Source select where x selects which input is being changed where: 1 = MAIN 2 = AUX nn selects which source is used for that particular input where: 00 = AES 01 = SPDIF 02 = Optical	-ACK:
SYS:nn	Sync source select where nn selects which sync source is used where: 00 = Input 1 01 = Input 2 02 = AES 03 = Wordlock 04 = Video	-ACK:
UID:	Unit id where	-UID:RB-DSD1

VER: Version number where -VER:x.xxx,y.yyy
Where x.xxx is the firmware version and y.yyy id the front panel firmware version number

Error Messages

The following error messages can be returned for illegal commands

- Err:01 = Return if command not found
- Err:02 = Return if missing parameter
- Err:04 = Return if parameter out of range



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SCI for RB-DSD1

The free of charge Sonifex SCI software allows the user to control the RB-DSD1 remotely. The interface has three tabs including a unit setup page, an indication page and a miscellaneous options page. The status of the connection, serial number and firmware versions are always visible at the bottom of the screen.

Please note: In order to edit the options on the Unit Setup and Miscellaneous pages, the RB-DSD1 must have the front panel SERIAL mode DIPswitch set to on (DIPswitch 8, see page 113).

Indication Page

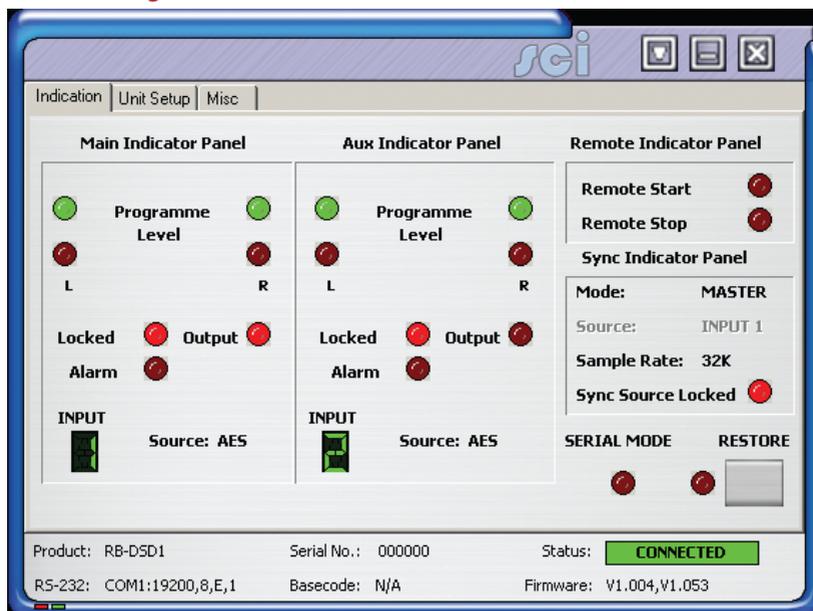


Fig 10-10: Indication Page

This page displays the current status of the unit. Each channel (Main and Aux) has an indicator panel which displays:

Program Level: The current audio status is displayed as it is on the front panel.

Locked LED: If the input is locked, this LED will be lit.

Alarm LED: If the input is alarmed, this LED will be lit.

Output LED: If the input is being used as the current output, this LED will be lit.

Input Segment Display: The number in this box indicates which input is currently being used for this channel's input.

Source Label: The source which is currently being used is displayed here.

Remote Start & Stop: The remote start and remote stop signals are displayed in the remote indicator panel. When either of these signals is active, the relevant led will be lit.

Synchronization options are displayed in the sync indicator panel:

Mode: This displays the selected sync mode.

Source: This displays the selected sync source. This is disabled in Master mode.

Sample Rate: This displays the current output sample rate.

Sync Source Locked LED: If the synchronization source is locked this LED will be lit.

Restore: The Restore button can be used to complete the restore function if the unit is in manual mode. The LED (to the left of the button) will flash if the unit is ready to be restored. If the LED is lit continuously, the unit is in auto mode.

Serial Mode: The serial mode LED indicates whether the unit is in serial mode.

Unit Setup Page

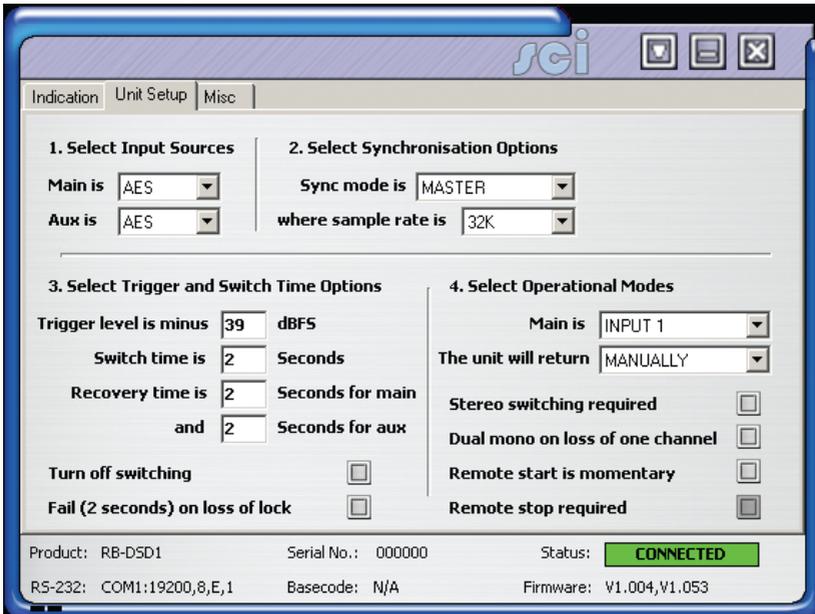


Fig 10-11: Unit Setup Page

The control page is where you can configure your unit. For ease of use, the options have been split into four sections:

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1. Select Input Sources

Select which source you would like to use for both the Main Input and the Aux Input from the drop down boxes labeled "Main is" and "Aux is" respectively.

2. Select Synchronization Options

Select which sync mode to use from the first drop down box, labeled "Sync Mode Is". If "MASTER" is chosen, then simply select your output sample rate from the drop down box labeled "where sample rate is". If "AUTO", "AUTOLOCK" or "SLAVE" are selected, then a new drop down box labeled "From" will appear which allows the user to select the synchronization source.

3. Select Trigger And Switch Time Options

Select the trigger level in the first box by typing a value in the box labeled "Trigger level is minus" and pressing return to confirm. The highest value you can enter is 39 and the lowest is 84. Please remember that the value entered is actually a negative value.

The switch time (the time it takes for a low level channel to be registered as a failure and consequently switched to Aux) is determined by the value in the box labeled "Switch time is". Again, type a value in the box and press return to confirm it. "2" is the minimum value accepted, "252" is the maximum.

Please note that if the unit is not in Serial Mode and all front panel time switches are off, you can set the default delay by typing in the desired value between 2 - 252 in the 'Switch Time is' box and press return.

The Recovery time (the time needed for a level to be confirmed as good after an alarm state) can be set in the box labeled "Recovery time is" for the main input and the box labeled "and" for Aux.

There are two tick options that can be selected. Click on the check box to select the option:

- Turn off switching
- Fail (2 seconds) on loss of lock

4. Select Operational Modes

Select which input is the main input by using the drop down box labeled "Main is".

Select how the unit will return from failure by using the drop down box labeled "The Unit Will Return".

There are four tick options that can be selected. Click on the check box to select the option:

- Stereo switching required
- Dual mono on loss of one channel - Dual mono the output when the left or right channel is lost in stereo switch mode
- Remote start is momentary, rather than latched
- Remote stop required - If remote start is momentary, then remote stop can be selected.

Miscellaneous Page

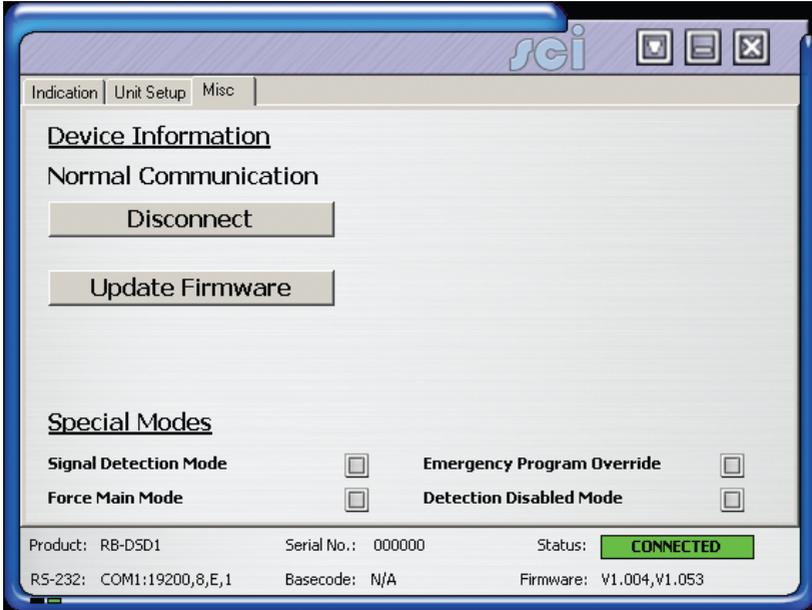


Fig 10-12: Miscellaneous Page

This page is used for the connecting and disconnecting SCi to the unit and for updating the firmware. The Special Modes are also selected from here.

Updating The Firmware

The RB-DSD1 firmware will at times be updated to add new features, or to correct any possible issues that may arise. Check for updates at:

<http://www.sonifex.co.uk/technical/software/>

To update the firmware click on the button labeled "Update Firmware" and then select the downloaded firmware file. Firmware files for the RB-DSD1 always have an ".ldr" extension. A progress bar will appear in SCi, indicating how much of the file has been uploaded to the unit.

When the unit switches to update mode, the front panel display leds will all be extinguished. The left program LED is then used to display the status of the upload:

Uploading the Code: The LED will begin to flash amber to confirm the unit is receiving the new firmware to RAM.

Copying Code To Flash Memory: The LED will be solid amber while the unit checks the integrity of the file and copies the file from RAM to flash memory.

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Successful Update: The LED will turn green for two seconds and the unit will automatically reset and begin running the new code.

Unsuccessful Update: The led will turn red for two seconds and the unit will return to running the last code used.

Special Modes

Signal Detection Mode

In this mode the three remote buttons are disabled (mode, source and restore). The recovery time is automatically set to zero for both channels. All other functions operate as in normal mode.

Force Main Mode

In this mode only the remote mode control input has any effect. When this control input is active (low) the output is forced to Input 2 (MAIN input) regardless of whether audio is present. The unit operates in auto mode and with input 2 as the MAIN input. All other functions operate as in normal mode.

Emergency Program Override Mode

In this mode all alarm LEDs are illuminated whenever the main input is active. This is to indicate that the emergency paging program is active on the MAIN input. When the paging program goes silent, the unit will revert back to the AUX input and all the alarm LEDs will switch off. All other functions operate as in normal mode.

Detection Disable/Enable Mode

In this mode the silence detection can be disabled and enabled remotely. The SCI "Detection is" drop down box and the remote source button is used to switch between the two modes. The "Detection is" drop down box toggles the state, while the remote source control (pin 14 on the remote connector) can only ENABLE the silence detection feature. Pin 10 on the remote connector is used to DISABLE silence detection. When the unit powers up Silence Detection is always enabled. The remote source select indicator (Pin 13 on the remote connector) is enabled when Silence Detection is disabled. In this mode Input 1 is always the MAIN input and Input 2 is always the AUX input. The AUX alarms are masked for both front panel indicators and remote alarm relay, for as long as the MAIN input is present.

When Special Modes are selected, certain controls are disabled or altered:

Signal Detection Mode: Main Input and return mode selection is disabled. Recovery times are set to zero.

Force Main Mode: Main Input and return mode selection is disabled.

Emergency Program Override Mode: Main Input selection is disabled.

Detection Enabled/Disabled Mode: The Main Input selection label becomes "Detection Is" and the options available change to "ENABLED" and "DISABLED".

Technical Specification For RB-DSD1

Audio Specification

Dynamic Range:	>138dB
Distortion and Noise:	<-137dB THD + N at 1kHz, ref 0dB FS
Input & Output Impedances:	110Ω ±20% AES/EBU balanced I/O 75Ω ±5% S/PDIF unbalanced I/O 75Ω ±5% TOSlink unbalanced I/O 50Ω BNC TTL word clock input
Signal Level:	Balanced: 3V/10V peak to peak min/max Unbalanced: Min 0.5V±20% peak to peak
Sample Frequencies:	32, 44.1, 48, 88.2, 96, 176.4 or 192kHz
Bit Depth:	Up to and including 24 bit

Front Panel Operational Controls & Indicators

Digital Input Select:	AES/EBU, S/PDIF or TOSlink optical via INPUT 1 or INPUT 2 push-buttons
Sync Input Select:	AES/EBU, wordclock, INPUT 1, INPUT 2 or video board, via front panel DIPSwitch
Sync Mode Select:	Master, slave, auto or auto lock, via SYNC push-button
Alarm Threshold:	-39dBfs to -84dBfs in 3dBfs steps via front panel DIPSwitches
Silence Detect Duration:	0 - 252 seconds in 2 second intervals via front panel DIPSwitches
Detection Type:	Mono or stereo, via front panel DIPSwitch
Restore Control:	Manual restore button & mode indication LED
Indicators:	Program and alarm indicators for left and right sources for both main and auxiliary inputs



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Rear Panel Operational Controls

Master Frequency Select:	32, 44.1, 48, 88.2, 96, 176.4 or 192kHz via rear panel DIPSwitches
Input Select:	Main input from INPUT 1 or INPUT 2 via DIPSwitch
Restore Mode:	Automatic or manual, via DIPSwitch
Remote Start:	Latched or momentary, via DIPSwitch
Channel Loss:	Mute channel or mix remaining, via DIPSwitch
Sync Loss:	Switch immediately or treat as silence delay, via DIPSwitch

Connections

Digital Inputs:	2 x AES/EBU XLR 3 pin female 2 x S/PDIF RCA phono 2 x TOSLink optical input
Digital Outputs:	2 x AES/EBU XLR 3 pin plug 2 x S/PDIF RCA phono socket 2 x TOSLink optical output
Sync Inputs:	1 x AES/EBU XLR 3 pin female 1 x Word Clock BNC 1 x Video Input (optional)
Remote I/O Port:	15 way D-type plug
Serial Port:	RS232, RJ45 socket
Mains Input:	Universal filtered IEC, continuously rated 85-264VAC @47- 63Hz, max 10W
Fuse Rating:	Anti-surge fuse 1A 20 x 5mm

Equipment Type

RB-DSD1:	Digital silence detection unit
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Physical Specifications

Dimensions (Raw): 48cm (W) x 10.8cm (D*) x 4.2cm (H) (1U)
19" (W) x 4.3" (D*) x 1.7" (H) (1U)

Dimensions (Boxed): 59cm (W) x 27.5cm (D) x 11cm (H)
23.2" (W) x 10.8" (D) x 4.3" (H)

Weight: Nett: 1.4kg Gross: 2.0kg
 Nett: 3.1lb Gross: 4.4lb



Accessories

RB-SYA: Analogue video sync board (NTSC, PAL & SECAM)

RB-SYD: Digital video sync board (SD-SDI & HD-SDI)

RB-RK3: 1U Rear panel rack kit for large Redboxes

* **Note** that this product is deeper than standard Redboxes



11 Connectors & Cabling

Many of the problems associated with installing and maintaining studio equipment are due to the use of poor cables or faulty connections. It is recommended that, wherever possible, pre-wired cables are purchased from recommended manufacturers. If you need bespoke cables making, please ensure that a qualified engineer carries out the work.

The main types of connectors used with the Redboxes are the following:

XLR 3 Pin Connectors

The following diagram shows the pin details for the 3 pin XLR sockets and plugs:

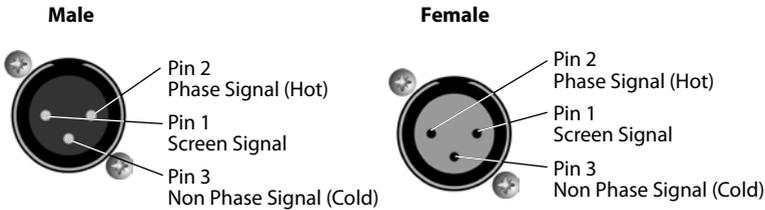


Fig 11-1: XLR Connectors.

RCA Phono Connectors

The following diagram shows the connection details for the RCA phono connectors:

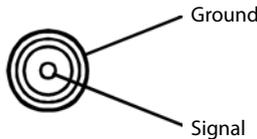


Fig 11-2: RCA Phono Connector.

1/4" Jack Connector

The following diagram shows the connection details for the jack connector:

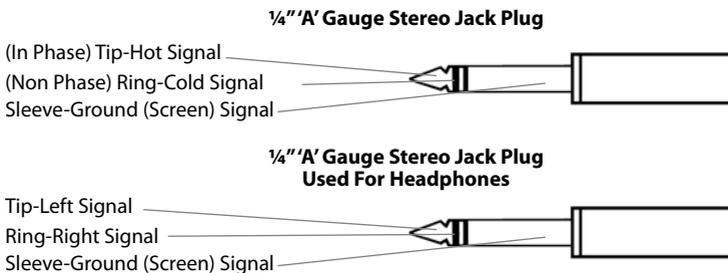


Fig 11-3: 1/4" Jack Connector.

D-Type Connectors

The following diagram shows the connection details for the D-type connectors as viewed from the rear of a Redbox :

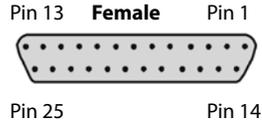
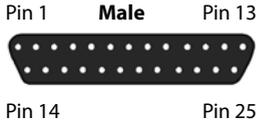


Fig 11-4: 25 Way D-Type Connectors.

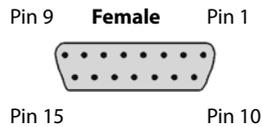
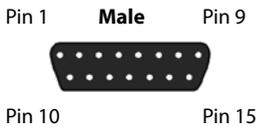


Fig 11-5: 15 Way D-Type Connectors.

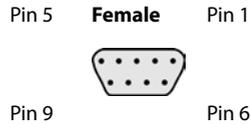
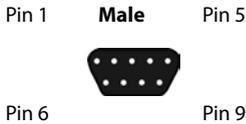


Fig 11-6: 9 Way D-Type Connectors.





12. Glossary	
1kHz	1000Hz or 1 kilohertz tone of 1kHz is often used for line-up and testing.
ADSL	Asynchronous Digital Subscriber Link - a broadband delivery system over standard telephone lines (between the exchange and the user).
AES	Audio Engineering Society.
AES/EBU	Professional digital audio standard covering frame format, connections and interfaces. Interface is usually on XLR sockets and plugs.
AM	Amplitude Modulation - older analogue modulation standard used on long, medium and short wave.
APT-X	Data reduction process for reducing the amount of storage or bit-rate need for audio.
ATRAC	Adaptive Transform Acoustic Coding - the data reduction of compression scheme used in Mini-disk machines.
Attenuation	The reduction of a signal level. Attenuation is usually measured in dB.
B-Channel	Bearer Channel - the main carrier channels used in ISDN.
Balance	The relative levels of the left and right channels of a stereo signal.
Balanced Audio	A way of sending audio over cable that protects it from induced interference.
CAR	See racks room.
CAT 5	Category 5 - type of cabling/connectivity standards used in computer networking.
CD	Compact Disc.
Cleanfeed	A cleanfeed is a signal produced by a telco module on a mixer which is used as the output to be fed back to a caller on a telephone line. The cleanfeed is a sum of all the other signals which constitute the programme output, except for the caller's audio. A cleanfeed signal will generally be of a better quality than a mix-minus signal.
Clipping	The onset of severe distortion in the signal path, usually caused by the peak signal voltage being limited.
CMRR	Common Mode Rejection Ratio. This is the ratio of the extent to which a differential amplifier will cancel noise, which is present on both inputs, compared to its ability to amplify the signal.
CODEC	COder-DECoder - used to change a signal from one format to another and back again.
Codecs	Using digital circuits being offered by telecom providers, usually ISDN (Integrated Services Data Networks) or other data transfer methods, audio can be transferred with good to excellent quality over what almost amount to dial-up telephone lines. The units operate by Coding the audio into a digital data stream that can be transferred over the digital circuit which is Decoded at the receiving end (Thus CoDec). By using single or multiple circuits, very high quality audio can be transferred from one place to another, even internationally.
Cross-talk	This is the amount of a signal from a bus which appears, or is induced, on a different signal. The problem is usually most prevalent with adjacent channels.
DA	Distribution Amplifier.
DAB	Digital Audio Broadcasting = now called Digital Radio.



DAT	Digital Audio Tape.
dB (decibel)	A ratio of two voltages or signal levels, expressed by the equation $dB=20\text{LOG}(V1/V2)$ Adding the suffix "u" denotes that the signal is relative to 0.775V RMS. Adding the suffix "v" denotes that the signal is relative to 1V RMS.
DCF	This is radio code signal sourced from Mainflingen, near Frankfurt in Germany, which can be used to automatically synchronize the Sentinel+ audio logger and PC time to an accurate atomic clock, European Time.
DDS & DDS-2	DDS and DDS-2 are world-wide standards established by Sony and Hewlett-Packard which introduce many levels of error checking as data is written to a DAT tape. DDS(-2) drives and tapes are used in security products all over the world. 120m DDS-2 DAT tapes are more accurate and reliable than DDS tapes and are able to store twice as much audio data.
DI	Direct Inject - a means of driving audio, usually from an instrument such as a guitar, to allow the audio signal to be fed directly into a sound desk input.
DLS	Dynamic Label segment - a text message scheme used in digital radio, usually to carry information about the programme. Like the radio-text scheme used in RDS.
DSP	Digital Signal Processing.
EBU	European Broadcasting Union.
EIN	Equivalent input noise. It is the ratio of output noise to the gain. It describes the level of noise which would need to be fed into an ideal amplifier to produce the measured output noise.
ENG	(Electronic news gathering). Machines that can record audio and video information digitally.
EON	(RDS) Enhance Other Networks - a scheme for switching a listener's radio to another Effective Radiated Power.
EQ	(Equalisation). This is a method of cutting or boosting selected bands of frequencies in a signal.
FM	Frequency Modulation - a way of sending audio or data over a radio carrier, the 88 - 108 Broadcast Band is often known as the FM Band.
FSD	Full Scale Digits - This is the analogue representation of the maximum digital value that the converter in question can see before digital clipping occurs.
3G	Third generation mobile telephone system that may allow broadband transmission of Global Positioning System.
Gain	The degree of amplification, or attenuation, applied to a signal.
GRAM	Gramophone Reproducer - a turntable with a 'pick-up' amplifier and remote start interface.
HDLC	High Speed Data Link.
Hybrid	See TBU
IEC	International Electrotechnical Commission - often used to refer to a mains connectivity standard, i.e. A type of plug/socket similar to that of a kettle, used on most pro-audio equipment.
ISDN	Integrated Services Digital Network - a system for sending high bandwidth material over standard telephone lines.

12 GLOSSARY



GLOSSARY

kbps	Kilo-bits per second - the number of 1s or 0s transmitted or transferred in one second.
kHz	Kilo Hertz (samples per second).
LAN	Local Area Network - scheme where several computers on the same premises are connected together so that they can exchange data.
LCD	Liquid Crystal Display (grey displays e.g. on calculators).
LED	Light Emitting Diode (small lights e.g. on computers).
LNB	Low Noise Block - the RF amplifier attached to a satellite dish.
MCR	See racks room.
MDU	Mains Distribution Unit - a panel of mains outlets for power distribution.
MF	Medium Frequency - another name for Medium Wave - roughly frequencies between 300kHz and 3MHz.
MHz	Mega-hertz - One million cycles per second.
Mix-minus	A mix-minus is similar to a cleanfeed except that the caller's voice is removed from the signal electronically. Due to phasing problems at the signal band edges, the mix-minus method often produces a lower grade signal.
Mono	Monophonic sound (system of broadcasting, recording or reproducing sound) using only one channel between source and loudspeaker.
MP3	MPEG 1 Layer 3 - see MPEG.
MPEG	(ISO MPEG) Motion Picture Experts Group. For audio, this is used to refer to a framing format standard. There are several layers and variants. Most common ones are listed below. MPEG 1 Layer 1 Used for Digital Compact Cassette. MPEG 1 Layer 2 (Musicam) used in digital radio, digital terrestrial television, ISDN and many hard-disk storage systems. MPEG 1 Layer 3 Known as MP3. A more elaborate version of MPEG 1 Layer 2 that allows for audio storage using low bit-rates.
MSF	This is radio code signal sourced from Rugby in England, which can be used to automatically synchronize the Sentinel+ audio logger and PC time to an accurate atomic clock, UK Time.
Musicam	Masking Pattern Universal Sub-Band Encoding.
MUX	Short for Multiplex - a 'package' of digital radio services.
Pan	This controls the levels sent to the left and right outputs and is an abbreviation of 'panorama'.
PFL	(Pre Fade Listen, or Cue). This is a method of auditioning audio material independently of the programme output, without routing the signal to air. The PFL button on each channel routes the signal to the PFL bus, where it can be monitored.
PI Code	(RDS) Programme Information Code used to identify an FM RDS station.
PPM	Peak Performance Meter (BBC Specification).
Promo	Something used to promote a product.
PS Name (RDS)	Programme Service Name - 8 characters used for naming an FM RDS station.
PSU	Power Supply Unit.
PTY	(RDS and DAB) Programme Type. A code transmitted to describe the overall programme content of a radio station (Static PTY) and/or the moment by moment content (Dynamic PTY). RDS has 32 codes although 3 are used in special circumstances only.



Racks Room	Often called the Master Control Room (MCR) or Central Apparatus Room (CAR), the racks room is an engineering area containing studio routing to transmitter equipment. Outside sources, phone-in equipment, off-air and transmitter status monitoring and logging equipment (or the transmitter itself!) are stored here with limited access.
RDS	Radio Data System.
Reverb	An echo (in very basic terms).
RF	Radio Frequency.
RJ-45	A connection standard used in computer networking.
S/PDIF	Sony/Phillips Digital Interface. Domestic digital audio interface standard using phono connectors or optically using fibre-optic connectors.
SPL	Sound Pressure Level.
TA	Traffic Announcement (flag) - a means of signalling to a radio with FM RDS indicating that traffic/travel news is being broadcast.
T/B	Talk-back (and reverse talk-back). This is the ability for two or more studios to communicate with each other off-air.
TBU	See Telephone Balance Unit
TCP/IP	Transmission Control Protocol / Internet Protocol. Computer interconnection methodology used on the internet and LANs.
Telephone Balance Unit (TBU) or Hybrid	Telephone hybrids provide the interface between professional audio equipment and the public telephone network. They provide protection for your equipment and the public telephone lines, allowing for varying line signals and line conditions. Automatically cancelling out the unwanted signal they also facilitate two-way communication down a single telephone line. In most European countries, including the UK, the TBU must be approved for connection to telephone company lines. Sonifex TBUs carry British Telecom and pan European approval.
THD	The Total Harmonic Distortion is the percentage presence of signals outside the measured reference frequency.
TMC	(RDS) Traffic Message Channel.
TP	Traffic Programme (flag) - a means of indicating to an RDS radio that a station will carry traffic announcements.
U	A height measurement for rack-mount equipment: 1U = 1.75 inches 2U = 3.5 inches 3U = 5.25 inches 4U = 7 inches
UPS	Uninterruptible Power Supply. A device which maintains power to equipment when the electricity supply fails.
VHF	Very High Frequency - frequencies between 30MHz and 300MHz.
VP	Vertical Polarisation i.e. of FM radio transmission.
VSWR	Voltage Standing Wave Ratio
VU	Voltage units
WAP	Wireless Application Protocol - a system for delivering mini-web pages to mobile phones from the Internet.
X21	A data communications protocol used in synchronous connections.
XLR	Professional, robust lockable audio connector - sometimes known as a 'Canon' after one of the earlier manufacturers.

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