HANDBOOK

# Redbox
## User Handbook No 1

| | |
|---|---|
| **RB-ADDA** | Combined A/D and D/A Converter (24 bit, 96kHz Capable) (1U) |
| **RB-ADDA2** | Combined A/D and D/A Converter (24 bit, 192kHz Capable) (1U) |
| **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 Splitter & Combiner |
| **RB-SYA** | Analogue Video Sync Board For RB-SC2 (PAL, NTSC, SECAM) |
| **RB-SYD** | Digital Video Sync Board For RB-SC2 (HD-SDI, SD-SDI) |
| **RB-SYE** | AES/EBU Sync Board For RB-TGHD(B or X) |
| **RB-SYW** | Word Clock Sync Board |
| **RB-DS2** | Stereo Delay Synchroniser & Time-Zone Delay (1U) |
| **RB-DS2R** | Remote Switch Panel For RB-DS2 (1U) |
| **RB-PD2** | Stereo Profanity Delay (1U) |
| **RB-DD4** | 4 Channel Digital Audio Delay Synchroniser (1U) |
| **RB-AEC** | Acoustic Echo Canceller (1U) |

Manufacturers of audio & video products for radio & TV broadcasters

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# Register Online for an Extended 2 Year Warranty

As standard, Sonifex products are supplied with a 1 year back to base warranty.

If you register the product online, you can increase your product warranty to 2 years and we can also keep you informed of any product design improvements or modifications.

Product: _____

Serial No: _____

To register your product, please go online to www.sonifex.co.uk/register

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# Product Warranty - 2 Year Extended

As standard, Sonifex products are supplied with a 1 year back to base warranty. In order to register the date of purchase and so that we can keep you informed of any product design improvements or modifications, it is important to complete the warranty registration online. Additionally, if you register the product on the Sonifex website, you can increase your product warranty to 2 years. Go to the Sonifex website at: http://www.sonifex.co.uk/technical/register/index.asp to apply for your 2 year warranty.

**Note:** For your own records the product serial number is recorded on the CE certification page of this handbook.

## Sonifex Warranty & Liability Terms & Conditions

### 1. Definitions

'the Company' means Sonifex Ltd and where relevant includes companies within the same group of companies as Sonifex Limited.

'the Goods' means the goods or any part thereof supplied by the Company and where relevant includes: work carried out by the Company on items supplied by the Purchaser; services supplied by the Company; and software supplied by the Company.

'the Purchaser' means the person or organisation who buys or has agreed to buy the Goods.

'the Price' means the Price of the Goods and any other charges incurred by the Company in the supply of the Goods.

'the Warranty Term' is the length of the product warranty which is usually 12 months from the date of despatch; except when the product has been registered at the Sonifex website when the Warranty Term is 24 months from the date of despatch.

'the Contract' means the quotation, these Conditions of Sale and any other document incorporated in a contract between the Company and the Purchaser.

This is the entire Contract between the parties relating to the subject matter hereof and may not be changed or terminated except in writing in accordance with the provisions of this Contract. A reference to the consent, acknowledgement, authority or agreement of the Company means in writing and only by a director of the Company.

### 2. Warranty

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 the Warranty Term 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);

(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 Your Product

Each product is shipped in protective packaging and should be inspected for damage before use. If there is any transit damage take pictures of the product packaging and 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.

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

| Item | Quantity |
|---|---|
| Product unit | 1 |
| IEC mains lead fitted with moulded mains plug | 1 |
| Handbook and warranty card | 1 |

If you require a different power lead, please let us know when ordering the product.

### Repairs & Returns

Please contact Sonifex or your supplier if you have any problems with your Sonifex product. Email technical.support@sonifex.co.uk for the repair/upgrade/returns procedure, or for support & questions regarding the product operation.

## CE Conformity

The products in this manual comply with the essential requirements of the relevant European health, safety and environmental protection legislation.

The technical justification file for this product is available at Sonifex Ltd.

The declaration of conformity can be found at:
http://www.sonifex.co.uk/declarations

## Safety & Installation of Mains Operated Equipment

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. 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.

### Voltage Setting Checks

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

### Fuse Rating

The product is supplied with a single fuse in the live conducting path of the mains power input. 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. The active fuse is fitted on the outside rear panel of the unit.

### Power Cable & Connection

An IEC power connector is supplied with the product which has a moulded plug attached.

The mains plug or IEC power connector is used as the disconnect device. The mains plug and IEC power connector shall remain readily operable to disconnect the apparatus in case of a fault or emergency.

The mains lead is automatically configured for the country that the product is being sent to, from one of:

| Territory | Voltage | IEC Lead Type | Image |
|---|---|---|---|
| UK & Middle East | 230V | UK 3 pin to IEC lead | |
| Europe | 230V | European Schuko round 2 pin to IEC lead | |
| USA, Canada and South America | 115V | 3 flat pin to IEC lead | |
| Australia & New Zealand | 230V | Australasian 3 flat pin to IEC lead | |

Connect the equipment in accordance with the connection details and before applying power to the unit, check that the machine has the correct operating voltage for your mains power supply.

This apparatus is of a class I construction. It must be connected to a mains socket outlet with a protective earthing connection.

**Important note:** If there is an earth/ground terminal on the rear panel of the product then it must be connected to Earth.

## WEEE Directive

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. All products manufactured by Sonifex Ltd have the WEEE directive label placed on the case. Sonifex Ltd will be happy to give you information about local organisations that can reprocess the product when it reaches its "end of use", or alternatively all products that have reached "end of use" can be returned to Sonifex and will be reprocessed correctly free of charge.

## Atmosphere/Environment

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

This apparatus shall not be exposed to dripping or splashing, and no objects filled with water, such as vases shall be placed on the apparatus.

## Fitting Redboxes

Redboxes can be fixed to the underside of a 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.

RK1

*Fig A: RB-RK1Small Redbox Front Rack-mount Kit .*

They can also be rack-mounted, with either the front, or rear of the Redbox positioned at the front of the rack (Note: this product is front rack-mounted as standard):

**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).

**Rear Mounting a Redbox**: For rear panel mounting you can use either the **RB-RK2** (in this case), or **RB-RK3**, depending on the size of your Redbox.

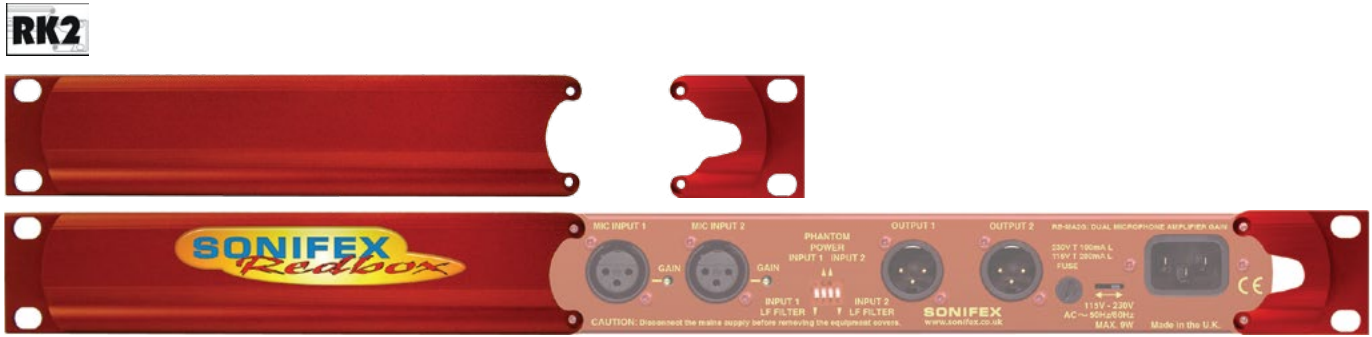



*Fig B: RB-RK2 Small Redbox Rear Rack-mount Kit.*

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

# 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:

**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 ±1dB.

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

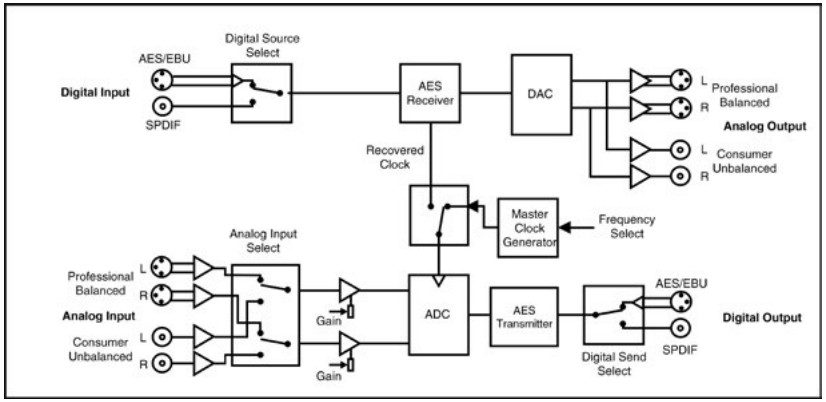



*Fig 1-2: RB-ADDA System Block Diagram*

## 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

**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 10kΩ bridging. Each XLR has the following connections:

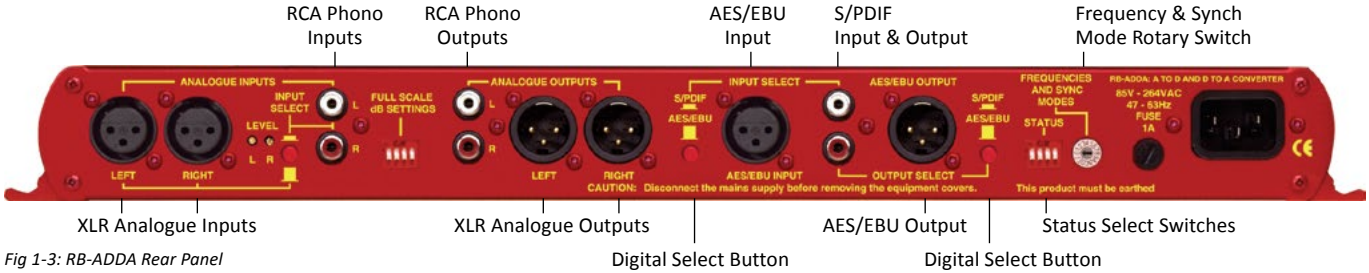



*Fig 1-3: RB-ADDA Rear Panel*

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 20kΩ.

***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 ± 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 75Ω.

**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 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 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)<br>2 x RCA phono (unbalanced) (L & R) |
| Digital Outputs: | 1 x AES/EBU XLR 3 pin plug<br>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<br>1 x S/PDIF RCA phono |
| Analogue Outputs: | 2 x XLR 3 pin male (balanced) (L & R)<br>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)<br>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 | Combined A/D and D/A converter |

| Physical Specifications | |
|---|---|
| Dimensions (Raw) | 48cm (W) x 10.8cm (D) x 4.2cm (H) (1U)<br>19” (W) x 4.3” (D) x 1.7” (H) (1U) |
| Dimensions (Boxed) | 58.5cm (W) x 22.5cm (D) x 7cm (H)<br>23” (W) x 8.9” (D) x 2.8” (H) |
| Weight | Nett: 1.6kg Gross: 2.2kg<br>Nett: 3.5lbs Gross: 4.8lbs |

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

## Introduction

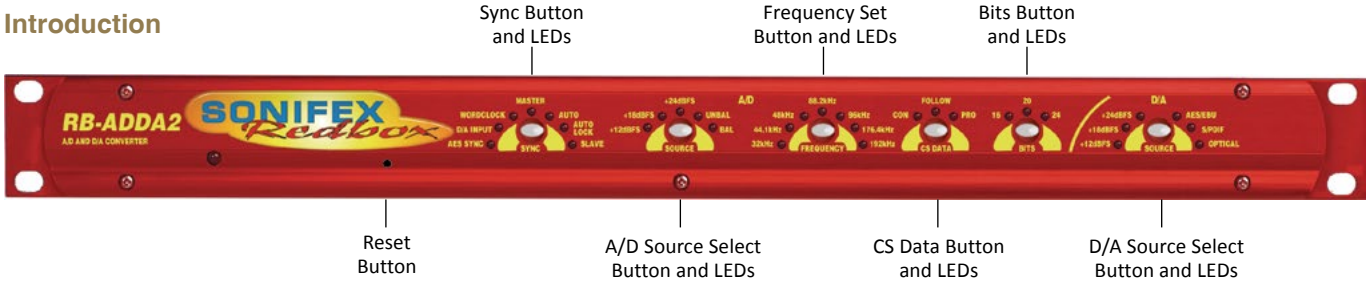



*Fig 2-1: RB-ADDA2 Front Panel*

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 ±3dB 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

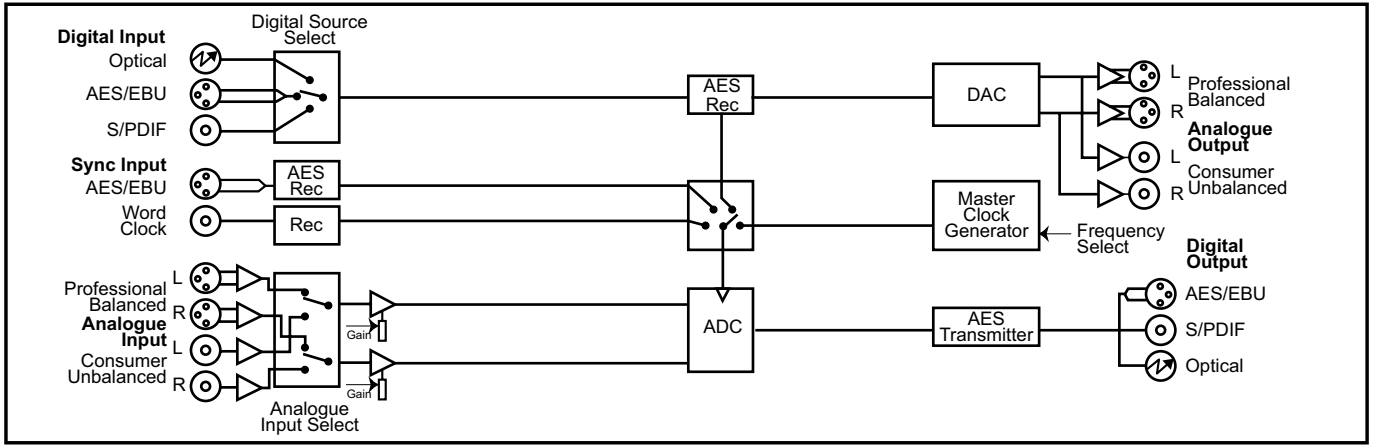



*Fig 2-2: RB-ADDA2 System Block Diagram*

## 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.

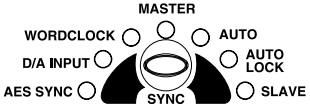



*Fig 2-3: SYNC Button & LEDs*

**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.

**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.

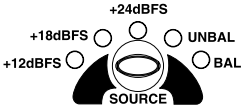



*Fig 2-4: A/D SOURCE Select Button & LEDs*

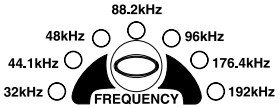



*Fig 2-5: FREQUENCY Set Button & LEDs*

**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.

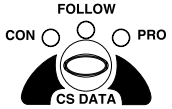



*Fig 2-6: CS DATA Button & LEDs*

**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.

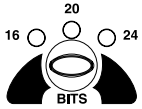



*Fig 2-7: BITS Button & LEDs*

**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.

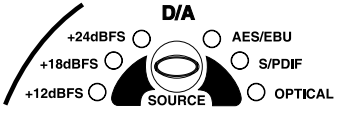



*Fig 2-8: D/A SOURCE Select Button & LEDs*

**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

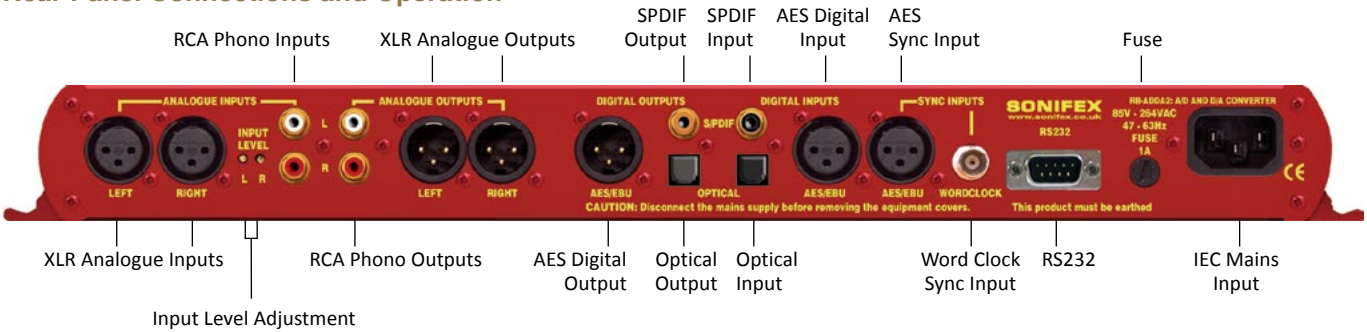



*Fig 2-9: RB-ADDA2 Rear Panel*

**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 10kΩ 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 20kΩ.

***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 ± 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 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 have an impedance of 75Ω.

***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<br>nn=00 - 16 Bits<br>nn=01 - 20 Bits<br>nn=02 - 24 Bits | - ACK: |
| ADC:nn | - Set ADC Channel Status Type<br>nn=00 - Send Consumer CS Data<br>nn=01 - Follow Sync Input (or use last set)<br>nn=02 - Send Professional CS Data | - ACK: |
| ADF:nn | - Set ADC Channel Status Type<br>nn=00 - Select 32kHz Master Clock Sample Rate<br>nn=01 - Select 44kHz Master Clock Sample Rate<br>nn=02 - Select 48kHz Master Clock Sample Rate<br>nn=03 - Select 88kHz Master Clock Sample Rate<br>nn=04 - Select 96kHz Master Clock Sample Rate<br>nn=05 - Select 176kHz Master Clock Sample Rate<br>nn=06 - Select 192kHz Master Clock Sample Rate | - ACK: |
| ADM:nn<br>For last three modes add | - Set ADC Mode & Select Sync Source<br>nn=00 - Master Mode<br>nn=04 - Auto Mode<br>nn=08 - Auto Lock Mode<br>nn=10 - Slave Mode<br>nn=01 - Select External AES Sync input as sync source<br>nn=02 - Select DA Stage input as sync source<br>nn=03 - Select Word Clock input as sync source | - ACK: |
| ADS:nn<br>And add | - Set ADC Source & Full Scale Input Level<br>nn=00 - Select Balanced XLR Inputs<br>nn=04 - Select Unbalanced Phono Inputs | - ACK: |

| | |
|---|---|
| | nn=01 - +12dBu represents Full Scale Digits (+2 for unbalanced)<br>nn=02 - +18dBu represents Full Scale Digits (+8 for unbalanced)<br>nn=03 - +24dBu represents Full Scale Digits (+14 for unbalanced) |
| Bnn: | - Baud Rate - ACK: (at old rate)<br>nn=11 (115200)<br>nn=57 (57600)<br>nn=38 (38400)<br>nn=19 (19200)<br>nn=96 (9600) |
| DAS:nn<br><br><br><br>And add- | - Set DAC Source & Full Scale Output Level - ACK:<br>nn=04 - Select AES Digital Audio Input<br>nn=08 - Select SPDIF Digital Audio Input<br>nn=10 - Select Optical Digital Audio Input<br><br>nn=01 - Full Scale Digits Sends +12dBu Out (+2 for unbalanced)<br>nn=02 - Full Scale Digits Sends +18dBu Out (+8 for unbalanced)<br>nn=03 - Full Scale Digits Sends +24dBu Out (+14 for unbalanced) |
| DWN: | - Download new code -OK: then erasing message<br>Then send S-records followed by<br>S-record terminator to<br>force programming of flash memory<br>and reboot |
| FPS: | - Request Front Panel Settings - FPS:aa_bb_cc_dd_ee_ff<br>Where aa = value from ADC Mode as ADM: above<br>bb = value from ADC Source as ADS: above<br>cc = value from ADC Frequency as ADF: above<br>dd = value from ADC Bits as ADB: above<br>ee = value from ADC CS Data Select as ADC: above<br>ff = value from DAC Source as DAS: above |
| LOC: | - Lock Front Panel - ACK: |
| MEM: | - Memory Dump - Memory, then 10 lines Madd:data<br>Where add is memory address starting at 060(hex) and data is 16 bytes of data, each byte shown as 2 hex digits |
| MFn: | - Limit Front Panel Frequency Selection - ACK:<br>Where n=0 for No Limit<br>n=1 for 48kHz Limit &<br>n=2 for 96kHz Limit |
| REG: | - Register Dump<br>- Registers, then 4 lines of 8 x Rrr:dd<br>Where rr is register number and dd is value shown as 2 hex digits |
| SER: | - Serial Number request - SER:012345 |
| SRQ: | - Status Request - STA:aa_dd_ss<br>Where aa = ADC Status<br>dd = DAC Status<br>ss = Internal Status |
| STK: | - Stack Dump - STK: aaaa, then 6 lines Sadd:data<br>Where aaaa= current Stack Pointer,<br>add is memory address starting at 0800(hex)<br>and data is 16 bytes of data, each byte shown as 2 hex digits |
| 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)<br>2 x RCA phono (unbalanced) (L & R) |
| Digital Outputs: | 1 x AES/EBU XLR 3 pin plug<br>1 x S/PDIF RCA phono socket<br>1 x TOSLink optical output |
| Sync Inputs: | 1 x AES/EBU XLR 3 pin female<br>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<br>1 x S/PDIF RCA phono<br>1 x TOSLink optical input |
| Analogue Outputs: | 2 x XLR 3 pin male (balanced) (L & R)<br>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 | Selectable +12dBu, +18dBu or +24dBu output level, ref |
| Level for FSD: | FSD, via D/A SOURCE push-button |

**Equipment Type**

| | | |
|---|---|---|
| RB-ADDA2 | RK3 | 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)<br>19” (W) x 6.2” (D*) x 1.7” (H) (1U) |
| Dimensions (Boxed) | 59cm (W) x 27.4cm (D*) x 10.8cm (H)<br>23.2” (W) x 10.8” (D*) x 4.3” (H) |
| Weight | Nett: 1.6kg Gross: 2.3kg<br>Nett: 3.5lbs Gross: 5lbs |

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

# 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

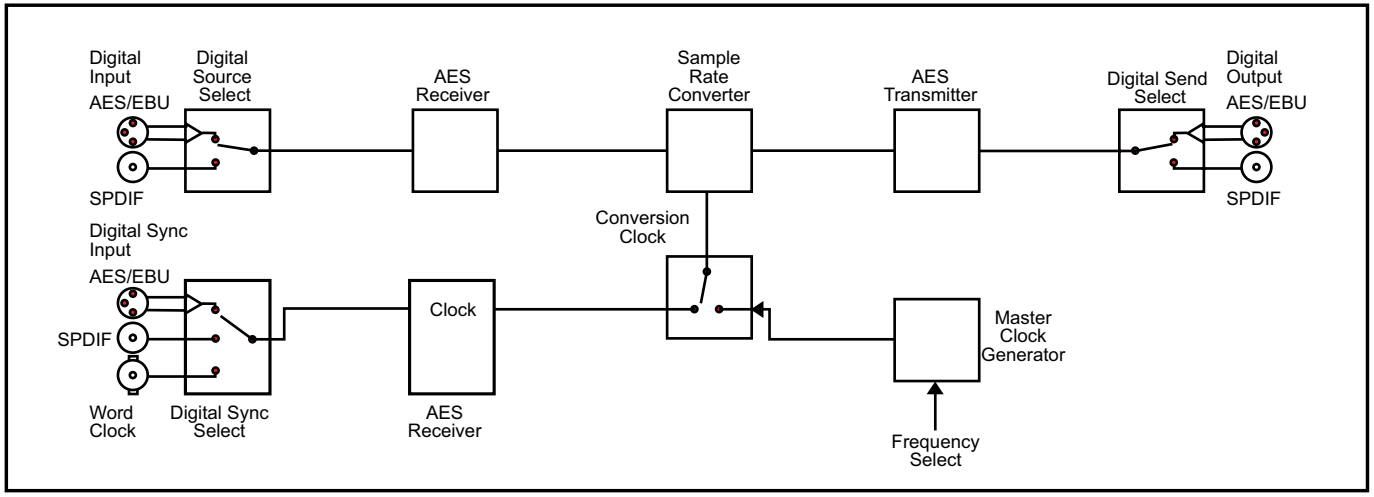



*Fig 3-2: RB-SC1 System Block Diagram*

## 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

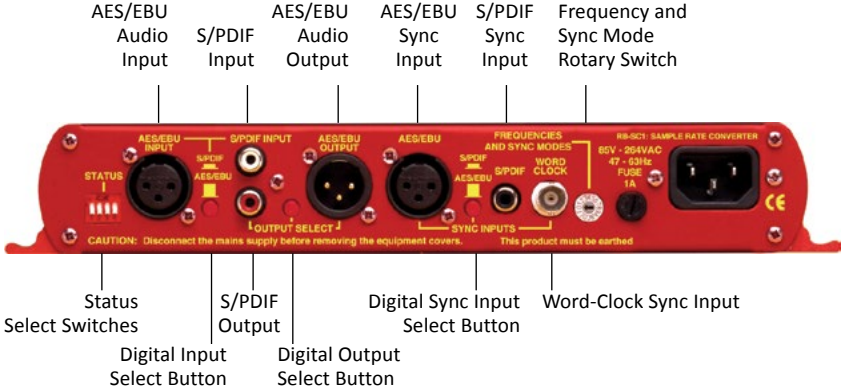



*Fig 3-3: RB-SC1 Rear Panel*

**Inputs and Outputs**

***AES/EBU Inputs***

The digital source and digital sync XLR 3 pin sockets both have an impedance of 110Ω. 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.

***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**

***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*

## 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<br>1 x S/PDIF RCA phono female<br>(Input button select between AES/EBU and S PDIF) |
| Sync Inputs: | 1 x AES/EBU XLR 3 pin female<br>1 x S/PDIF RCA phono female<br>1 x TTL BNC female<br>(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<br>1 x S/PDIF RCA phono female<br>(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 1A 20 x 5mm |
| 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 | |

| Physical Specifications | |
|---|---|
| Dimensions (Raw) | 28cm (W) x 10.8cm (D) x 4.2cm (H) (1U)<br>11” (W) x 4.3” (D) x 1.7” (H) (1U) |
| Dimensions (Boxed) | 36cm (W) x 20.5cm (D) x 6cm (H)<br>14.2” (W) x 8” (D) x 2.4” (H) |
| Weight | Nett: 1.0kg Gross: 1.4kg<br>Nett: 2.2lbs Gross: 3.1lbs |

# 4 RB-SC2 Dual Sample Rate Converter

## Introduction

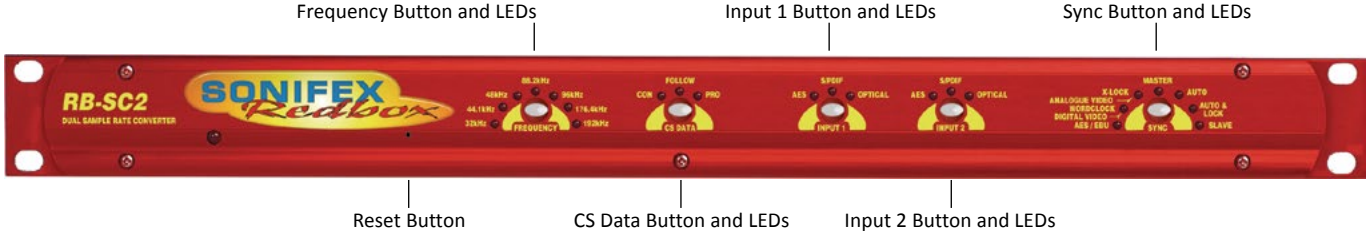



*Fig 4-1: RB-SC2 Front Panel*

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).

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.

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.

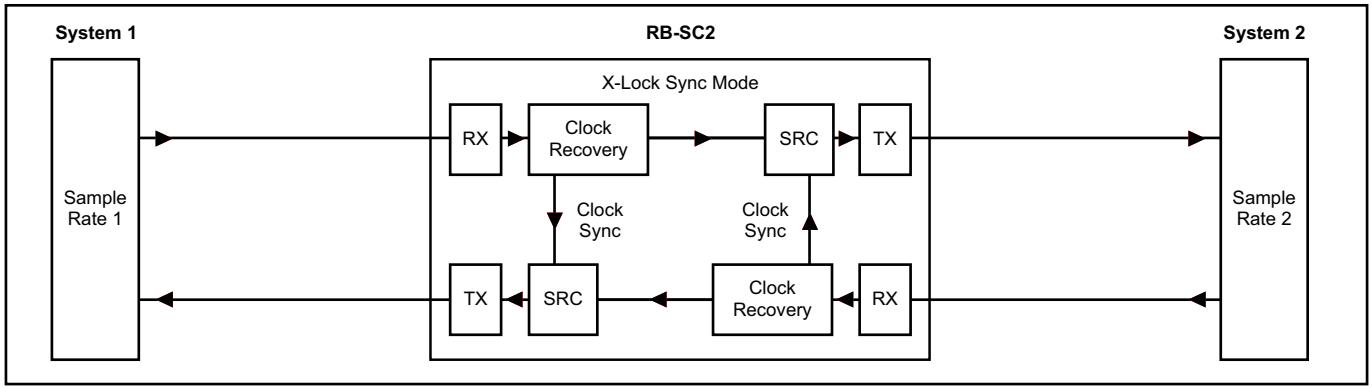



*Fig 4-2: RB-SC2 X-Lock Sync Mode*

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. 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 avaiable using the SYNC button (see page 25).

Follow these instructions to fit either of the sync boards.

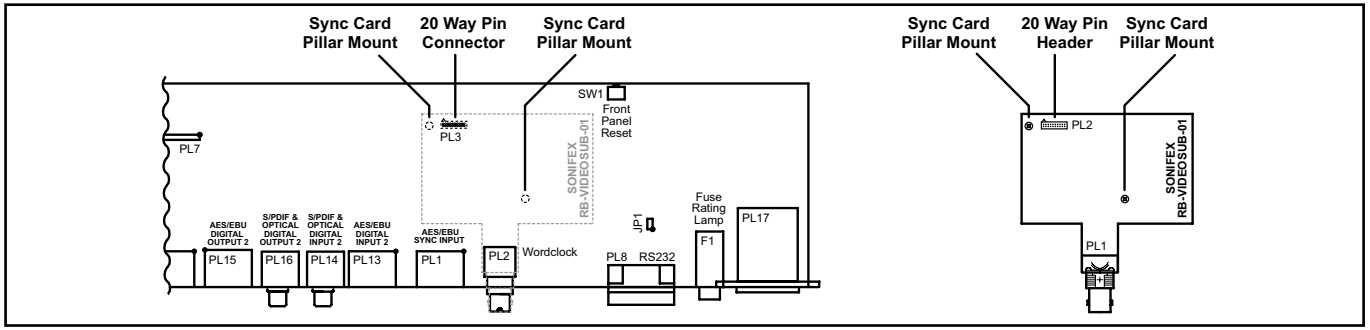



*Fig 4-3: RB-SC2 Optional Video Sync Installation*

## System Block Diagram

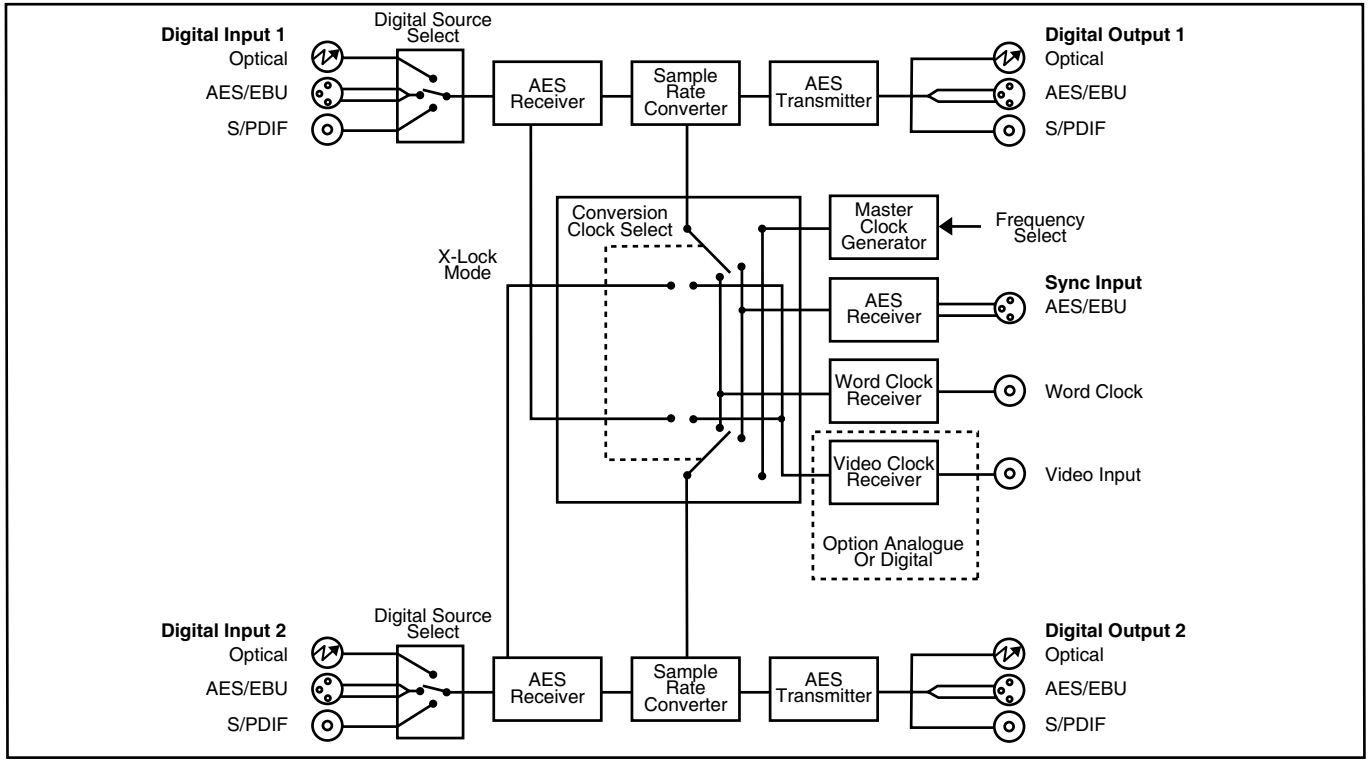



*Fig 4-4: RB-SC2 System Block Diagram*

## 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.

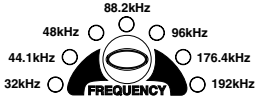



*Fig 4-5: FREQUENCY Set Button & LEDs*

**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 4-6: 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.

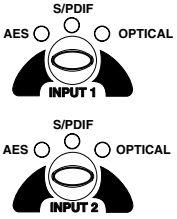



*Fig 4-7: INPUT 1 & INPUT2 Source Select Button & LEDs*

**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 and this button is also used to select the 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.

**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).

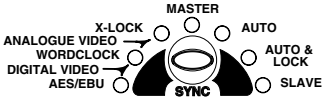



*Fig 4-8: SYNC Button & LEDs*

## Rear Panel Connections and Operation

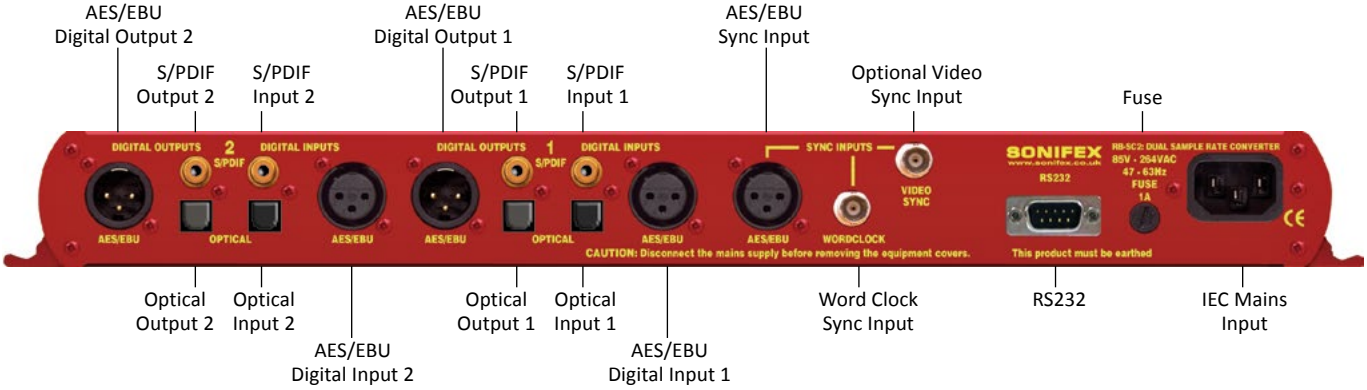



*Fig 4-9: RB-SC2 Rear Panel*

**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 4-10: 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.

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<br>nn=11 (115200)<br>nn=57 (57600)<br>nn=38 (38400)<br>nn=19 (19200)<br>nn=96 (9600) | - ACK: (at old rate) |
| CSD:nn | - Set Channel Status Type<br>nn=00 - Send Consumer CS Data<br>nn=01 - Follow Input (or use last set)<br>nn=02 - Send Professional CS Data | - ACK: |
| DWN: | - Download new code<br>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<br>Where a = Lock Status (0=unlocked, 1=locked)<br>b = Master Frequency Limit (0=no limit, 1=48kHz, 2= 96kHz) | - FPL:a_b |
| FPS: | - Request Front Panel Settings<br>Where aa = value of Sync Mode as MOD: below<br>bb = value of Master Frequency as FRQ: below<br>cc = value of Sync Frequency as FRQ: below<br>dd = value of CS Data as CSD: above<br>ee = value of Input 1 Source Select as SS1: below<br>ff = value of Input 2 Source Select as SS2: below | - FPS:aa_bb_cc_dd_ee_ff |

| | | |
|---|---|---|
| FRQ:nn | - Set Master Clock Frequency<br>nn=00 - Select 32kHz Master Clock Sample Rate<br>nn=01 - Select 44kHz Master Clock Sample Rate<br>nn=02 - Select 48kHz Master Clock Sample Rate<br>nn=03 - Select 88kHz Master Clock Sample Rate<br>nn=04 - Select 96kHz Master Clock Sample Rate<br>nn=05 - Select 176kHz Master Clock Sample Rate<br>nn=06 - Select 192kHz Master Clock Sample Rate | - ACK: |
| LOC: | - Lock Front Panel | - ACK: |
| MEM: | - Memory Dump<br>Where add is memory address starting at 060(hex)<br>and data is 16 bytes of data, each byte shown as<br>2 hex digits | - Memory, then 10 lines<br>Madd:data |
| MFQ:nn | - Limit Front Panel Frequency Selection<br>Where nn=0 for No Limit<br>nn=1 for 48kHz Limit &<br>nn=2 for 96kHz Limit | - ACK: |
| MOD:nn | - Set Mode & Select Sync Source<br>nn=00 - Master Mode<br>nn=08 - Auto Mode<br>nn=10 - Auto Lock Mode<br>nn=20 - Slave Mode<br>For last three modes add<br>nn=01 - Select External AES Sync input as sync source<br>nn=02 - Select Word Clock input as sync source<br>nn=03 - Select Other Digital input as sync source (X-Lock)<br>nn=04 - Select Optional Video input as sync source | - ACK: |
| REG: | - Register Dump<br>Where rr is register number and dd is value shown<br>as 2 hex digits | - Registers, then 4 lines of 8 x Rrr:dd |
| SER: | - Serial Number request | - SER:012345 |
| SRQ: | - Status Request<br>Where aa = SRC 1 Status<br>bb = SRC 2 Status<br>ss = Sync Status | - STA:aa_bb_ss |
| SS1:nn | - Select Channel 1 Digital Audio Source<br>nn=00 for AES/EBU<br>nn=01 for S/PDIF &<br>nn=02 for Optical | -ACK: |
| SS2:nn | - Select Channel 2 Digital Audio Source<br>nn=00 for AES/EBU<br>nn=01 for S/PDIF &<br>nn=02 for Optical | -ACK: |
| STK: | - Stack Dump<br>Where aaaa= current Stack Pointer,<br>add is memory address starting at 0800(hex)<br>and data is 16 bytes of data, each byte shown as<br>2 hex digits | - STK: aaaa, then 6 lines Sadd:data |
| UNL: | - Unlock Front Panel | - ACK: |
| UID: | - Unit ID Request | - UID:SC2 |
| VER: | - Firmware Version Request | - VER:1.23 or BOOT:1.23 |

*Fig 4-11: Serial Interface Commands and Responses*

**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

| Audio Specification | |
|---|---|
| Dynamic Range: | 138dB typical A-Weighted. |
| Distortion & Noise: | -134dB THD + N at 1kHz, ref 0 dBFS |
| Sample Freq Range: | 32kHz – 196kHz |
| Input Sample Width: | Up to and including 24 Bits. |
| Output Sample Width: | 24 Bits. |

| Connections | |
|---|---|
| Digital Inputs: | 2 x AES/EBU XLR 3 pin female<br>2 x S/PDIF RCA phono<br>2 x TOSLink optical input |
| Digital Outputs: | 2 x AES/EBU XLR 3 pin plug<br>2 x S/PDIF RCA phono socket<br>2 x TOSLink optical output |
| Sync Inputs: | 1 x AES/EBU XLR 3 pin female<br>1 x Word Clock BNC<br>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 | RK3 | Dual Stereo Sample Rate Converter, 24 bit 192kHz |

| Physical Specifications | |
|---|---|
| Dimensions (Raw) | 48cm (W) x 10.8cm (D*) x 4.2cm (H) (1U)<br>19” (W) x 4.3” (D*) x 1.7” (H) (1U) |
| Dimensions (Boxed) | 59cm (W) x 27.5cm (D) x 11cm (H)<br>23.2” (W) x10.8” (D) x 4.3” (H) |
| Weight | Nett: 1.4kg Gross: 2.0kg<br>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 |

# 5 RB-DAC1 Digital to Analogue Converter

## Introduction

*Fig 5-1: RB-DAC1 Front Panel*

24 BIT 192 kS/s

Using 24 bit, 192kHz 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

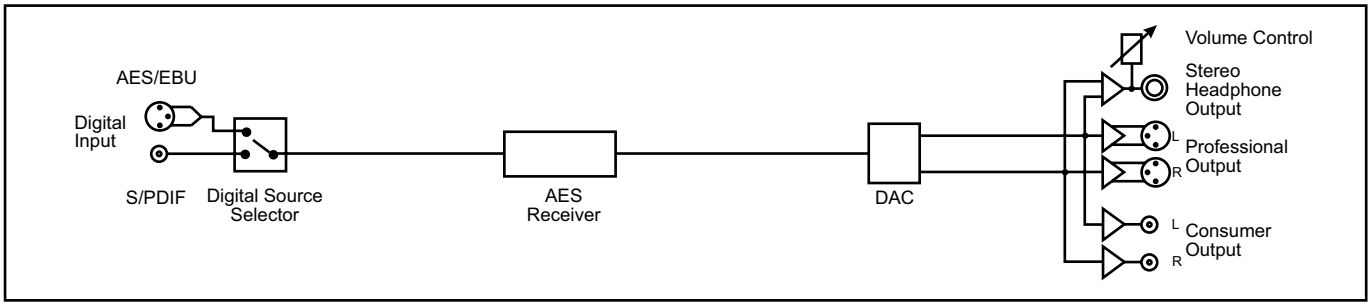



*Fig 5-2: RB-DAC1 System Block Diagram*

## 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 ¼" 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 +20dBu.

## Rear Panel Connections and Operation

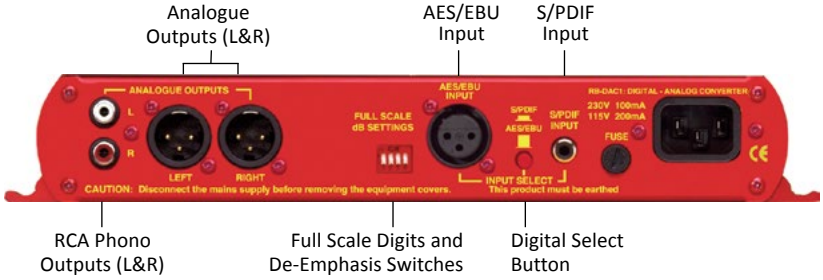



*Fig 5-4: RB-DAC1 Rear Panel*

**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<br>1 x S/PDIF RCA phono |
| Analogue Outputs: | 2 x XLR 3 pin male (balanced) (L & R)<br>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 18W max |
| Fuse Rating: | Anti-surge fuse 100mA 20 x 5mm (230VAC)<br>Anti-surge fuse 250mA 20 x 5mm (115VAC) |

| Audio Specification | Balanced Output | Unbalanced Output | Headphone Output |
|---|---|---|---|
| Maximum Output Level: | +24dBu | +15.5dBu | +20dBu (ref. <0.01% THD) |
| Output Impedance: | <50Ω | <75Ω | Drives 150mW into 32Ω to 600Ω professional headphones |
| Dynamic Range: | 115dB | 115dB | 100dB |
| Gain Range: | Selectable 12dBu, 18dBu or 24dBu output level, ref FSD | Selectable 3.5dBu, 9.5dBu or 15.5dBu output level, ref FSD | -80dBu (volume 0%) to +20dBu (volume 100%) ref FSD |
| Distortion & Noise: | <0.005% THD+N @1kHz, ref +8dBu | <0.005% THD+N @1kHz, ref -6dBV | <0.005% THD+N @1kHz, ref +8dBu |
| Sample Freq. Range: | 32kHz - 192kHz | | |

| 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 | RK1 RK2 |

| Physical Specifications | |
|---|---|
| Dimensions (Raw): | 48cm (W) x 10.8cm (D) x 4.2cm (H) (1U)<br>19" (W) x 4.3" (D) x 1.7" (H) (1U) |
| Dimensions (Boxed): | 58.5cm (W) x 22.5cm (D) x 7cm (H)<br>23" (W) x 8.9" (D) x 2.8" (H) |
| Weight: | Nett: 1.4kg Gross: 2.0kg<br>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

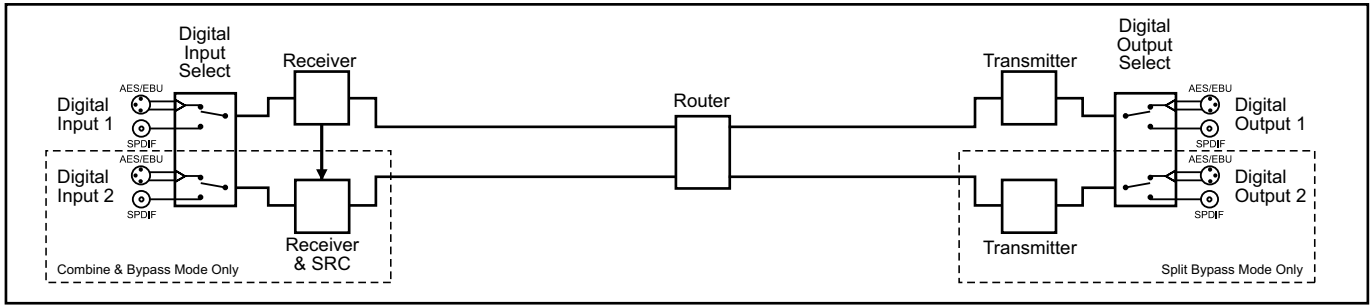



*Fig 6-2: RB-SP1 System Block Diagram*

## Front Panel Indicators & Controls

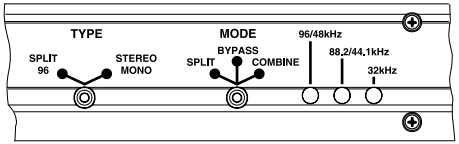



*Fig 6-3: RB-SP1 Front Panel*

**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.

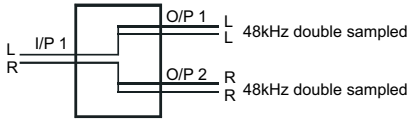
**Bypass Mode** – In Bypass mode, input 1 is routed to output 1 and input 2 is routed to output 2. Input 1 is used as the master clock input. If input 2 is at a different sample rate to input 1 then input 2 is sample rate converted to match input 1. If the signal from input 2 is lost then output 2 will 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**

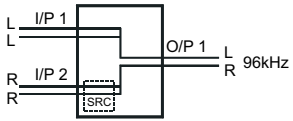
96kHz




**Combine**

48kHz double sampled

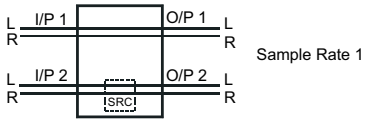
48kHz double sampled




## Bypass

Sample Rate 1

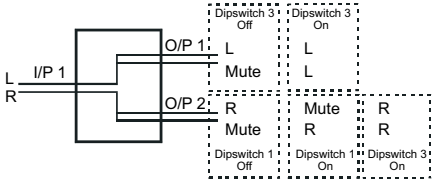
Sample Rate 2




## Stereo/Mono

**Split**

Any Sample Rate




**Combine**

Sample Rate 1

Sample Rate 2

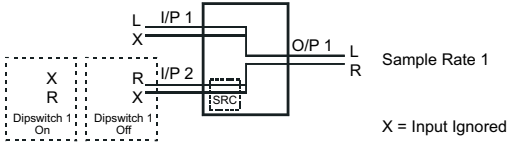



*Fig 6-4: RB-SP1 Type and Mode Flow Diagrams*

## Rear Panel Connections and Operation

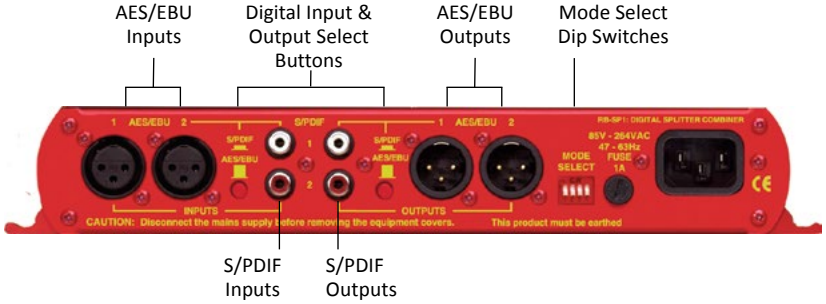



*Fig 6-5: RB-SP1 Rear Panel*

**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 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)<br>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<br>2 x S/PDIF RCA phono female<br>(Input button select between AES/EBU and S/PDIF) |
| Audio Outputs: | 2 x AES/EBU XLR 3 pin male<br>2 x S/PDIF RCA phono female<br>(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)<br>19” (W) x 4.3” (D) x 1.7” (H) (1U) |
| Dimensions (Boxed) | 58.5cm (W) x 22.5cm (D) x 7cm (H)<br>23” (W) x 8.9” (D) x 2.8” (H) |
| Weight | Nett: 1.6kg Gross: 2.2kg<br>Nett: 3.5lbs Gross: 4.8lbs |

# 7 Redbox - Synchronisation Add-On Boards

## Introduction

There are two optional video sync boards which can be used to synchronise the outputs of the RB-SC2 to a 48kHz sample rate: the RB-SYA and the RB-SYD.
There are four optional video sync boards which can be used to synchronise the outputs of the RB-TGHD(B & X): the RB-SYA, RB-SYD, RB-SYE and the RB-SYW.

## RB-SYA Analogue Video Sync Board

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

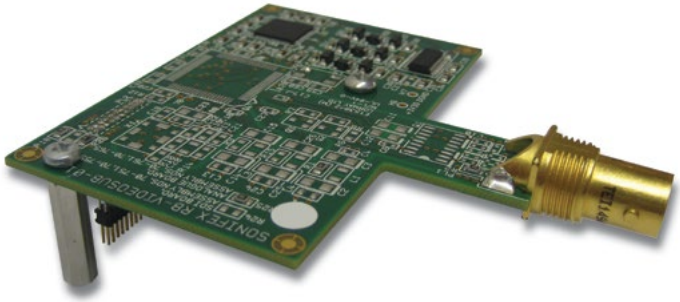


*Fig 7-1: RB-SYA Analogue Video Sync Board For RB-SC2 (PAL, NTSC, SECAM)*

## RB-SYD Digital Video Sync Board

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).

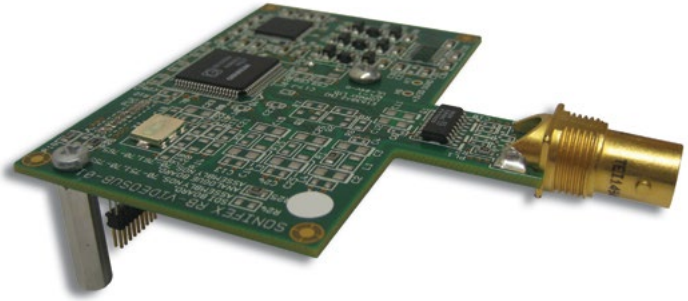


*Fig 7-2: RB-SYD Digital Video Sync Board For RB-SC2 (HD-SDI, SD-SDI)*

## RB-SYE Sync Board

The AES/EBU sync board will accept a digital audio input signal with a sample frequency between 32kHz and 192 kHz. When using the RB-SYE sync board, the Channel Status information that is encoded in the input data signal is copied to all digital output channels on the RB-TGHD.

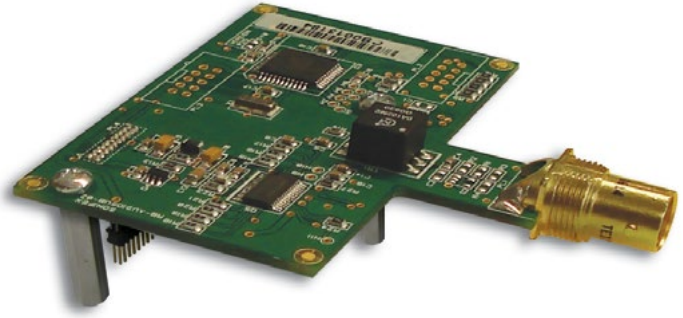


*Fig 7-3: RB-SYE Sync Board*

## RB-SYW Sync Board

The Audio Clock sync board will accept a distributed clock running at the desired sample frequency between 32 kHz and 192 kHz. Thesignal can be differential or single ended TTL level.

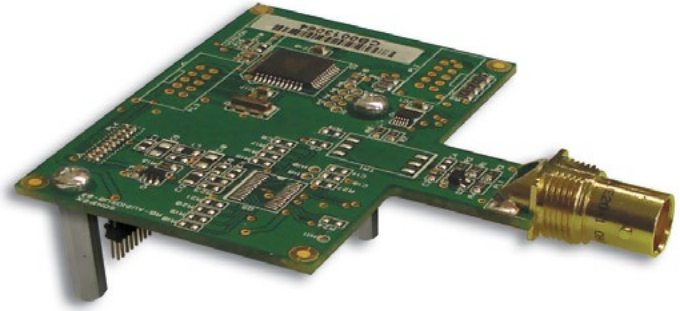


*Fig 7-4: RB-SYW Sync Board*

# 8 RB-DS2 Stereo Delay Synchroniser & Time Zone Delay

## Introduction

*Fig 8-1: 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 16GB of memory to be accessed providing delay times of over 24 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 ±3dB 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. Alternatively, 6 of the GPI/O ports can be configured to provide delay synchronous signalling where an input is delayed by the current delay before driving a corresponding output. The RB-DS2 also has an RS232 serial port for remotely controlling the unit. 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 24 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 16Gbytes - see page 54 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. There are 2 extended modes, FAT and RAW.

**Extended FAT Mode**

This mode supports cards up to 2Gbytes in size and 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.

**Extended RAW Mode**

This mode supports cards up to 16Gbytes in size and the delay synchroniser can use the full card capacity for the delay. The card does not require formatting in this mode as no filing system is created, and the contents will not be available to read in a PC card reader. This means that maximum delays for cards up to 2Gbytes will be slightly larger when using this mode compared to Extended FAT mode.

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

## System Block Diagram

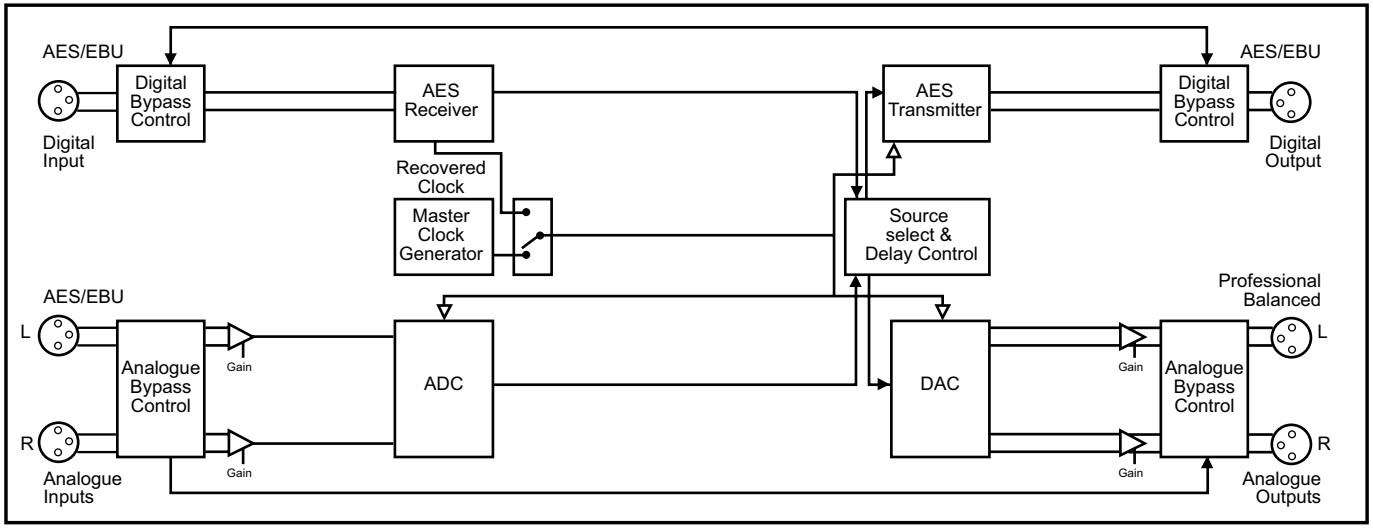



*Fig 8-2: RB-DS2 System Block Diagram*

## 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.

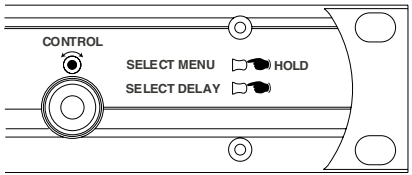



*Fig 8-3: Front Panel CONTROL*

**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 8-4: 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 10kΩ 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 ±3dB.

**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.

**XLR Analogue Outputs (Left and Right)**
The XLR 3 pin output plugs 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

**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. When the Remote Loading function is disabled, input and outputs 1-6 can be configured to provide delayed GPI/O (see System Menu: Remotes). The connector 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
Pin10: Open collector output 1
Pin11: Open collector output 2
Pin12: Open collector output 3
Pin13: Open collector output 4
Pin14: Open collector output 5
Pin15: 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 8-5) 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 43.

*Fig 8-5: The Default Screen*

### The Main Menu

To access the main menu, press and hold the rotary CONTROL knob and follow the instructions on page 44.

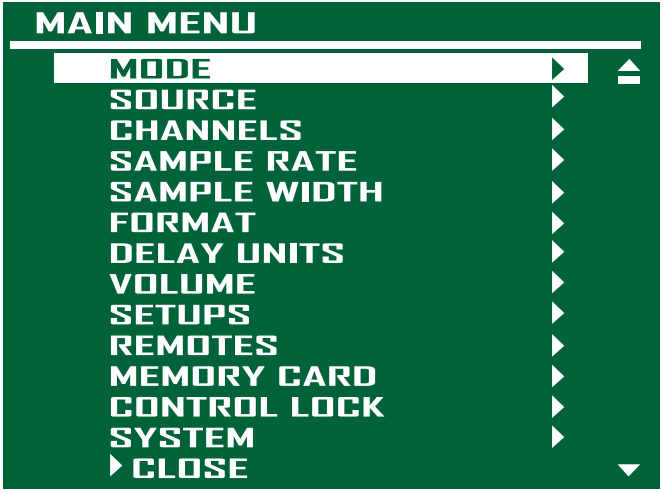



*Fig 8-6: The Main Menu*

**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, EXTENDED CF (FAT) and EXTENDED CF (RAW). 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 8-7, 8-8 and 8-9 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 CF (FAT) – This mode selects the Compact Flash™ memory card loaded in the internal memory card slot. This mode supports Compact Flash™ cards up to 2Gbytes and uses a PC readable filing system. If this Extended mode is selected with no Compact Flash™ memory installed, a message box will be display and the audio output will not be interrupted.
- Extended CF (RAW) – This mode selects the Compact Flash™ memory card loaded in the internal memory card slot. This mode supports Compact Flash™ cards up to 16Gbytes. If this 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. If the digital input signal is removed and a change in sample rate is not detected, the unit will automatically revert to an internally generated master clock to maintain the current delay, and the source of the delay will change to the analogue inputs. When a valid digital input signal is restored, the source for the delay returns to the digital inputs.

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.

**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.

**Remotes**

This selection shows the Remotes sub-menu. The sub-menu items are DELAYED REMOTE 1 to DELAYED REMOTE 6. Each item allows the corresponding remote to act as a delayed input to output using the current delay. From the remotes sub-menu select the required remote number and then select DISABLED or ENABLED. The remote inputs are on pins 1-6 on the rear panel mounted 15-way D-type connector, and the corresponding remote outputs are on pins 10-15. The remote inputs are activated by connecting the required input pin to pin 9, the ground pin. Then, after the current delay period has elapsed, the corresponding open collector output will be asserted (active low).

**Please note:** The remote loading option in the Setup sub-menu must be disabled for the delayed remotes to work. The original remote functions for outputs 1 and 2 are still enabled by default. These are as follows:
**Remote output 1** is a power fail indicator – the remote is always asserted whilst power is applied to the unit.
**Remote output 2** is a bypass indicator – the remote is asserted when the bypass function is inactive.
If the Delayed Remote function is enabled on remotes 1 or 2, the default function on that remote is automatically 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 in Extended FAT mode and 16Gbytes in Extended RAW mode. A memory card is only used in the Extended Modes.

***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 when in Extended FAT mode. When Extended RAW mode is selected, the memory card is deleted and the read and write positions on the card are reset. 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 PC readable files on the memory card in Extended FAT mode.

**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-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<br>nn = 11 (115200 baud)<br>nn = 57 (57600 baud)<br>nn = 38 (38400 baud)<br>nn = 19 (19200 baud)<br>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<br>n.n = bootstrap version | BSV:Vn.n |
| DLY:c,n | Set delay where<br>c = 0 (left/stereo channel)<br>c = 1 (right channel)<br>n = delay | ACK: or NAK: |
| DWN: | **Download firmware<br>new firmware installed when<br>can start)<br>‘S-Record’ terminator received | ACK: or NAK:<br>(ACK: indicates download |
| FMC: | Format memory card | ACK: or NAK: |
| LCS:n | Load configuration setup where<br>n = setup number (1-8) | ACK: or NAK: |
| LOC: | Enable control lock | ACK: or NAK: |
| MCD: | Request memory card details where<br>x = model number<br>y = capacity<br>z = size for delay | MCD:x,y,z or NAK: |
| SAM:n | Set analogue mute where<br>n = 0 (off)<br>n = 1 (on) | ACK: or NAK: |

| Command | Description | Response |
|---|---|---|
| SCH:n | Set channels where<br>n = 0 (mono)<br>n = 1 (dual mono)<br>n = 2 (stereo) | ACK: or NAK: |
| SDM:n | Set digital mute where<br>n = 0 (off)<br>n = 1 (on) | ACK: or NAK: |
| SDU:n | Set delay units where<br>n = 0 (samples)<br>n = 1 (fields)<br>n = 2 (frames)<br>n = 3 (time) | ACK: or NAK: |
| SER: | **Serial number request where<br>nnnnnn = serial number | SER:nnnnnn |
| SFM:n | Set format where<br>n = 0 (25 frames per second)<br>n = 1 (24 frames per second)<br>n = 2 (30 frames per second)<br>n = 3 (30 drop frame) | ACK: or NAK: |
| SIP:n | Set input peak digits where<br>n = 0 (+12dB)<br>n = 1 (+18dB)<br>n = 2 (+24dB) | ACK: or NAK: |
| SMD:n | Set mode where<br>n = 0 (normal mode)<br>n = 1 (extended FAT mode)<br>n = 2 (extended RAW mode) | ACK: or NAK: |
| SOG:n | Set output gain where<br>n = 0 (-6dB)<br>n = 1 (-5dB)<br>n = 2 (-4dB)<br>n = 3 (-3dB)<br>n = 4 (-2dB)<br>n = 5 (-1dB)<br>n = 6 (-0dB)<br>n = 7 (+1dB)<br>n = 8 (+2dB)<br>n = 9 (+3dB)<br>n = 10 (+4dB)<br>n = 11 (+5dB)<br>n = 12 (+6dB)<br>n = 13 (+7dB)<br>n = 14 (+8dB)<br>n = 15 (+9dB)<br>n = 16 (+10dB)<br>n = 17 (+11dB)<br>n = 18 (+12dB)<br>n = 19 (+13dB)<br>n = 20 (+14dB)<br>n = 21 (+15dB)<br>n = 22 (+16dB)<br>n = 23 (+17dB)<br>n = 24 (+18dB)<br>n = 25 (+19dB)<br>n = 26 (+20dB)<br>n = 27 (+21dB)<br>n = 28 (+22dB)<br>n = 29 (+23dB)<br>n = 30 (+24dB) | ACK: or NAK: |
| SRL:n | Set remote setup loading where<br>n = 0 (off)<br>n = 1 (on) | ACK: or NAK: |
| SRQ: | Status request where<br>i = bypass status<br>J = left/stereo delay<br>k = right delay<br>l = control lock status<br>m = analogue mute status<br>n = channel setting<br>o = digital mute status<br>p = delay units setting<br>q = format setting<br>r = input volume setting<br>s = mode setting<br>t = output volume setting<br>u = remote loading setting | ACK: or NAK: |

| Command | Description | Response |
|---|---|---|
| | v = source setting<br>w = system date ddmmyy<br>x = sample rate setting<br>y = system time hhmmss<br>z = sample width setting | |
| SSC:n | Set source where<br>n = 0 (analogue)<br>n = 1 (analogue sync)<br>n = 2 (digital) | ACK: or NAK: |
| SSD:ddmmyy | Set system date where<br>dd = day (1-31)<br>mm = month (1-12)<br>yy = year (00-99) | ACK: or NAK: |
| SSR:n | Set sample rate where<br>n = 0 (32kHz)<br>n = 1 (44.1kHz)<br>n = 2 (48kHz)<br>n = 3 (64kHz)<br>n = 4 (88.2kHz)<br>n = 5 (96kHz) | ACK: or NAK: |
| SST:hhmmss | Set system time where<br>hh = hours (0-23)<br>mm = minutes (0-59)<br>ss = seconds (0-59) | ACK: or NAK: |
| SSW:n | Set sample width where<br>n = 0 (16bit)<br>n = 1 (24bit) | ACK: or NAK: |
| UID: | **Unit ID request | UID:DS2 |
| UNL: | Disable control lock | ACK: or NAK: |
| VER: | Version request where<br>y.yyy = firmware version<br>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.

## Technical Specifications RB-DS2

| Audio Specification | |
|---|---|
| Maximum Input Level: | +27dBu |
| Input Impedance: | > 10kΩ 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: | < 50Ω |

| | |
|---|---|
| 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<br>20 samples at 64kHz, 88.2kHz, 96kHz |
| Maximum Delays: | See Figures 8-7, 8-8 and 8-9. |

| **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 |

| **Equipment Type:** | |
|---|---|
| RB-DS2: RK3 | Stereo Delay Synchroniser & Time-Zone Delay |

| **Physical Specifications** | |
|---|---|
| Dimensions (Raw) | 48cm (W) x 15.8cm (D *) x 4.2cm (H) (1U)<br>19” (W) x 6.2” (D*) x 1.7” (H) (1U) |
| Dimensions (Boxed) | 59cm (W) x 27.4cm (D*) x 10.8cm (H)<br>23.2” (W) x 10.8” (D*) x 4.3” (H) |
| Weight | Nett: 1.6kg Gross: 2.2kg<br>Nett: 3.5lbs Gross: 4.8lbs |

* Note that this product is deeper than standard Redboxes.

## 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.

| 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 8-7: 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.*

| 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 8-8: Maximum Delays in Extended FAT Mode with 1 GB Compact Flash™ Memory Card Installed*

**Note:** *1. A maximum of 2GB can be installed using Extended FAT mode.*
*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.*

| 16GB | | 32 kHz | | 44.1 kHz | | 48 kHz | |
|---|---|---|---|---|---|---|---|
| | | 16 bit | 24 bit | 16 bit | 24 bit | 16 bit | 24 bit |
| Samples | | 4000000000 | 2000000000 | 4000000000 | 2000000000 | 4000000000 | 2000000000 |
| Fields | 25fps | 6250000 | 3125000 | 4535147 | 2267573 | 4166666 | 2083333 |
| | 24fps | 6000000 | 3000000 | 4353741 | 2176870 | 4000000 | 2000000 |
| | 30fps | 7500000 | 3750000 | 5442176 | 2721088 | 5000000 | 2500000 |
| | 30df | 7492506 | 3746253 | 5436739 | 2718369 | 4995004 | 2497502 |
| Frames | 25fps | 3125000 | 1562500 | 2267573 | 1133786 | 2083333 | 1041666 |
| | 24fps | 3000000 | 1500000 | 2176870 | 1088435 | 2000000 | 1000000 |
| | 30fps | 3750000 | 1875000 | 2721088 | 1360544 | 2500000 | 1250000 |
| | 30df | 3746253 | 1873126 | 2718369 | 1359184 | 2497502 | 1248751 |
| Time (hh:mm:ss) | | 34:43:20 | 17:21:40 | 25:11:42 | 12:35:51 | 23:08:53 | 11:34:26 |

| 16GB | | 64 kHz | | 88.2 kHz | | 96 kHz | |
|---|---|---|---|---|---|---|---|
| | | 16 bit | 24 bit | 16 bit | 24 bit | 16 bit | 24 bit |
| Samples | | 4000000000 | 2000000000 | 4000000000 | 2000000000 | 4000000000 | 2000000000 |
| Fields | 25fps | 3125000 | 1562500 | 2267573 | 1133786 | 2083333 | 1041666 |
| | 24fps | 3000000 | 1500000 | 2176870 | 1088435 | 2000000 | 1000000 |
| | 30fps | 3750000 | 1875000 | 2721088 | 1360544 | 2500000 | 1250000 |
| | 30df | 3746253 | 1873126 | 2718369 | 1359184 | 2497502 | 1248751 |
| Frames | 25fps | 1562500 | 781250 | 1133786 | 566893 | 1041666 | 520833 |
| | 24fps | 1500000 | 750000 | 1088435 | 544217 | 1000000 | 500000 |
| | 30fps | 1875000 | 937500 | 1360544 | 680272 | 1250000 | 625000 |
| | 30df | 1873126 | 936563 | 1359184 | 679592 | 1248751 | 624375 |
| Time (hh:mm:ss) | | 17:21:40 | 08:40:50 | 12:35:51 | 06:17:55 | 11:34:26 | 05:47:13 |

*Fig 8-9: Maximum Delays in Extended RAW Mode with 16 GB Compact Flash™ Memory Card Installed.*

***Note:*** *1. A maximum of 16GB can be installed using Extended RAW mode.*
*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.*

# 9 RB-DS2R Remote Switch Panel For RB-DS2

## Introduction

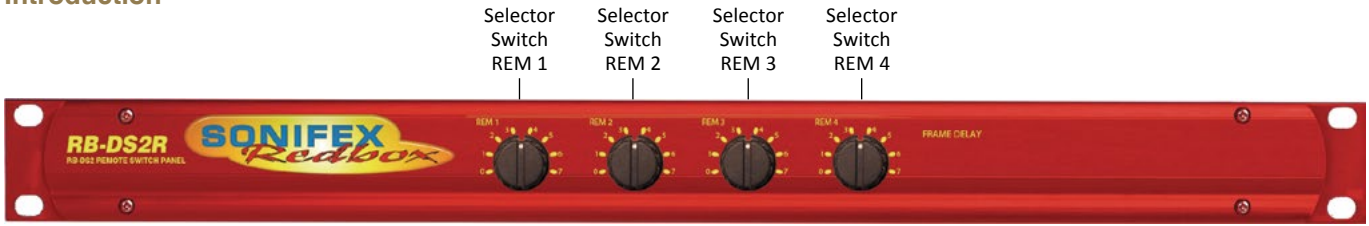



*Fig 9-1: RB-DS2R Front Panel*

The RB-DS2R is a remote panel for controlling up to 4 separate RB-DS2 units from a single 1U 19" rack-mount panel.

The unit has 4 rotary switches, each with 8 available selections numbered 0 to 7, used for selecting the delay time in frames. On the rear of the unit are 4 x 15-way D-type connectors used for connecting directly to the remote input of 4 x RB-DS2 units. The RB-DS2R is a passive unit, i.e. there is no power supply.

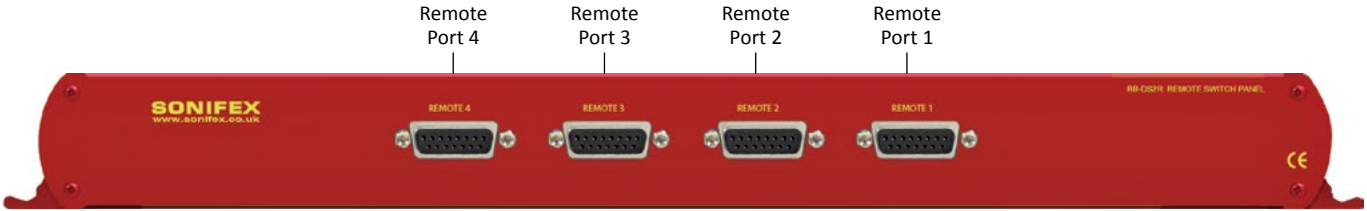



*Fig 9-2: RB-DS2R Rear Panel*

## Technical Specification For RB-DS2R

| Rear Panel Connections | |
|---|---|
| Remote I/O Port: | 4 x 15-way ‘D’-type socket, 8 GPI outputs |

| Front Panel Controls | |
|---|---|
| Selector Switches: | 4 x 8 way rotary selectors |

| Equipment Type | |
|---|---|
| RB-DS2R: | Remote switch panel for 4 x RB-DS2 RK3 |

| Physical Specification | |
|---|---|
| Dimensions (Raw): | 48cm (W) x 10.8cm (D) x 4.2cm (H) (1U)<br>19” (W) x 4.3” (D) x 1.7” (H) (1U) |
| Dimensions (Boxed): | 58.5cm (W) x 22.5cm (D) x 7cm (H) (1U)<br>23” (W) x 8.9” (D) x 2.8” (H) (1U) |
| Weight | Nett: 1.2kg Gross: 1.6kg<br>Nett: 2.6lbs Gross: 3.5lbs |

# 10 RB-PD2 Stereo Profanity Delay

## Introduction

*Fig 10-1: 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.

*(*at 32kHz 16bit)*

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.

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 ±3dB 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.

**Important Note**

Certain newer Compact Flash (CF) cards can cause the RB-PD2 to reboot, reset or 'hang-up'. This can be resolved by repowering the unit with the CF card already inserted.

## System Block Diagram

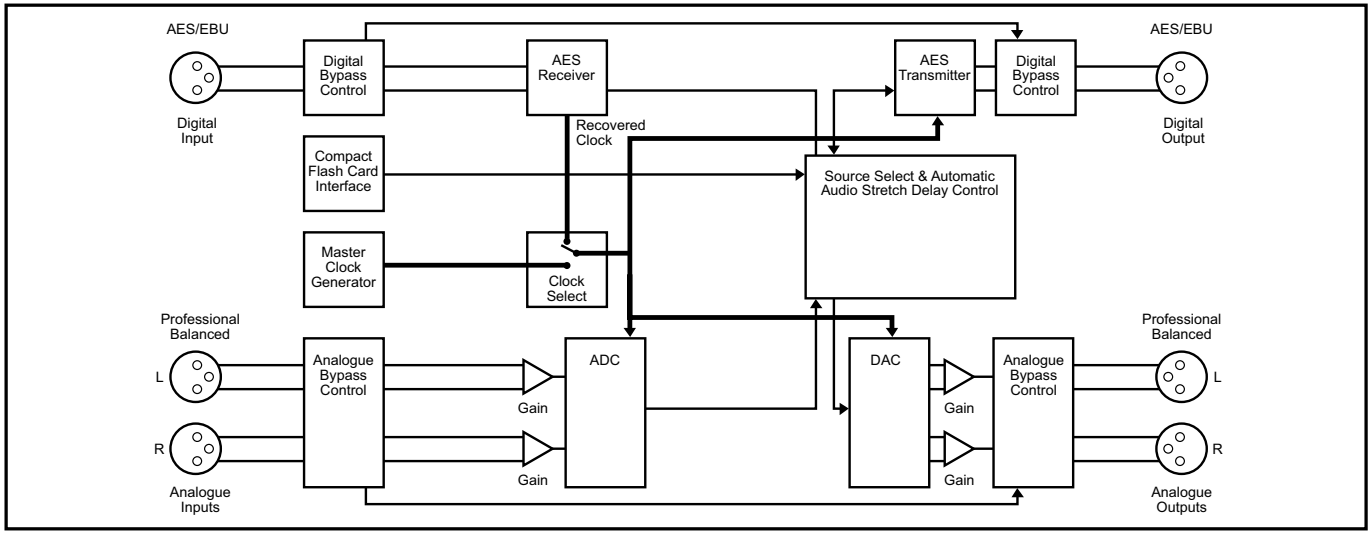



*Fig 10-2: RB-PD2 System Block Diagram*

## 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 71: Sample Rate & Sample Width). Also, you can make a recording directly on the RB-PD2 (see page 71: 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.

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 10-3: 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 69).

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".

**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 69).

*Fig 10-4: 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 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 70) 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 10-5: 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 10kΩ 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 ±3dB.

**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.

**XLR Analogue Outputs (Left and Right)**
The XLR 3 pin output plugs 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

**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 which can be programmed to your requirements. The inputs are used to remotely activate a pre-assigned function andthe 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
Pin10: Open collector output 1
Pin11: Open collector output 2
Pin12: Open collector output 3
Pin13: Open collector output 4
Pin14: Open collector output 5
Pin15: 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

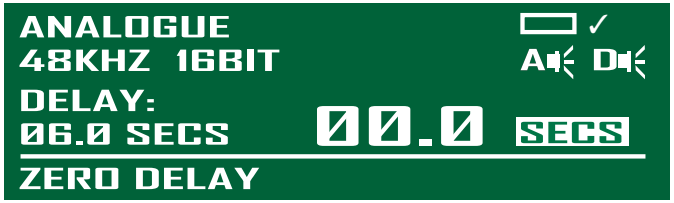



*Fig 10-6: The Default Screen*

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 64.

### The Main Menu

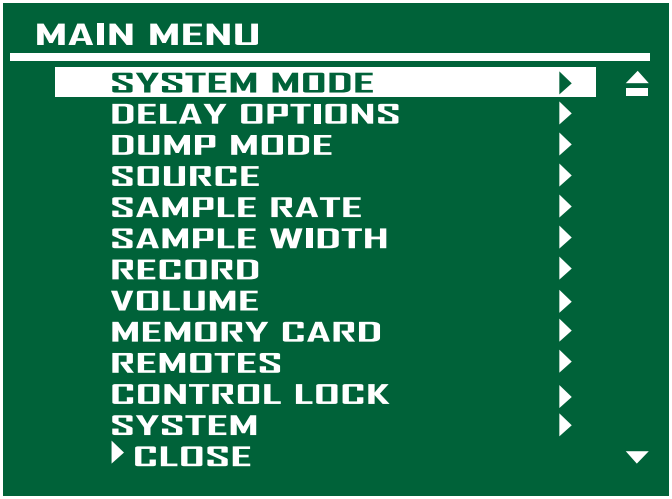



*Fig 10-7: The Main Menu Screen*

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 66.

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

**SYSTEM MODE**
This selection shows the SYSTEM MODE sub-menu. The sub-menu items are NORMAL MODE and REVIEW MODE.

**Normal Mode**
This option selects the normal profanity delay operating mode of the unit. This is the default mode.

**Review Mode**
This option selects the review mode. This mode allows constant live monitoring of an audio feed, with the ability to repeat sections of audio by increasing the delay by a preconfigured step. See the Review Mode section for details.

**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***

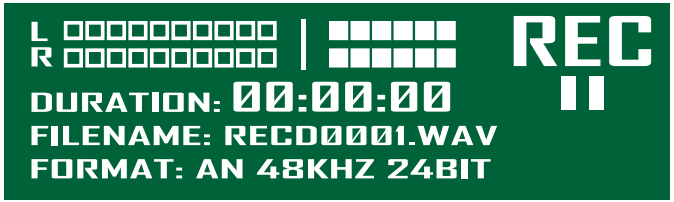



*Fig 10-8: The Record Screen*

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 is 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.

## Review Mode

Review mode changes the functionality of the unit. It allows live monitoring of an audio input with the ability to repeat sections of audio by increasing the delay immediately by a preconfigured step. The delay can be increased multiple times until either the maximum possible delay is reached or no further delays increases by the preconfigured step are possible. Review mode is enabled via: [MAIN MENU > SYSTEM MODE > REVIEW MODE].

The default screen shows the selected step size in seconds on the left hand side of the display and the current delay in a larger font on the right hand side. The current status is shown at the bottom of the display, and shows “MONITORING LIVE” when the output is not delayed, and “REVIEW ACTIVE” when the delay is active.

The step size is changed directly from the main screen. Briefly press the rotary control to enable the step edit mode. The decimal point on the

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

Pressing the DUMP button increases the delay by the current step size. When the delay is active, the EXIT DELAY button illumination is turned off, and the DUMP button illumination is turned on. When no more delay increases are available at the current step size, the DUMP button will flash. The maximum delays for the indicated sample rates are shown below:

32 kHz sample rate - 55.2 seconds for 16 bit, 27.6 seconds for 24 bit
44.1 kHz sample rate - 40.0 seconds for 16 bit, 20.0 seconds for 24 bit
48 kHz sample rate - 36.8 seconds for 16 bit, 18.4 seconds for 24 bit

To return to live monitoring, press the exit delay button. The DUMP button illumination is turned off and the EXIT button illumination is turned on.

The current delay can be temporarily bypassed by pressing the BYPASS button. This allows the live audio to be temporarily monitored while the delay is still active. Both the DUMP and EXIT DELAY buttons are still active while the bypass is active.

The BUILD DELAY and COUGH buttons are disabled when the review mode is selected and the menu options DELAY OPTIONS, DUMP MODE, SOURCE, RECORD, MEMORY CARD and REMOTES are omitted as they do not apply. The source selection is pre-set to Analogue in review mode.

## 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:<br>nn = 11 (115200 baud)<br>nn = 57 (57600 baud)<br>nn = 38 (38400 baud)<br>nn = 19 (19200 baud)<br>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: |
| 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:<br>c = "\path\filename"<br>(c="\filename" for files in root directory) | ACK: or NAK: |
| DLY:n | Set required delay where:<br>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:<br>n = 0 (5mins 30secs)<br>n = 1 (5mins 00secs)<br>n = 2 (4mins 30secs)<br>n = 3 (4mins 00secs)<br>n = 4 (3mins 30secs)<br>n = 5 (3mins 00secs)<br>n = 6 (2mins 30secs)<br>n = 7 (2mins 20secs)<br>n = 8 (2mins 10secs)<br>n = 9 (2mins 00secs)<br>n = 10 (1min 50secs)<br>n = 11 (1min 40secs)<br>n = 12 (1min 30secs)<br>n = 13 (1min 20secs)<br>n = 14 (1min 10secs)<br>n = 15 (1min 00secs) | ACK: or NAK: |
| DST: | Dump start | ACK: or NAK: |
| DWN: | **Download firmware.<br>New firmware installed when 'S-Record' terminator received | ACK: or NAK:<br>(ACK: indicates download can start) |
| FMC: | Format memory card | ACK: or NAK: |
| GDL:c | Get directory list where:<br>c = "\path" for base directory of list (c="\" for root directory)<br>z = comma separated list of directories | GDL:z |
| GFL:n_c | Get file list where:<br>n = 1 (list wav files)<br>n = 2 (list wav files with same settings as current configuration)<br>n = 3 (list all files)<br>c = "\path" for base directory of list (c="\" for root directory)<br>z = comma separated list of files | GFL:z |
| LOC: | Enable control lock | ACK: or NAK: |
| MCD: | Request memory card details where:<br>y = model number<br>z = capacity in MBytes | MCD:y,z or NAK: |
| NRC: | Initialise new recording | ACK: or NAK: |
| REC:n | Record control where:<br>n = 1 (start)<br>n = 0 (stop) | ACK: or NAK: |
| RIN:c | Set record initials where:<br>c = 4 character initials | ACK: or NAK: |
| RSC:n | Set record source where:<br>n = 0 (analogue)<br>n = 1 (analogue sync)<br>n = 2 (digital) | ACK: or NAK: |
| RSR:n | Set record sample rate where:<br>n = 0 (32 kHz)<br>n = 1 (44.1 kHz)<br>n = 2 (48 kHz) | ACK: or NAK: |
| RSW:n | Set record sample width where:<br>n = 0 (16 bit)<br>n = 1 (24 bit) | ACK: or NAK: |
| SAM:n | Set analogue mute where:<br>n = 0 (off )<br>n = 1 (on) | ACK: or NAK: |
| SDB:n | Set dump buffer length where: | ACK: or NAK: |

| | | |
|---|---|---|
| | n = dump length in milliseconds<br>(100ms resolution) | |
| SDF:c | Set dump play file where:<br>c = “\path\filename” (c=“\filename” for files in root directory)<br>z = error message | ACK: or NAK:z |
| SER: | **Serial number request where:<br>z = serial number (6 digits) | SER:z |
| SDM:n | Set digital mute where:<br>n = 0 (off )<br>n = 1 (on) | ACK: or NAK: |
| SIP:n | Set input peak digits where:<br>n = 0 (+12dB)<br>n = 1 (+18dB)<br>n = 2 (+24dB) | ACK: or NAK: |
| SMD:n | Set dump mode where:<br>n = 0 (dump buffer)<br>n = 1 (play file) | ACK: or NAK: |
| SMS:n | Set start delay mode where:<br>n = 0 (build using audio stretch algorithm)<br>n = 1 (play file)<br>z = error message | ACK: or NAK:z |
| SOG:n | Set output gain where:<br>n = 0 (-6dB)<br>n = 1 (-5dB)<br>n = 2 (-4dB)<br>n = 3 (-3dB)<br>n = 4 (-2dB)<br>n = 5 (-1dB)<br>n = 6 (-0dB)<br>n = 7 (+1dB)<br>n = 8 (+2dB)<br>n = 9 (+3dB)<br>n = 10 (+4dB)<br>n = 11 (+5dB)<br>n = 12 (+6dB)<br>n = 13 (+7dB)<br>n = 14 (+8dB)<br>n = 15 (+9dB)<br>n = 16 (+10dB)<br>n = 17 (+11dB)<br>n = 18 (+12dB)<br>n = 19 (+13dB)<br>n = 20 (+14dB)<br>n = 21 (+15dB)<br>n = 22 (+16dB)<br>n = 23 (+17dB)<br>n = 24 (+18dB)<br>n = 25 (+19dB)<br>n = 26 (+20dB)<br>n = 27 (+21dB)<br>n = 28 (+22dB)<br>n = 29 (+23dB)<br>n = 30 (+24dB) | ACK: or NAK: |
| SRI:i_n | Set function for remote input where:<br>i = remote input id (0-7)<br>n = 0 (no action)<br>n = 1 (build delay)<br>n = 2 (exit delay)<br>n = 3 (activate dump)<br>n = 4 (activate drop)<br>n = 5 (activate cough)<br>n = 6 (start new recording)<br>n = 7 (mute all audio outputs)<br>n = 8 (activate audio bypass) | ACK: or NAK: |
| SRO:o_n | Set function signal for remote output where:<br>o = remote output id (0-5)<br>n = 0 (no action)<br>n = 1 (delay is at zero)<br>n = 2 (delay building)<br>n = 3 (delay is greater than safe period)<br>n = 4 (delay is at selected value)<br>n = 5 (delay exiting)<br>n = 6 (dump active)<br>n = 7 (drop active)<br>n = 8 (cough active) | ACK: or NAK: |

| | | |
|---|---|---|
| | n = 9 (record active)<br>n = 10 (audio outputs muted)<br>n = 11 (audio bypass active) | |
| SRQ:n | Status request where:<br>e = unit status (8 hex digits)<br>f = current delay (8 hex digits)<br>g = selected delay (8 hex digits)<br>h = start mode setting<br>i = start play filename<br>j = safe period setting<br>k = dump mode setting<br>l = dump buffer length setting<br>m = dump play filename<br>n = source setting<br>o = sample rate setting<br>p = sample width setting<br>q = record source setting<br>r = record sample rate setting<br>s = record sample width setting<br>t = input peak digits setting<br>u = output gain setting<br>v = analogue mute status<br>w = digital mute status<br>x = bypass status<br>y = control lock status<br>z = delay build/exit speed setting | STA:e_f_g_h_i_j_k_l_m_n_o_p_q_r_s_t_u_v_w_x_y_z |
| SSC:n | Set source where:<br>n = 0 (analogue)<br>n = 1 (analogue sync)<br>n = 2 (digital) | ACK: or NAK: |
| SSF:c | Set delay start play file where:<br>c = "\path\filename"<br>(c="\filename" for files in root directory)<br>z = error message | ACK: or NAK:z |
| SSP:n | Set safe period length where:<br>n = dump length in milliseconds (100ms resolution) | ACK: or NAK: |
| SSR:n | Set sample rate where:<br>n = 0 (32 kHz)<br>n = 1 (44.1 kHz)<br>n = 2 (48 kHz) | ACK: or NAK: |
| SSW:n | Set sample width where:<br>n = 0 (16bit)<br>n = 1 (24bit) | ACK: or NAK: |
| UID: | **Unit ID request | UID:PD2 |
| UNL: | Disable control lock | ACK: or NAK: |
| VER: | Version request where<br>y.yyy = firmware version<br>zzzz = display version | VER:Vy.yyy,Vzzzz |
| VRS: | Request remote settings where:<br>m = remote input 1 setting<br>n = remote input 2 setting<br>o = remote input 3 setting<br>p = remote input 4 setting<br>q = remote input 5 setting<br>r = remote input 6 setting<br>s = remote input 7 setting<br>t = remote input 8 setting<br>u = remote output 1 setting<br>v = remote output 2 setting<br>w = remote output 3 setting<br>x = remote output 4 setting<br>y = remote output 5 setting<br>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) |

| **Equipment Type:** | | |
|---|---|---|
| RB-PD2: | RK3 | Stereo Profanity Delay |

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

* Note that this product is deeper than standard Redboxes

# 11 RB-DD4 4 Channel Digital Audio Delay

*Fig 11-1: RB-DD4 Front Panel*

24 BIT 192 kS/s

The RB-DD4 4 channel digital audio delay allows you to delay 4 mono channels of audio independently or together. Each channel delay is user selectable from multiples of common video frame rates, or a user defined value set via the serial interface. The unit is perfect for synchronizing audio to video which has been delayed by processing latency.

Using a front panel button, you can select which channel needs to be delayed. There is also an 'ALL' option which allows the selected delay to be applied to all channels. Then using another front panel button you can select the length of one frame of delay and the multiple of frames to delay by.

The connectivity is incredibly flexible, allowing three different types of connection to each input and output including AES/EBU, S/PDIF and TOSLink. All three different types of output can be used simultaneously. There is a monitor socket on the front panel which allows you to listen to each mono channel, by front panel selection. Pairs of channels can be monitored (1 & 2 or 3 & 4) using a rear panel stereo option. There is also an option to attenuate the monitor by 12dB selectable by rear panel DIPswitch. Audio presence is detected and displayed for each channel around the INPUTS 1 & 2 and INPUTS 3 & 4 buttons.

The flexibility continues with many audio synchronization options. The digital audio output can be synchronized to either input, an additional AES/EBU reference input, a TTL wordclock BNC input or an analogue/SDI video feed if used with an additional RB-SYA or RB-SYD board. Also the output can be synchronized to an on-board master clock, with a selectable frame rate. There are warning indicators on the front panel for loss of lock on both inputs and for the selected external synchronization. Selectable synchronization 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.

A powerful feature of the RB-DD4 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 program the unit for your specific application. A rear panel DIPswitch configures the unit to be controlled serially. Contact Sonifex for further information if you have a particular requirement that isn't catered for by the RB-DD4 as standard.

The RB-DD4 has been designed to have a passive signal path through the main input, so if power to the unit fails, signal inputs 1 & 2 are routed to outputs 1 & 2 and signal inputs 3 & 4 are routed to outputs 3 & 4. 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. Please note that this is not true for the TOSLink outputs which are muted.

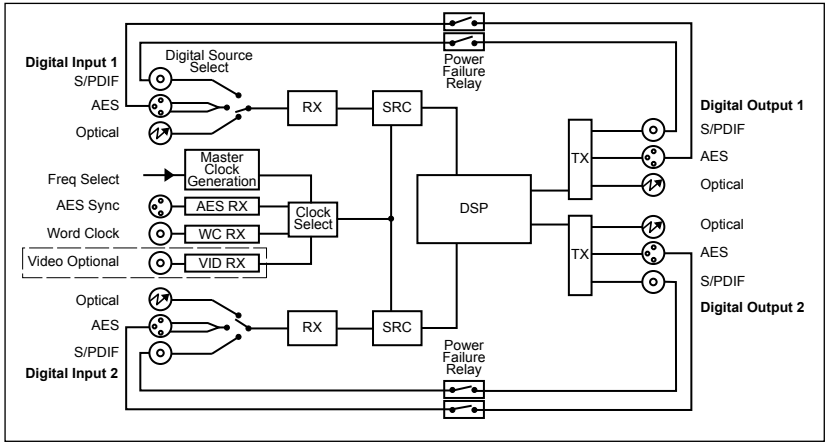



*Fig 11-2: RB-DD4 Block Diagram*

## Front Panel Controls and Indicators

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

*Fig 11-3: Front Panel Controls and Indicators*

*Fig 11-4: DELAY Button*

**DELAY Unit & Quantity Select and Indicators**

This button allows you to select the length of a delay unit and the multiple, which when multiplied together give the total delay length.

To set the Delay Unit, press and hold the DELAY button. A single given LED is lit to indicate which unit has been selected. Pressing the button moves the selection in a clockwise direction.

To set the multiple/quantity of units to delay, press and hold the DELAY button again. The LEDS around the button indicate the chosen delay multiple and the total multiple is found by summing the indicated numbers. Pressing the button increments the multiple by one. (See page 83 for more info on Delay Button Modes).

*Fig 11-5: CHANNEL SELECT Button*

**Channel Select**

This button selects and indicates the currently selected input channel. There is an option for each individual channel and an option for all channels. As each different channel is selected, the currently selected delay multiple for that channel is displayed on the DELAY button LEDs. Any change of the multiple or delay unit length is applied to the currently selected channel, or all of them if the ALL option is selected. Pressing the button moves the selection in a clockwise direction.

*Fig 11-6: INPUTS 1 & 2 Button*

**INPUT 1 & 2 Source Select, Indicators & Input Presence LEDs**

This button allows you to select which input source you would like to use for the selected channel. The three LEDs above the button illustrate which source is selected. The button itself is illuminated when the input is locked.

The bicolour LEDs, marked '1' and '2' , show input presence and give an indication of the input level using the AES digital standard with the following colours:

-INF < -52dBFS = OFF

-52dBFS < -3dBFS = GREEN

-3dBFS <0dBFS = ORANGE

**INPUT 3 & 4 Source Select, Indicators & Input Presence LEDs**

The operation of this button is identical to above.

*Fig 11-7: MONITOR SELECT Button*

**MONITOR SELECT button**

Press the MONITOR SELECT button to choose an output to monitor in the headphones. For monitor selection the button illuminates read and the LEDS around the button display in red which channel has been selected.

The button also displays external signal status. If the selected synchronisation source is unlocked, the button flashes green and red.

*Fig 11-8: Headphone Output & Level Controls*

**Headphone Output**

The front panel headphone output is a ¼" (6.35mm) stereo jack socket capable of delivering over 80mW into 32Ω - 600Ω professional headphones at full volume. Higher impedance headphones may be used at reduced levels. Lower impedance headphones should not be used.

If the output sounds bad or disturbed at any time, use the monitor attenuation DIPSwitch 10 to reduce the headphone output to an acceptable level - this will depend on the impedance of the headphones that you are using.

**LEVEL Control**

The front panel LEVEL control is a potentiometer that adjusts the level of the monitor output and provides a gain range of -70dB to +12dB.

*Fig 11-9: Reset Button*

**Reset Button**

In the unlikely event that the RB-DD4 unit fails to respond, press the reset button to reboot the unit (see Fig 1-9 for location).

## Delay Button Modes

The Delay button operates in two different modes:

**Delay Unit Select Mode**

To select the delay unit press and hold the delay button for two seconds and then release. The button turns green. Momentary pressing of the button moves the Delay Unit selection in a clockwise direction. There are eight different options for the delay unit with each number corresponding to a frame per second value with the delay unit length being the reciprocal of this value. The available choices (in frames per second) are:

23.98, 24, 25, 29.97, 30, 50, 59.94 & 60

Once the delay unit length has been chosen, press and hold the button for two seconds to return to Delay Multiple mode.

**Delay Multiple Mode**

This is the default mode for this button and is used to select the quantity, or multiple, of units to delay by. Press this button to increment the multiple value from 1 unit to 19. The quantity is displayed as the sum of the LEDS around the button. The following explains how the LEDs display the number:

| | Led Number | | | | | | |
|---|---|---|---|---|---|---|---|
| **Multiple** | **1** | **2** | **3** | **4** | **5** | **10** | **U D** |
| **Minimum** | **Off** | **Off** | **Off** | **Off** | **Off** | **Off** | **Off** |
| 1 | On | Off | Off | Off | Off | Off | Off |
| 2 | Off | On | Off | Off | Off | Off | Off |
| 3 | Off | Off | On | Off | Off | Off | Off |
| 4 | Off | Off | Off | On | Off | Off | Off |
| 5 | Off | Off | Off | Off | On | Off | Off |
| 6 | On | Off | Off | Off | On | Off | Off |
| 7 | Off | On | Off | Off | On | Off | Off |
| 8 | Off | Off | On | Off | On | Off | Off |
| 9 | Off | Off | Off | On | On | Off | Off |
| 10 | Off | Off | Off | Off | Off | On | Off |
| 11 | On | Off | Off | Off | Off | On | Off |
| 12 | Off | On | Off | Off | Off | On | Off |
| 13 | Off | Off | On | Off | Off | On | Off |
| 14 | Off | Off | Off | On | Off | On | Off |
| 15 | Off | Off | Off | Off | On | On | Off |
| 16 | On | Off | Off | Off | On | On | Off |

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| 17 | Off | On | Off | Off | On | On | Off |
| 18 | Off | Off | On | Off | On | On | Off |
| 19 | Off | Off | Off | On | On | On | Off |
| User Defined | Off | Off | Off | Off | Off | Off | On |

When all the LEDs are OFF, the channel uses the minimum delay which is dependant on the input sample and output sample rates.

| | | Input Samplerate (Hz) | | | | | | |
|---|---|---|---|---|---|---|---|---|
| | | **32k** | **44.1k** | **48k** | **88.2k** | **96k** | **176.4k** | **192k** |
| **Output Samplerate (Hz)** | **32k** | 4.594 msec | 3.680 msec | 3.484 msec | 2.472 msec | 2.374 msec | 1.869 msec | 1.819 msec |
| | **44.1k** | 4.247 msec | 3.333 msec | 3.137 msec | 2.125 msec | 2.027 msec | 1.522 msec | 1.473 msec |
| | **48k** | 4.172 msec | 3.259 msec | 3.063 msec | 2.051 msec | 1.953 msec | 1.447 msec | 1.398 msec |
| | **88.2k** | 3.788 msec | 2.875 msec | 2.678 msec | 1.667 msec | 1.569 msec | 1.063 msec | 1.014 msec |
| | **96k** | 3.751 msec | 2.837 msec | 2.641 msec | 1.629 msec | 1.531 msec | 1.025 msec | 0.976 msec |
| | **176.4k** | 3.559 msec | 2.645 msec | 2.449 msec | 1.437 msec | 1.339 msec | 0.833 msec | 0.784 msec |
| | **192k** | 3.540 msec | 2.626 msec | 2.430 msec | 1.419 msec | 1.320 msec | 0.815 msec | 0.766 msec |

*Fig 11-10: Minimum delay values (at 0 frames) due to inherent delay at different input to output sample rates within the unit, measured in ms*

## Rear Panel Controls & Connectors

### Rear Panel DIPSwitches

The SETTINGS DIPSwitch block on the rear panel is used to configure the RM-DD4:

*Fig 11-11: RB-DD4 Rear Panel DIPSwitch Block*

**Master Mode Sample Rate Selection (DIPSwitches 1-3)**

These DIPSwitches allow you select which sample rate the output is set to when the unit is synchronised to the Master Mode:

| 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 |

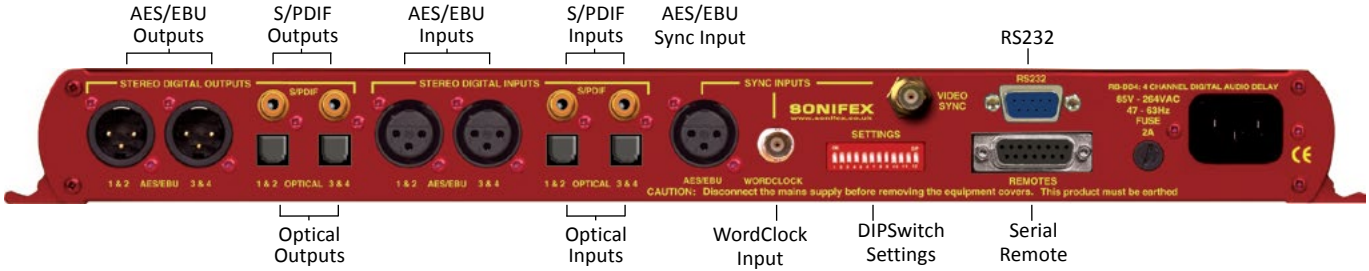



*Fig 11-12: RB-DD4 Rear Panel*

**Synchronisation Source Selection (DIPSwitches 4-6)**
These DIPSwitches allow you select which input sync source is used to synchronise the unit:

| Synchronisation Source | DIPSwitch 4 | DIPSwitch 5 | DIPSwitch 6 |
|---|---|---|---|
| Input 1 & 2 | OFF | OFF | OFF |
| Input 3 & 4 | ON | OFF | OFF |
| AES/EBU Sync Input | OFF | ON | OFF |
| Word Clock Input | ON | ON | OFF |
| Video Sub Board | OFF | OFF | ON |

**Synchronisation Mode Selection (DIPSwitches 7-8)**
These DIPSwitches allow you select the active sync mode:

| Synchronisation Mode | DIPSwitch 7 | DIPSwitch 8 |
|---|---|---|
| Master Mode | OFF | OFF |
| Auto Sync Mode | ON | OFF |
| Auto Lock Mode | OFF | ON |
| Slave Mode | ON | ON |

**Stereo/Mono Operation of the Headphone Monitor (DIPSwitch 9)**This defines whether the monitor operates as a stereo pair or as mono channels 1 & 2.

| Mode | DIPSwitch 9 | Description |
|---|---|---|
| Stereo | ON | Also when ON, the headphone monitor outputs a stereo signal made up of either Input 1 & 2 or Input 3 & 4. |
| Mono | OFF | In Mono Mode the selected signal is sent to left and right earpieces of the headphone output. |

**Monitor Attenuation (DIPSwitch 10)**
This defines whether the monitor signal is attenuated by 12dB. This is useful if you're using low impedance headphones which are too loud in everyday use.

| Mode | DIPSwitch 10 | Description |
|---|---|---|
| Attenuated | ON | When ON, the monitor signal is attenuated. |
| Unattenuated | OFF | When OFF, the monitor signal is unaffected. |

**Serial Mode (DIPSwitch 11)**
This defines whether the unit is in serial mode. In serial mode the unit is controlled by the serial port, not by its DIPSwitch settings. For example for use with the Sonifex SCi software.

| Mode | DIPSwitch 11 | Description |
|---|---|---|
| Serial Control | ON | When ON, the unit is in serial mode. |
| DIPSwitch Control | OFF | When OFF, the unit is in normal operation. |

**Boot Mode (DIPSwitch 12)**
With this DIPSwitch ON, the unit powers up into 'Boot Mode'. In this mode, the firmware in the unit can be upgraded using the SCi software. Note that this would be useful if a firmware update to the unit was interrupted, or corrupted which left the unit in an inoperable condition.

| Mode | DIPSwitch 12 | Description |
|---|---|---|
| Boot Mode | ON | When ON, the unit is in Boot Mode. |
| Normal Operation | OFF | When OFF, the unit is operates normally. |

## RB-DD4 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 wordclock TTL BNC input has an impedance of 75Ω.

**Video Sync Input**

The optional video sync input is presented as a 75Ω BNC connector. See page viii for more information on the video sync boards available.

**Serial RS232 Connector**

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.

**Remotes Connector**

The remotes connector is a 15-way 'D' type socket that is currently reserved for future development. Displayed below are the pin connections and a description of what is available:
Pin 1 – Relay 1 Normally Open
Pin 2 – Relay 1 Normally Closed
Pin 3 – Relay 2 Normally Open
Pin 4 – Relay 2 Normally Closed
Pin 5 – Make to Digital Ground Input
Pin 6 – Internal Open Collector to Digital Ground Output
Pin 7 – Make to Digital Ground Input
Pin 8 – Digital Ground
Pin 9 – Relay 1 Common
Pin 10 – Make to Digital Ground Input
Pin 11 – Relay 2 Common
Pin 12 – 5 V Supply Maximum 200 mA
Pin 13 – Internal Open Collector to Digital Ground Output
Pin 14 – Make to Digital Ground Input
Pin 15 – Internal Open Collector to Digital Ground Output

## RB-DD4 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 have an unweighted dynamic range of at least 138dB and a THD+N noise of or better than –137dB. The data at the outputs is presented as 24 bit wide.

## Serial Port Control

The Serial Port allows the RB-DD4 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 11-13: 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-DD4. Commands are not case sensitive.

Responses are CR & LF terminated.

After the RB-DD4 has been powered-up, an initialisation string is sent

“Initialising DD4”.

Following are the commands and the expected responses:

| Command | Description | Response |
|---|---|---|
| Bnn: | Baudrate Change<br>nn is the new baudrate value where:<br><br>nn = 11 = 115200kbps<br>nn = 57 = 57600kbps<br>nn = 38 = 38400kbps<br>nn = 19 = 19200kbps<br>nn = 96 = 9600kbps | -ACK: |
| CHN:nn | Channel Select<br>Where nn represents the channel which is selected where:<br>00 = Input 1<br>01 = Input 2<br>02 = Input 3<br>03 = Input 4<br>04 = All Inputs | -ACK: |
| DLY: | Delay Setup a_b_cc_dddddd<br>a: Channel to apply delay to where<br>0 = Input 1<br>1 = Input 2<br>2 = Input 3<br>3 = Input4<br>4 = All Inputs<br>b: Chosen delay Unit where<br>0 = Samples<br>1 = Milliseconds<br>2 = Fields<br>3 = Frames<br>4 = Lines<br>cc: Choose video standard where<br>00 = 625/29.97i<br>01 = 525/25i<br>02 = 720/60p<br>03 = 720/59.94p<br>04 = 720/50p<br>05 = 720/30p<br>06 = 720/29.97p<br>07 = 720/25p<br>08 = 720/24p | -ACK: |

| Command | Description | Response |
|---|---|---|
| | 09 = 720/23.98p<br>0A = 1035/60i<br>0B = 1035/59.94i<br>0C = 1080/60i<br>0D = 1080/59.94i<br>0E = 1080/50i<br>0F = 1080/30p<br>10 = 1080/29.97p<br>11 = 1080/25p<br>12 = 1080/24p<br>13 = 1080/23.98p<br>14 = 1080/30pSF<br>15 = 1080/29.97pSF<br>16 = 1080/25pSF<br>17 = 1080/24pSF<br>18 = 1080/23.98pSF<br>19 = 1080/60p<br>1A = 1080/59.94p<br>1B = 1080/50p<br>dddddd: Number of samples in hex | |
| **DWN:** | **Download firmware**<br>**Initiates a firmware upgrade** | **-ACK:** |
| FPS:<br>-FPS:aa_bb_cc_dd_ee_ff_gg_hh_ii_jj_kkkkkk_llllll_mmmmmm_nnnnnn_oooooo | Front panel and unit status<br>aa: Input 1 & 2 source selection<br>bb: Input 3 & 4 source selection<br>where 00 = AES, 01 = SPDIF,02 = TOSLINK<br>cc: Sync mode selection<br>where 00 = Master, 01 = Auto, 02 = Auto lock, 03 = Slave mode<br>dd: Serial Flag indication<br>where 00: Serial Mode Off, 01 = Serial mode On<br>ee: Frequency<br>where 00 = 32k,01 = 44.1k,02 = 48k,03 = 88.2k,04 = 96k,05 =176.4k,06 = 192k<br>ff: Monitor Channel<br>where 00 = OUTPUT1,01 = OUTPUT2, 02 = OUTPUT3, | |

| Command | Description | Response |
|---|---|---|
| | 03 = OUTPUT4<br>gg: Sync From<br>where 00 = Input 1, 01 = Input 2, 02 = AES Sync, 03 = Word Clock, 04 = Video Sync<br>hh: Rear dip switch settings<br>where hh is a hex value built from the sum of all applicable from:<br>01: Stereo Monitor<br>02: Monitor Attenuation<br>ii: Current video standard and delay unit<br>where ii is a hex value built from the sum of:<br>If in standard mode, Interval<br>60 = 23.98<br>61 = 24.0<br>62 = 25.0<br>63 = 29.97<br>64 = 30.0<br>65 = 50.0<br>66 = 59.94<br>67 = 60.0<br>If in Serial Mode, Video standard<br>00 = 625/29.97i<br>01 = 525/25i<br>02 = 720/60p<br>03 = 720/59.94p<br>04 = 720/50p<br>05 = 720/30p<br>06 = 720/29.97p<br>07 = 720/25p<br>08 = 720/24p<br>09 = 720/23.98p<br>0A = 1035/60i<br>0B = 1035/59.94i<br>0C = 1080/60i<br>0D = 1080/59.94i<br>0E = 1080/50i<br>0F = 1080/30p<br>10 = 1080/29.97p<br>11 = 1080/25p | |

| Command | Description | Response |
|---|---|---|
| | 12 = 1080/24p<br>13 = 1080/23.98p<br>14 = 1080/30pSF<br>15 = 1080/29.97pSF<br>16 = 1080/25pSF<br>17 = 1080/24pSF<br>18 = 1080/23.98pSF<br>19 = 1080/60p<br>1A = 1080/59.94p<br>1B = 1080/50p<br>Delay unit<br>00 = Samples<br>20 = Milliseconds<br>40 = Fields<br>60 = Frames<br>80 = Lines<br>jj: Current channel<br>where 00 = Input1, 01 = Input 2, 02 = Input 3, 03 = Input 4 and 04 = All inputs<br>kkkkkk: Current delay on Input1<br>where kkkkkk is the number of samples in hex. If most significant bit is set, User Defined mode is selected.<br>llllll: Current delay on Input2<br>where llllll is the number of samples in hex. If most significant bit is set, User Defined mode is selected.<br>mmmmmm: Current delay on Input3<br>where mmmmmm is the number of samples in hex.<br>If most significant bit is set, User Defined mode is selected<br>nnnnnn: Current delay on Input4<br>where nnnnnn is the number of samples in hex. If most significant bit is set, User Defined mode is selected.<br>oooooo: Current all input delay<br>where oooooo is the number of samples in hex.<br>If most significant bit is set, User Defined mode is selected. | |

| Command | Description | Response |
|---|---|---|
| FRQ:nn | Output samplerate selection<br>nn selects which samplerate is selected for the output where:<br>00 = 32k<br>01 = 44.1k<br>02 = 48k<br>03 = 88.2k<br>04 = 96k<br>05 = 176.4k<br>06 = 192k | -ACK: |
| MAT:nn | Monitor attenuation<br>nn selects between the two modes<br>00 = No attenuation<br>01 = 12 dB of attenuation | -ACK: |
| MOD:nn | Sync mode selection<br>nn selects the synchronization mode where:<br>00 = Master mode<br>01 = Auto mode<br>02 = Auto Lock mode<br>03 = Slave | -ACK: |
| MON:nn | Select Monitor Channel<br>nn selects which channel is monitored where<br>00 = Output 1<br>01 = Output 2<br>02 = Output 3<br>03 = Output 4 | -ACK: |
| MOS:nn | Mono or stereo selection<br>nn selects between the two options where:<br>00 = Mono mode<br>01 = Stereo mode | -ACK: |
| SRQ: Status Request | | -SRQ:aa_bb_cc_dd_eeee_ffff<br>aa: Input1 Lock status<br>bb: Input2 Lock status<br>where 01 = locked and 00 = unlocked<br>cc: Sync Flash<br>where 01 = Flashing and 00 = Not flashing |

| Command | Description | Response |
|---|---|---|
| | dd: Presence<br>where dd is a hex value built from the sum of:<br><br>01 = Green On Input 1<br>02 = Green On Input 2<br>04 = Green On Input 3<br>08 = Green On Input 4<br>10 = Red On Input 1<br>20 = Red On Input 2<br>40 = Red On Input 3<br>80 = Red On Input 4 | |
| SSx:nn | Source select<br>x selects which input is being changed where:<br>1 = Input 1 & 2<br>2 = Input 3 & 4<br>nn selects which source is used for that particular input where:<br>00 = AES<br>01 = SPDIF<br>02 = Optical | -ACK: |
| SYS:nn | Sync source select<br>nn selects which sync source is used where:<br>00 = Input 1 & 2<br>01 = Input 3 & 4<br>02 = AES<br>03 = Wordlock<br>04 = Video | -ACK: |
| UID: | Unit id | -UID:RB-DD4 |
| VER: | Version number<br>Where x.xxx is the firmware version and y.yyy id the front panel firmware version number | -VER:x.xxx,y.yyy |

**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

## SCi for the RB-DD4

SCi is the free of charge remote control software available from the Sonifex website.

SCi allows the user to control the RB-DD4 remotely. The interface has three tabs including a Status page, a Unit Setup page and a Miscellaneous option page. The status of the connection, serial number and firmware versions are always visible at the bottom of the interface.

## Status Page

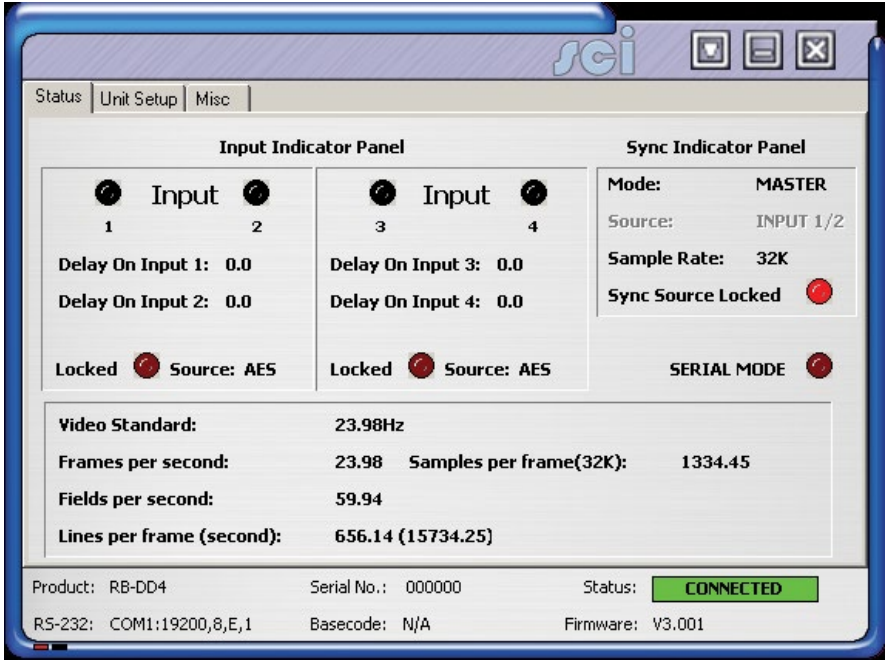



*Fig 11-14: Status Page*

This page displays the current status of the unit. Each set of inputs has an indicator panel which displays:

**Presence Level:** The current audio status is displayed as it is on the front panel. (Please refer to page 82).
**Delay on Input Labels:** Displays the delay applied to each input channel
**Locked LED:** If the input is locked, this LED is lit.
**Source Label:** The source which is currently being used is displayed here.

### 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.

The serial mode LED indicates whether the unit is in serial mode.
All the current delay information is displayed in lower panel.

## Unit Setup Page

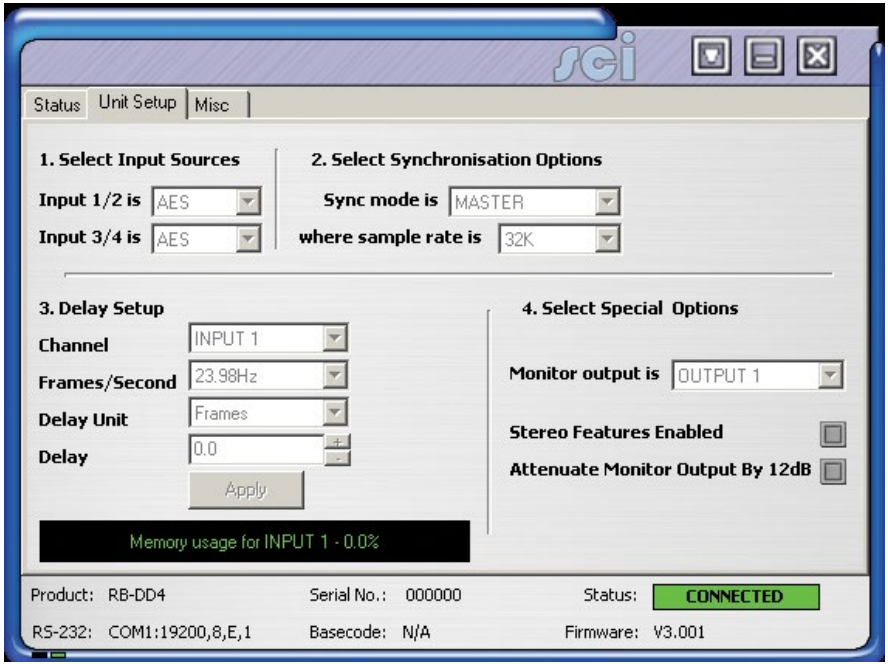



*Fig 11-15: Unit Setup Page*

The control page is where the user will set up their unit. For ease of use, the options have been split into four sections which the user visits in numbered steps:

**1. Select Input Sources**
Select which source you would like to use for both the Input 1 and input 2 from the drop down boxes labelled “INPUT 1/2 is” and “INPUT 3/4 is” respectively.

**2. Select Synchronization Options**
Select which sync mode to use from the first drop down box, labelled “Sync mode is”. If “MASTER” is chosen, then simply select your output sample rate from the drop down box labelled “where sample rate is”. If “AUTO”, “AUTOLOCK” or “SLAVE” are selected, then a new drop down box labelled “from” will appear which allows the user to select the synchronization source.

**3. Delay Setup**
Select all appropriate options from the drop down boxes and then type the delay value you require in the Delay edit box. Press the “Apply” to confirm this delay. Please note that changing the frame rate can affect the total allowed delay. The memory usage bar indicates how much of the allotted memory has been used for each channel. The maximum delay achievable is 8 seconds per channel at 32kHz.

**4. Select Special Options**
Select which output is monitored by using the drop down box labelled “Monitor output is”.

There are two tick options that can be selected. Select each one by clicking on the check box so that it displays a tick. The options are:

- Stereo features enabled
- Attenuate monitor output by 12dB

If serial mode is selected, all the controls are enabled. The controls are disabled if serial mode is switched off.

## Miscellaneous Page

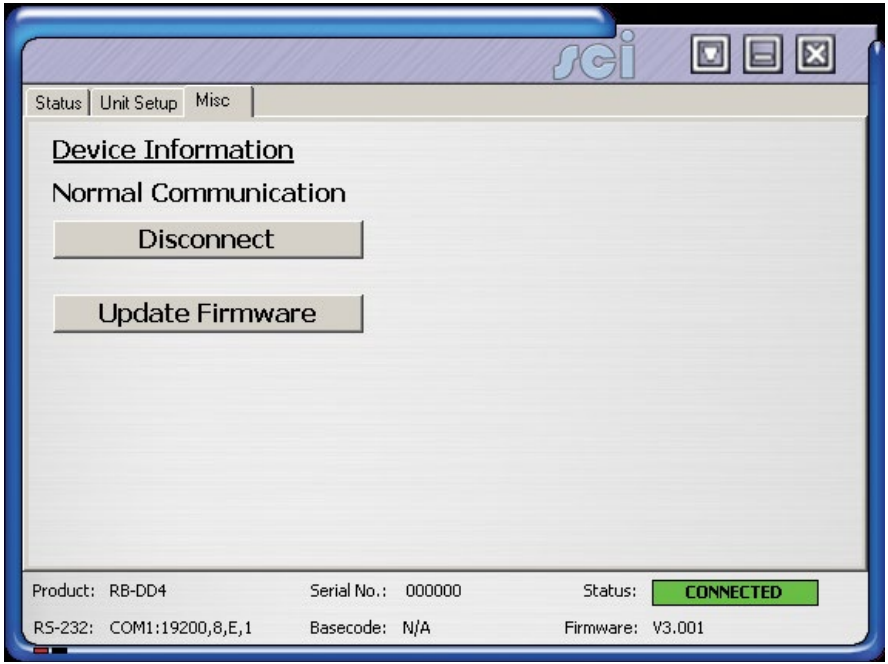



*Fig 11-16: Miscellaneous Page*

This page is used for the connecting and disconnecting SCi to the unit and updating the firmware. Any special modes are also selected from here.

## Updating the Firmware

The RB-DD4 firmware will at times be updated to add new features or 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-DD4 always have an “.dwn” extension. A progress bar appears 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 are all extinguished. The left program LED is then used to display the status of the upload:

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

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

**Successful Update:** The LED turns green for two seconds and the unit automatically resets and begins to run the new code.

**Unsuccessful Update:** The LED turns red for two seconds and the unit returns to running the last code.

# Technical Specification For RB-DD4

| 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<br>75Ω ±5% S/PDIF unbalanced I/O<br>75Ω ±5% TOSlink unbalanced I/O<br>50Ω BNC TTL word clock input |
| Signal Level: | Balanced: 3V/10V peak to peak min/max<br>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<br>via INPUTS 1 & 2 or INPUTS 3 & 4 push-buttons |
| Delay Control: | Delay time selection system via front panel push button |
| Monitor Volume Control Range: | -70dB to +12dB gain |
| Indicators: | Input presence indicators via bicolour LEDS around each push button |

| Rear Panel Operational Controls | |
|---|---|
| Master Select: | 32, 44.1, 48, 88.2, 96,176.4 or 192kHz<br>frequency via rear panel DIPSwitches |
| Sync Source Select: | INPUTS 1 & 2, INPUTS 3 & 4, AES Sync, Word Clock or optional Video Sync Board |

| Video Sync via rear panel DIP switches | |
|---|---|
| Sync Mode Select: | Master, Auto Sync, Auto Lock, Slave via rear panel DIPSwitches |
| Stereo Features: | Stereo monitor outputs via rear panel DIPSwitches |
| Monitor Attenuation: | 12dB Monitor attenuation via rear panel DIPSwitches |
| Serial Mode: | Enter serial control mode via rear panel DIPSwitches |
| Boot Mode: | Boot up base code or firmware via rear panel DIPSwitches |

| Connections | |
|---|---|
| Digital Inputs: | 2 x AES/EBU XLR 3 pin female<br>2 x S/PDIF RCA phono<br>2 x TOSLink optical input |
| Digital Outputs: | 2 x AES/EBU XLR 3 pin plug<br>2 x S/PDIF RCA phono socket<br>2 x TOSLink optical output |
| Sync Inputs: | 1 x AES/EBU XLR 3 pin female<br>1 x Word Clock BNC<br>1 x Video Input (optional) |
| Remote I/O Port: | 15 way D-type socket |
| Serial Port: | RS232, 9 Way D-Type socket |
| Mains Input: | Universal filtered IEC, continuously rated<br>85-264VAC@47- 63Hz, max 10W |
| Fuse Rating: | Anti-surge fuse 2A 20 x5mm |

| Equipment Type | |
|---|---|
| RB-DD4: | 4 Channel Digital Audio Delay Synchroniser |

| Physical Specifications | |
|---|---|
| Dimensions (Raw): | 48cm (W) x 10.8cm (D*) x 4.2cm (H) (1U)<br>19” (W) x 4.3” (D*) x 1.7” (H) (1U) |
| Dimensions (Boxed): | 53cm (W) x 25.5cm (D) x 6cm (H)<br>21” (W) x 10” (D) x 2.4” (H) |
| Weight: | Nett: 1.4kg Gross: 2.0kg<br>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

# 12 RB-AEC Acoustic Echo Canceller

## Introduction

*Fig 12-1: The RB-AEC Acoustic Echo Canceller*

### Category

Synchronisers, Delays & Silence Detectors.

### Product Function

Remove acoustic echo in a presenter's earpiece caused by microphones picking up audio from loudspeakers in delay.

### Typical Applications

In a TV production environment where presenters are fed a signal which has some form of acoustic delay or echo. Any situation where adaptive echo cancellaton is required.

### Features

The RB-AEC 1U rack-mount is an acoustic echo canceller primarily designed for the benefit of studio personnel for television and radio. When a studio presenter's microphone signal is played out through a monitor speaker in the control room, it can be picked up by the control room microphone(s) and returned to the presenter's earpiece as an undesirable echo.

In circumstances where green screen video processing is taking place, the delay can be greater than 200ms. Additionally, the dimensions, occupancy and distance between mouth and microphone can further influence the echo. The RB-AEC is used to remove the entire control room monitor speaker output from the presenter's feed by adapting to the environment in which the control room microphones are placed. Although acoustic echo cancellation is more commonly implemented in telephony systems, the Sonifex RB-AEC is designed to produce broadcast quality cancellation.

Much like during a conference call configuration between two rooms, each room has a microphone and speaker to conduct a conversation. When an occupant of one room speaks, it takes a certain length of time before it is received in the second room. Without a suitable solution this 'delayed' signal can then be captured by the microphone in the second room and returned back to the first room as an echo.

In the particular example of TV production, as well as the processing/ transmission delay, sound reflections from the control room monitor speaker into the control room microphone(s) cause the studio earpiece to suffer further delay. The sound reflections in the control room vary with the contents of the room including any personnel present. Also, different frequencies produce varying reflections across various types of surfaces and magnitudes within the room. For a 15m distance between speaker and microphone the delay is as much as 40ms.The DSP solution offered by the RB-AEC can dynamically compensate for varying configurations.

## Operation of the RB-AEC

The post-processed transmission output program from the studio (A) is sent to the RB-AEC as an analogue or digital audio signal (the stereo input is auto-sensing) which acts as a mix-minus to the input signal (B) from the Control Room. The RB-AEC removes the unwanted acoustic echoes so that the audio sent to the presenter's earpiece (C) is free of echoes and reflection artifacts.

The RB-AEC is a high performance, reliable single channel acoustic echo canceller designed to be a transparent addition to an exisiting studio environment.

The canceller will auto-detect analogue and digital inputs with sample rates up to 192kHz. Digital output is available when using a digital input and its sample rate is set to the rate detected at the input.

The settings for the RB-AEC are adjusted using the onboard webserver interface.

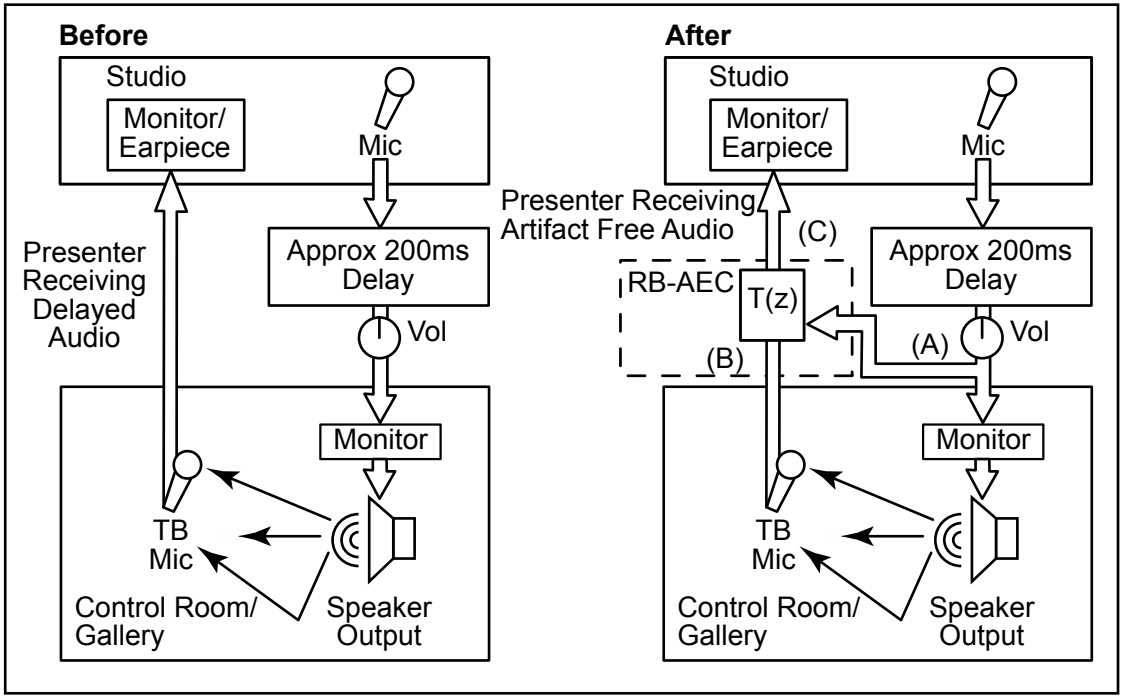



*Fig 12-2: The RB-AEC Operational Block Diagram*

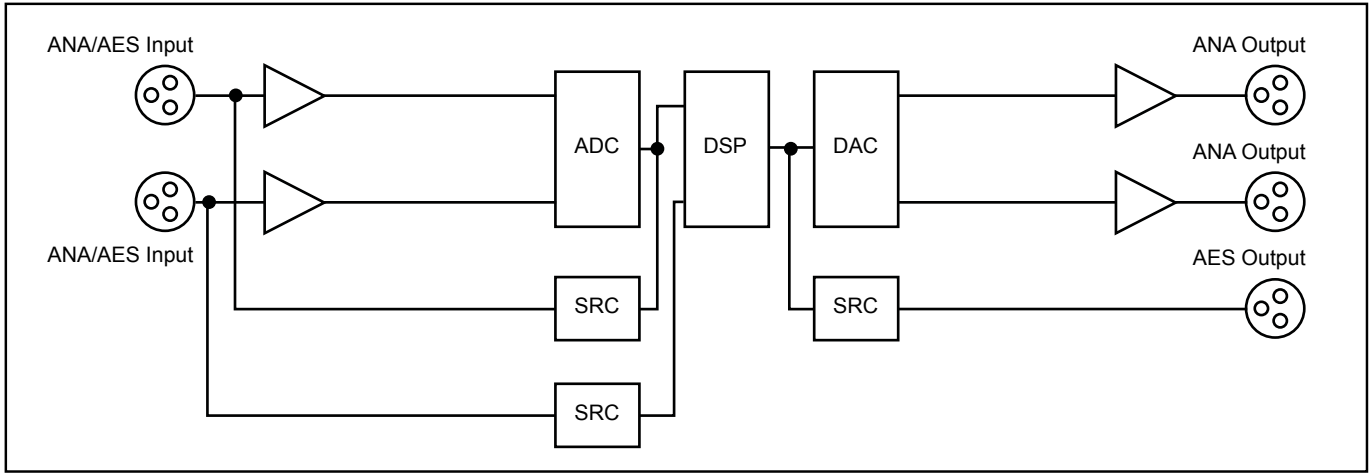



*Fig 12-3: The RB-AEC Functional Block Diagram*

## Front Panel Controls and Indicators

*Fig 12-4: The RB-AEC Front Panel*

**Power LED**

The POWER LED illuminates whilst internal power is present within the unit. If this indicator is not on, the most likely reason is simply the absence of mains power, but under fault conditions it may also indicate a ruptured mains fuse or a problem with the internal power supply module.

**Reset Button**

In the unlikely event that the unit fails to respond, press and hold the reset button for 3 seconds to reboot the unit

## Rear Panel Connections and Controls

### Inputs

**Far End Input**

An XLR 3 pin socket is used for the FAR END input channel and it can take an analogue or AES/EBU digital feed. The input will automatically detect whether the source is analogue or digital. It is electronically balanced, with the following connections:

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

The source of this XLR input must be from the audio feed that is the source of the echo. This is used by the cancelling algorithm to calculate what to remove from the near end.

**Near End Input**

An XLR 3 pin socket is used for the NEAR END input channel and it can take an analogue or AES/EBU digital feed. The input will automatically detect whether the source is analogue or digital. It is electronically balanced, with the following connections:

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

The source of this XLR input must be from the audio feed that is contaminated with the echo. It is from this feed, that the cancelling algorithm will be subtracting it's echo estimation to produce an echo free result.

### Outputs

**Analogue Outputs 1 & 2**

The analogue outputs consist of two XLR male connectors professionally balanced with following connections:

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

*Fig 12-5: The RB-AEC Rear Panel*

Both channels contain the same signal. When using analogue as the input type, only analogue outputs are enabled.

**Digital Output**

The digital output is connected using a male XLR connector on the rear panel. This XLR 3 pin plug has an impedance of 110Ω with the following connections:

Pin 1: Screen.

Pin 2: Phase.

Pin 3: Non-phase.

When using analogue as the input type, this output is disabled.

**General Note About Inputs & Outputs**

The analogue and digital outputs will both be present when digital inputs are used. The digital output is disabled when an analogue input is used because it detects the clock from the digital input to detect the input/ output sample rate. However, it it is possible to use different analogue and digital input configurations for the Near End and Far End inputs.

**Settings DIPSwitch**

Unit Settings are configured using a 4 way DIPSwitch:

| Settings DIPSwitch | |
|---|---|
| 1 | Far end digital input termination. ON = Terminated, OFF = Unterminated |
| 2 | Near end digital input termination. ON = Terminated, OFF = Unterminated |
| 3 | ON = Bootstrap mode, OFF = Normal mode |
| 4 | ON = Disable cancellation (Bypass), OFF = Enable cancellation |

***Input Termination***

110Ω input termination for the AES/EBU digital inputs are controlled by the settings of DIPSwitches 1 and 2, as follows:

**SW1:** When set to ON, Far end digital input is terminated with 110Ω.
**SW2:** When set to ON, Near end digital input is terminated with 110Ω.

***Bootstrap Mode***

To enable bootstrap mode, power the unit off, switch DIPSwitch 3 to ON and power up the unit. The RB-AEC can now be programmed with new firmware.

**Warning:** This removes the main firmware from the unit and should not be necessary unless the unit has been loaded with incorrect or corrupt firmware. Once in Bootstrap Mode you will need to load main firmware into the device using a standalone application. Contact Sonifex for more information.

***Bypass Cancellation***

To test whether the product is effective in your application, you can configure it and then toggle DIPSwitch 4 to listen to the effect with (OFF) and bypassed without (ON) cancellation.

**GPIO Connector**

This a 9 way D-type socket with the following connection details:

| GPIO Pin | | | |
|---|---|---|---|
| **Pin No** | **Signal** | **I/O** | **Description** |
| Pin 1 | GPO1 | O | General purpose output 1 - Power good |
| Pin 2 | GPO2 | O | General purpose output 2 |
| Pin 3 | GPO3 | O | General purpose output 3 |
| Pin 4 | GPI1 | I | General purpose input 1 - Mute input detect |
| Pin 5 | GPI2 | I | General purpose input 2 |
| Pin 6 | GPI3 | I | General purpose input 3 |
| Pin 7 | 12VDC | - | 50mA fused +12V power supply |
| Pin 8 | DGND | - | Ground return for external circuits |
| Pin 9 | DGND | - | Ground return for external circuits |

***Power Good***

Pin 1 shows a power fail condition. This open collector output is normally HIGH. Should a power failure occur, this level will go low (DGND).

***Mute Input Detect***

Setting the mute input detect pin 4 to DGND 0V will disable adaption. This may prevent the canceller from becoming unstable when the input is muted.

**Ethernet Port**

The Ethernet port is connected using a standard RJ-45 connector on the rear of the panel. It is a 10/100Mbps link. The unit can be controlled remotely over the Ethernet connection using the built-in webserver. By default the unit is configured for dynamic addressing using DHCP and Auto-IP. If a static IP address is required then this must be configured through the webserver.

The connections for the RJ45 connector are as follows:

| Ethernet Connector (RJ-45) | |
|---|---|
| **Pin No.** | **Function** |
| Pin 1 | Transmit data (+) |
| Pin 2 | Transmit data (-) |
| Pin 3 | Receive data (+) |
| Pin 4 | No connection |
| Pin 5 | No connection |
| Pin 6 | Receive data (-) |
| Pin 7 | No connection |
| Pin 8 | No connection |

## Mains Power

Power is applied via a standard three-pin IEC male socket. Mains voltages between 85V and 264V AC and frequencies between 47 and 63Hz are accepted without adjustment.

A 2A, 5 x 20mm SB fuse is used. The Earth pin MUST be connected to ensure safety.

## Applications

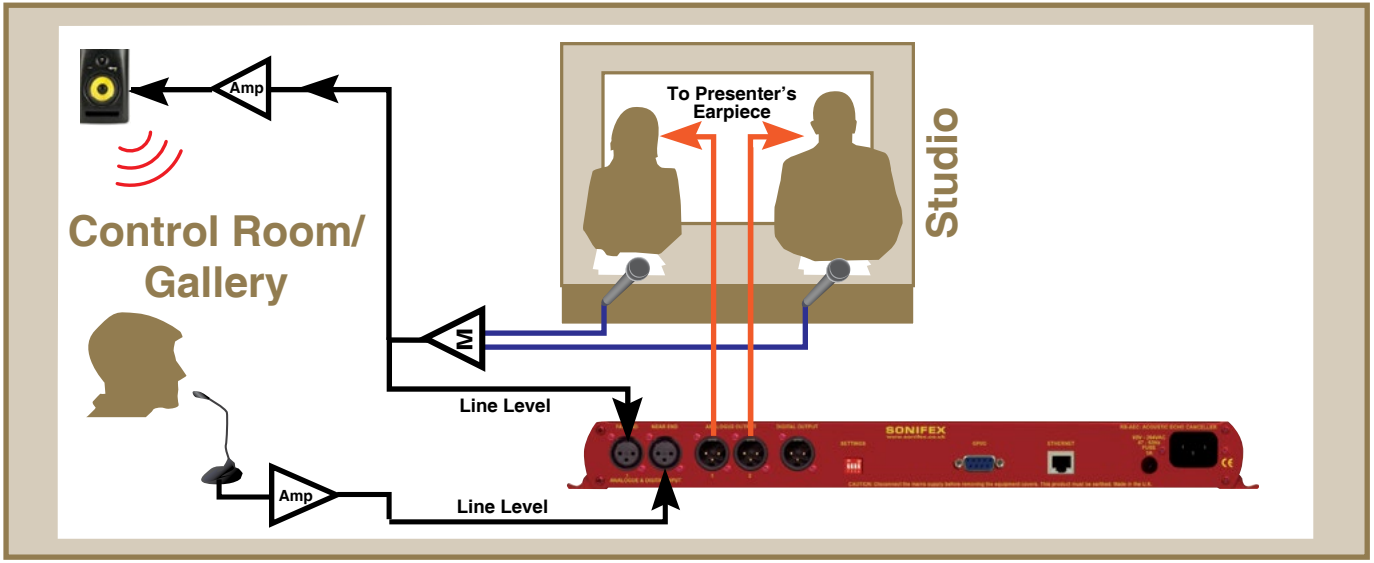
### How to Remove Delayed Presenter's Audio From Their Earpiece

#### The Problem:

Microphone audio from the presenter(s) is played aloud in the control room/gallery, where a live microphone allows the director to talk to the presenter(s) via their earpiece(s). This live microphone also inadvertently picks up the presenter's microphone audio and feeds it back to them in delay, causing a disconcerting self-echo in their earpiece.

#### The Solution:

1. Connect the director's microphone signal (at line level) to the RB-AEC near-end input.
2. Connect the gallery monitor signal (at line level, containing the presenter's microphone audio) to the RB-AEC far-end input.
3. Connect the RB-AEC output(s) to the presenter's earpiece(s). Turn on the RB-AEC.
4. Allow the presenter(s) to talk for 1 minute without the director using their microphone (though it must still be active). The RB-AEC will learn the environment and the echo will gradually disappear from the presenter's earpiece(s). The coefficients are stored and loaded on power-up. This process will not need repeating unless the environment changes.



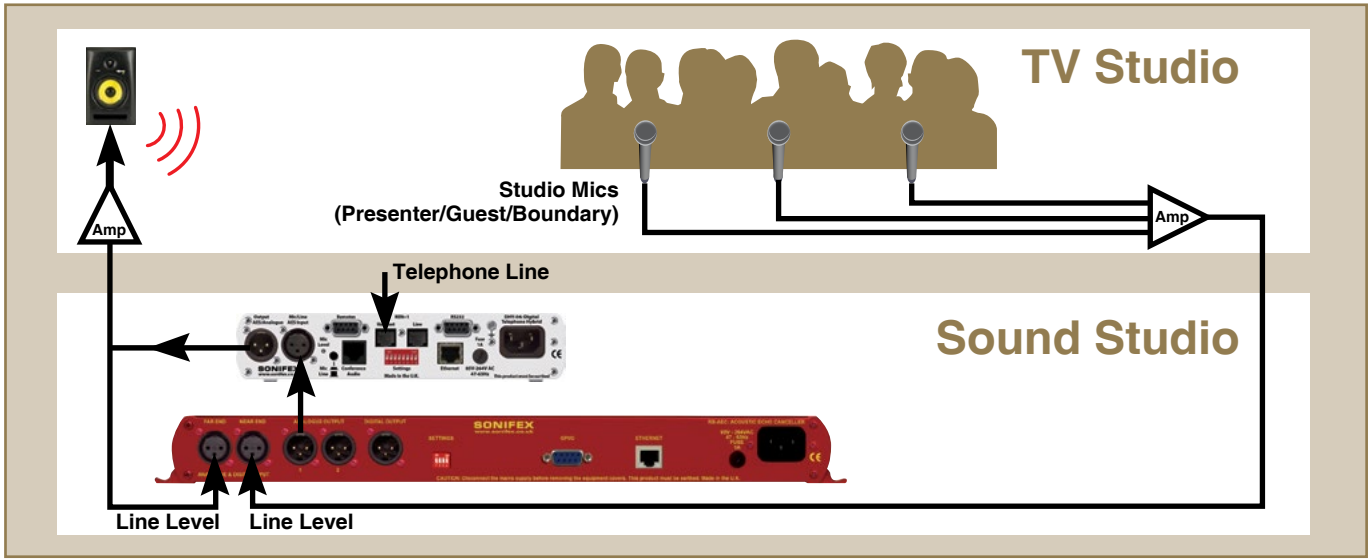
## How to Remove Delayed Caller Audio From The Telephone Line

### The Problem:

Caller audio from the telephone hybrid is played aloud in the TV studio, where live microphones are fitted to presenters and guests (boundary microphones may also be present). These live microphones also inadvertently pick up the caller's audio and feed it back to them in delay, causing an undesirable self-echo.

### The Solution:

1. Connect the mixed studio microphones signal (at line level) to the RB-AEC near-end input.
2. Connect the TBU output to the RB-AEC far-end input.
3. Connect the RB-AEC output(s) to the hybrid's input (in line mode). Turn on the RB-AEC.
4. Make a test call and allow the caller to talk for 1 minute, their voice played aloud through the studio monitor and being picked up by all studio microphones (which should be in-position and active but not being used). The RB-AEC will learn the environment and the echo will gradually disappear from the phone line. The coefficients are stored and loaded on power-up. This process will not need repeating unless the environment changes.



## Calibration

When using the RB-AEC for the first time, the unit must first be calibrated to suit the environment in which it will be used. Using the webserver, select the 'Config' tab.

1. Select the desired sample rate.
2. Measure the distance between the speaker and microphone in centimetres. Halve this number and enter it here. Example: For a distance of 100cm, enter 50. Otherwise leave this value at zero.
3. Tick the box labelled 'Load Saved Coefficients on Startup'. This will ensure that any saved coefficients are used next time the unit is switched on.
4. Set the presenter's(Far End) silence detect threshold. Levels below this value will disable the canceller.
5. You also have the option of enabling the noise gate, setting the threshold and changing the release time.
6. Submit your settings
7. For calibration, ensure the control room is silent and speak into the presenter's microphone while monitoring using the presenter's earpiece for around one minute or until the echo has subsided to a satisfactory level.
8. Save your coefficient values so they can be used as a starting point for next time the RB-AEC is used.

## Webserver & Unit Discovery

The RB-AEC uses the Zeroconf networking methodology to allow the unit to be a "plug and play" device. Just simply connect it to the network and use it. The unit is assigned an IP address using a DHCP server, or a self-assigned address using AUTOIP when one isn't available. The RB-AEC can also use Bonjour discovery on a network.

Once the RB-AEC has been connected to the network, it will try to obtain an IP address from a DHCP server on the network. If none is found then the unit will use Auto-IP to assign itself an IP address.

There is no direct method for you to determine exactly which IP address has been assigned to the unit. However, using a Bonjour discovery application or browser plug-in, the unit can be discovered and connected to with relative ease.

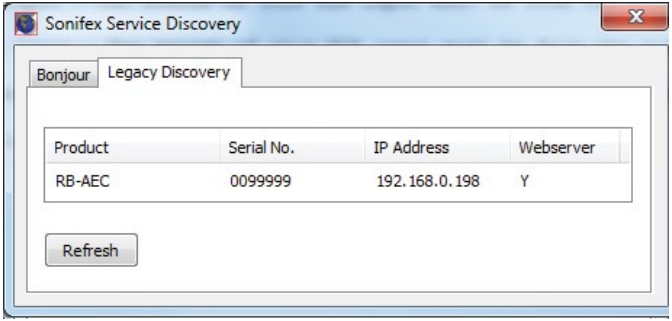



*Fig 12-6: The RB-AEC Sonifex Service Discovery Tool*

The Sonifex device can be detected using the Sonifex Discovery Tool and the device's webserver may be launched.

Bonjour is available as standard on MAC OS and as plug-in for Internet Explorer for the 'Bonjour for Windows' download. You can also use the discovery application provided by Sonifex and available for download on the Sonifex website www.sonifex.co.uk

**Running The Webserver**

Use Google Chrome or Mozilla Firefox browsers and simply type in the RB-AEC IP address.

## Home Page

Welcome to the RB-AEC1 webserver interface.

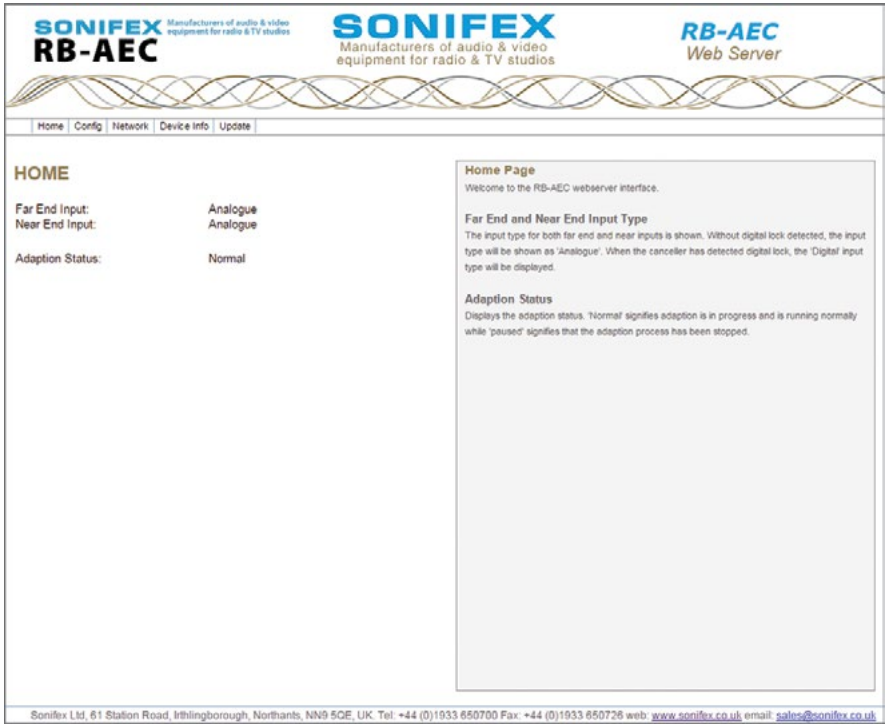



*Fig 12-7: The RB-AEC Webserver Interface - Home*

The input type for both far end and near inputs is shown. Without an AES/EBU digital lock detected, the input type will be shown as 'Analogue'. When the unit has detected digital lock, the 'Digital' input type will be displayed

The input type for both far end and near inputs is shown. Without digital lock detected, the input type will be shown as 'Analogue'. When the canceller has detected digital lock, the 'Digital' input type will be displayed.

### Far End and Near End Input Type

The input type for both far end and near inputs is shown. Without digital lock detected, the input type will be shown as 'Analogue'. When the canceller has detected digital lock, the 'Digital' input type will be displayed.

### Adaption Status

Displays the adaption status. 'Normal' signifies adaption is in progress and is running normally while 'paused' signifies that the adaption process has been stopped.

## Configuration Page

This page shows the configuration options for the RB-AEC.

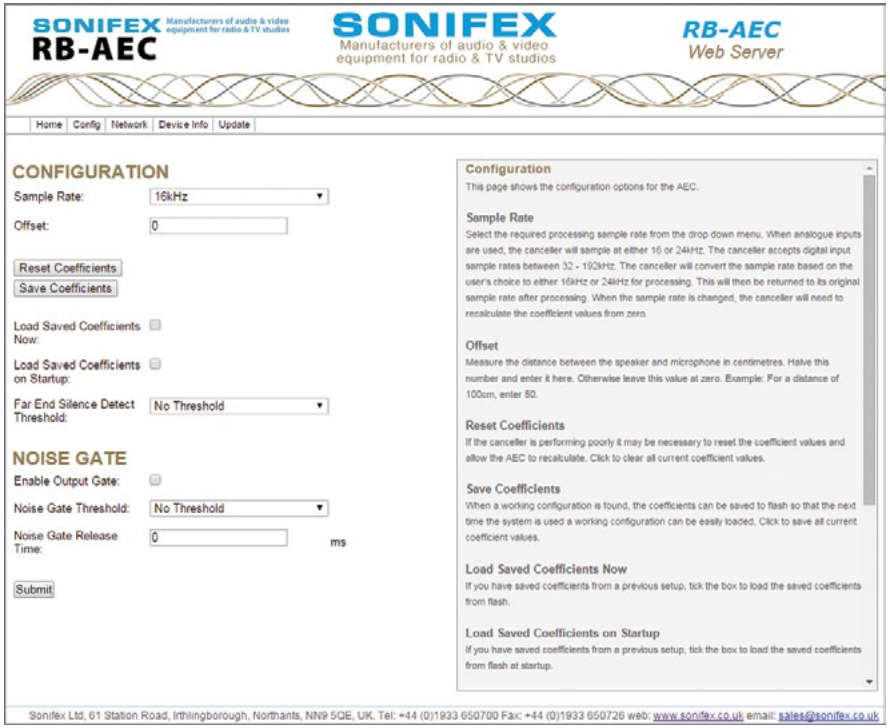



*Fig 12-8: The RB-AEC Webserver Interface - Configuration*

**Sample Rate**

Select the required processing sample rate from the drop down menu. When analogue inputs are used, the canceller will sample at either 16 or 24kHz. The canceller accepts digital input sample rates between 32 - 192kHz. The canceller will convert the sample rate based on the user's choice to either 16kHz or 24kHz for processing. This will then be returned to its original sample rate after processing. When the sample rate is changed, the canceller will need to recalculate the coefficient values from zero.

**Offset**

Measure the distance between the speaker and microphone in centimetres. Halve this number and enter it here. Otherwise leave this value at zero. Example: For a distance of 100cm, enter 50.

**Reset Coefficients**

If the canceller is performing poorly it may be necessary to reset the coefficient values and allow the AEC to recalculate. Click to clear all current coefficient values.

**Save Coefficients**

When a working configuration is found, the coefficients can be saved to flash so that the next time the system is used a working configuration can be easily loaded. Click to save all current coefficient values.

**Load Saved Coefficients Now**
If you have saved coefficients from a previous setup, tick the box to load the saved coefficients from flash.

**Load Saved Coefficients on Startup**
If you have saved coefficients from a previous setup, tick the box to load the saved coefficients from flash at startup.

**Far End Silence Detect Threshold:**
This value sets the threshold at which the adaption process is paused during periods where no-one is speaking at the far end. Select a threshold from the drop down list. -6dBFS will set the threshold to half of the maximum value, while 'No Threshold' will allow the adaption to continue all the time.

**Enable Noise Gate:**
When enabled, the noise gate will lower the output level based on the threshold set below.

**Noise Gate Threshold:**
This threshold sets the point at which the noise gate is enabled or disabled.

**Noise Gate Release Time:**
After the noise gate has been triggered, a length of time can be set before the gate can be released. This helps prevent the gate from being activated during speech. Set a time in milliseconds between 0 and 500.

## Network Page

Displays the Hostname, Static IP Address, Static subnet mask, Gateway IP address, and whether DHCP and Auto IP are enabled.

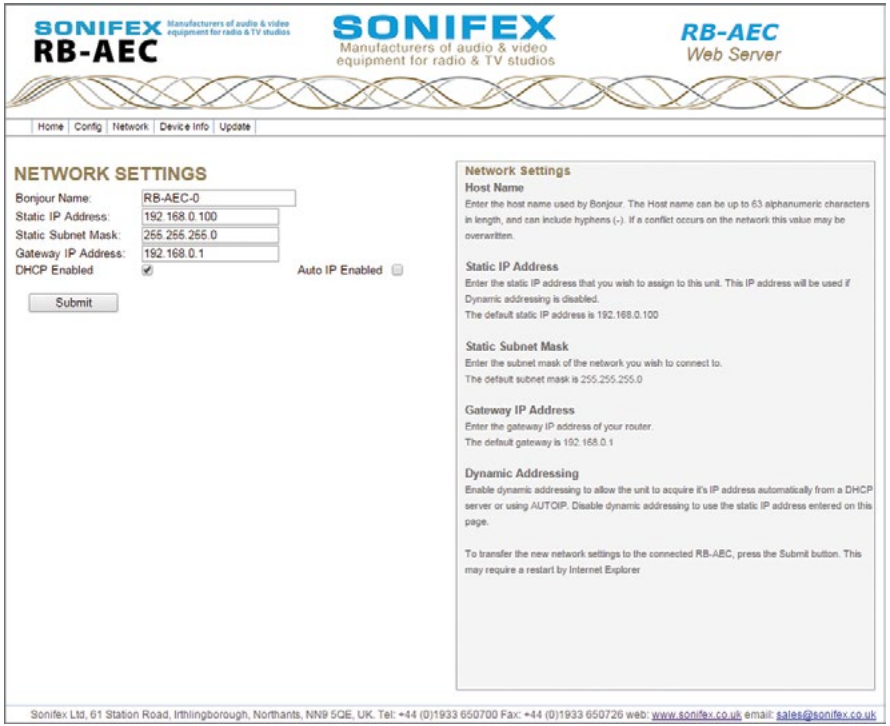



*Fig 12-9: The RB-AEC Webserver Interface - Network*

## Network Settings

**Host Name**

Enter the host name used by Bonjour. The Host name can be up to 63 alphanumeric characters in length, and can include hyphens (-). If a conflict occurs on the network this value may be overwritten.

**Static IP Address**

Enter the static IP address that you wish to assign to this unit. This IP address will be used if Dynamic addressing is disabled.

The default static IP address is 192.168.0.100

**Static Subnet Mask**

Enter the subnet mask of the network you wish to connect to.

The default subnet mask is 255.255.255.0

**Gateway IP Address**

Enter the gateway IP address of your router.

The default gateway is 192.168.0.1

**Dynamic Addressing**

Enable dynamic addressing to allow the unit to acquire it's IP address automatically from a DHCP server or using AUTOIP. Disable dynamic addressing to use the static IP address entered on this page.

To transfer the new network settings to the connected RB-AEC, press the Submit button. This may require a restart by Internet Explorer

## Device Info Page

This page shows general information about the connected RB-AEC.

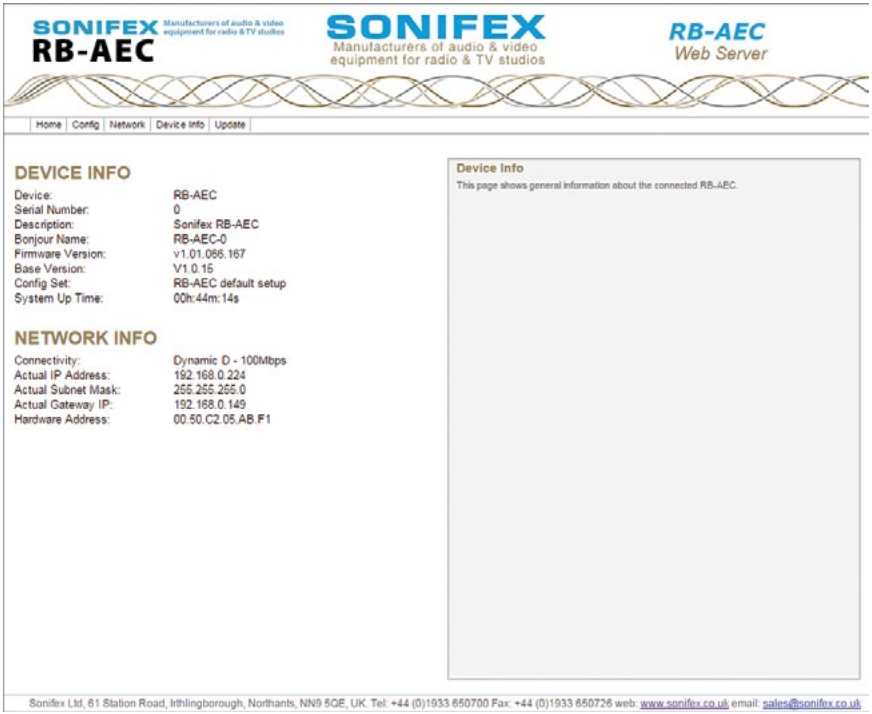



*Fig 12-10: The RB-AEC Webserver Interface - Device Info*

## Update Page

To update the RB-AEC firmware, download the latest file from the Sonifex website, select Choose File to find it and click Update to load it into the RB-AEC.

*Fig 12-11: The RB-AEC Webserver Interface - Update*

**Update**

The version of firmware currently running on this unit is V1.0.6.

To find out if there is new firmware for this unit, check the Sonifex website. If an update is available, download the latest version and save the file to your computer. Browse your computer to locate and select the file, and press the update button. Once the update has started, this page will update automatically.

The file must be named according to the following convention “RB-AEC” followed by any version or other info then a “.DWN” suffix - e.g “RB-AEC v1_02 special release.dwn”

## Technical Specification For RB-AEC

| Audio Specification | |
|---|---|
| Audio Input (Near End/Program): | 1 x mono analogue or AES/EBU digital on XLR 3-pin female (autoselecting) |
| Audio Input (Far End/ From Control Room): | 1 x mono analogue or AES/EBU digital on XLR 3-pin female (autoselecting) |
| Max Level (0dB Input Gain): | +18dBu (analogue)<br>0dBFS (digital) |
| CMRR: | >60dB typical |
| Input Impedance: | 20kΩ (analogue) 110Ω (digital with termination switchable) |
| AES/EBU Input: | 32kHz to 192kHz |
| Audio Outputs (Analogue): | 2 x mono analogue on XLR 3-pin male |
| Audio Outputs (AES/EBU): | 1 x stereo digital AES/EBU on XLR 3-pin male |
| Maximum Output Level: | +18dBu (analogue)/<br>0dBFS (digital) |
| Output Impedance: | <50Ω (analogue)/110Ω (digital) |
| AES/EBU Output Sample Rates: | Selectable 32kHz - 192kHz |
| Distortion: | fs = 16kHz, THD+N<0.01%, 8dBu, 20Hz-8kHz, unity gain, 8kHz BW<br>fs = 24kHz, THD+N<0.01%, 8dBu, 20Hz-12kHz, unity gain, 12kHz BW |
| Noise: | -84dB RMS, unity gain ref +8dBu output |
| Frequency Response: | 20Hz-9.5kHz +0/-0.5dB |
| Rejection Ratio (Input to Output): | Typically 35dB on complex waveforms, reference peak level of 0dB |
| Remote I/O Port: | 9 way D-type socket |
| Ethernet Port: | 1 x RJ45 with status LEDs |
| Mains Input: | Filtered IEC, continuously rated 85 264VAC @ 47-63Hz, 10W max |
| Fuse Rating: | Anti-surge fuse 1A 20 x 5mm |

| Controls | |
|---|---|
| Configuration: | 1 x Ethernet port, webserver<br>1 x rear panel 4-way DIPSwitch |
| Reset: | 1 x front panel recessed button |

| Equipment Type | |
|---|---|
| RB-AEC: | Acoustic echo canceller |

| Physical Specification | |
|---|---|
| Dimensions (Raw): | 48cm (W) x 10.8cm (D) x 4.2cm (H)(1U)<br>19” (W) x 4.3” (D) x 1.7” (H) (1U) |
| Dimensions (Boxed): | 58.5cm (W) x 22.5cm (D) x 7cm (H)<br>23” (W) x 8.9” (D) x 2.8” (H) |
| Weight: | Nett: 1.5kg Gross: 2.0kg<br>Nett: 3.3lbs Gross: 4.4lbs |

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