SONIFEX

Manufacturers of Audio & Video
Products For Radio & TV Broadcasters

New Products
Update
September 2011

SONIFEX - ONLY THE BEST -

This edition contains all updates to the 2010/11 catalogue

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Redbox - Audio Distribution Amplifiers

RB-DA6R

6 Way Stereo Distribution Amplifier With RJ45 Connectors

StudioHub+ RK3

The RB-DA6R is a 1U rack-mount high performance 6 way stereo distribution amplifier for splitting a source into a number of different outputs. The amplifier provides multiple balanced audio outputs using RJ45 connectors wired to the StudioHub+™ standard *.

By using RJ45 outputs, it allows simpler CAT5 cabling to used to connect the amplifier to other products.

The RB-DA6R has one stereo input which is switchable via a rear panel push switch between paralleled balanced inputs (2 x XLR sockets or 1 x stereo input on RJ45) and unbalanced inputs (1 pair of stereo phono sockets).

The unit has 6 stereo outputs on 6 x RJ45 connectors. The unit can also be configured so that 1 mono input can be distributed to 12 outputs by use of a switch which is recessed on the front panel to prevent it being accidentally knocked.

The inputs and outputs are electronically balanced and can be wired unbalanced. Each output is individually buffered so that a short circuit on one output won't affect the others. The left and right input gain controls (normalising) are pre-set potentiometers accessible through the front panel.

The output gain may be varied from -8dB to 18dB which is useful for normalising consumer and professional signals to give outputs of -15dBu and 0dBu respectively.

Audio Specification For RB-DA6R	
Maximum Input Level:	+28dBu
Maximum Output Level:	+28dBu
Frequency Response:	20Hz to 20kHz ±0.1dB (600Ω load, ref 1kHz)
Gain Range:	Adjustable 8dB loss to 18dB gain (L & R adjust)
Common Mode Rejection:	>66dB typically
Input impedance:	>20kΩ bridging (balanced) >10kΩ (unbalanced)
Output impedance:	<50Ω
Distortion:	0.01% THD @ 1kHz, ref +8dBu output
Noise:	-100dB unity gain, ref +8dBu output
Connections	
Inputs:	2 x XLR 3 pin female, 1 x RJ45 socket balanced) 2 x phono sockets (unbalanced)
Outputs:	6 x RJ45 sockets (all balanced, can be unbalanced)
Mains Input:	Filtered IEC, switchable 110-120V, or 220-240V, fused, 6W max
Fuse Rating:	Anti-surge fuse 100mA 20 x 5mm (230VAC) Anti-surge fuse 250mA 20 x 5mm (115VAC)
Equipment Type	
RB-DA6R:	6 way stereo distribution amplifier with RJ45 connectors
Physical Specification	
Dimensions (Raw):	48cm (W) x 10.8cm (D) x 4.2cm (H) (1U) 19" (W) x 4.3" (D) x 1.7" (H) (1U)
Dimensions (Boxed):	58.5cm (W) x 22.5cm (D) x 7cm (H) 23" (W) x 8.9" (D) x 2.75" (H)
Weight:	Nett: 1.3kg Gross: 1.9kg Nett: 2.9lbs Gross: 4.2lbs
Accessories	
RB-RK3:	1U Rear panel rack kit for large Redboxes

RB-DA6RG

6 Way Stereo Distribution Amplifier With RJ45 Connectors & Output Gain Control

StudioHub+ RK3

The RB-DA6RG is a 1U rack-mount high performance 6 way stereo distribution amplifier for splitting a source into a number of different outputs. It is identical to the RB-DA6R with the addition of individual output gain adjustment, instead of global stereo gain adjustment.

Audio Specification For RB-DA6RG	
As per RB-DA6R with the following difference:	
Gain Range:	Adjustable 8dB loss to 18dB gain (12 adjustable pots)

** StudioHub+™ is a registered trademark of Radio Systems Inc*

Redbox - Audio Distribution Amplifiers

RB-DDA22

Digital Audio Distribution Amplifier With Multiple Outputs

* StudioHub+[TM] is a registered trademark of Radio Systems Inc

The RB-DDA22 is a multiple input and multiple output digital audio distribution amplifier designed to accept one of 5 different digital inputs and output to 22 digital audio outputs of 5 different connector types. It is used for distributing digital audio data in many formats, repeating both the audio data and the status information of the input whilst re-clocking the data and re-normalising to the appropriate standard digital audio levels.

Input Signals:

- 1 x XLR 3 female AES/EBU digital audio input
- 1 x BNC AES-3id digital audio input
- 1 x RJ45 AES/EBU digital audio input
- 1 x RCA Phono S/PDIF input
- 1 x TOSLink digital audio optical input

Output Signals:

- 6 x XLR-3 male AES/EBU digital audio outputs
- 6 x BNC AES-3id digital audio outputs
- 6 x RJ45 AES/EBU digital audio outputs
- 2 x RCA Phono S/PDIF outputs
- 2 x TOSLink optical outputs

The RJ-45 connectors are wired to be StudioHub+ compatible, a format defined by Radio Systems Inc.

Applications include distributing audio from a digital mixing desk to multiple digital recorders, or feeding multiple studios with an output from a USB audio player.

It can accept input sample rates in the range of 30kHz - 200kHz, and bit rates of 16, 20 and 24 bit. So, it can be used for standard CD signal distribution at 16 bit 44.1kHz, as well as for high quality 24 bit 96kHz or 192kHz recording.

The front panel has an INPUT SELECT button and 7 indicator LEDS. The button is used to select the input to be used and shows the mode of operation of the unit. The LEDs will indicate, depending on the mode of operation, the input selected, valid signal presence, or the operational frequency.

In 'operating mode', the central button LED is unlit. The unit selects this mode when a valid input signal has been present and no button presses have been made for 8 seconds. The LEDs around the button illuminate amber to directly indicate the following frequencies 32, 44.1, 48, 88.2, 96, 176.4 & 192 kHz. If the input is at a non-standard frequency then the unit will indicate both the frequencies either side of the incoming frequency – e.g. if the input signal has a frequency of 64kHz then both the 48kHz and 88.2kHz LEDs will be illuminated. If the frequency is below 32kHz or above 192kHz then the appropriate LED will flash.

Pressing the INPUT SELECT central button illuminates it red and allows you to select the required input. Pressing the button will step the selected input to the next input selection including auto mode.

In Input Select Mode the LEDs have the following states:

Off – Signal not selected & no valid signal.
Green – Signal not selected & valid signal.
Red – Signal selected & no valid signal.
Amber – Signal selected & valid signal.
present or Auto Mode Selected.

Note: Input Select mode is chosen automatically when the selected input is not present.

In this special input select mode the unit will hunt through all the inputs until it finds a valid signal. If multiple signals are present you can force selection of the next input by holding the input select button down for 2 seconds, when the next valid input will be selected.

Technical Specification For RB-DDA22	
Audio Specification	
Input & Output Impedances:	110Ω ±20% AES/EBU balanced I/O 75Ω ±5% S/PDIF unbalanced I/O
Signal Level:	Balanced: 3V/10V peak to peak min/max Unbalanced: Min 0.5V±20% peak to peak
Sample Frequencies:	32, 44.1, 48, 88.2, 96,176.4 or 192kHz
Bit Depth:	Up to and including 24 bit
Front Panel Operational Controls & Indicators	
Digital Input Select:	AES/EBU (XLR), AES/EBU (BNC), AES/EBU (RJ45), S/PDIF or TOSlink optical
Indicators:	Input presence indicators via tricolour LEDS around the input select button
Connections	
Digital Inputs:	1 x AES/EBU XLR 3 pin socket 1 x AES/EBU BNC 1 x AES/EBU RJ45 socket (StudioHub+[TM]) 1 x S/PDIF RCA phono socket 1 x TOSLink optical input
Digital Outputs:	6 x AES/EBU XLR 3 pin plug 6 x AES/EBU BNC 6 x AES/EBU RJ45 socket (StudioHub+[TM]) 2 x S/PDIF RCA phono socket 2 x TOSLink optical output
Mains Input:	Universal filtered IEC, continuously rated 85-264VAC@47- 63Hz, max 10W
Fuse Rating:	Anti-surge fuse 1A 20 x 5mm
Equipment Type	
RB-DDA22:	Digital audio distribution amplifier with multiple outputs
Physical Specification	
Dimensions (Raw):	48cm (W) x 10.8cm (D) x 4.2cm (H) (1U) 19" (W) x 4.3" (D) x 1.7" (H) (1U)
Dimensions (Boxed):	58.5cm (W) x 22.5cm (D) x 7cm (H) 23" (W) x 8.9" (D) x 2.75" (H)
Weight:	Nett: 1.4kg Gross: 2.0kg Nett: 3.0lbs Gross: 4.4lbs
Accessories	
RB-RK3:	1U Rear panel rack kit for large Redboxes

Redbox - Synchronisers, Delays & Silence Detectors

RB-DSD8

8 Channel Silence Switcher

RK3

The RB-DSD8 8 channel silence switcher works in a similar way to the Sonifex RB-SD1 and RB-DSD1 but allows for 4 stereo channels of audio. These stereo audio channels can be either analogue or digital and can be used independently to give 4 stereo silence detectors or they can be linked to handle multichannel audio inputs, e.g. for 5.1 and 7.1 surround systems. Designed to switch from one input (or set of inputs) to another in the event of loss of audio, the unit is ideal at transmitter sites, or after the master output of a studio, to switch in another audio source, or simultaneous broadcast, should a master source fail.

The unit can switch:

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

The 2 x 8 channel audio inputs can accept both digital and analogue connections, with the unit automatically recognising a digital input. For the 8 channel outputs, by setting the appropriate DIPSwitches, each stereo output can be designated as either an analogue pair or as a digital output, thus making the RB-DSD8 incredibly flexible and suitable for many different applications. The unit level settings are in dBFS but when using analogue signals the equivalent full scale value can be set to +24dBu, +18dBu, or +12dBu by DIPSwitches.

Each stereo pair has individual settings and controls and when stereo signals are linked, the foremost pair determines the switching characteristics and controls to be used. Each stereo pair has an AES LED that shows the status of the digital audio on that channel and a Selection LED to show that input is currently being sent to each respective output. Two Presence LEDS for the left and right inputs of each stereo pair indicate the input level of the channels.

The unit can switch between sources manually or automatically at the push of a button. If switching manually, silence detection is disabled and the user chooses when to switch using the main or backup buttons. If switching automatically, the unit switches between the two sources automatically upon the detection of silence. Each pair can be set to switch manually or automatically and the current setting is indicated by the Mode LED. Link/Select buttons are used to group channels together to access multichannel operation and switch simultaneously. Each pair has a Link/Select button which illuminates blue when active. Pressing and holding the first Link/Select button with any other Link/Select button causes all inputs up to that point to be selected.

The RB-DSD8 has a 'slave mode' that allows you to connect two RB-DSD8 units and control them simultaneously from one unit.

The silence detect level is adjustable between -39dBFS and -84dBFS in 3dBFS steps via DIPSwitches and this level is compared to peak signals. The silence interval can be adjusted between 2 seconds and 254 seconds in 2 second steps via DIPSwitches.

A powerful feature of the RB-DSD8 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 set up the unit for your specific application. A DIPSwitch configures the unit to be controlled serially which is indicated by a front panel LED. You can control the unit remotely using either USB or Ethernet.

The RB-DSD8 has been designed with dual redundant power supplies. This means that if either power supply fails, the other is ready to take over. In the extremely unlikely event that both fail, the unit has been designed with a passive signal path through the main input. 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.

Contact Sonifex for further information if you have a particular requirement that isn't catered for by the RB-DSD8 as standard.

Clocking & Synchronisation

All digital input signals are routed to a sample rate converter allowing mixed incoming sample rates to be used. The output sample rates are selectable from a predefined master clock of 32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz or the clock can be derived from a sync input. When analogue inputs are selected, the analogue to digital converters are also clocked at that sample rate.

DIPSwitches choose the synchronisation mode and the sync source from TTL wordclock or AES/EBU through the dual-purpose synchronising input as standard. A front panel indicator shows the status of the synchronization input. Selectable sync modes are as follows:

Redbox - Synchronisers, Delays & Silence Detectors

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 Lock Mode

In this mode no output is 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 is set by, and locked to, the internal on-board clock generator at the closest frequency available to the previous sync input.

Slave Mode

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

Audio Specification For RB-DSD8	
Audio Specification - Digital	
Dynamic Range:	>138dB
Distortion & Noise:	<-137dB THD + N at 1kHz, ref 0dBFS
Input & Output Impedances:	110Ω ± 20% AES/EBU balanced I/O 50Ω BNC TTL word clock input
Signal Level:	Balanced: 3V/10V peak to peak min/max
Sample Rates:	32, 44.1, 48, 88.2, 96,176.4 or 192kHz
Bit Depth:	Up to and including 24 bit
Audio Specification - Analogue	
Maximum Input Level:	+24dBu
Input Impedance:	>20kΩ bridging balanced
Dynamic Range:	>110dB
Distortion & Noise:	>82dB THD + N at 1kHz
Common Mode Rejection:	>60dB , ref 0dBu
Front Panel Operational Controls	
Switch Mode Select:	Via AUTO, MANUAL or SLAVE push-buttons
Manual Source Select:	Via MAIN and BACKUP push-buttons
Group Selection :	Via LINK/SELECT push-buttons
Front Panel Indicators	
Presence LEDs:	For all input channels
Link LEDs:	Show which channels are controlled concurrently
Mode LEDs:	Indicate the current mode selected for each group
Selection LEDs:	Indicate whether MAIN or BACKUP is selected
AES LEDs:	Show the state of the digital input to each group
PSU LEDs:	Show the state of each power supply
Remote Control LED:	Show if remote control is selected
External Sync LED:	Show the state of any sync inputs used.
Rear Panel - Operational Controls	
Silence Threshold:	-39dBfs to -84dBfs in 3dBfs steps, via rear panel DIPSwitches
Silence Duration:	0 - 254 seconds in 2 second intervals duration, via rear panel DIPSwitches
Stereo/Mono Switching:	Stereo or mono, via rear panel DIPSwitch
Master Select:	32, 44.1, 48, 88.2, 96,176.4 or 192kHz Output sample rate, via rear panel DIPSwitches
Ignore Silence:	Each channel can be set to ignore silences, via rear panel DIPswitches
Remote Control Enable:	Enabled or disabled, via rear panel DIPswitch
Sync Mode & Source Select:	Sync in master mode or sync from MAIN input1, AES or wordclock sync input in auto or slave mode, via rear panel DIPswitches
Remote Start:	Latched or momentary, via DIPswitch
Input Lock Loss:	Switch immediately or treat as silence delay, via rear panel DIPswitch
Digital or Analogue Output:	Digital or analogue, via rear panel DIPswitches
Full Scale Line Up:	24, 18 or 12 dBu = 0dBFS, via rear panel DIPswitches
Boot Mode:	Boot in boot or normal via rear panel DIPswitch
Connections	
Digital/Analogue Inputs:	2 x 8 channel inputs on 2 x 25 pin D-type male
Digital/Analogue Outputs:	1 x 8 channel outputs on 1 x 25 pin D-type female
Sync Inputs:	1 x BNC (Wordclock or AES)
Remote I/O Port:	25 way D-type female
SCi port:	USB or ethernet
Mains Input:	2 x Universal filtered IEC, continuously rated 85-264VAC @47- 63Hz, max 60W
Fuse Rating:	2 x Anti-surge fuse 2A 20 x 5mm
Equipment Type	
RB-DSD8:	8 channel silence switcher
Physical Specifications	
Dimensions (Raw):	48cm(W) x 22cm(D) x 4.2cm(H) 1U 19" (W) x 8.7" (D) x 1.7" (H) 1U
Dimensions (Boxed):	55cm(W) x 28cm(D) x 17cm(H) 21.7"(W) x 11"(D) x 6.7"
Weight:	Nett: 2.3kg Gross: 3.8kg Nett: 5.1lb Gross: 8.4lb

* Note that this product is deeper than standard Redboxes

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

RB-DEED8

Dolby® E Encoder 8 Channel, Digital Inputs

RK3

Dolby® and the double-D symbol are registered trademarks of Dolby Laboratories

The RB-DEED8 is a Dolby® E Encoder. It encodes up to 8 digital audio channels into a single Dolby bitstream output. There is also an extra Dolby output for connection to monitoring equipment.

Each digital input channel has independent level control, which can be adjusted from -24dB through to +24dB in 0.5dB steps. The digital inputs and outputs can be selected to be either balanced or unbalanced AES/EBU specification, and can be connected through either BNC or D-type connectors. These I/O connections are paralleled, allowing one type to be used per input or output.

Metadata can be input to the encoder using RS-485 via the external 9-pin D-type socket on the rear panel and can also be viewed using the SCi software.

A video input is used to genlock the audio output and Dolby® E encoding to a known video reference. This input autodetects between NTSC, PAL or Tri-level sync.

8 general purposes inputs and 8 outputs are available on a rear panel 25 way D-type socket whose function can be programmed using the menu, e.g. alarm outputs for loss of input or encoder errors.

The unit is controlled locally using the front panel display and control knob but can be remote controlled via an ethernet or RS-232 serial connection using the Sonifex SCi software.

Redbox - Dolby® Encoders & Decoders

RB-DEED8

Dolby® E Encoder 8 Channel, Digital Inputs (continued...)

Audio Specification For RB-DEED8	
Output Sample Rate:	48kHz
Input Sample Rates:	CH1/2 & CH3/4: 32-192kHz CH5/6 & CH7/8: 32-48kHz
Input & Output Impedance:	75Ω/110Ω selectable
Signal Level (un-terminated):	Unbalanced: 1Vp-p +/- 20% Balanced: 6.6Vp-p +/- 20%
Digital Audio I/O: Inputs:	8 input channels via 4 x BNCs or 25 way D-type socket (AES3)
Dolby Output 1:	1 x Dolby bitstream output via BNC
Dolby Output 2:	1 x Dolby bitstream output via BNC or 25 way D-type socket (AES3)
Digital Audio Connectors:	6 x BNC 1 x 25-way D-type socket
Metadata:	SMPTE-RDD06, 9 way D-type socket

Video Reference	
Reference Type:	Autodetect NTSC, PAL or Tri-level sync
Impedance:	75Ω
Connector:	1 x BNC
LTC Input:	1 x BNC (not used)
Operational Control	
Display:	Vacuum fluorescent display
System Navigation:	Rotary selector with integral push-switch
Additional Connections	
Ethernet Port:	10/100Mbps, RJ-45
Remote Input/Output Port:	25-way 'D'-type socket
Serial Port:	RS232, 9 way D-type
Power Supply:	Universal filtered IEC, continuously rated 85-264VAC @47-63Hz, fused
Fuse Rating:	Anti-surge fuse 1A 20 x 5mm
Physical Specifications	
Dimensions (Raw):	48cm (W) x 15.8cm (D*)x 4.2cm (H) 19" (W) x 6.2" (D*) x 1.7" (H) (1U)
Dimensions (Boxed):	59cm (W) x 27.5cm (D) x 11cm (H) 23.2" (W) x 10.8" (D) x 4.3" (H)
Weight:	Nett: 1.8kg Gross: 2.3kg Nett: 4.0lb Gross: 5.1lb

* Note that this product is deeper than standard Redboxes

Equipment Type	
RB-DEED8	Dolby® E encoder 8 channel, digital inputs
Accessories	
RB-RK3:	1U Rear panel rack kit for large Redboxes

RB-DEDD8

Dolby® E & Dolby Digital Decoder 8 Channel, Digital Outputs

RK3

The RB-DEDD8 is a Dolby® E & Dolby Digital Decoder. It decodes an incoming Dolby bitstream and transmits the decoded outputs on BNC or D-type connectors on the rear panel. The Dolby input is also looped through to allow connection to other Dolby receiver equipment.

Each digital output channel has independent level control, which can be adjusted from -24dB through to +24dB in 0.5dB steps. The digital inputs and outputs can be selected to be either balanced or unbalanced AES/EBU specification, and can be connected through either BNC or D-type connectors. These I/O connections are paralleled, allowing one type to be used per input or output.

The metadata output from the decoder is transmitted using RS-485 via the external 9-pin D-type socket on the rear panel and can also be viewed using the SCi software.

A video input is used to genlock the audio outputs and Dolby® E decoding to a known video reference. This input autodetects between NTSC, PAL or Tri-level sync.

8 general purposes inputs and 8 outputs are available on a rear panel 25 way D-type socket whose function can be programmed using the menu, e.g. alarm outputs for loss of input or decoder errors.

The unit is controlled locally through the front panel display but can be remote controlled via an ethernet or RS-232 serial connection using the Sonifex SCi software.

Audio Specification For RB-DEDD8	
Output Sample Rate:	48kHz
Input Sample Rates:	32-48kHz
Input & Output Impedance:	75Ω/110Ω selectable
Signal Level (Un-terminated):	Unbalanced: 1Vp-p ±20% Balanced: 6.6Vp-p ±20%
Digital Audio I/O: Dolby Input:	1 x Dolby bitstream input via BNC
Outputs:	8 x Digital audio output channels & 2 x Auxiliary audio output (stereo downmix) channels via 5 x BNCs or 25 way D-type socket
Dolby Output:	1 x Dolby bitstream output via BNC or D-type
Digital Audio Connectors:	7 x BNC 1 x 25 way D-type socket
Metadata:	SMPTE-RDD06, 9 way D-type socket

Video Reference	
Reference Type:	Autodetect NTSC, PAL or Tri-level sync
Impedance:	75Ω
Connector:	1 x BNC
LTC Output:	1 x BNC (not used)
Operational Control	
Display:	Vacuum fluorescent display
System Navigation:	Rotary selector with integral push-switch
Additional Connections	
Ethernet Port:	10/100Mbps, RJ-45
Remote Input/Output Port:	25-way 'D'-type socket
Serial Port:	RS232, 9 way D-type socket
Power Supply:	Universal filtered IEC, continuously rated 85-264VAC @47-63Hz, fused
Fuse Rating:	Anti-surge fuse 1A 20 x 5mm
Physical Specifications	
Dimensions (Raw):	48cm (W) x 15.8cm (D*)x 4.2cm (H) 19" (W) x 6.2" (D*) x 1.7" (H) (1U)
Dimensions (Boxed):	59cm (W) x 27.5cm (D) x 11cm (H) 23.2" (W) x 10.8" (D) x 4.3" (H)
Weight:	Nett: 1.8kg Gross: 2.3kg Nett: 4.0lb Gross: 5.1lb

* Note that this product is deeper than standard Redboxes

Equipment Type	
RB-DEDD8	Dolby® E & Dolby Digital decoder 8 channel, digital outputs
Accessories	
RB-RK3:	1U Rear panel rack kit for large Redboxes

Dolby® and the double-D symbol are registered trademarks of Dolby Laboratories

Redbox - Video Embedders & De-embedders

RB-VHEDD8

3G/HD/SD-SDI Dolby® E Encoder & Embedder

RK3

The RB-VHEDD8 is an SDI audio de-embedder and re-embedder with Dolby® E Encoding capabilities. Dolby E encodes up to 8 channels of audio into two channels of an AES digital audio stream which is then embedded onto any of the available groups within each of the two video output paths. The encoded Dolby E bitstream is also available via a dedicated output on the rear panel.

The audio inputs to the Encoder can be selected to come from the external digital audio inputs via the BNC or D-type connections on the rear panel or from embedded audio contained in the incoming SDI input. The outputs from the de-embedder can also be re-embedded into the video outputs, along with the encoder inputs.

The metadata used for the encoding process can be selected to come from either the external 9-pin D-type on the rear panel, from metadata embedded into the vertical blanking area of the video input (SMPTE 2020), or by settings stored internally.

The unit is controlled locally through the front panel display but can be remote controlled via an ethernet connection using the Sonifex SCi software. The embedding channel routing is controlled using these methods also.

It has a triple rate SDI receiver with automatic input rate detection and equalisation along with two re-clocked and individually buffered SDI outputs. It supports the full range of 3G, HD and SD standards from NTSC and PAL up to 1080p 60Hz.

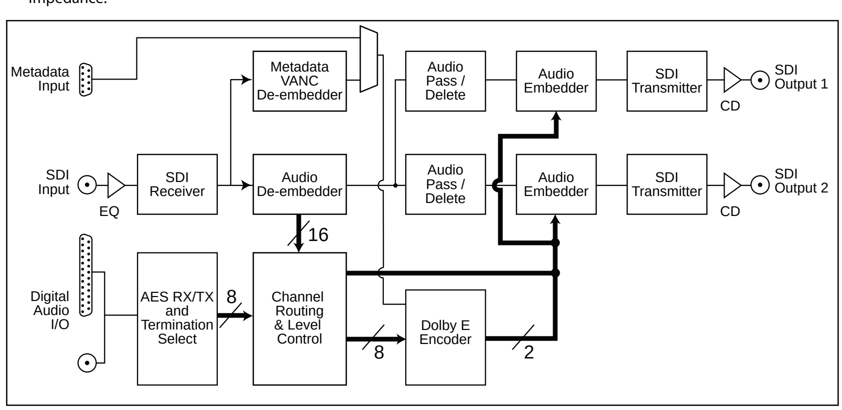
Each digital input channel has independent level control, which can be adjusted from -24dB through to +24dB in 0.5dB steps. The digital audio I/O connections are transformer-coupled balanced line interfaces and can be configured to be either 75Ω (AES 3ID) or 110Ω (AES 3) impedance through either a BNC or via the D-type connector. These connections are paralleled, allowing one type to be used per input or output.

8 general purposes inputs and 8 outputs are available on a rear panel 25 way D-type socket whose function can be programmed using the menu, e.g. alarm outputs for loss of input or encoder errors.

Audio Specification For RB-VHEDD8	
SDI Specification	
SDI Input:	1 x BNC, 3G/HD/SD-SDI
SDI Outputs:	2 x BNC, 3G/HD/SD-SDI, re-clocked
Impedance:	75Ω
Output Alignment Jitter:	<0.2UI (3G <0.3UI)
Output Level:	800mV ±10%
Return Loss:	<15dB at 1500MHz
SDI Supported Standards:	270Mbps, SMPTE-259M-C (SD) 1.485 or 1.4835Gbps, SMPTE-292M (HD) 2.97 or 2.967Gbps, SMPTE-424M (3G), SMPTE-425M-A
Supported Video Formats:	525/59.94 (SMPTE-125M) 625/50 (ITU-R BT.656) 720p/23.98, 24, 25, 29.97, 30, 50, 59.94, 60 (SMPTE-296M) 1035i/59.94, 60 (SMPTE-260M) 1080i/50, 59.94, 60 (SMPTE-274M) 1080p/23.98, 24, 25, 50, 59.94, 60 (SMPTE-274M) 1080pSF/23.98, 24, 25, 29.97, 30 (RP-211) 1080i/50 (SMPTE-295M) 1080p/50 (SMPTE-295M)
Embedded Audio:	24-bit, 48kHz synchronous SMPTE-272M-C SMPTE-299M
Metadata:	SMPTE-2020M SMPTE-RDD06, 9-Pin D-type socket
Audio Specifications	
Output Sample Rate:	48kHz
Input Sample Rates:	CH1/2 & CH3/4: 32-192kHz CH5/6 & CH7/8: 32-48kHz
Input & Output Impedance:	75Ω/110Ω selectable
Signal Level (Un-terminated):	Unbalanced: 1Vp-p ±20% Balanced: 6.6Vp-p ±20%
Digital Audio I/O: Inputs:	8 x digital audio input channels via 4 x BNCs or 25 way D-type socket (AES3)
Outputs:	2 output channels via 1 x BNC or 25 way D-type socket (AES3)
Digital Audio Connectors:	5 x BNC 1 x 25-way D-type socket
LTC Input:	1 x BNC (not used)
Operational Control	
Display:	Vacuum fluorescent display
System Navigation:	Rotary selector with integral push-switch
Additional Conections	
Ethernet Port:	10/100Mbps, RJ-45
Remote Input/Output Port:	25-way 'D'-type socket
Power Supply:	Universal filtered IEC, continuously rated 85-264VAC @47-63Hz, fused
Fuse Rating:	Anti-surge fuse 1A 20 x 5mm
Physical Specifications	
Dimensions (Raw):	48cm (W) x 15.8cm (D*)x 4.2cm (H) 19" (W) x 6.2 " (D*) x 1.7" (H) (1U)
Dimensions (Boxed):	59cm (W) x 27.5cm (D) x 11cm (H) 23.2" (W) x 10.8" (D) x 4.3" (H)
Weight:	Nett: 1.8kg Gross: 2.3kg Nett: 4.0lb Gross: 5.1lb

* Note that this product is deeper than standard Redboxes

Equipment Type	
RB-VHEDD8	3G/HD/SD-SDI Dolby® E encoder & embedder
Accessories	
RB-RK3:	1U Rear panel rack kit for large Redboxes



RB-VHEDD8 Block Diagram

VIDEO EMBEDDERS & DE-EMBEDDERS

Redbox - Video Embedders & De-embedders

RB-VHDDD8

3G/HD/SD-SDI Dolby® E & Dolby Digital Decoder & De-Embedder

The RB-VHDDD8 is an SDI audio de-embedder and re-embedder with Dolby E & Dolby Digital Decoding capabilities. It de-embeds and decodes a selected Dolby E or Dolby Digital bitstream embedded in the video input. The outputs from the decoder and the de-embedder can then be re-embedded onto either of the two SDI outputs and also transmitted on a BNC or D-type situated on the rear panel. The encoded Dolby bitstream is also available via a dedicated output on the rear panel.

The metadata output from the decoder is transmitted using RS-485 via the external 9-pin D-type on the rear panel and can also be embedded into the vertical blanking space (SMPTE-2020) onto either of the two SDI outputs.

The unit is controlled locally through the front panel display but can be remote controlled via an ethernet connection using the Sonifex SCi software, through which metadata can also be viewed.

It has a triple rate SDI receiver with automatic input rate detection and equalisation along with two re-clocked and individually buffered SDI outputs. It supports the full range of 3G, HD and SD standards from NTSC and PAL up to 1080p 60Hz.

Each digital output channel has independent level control which can be adjusted from -24dB through to +24dB in 0.5dB steps. The digital audio output connections are transformer-coupled balanced line interfaces and can be configured to be either 75Ω (AES 3ID) or 110Ω (AES 3) output impedance through either a BNC or via the D-type connector. These output connections are paralleled, allowing one type to be used per output.

8 general purposes inputs and 8 outputs are available on a rear panel 25 way D-type socket whose function can be programmed using the menu, e.g. alarm outputs for loss of input or decoder errors.

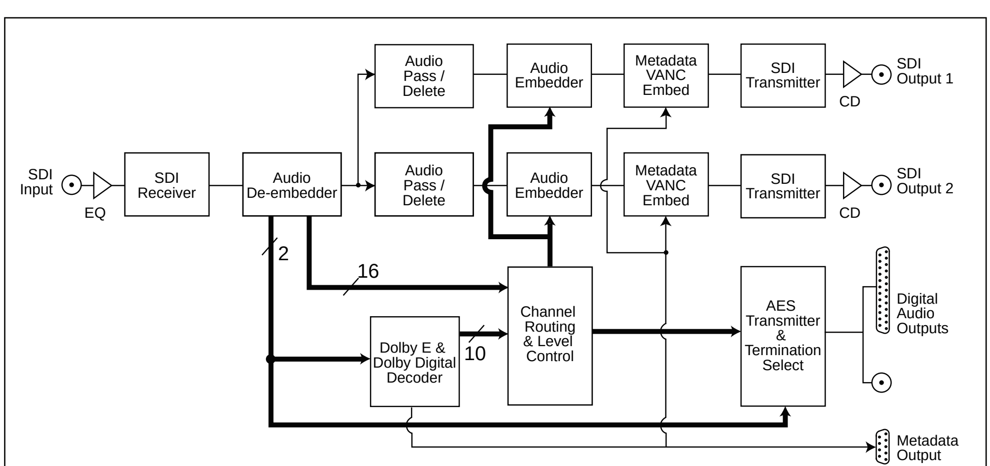
Audio Specification For RB-VHDDD8	
SDI Specification	
SDI Input:	1 x BNC, 3G/HD/SD-SDI
SDI Outputs:	2 x BNC, 3G/HD/SD-SDI, re-clocked
Impedance:	75Ω
Output Alignment Jitter:	<0.2UI (3G <0.3UI)
Output Level:	800mV ±10%
Return Loss:	<15dB at 1500MHz
SDI Supported Standards:	270Mbps, SMPTE-259M-C (SD) 1.485 or 1.4835Gbps, SMPTE-292M (HD) 2.97 or 2.967Gbps, SMPTE-424M (3G), SMPTE-425M-A
Supported Video Formats:	525/59.94 (SMPTE-125M) 625/50 (ITU-R BT.656) 720p/23.98, 24, 25, 29.97, 30, 50, 59.94, 60 (SMPTE-296M) 1035i/59.94, 60 (SMPTE-260M) 1080i/50, 59.94, 60 (SMPTE-274M) 1080p/23.98, 24, 25, 50, 59.94, 60 (SMPTE-274M) 1080pSF/23.98, 24, 25, 29.97, 30 (RP-211) 1080i/50 (SMPTE-295M) 1080p/50 (SMPTE-295M)
Embedded Audio:	24-bit, 48kHz synchronous SMPTE-272M-C SMPTE-299M
Metadata:	SMPTE-2020M SMPTE-RDD06, 9-Pin D-type
Audio Specifications	
Output Sample Rate:	48kHz
Output Impedance:	75Ω/110Ω selectable
Signal Level (Un-terminated):	Unbalanced: 1Vp-p +/- 20% Balanced: 6.4Vp-p +/- 20%
Digital Audio Outputs:	12 output channels via 6 x BNC or 25 way D-type socket (AES3)
Digital Audio Connectors:	6 x BNC 1x 25-way D-type socket
Operational Control	
Display:	Vacuum fluorescent display
System Navigation:	Rotary selector with integral push-switch
Additional Conections	
Ethernet Port:	10/100Mbps, RJ-45
Remote Input/Output Port:	25-way 'D'-type socket
Power Supply:	Universal filtered IEC, continuously rated 85-264VAC @47-63Hz, fused
Fuse Rating:	Anti-surge fuse 1A 20 x 5mm
Physical Specifications	
Dimensions (Raw):	48cm (W) x 15.8cm (D*)x 4.2cm (H) 19" (W) x 6.2" (D*) x 1.7" (H) (1U)
Dimensions (Boxed):	59cm (W) x 27.5cm (D) x 11cm (H) 23.2" (W) x 10.8" (D) x 4.3" (H)
Weight:	Nett: 1.8kg Gross: 2.3kg Nett: 4.0lb Gross: 5.1lb

* Note that this product is deeper than standard Redboxes

Equipment Type	
RB-VHDDD8	3G/HD/SD-SDI Dolby® E & Dolby Digital decoder & de-embedder

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

RB-VHDDD8 Block Diagram



Talkback & Communications

CM-TBU

Line Powered Telephone Balance Unit

The CM-TBU line-powered telephone balance unit is compatible with all analogue direct exchange lines and provides a 4-wire communications system to interface with the telephone network.

Front view of the CM-TBU

Rear view of the CM-TBU

The high degree of separation between send and receive signals makes it suitable for use in telephone IFB (interrupted foldback) applications and the high drive capacity at the 4-wire output enables a presenter's earpiece to be connected directly to the unit without an external amplifier.

This extremely compact unit is powered from the telephone line and provides an interface to a 4-wire circuit with separate level control of send and receive signals.

Optimum rejection of the input signal on the 4-wire output is achieved in a bridge circuit by adjusting three elements (NULL, R-BAL and C-BAL) via potentiometers which simulate the complex line impedance. This can be used to compensate for local line variations or to adapt to the telephone systems of other countries, where line characteristics may differ. Optimization of the sidetone rejection does not involve the use of any test equipment and can be easily carried out while the system is in use.

Although the signal level being sent to the line can be manually adjusted using the 'SEND LEVEL' control over a wide range, the level control is followed by a limiter that prevents the telephone line signal level becoming overloaded or distorted. The limiter drives a 'LIMIT' LED to indicate the onset of limiting.

Although the output stage can drive a presenter's earpiece in a telephone IFB application, the 'RECEIVE LEVEL' control may not be accessible to the presenter, who is normally situated some distance from the unit. The presenter may then require a local control of the earpiece signal level. The CM-TBU can be used to supply the correct signal level to a suitable battery powered earpiece belt-pack unit.

To enable communication between the 4-wire circuit and the telephone network, once the 4-wire and telephone line cable connections are made to the unit, the 'LINE CONNECT' switch can be pressed to power the unit from the line. This is indicated by the 'ON' LED, and can either be done after an outgoing call has been dialled on a telephone set connected to the unit, or to answer an incoming call after the 'RING' LED is seen to flash. Note that a telephone set is not required for incoming calls unless an audible ring is required. If the sidetone level at the 4-wire output is found to be excessive, the outgoing signal level can be reduced usingthe 'SEND LEVEL' control or the balance controls can be adjusted to minimize it.

The unit is supplied with a connector and cable kit that enables connections to be made to both UK Telecom or the universal RJ11 sockets used in most telephone networks around the world.

Features of the CM-TBU

- Isolated, full-duplex 4-wire interface to direct non-digital telephone exchange lines.
- Line powered, requiring no battery or external power.
- Simple optimization of sidetone rejection with any country's telephone system.
- LEDs indicating 'Ring', 'Line Hold' and 'Limit' conditions.
- Input level control with line-sensing limiter and limit indicator.
- High drive output with level control for direct feed to presenter's earpiece, etc.
- Loop-through RJ11 line sockets provide universal connection to line and telephone set.
- Connection to the telephone set is maintained while the unit is in use.
- Small, rugged extruded aluminium case with XLR3 male & female 4-wire connectors.

Technical Specification For CM-TBU	
4-Wire Input	
Input Impedance:	10kΩ, transformer coupled
Input Connector:	3 pin XLR female connector
Input Level Range:	-12dBU to +4dBU before limiting when connected to an average line
4-Wire Output	
Output Impedance:	150Ω, transformer coupled
Output Connector:	3 pin XLR male connector
Output Level Range:	-6dBU to +6dBU, for average line level
Sidetone Rejection:	30dB to 40dB average, depending on line characteristics
2-Wire	
Off Hook Voltage:	6V minimum
2 Wire Connectors:	RJ11 socket - line RJ11 socket - handset
Front Panel Operational Controls	
Line Connect:	Push button with indicator
Send Level:	Small rotary control
Receive Level:	Large rotary control
Null Balance:	Recessed preset potentiometer
R Balance:	Recessed preset potentiometer
C Balance:	Recessed preset potentiometer
Ring LED:	Indicates incoming ringing
Limit LED:	Indicates limiter active
On LED:	Indicates connection to the telephone line
Equipment Type	
CM-TBU	Line powered telephone balance unit
Physical Specification	
Dimensions:	77mm (W) x 83mm (D) x 42mm (H) 3.0" (W) x 3.3" (D) x 1.7" (H)
Weight:	Nett: 0.25kg Gross: 0.75kg Nett: 0.60lbs Gross: 1.7lbs

Talkback & Communications

CM-TLL

Line Powered Telephone Line Listen Unit

The CM-TLL provides a low-loss interface to receive audio from a telephone line. The line-powered unit is compatible with all analogue direct exchange lines and includes an LED indication of incoming calls. The unit is intended to be used with an earpiece amplifier to receive an audio feed by telephone from a studio.

The line and telephone set are connected to the unit via the RJ11 connectors in order to make an outgoing call. A telephone may not be required if calls are only incoming, because telephone line ringing is indicated by a flashing LED built into the unit. In either case, the 'LINE CONNECT' switch is pressed to make the line connection. The 'ON' LED indicates that the unit is powered and that audio is routed to the line via the XLR plug.

Front view of the CM-TLL

Rear view of the CM-TLL

Technical Specification For CM-TLL	
4-Wire Output	
Output Impedance:	150Ω, transformer coupled
Output Connector:	3 pin XLR male connector
Output Level:	Typically 0dBU for average line level
2-Wire	
Off Hook Voltage:	6V minimum
2 Wire Connectors:	RJ11 socket - line
	RJ11 socket - handset
Front Panel Operational Controls	
Line Connect:	Push button with indicator
Ring LED:	Indicates incoming ringing
On LED:	Indicates connection to the telephone line
Equipment Type	
CM-TLL	Line powered telephone line listen unit
Physical Specification	
Dimensions:	77mm (W) x 83mm (D) x 42mm (H) 3.0" (W) x 3.3" (D) x 1.7" (H)
Weight:	Nett: 0.20kg Gross: 0.70kg Nett: 0.50lbs Gross: 1.6lbs

Features of the CM-TLL

- Audio interface to direct analogue exchange lines - receive calls with electrical isolation from the line.
- Line powered, requiring no battery or external power.
- LEDs indicating 'Ring' and 'On Hold' conditions.
- Loop-through line sockets provide in-line connection with existing telephone.
- Existing telephone remains connected when the unit is in use.
- Line connections to British or International sockets via supplied cable kit.
- Connection to the telephone set is maintained while the unit is in use.
- Small, rugged extruded aluminium case with XLR3 male 4-wire connector.

Studio Illuminated Signs

signal LED

Changes to SignalLED signs sold after 1st December 2010

New Remote Programming Controller

New Programming Method

New Remote Programming Controller

The signs are preconfigured to display certain colours when the control inputs are active. Information on the colours shown can be found on the Sonifex website. Prior to 1st December 2010, the display mode and colour options could then be varied using an 8 way DIPswitch, accessible during installation. From 1st December 2010, this is different - the signs are still preconfigured, but to vary the colours and switching modes, the LD-RPC Remote Programming Controller must be used. This allows a much simpler way of programming any sign using a simple hand-held remote control. Please note that the control is sold separately and one control can program multiple signs.

Radio Talkshow System

Phone In 6

Radio Talkshow Telephone Switching System

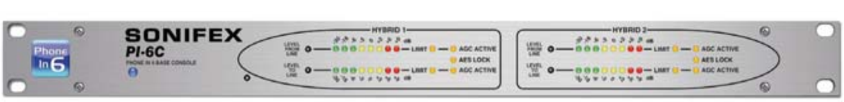
PI-6R Remote Control Front Panel

Talkshows have never been so easy

The Phone In 6 is a telephone switching system for radio talkshows. It consists of a desktop Remote Control Panel for call control, and a rackmount Base Console which contains all of the audio and telephony connections, including two superb quality digital telephone hybrids. Connected via ethernet, the units make up a simple, easy to use talkshow system.

The Base Console can be loaded with 2, 4, or 6 phone line modules, where each module can have 2 PSTN (normal telephone) lines or 1 ISDN BRI line. The ISDN basic rate interface allows 2 calls to be handled, 1 on each B channel. The PSTN module has 2 interfaces each using a modern digital hybrid interface, which is settable by software to handle a variety of PSTN and PABX systems with varying impedances, call connection & disconnection tones.

The Phone In 6 uses an echo-cancellation DSP algorithm and impedance matching to give around 70dB cancellation, which provides excellent separation of caller & line send audio and elimination of feedback, distortion & echo on the incoming calls. This is close to being the best performance possible on a telephone line & uses the same enhanced echo cancellation algorithms as used on the DHY-03, the best performing telephone hybrid in the world.



PI-6C Base Console Front & Rear

Features List

- Simple installation using CAT5 cabling for the remote control panel.
- Automatic call answering.
- Automatic call disconnection.
- Self-op or call screening modes.
- Caller conferencing.
- Headset included with system.
- Simple switch button control of callers.
- Receive, route, drop or make calls.
- 12 digit keypad to make calls, recall favourite settings and for speed-dialling.
- Two line LCD display.
- Phonebook for automatic dialling.
- Responsive illuminated switch buttons.
- Programmable GPIO.
- Ethernet control/connectivity.
- Two superb quality digital telephone hybrids.
- Modular PSTN & ISDN telephony inputs.
- Hardware metering of send & receive levels.
- Send & receive level adjustment.
- Wide range of impedance matching options.
- Music on hold input.
- AGC, automatic gain control of send & receive signals.
- Echo cancellation.
- Noise gate.
- Automatic caller ducking.
- Option for AES/EBU inputs & outputs.

Radio Talkshow System

Phone In 6

Radio Talkshow Telephone Switching System (continued...)

Easy to Use Menu System

The system can be set up to operate in the manner required using the menu system on the Remote Control Panel. Some of the settings are specific to the install, e.g. country type for the hybrid settings, ring cadences & disconnect tones supplied by the network provider. Other settings can be program specific, e.g. call screening/ self-op mode. The latter settings can change with the presenter, or as a particular show changes, so multiple sets of these parameters can be saved into permanent memory within the system which you can easily store and change by using the * key and a single digit number to update those parameters. Use the keyboard to direct dial a call using DTMF dialling, use the hash (#) key to use the phonebook entries or ## to repeat the last used number.

Conferencing Calls

Normally 2 calls can be handled independently, each with their own caller output and cleanfeed input. A conference mode is available where up to 2 calls can be mixed within the unit which are presented as a single audio stream on the output with a single cleanfeed input. In conference mode the Phone In 6 mixes the other caller audio into the sent signal so that each caller can hear the station output and the other caller in the conference call.

Call Handling - Self-op & Call Screening Modes

The Phone In 6 provides two different ways of operating: 'Self-op' where the show is entirely controlled by the presenter or using 'Call-screening' where an assistant or producer deals with incoming calls manually and then places the calls to the presenter via the on-air buttons.

For more information, check www.sonifex.co.uk

PI-6R Remote Control Panel Rear

Technical Specification Phone In 6 System

Audio Inputs	
Input Impedance – Line Mode (Mix-Minus Audio to Caller):	>10kΩ balanced 0dB, optimum working input
Input Level Range:	Adjustable 0 to +12dBu
ADC Signal to Noise:	Better than –89dbFS (RMS A-weighted at 24bit)
ADC Dynamic Range:	>96dB
ADC Distortion & Noise:	>87dB THD + N at 1kHz
ADC Frequency Response:	20Hz to 3.8kHz
Optional Digital Audio:	AES/EBU 110 Ω balanced inputs (IEC60968)
Sample Rates:	32kHz to 96kHz
0dBFS Reference Level:	12dBu or 18dBu
Audio Outputs	
Output Impedance (Received Audio From Caller):	<50Ω balanced floating 0dB, optimum working input
Output Level Range:	Adjustable -6 to +6dBu
DAC Signal to Noise:	Better than –85dbFS (RMS A-weighted at 24bit)
DAC Dynamic Range:	>97dB
DAC Distortion & Noise:	>83dB THD + N at 1kHz
DAC Frequency Response:	20Hz to 3.8kHz
Optional Digital Audio:	AES/EBU 110Ω balanced outputs (IEC60968)
Sample Rates:	32kHz to 96kHz
0dBFS Reference Level:	12dBu or 18dBu

ISDN Telephone Connection	
ISDN Interface:	S0 (BRI) / I.430
D Channel Protocol:	DSS1, National 1, 5ESS, JATE (INS64), AUSTEL, X.31, VN 4, TPH 1962
B Channel Protocol:	G.711
Regulatory Approval:	CE
PSTN Telephone Connection	
Send to Line Limiting Input:	+4dBu
Bandwidth to Telephone Line:	125Hz – 3.8kHz, -3dB ref 1kHz
Telephone Line Impedance:	Nominally 600Ω - set complex impedances via country code
Telephone Line Impedance Range:	300Ω to 1500Ω
Telephone Rejection:	78dB on 1kHz tone, typically 75dB on complex waveforms, reference peak level of 0dB
Ring Detector Sensitivity:	1 ring to 8 rings
Connections (PI-6C Base Console)	
Music On Hold Input:	3 pin XLR socket, balanced
Hybrid 1 & 2 Inputs:	2 x 3 pin XLR socket, balanced
Hybrid 1 & 2 Outputs:	2 x 3 pin XLR plug, balanced
Link to RCP:	RJ45 socket, link using CAT5 cable
Ethernet Port:	RJ45 socket
RS232 Serial Comms Port:	9-way 'D'-type socket
GPOI/O Remote I/O Port:	9-way 'D'-type socket
Mains Input:	Filtered IEC, continuously rated 85-264V AC @ 47-63Hz, fused 1A, max 10W

Connections (PI-6R Remote Control Panel)		
Local Headset:	2 x 3.5mm jack socket (mic input & headphones output)	
Link to BC:	RJ45 socket, link using CAT5 cable	
Ethernet Port:	RJ45 socket	
Mains Input:	Filtered IEC, continuously rated 85-264V AC @ 47-63Hz, fused 1A, max 10W	
Physical Specifications		
PI-6C Base Console		
Dimensions (Boxed):	60cm (W) x 34cm (D) x 7cm (H) 23.6" (W) x 13.4" (D) x 2.8" (H)	
Weight:	Nett: 2.2kg Nett: 4.8lbs	Gross: 3.2kg Gross: 7.0lbs
PI-6R Remote Control Panel		
Dimensions (Boxed):	32cm (W) x 29cm (D) x 15cm (H) 12.6" (W) x 11.4" (D) x 5.9" (H)	
Weight	Nett: 1.8kg Nett: 4.0lbs	Gross: 2.9kg Gross: 6.4lbs
Equipment Type		
PI-6PSTN4	Phone In 6 system, 1 x PI-6C, 1 x PI-6R & 2 x PI-PSTN2	
PI-6PSTN6	Phone In 6 system, 1 x PI-6C, 1 x PI-6R & 3 x PI-PSTN2	
PI-6ISDN6	Phone In 6 system, 1 x PI-6C, 1 x PI-6R & 3 x PI-ISDN2	
PI-6C	Phone In 6 Base Console	
PI-6R	Phone In 6 Remote Control Panel	
PI-PSTN2	Phone In 6 PSTN Card	
PI-ISDN2	Phone In 6 ISDN Card	

Pro Audio Streamers

Pro AudioStreamers

IP Streamers

The Pro Audio Streamers are a range of three IP to audio and audio to IP streamers which have professional analogue and digital inputs and outputs. They allow audio to be streamed around a building, wan or lan using IP audio and CAT5 cabling infrastructure. Typical applications include:

- As a backup STL (studio to transmitter link).
- For audio confidence monitoring in remote locations, such as at a radio transmitter site.
- For distribution of audio and music around a building, such as for passing audio to speakers in a conference room.
- As an internet-based IP music distribution system.
- As a tannoy, paging or IP based public address system.
- For in-house audio applications and distribution.
- For streaming internet radio in bars and clubs.
- For radio and music channels in hotel rooms distributed via an IP network.
- For playing audio from a PC, a jukebox application, or from a USB stick.

There are three products in the range providing encoding/decoding and streaming:
The **PS-SEND** converts an audio input to an IP stream.
The **PS-PLAY** reads an IP stream and outputs to balanced and unbalanced audio line levels.
The **PS-AMP** reads an IP stream and outputs audio to stereo speakers.

PS-SENDS Audio to IP Streaming Encoder

PS-PLAYS IP to Audio Streaming Decoder

PS-AMPS IP to Speakers Streaming Decoder

Pro Audio Streamers

PS-SEND Audio to IP Streaming Encoder

The PS-SEND is a freestanding audio to IP converter which is also available in a 1U rack-mount as the PS-SENDS.

It receives audio from a number of user selectable external stereo sources including balanced and unbalanced analogue audio, AES/EBU, S/PDIF & TOSlink digital audio. Once an audio source is selected, the unit encodes the audio in real time and sends it to the network as an encoded stream. The audio stream can be distributed over an IP-based network to one or more PS-PLAY or PS-AMP units or other proprietary servers such as those for Icecast or Shoutcast.

All the configuration settings for the unit are accessed via a local web-server built into it. The type of encoding and the transport mechanism are defined by selecting the connection from a pre-defined list. The PS-SEND encodes an audio source into an MP3 (from analogue or digital inputs), G.711 or PCM (from analogue inputs only) audio stream using HTTP, RTP, raw UDP or raw TCP protocols, including multicast support and the following encoder types are available: Mpeg1 & 2 Layer3, PM3 VBR (variable bit-rate), PCM linear and A-law, U-law, with 8kHz-48kHz sample rates . The unit can configure its own IP address using DHCP/BOOTP, IPzator or AutoIP. A readout of the set IP address can be heard on every reset using SONICIP technology, if selected. Remote level monitoring is also possible using SNMP traps.

Two red and green front panel LEDs indicate what state the unit is currently in, be it normal operational mode or bootstrap mode, and also indicates the current network connection status. A blue LED denotes power to the unit. The input being routed to the IP stream can be monitored on the front panel ¼" (6.35mm) stereo jack socket in combination with a headphone volume knob.

The rear panel has 2 x RJ45 connectors, one for the 10/100Mbit Ethernet interface and one for GPI connections. The PS-SEND has 6 x GPIs which can be used to trigger the sending of the audio stream and which can also be used to trigger remote events using an output relay on the PS-PLAY and PS-AMP. There is a 9 way D-type RS232 serial connection for control of the unit by automation systems and firmware updates. The unit can be remote controlled via serial connection, TCP or UDP.

Power to the unit is via a universal supply 85V - 264V fused IEC mains socket.

PS-SEND Audio to IP Streaming Encoder

Technical Specification for PS-SEND	
Analogue Inputs:	2 x XLR 3 pin (balanced)(L&R) 2 x RCA phono (unbalanced)(L&R)
Analogue Max Input Level:	18dBu XLR balanced 8dBu RCA phono unbalanced
Input Impedance:	20kΩ bridging (analogue balanced)
Input Impedance:	20kΩ (analogue unbalanced)
Analogue Input SNR:	74dB
Input THD:	0.02% Relative
Interchannel Isolation (Cross Talk):	80dB (Ref FSD)
Digital Inputs:	1 x AES/EBU XLR 3 pin female 1 x S/PDIF RCA phono 1 x TOSLink optical input
Analogue Outputs:	1 x 6.35mm (¼") jack headphone socket
Headphones Output:	Drives 150mW into 32Ω to 600Ω stereo headphones
GPIs (General Purpose Inputs):	6 x GPIs, selectable via webpage control on RJ45 socket
Serial Port:	1 x 9 way D-type socket, used to send control commands and update firmware
Ethernet Port:	1 x RJ45 socket. Remote control commands can be sent via TCP or UDP as well as firmware updates
Mains Input:	Filtered IEC, 85 - 264VAC, 47 - 63 Hz, 10W, max
Fuse Rating:	Anti-surge fuse 1A 20 x 5mm
Audio Codec Specifications PS-SEND	
G.711 (U Law/A Law 8kHz to 48kHz sampling rate)	
WAV (IMA ADPCM+ 16bit PCM uncompressed: 8kHz to 48kHz)	
MP3 Layer 1 (32, 44.1 and 48 kHz, CBR +VBR +ABR)	
MP3 Layer 2 (16, 22.05 and 24 kHz, CBR +VBR +ABR)	
General Features	
Supported network transport protocols	
RTP - UDP	
HTTP - TCP	
SIP	
Raw UDP	
Raw TCP	
Can also act as Icecast/Shoutcast source	
SNMP - traps for remote management	
DHCP, BOOTP, IPZator or AUTOIP - Dynamic IP address resolution	
SonicIP IP Address readout	
Physical Specification	
PS-SEND	
Dimensions (Raw):	22cm (W) x 13.7cm (D) x 4.3cm (H) 8.67" (W) x 5.39" (D) x 1.7" (H)
Dimensions (Boxed):	34cm (W) x 27cm (D) x 6cm (H) 13.4" (W) x 10.6" (D) x 2.4" (H)
Weight:	Nett: 0.84kg Gross: 1.64kg Nett: 1.85lbs Gross: 3.6lbs
PS-SENDS	
Dimensions (Raw):	48.3cm (W) x 13.7cm (D) x 4.3cm (H) 19" (W) x 5.39" (D) x 1.7" (H)
Dimensions (Boxed):	58.8cm (W) x 27cm (D) x 6.8cm (H) 23" (W) x 10.6" (D) x 2.7" (H)
Weight:	Nett: 1.34kg Gross: 2.14kg Nett: 2.95lbs Gross: 4.7lbs

PRO AUDIO STREAMERS

Pro Audio Streamers

PS-PLAY IP to Audio Streaming Decoder

The PS-PLAY is a freestanding IP to audio converter which is also available in a 1U rack-mount as the PS-PLAYS.

PS-PLAY IP to Audio Streaming Decoder

It takes an IP audio feed and converts it to one of a number of different stereo outputs: balanced and unbalanced analogue audio, AES/EBU, S/PDIF & TOSlink digital audio outputs.

As for the PS-SEND, all the configuration settings for the unit are accessed via a local web-server built into it. The unit can decode one of a number of audio streams, such as those generated by the PS-SEND (MP3, G.711 and PCM) and including AAC+V2, Ogg Vorbis and WMA audio files from external USB as well as from sources such as Shoutcast, Icecast (Internet radio), VLC and from RTP servers. The unit can receive streams from HTTP (TCP/IP) and RTP (UDP) protocols, as well as raw TCP and UDP packets. The unit can also configure its own IP address using DHCP/BOOTP, IPzator or AutoIP. A readout of the set IP address can be heard on every reset using SONICIP technology, if selected.

Two red and green front panel LEDs indicate what state the unit is currently in, be it normal operational mode or bootstrap mode, and also indicates the current network connection status. A blue LED denotes power to the unit. The audio output can be monitored on the front panel ¼" (6.35mm) stereo jack socket in combination with a headphone volume knob. The analogue audio outputs can be switched to be either a fixed level output or to be controlled by the front panel volume knob. The volume can also be adjusted, as well as many other features, using an infra-red remote control (available separately).

The PS-PLAY can be configured with up to 3 sources. The sources are prioritized in number order. If one has failed, the next one will attempt to play. If all fail, an external USB drive will be used as a back-up source. The external USB plugs into the USB socket on the front of the unit. This enables the PS-PLAY to act as a USB audio player, playing any of the audio formats previously mentioned.

The rear panel has 2 x RJ45 connectors, one for the 10/100Mbit Ethernet interface and one for GPIO connections. The PS-PLAY has 2 output relay contacts which can triggered remotely over IP from a connected PS-SEND unit. There is a 9 way D-type RS232 serial connection for control of the unit by automation systems and for firmware updates. The unit can be remote controlled via serial connection, TCP or UDP and remote management of the unit is also possible using SNMP traps.

Power to the unit is via a universal supply 85V - 264V fused IEC mains socket.

Technical Specification for PS-PLAY

Analogue Outputs:	2 x XLR 3 pin (balanced)(L&R) 2 x RCA phono (unbalanced)(L&R) 1 x ¼ inch (6.35mm) stereo jack headphone socket
Analogue Max Output Level:	18dBu XLR balanced 8dBu RCA phono unbalanced
Input Impedance:	<50Ω (analogue balanced)
Input Impedance:	<75Ω (analogue unbalanced)
Output SNR:	94dB
Output THD:	0.03% Relative
Interchannel Isolation (Cross Talk):	80dB (Ref FSD)
Analogue Output Gain Range:	-60dB to 18dB via front panel control knob, or optional IR controller
Digital Outputs:	1 x AES/EBU XLR 3 pin female 1 x S/PDIF RCA phono 1 x TOSLink optical input
Headphones Output:	Drives 150mW into 32Ω to 600Ω stereo headphones
USB Port:	1 x USB A socket
GPOs (General Purpose Outputs):	2 x switchable relay contacts (simultaneously switched) controlled from PS-SEND
Serial Port:	1 x 9 way D-type socket, used to send control commands and update firmware
Ethernet Port:	1 x RJ45 socket. Remote control commands can be sent via TCP or UDP as well as firmware updates.
IR Remote Receiver:	Remote commands can be sent using optional remote control via built in IR sensor
Mains Input:	Filtered IEC, 85 - 264VAC, 47 - 63 Hz, 10W, max
Fuse Rating:	Anti-surge fuse 1A 20 x 5mm

Audio Codec Specifications PS-PLAY

- G.711 (U Law/A Law 8kHz to 48kHz sampling rate)
- WAV (IMA ADPCM+ 16bit PCM uncompressed: 8kHz to 48kHz)
- MP3 Layer 1 (32, 44.1 and 48kHz, CBR +VBR +ABR)
- MP3 Layer 2 (16, 22.05 and 24kHz, CBR +VBR +ABR)
- AAC+ (HE-AAC v2 Level 3, incl.SBR and PS)
- Ogg Vorbis (floor 1)
- WMA 4.0/4.1/7/8/9 all profiles (5-384kbps)

General Features

Supported network transport protocols

- RTP - UDP
- HTTP - TCP
- SIP
- Raw UDP
- Raw TCP
- SNMP - traps for remote management
- DHCP, BOOTP, IPZator or AUTOIP - Dynamic IP address resolution
- SonicIP IP Address readout

Physical Specification

Dimensions (Raw):	22cm (W) x 13.7cm (D) x 4.3cm (H) 8.67" (W) x 5.39" (D) x 1.7" (H)	
Dimensions (Boxed):	34cm (W) x 27cm (D) x 6cm (H) 13.4" (W) x 10.6" (D) x 2.4" (H)	
Weight:	Nett: 0.84kg Nett: 1.85lbs	Gross: 1.64kg Gross: 3.6lbs

Pro Audio Streamers

PS-AMP IP to Speakers Streaming Decoder

The PS-AMP is a freestanding unit which converts an IP audio stream directly to speaker outputs.

PS-AMP IP to Speakers Streaming Decoder

It is also available in a 1U rack-mount as the PS-AMPS. The PS-AMP has the same feature-set as the PS-PLAY except that there are no individual audio outputs other than the speaker terminals. The PS-AMP uses an integrated 2 x 15W D-class amplifier to deliver audio directly to a pair of connected speakers.

For further information on features please refer to the PS-PLAY.

Technical Specification for PS-AMP	
Analogue Outputs:	2 x speaker connectors (2 each black and red terminals) 1 x ¼" (6.35mm) stereo jack headphone socket
Headphones Output:	Drives 150mW into 32Ω to 600Ω stereo headphones
Headphone Level Range:	-60dB – 18dB via front panel control knob or optional IR controller
USB Port:	1 x USB A socket
GPOs (General Purpose Outputs):	2 x switchable relay contacts (simultaneously switched) controlled from PS-SEND
Serial Port:	1 x 9 way D-type socket, used to send control commands and update firmware
Ethernet Port:	1 x RJ45 socket. Remote control commands can be sent via TCP or UDP as well as firmware updates.
IR Remote Receiver:	Remote commands can be sent using optional remote control via built in IR sensor
Speaker Power:	15W per channel into 8Ω @ 10% THD+N
Mains Input:	Filtered IEC, 85 - 264VAC, 47 - 63 Hz, 60W max
Fuse Rating:	Anti-surge fuse 1A 20 x 5mm
Audio Codec Specifications PS-AMP	
G.711 (U Law/A Law 8kHz to 48kHz sampling rate)	
WAV (IMA ADPCM+ 16bit PCM uncompressed: 8kHz to 48kHz)	
MP3 Layer 1 (32, 44.1 and 48kHz, CBR +VBR +ABR)	
MP3 Layer 2 (16, 22.05 and 24kHz, CBR +VBR +ABR)	
AAC+ (HE-AAC v2 Level 3, incl.SBR and PS)	
Ogg Vorbis (floor 1)	
WMA 4.0/4.1/7/8/9 all profiles (5-384kbps)	
General Features	
Supported network transport protocols	
RTP - UDP	
HTTP - TCP	
SIP	
Raw UDP	
Raw TCP	
SNMP - traps for remote management	
DHCP, BOOTP, IPZator or AUTOIP - Dynamic IP address resolution	
SonicIP IP Address readout	
Physical Specification	
Dimensions (Raw):	22cm (W) x 13.7cm (D) x 4.3cm (H) 8.67" (W) x 5.39" (D) x 1.7" (H)
Dimensions (Boxed):	34cm (W) x 27cm (D) x 6cm (H) 13.4" (W) x 10.6" (D) x 2.4" (H)
Weight:	Nett: 0.84kg Gross: 1.64kg Nett: 1.85lbs Gross: 3.6lbs

S0 Mixing Console

S0 Radio Broadcasting Mixer

A simple radio mixer for novice and professional users…

The S0 is a high quality yet simple to operate radio broadcast mixer ideally suited to community radio stations, for educational purposes and for internet radio. Easy to understand, the S0 includes a telephone hybrid for making and recording telephone calls and a 3.5mm stereo jack for plugging in an mp3 player.

S0 Radio Broadcast Mixer Iso View

The addition of a USB port allows for recording to a PC and for playing a PC automation system directly through the mixer. The headphone outputs have a built in limiter to offer hearing protection and the studio speakers mute when a microphone fader is open, with automatic mic live sign switching. The S0 allows presenters and DJs to be up and running quickly with a fully featured radio studio mixer.

The Sonifex S0 mixer is a compact, low cost, fixed format broadcast mixing console designed for on-air radio use. It uses the same high quality circuitry and components as the Sonifex S2 and S1 mixers to provide an audio experience second to none. The S0 can be fitted flush into a desk-top or can be rack mounted directly using the front panel mounting holes.

The uncomplicated and intuitive front panel layout ensures that the unit appeals to novices and broadcast professionals alike, whilst a range of user configurable options allows for flexible operation.

The console consists of nine input channels:

- 4 x mono balanced XLR mic/stereo unbalanced RCA phono line inputs.
- 4 x dual stereo unbalanced RCA phono line inputs.
- 1 x telephone balance unit (with line and handset ports)/stereo unbalanced RCA phono auxiliary input with a parallel 3.5mm stereo jack input on the front panel.

Providing in total 4 mic, 12 stereo line, 1 TBU & 1 stereo auxiliary inputs which you can switch between.

Input Channels

Input source buttons at the top of each channel strip are used to select the required mono or stereo source. The mixer has two main stereo buses, PGM (Program) and REC (Record), so each channel also has PGM and REC buttons to independently select which mixer bus the selected input is routed to.

The XLR microphone inputs on channels 1 to 4 have individually selectable +48V

S0 Radio Broadcast Mixer Rear.

S0 MIXING CONSOLE

S0 Mixing Console

phantom power and a gain calibration potentiometer providing up to 65dB of gain for the pre-amp. Input channel 1 also serves as the microphone input for a dedicated talkback channel.

The stereo RCA phono line inputs on channels 1 to 8 have 10dB of gain at the input to compensate for unbalanced consumer inputs.

Input channel 9 is a TBU and stereo auxiliary input channel. The TBU allows direct connection to a telephone line and allows calls to be made and received through the mixer using the handset connection. The auxiliary channel can switch between 2 independent inputs, one on the rear panel and one on the front panel.

Any channel which has the fader up is routed to the selected output, either PGM or REC or both.

Gain for each channel is trimmed by the front panel TRIM control providing ±15dB of gain. A PAN/BAL(ance) control is available to facilitate stereo imaging.

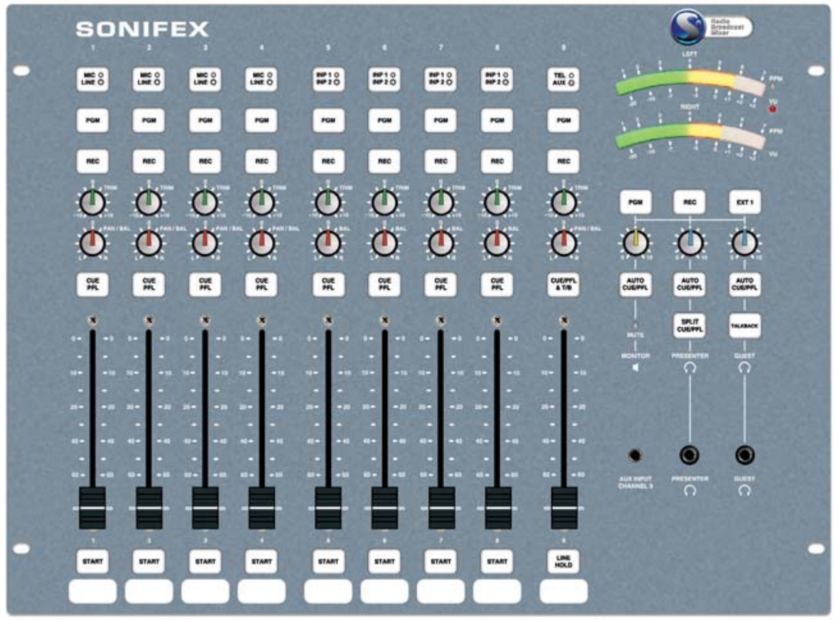
The use of VCAs controlled by the ALPS long throw 100mm faders gives a smooth, repeatable response and ensures tight stereo tracking while eliminating mechanical and electronic noise.

USB Audio for Playback and Recording
The S0 has the option to send and receive audio over USB. This allows the audio on the REC bus to be sent to a PC for recording or monitoring purposes. Also, the S0 can receive a USB audio stream from a PC and route it to the auxiliary inputs on channel 8. Alternatively, this signal can be routed to channels 5, 6 or 7 if required.

Output Channels

The S0 has separate stereo PGM and REC bus outputs. The PGM bus is output on balanced stereo XLRs and the REC bus is output on unbalanced stereo RCA phono connectors. There are monitor outputs for presenter headphones, guest headphones and loudspeakers.

Monitoring & Headphone Limiter
The monitor loudspeakers, presenter headphones and guest headphones are on 6.35mm stereo jack sockets and the headphones can be plugged in to the front and rear of the mixer. The monitor



S0 Radio Broadcast Mixer Top View

loudspeaker and headphone levels are independently variable between 0 (cut off) and 10 (max).

With the concerns over listening levels being too high in headphones, the addition of an adjustable limit level potentiometer on the rear panel of the mixer is a great idea which limits the maximum level of the audio routed to the presenter and guest headphone outputs. An illuminated MUTE LED shows when a live microphone channel has muted the loudspeakers and there is a MUTE contact output available to illuminate a 'MIC LIVE' light via a 6.35mm stereo jack socket on the rear panel.

A three way electronically interlocking illuminated switch bank selects the source routed to the loudspeaker and headphone outputs from either PGM, REC or from an additional unbalanced stereo RCA phono input EXT 1. This external input can be used for monitoring an off air signal or studio output.

Green illuminated AUTO CUE/PFL (pre fade listen) buttons adjacent to each level control allow the automatic monitoring of any channel that has been selected to pre-fade, either to the monitors or headphones.

For the presenter headphones, SPLIT CUE/PFL can be selected which places the

- User friendly broadcast mixer.
- Clear, simple layout with no jargon.
- Designed for school & community radio.
- Nine multi-function channel mixer.
- Built in telephone interface.
- Built in headphone volume limiter.
- Large, simple LED volume display.
- Remote output for fader starts.
- Speaker muting when 'Mics' are on.
- External mic-light switching output.
- 'Programme' and 'record' outputs.
- 'Aux' input for iPod or MP3 players.
- Four microphone/line channels.
- Four stereo line channels.
- Switchable telephone/AUX channel.
- Stereo USB audio to and from a PC.
- Guest headphone 'talkback'.
- Reliable, low cost mixing solution.
- Rack or flush mountable.
- Integrated AC power supply.

selected source in mono in the left ear, and pre-fade in mono in the right ear.

Metering
A pair of bright 21 segment LED meters can be configured to show either VU or PPM metering. The meters follow the selection of the presenter headphones including any pre-fade or split pre-fade function.

S0 Mixing Console

Talkback

A separate TALKBACK button is provided to allow the presenter to talk to a guest on the guest headphones. The S0 uses input channel 1 as the talkback source.

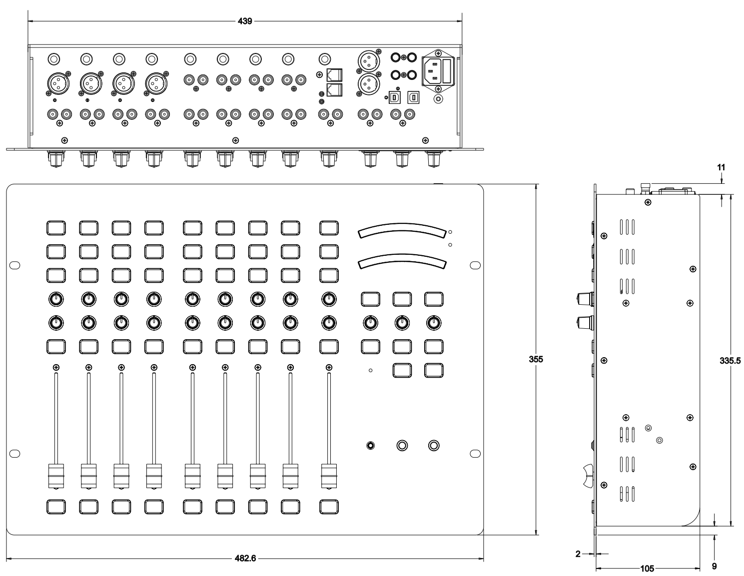
Channel Remotes

The remote outputs for each channel are highly configurable providing a comprehensive range of options to interface external equipment to the mixer. Each channel input has its own START and STOP remote controls which can be triggered when the channel is routed to the PGM or REC bus and the fader is opened or closed. The remotes can be set-up to be either pulsed or continuous latched outputs, and if in pulsed mode the START button has the capability to produce repeated start pulses.

Configuration Settings

The S0 has a range of software configurable options which can be used to customize the operation of the mixer. It is possible to enable or disable specific inputs, enable phantom power for the microphone channels and limit which buses each channel can be routed to. Other settings control auto cancelling of PRE FADE when the channel fader is opened and all the remote output configurations.

The S0 has an integral universal switch mode power supply, which uses an IEC mains inlet.



S0 Radio Broadcast Mixer Dimensions

S0 Radio Broadcast Mixer Side View

Technical Specification for S0

Input / Output Impedances	
Mic Input:	> 1k5Ω electronically balanced
Stereo Line Inputs:	> 20kΩ electronically unbalanced
PGM Output:	< 50Ω electronically balanced
REC Output:	< 75Ω unbalanced
Monitor Output:	< 75Ω unbalanced
Input / Output Gain Range	
Mic Input:	Preset pot +24dB to +67dB ref -50dBu, TRIM pot ± 15dB
Line Input:	+10dB ref 0dBu at PGM output, TRIM pot ± 15dB
Frequency Response	
Mic Input:	40Hz to 20kHz –1dB, +0dB
Line Input:	20Hz to 20kHz –0.5dB, +0dB
Noise (20Hz to 20kHz)	
Mic Input E.I.N.:	-130dB with 150Ω source.
Stereo Inputs	-92dBu ref 0dB (fader down, no routing)
Distortion	
Total Harmonic Distortion:	0.015% at 1kHz, 0.015% at 10kHz ref +8dBu

Range	
Pan Range:	Off/-3dB Centre/Off
Balance Range:	± 6dB
Common Mode Rejection Ratio	
Mic Input:	> 60dB typically
Output	
Maximum PGM Output:	+26dBu balanced
Maximum REC Output:	+16dBu unbalanced
Headphone Output Load:	> 16Ω, recommended 250Ω
Input & Output Connections	
Audio Inputs:	4 x Microphone XLR-3 pin sockets 12 x Pair stereo line RCA phono sockets 1 x Pair stereo aux RCA phono sockets 1 x Stereo aux 3.5mm jack socket
Audio Outputs:	
PGM:	2 x XLR-3 pin plug (balanced)
REC:	1 x Stereo pair RCA phono sockets
Monitor Inputs:	1 x Stereo pair RCA phono sockets
Monitor Outputs:	5 x 6.35mm (1/4") stereo jack sockets (2 x presenter, 2 x guest, 1 x loudspeaker)

Remote Outputs:	9 x 6.35mm (1/4") stereo jack sockets (one per channel) 1 x 6.35mm (1/4") stereo jack socket for light control
Telephone:	2 x RJ11 6/4 (1 x line, 1 x handset)
USB Audio:	1 x Type-B receptacle
USB Serial:	1 x Type-B receptacle
Mains Input:	Filtered IEC, continuously rated 85-264VAC, 47-63Hz, 45W nominal, >50W peak
Fuse Rating:	Anti-surge fuse 2A 20 x 5mm
Equipment Type	
S0:	S0 radio broadcast mixer
Physical Specification	
Dimensions (Raw):	48.3cm (W) x 35.6cm (D) x 12.5cm (H) 19" (W) x 14" (D) x 4.9" (H)
Dimensions (Boxed):	67cm (W) x 44cm (D) x 25cm (H) 26.4" (W) x 17.3" (D) x 9.84" (H)
Cut-Out Dimensions:	44cm (W) x 34.7cm (D) 17.32" (W) x 13.66" (D)
Weight:	Nett: 8.5kg Gross: 10.12kg Nett: 18.7lb Gross: 22.26lb

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