Soundcraft Vi5000™
Soundcraft Vi7000™

User Guide v1.8



Soundcraft
by HARMAN

IMPORTANT INFORMATION



IMPORTANT

Please read this manual carefully before using your mixer for the first time.

This equipment complies with the EMC directive 2014/30/EU
and LVD 2014/35/EU
This product is approved to safety standards
EN 62368-1:2014 + A11:2017
And EMC standards
EN55032:2015
EN55035:2017

Warning: This equipment is compliant with Class A of EN 55032. In a residential environment this equipment may cause radio interference. This product must not be used in residential environment.

Warning: Any modification or changes made to this device, unless explicitly approved by Harman, will invalidate the authorisation of this device. Operation of an unauthorised device is prohibited under Section 302 of the Communications act of 1934, as amended, and Subpart 1 of Part 2 of Chapter 47 of the Code of Federal Regulations.

NOTE: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy, and if it is not installed and used in accordance with the instruction manual, it may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference, in which case the user will be required to correct the interference at his own expense.

© Harman International Industries Ltd. 2020 All rights reserved
Parts of the design of this product may be protected by worldwide patents.
Part No. 5061286
Rev 1.8
E&OE March, 2023

Soundcraft is a trading division of Harman International Industries Ltd. Information in this manual is subject to change without notice and does not represent a commitment on the part of the vendor. Soundcraft shall not be liable for any loss or damage whatsoever arising from the use of information or any error contained in this manual. No part of this manual may be reproduced, stored in a retrieval system, or transmitted, in any form or by any means, electronic, electrical, mechanical, optical, chemical, including photocopying and recording, for any purpose without the express written permission of Soundcraft.

Harman Professional, Inc., 8500 Balboa Blvd., Northridge CA
91329, USA
Telephone +1-818-841-4600
email:
HProTechSupportEMEA@harman.com,
HProTechSupportUSA@harman.com

CONTENTS

1.0 INTRODUCTION
- 1.1: Saftey Notices & Warnings
- 1.2: Product Warranty

2.0: SPECIFICATIONS
- 2.1: Console block diagram

4.0: HARDWARE OVERVIEW
- 4.1: Local I/O
- 4.2: MADI / DANTE Sources / Switching

5.0 OPERATIONS OVERVIEW
- 5.1: Manual Conventions
- 5.2: Vistonics II
- 5.3: Console Bays
- 5.4: FaderGlow™
- 5.5: Buttons
- 5.6: Encoders
- 5.7: Gangs
- 5.8: Labelling

6.0: INPUTS
- 6.1: Navigation - Fader Pages
- 6.2: The Channel Strip
- 6.3: Encoder Mode
- 6.4: VST Control
- 6.5: Touchscreen / Signal Path
  - 6.5.1: Input Section
  - 6.5.2: EQ
  - 6.5.3: Dynamics
  - 6.5.4: Busses
  - 6.5.5: Output Section

7.0: OUTPUTS
- 7.1: LCR Mix Masters
- 7.2: Bus Configuration
- 7.3: Bus Master Control
  - 7.3.1: Master Vistonics Control
  - 7.3.2: Channel Strip
  - 7.3.3: Processing Chain
- 7.3.4: Graphic EQs

8.0: MATRIX SYSTEM

9.0: MUTE & VCA GROUPS
- 9.1: Mute Groups
- 9.2: VCA Groups
- 9.3: Aux VCA Groups

10.0: MONITORING
- 10.1: Console Controls
- 10.2: Setup
  - 10.2.1: MON Setup

11.0: SOLO SYSTEM
- 11.1: Primary Behaviour
- 11.2: Input Priority Mode
- 11.3: Autocancel Behaviour
- 11.4: Follow Output Solo Mode

12.0: METERING
- 12.1: Inputs
- 12.2: Busses

13.0: EDIT SYSTEM (COPY, PASTE, AND LIBRARY)
- 13.1: Copy, Paste, Undo
- 13.2: Library Basics
- 13.3: Library File Screen

14.0: SNAPSHOTS, CUELISTS, GLOBAL FILTER
- 14.1: Console Controls
- 14.2: Cuelist Control
- 14.3: Cue Details
  - 14.3.1: Cue Number & Timecode
  - 14.3.2: Snapshot/Cue Name
  - 14.3.3: MIDI
  - 14.3.4: GPIO
- 14.4: Snapshot Filters
  - 14.4.1: Snapshot Filter Scope
  - 14.4.2: Global Filter

15.0: TALKBACK & OSCILLATOR
- 15.1: Console Controls
- 15.2: Oscillator Setup & Use
- 15.3: Talkback Setup
- 15.4: Talkback Return Setup

16.0: MAIN MENU
- 16.1: Main (Screen, Security)
- 16.2: Shows
  - 16.2.1: Show File Data
- 16.3: GPIO
- 16.4: Sync
- 16.5: Tielines
- 16.6: FX
- 16.7: MIDI
- 16.8: Log
- 16.9: Settings
- 16.10: System

17.0: LEXICON FX
- 17.1: Using The FX Processors
  - 17.1.1: Tap Tempo Functionality
- 17.2: FX Algorithms & Parameters
  - 17.2.1: Reverbs
  - 17.2.1: Delays
  - 17.2.1: Misc

18.0: VM² WIreless Status Monitoring / HiQnet
- 18.1: HiQnet Network Configuration
  - 18.1.1: VM ² Device List
- 18.2: Status Displays
  - 18.2.1: Channel Strip Status
  - 18.2.2: VST Status

APPENDIX 01: Dante User Guide

APPENDIX 02: ViSi Connect Brochure



Welcome To Soundcraft Vi

Live sound mixing - the way it should be

Our patented Vistonics was designed from a simple start point - create the most intuitive digital mix interface available. Vistonics simplifies the process and puts the rotary encoders and switches right on the touch screen.

With an adaptive graphics and relevant controls combined in one area, the operator is immediately presented with a simple, clear mix system that streamlines workflow and enhances the creative process.

Each Vistonics interface controls eight input channels and comprises a touch screen with 16 rotary encoders and 16 switches. A simple touch in the desired part of the on-screen channel strip is all it takes to access channel functions including routing, input gain, digital gain trim, delay, high and low pass filters, 4-band fully parametric EQ, compressor, limiter, gate, de-esser and pan, with immediate access to a sophisticated visual status display and straightforward controls.

Fader layers - like you've never seen them before

In the heat of the moment, with the faders flying, fader layers can cause all kinds of headaches. Soundcraft® FaderGlow solves the problem associated with multiple fader layers. At a single glance, illuminated faders in colours that match those on the Vistonics™; display show immediately what function a fader is controlling.

When controlling inputs, no illumination is applied to the motorised faders. But when the Aux, Group or Matrix bus master is soloed with the Follow Solo function switched on, the fader becomes a 'contributor' for the soloed bus, lighting up in orange, green or blue accordingly.

A console focused on the operator

User-configurable fader layers allow an engineer to map out channels on any of 5 user layers so that a combination of different inputs can be placed on one layer.

Output fader layers can also be customised; for example, stereo aux masters can be placed alongside VCAs or matrix masters for faster access to priority outputs.

Every aspect of the control surface, mechanical and operational, has been engineered around the operator to provide an unimpeded workflow and a more creative mix.

The angle of the touch screens ensures that the engineer can always view display data clearly during the show. The brightness and contrast of the displays, and the illumination of the control surface itself, are designed to minimise strain on the eye. By optimising the density of faders and controls on the operating surface, the engineer is able to reach all critical areas of the console comfortably from a central point, without straining or leaning over.

SAFETY NOTICES

For your own safety and to avoid invalidation of the warranty please read this section carefully.

THIS UNIT MUST BE EARTHED

Under no circumstances should the mains earth be disconnected from the mains lead.



The wires in the mains lead are coloured in accordance with the following code:

Earth: Green and Yellow (Green/Yellow - US)
Neutral: Blue (White - US)
Live: Brown (Black - US)

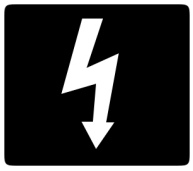
As the colours of the wires in the mains lead may not correspond with the coloured markings identifying the terminals in your plug, proceed as follows:

The wire which is coloured Green and Yellow must be connected to the terminal in the plug which is marked with the letter E or by the earth symbol.

The wire which is coloured Blue must be connected to the terminal in the plug which is marked with the letter N.

The wire which is coloured Brown must be connected to the terminal in the plug which is marked with the letter L.

Ensure that these colour codings are followed carefully in the event of the plug being changed.



The internal power supply unit contains no user serviceable parts. Refer all servicing to a qualified service engineer, through the appropriate Soundcraft dealer.

WARNINGS

• **Read** these instructions.
• **Keep** these instructions.
• **Heed** all warnings.
• **Follow** all instructions.
• **Clean** the apparatus only with a dry cloth.
• **Do not** install near any heat sources such as radiators, heat resistors, stoves, or other apparatus (including amplifiers) that produce heat.
• **Do not** block any ventilation openings. Install in accordance with the manufacturer's instructions.
• **Do not** use this apparatus near water.
• **Do not** defeat the safety purpose of the polarized or grounding type plug. A polarized plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding prong. The wide blade or the third prong are provided for your safety. When the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.
• **Protect** the power cord from being walked on or pinched particularly at plugs, convenience rececta-cles and the point where they exit from the apparatus.
• **Only** use attachments/accessories specified by the manufacturer.
• **Unplug** this apparatus during lightning storms or when unused for long periods of time.
• **Refer** all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way such as power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally or has been dropped.
• **Use** only with the cart, stand, tripod, bracket or table specified by the manufacturer, or sold with the apparatus. When the cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.
• **Rack mounting (for local rack and active breakout box)**: In case of rack mounting, use M6 x 16mm standard rack mount screws, with matching cage nuts (not supplied) for fixing the unit into the rack. In the USA, use standard rack mounting screws size 10-32 or 12-24. Use at least 4 screws on each side, equally spaced.
• **No** naked flame sources, such as lighted candles or cigarettes etc., should be placed on the apparatus.
• **Warning**: To reduce the risk of fire or electric shock, do not expose this apparatus to rain or moisture. Do not expose the apparatus to dripping or splashing and do not place objects filled with liquids, such as vases, on the apparatus.
• **Warning (The following statement applies to Local Racks shipped in the EU only.)**: In order to maintain compliance with EU Electromagnetic Compatibility requirements, the C-section metal plates covering the gaps between the DSP and I/O cards shall be kept in place and re-installed in the event of card replacement.
• **No user serviceable parts**. Refer all servicing to a qualified service engineer, through the appropri-ate Soundcraft dealer.
• **Ventilation** should not be impeded by covering the ventilation openings with items such as newspapers, table cloths, curtains etc.
• **It is recommended** that all maintenance and service on the product should be carried out by Sound-craft or its authorised agents. Soundcraft cannot accept any liability whatsoever for any loss or damage caused by service, maintenance or repair by unauthorised personnel.

WARRANTY

1 Soundcraft is a trading division of Harman International Industries Ltd.
End User means the person who first puts the equipment into regular operation.
Dealer means the person other than Soundcraft (if any) from whom the End User purchased the Equipment, provided such a person is authorised for this purpose by Soundcraft or its accredited Distributor.
Equipment means the equipment supplied with this manual.

2 If within the period of twelve months from the date of delivery of the Equipment to the End User it shall prove defective by reason only of faulty materials and/or workmanship to such an extent that the effectiveness and/or usability thereof is materially affected the Equipment or the defective component should be returned to the Dealer or to Soundcraft and subject to the following conditions the Dealer or Soundcraft will repair or replace the defective components. Any components replaced will become the property of Soundcraft.

3 Any Equipment or component returned will be at the risk of the End User whilst in transit (both to and from the Dealer or Soundcraft) and postage must be prepaid.

4 This warranty shall only be available if:
a) The Equipment has been properly installed in accordance with instructions contained in Soundcraft's manual.
b) The End User has notified Soundcraft or the Dealer within 14 days of the defect appearing; and
c) No persons other than authorised representatives of Soundcraft or the Dealer have effected any replacement of parts maintenance adjustments or repairs to the Equipment; and
d) The End User has used the Equipment only for such purposes as Soundcraft recommends, with only such operating supplies as meet Soundcraft's specifications and otherwise in all respects in accordance Soundcraft's recommendations.

5 Defects arising as a result of the following are not covered by this Warranty: faulty or negligent handling, chemical or electro-chemical or electrical influences, accidental damage, Acts of God, neglect, deficiency in electrical power, airconditioning or humidity control.

6. The benefit of this Warranty may not be assigned by the End User.

7. End Users who are consumers should note their rights under this Warranty are in addition to and do not affect any other rights to which they may be entitled against the seller of the Equipment.

SPECIFICATIONS

Frequency Response
Stagebox Mic input to Line output: +0/-1dB, 20Hz-20kHz
AES/EBU In to AES/EBU Out: +0/-0.2dB, 20Hz-20kHz

T.H.D. & NOISE
Stagebox Mic In (min gain) to Local Line Out:
22Hz-22kHz, <0.003% @ 1kHz
Stagebox Mic In (max gain) to Local Line Out:
22Hz-22kHz, <0.020% @ 1kHz
Local Line In to Line Out:
22Hz-22kHz, <0.003% @ 1kHz
Mic Input E.I.N (22Hz-22kHz bandwidth, unweighted):
<-126dBu (150Ω source)
Residual Noise, Stagebox line output; no inputs routed:
Mix fader @0dB: -91dBu
CMRR, Stagebox Mic input: 80dB @ 1kHz

Sampling Frequency
48kHz or 96kHz

Latency
Stagebox Mic Input to Local Line output:
<2ms @48kHz; <1ms @96kHz

AES/EBU Input Sample Rate
32–108kHz (with SRC enabled)

DSP resolution
40-bit floating point

Clocks
Internal clock accuracy: <+/-50ppm
Internal clock jitter: <+/-5ns
External Sync: BNC Wordclock

Input & Output Levels
Mic Inputs: +28dBu max
Line Inputs: +22dBu max
Line Outputs: +22dBu max
Nominal Operating Level: +4dBu (-18dBFS)

Input & Output Impedances
Mic Inputs: 10kΩ
All other analogue Inputs: >10kΩ
Line Outputs: <75Ω
AES/EBU Outputs: 110Ω

Oscillator
20Hz to 20kHz/Pink/White Noise, variable level

EQ (Inputs and bus Outputs)
HF: 20Hz-20kHz, +/-18dB, Q= 0.3-8.7 or shelving
Hi-Mid: 20Hz-20kHz, +/-18dB, Q=0.3-8.7
Lo-Mid: 20Hz-20kHz, +/-18dB, Q=0.3-8.7
LF: 20Hz-20kHz, +/-18dB, Q= 0.3-8.7 or shelving

Stagebox HP Filter: 80Hz fixed, 12dB per octave
Channel HP filter: 20Hz-600Hz, 18dB per octave
Channel LP filter: 1kHz-20kHz, 18dB per octave

Metering
Internal 20-segment LED bargraphs plus 9-segment gain reduction meters for all inputs and Outputs.
Peak hold: Variable from 0-2s.

Power / Operation
Mains Voltage operating range: 90-264V, 47-63Hz, autoranging
Mains Power Consumption: 300W
Operating Temperature Range: 0°C – 45°C
(32°F – 113°F)
Relative Humidity: 0% – 90%, non-condensing Ta=40°C
(104°F)
Storage Temperature Range: -20°C to 60°C
(-4°F – 140°F)

E & OE.
Soundcraft reserves the right to change specifications without notice.

HARDWARE

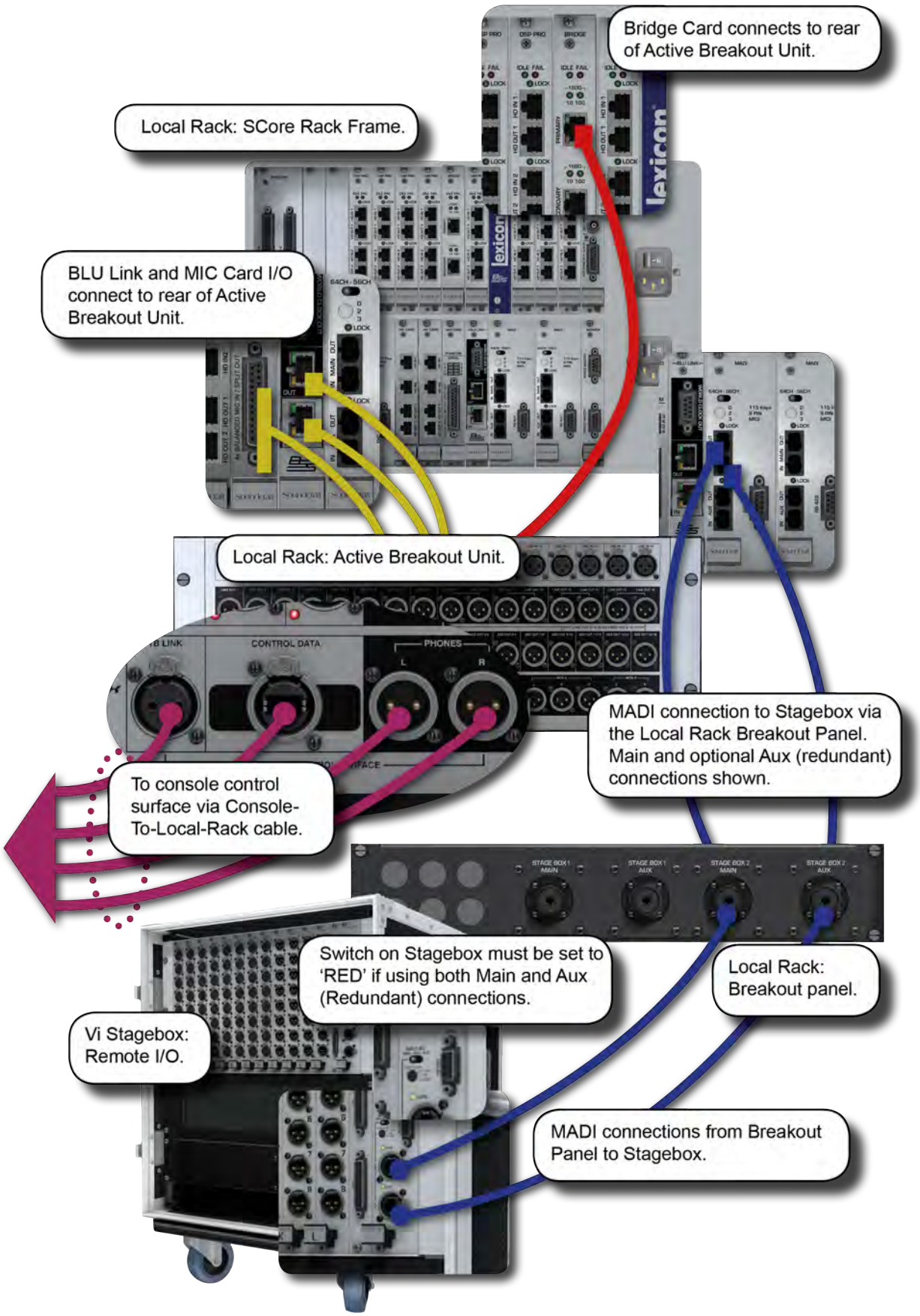
The Vi5000 and Vi7000 systems feature Local Rack and Stagebox inputs and outputs (I/O). The Local Rack houses the system DSP (audio processing), data connections, and some direct digital and analogue audio connections. It is designed to remain local to the control surface and connects via the supplied Console-To-Local-Rack cable.
The Stagebox connects to the Local rack via MADI and is designed to provide physical inputs and outputs on-stage. The system can connect to multiple Stageboxes via multiple MADI cards. Many different types of I/O are available for the system with the ViSi Connect series of optional interface cards designed to fit either the Local Rack or the Compact Stagebox (and in some cases, the Vi Stagebox).







HARDWARE OVERVIEW



HARDWARE > CONTROL SURFACE CONNECTIONS

The rear of the control surface has essential audio connections, data connection to the Local Rack, MIDI, USB, and HiQnet (Ethernet) connections, as well as the dual-redundant power supply connections and switches.





HARDWARE > CONTROL SURFACE CONNECTIONS



MIDI I/O

Musical Instrument Digital Interface

One MIDI input and two MIDI outputs (5-pin DIN), used mainly for triggers to and from the Snapshot / Cue system. See Section 14 for more detail.



HiQnet

Harman control protocol / standard 1GB Ethernet connection.

The console's VM² system allows status monitoring and (where compatible) control of wireless microphone systems. See section 18 for more detail. This is an XLR-Housed EtherCon connector. This port is also used for connecting a wireless access point for iPad remote control using the ViSi Remote app, and for connecting the Soundcraft Realtime Rack Plug-in engine for storage and recall of plug-in snapshots.



USB

Connection for USB memory devices and/or USB keyboard

USB memory sticks can be used to save and load console Snapshot, show, and library data. See sections 13 and 14 for more detail. A USB keyboard can also be connected here and will work in parallel with the on-screen keyboard.



TB Mic

Talkback microphone Input.

Local talkback microphone input - balanced XLR with phantom power available (switched next to connector). This connection is a parallel connection with the Talkback microphone connector in the TALKBACK/OSC section of the Master Bay section. See section 15 for more detail.



Local Rack Connections

Four connections to the system's Local Rack.

All connections must be made via the supplied Console-To-Local-Rack cable (part number 5058753).



Power

Primary and Redundant IEC power connections.

100V - 240V mains power connections. The primary power connector (PSU1) feeds power to the primary PSU. Power to the Redundant (PSU2) connector is optional and will provide backup switching should the primary supply fail.

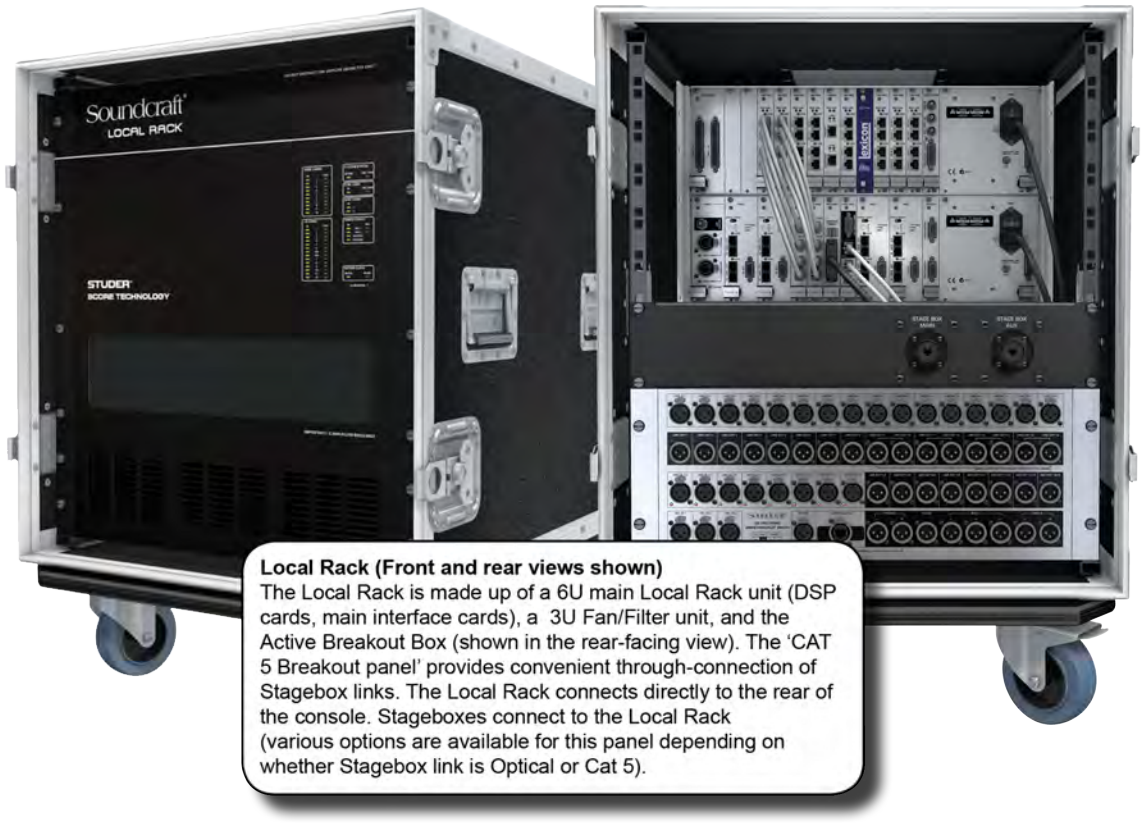
HARDWARE > LOCAL RACK

The rack consists of a 6U processing and I/O unit, developed by Studer, called the SCore. Below this is a 3U low-noise cooling fan unit. There is also an Active Breakout Unit, which provides direct audio connections for local use and for connection to the Control Surface, and a another Breakout Panel for convenient MADI connections for Stageboxes. The SCore itself consists of two sections: the upper 3U section houses the DSP mixing core, and the lower 3U section houses the local audio I/O and also the connections to the remote Stage Box.

The audio processing inside the SCore is independent of the control surface. This means that the audio will continue to pass even if the control surface is switched off or disconnected from the core.

How It Works

The control surface sends control data to the Bridge Card, which is a communication and processing hub situated in the centre of the top 3U section of the Local Rack. The Bridge Card interprets the control data and sends internal data to the five or eight DSP Pro cards and the FX Card. These cards actually process the audio. The audio input and output connections to the DSP cards are made via short CAT 5 patch cables, which connect the DSP cards with the I/O rack in the lower 3U section of the core.



Local Rack (Front and rear views shown)
The Local Rack is made up of a 6U main Local Rack unit (DSP cards, main interface cards), a 3U Fan/Filter unit, and the Active Breakout Box (shown in the rear-facing view). The 'CAT 5 Breakout panel' provides convenient through-connection of Stagebox links. The Local Rack connects directly to the rear of the console. Stageboxes connect to the Local Rack (various options are available for this panel depending on whether Stagebox link is Optical or Cat 5).

HARDWARE > LOCAL RACK > FANS/FILTERS

The Local Rack and Stagebox have integral fan units which are essential for reliable operation.
The fan status is monitored from the console GUI and this will provide warning if there is a problem with any of the fan motors.

The fan units have a manual fan speed adjustment facility which allows the fan speed to be reduced in environments where acoustic noise is an issue. Please note that reducing the speed from the factory-set value is ONLY allowable when the console is operated in an air-conditioned environment where the ambient temperature is guaranteed to be maintained at 20degC maximum.

To gain access to the adjustment trimmer, remove the small 2-part plastic blanking plug located to the left of the fan filter grille. Insert a small screwdriver into the hole to adjust: turn anticlockwise to reduce the speed.
Contact Soundcraft for advice before changing fan speed if unsure.



Ensure fan units on Local Rack and Stagebox are connected and operational.

CLEAN FILTERS REGULARLY!

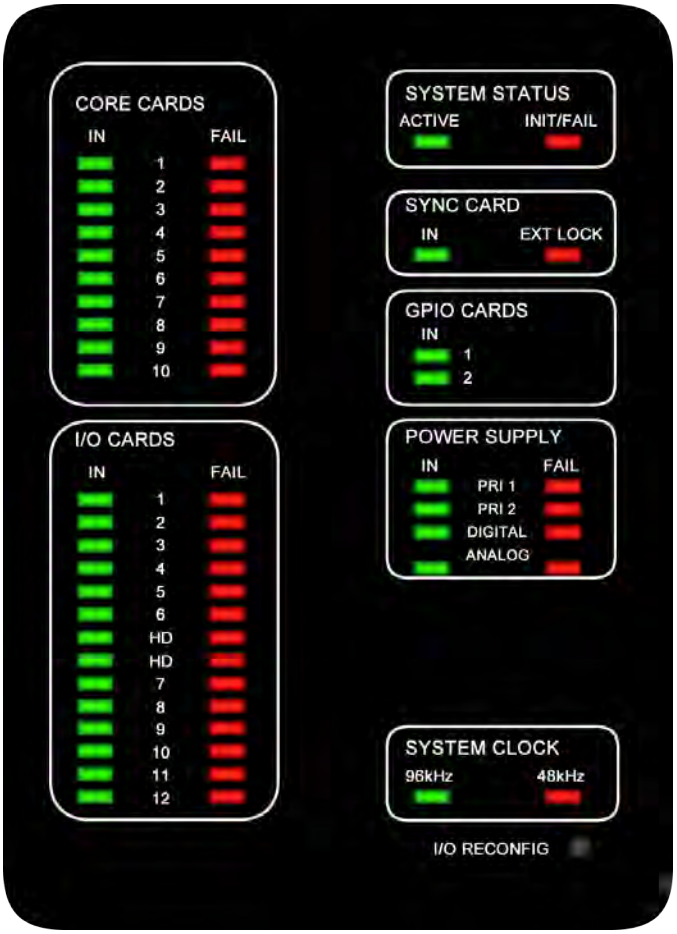


4.2.2: SCore FRONT PANEL

HARDWARE > LOCAL RACK > SCore FRONT PANEL

The status LEDs on the front panel of the SCore unit give an at-a-glance indication that the Local Rack is functioning correctly, or that an error condition has occurred. An illuminated green LED indicates the presence of a card, an illuminated red LED warns of a card error.

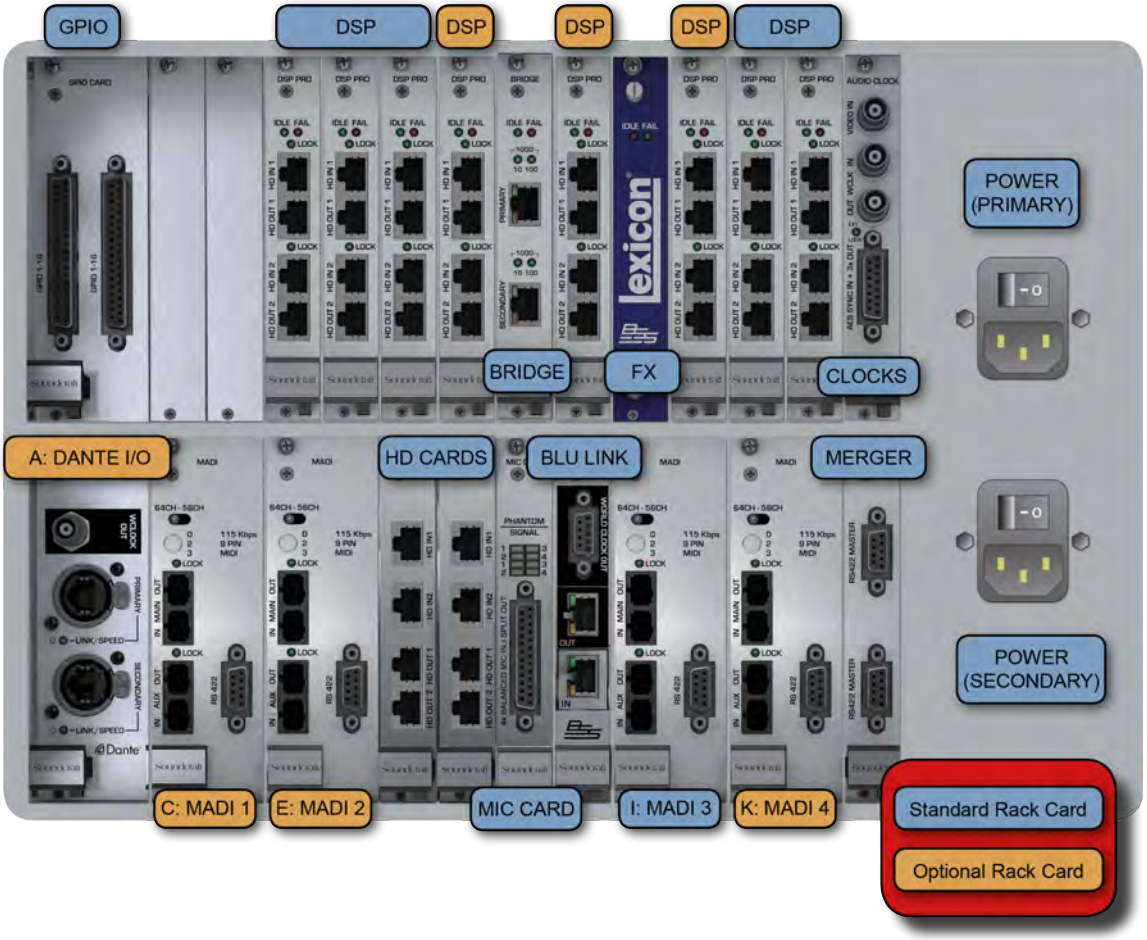
Note that there is a RECONFIG button behind a small hole at the bottom right of the panel. When pressed with a small screwdriver or similar a reset is applied to the lower row of cards in the local rack: this will force a re-polling of the loaded cards to reconfigure the console. This needs to be done only if the card configuration has been changed.



HARDWARE > LOCAL RACK > SCore CARD FRAME

The rear panel of the SCore shows the Score Card Frame. Your console's Card Frame will be populated according to your own specification. The diagram shows a 96kHz-capable system with four MADI cards installed. A 48kHz-only system has five DSP cards installed in the top section, with three additional 'blank' spaces (left of bridge card, left of FX card, right of FX card). 'Standard' cards are shown with a blue background, optional cards (including the three 96kHx-system DSP cards) are shown with an orange background.

There are a number of connections that must be made in the Score Rack Frame, and between the SCore Rack Frame and the Active Breakout Unit, in order for the system to operate correctly. These are shown in section 4.2.4.



There are 12 single slots for I/O cards, identified with letters A-L, the HD cards are not included in these 12 slots. Double-slot cards such as MADI and Dante take the letter of their left-hand single slot.

GPIO Card with Relay Contacts
Provides a set of GPIO contacts, controlled by the GPIO page of the control surface, The inputs are opto-isolated and the outputs are SPST relay contacts.

DSP Cards
Contains the DSP processing for audio processing in the SCore. A 96kHz-capable system will have eight DSP cards while a 48kHz-only system will have five DSP cards. The left-most DSP cards (1 & 2) must be connected to the HD cards 1 & 2: HD IN 1 > HD OUT 1; HD IN 2 > HD OUT 2 and so on.

Bridge Card
Provides the connection from the Local Rack to the Control Surface. Houses a QNX embedded processing system that controls all the audio processing in the rack. The Primary Bridge Card connection should be connected to the Primary Bridge Card connection on the rear of the Active Breakout Unit.

HARDWARE > LOCAL RACK > SCore CARD FRAME

FX Card
Provides the additional DSP and algorithms required to run the eight Lexicon digital FX processors.

Audio Clock Card
Provides clock connection to external equipment, both in and out. The wordclock input options are Video clock, standard wordclock or AES/EBU sync clock. Any of these can be used as a wordclock source for the console. The console permanently outputs standard wordclock and AES/EBU wordclock for sync purposes.

DANTE Card
DANTE audio networking I/O - normally used for recording. See the appendix 2 at the end of this manual for operation of the DANTE card.

MADI Cards
Up to 64 channels of MADI digital audio inputs and outputs, either via optical or CAT-5e-type connections, depending on the options fitted to the console. Each card has redundant (AUX) connections.
The left-hand (first two) MADI card slots are recording and/or additional connection options. The right-hand MADI cards are for connection to Vi Stageboxes. For convenience, these should be connected to the Stagebox breakout panel.

HD Cards
Communicate with the first two DSP Pro cards on the top row of cards. They pass digital audio along with control data and status information about the cards loaded into the frame. Each pair of links can handle up to 192 channels in both directions, giving a total I/O capacity for the rack of 384 channels. The HD card is responsible for working out what cards are loaded into the system and configuring it accordingly. The RECONFIG button on the front panel of the Local Rack will force a re-polling of the loaded cards to reconfigure the console.

MIC Card
Provides 4 x Mic input for Talkback and local Mic input. The card has 8 indicators, 4 indicators for signal present at the mic input and 4 indicators for the phantom power status. The D-Sub connector should be connected to the corresponding I/O MIC CARD S-Sub connector on the rear of the Active Breakout Unit.

BLU Link Card
The console uses the BLU Link digital audio networking protocol to connect local audio I/O to the Active Breakout Unit. The BLU Link input and output on the BLU Link card should be connected to the BLU Link output and input (respectively) on the rear of the Active Breakout Unit. Note that the BLU Link card that is connected to the Active Breakout Box is not intended to be connected to other Harman BLU Link audio devices. Other BLU link cards may be fitted in the local rack's configurable slots if connection to a BLU Link network or device is required.

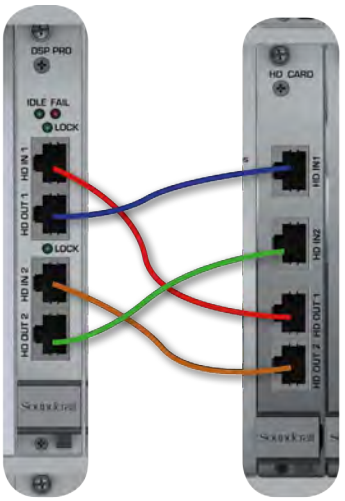
Merger Card
The D-type connectors on this card are not used. This card is used only for internal communications.

4.2.4: LOCAL RACK CONNECTIONS

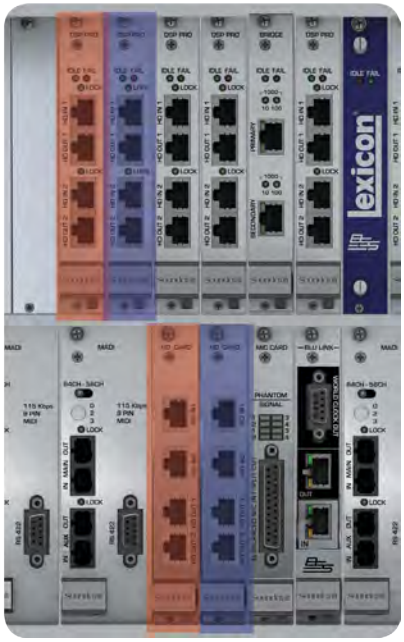
HARDWARE > LOCAL RACK > CONNECTIONS

The SCore Rack Frame cards should be connected correctly before operating the system, using the supplied cables.

1) Connect the DSP cards 1 and 2 (left-most DSP cards in frame) to the HD cards as shown.
2) Connect the Primary Bridge Card output, the MIC Card D-Sub connector, and the BLU Link Card input and output to the corresponding connections on the rear of the Active Breakout Unit (BLU Link in > out; Blu Link out > in).
3) Connect the required Stagebox MADI outputs to the rear of the Stagebox Breakout panel, then connect the Stagebox Breakout Panel connections to the Stageboxes.
4) Connect the TB LINK, CONTROL DATA, and PHONES L & R outputs from the front panel of the Active Breakout Unit to the corresponding connections on the Control Surface.



The left-most DSP cards (1 & 2) must be connected to the HD cards 1 & 2: HD IN 1 > HD OUT 1; HD IN 2 > HD OUT 2 and so on.



Connect the corresponding MADI outputs from the Active Breakout Unit to the rear of the Breakout Panel.

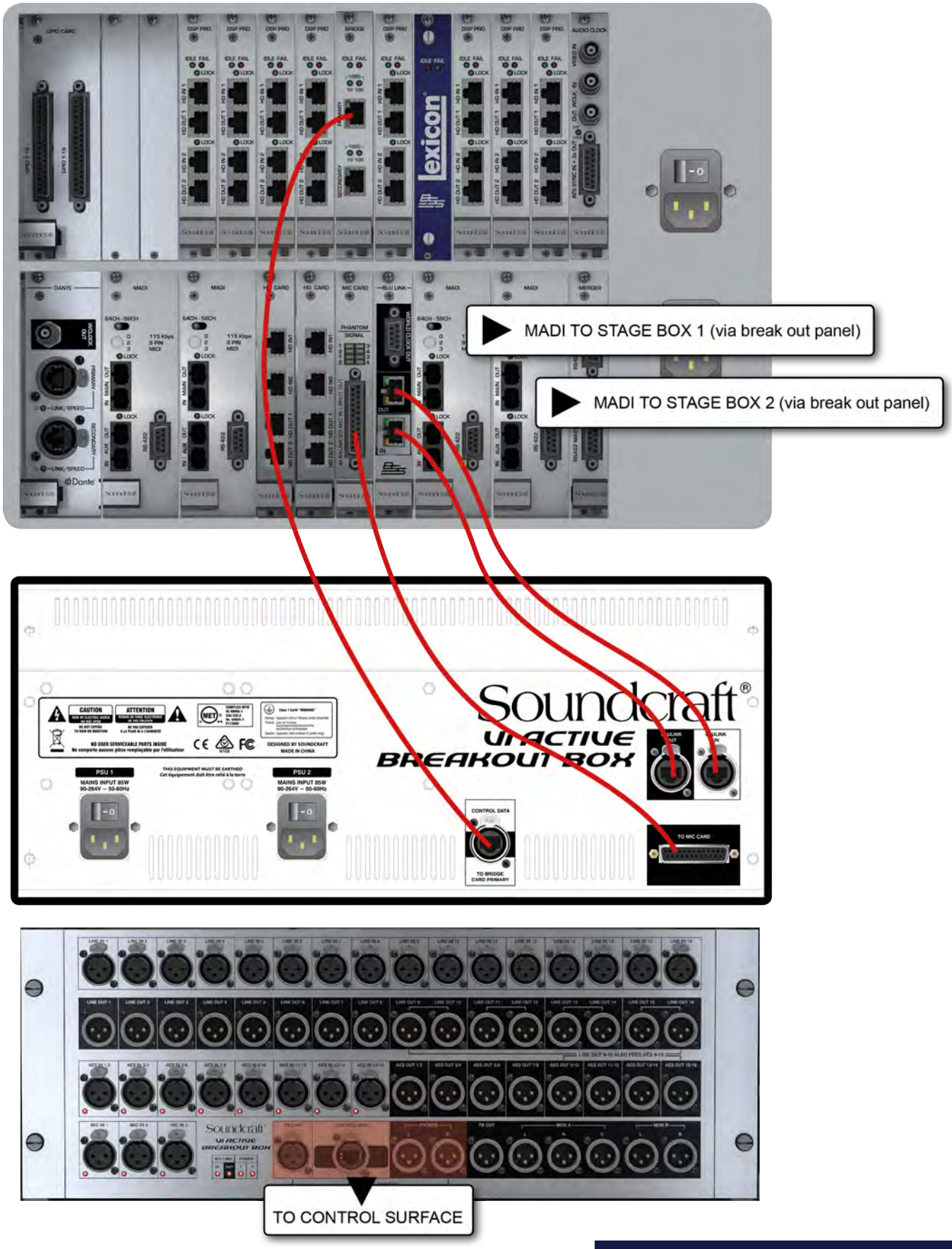


Blank panel holes for customer use.

STAGE BOX connections for MAIN and AUX (Redundancy) MADI connections to stage boxes. (Cat 5 panel shown: alternative panels are available)

4.2.4: LOCAL RACK CONNECTIONS

HARDWARE > LOCAL RACK > CONNECTIONS



HARDWARE > Vi STAGE BOX

The 9U Vi Stagebox houses 64 mic/line inputs and 32 analogue line outputs. Mic/line amp gain, 48 V phantom power and 100 Hz lo-cut filter before the A-D converters can all be controlled remotely from the desk. The Stagebox is available in Standard 64/32 configuration, or can be reconfigured as part-fitted or with alternative cards as required.



The Stage Box contains 12 Slots for 6U-high Audio I/O Cards, a GPIO/LED Card and the MADI HD Card which contains the Stage Box-to-Local Rack connection. Slots are labelled from left to right A-L, and the connectors on the cards are numbered from top to bottom 1-8. These labelling references are used by the patching system (see chapter 11) when the user wishes to patch the connectors to input channels or output busses.

The primary power supplies and an additional cooling fan with filter can be found on the rear panel.

Primary power supply
The primary power supply connects directly to the IEC inlet and provides a full range AC inlet, converting 100V to 240V ac to 24V DC. The Stage box is fitted with two power supplies, providing redundancy for those that require it

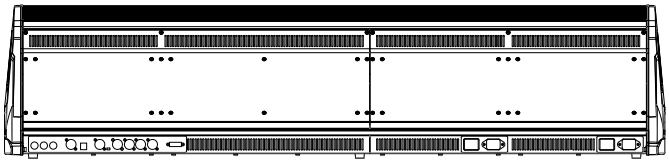
Redundant MADI Cable Operation
The system can be used with either a single cable connected to the MAIN or AUX ports (usually MAIN), or, for additional security, two cables can be used to provide fully-automatic redundant operation. The 3-position INPUT SEL toggle switch on the Stagebox MADI card determines the mode of operation.

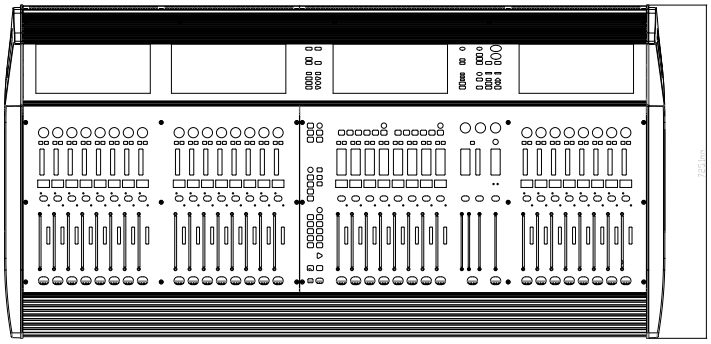
For single cable operation: the switch must be set to either ‘MAIN’ or ‘AUX’, depending on which socket is being used. For dual redundant operation: the switch MUST be set to the ‘RED’ position.

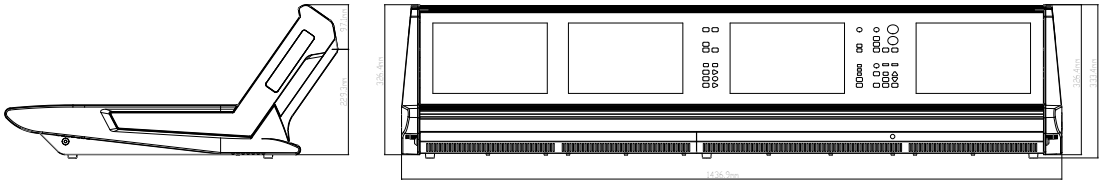
In redundant mode, the system will automatically switch to the other cable if a fault is detected on one cable. Note that it will then stay on the second cable even if the connection is restored on the first cable. This is to prevent intermittent switching.

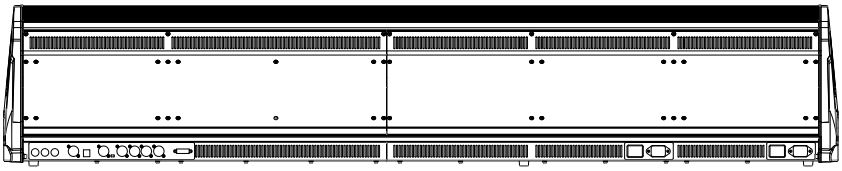
For More information of available Stageboxes, cards, and so on, please see Appendix 1 at the end of this manual.

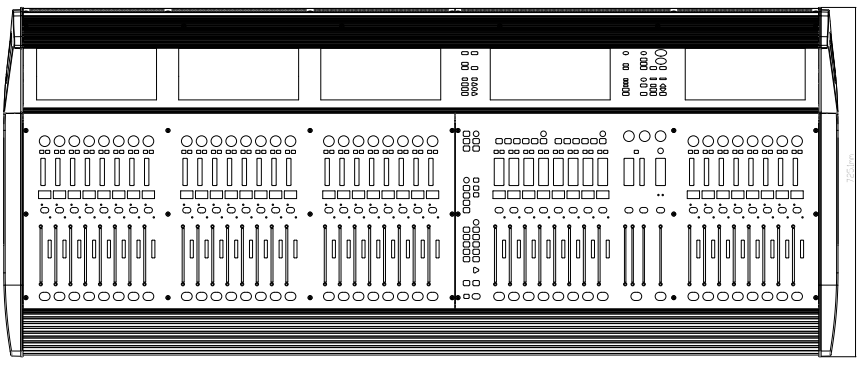
HARDWARE > DIMENSIONS / DRAWINGS

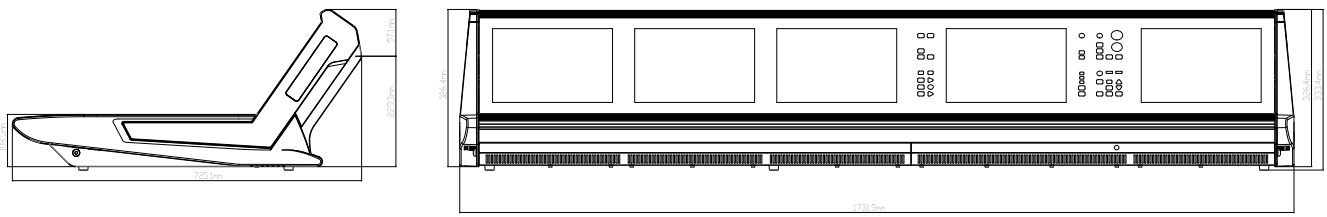












OPERATIONS OVERVIEW



The illustrations in this manual show the Vi5000 console. The Vi7000 has an extra eight-wide Inputs Bay. Where relevant, the text / illustrations will note operational differences between the Vi5000 and the Vi7000 models.

OPS OVERVIEW > CONVENTIONS & COLOURS



This manual, and the console uses certain conventions to make things significantly easier for the user. This includes the unique FaderGlow technology that can dynamically colour code console faders depending on their current assignment.

Conventions used in this manual

Three types of brackets are used to indicate the type of control being referred to.

[ ] is used to indicate a panel-mounted key or encoder.

{ } is used to indicate a Vistonics™ (VST) key or encoder.

( ) is used to indicate a button on a touch-screen.

COLOURS

The following table shows colours used in the VISTONICS screens, and in the headers and footers of the pages in this manual:

Audio Processing

Input Functions	Blue
Equaliser	Red
Filter	Blue
Gate,Comp,Lim,De-ess	Green
Pan, Dir Out, Insert	Yellow
Oscillator	Gold
Monitoring	Lilac

Busses

Aux	Orange
Audio Group	Green
Matrix	Cyan

VCA/MG Indication

VCA 1..8	Blue
VCA 9..16	Pink
Aux VCA Master 1..16	White
Mute Group	Red

Patching & Misc

Input Patch	Blue
Output Patch	Red

OPS OVERVIEW > VISTONICS



Vistonics II Input Bay Screen Areas And Fields

Vistonics II is a unique control and display technology derived from the Studer Vistonics technology and exclusive to Soundcraft. The console uses Vistonics II as a core technology for console operation because of its significant user-interface benefits.

The Vistonics screens are divided into two main areas. The top section is the touch screen area and serves a number of functions. For instance, in the default input channel mode, the input touch screens show the Input, EQ, Dynamics, assigned internal insert FX, Bus outputs, and panning sections of the input channel strip. Touching one of these opens that channel section onto the Vistonics VST area.

The VST area is the lower section of the Vistonics screens, each with two rows of eight VST Fields. Each VST Field contains an encoder, a button, and a display area. VST Fields are normally used for individual parameter control. For instance, the default input Vistonics screen uses the upper row of encoders to control a channel's contribution level to the Aux 1 bus, and the upper row of buttons to turn those contributions on and off.



Buttons On/Off

If a function is assigned to a button it is displayed as shown. The 'active' state shows the button highlighted in a lighter colour.



Open Touch Page

Open a configuration window such as a labelling or patching screen

If the button can open a configuration page in the Touch area it shows a '+' sign on the button. If the configuration page is already open, the button will be labelled with a '-' sign. Pressing the button in that state will exit the configuration page.

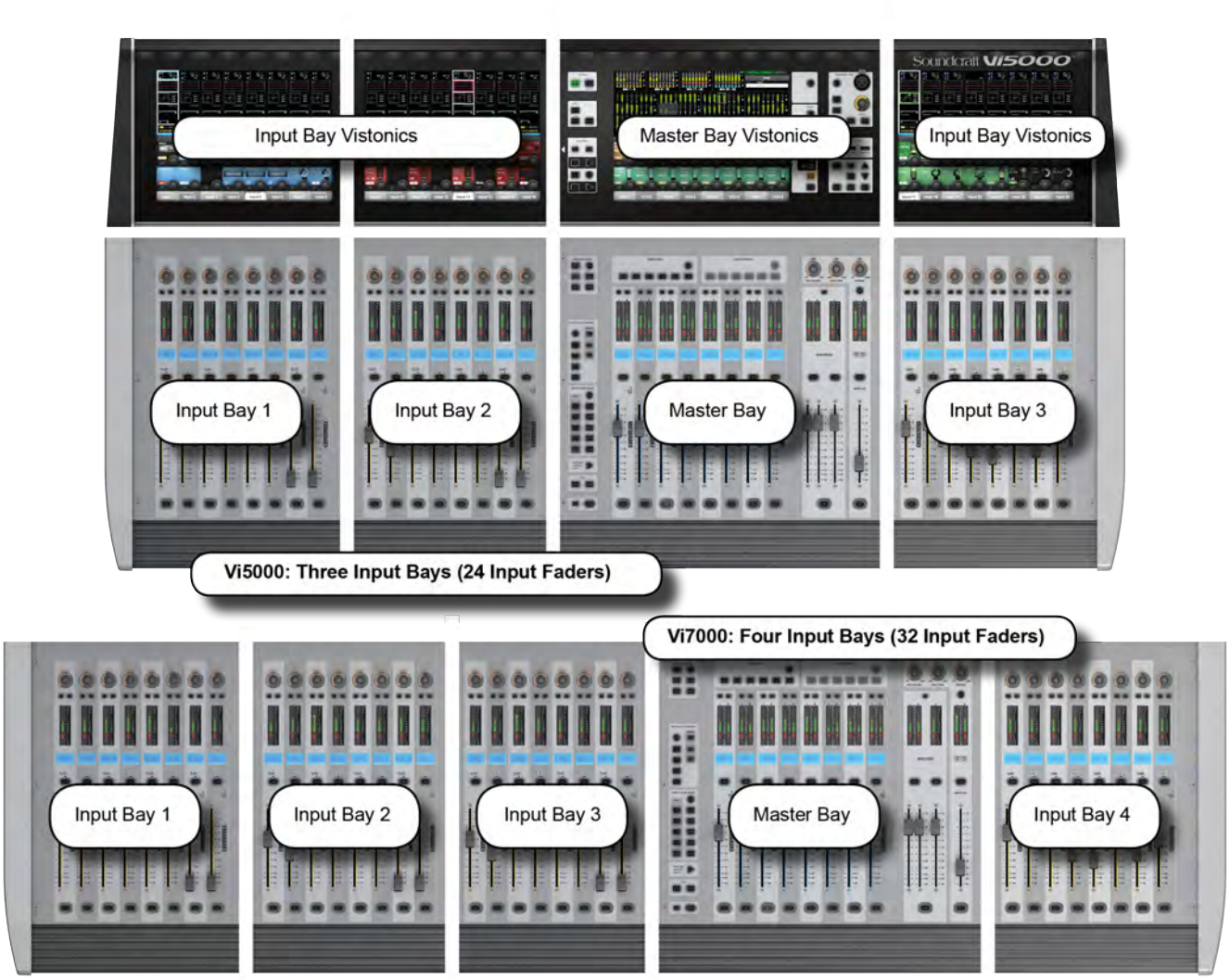


Inactive Fields

If an audio function block is disabled, with the background of the field changed to grey, the button indication will change to a darker colour.

Please note, to allow the presetting of controls, it is still possible to adjust parameters when they are greyed out.

OPS OVERVIEW > CONSOLE BAYS



The Vi5000 is divided into four main bays, and the Vi7000 is divided into five. Both have a single Master Bay, a right-hand Input Bay (which has additional fast-assign capability - see Section 5.3.1), and several left-hand Input Bays (two for the Vi5000 and three for the Vi7000).

Input bays normally control input channels, however, they can have other functions mapped to them depending on how the console is being used. For example, you can map the bus masters, graphic EQ bands, matrix mix sources, aux bus contributions, and more to the input faders.

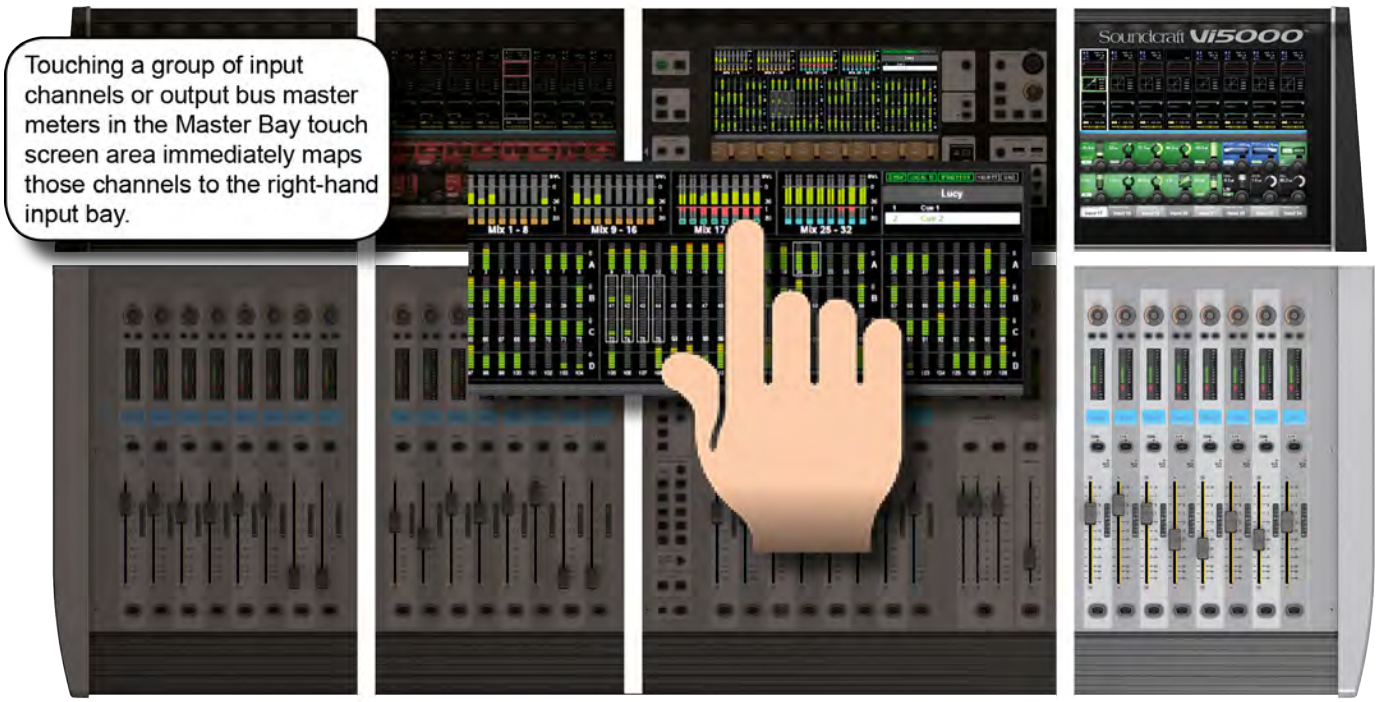
The input bay vistonics touch screen areas normally show eight channel-strip processing blocks for the eight corresponding faders. The input bay VST areas normally show Aux 1 and 2 contributions, though the functions of these encoders are also determined by the input channel VISTONICS button group.

The Master Bay contains assignable faders for the bus masters, plus the mix master (L, R, C) faders and the assignable monitor fader.

The Master Bay touch screen default display is an overview of metering for all input and output channels, as well as system message area and cue list.

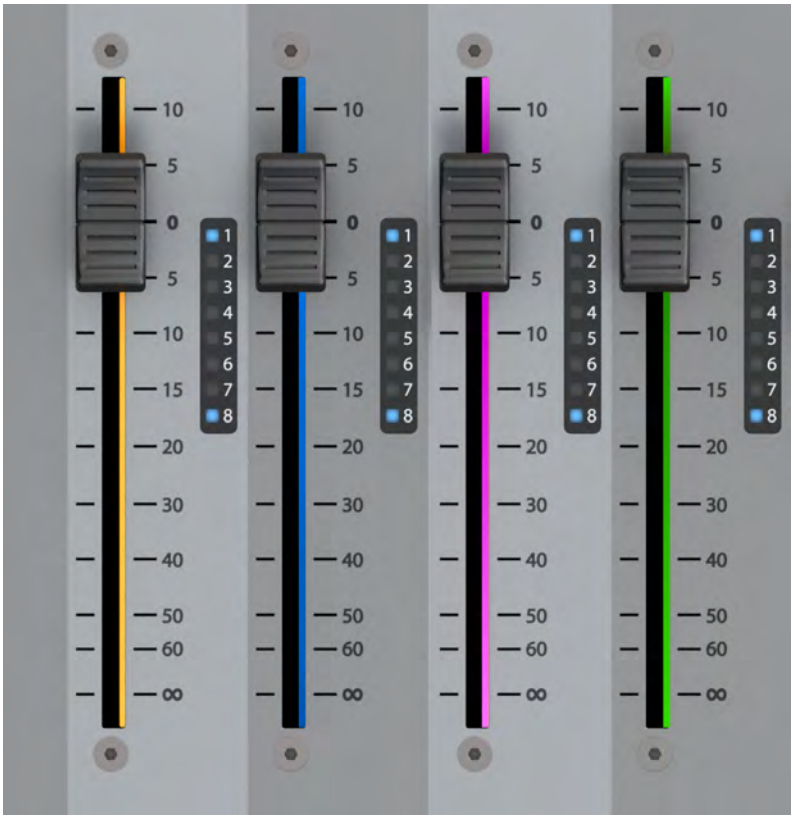
The Master Bay VST area's default mapping is the first 16 bus masters. The next 16 masters can be accessed with the [PAGE 2] button in the Master Bay VISTONICS button group.

OPS OVERVIEW > INPUT BAY 3/4 NAVIGATION



The Master Bay touch screen meters can be used as a navigation device - simply touch a block of eight meters and those channels (both inputs and outputs) are mapped to the right-hand input bay. This is excellent for fast access and for dual operator scenarios.

OPS OVERVIEW > FADERGLOW™



Soundcraft FaderGlow™ (Pat. Pend.) is a unique feature that gives the user an additional level of status indication, and can significantly reduce operating errors.

On the console, several different functions can be assigned to a particular fader, it can therefore be easy to forget which function is currently being controlled, especially when grabbing a fader in a hurry. The main principle of FaderGlow is therefore to indicate the actual function type that is currently assigned to a particular fader. Soundcraft FaderGlow is fitted to all Faders that can change their function.

FADER OPERATION Soundcraft FaderGlow™ COLOUR

Channel level	NONE
AUX send level	ORANGE
GRP (fader closed)	GREEN
MTX contribution level	CYAN
VCA Master 1..8	BLUE
VCA Master 9..16	PINK
Aux VCA Master 1..16	WHITE

OPS OVERVIEW > CONSOLE BUTTONS



There are a wide variety of button groups on the console. Below are described all button groups and selected individual buttons. Please refer to the relevant reference section for detailed information on the operation of all buttons.

Most buttons on the console can be latching (stay on) or Momentary (on when pressed, off when you let go) If you hold a button down it will have a momentary action. If you press a button and let go within half a second it will 'latch'.



System Button Group

[MUTE ALL O/P] (Mute All Outputs) and [POWER ON] buttons.



Edit Button Group

[COPY], [PASTE], and [UNDO] buttons.

These are used for copying and pasting whole channels, processing blocks, and individual parameters either elsewhere on the console or to and from the console's powerful library system. For more information see the Edit System reference chapter 13.

OPS OVERVIEW > CONSOLE BUTTONS



Input Channel VISTONICS Group

The input channel VISTONICS control buttons provide convenient ways of assigning bus master functions to the input channel VST encoders, such as selections for individual bus 'tap' points, and getting the input bus contribution shown on the VST screen to follow the master bus selection in the Master Bay.

For more information, see the Inputs reference chapter 6.



Output Vistonics Group

These buttons provide control over the VST section functionality of the Master Bay vistonics screens. For example, the [PAGE A] and [PAGE B] buttons switch the encoder section of the screen between busses 1-16 and 17-32. For More information, see Outputs Reference chapter 7.



Encoder Mode Buttons

These buttons assign various functions to the input channel strip encoders. For more information see the Inputs reference chapter 6.



Mute & VCA Groups

Control and assign the four mute groups and available 16 VCAs. For more information see the Mute & VCA Groups reference chapter 9.

OPS OVERVIEW > CONSOLE BUTTONS



Input Fader Pages

These buttons control navigation through five basic input channel Pages/Layers, including the 'All Busses' view, plus five user-definable pages. These buttons only affect the input channel bays. For more information see the Input Channels reference chapter 6.



Output Fader Pages

Navigate through eight-wide pages of the console's output busses and VCA masters. These buttons only affect the Master Bay channel strips. For More information see the Output Channels reference chapter 7.



Next & Last

[NEXT] and [LAST] buttons for moving through the loaded Cuelist. These buttons have to be enabled in the Settings page of the main Menu. For more information see the Snapshots, Cuelist, and Global Filter reference chapter 14.



Snapshot Control

Full control over Snapshot and Cuelist storage, navigation, and recall. For more information see the Snapshots, Cuelist, and Global Filter reference chapter 14.



Talkback & Oscillator

Most of these buttons are dedicated to activating preset Talkback routings. To access Oscillator functions, use the [SETUP] button. For more information see the Talkback And Oscillator reference chapter 15.



[SETUP] Buttons

[SETUP] buttons open the relevant setup screen in the Master Bay Vistonics screen.

While a setup screen is active, pressing the [SETUP] button again will close that setup screen. For more information see the chapter relevant to each [SETUP] button.



[SOLO/SEL] Buttons

Solo or select a channel.

The [SOLO/SEL] keys operate in two modes. The default function is to enable the SOLO path from its channel or bus to the monitoring system. However, if a touch-area page is open, pressing a [SOLO/SEL] from another channel (within its own bay of eight channels) moves the touch-area page to this new channel. [SOLO/SEL] buttons also perform other selection functions depending on the console mode. These are described in the relevant reference chapter.



[GANG] Buttons

Special Ganging functionality.

For linking channels, processing blocks, and parameters, for convenient control. See the Gang section in this operations overview chapter for more information.



[SOLO CLEAR] Button

Clear any currently active console Solos.

OPS OVERVIEW > CONSOLE BUTTONS



Lower [FLW] Button

Input faders follow master bus selection.

The fader FOLLOW OUTPUT SOLO [FLW] button located near the bottom of the Master Bay buttons strip (not the VISTONICS encoder [FLW] buttons) causes the input faders to become contributions to an output bus when its [SOLO/SEL] button is pressed (often known as 'Sends on Faders mode').

Matrix (MTX) [SOLO/SEL] buttons work differently. If a Matrix Output Solo/Sel is pressed, regardless of the setting of Follow Solo modes, the Output Solo will be activated, the EQ/Dyn/Misc touch screen for the Matrix Output displayed, and the channel faders will be assigned to the contribution levels from the Outputs to the Soloed Matrix Output.

Note: the lower [FLW] key has an additional function in allowing VCAs to control the Aux sends of input channels. See the Mute & VCA reference chapter 9 for details.



Main Menu Button

Brings up the Main Menu screen in the Master Bay.

Settings and preferences for a variety console functions are arranged in tabbed pages in the main menu. For more information, see the Main Menu reference chapter 16.



Input Priority

Allow an Output Solo to remain active, whilst an Input Solo is temporarily activated 'over the top' of it.

For more information see the SOLO System reference chapter 8.

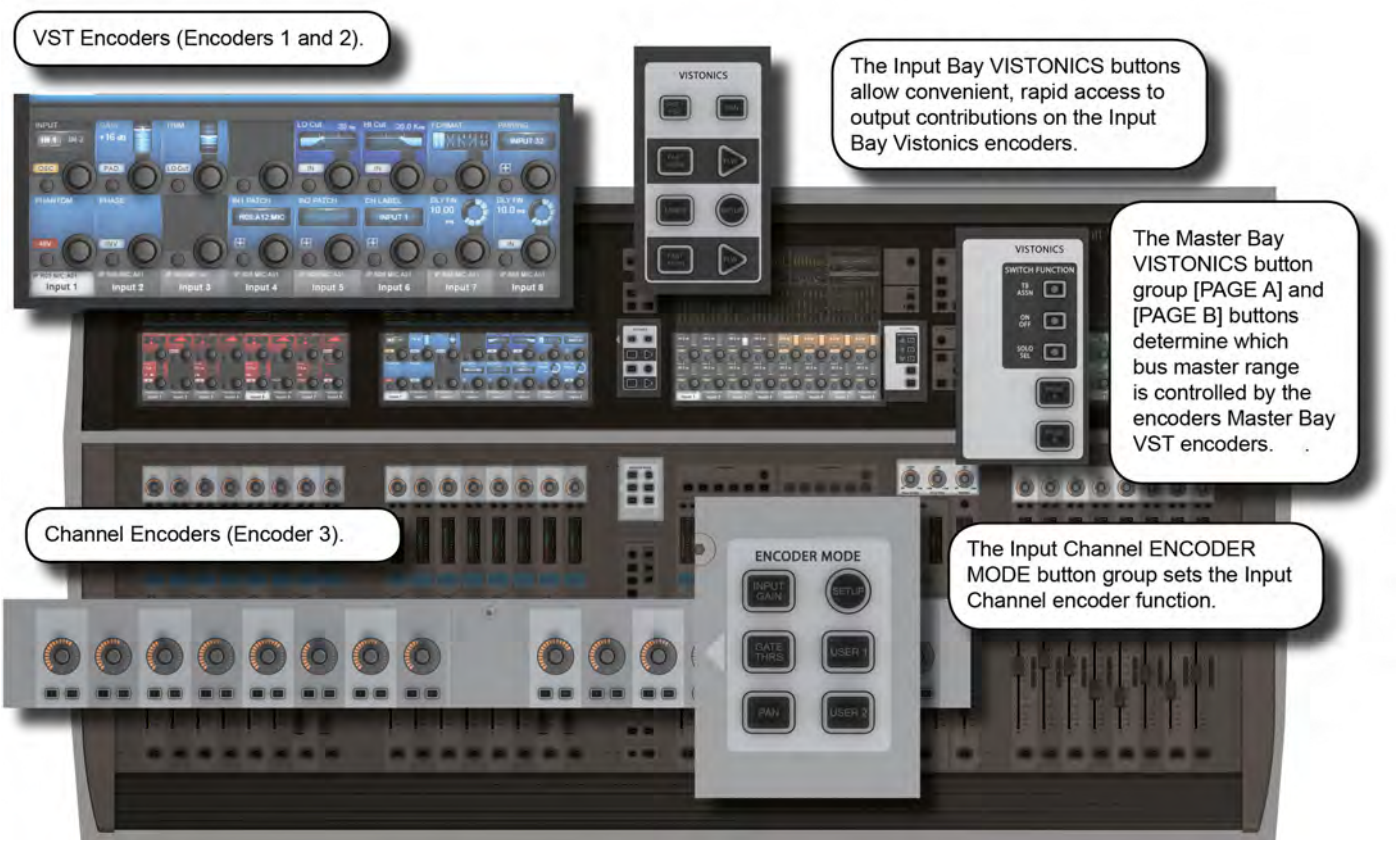


Lock Meters

Locks the Meter section of the Master Bay touch screen onto the screen.

The meter section will be safe from output selections bringing the channel up on the touch screen, for example. [SETUP] buttons still takeover the screen though.

OPS OVERVIEW > ENCODERS



Each input channel strip contains three encoders: encoder 1 and encoder 2 are located in the VST fields in the lower screen area, while the channel encoder is located at the top of the fader area, and has an LED ring to indicate its parameter state. Each of these encoders can control different parameters, depending on the settings of other parts of the console. There are user-assignable settings for all encoders.

The master section has 16 VST encoders and 4 panel-mounted encoders with LED rings: the TB/OSC Level Control encoder, and the Sold Blend, Solo Trim & Phones Volume encoders. These last four are dedicated to their respective functions.

Channel encoders always control a parameter on their own channel strip. The function of the channel encoders can be globally selected via the [INPUT GAIN], [GATE THRS], [PAN], and [USER 1-2] keys on the Encoder Mode panel. See the Inputs reference chapter 5 for more information.

The VST Input Bay encoders default to Aux 1 and 2, though can be assigned a variety of functions depending on the console mode. If any touch field is activated, the 16 VST encoders are assigned with expanded channel function parameters. See the Inputs reference chapter 5 for more information.

The default setting for the Master VST encoders is as the output level controls for Master Outputs 1-16. This can also be selected by pressing the [PAGE A] key on the Master Vistonics Mode Panel. Pressing [PAGE B] will cause the Master VST encoders to be assigned as the output level controls for Master Outputs 17-32.

The Master VST encoders can also be assigned to Master Output Expanded Functions (e.g. EQ, Dynamics, etc.). When a Master Output [SOLO/SEL] key is touched, it opens the Processing Area in the Master VST screen. If then a particular touch-area is touched, the VST encoders are assigned to appropriate expanded functions. For more information see Outputs reference chapter 6.

OPS OVERVIEW > GANGS



Gang is a very helpful feature to speed up operations that influence functions on multiple input channels, or on output busses, in the same way. Any parameter change on ganged channels will be applied to all other ganged channels as an offset. For example, adding 4dB of level to an aux send on one channel will add 4dB of level to the corresponding Aux sends on all other ganged channels - it will not 'copy' the actual level to the other ganged channels.

Button presses will change any corresponding buttons in ganged channels that are not currently in the new state, to the new state. From that point on, further presses will result in all switches changing state together.

Gang mode is activated and deactivated with the [GANG] button. You add or remove channels to or from the gang with the channel [SOLO/SEL] buttons.

Entering Gang Mode does not cancel any solos of any type that are active at the time. The Solo system continues to work as it was when Gang Mode was switched ON. The amber 'Solo' illumination of the Solo/Sel switches cannot be seen whilst gang mode is ON.



Gang Button

Activate or deactivate Gang Mode.

After activating Gang Mode, hold the [GANG] button down to Gang all channels.
You can leave Gang Mode pressing the [Gang] button. If you haven't 'cleared down' the gang then it will be restored the next time you re-enter Gang Mode.



Select (Gang)

Select and deselect ganged channels.

When in Gang Mode you can add single channels by pressing individual channels' [SOLO/SEL] buttons OR you can add a range of channels by pressing the first and last channels' [SOLO/SEL] buttons together.
Press & Hold any [SOLO/SEL] button to clear down the entire gang (recommended after use to avoid accidental operation of Gang mode).
When in a gang, a channel's [SOLO/SEL] button will glow blue.

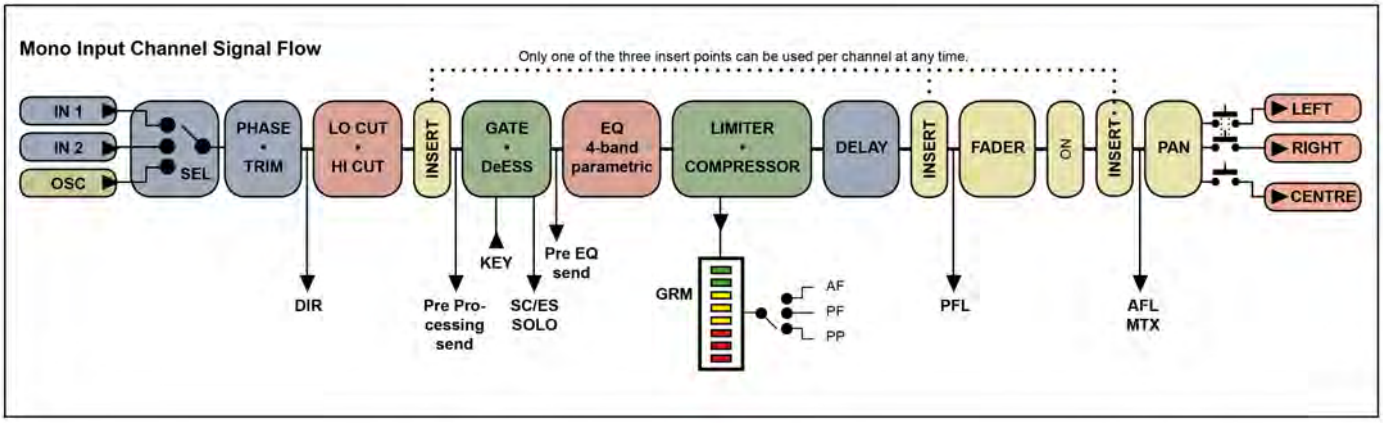


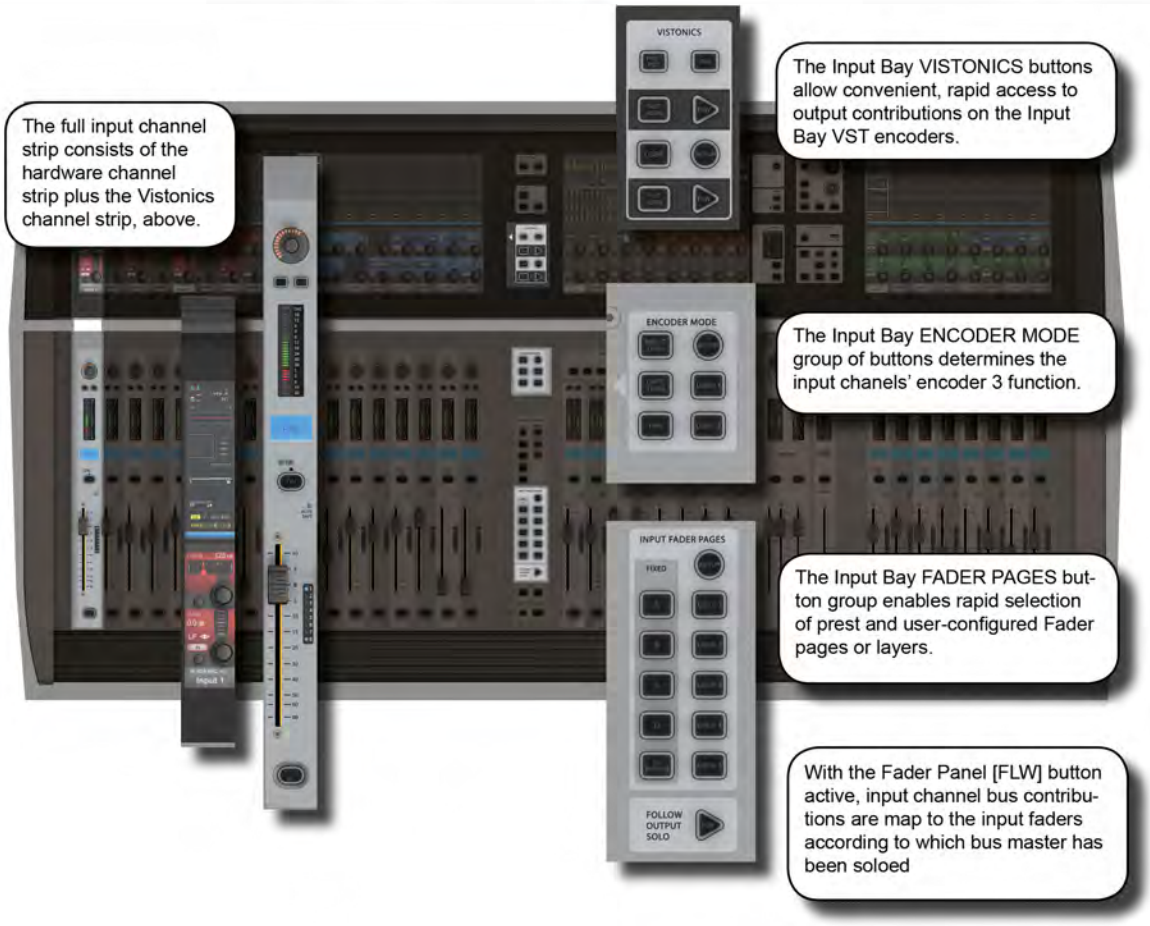
The console labelling system is displayed on a touch-screen area whenever the labelling of a channel, output, cue, file, or other named parameter or item is required. For example, you can label an input channel by touching the channel's input area on the touch screen then pressing the {CH LABEL} button in the VST area.

Labelling can be done with the displayed on-screen keyboard or an external USB keyboard. The on-screen keyboard is context sensitive and shows only the allowed character and symbols.

Long Labels are displayed along the bottom of each touchscreen and can contain up to 10 characters. Short labels are displayed on the LCD displays located on the Fader panel and can contain up to six characters.

INPUT CHANNEL



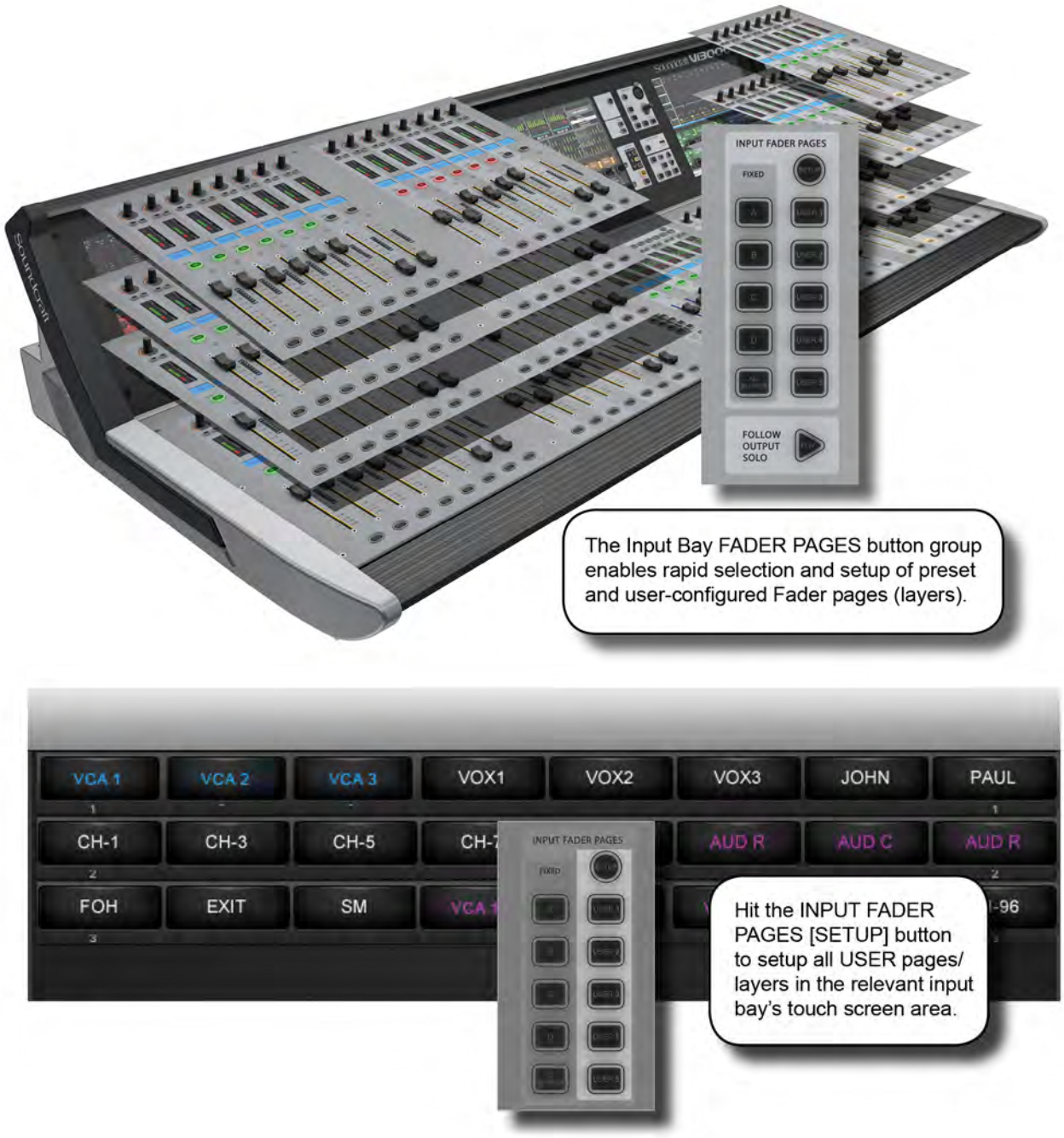


Bays 1, 2, and 4 of the console (Input Bays 1, 2, and 3) are normally dedicated to controlling input channels. However, other functions can be mapped to these channels, such as the Master Busses, Matrix feeds, Graphic EQ, and so on.

There are four main input channel control areas on the console:

- The main channel strips, including the Vistonics screen sections, are for control of input channel parameters, routing, and patching.
- The INPUT FADER PAGES buttons are for navigating the available input channels by choosing preset and user-definable fader pages.
- The input channel VISTONICS control buttons provide convenient ways of assigning bus master functions to the input channel VST encoders.
- The ENCODER MODE buttons provide fast switching of the channel strip encoder functions.

INPUT CHANNEL > FADER PAGES



This section is used for navigating Fader Pages (or layers). There are four main layers of input channel strips, and a further five user-definable layers available. The [ALL BUSSES] button maps the output busses to the input channel strips, as described further in the Outputs reference section.

In addition, you can touch the input meter fields in the Master Bay Vistonics to remap those input channels to the right-most input bay.

INPUT CHANNEL > FADER PAGES



Fixed Fader Layers

Select four pages/layers of input channel strips.

Each button maps Pages of 24 (Vi 500) or 32 (Vi 7000) channels to the three/four input bays, so for the Vi 5000 input fader layer [A] is channels 1-24; [B] is channels 25-48; [C] is channels 49-72; and [D] is channels 73-96. For the Vi 7000 it's: [A] 1-32; [B] 33-64; [C] 65-96, [D] 97 - 128.



User Pages Setup

Set up user Fader Pages using the fader bay touch screens.

Press [SETUP], then select a User page. The touch screens will display five rows of channel numbers that correspond to the five User Pages. Touch a user page 'slot' to select a channel for that slot. You do not need to select User Fader Layer buttons - all layers are shown on-screen in set-up mode.



User Fader Layers

User-Defined input channel strip pages/layers

Select one of these buttons to map that user-defined fader-layer to the console. The User layers are defined by pressing the FADER PAGES [SETUP] button.

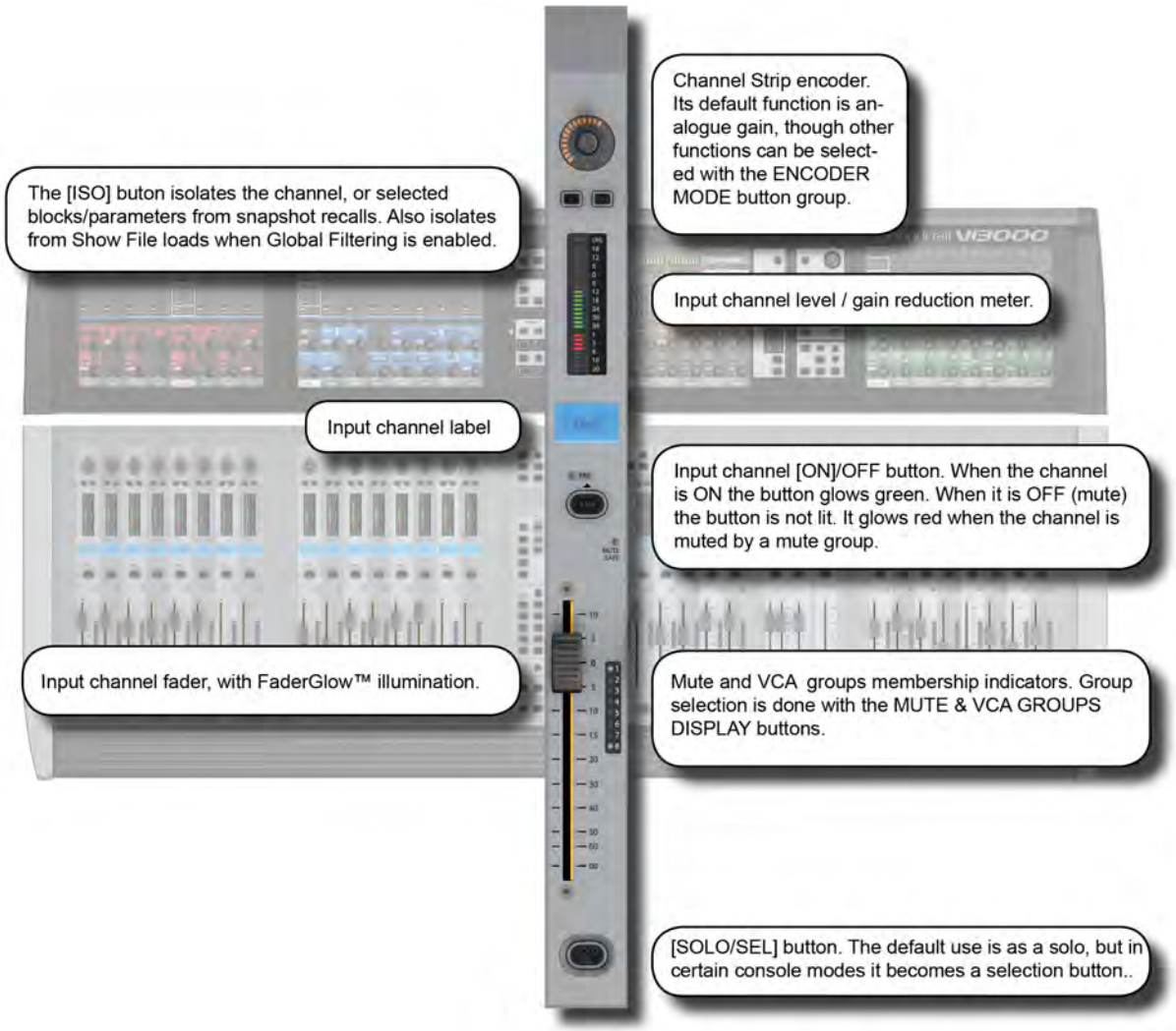


All Busses Mode

Map the output busses to the input bay fader strips.

See Outputs Reference section 7 for more detail on All Busses mode.

INPUT CHANNEL > CHANNEL STRIP





Input Channel Strip Encoder

Rotary control whose function is defined by the input ENCODER MODE buttons. The Default function is control over the channel's analogue input gain.

For more information see the Inputs Encoder Mode reference section 6.3



Input Channel Strip F

Switch between assigned inputs 1 and 2

When the F key is lit, the input channel is using its IN 2 Patch. To assign IN 1 and IN 2, see section 6.5.1.

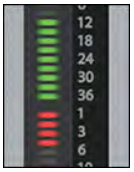
INPUT CHANNEL > CHANNEL STRIP



Input Isolate

Used to isolate the channel (and selections) from snapshot recall and (when Global Filtering is active) from Show file loads.

See the Snapshot and Global Filtering reference section 14 for more information.



Level and GR Meter

Level and gain reduction metering for the input channel.

See the metering reference section 12 for more information.



Input Channel Label

Displays the short label/name of this channel.

The long and short channel labels are set in the channel's 'Input' Vistonics parameter block. See reference section 6.5.1 for more information.



'Pre' Indicator

Bus send routing indicator.

When Bus sends are assigned to channel faders, using FLW mode, the PRE indicator lights to indicate when the send is set to a Pre-fader source.



Channel ON/OFF

Channel [ON]/OFF button. 'OFF' is effectively a mute control.

The button glows green when the channel is 'On'. If the channel is off there is no glow colour. However, if the channel is muted because a mute group has been activated, the [ON] button will glow red.



MUTE SAFE indicator

Indicates when the channel has been set to MUTE SAFE mode from the Monitor Setup menu (refer to Chapter 10).

When set to MUTE SAFE, a channel can still be manually muted by pressing the ON/OFF button, but cannot be muted by 'remote' actions such as Mute or VCA group operation, snapshot recalls or activation of a Solo in Place (SIP) from another channel.

INPUT CHANNEL > CHANNEL STRIP



Mute/VCA Group Indicators

Shows the Mute or VCA Groups that is channel belongs to.

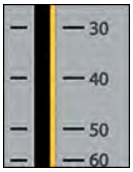
The actual target of these LEDs (Mutes, VCAs 1-8, or VCAs 9-16) is determined by the DISPLAY buttons in the MUTE & VCA GROUPS button group. See the Mute & VCA Groups reference section 9 for more information.



Channel Fader

Control the channel output level.

The input channel faders can also be assigned to follow the bus master selection in the Master Bay with the fader [FLW] button and become level controls for contributions to that bus.



FaderGlow

The channel fader is colour-coded depending on its function.

Input channel levels are not colour-coded. The other FaderGlow colours are Aux Sends: ORANGE; Groups: GREEN; Matrix Contribution: CYAN; VCA Master 1-8: BLUE; VCA Master 9-16: PINK.



Solo / Select

Default is the channel Solo function (listen to this channel only on the monitor bus).

If a touch-panel screen is open for the for another channel in the same bay, the [SOLO/SEL] button moves the touch screen to this channel. The button has other functions in other console modes such as selecting channel for a Copy or Paste edit function. Other functions for the [SOLO/SEL] button are detailed in the appropriate reference sections. For more on the console Solo System see reference section 11 for more information.

INPUT CHANNEL > ENCODER MODE



These buttons determine the function of the input channel encoder at the top of the main channel strip.

If [ALL BUSSES] is active, the channel encoders are disabled and have no function (the previous function is remembered however).

You can assign a range of input channel parameters to the [User 1] and [User 2] assignments by entering setup mode with the Encoder Mode [SETUP] button.



Mode: Input Gain

(Default) Switch the input Encoders to control analogue input gain.



Mode: Gate Threshold

Switch the input Encoders to control Gate Threshold.

INPUT CHANNEL > ENCODER MODE



Mode: Pan

Switch the input Encoders to control Pan.



Setup

Assign input channel parameters to the User 1 and 2 Encoder Mode buttons

To assign User 1 and 2 Encoder Modes, press [SETUP], then use the VST encoders to choose from a range of input channel parameters, including EQ and dynamics controls. Press the VST Encoder [SET] button to confirm the selection and press [SETUP] again to exit Setup Mode.



Mode: User 1

Switch the input Encoders to control the parameter assigned to the [User 1] button.

The default assignment is Compressor Threshold.



Mode: User 2

Switch the input Encoders to control the parameter assigned to the [User 2] button.

The default assignment is Digital Trim.

INPUT CHANNEL > VISTONICS / VST CONTROL



These buttons control the functionality of the input channel VST encoders.

You can easily and rapidly assign bus contributions and bus panning to the encoders using this button group. You can assign any group or auxiliary output to the two [USER] rows using the [SETUP] button.



Pre/Post

Configure Aux sends from channels (when they are assigned to the two VST encoder rows), as one of four sources: Pre-EQ, pre-compressor, pre-fader or post-fader.



VST Mode: Pan

Activate aux panning for encoders where applicable.

If a stereo Aux is assigned to a VST encoder row, and if the [PAN] key is active, the encoder will control the Aux Pan rather than the contribution level. If both Auxes assigned to Row 1 & 2 are Stereo Auxes, both Rows 1 and 2 will change to the PAN function across the desk. If only one of the two rows is a Stereo Aux, then only this row will change to PAN. If neither row has a Stereo Aux assigned, the [PAN] switch will have no function.



Fast Assign

Provides a very fast way to temporarily assign a bus function to a VST encoder row.

Press and hold one of the [FAST ASSN] keys, then press one of the Output Masters' [SOLO/SEL] keys. The relevant row of VST encoders will now be assigned to that Output master, and the [FAST ASSN] key in question will illuminate. There will be no influence on audio, the Output solo is not activated.

INPUT CHANNEL > VISTONICS / VST CONTROL



FLW: Encoders Follow Masters

Activate the FOLLOW SOLO function for the respective VST encoder rows.

Pressing a bus master [SOLO/SEL] button will automatically assign the soloed bus master to this row, overriding the default or the [USER] layer. Note that only one [FLW] can be active at a time.



VST Mode: USER

Assign the two VST Encoder rows as user-definable output controls.

The default assignments are Aux 3 and Aux 4. Use the VST User Mode [SETUP] button to define alternative allocations.



SETUP

Define a VST Encoder mode.

To define the VST Encoder mode, press [SETUP]. You can now use the VST encoders to choose any AUX or GRP control. Press the VST Encoder button (SET) to allocate the chosen control. Press [SETUP] again to exit Setup mode.

INPUT CHANNEL > TOUCHSCREEN > INPUT BLOCK



A mono input channel's input section touch screen area is at the top of the Vistonics input channel strip area, colour-coded blue. The input section consists of that channel's input source selection, pre-amp and gain stages control, channel pairing selection for stereo (and other format) channel creation, channel labelling, and channel delay adjustment.

INPUT CHANNEL > TOUCHSCREEN > INPUT BLOCK



Channel Input Select

Select either Input 1 or Input 2 as the channel input.

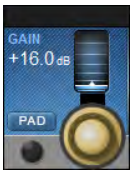
The two physical channel inputs are assigned in the IN1 PATCH and IN2 PATCH fields. You can also use the channel strip 'F' button to switch between IN 1 and IN 2.



Oscillator Input Select

Assign the console's central oscillator to the input of this channel.

When OSC is active, the normal input selections are disabled and the corresponding input touch field is highlighted in orange.



Input Gain

Adjust the analogue input gain.

This is an analogue gain stage, before analogue-to-digital conversion.



Input Pad

Reduce the input sensitivity by 20dB.

When the Pad is active, the PAD text in the input touch field is highlighted in blue.



Input Trim

Adjust the digital input gain.

This is the post analogue-to-digital conversion gain, in the digital domain.



Analogue Lo Cut filter

Insert the analogue (pre AD converter) 80Hz low cut filter (only analogue inputs).

When the analogue LO Cut filter is engaged, the LO Cut symbol in the input touch field is highlighted in blue. This filter affects analogue mic inputs only.



LO CUT Cut-Off Frequency

Adjusts the LO CUT cut-off frequency

This is part of the digital (post AD) LO CUT / HIGH CUT section and affects all inputs. The affected frequency range is shown graphically in the EQ section of the touch screen.



LO CUT IN

Switch the digital LO CUT filter in and out

When the digital LO CUT filter is switched in, the LO CUT indicator above the response graph in the EQ touch section is highlighted in blue.



HI CUT Cut-Off Frequency

Adjusts the HI CUT filter cut-off frequency.

This is part of the digital (post AD) LO CUT / HIGH CUT section and affects all inputs. The affected frequency range is shown graphically in the EQ section of the touch screen.



HI CUT IN/OUT

Switch the digital HI CUT filter in and out

When the digital HI CUT filter is switched in, the HI CUT indicator above the response graph in the EQ section of the touch screen is highlighted in blue.



Channel Pairing Selection

Activate channel pairing selection on touch screen.

The touch screen will show pairing candidate options for the channel - these will be consecutive channel numbers (horizontal neighbours) and the corresponding channels on higher and lower layers (vertical neighbours). All channel parameters will be copied to the selected pairing candidate and the pairing FORMAT field will be displayed.



Pairing Format Selection

Adjusts stereo format of paired channels.

The choices are LR, RL, LL, RR, MONO. If the channel is not paired, the FORMAT field will not be displayed.

INPUT CHANNEL > TOUCHSCREEN > INPUT BLOCK



Phantom Power

Apply Phantom Power (+ 48V) to a mic input XLR patched to this input.

When phantom power is applied to this channel's input, the '48V' indicated (touch screen input section) is highlighted in red.



Phase Invert

Invert the phase of audio through this channel.

When phase invert is active, the phase symbol in the input section of the touch screen is highlighted blue.



Input Patch

Activate input patching selection on touch screen.

The input patch selection screen will give you access to all available inputs / sources, both external (local I/O, Stage Box, MADI, and so on) and internal (Lexicon effects). If an input channel is paired, you will also have to option to select L or R assignment from the same screen. The IN 1 and IN 2 field text areas will show the selected source.



Channel Label

Activate channel labelling functionality on main touch screen.

This control brings up the standard ASCII keyboard display and colour selector in order to label the channel. You can specify short and long channel names, and a label colour. You can also use a connected USB keyboard if available.



Delay Fine Adjust

Fine adjustment of the channel delay time.

Adjusts overall delay time in small steps. The delay units (feet, meters, seconds etc) of adjustment can be changed in the main Settings screen, via the [MENU] button.



Delay Adjust

Adjustment of the channel delay time.

Adjusts overall delaytime. The delay units (feet, meters, seconds etc) can be changed in the main Settings screen, via the [MENU] button.

INPUT CHANNEL > TOUCHSCREEN > EQ BLOCK



A mono input channel's parametric EQ section touch screen area is near the top of the Vistonics input channel strip area, colour-coded red. The EQ section consists of four full-range parametric bands with shelving response available to the LF and HF bands..

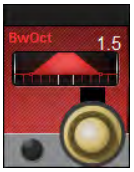


Frequency Control

Adjust the centre frequency (band pass) or cut-off frequency (shelving) of the EQ.

All changes are reflected in the graphic Vistonics touch screen EQ section. Any band currently being adjusted is shown highlighted in white. All four bands feature full-range frequency control.

INPUT CHANNEL > TOUCHSCREEN > EQ BLOCK



EQ Bandwidth / Q Control

Adjust the bandwidth or Q (inverse) of the current filter.

You can chose whether the adjustment unit is Bandwidth or Q via the main settings screen by hitting the [menu] button.



Shelf Select

Choose a shelving response for the filter.

The shelf response is available to the high (HF) and low (LF) EQ bands.



Gain Control

Adjust the gain applied to the defined filter band.



EQ Band In

Switch the selected EQ band into the Equaliser signal path.

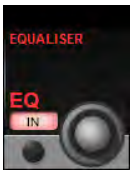
Please note, there is also an EQUALISER IN button to switch the equaliser section into the channel signal path.



Flatten All

Flatten settings for the whole equaliser section.

The button has to be pressed and held for 2 sec to activate the function.



EQ In

Switch the whole equaliser section into and out of the channel signal path.

Respects in/out settings on individual bands. When the Equaliser section is 'In' the EQ graphic on the Vistonics input channel touch screen area will be highlighted in red.

INPUT CHANNEL > TOUCHSCREEN > DYNAMICS BLOCK



The dynamics section has three main components - The first can either be a GATE or DeEsser section; the other two are a COMPRESSOR and a LIMITER. All three sections can be turned on or off independently.

The dynamics section of the Vistonics touch screen shows gain reduction as a compression transform, and Gate/ DeEss selection. It also shows either the Gate status, or a gain reduction meter, depending on the Gate / DeEss selection.

Gate

A gate is a threshold driven gain reduction process normally used to reduce the level of, or silence, a signal when it falls below the threshold. The gate section includes an assignable side-chain input with filters, a key signal solo function, and a ducking mode.



Gate / DeEsser Selection

Select either the GATE or DeEsser function for the first dynamics process.

The top line of dynamics controls adapt to this selection.



Gate Threshold Set

Adjust the gate threshold level.



Gate In

Include the Gate in the dynamics signal path.

When Gate In is selected, the word 'GATE' is highlighted in the touch screen dynamics section.



Gate Attack Time

Adjust the attack time of the Gate.



Gate Duck Function

Select the Gate DUCK mode.

DUCK mode in inverse gate mode which reduces level when the side chain input EXCEEDS the threshold. This can be used for 'ducking' a music bed under a DJ vocal, for instance.

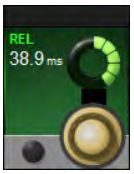
INPUT CHANNEL > TOUCHSCREEN > DYNAMICS BLOCK



Gate HOLD function

Adjust the Gate Hold Time

This is the time an activated gate will remain fully closed, before entering the 'release' phase.



Gate Release Time

Adjust the Gate Release Time

This is the time an activated gate will take to open, from its 'hold' state.



Gate Range

Adjust the Gate Range.

This is the difference between the normal signal level and the gated signal level.



Gate Side Chain Solo

Switches the Gate Side Chain signal to the Solo buss.

For monitoring the signal being used as the gate 'key' (The signal used for testing gate threshold).



Lo Cut Frequency

Adjust the cut-off frequency of the gate sidechain lo-cut filter.

Together with the gate hi-cut filter, this creates a band-pass filter for side chain control of the gate.



Sidechain Filter In

Inserts the gate Hi and Lo Cut side chain filters into the side chain signal path.

Together the gate lo-cut and high-cut side chain filters, create a band-pass filter for side chain control of the gate.

INPUT CHANNEL > TOUCHSCREEN > DYNAMICS BLOCK



Gate Hi Cut Filter

Adjust the cut-off frequency of the gate sidechain hi-cut filter.

Together with the gate lo-cut filter, this creates a band-pass filter for side chain control of the gate.



External Key Assign

Activate the External Key Assign screen in the main Vistonics touch screen panel.

All external I/O options for the external key become available for selection.

DE-ESS Function

The de-esser is a dynamically controlled filter normally used to reduce the sibilance ('sss' components) in a singer's voice. If the de-esser is active, the signal level will be reduced only in the band set by the filters, when the signal in this band exceeds the required threshold.



De-Esser Sensitivity

Adjust the sensitivity of the De-esser process.

The de-esser can be used instead of the Gate, but not at the same time. The Gate-DeEss control is used to select which type of processing is required.



De-Esser In

Activates the De-esser process



De-Esser Range

Adjusts the De-esser attenuation.

The maximum amount by which the signal in the 'Ess' band is attenuated when the DeEsser is activated.

INPUT CHANNEL > TOUCHSCREEN > DYNAMICS BLOCK



Ess Band Solo

Solo the De-esser dynamic filter.

Helps pinpoint the 'Ess band' most suited to the current signal.



De-Ess Filter Frequency

Adjust the centre frequency of the De-ess dynamic filter.



De-Ess Filter Bandwidth

Adjust the bandwidth or Q of the De-esser dynamics filter

You can set the unit of adjustment (octave bandwidth or Q) in the main Settings screen, activated with the [menu] button.

Compressor / Limiter

A compressor is a threshold driven process used to reduce the dynamic range of a signal by applying gain reduction when the signal level exceeds the threshold and applying 'make-up gain' to keep the overall level consistent. The compressor includes an automatic make-up gain function.
A Limiter is a dynamics process used to limit peak signal level by reducing signal level as it approaches or exceeds the threshold.



Compressor Threshold

Adjust the threshold of the compressor

When the signal exceeds the threshold, it will be attenuated according to the compression settings.



Compressor In

Activate the compressor process.

The compression process is activated, the compression transform graphic in the dynamics touch screen area will be highlighted in green.

INPUT CHANNEL > TOUCHSCREEN > DYNAMICS BLOCK



Compressor Attack

Adjust the compressor attack time.

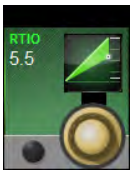
The time it takes for the compression function to reach full attenuation after the threshold has been reached. The slower the attack time, the 'softer' the compression.



Compressor Release Time

Adjust the compressor release time.

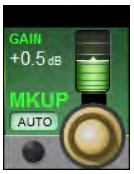
The time it takes for the compression function to return to zero attenuation after full attenuation has been reached.



Compression Ratio

Adjust the compression ratio.

The ratio between the normal signal level and the attenuated (compressed) signal level. The higher the ratio, the more compression is applied.



Compressor Makeup Gain

Adjust the post-compression makeup gain.

Gain applied after the compressor to account for level lost in compression.



Compressor Automatic Makeup Gain

Activate automatic makeup gain.

When activated, this will apply makeup gain after the compressor automatically, depending on the settings of the Threshold and Ratio controls.



Limiter threshold

Adjust Limiter threshold.

Threshold at which the limiter is activated.

INPUT CHANNEL > TOUCHSCREEN > DYNAMICS BLOCK



Limiter In

Activate Limiter

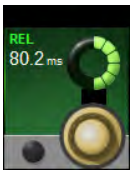
Include the Limiter in the dynamics section.



Limiter Attack Time

Adjust the Limiter attack time.

The time taken for the limiter to reach full attenuation.



Limiter Release Time

Adjust the Limiter release time.

The time taken for the limiter to return to zero attenuation after limiting.

INPUT CHANNEL > TOUCHSCREEN > BUSSES BLOCK



The Busses section of the input channel vistonics touch screen signal path provides access to all Aux and Group contributions in the VST area, The Touch Screen block graphic shows activated Aux levels in orange and group routings in Green. The controls vary depending on whether you are working on Auxiliaries or groups.



Auxiliary Send Level

Adjust the send level from the input channel to the numbered Aux Buss.

The Orange bars in the buss touch screen areas show also aux send levels. The control display includes post / pre-fade routing of the aux buss.



Auxiliary Send On

Enables the numbered aux send from the input channel.

The corresponding aux level bar in the buss touch screen area is highlighted orange when the aux send is enabled.

INPUT CHANNEL > TOUCHSCREEN > BUSSES BLOCK



Stereo Aux Send Balance

Adjust balance of the numbered stereo aux send.

The Stereo symbol above the encoder shows that this aux send has been switched to stereo mode. The Pan for the send will become visibke in place of the send level if the global PAN mode switch is activated in the Master Bay.. If the aux has been set to follow the input channel panning, there will be no balance control visible when PAN mode is activated.



Group Buss Routing

Route the input channel signal to the numbered group buss.

The corresponding group assign bar in the buss touch screen area is extended and highlighted in blue when the numbered groups buss routing is enabled for this channel.



Stereo Group Buss Routing

Route the input channel signal to the numbered stereo group buss.

The Stereo symbol above the encoder indicates that this group has been paired with its neighbour to create a stereo group buss that follows channel panning. The neighbouring group display will be blank.



Buss 'Blank'

Shows this buss has been assigned as a Matrix buss.

INPUT CHANNEL > OUTPUT BLOCK: PAN, INSERT, DIR OUT, DELAY



This page contains the output functions of the input channel including the panning, the routing to the mix masters, and the insert point and the direct out parameters. The Pan can work in LR or in LCR mode. In LCR mode an additional width function is available.



Panning

Adjusts panning to the Master L, R, and C busses, depending on pan mode chosen.

If the channel is paired, the balance can be adjusted. If the Pan {MODE} is set to LCR, an additional WIDTH field becomes available.



Panning On

Activate or deactivate the panning controls.

If PAN is set to OFF, then the level to all master busses is the same as if pan controls are 'middled'. That is -3dB for mono channel panning and 0dB for paired channel balance. If PAN is set to ON then the level to the master busses varies between 0dB and -infinity. For balance that becomes +3dB to -infinity.

INPUT CHANNEL > OUTPUT BLOCK: PAN, INSERT, DIR OUT, DELAY



LR On

Routes the channel signal to the Left and Right masters.



Centre On

Routes the channel signal to the Centre master bus.



Panning Mode

Set either LR or LCR Mode.

In LR Mode only Pan is available, though routing to the C master bus is still possible. With LCR mode enabled, the width parameter becomes available.



Insert Patch

Select and assign physical I/O, internal FX processor, or BSS DPR 901 ii EQ for the insert send and return.

Insert points provide a send and return path that interrupts the channel signal flow.

In any Insert Select page, all available insert points can be patched to physical I/O by touching the on-screen (SETUP) button. You can return to insert assignment by touching the (SETUP) button again. In Insert SETUP mode you can choose send and return I/O from any of the available physical inputs and outputs, and you can pair inserts.

You can assign input channel inserts to any unused insert point or to an available internal FX unit or BSS DPR 901 ii EQ. If you try to select an insert point that has already been assigned, you will be offered the option to 'steal' that insert.



Insert Point

Select the point in the input channel's signal path for the insert point 'tap'.

Options are post-HPF/Pre-processing, post-processing/pre-fader, or post-fader.

INPUT CHANNEL > OUTPUT BLOCK: PAN, INSERT, DIR OUT, DELAY



Insert Trim

Adjust the actual send signal level.



Insert On

Activate the insert point.



Direct Output On

Activate the Direct Output for this input channel.



Direct Output Gain

Adjust the gain of the channel's Direct Output.



Direct Output Point

Select the point in the input channel's signal path for the Direct Output 'tap'.

Options are pre filters, post-filters/Pre-processing, post-processing/pre-fader, or post-fader.



Direct Output Patch

Select and assign physical I/O for the direct output.

Opens a patching screen in the corresponding Vistonics touch screen area.

INPUT CHANNEL > BSS DPR 901 ii DEQ



The console has 16 internal BSS DPR 901 ii Dynamic EQ processors, assignable to the insert points on input channels, output busses, and the Master LCR mix outputs. To assign a 901 EQ, touch the channel's output block in the Vistonics touch screen, then choose INSERT to access the assignment screen.

If a BSS 901 is assigned to an Input channel or output buss, the processing block will appear next to the Dynamics block in the touch screen.

The BSS 901 is a four-band parametric Dynamic Equaliser which allows frequency-selective compression and expansion. It's essentially a parametric EQ where the boost/cut is controlled by a threshold-driven, compression/expansion control parameter. Time constants are controlled by the dynamic and harmonic content of the signal, though there is a FAST/SLOW Release for each band. An ABOVE or BELOW threshold mode allows compression or expansion actions to be targeted at quiet or loud signals, making the processor an invaluable problem-solving tool.



DEQ Band In/Out

Turn that DEQ Band On or Off.



DEQ Frequency

Adjust the centre frequency for this DEQ band. .

INPUT CHANNEL > BSS DPR 901 ii DEQ



DEQ Sidechain Solo (SC Solo)

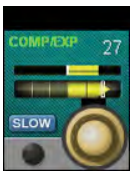
Listen to just the DEQ sidechain via the Solo buss / monitoring system..



DEQ Q / Bandwidth

Adjust the Q or Bandwidth of the filter.

The Q control can be selected between Q-factor and Bandwidth in Octaves, using a setting in the Menu-Settings page (see section 16.9).



DEQ Compression / Expansion

Adjust the compression / expansion amount for this DEQ band.

When the threshold has been crossed (EQ band activated) this band will either compress (cut) or expand (boost) the selected band, according to time constants, up to the amount set by this control.



DEQ Release Fast / Slow

Set the release time for this band.

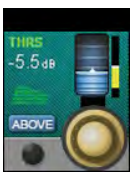
When the band is deactivated, the compression or expansion will reduce over the Release time period, back to zero compression or expansion. A fast Release means that the signal will return to normal faster.



DEQ Above/Below Threshold

Choose the active ‘polarity’ of this DEQ band.

If this parameter is set to ‘ABOVE’ the band will become active (compression / expansion triggered) when the sidechain signal rises above the threshold. If this parameter is set to ‘BELOW’ the band will become active when the sidechain signal falls below the threshold. Choose ABOVE mode to deal with problems that only become apparent at high signal levels, and BELOW mode to deal with problems at low levels.



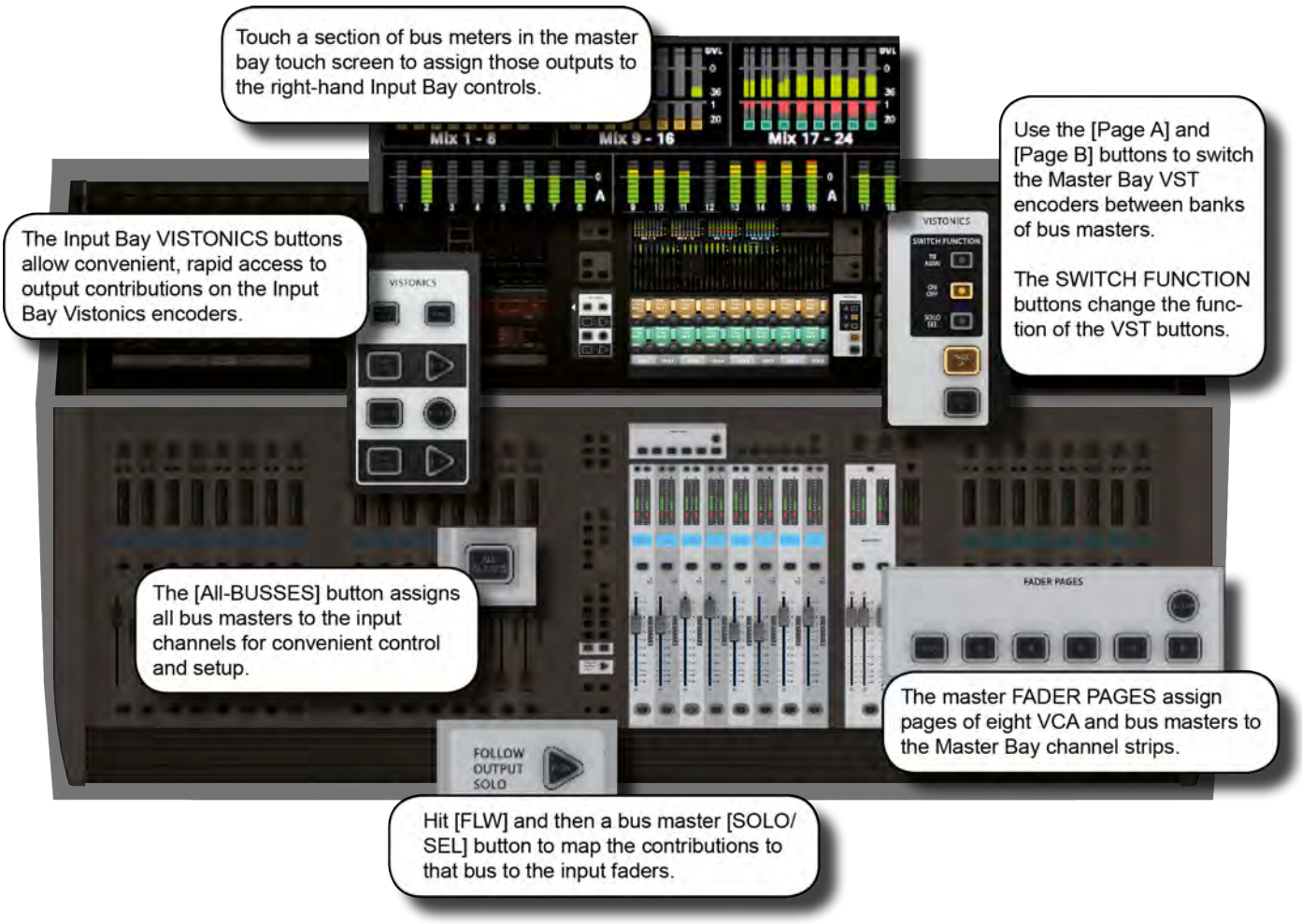
DEQ Threshold

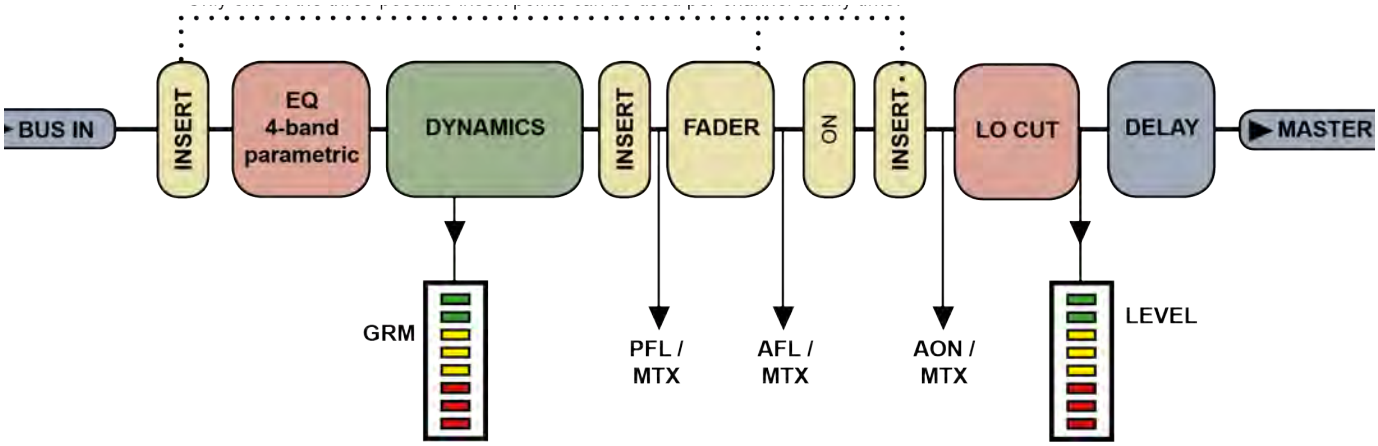
Set a threshold level for this DEQ band.

The threshold is the level at which this DEQ band will become active or inactive, depending on the ABOVE/BELOW parameter. The Threshold meter indicates when the signal crosses the threshold by changing colour to red, assisting adjustment of correct threshold setting.

OUTPUTS

Auxiliaries, Groups, and LCR Mix busses





OUTPUTS

There are 32 configurable output busses on the console, plus the Master Left, right, and Centre (LRC) busses. You can choose to assign any of the 32 busses as mono OR stereo Auxiliaries or Groups with full processing. You can create stereo busses with no reduction in the total number of available busses (32 busses are available regardless of whether they are mono or stereo).

Up to 16 busses can be configured as mono or stereo Matrix (MTX) outputs with full processing, each with up to 24 or 32 (Vi 5000, Vi7000 respectively) configurable sources (including busses) - effectively creating up to 16 24- or 32-source mixes. Most of the references in this chapter refer to the Auxiliary and Group assignments. For the full Matrix system reference please see the Matrix reference chapter 8.

In normal operation (Output Fader Pages A, B, C, and User views) bus master output controls are mapped to the VST encoders in the Master Bay Vistonics screen over two pages. Use the [PAGE A] and [PAGE B] buttons to switch between those pages.

OUTPUTS > LCR MIX MASTER BUSSES



LRC Mix Outputs

The LRC Bus Master faders are located in the MASTERS section in the console Master Bay. These are dedicated faders and controls, and are never used for anything else.

To access the LRC Bus mastering page, simply press the MASTERS [SEL] button. The processing page will appear on the Master Bay Vistonics touch screen area.

The LRC Master busses can have their Parametric and/or Graphic Equaliser sections linked for easier adjustment. See below for more detail.

OUTPUTS > LCR MIX MASTER BUSSES



Isolate

Isolate the LRC Masters from snapshot recall or Show Recall (with global filtering active).

See the Snapshots / Global Filtering reference for more detail.



On

Bus On/Off (Mute).



LCR Select

Access the LCR Master Busses processing page.

Gives access to the processing blocks in the Master LRC outputs via the touch screen area of the Master Bay Vistonics.



LCR Fader

Control the output levels of the LRC Masters

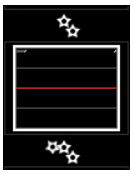
The output levels of the LRC Master outputs are always controlled by their dedicated faders. Those faders are never used for anything else.



L/C/R Link setup

Set up LRC Master bus EQ and GEQ linking.

Left and Right busses can be linked, or the Centre bus can be added to the linked L and R so that all three busses can be adjusted together. It is not possible to link Left and Centre or Right and Centre. The default state, loaded with Default Show Files, is for LINK to be set to ON for LCR busses, both EQ and GEQ.

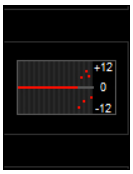


Access EQ

Adjust the parametric EQ for this Master.

The output bus parametric EQ is the same is used for the input channel EQ. Please see the input channel reference for more detail.

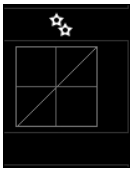
OUTPUTS > LCR MIX MASTER BUSSES



Access L/C/R GEQ

Maps the BSS Graphic equaliser for this block to the console faders.

See the BSS Graphics Equaliser reference later in this chapter for more Detail.



L/C/R Dynamics

Access the L/C/R bus dynamics.

The controls are identical to those of the input channels, except there are no Gate or De-Esser modes available for Master LRC busses.



L/C/R BSS DEQ

Access the BSS DPR901ii Dynamic EQ, if assigned.

The EQ controls will appear in the VST section of the Master Bay touch screen. You can assign any of the available BSS 901 processors to an LCR Master insert point via the output block. See section 6.6 for more detail.



L/C/R Insert/Delay etc (Output Block)

The controls for the Insert/Delay section are similar to those of the input channels.

Extra functions unique to the insert section of the output channels are Low Cut Filter, Phase, and Delay. However, these are the same as those in the Input section of the input channels.

OUTPUTS > BUS CONFIGURATION



To configure the console busses and map the bus masters to the console input channel strips, you simply press the [ALL BUSS] button in the Input Fader Pages controls or assign a bank of eight bus masters to input bay 3 (right hand bay) by touching the relevant metering group in the Master Bay touch screen. You then patch physical outputs and name the busses by touching the Panning/Insert processing block in the touch screen area and pressing the {BUS OUT} and {OUT LABEL} buttons in the VST area.

You can name the output busses and assign the physical output by touching the final PAN/INSERT/DELAY processing block and pressing the {BUS OUT} and {OUT LABEL} buttons in the VST area.

OUTPUTS > BUS CONFIGURATION



Bus Type

Set the Bus Type for this Bus.

Choose from AUX, GRP or MTX with the encoder. If Aux is selected and if the format field is set to stereo then the {CHPAN} field is enabled.



Pan Follows Channel

Set 'Pan follows channel' for a stereo bus.

Set this so the panning into a stereo auxiliary follows the channel panning rather than the Aux panning.

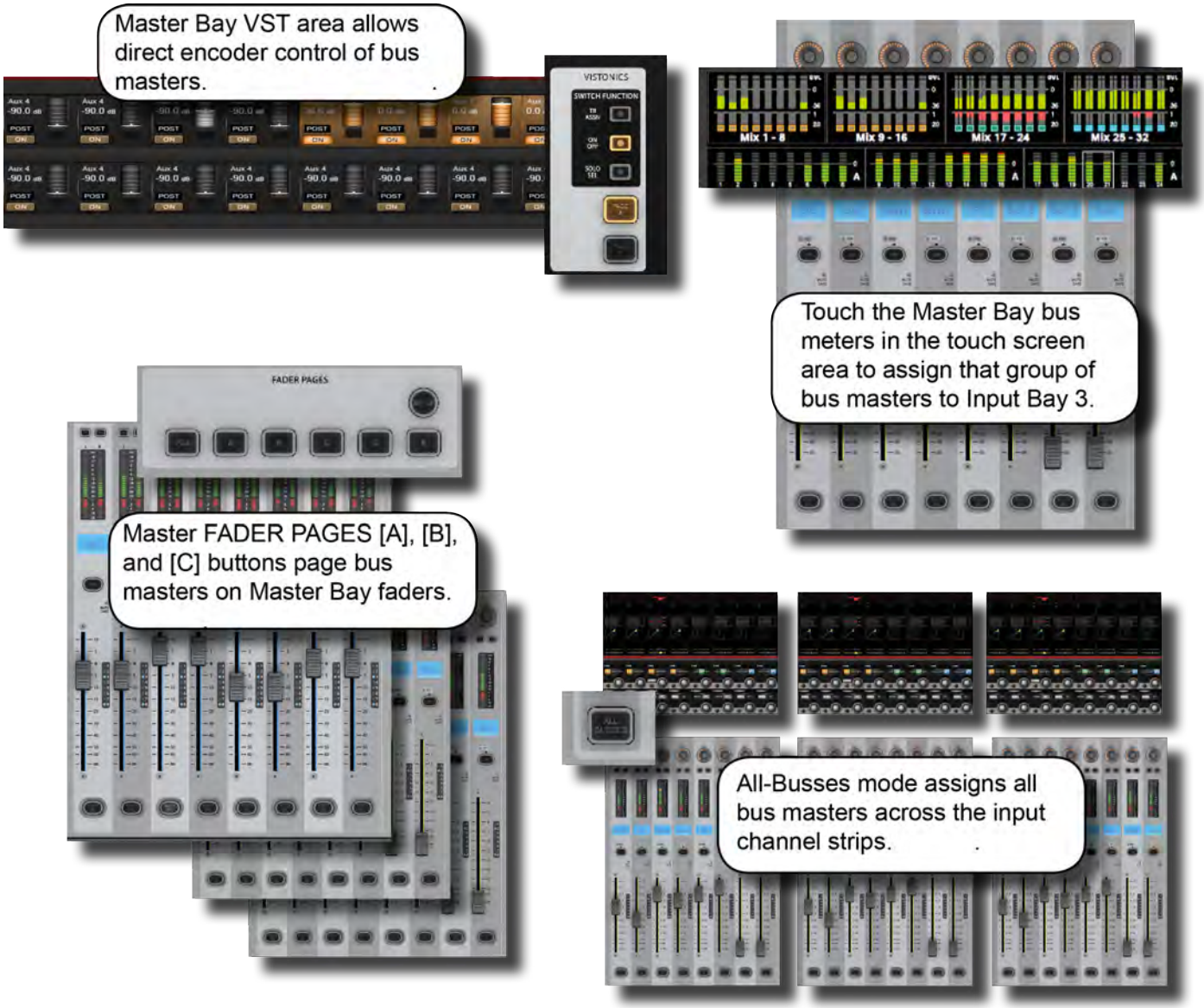


Bus Format

Choose a format (mono or stereo) for the bus.

For Stereo Auxiliary busses, the CHPAN option will become available.

OUTPUTS > BUS MASTER CONTROL



Outputs Controls

There are four ways of controlling the console's bus masters and two ways of accessing the output bus processing and parameters:

Master Bay VST Encoders

Bus master levels are available on the VST Encoders in the Master Bay, with groups of bus masters selected via the [Page A] and [Page B] Vistonics control buttons. The [TB ASN] (Talkback Assign), [ON/OFF], and [SOLO/SEL] Vistonics control buttons select the VST button function.

OUTPUTS > BUS MASTER CONTROL

Master Bay output channel strips

Bus masters levels can be mapped to the eight Master Bay faders and associated channel strip controls by selecting Master Bay FADER PAGES buttons [A] (Busses 1-8), [B] (Busses 9-16), [C] (Busses 17-24), and [D] (Busses 25-32).

All Busses Mode

If the All Busses mode is active, achieved by pressing the [ALL BUSS] button in the Master Bay INPUT FADER PAGES controls, the input strips on all of the input bays will be switched to control the 32 output busses and the faders will control the bus master output levels.

Right-Hand Input Bay - Fast-Select

By touching any of the channel metering areas in the Master Bay touch screen area you assign that bank of eight to the right-hand input bay. Three banks of eight outputs are located on the top row on the Master Bay touch screen.

Bus Parameter Control

With the console in All Busses mode, the console's Vistonics screens give direct touch access to the processing chains of all 32 output buss channel strips.

Alternatively, with the Master Bay Vistonics button function set to [SOLO/SEL], pushing any bus master channel strip [SOLO/SEL] button will display that bus output's processing chain in the Master Bay touch screen area.

In both cases, selecting a processing block will bring that block's parameters up on the VST area of the Vistonics screen.

You can assign Lexicon FX and the BSS DPR901ii Dynamic EQ processors to any output busses via the output (INSERT/DELAY) processing block.

OUTPUTS > MASTERS > VISTONICS



With the Bus Master Controls in the VST fields of the Master Bay, a number of primary functions are available. Use the Master Bay VISTONICS button group to control which functions and which masters are available.



Talkback Assign Mode

Assign the {TB ASSN} function to the bus masters VST buttons.

See the Talkback and Oscillator chapter for more information.



On/Off Mode

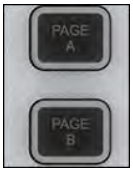
Assign the {ON} function to the bus masters VST buttons.



Solo/Sel Mode

Assign the {SOLO} function to the bus masters VST buttons and let the Bus channel strip [SOLO/SEL] buttons access the bus parameters via the Master Bay touch screen.

OUTPUTS > MASTERS > VISTONICS



Masters Bank Pages

Switch between banks of bus masters in the Master Bay Vistonics VST area.

Page A assigns bus masters 1-16 to the 16 Vistonics encoders, Page B assigns bus masters 17-32 to these controls.



Bus Level

Control the level of the bus output.

May duplicate a fader-controlled bus master, if the ALL BUSSES fader page is active on the input bays, or one of the output fader pages A-D is active on the master bay.



Bus Solo

Solo this bus output.

See the Solo system chapter for more information.



Bus ON

Mute or Unmute this Bus.

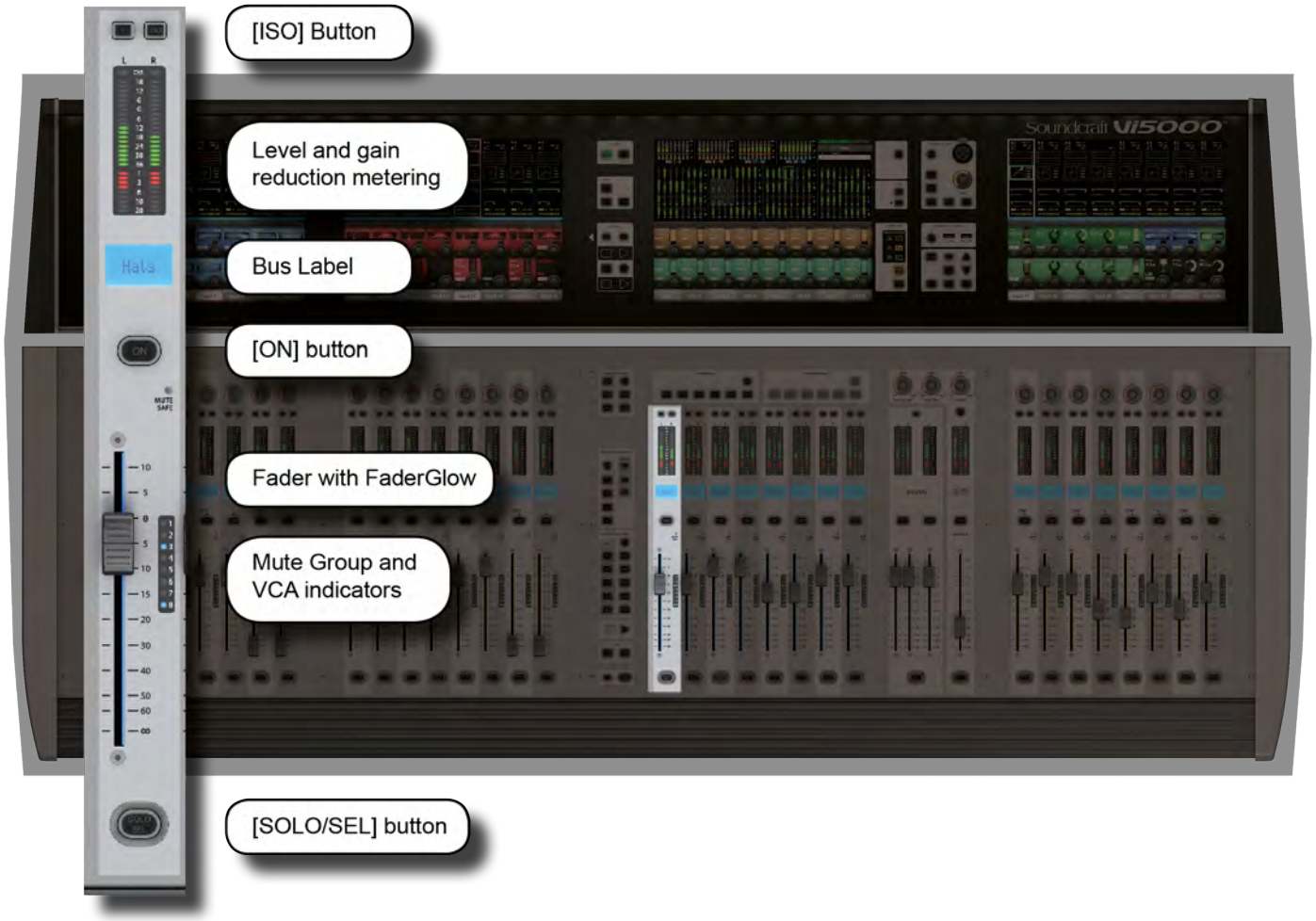


Talkback Assign

Assign this bus to the Talkback system.

When TB ASSN is enabled on this bus, pressing the INT button in the Talkback/Osc section of the control surface will route talkback to this bus. See the Talkback and Oscillator chapter for more information.

OUTPUTS > MASTERS > CHANNEL STRIP



The Master Bay channel strips are for output Bus and VCA control, determined by the Master Bay Fader Pages selectors. The [VCA] button maps VCA masters 1-8 to the Master bay faders. Buttons [A] through [D] map busses 1-32 to the Master Bay channel strips in groups of eight. Buttons [E] maps VCA masters 9-16 to the eight master bay faders. For more information on VCAs, please see the Mute Groups & VCAs reference section 9. However, using the [ALL-BUSSES] mode and Master Bay meter selection input channel strips become bus master strips.

OUTPUTS > MASTERS > CHANNEL STRIP



ISO: Isolate Bus

Isolate this bus from snapshot recall and Show recall (when Global Filtering is active)

See the Snapshot and Global Filtering chapter for more information.



Mute Safe LED

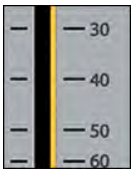
Indicate when Mute Safe is active for this bus.

When the Mute Safe system is enabled (in Monitoring section [SETUP]), then the channel strip SOLO/SEL button activates the Mute Safe status for that channel. When set to Mute Safe, the bus can still be muted by operation of the ON/OFF switch, but cannot be remotely muted by a Mute or VCA group, snapshot automation or Solo-in-Place muting.



Bus Fader

Control the bus master output level.



FaderGlow

The Bus Master fader is colour-coded depending on its function.

The Bus Master FaderGlow colours are Aux Sends: ORANGE; Groups: GREEN; Matrix Contribution: CYAN; VCA Master 1-8: BLUE; VCA Master 9-16: PINK.



Bus MUTE/VCA Display

Shows mute group and VCA group assignment for this Bus Master.



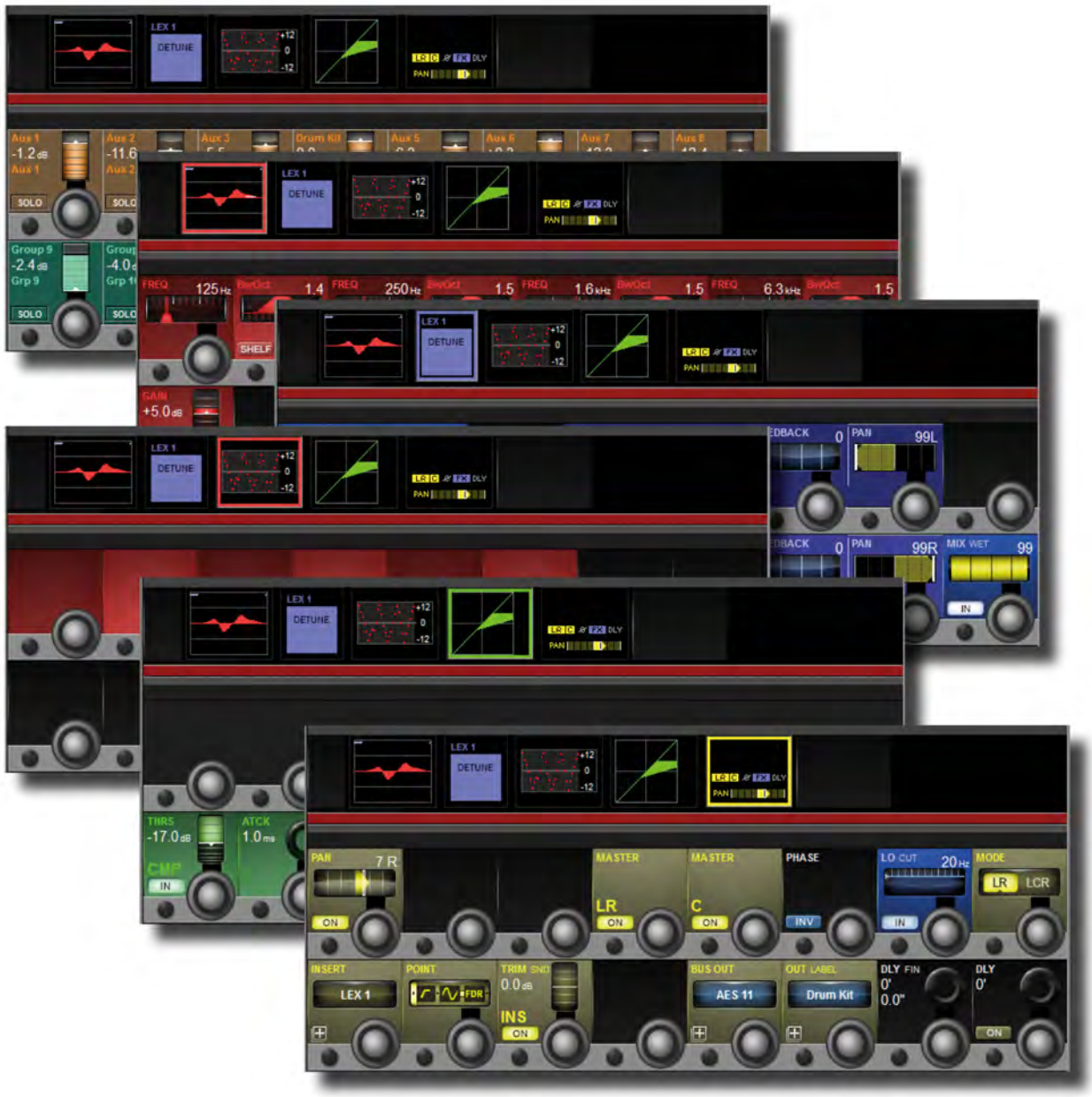
Bus SOLO/SEL

Solos the Bus Master.

Soloing a Bus Master also opens the Bus processing strip on the Master Vistonics screen, allowing EQ, Dynamics and patching to be edited for the selected bus. If FLW mode for faders or Vistonics encoders is active, pressing SOLO will assign the send to the selected bus to the faders or the Vistonics encoders, as appropriate (often described as Sends on Faders mode). The button has other functions in other console modes such as selecting channel for a Copy or Paste edit function. Other functions for the [SOLO/SEL] button are detailed in the appropriate reference sections.

7.3.3: MASTER PROCESSING

OUTPUTS > MASTERS > PROCESS



The bus output processing chain has a parametric EQ block, a BSS Graphic Equaliser block (GEQ), a dynamics block, and a Pan/Insert block. You can also label the bus, assign a physical output, and set a delay inside the Pan/Insert block.

7.3.3: MASTER BUS PROCESSING ▶

OUTPUTS > MASTERS > PROCESS



Bus Channel Strip EQ

Adjust the parametric EQ for this Master.

The output bus parametric EQ is the same is used for the input channel EQ. Please see the input channel reference for more detail.



Bus Channel Strip GEQ

Maps the BSS Graphic equaliser for this block to the console faders.

See the BSS Graphics Equaliser reference later in this chapter for more Detail.



Bus Channel Strip Dynamics

The controls are identical to those of the input channels, except there are no De-Esser modes available for output busses and no Gate or DeEsser modes available if the console is running at 96kHz sample rate.

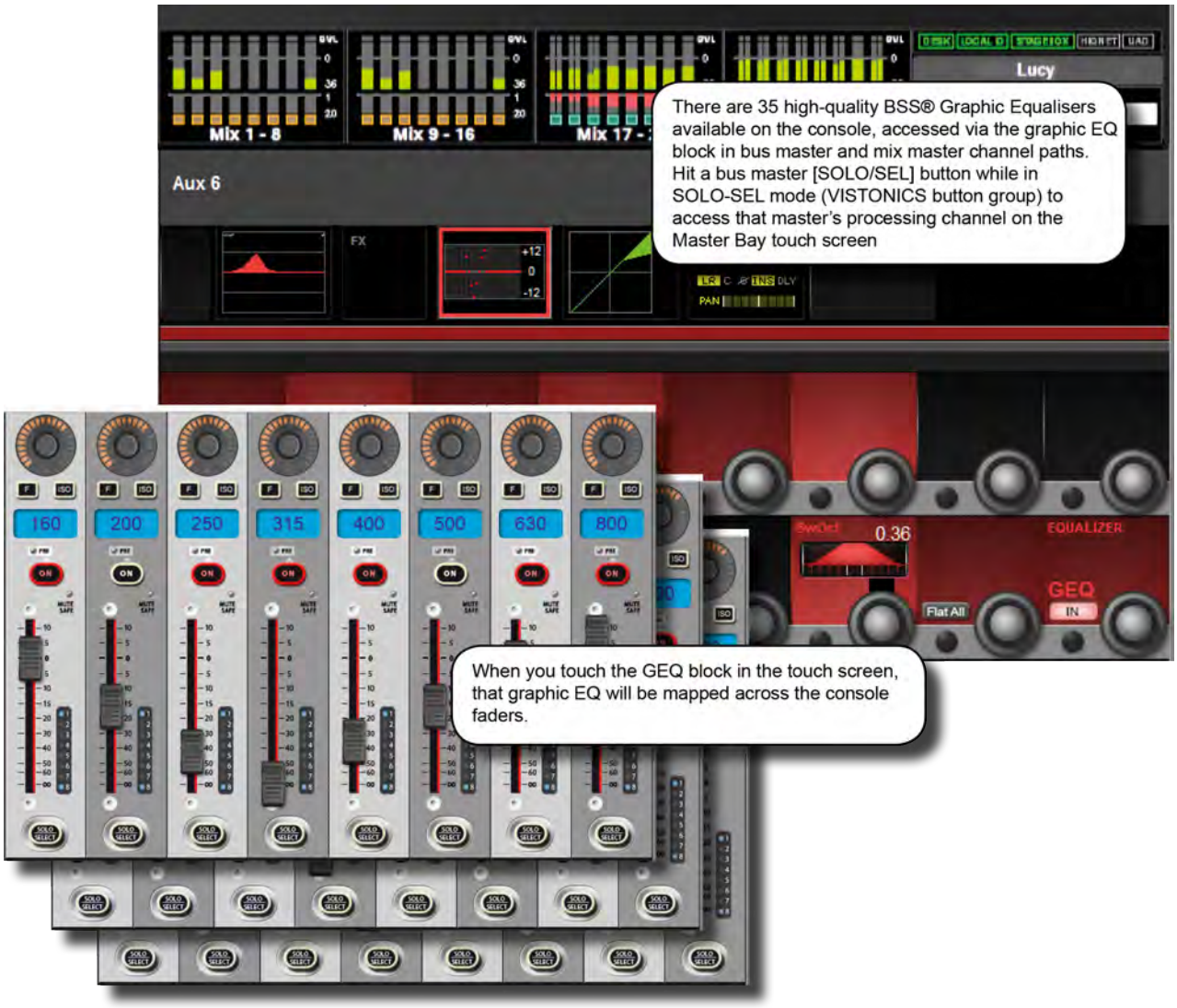


Bus Channel Strip Insert, Delay

The controls for the insert section are similar to those of the input channels.

Patch and name the buss, as well as control panning insert configuration, delay, and more. Extra functions unique to the insert section of the output channels are Low Cut Filter, Phase, and Delay. However, these are the same as those in the Input section of the input channels except that there is up to 1000ms of delay available on busses..

OUTPUTS > GRAPHIC EQUALISERS (GEQ)



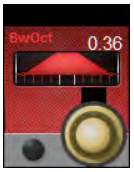
The console has a total of 35 high-quality BSS Graphic Equalisers (GEQ) - one for each of the 32 assignable Busses, and one for each of the LCR Master outputs. The GEQs are all 30-Band EQs with an adjustable bandwidth control.

To access a GEQ, simply select a GEQ processing block from an output's touch-screen parameter display.

Bus parameter displays can be accessed either by entering All Busses mode with the [ALLBUSS] button and selecting the GEQ block from the corresponding channel's touch screen area, or by selecting a bus [SOLO/SEL] button while the Master Bay Vistonics button assign is set to Solo/Select. For more information see the Bus Output reference.

When a GEQ is selected, Bandwidth, Flatten, and In controls are mapped to the right-most VST controls, and the Graphic EQ bands are mapped to the console faders. There are two GEQ control modes. The Large GEQ mode maps all 30 bands across the consoles faders. The Small GEQ mode maps eight GEQ bands to the Master Bay faders - scrollable with the Output Fader Page buttons [A] to [E]. The GEW Control mode is set in the Main Menu Settings Tab.

OUTPUTS > GRAPHIC EQUALISERS (GEQ)



Bandwidth

Adjust the bandwidth of the 30 GEQ bands.

The Bandwidth units can either be Q or Octaves, depending on the {EQ UNIT} setting in the Main Menu (Settings Tab). The narrowest bandwidth is intended for stage-monitor applications, whereas the widest bandwidth is preferred for system tuning for front-of-house applications.



Flatten All

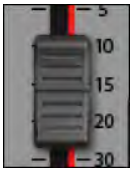
Flatten levels of all bands of the currently active GEQ to 0dB.

Press and hold for 2 secs to activate.



GEQ In

Activate or deactivate the currently selected GEQ.



GEQ Fader On Button

Indicates which GEQ bands are non-zero, and flattens bands.

When you adjust the level of a GEQ band, the corresponding [ON] button is highlighted in red. A subsequent press of the [ON] button will flatten/null the band and turn the [ON] light off.

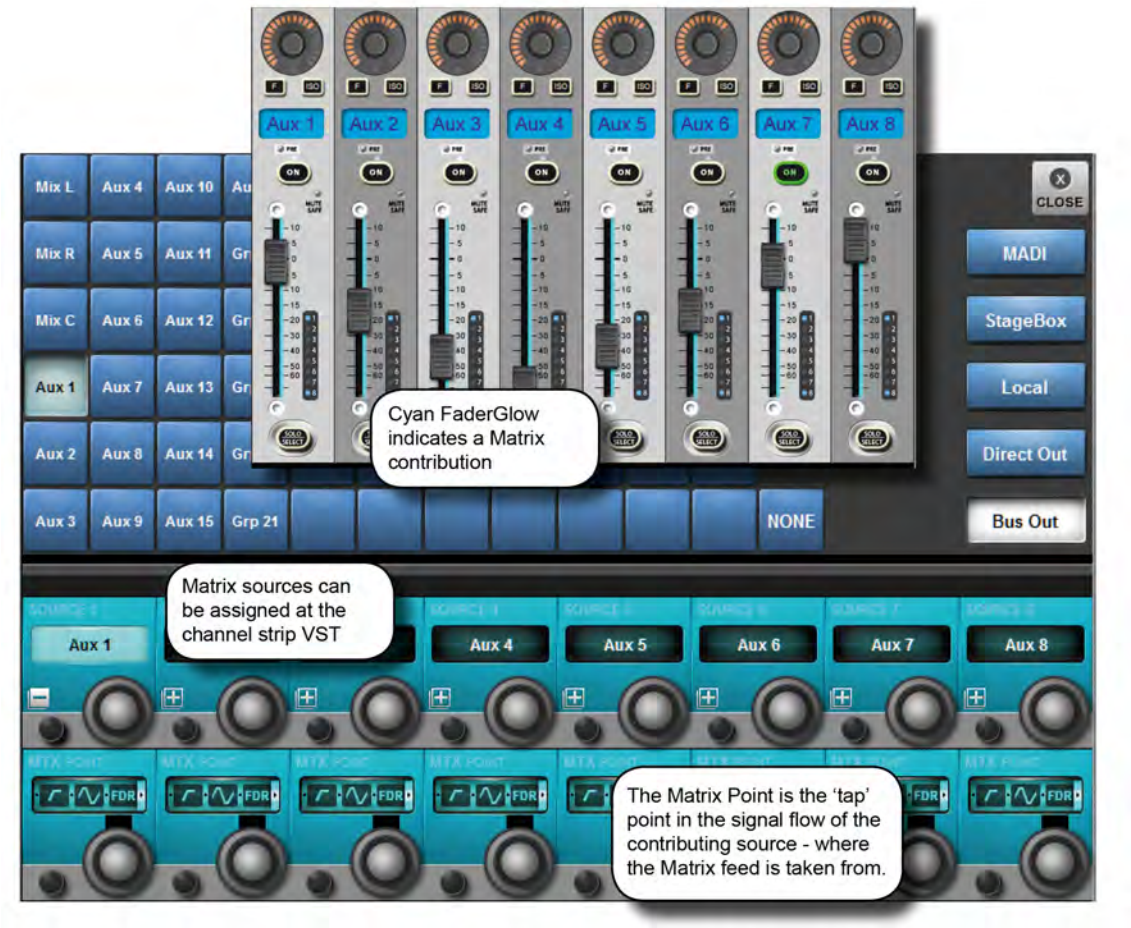


GEQ Fader On Button

Indicates which GEQ bands are non-zero, and flattens bands.

When you adjust the level of a GEQ band, the corresponding [ON] button is highlighted in red. A subsequent press of the [ON] button will flatten/null the band and turn the [ON] light off.

CONSOLE MATRIX SYSTEM



Instead of a simple Output matrix, the console has a built-in, freely-configurable matrix that can have up to 16 Outputs (mono or stereo) with full processing including parametric EQ, dynamics, delay, and graphic EQ. Each matrix output is a mix of up to 24 (Vi5000) or 32 (Vi7000) configurable sources that can be signals from busses, channel direct outputs, or inputs directly from the patching system.

The sources for each matrix mix can be individually patched, so the matrix can be utilized as 16 different mixers, each with up to 24 or 32 inputs, with output processing.

In order to use a matrix, the output bus must first be configured as a matrix output. This can be done easily either via All-Busses Mode (press the [All BUSSES] button in the Input Fader Pages section) or by touching the relevant meter group in the Master Bay touch screen. In this case, the buss masters and assignments will be mapped to the right-hand input bay.

Matrix Control

To map matrix contributions to the input bays, first ensure that the [ALL BUSSES] key is NOT selected, then Press the [SOLO/SEL] on a matrix master. All the input bays will change to the matrix contributions view and Faderglow™ illuminates with the matrix colour (cyan).

Press the Matrix source '+' Vistonics button in one of the input screens above the fader you wish to use for the source. This opens a patching screen that allows the source to be chosen from the categories on the right side of the screen: MADI, Stagebox, Local, Direct Out or Bus Out.

Once the source is selected, its level to the matrix bus out will be controlled by the channel fader and [ON] button. With some types of source (Direct and Bus Out) is possible to choose the point in the signal path of the source channel or bus from where the feed to the Matrix bus will be taken.

CONSOLE MATRIX SYSTEM



Matrix Source ON

Turn on (or off) the corresponding matrix source.



Matrix Source Fader

Adjust the level of this source as a contribution to the matrix mix.



Matrix Source Solo/Sel

Cancels the Matrix mapping and returns channels to normal operations.

The Solo/Sel button on a Matrix contribution strip always remains as an input channel solo for the channel that was previously on this strip before Matrix solo was activated.



Matrix Pan Encoder

Pan control for stereo Matrix sources.

If the Matrix bus is stereo, the Pan control will appear above each Matrix Source fader allowing the selected source to be positioned within the stereo image of the Matrix bus. If the Matrix is mono, this control will have no function.



Matrix Source

Choose a source for this matrix input.

The button opens up a patch selection screen with all available matrix sources, including output busses, physical console inputs, and input channel direct outs.

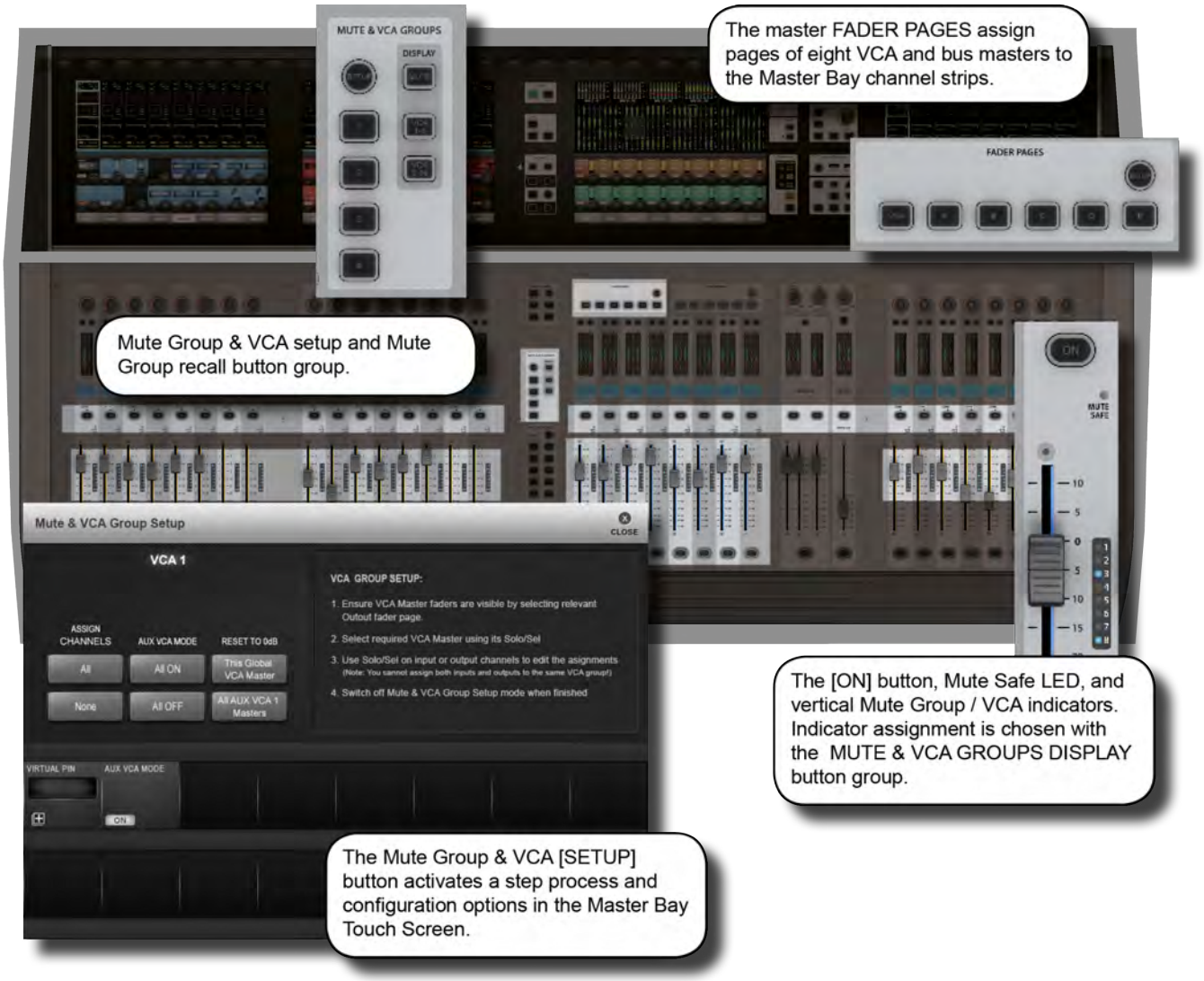


Matrix X Point

Select the point/tap where the selected matrix source is taken from.

For example, selecting a buss master as a matrix source offers the choice of pre-fade, post-fader, or post-ON.

MUTE & VCA GROUPS



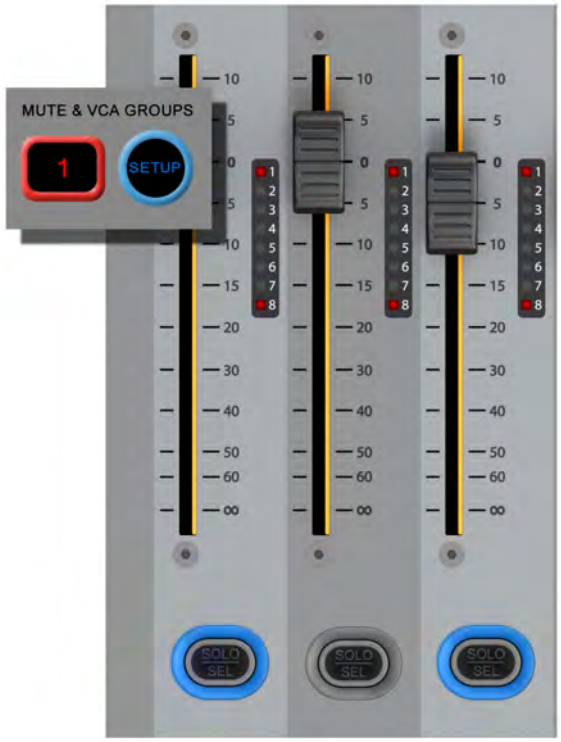
The console supports up to 4 mute Groups (MG) and 16 primary VCA masters that can include both input and output channels.

There is a secondary level of VCA functionality, most useful for monitor engineers, which allows the user to assign up to 16 VCA masters to each auxiliary mix when the auxiliaries are controlled for the channel faders. This means there are 33 possible sets of 16 VCA groups. A more detailed reference for this functionality is included later in the this chapter (section 9.3).

Each input and output strip has a VCA/mute group display next to its fader made up of eight LED indicators. These show which mute and VCA groups each channel belongs to in three modes: mute groups 1-4, VCAs 1-8, and VCAs 9-16. The mode selection for the indicators is selected in the DISPLAY section of the MUTE & VCA GROUPS buttons located on the left side of the Master Bay fader panel.

MUTE & VCA GROUPS > MUTE GROUPS

Mute Groups Operation



There are four console mute groups, activated with the [1], [2], [3], and [4] buttons in the MUTE & VCA GROUPS button group. Both input and output channels can be assigned to mute groups.

When a channel mute is activated by a mute group, the channel's ON button glows red.

Mute group assignment is started by pressing the MUTE & VCA GROUPS [SETUP] button (see below).



Mute Setup

Press the MUTE & VCA GROUPS [SETUP] button, then chose a Mute Group button to configure that group.

The [SETUP] button will glow blue when pressed. The chosen mute group master button - [1], [2], [3], or [4] - will glow red when you select that group. Press the [SOLO/SEL] key of any channels (input and/or outputs) to be assigned to the selected mute group, then press [SETUP] again to exit the configuration mode.



Mute Group 'n'

Press to activate / select a Mute Group.

In normal use, the mute groups selectors will activate the chosen mute group (mute those channels). In setup mode they select a mute group to be set up.

MUTE & VCA GROUPS > VCA GROUP OPERATION

VCA Groups configuration is started by pressing the MUTE & VCA GROUPS [SETUP] button. When you press a VCA Master [SOLO/SEL] button you can assign channels to that VCA group with their individual [SOLO/SEL] buttons, and also access VCA group options - including labelling - in the Master Bay Vistonics. See below for more information.

You can select EITHER output channels OR input channels to be part of a normal VCA group.

In order to assign VCAs, the VCA masters [SOLO/SEL] buttons have to be made available by selecting the appropriate Master Bay Fader Pages layer - [VCA] for VCA masters 1-8, or [E] for VCA masters 9-16.

VCA Master level at setup

The assignment of VCAs should be done with the channel or group faders and the VCA master, to which assignment is being made, being at or near a nominal operating level. That is, don't assign a VCA master to a channel or group if the VCA master is at -40dB while the channel or group is at 0dB. There are 0dB buttons for VCA masters available in VCA group configuration mode.



VCA and 'E' Buttons

Fader Pages buttons for mapping VCAs to the Master Bay faders.

The [VCA] button maps VCA masters 1-8 to the eight Master Bay faders. The [E] button maps VCA masters 9-16 to the eight Master Bay faders.
VCA Masters 1-16 can be assigned to any Output Fader page from A to E, along with Bus Output faders using the User Defined fader page setup. The VCA page cannot be remapped and always contains VCA 1-8.



VCA Masters [ON] Buttons

Master On/Off button for all of the VCA's member channels.

If a channel that was previously on is turned off by the action of a VCA Master [ON]/Off button, the channel's [ON] button will illuminate in red to distinguish this condition from a manually 'off' channel.
The VCA Master On/Off switch itself has only 2 states, and always illuminates red when off, and green when on.



VCA Group Fader

An offset dB level for all channels that are members of the VCA Group.

- When a VCA master fader is moved, the contributing channel fader levels remain unchanged.
- If a channel is assigned to more than one VCA Group, the resultant offset applied to the channel is calculated as the arithmetic sum of the dB values of each VCA Master.
- Any VCA master fader reaching –infinity dB (fader at lowest position) will set all member channels to –infinity dB, regardless of other Master fader settings.
- The maximum gain applied to a member channel as the result of the channel fader setting plus offsets from VCA Master faders is limited to +10dB.



VCA Master Solo/Sel

A master Solo button for all of the VCA's member channels.

Depending on the setting of the AUTO button in the Monitor Setup page (SOLO section), activating a VCA Solo will result in PFL (if the group only has one member channel) or AFL (if the group has more than one member channel). Otherwise the VCA solo will follow the setting of the PFL/AFL switch in the Monitor Setup page.

MUTE & VCA GROUPS > VCA GROUP OPERATION



VCA Setup

Press the MUTE & VCA GROUPS [SETUP] button, then chose a VCA [SOLO/SEL] to configure that VCA group.

The [SETUP] button will glow blue when pressed. The [SOLO/SEL] button of the chosen VCA Master will also glow blue. You can then press the [SOLO/SEL] buttons on any channels you wish to make part of that VCA group. Then either press the [SETUP] button again to exit the configuration mode, or press another VCA Master [SOLO/SEL] button to configure a new VCA group.
During configuration of VCA groups, a number of options become available in the Master Bay touch screen and VST area (see below).



Bulk Assign VCA

Select all input channels or no channels as members of the VCA Group.

Select the VCA Group you require by pressing its SOLO/Sel button first.



Aux VCA Mode Bulk On/Off

Switch all VCA Masters in the Master bay between Aux VCA Mode and Normal mode. (See Aux VCA Mode reference).



Reset VCA Masters

Set VCA Masters to 0dB.

Either set the currently selected master to 0dB, or all aux send VCA masters (see Aux VCA Mode reference).



Label VCA Group

Opens up a standard labelling keyboard in the Master Bay touch screen area for labelling the VCA master.

Identification colours can also be chosen from the labelling keyboard.

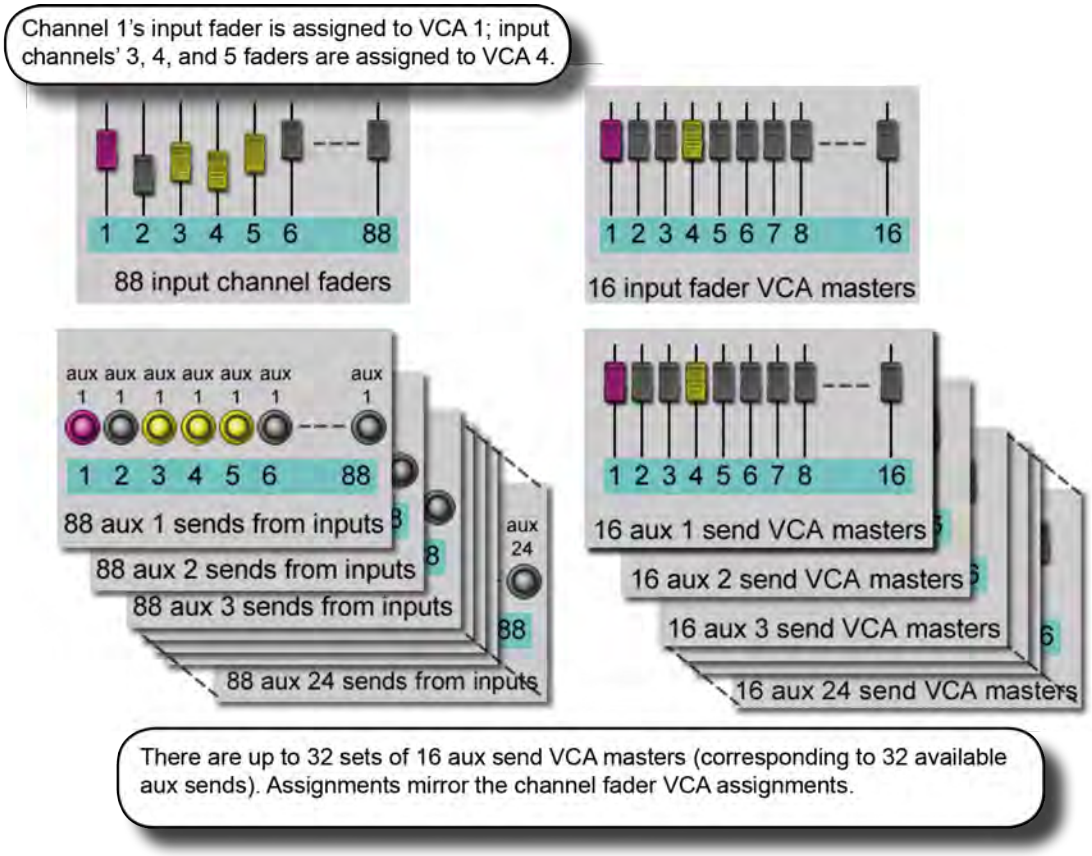


Aux VCA Mode [On]/Off

Set the currently selected Aux VCA Master on or off (see Aux VCA Mode reference).

When in fader [FLW] mode and an Aux master is selected, VCA Masters set to 'ON' will control the Aux VCA group, not the 'normal VCA group. Aux VCA group master faders are highlighted with white FaderGlow, while 'normal' channel VCAs are highlighted with blue FaderGlow.

MUTE & VCA GROUPS > AUX VCA GROUPS



The VCA Groups on the console are normally used to control groups of the input channel faders, for use by FOH engineers. In this case they are used to group mono and/or stereo channels together under control of a single master fader, for easier control during a mix.

For the Monitor mix engineer, controlling channel faders is of secondary importance to controlling Aux sends from channels, and so the functionality of VCA Groups has been extended to control of groups of Aux sends as well as channel faders.

Because there can be up to 32 Auxiliary sends configured on the console, this means that there are up to 32 sets of VCA groups (each with up to 16 Group Masters), in addition to the set of main channel fader VCA groups.

VCA control of Auxes is only available by activating the Follow Output Solo [FLW] key next to the master faders. The pair of [FLW] keys for the VST Encoder Rows will not access this function.

It is possible to control only one Aux mix via its VCA masters at any one time.

When setting up Auxiliary VCAs, it is possible to turn the Aux VCA masters mode on or off for each VCA Master, so 'normal' VCAs can still be available.

MUTE & VCA GROUPS > AUX VCA GROUPS

Auxiliary VCA Procedure

1) Select the Aux bus required by selecting the required master fader page [A]-[D].

2) Activate the [FLW] key next to the master faders.

3) Press the [SOLO/SEL] key under the required Aux master fader (or use the solo switches in the Master VST area).

4) Press [VCA] (for VCA 1-8) or [E](for VCA 9-16).

5) The FaderGlow for any active VCA master faders will change to white and the faders will move to show the offsets being applied to the Aux feeds from those channels assigned to the VCAs. At this point, the VCA Master [SOLO/SEL] keys have no function. The VCA [ON] switch controls the Aux send On/Off on the member channels.

6) To select another Aux for adjustment, press the required bank key [A]-[D], solo the required aux (or solo directly on the VST screen), and press [VCA] or [E]. To exit, press [SOLO CLEAR] and de-select [FLW].

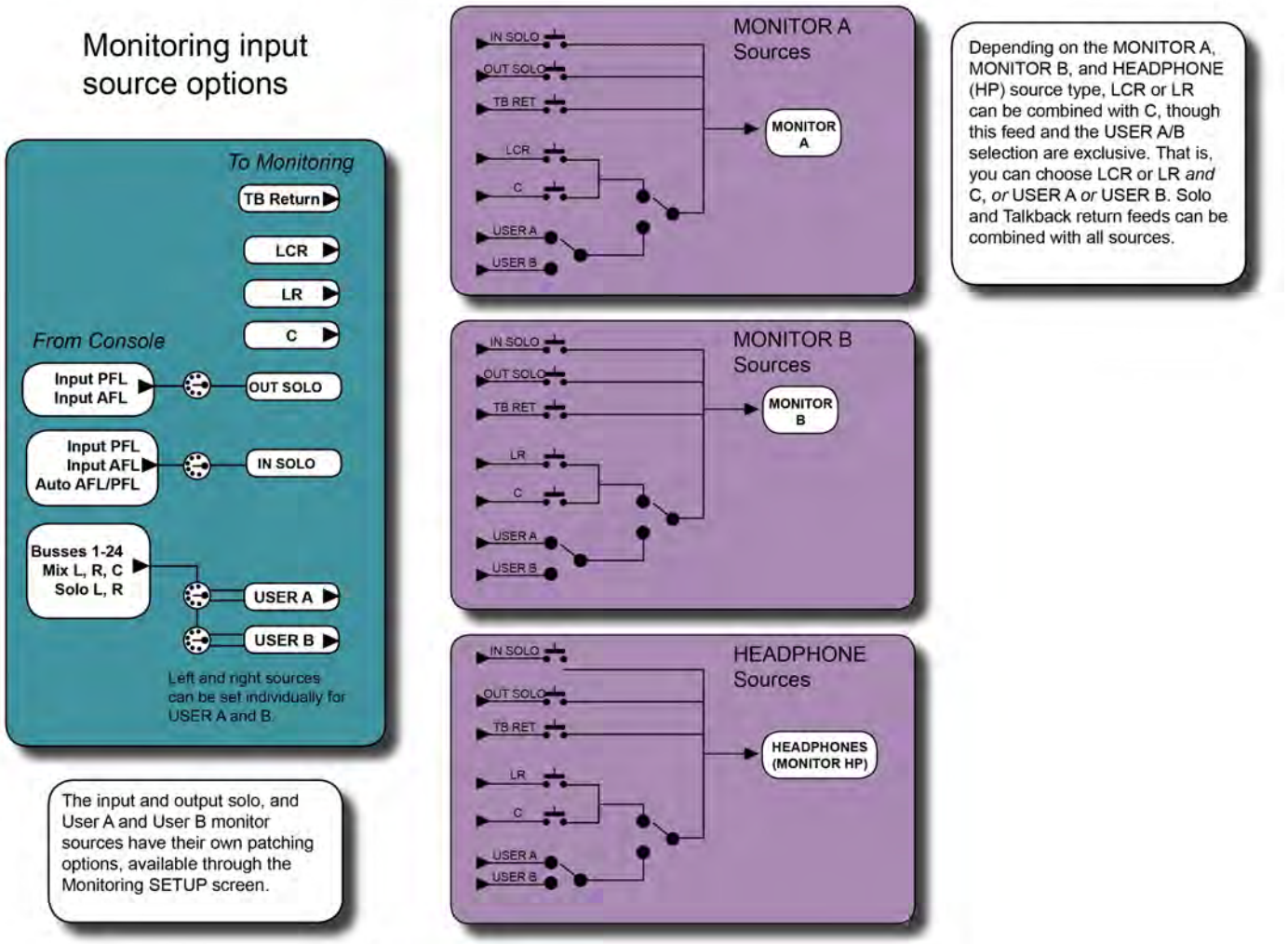
Note: To set up Auxiliary VCAs, press the MUTE & VCA GROUP [SETUP] button after step 4 and assign member channels to that Aux VCA group as per the normal VCA group setup procedure (use the channel [SOLO/SEL] buttons to choose members, then press [SETUP] to exit). Note that Aux VCA Groups have to share the same input assignments as the ‘normal’ channel VCAs.

MONITORING OVERVIEW

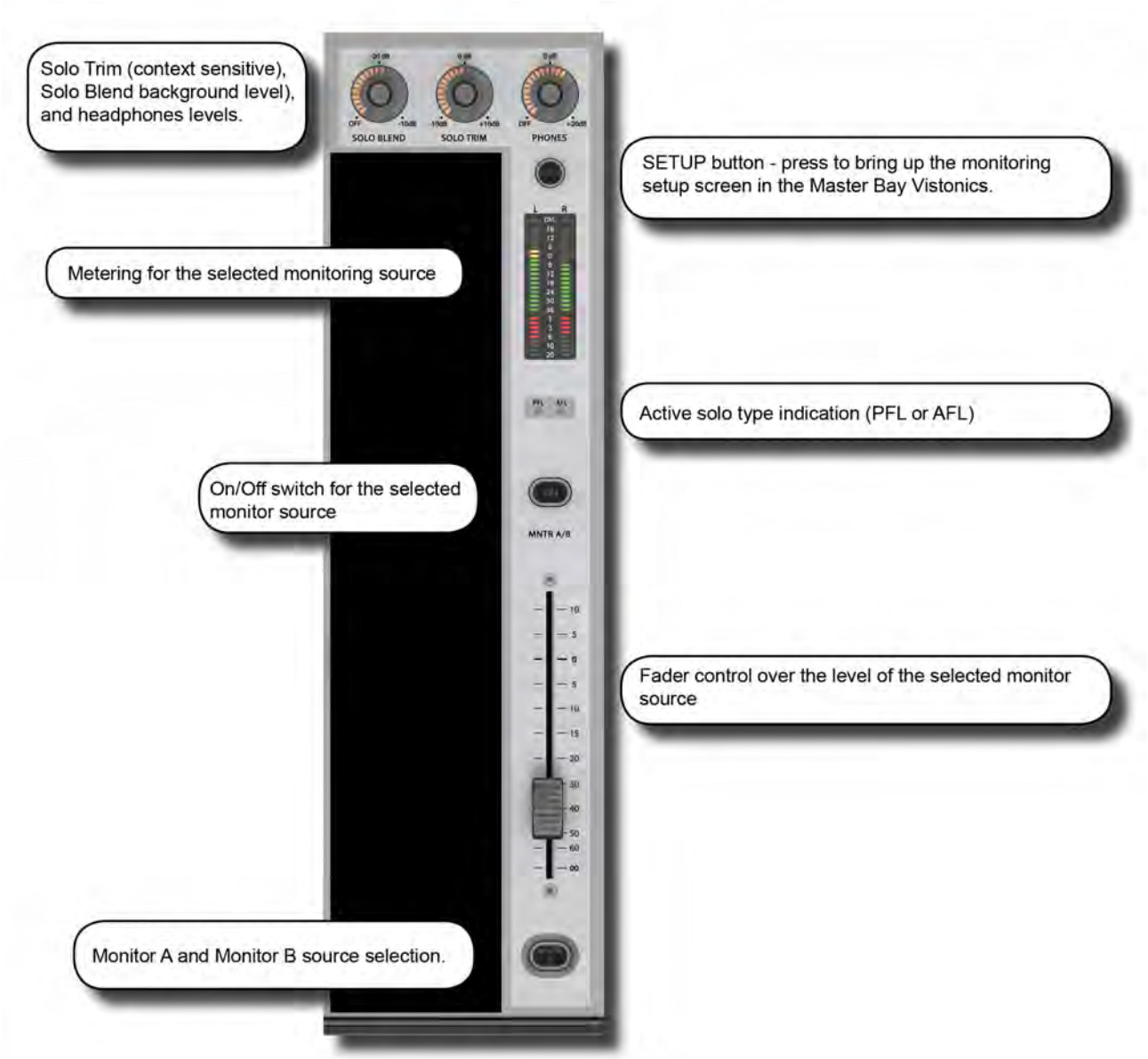


The monitoring system has three main sources - Monitor A, Monitor B, and Headphones. These are fed by a choice of sub-sources, including Solos, LCR, Talkback return, and two user-defined stereo sources where the left and right signals can be patched independently

MONITORING OVERVIEW



MONITORING > CONTROLS



The following console monitoring controls are located to the right of the master faders in the Master Bay, except for [INPUT PRI] (Solo Input Priority), which is located with the SOLO button group on the upper section, to the right of the Master Bay.

There are three main monitoring outputs - A, B, and headphones. Monitor A is an LCR output, Monitor B is an LR output, and the headphones output is LR to the headphone socket on the front of the console. You can select a variety of sources for the monitoring outputs, including two user-definable sources, USER A and USER B, with separate left and right assignments.

Solos can be blended with the main monitor feed for contextual solos, and an Auto function can be set to select PFL or AFL depending on your solo selections.

Solo system control is described in the Monitoring SETUP section. The console's Solo system logic and general use is described in the next chapter (11).

MONITORING > CONTROLS



MNTR B - Monitor A/B Select

Select either the A or the B monitor to be displayed on the meter, to be controlled by the fader, and to be switched on and off by the [ON] switch.

Both Monitor A and B sources are always active - (respecting their individual On/Off status'). The MNTR B button changes metering and control assignment.



Monitor Fader

Control monitoring volume level.



AFL and PFL indication

These two LEDs show if an active solo is a PFL or an AFL.



Monitoring Setup

Display the Monitor Setup Page on the master screen.

The [SETUP] key glows blue when it is active.



Headphones Level

Control the headphones volume level.

The headphone output socket is located under the armrest at the front of the console, and is designed for headphones with impedances in the range 50-600 Ohms.



Solo Trim

Trims source SOLO Level.

This control is SOLO context sensitive. It is possible to set a different trim for each of the 32 outputs in addition to a global input solo trim level.

MONITORING > CONTROLS



Solo Blend

Adjust the background level of the monitoring source signal.

The level of the non-solo source adjusted from OFF to -10dB to allow solos to be heard in the context of a mix that has a reduced level.



Solo Input Priority

Activate Input Priority Mode

Allows an Output Solo to remain active, whilst an Input Solo is temporarily activated 'over the top' of it. When the Input Solo is deactivated, the Output Solo's audio will return to the Solo Bus. For more detail, see the Solo System reference chapter 11.

MONITORING > SETUP



All monitoring source assignments for MONITOR A, B, and HP are assigned in the monitoring setup page, as well as monitoring delay.



Monitoring Input Solo Source

Choose either input channel PFL, AFL, or 'AUTO' as the Input Solo source.

The AUTO option works by automatically selecting PFL when only one solo is active and AFL when more than one solo is active.



Output solo source

Choose either output buss PFL or AFL as the Output Solo Source.

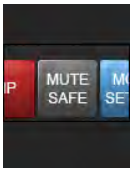
MONITORING > SETUP



Solo-In-Place Mode

Activate the SOLO-IN-PLACE mode.

This is a destructive mode for use only during sound-checks or rehearsals. When a channel is soloed in SIP mode, all other channels are muted so that only the soloed channel is heard, in its stereo position, at the console's mix outputs.



Mute Safe Mode

Enable the Mute Safe (SIP isolation) configuration mode.

If (MUTE SAFE) is active the Mute safe state from the Input channels can be toggled with the Channel's SEL Key. The state for a given channel is indicated by that channel's Mute Safe LED. NOTE: This configuration mode is disabled when the setup page is exited. When a channel or bus is set to Mute Safe state, the channel can still be muted by manual operation of the ON/OFF button, but it is not possible to mute the channel by Mute or VCA group operation, snapshot automation or Solo-in-Place of another channel.



MON SETUP Mode

Activate the Monitor Setup sub-page.

The Monitor Setup sub-page parameters become available in the VST area of the master screen. See below for more details.



Auto Cancel Mode

Intelligent auto-cancelling of Solos, depending on Input Priority [INPUT PRI] Mode.

With Input Priority OFF, input and output solos cancel other active solos of any type. With Input Priority ON an input solo will cancel active input or VCA solos and temporarily override active output solos, and pressing another output solo will cancel other active output solos.
Autocancelling can be overridden by holding down an input or output solo and pressing other solos of the same type. See Section 11 for more detail on Input Priority,



LCR Source Select

Select the mix master output sources for the monitor outputs A, B, and HP

For the Monitor A (LCR) output you can choose LCR and/or C; for the Monitor B and Headphone outputs you can choose LR and/or C. This selection cannot be combined with the User A or User B sources, but solo and talkback options are concurrent.



User A / B Source Select

Select the user-defined output source for the monitor outputs A, B and HP.

Choose EITHER User A or User B as a main source for the monitoring output. This selection cannot be combined with the LCR, LR, or C sources, but solo and talkback options are concurrent. Each of the monitoring outputs has its own individual USER A / B assignment. User A and B are useful in stage monitoring applications to allow a favourite output bus to be the default monitoring source when no solos are active.

MONITORING > SETUP



Input Solo Select

Route the Input Solo Signal to the Monitor A, B, and headphone outputs.

The Input Solo signal is defined in the SOLO section of the monitor setup screen. Each of the monitoring outputs has its own individual IN SOLO assignment.



Output Solo Select

Route the Output Solo Signal to the Monitor A, B, and Headphone outputs.

The Output Solo signal is defined in the SOLO section of the monitor setup screen. Each of the monitoring outputs has its own individual OUT SOLO assignment.



Talkback Return Select

Route the Talkback Return Signal to the Monitor A, B, and Headphone outputs.

Each of the monitoring outputs has its own individual Talkback Return assignment.
Routing talkback return to the monitor or phones outputs allows a continuous feed of a return TB signal to be heard, regardless of solo or other souce activation, and therefore can be used as a 'shout' feed without needing to use an input channel.



Monitor Follow A

Force the Monitor B and/or Headphone source assignments to be the same as the MONITOR A assignment.



Monitor output format

Selects one of the stereo format options to be the listening format for the Monitor A, B, and Headphone outputs.

LR - left source to left monitor out and right source to right monitor out • RL - left source to right monitor out, right source to left monitor out • LL - left source to left and right monitor outputs • RR - right source to left and right monitor outs • Mono - left and right source is summed and fed to left and right monitor. The Centre signal is not affected. Each of the monitoring outputs has its own individual FORMAT selection.



Monitoring delay - Fine

Fine Control of the monitoring delay time at 0.02ms per step.

MONITORING > SETUP



Monitoring delay

Adjust the monitoring delay.

The monitor delay allows the headphones and/or monitor speakers to be time-aligned to the output from the main PA system, when working at a distance from the PA speakers. The delay applies to all three monitor outputs (Monitor A, B, and Headphones). Up to 2 seconds of delay is available. Resolution is 1ms per step.



Monitoring delay ON

Activate/Deactivate the monitoring delay.

MONITORING > MON SETUP



Mon Setup is a sub-screen of the Monitoring setup screen for setting up fundamental configurations such as physical patching and naming for the User sources.



User A and B Patching

Assign the input, individually, for the User A and User B monitoring sources.

When selected, the source assignment screen is displayed on the touch screen. You can assign any available input, individually, to the left and right User channels - including console I/O, group busses, mix buss channels, and solo channels.



User A and B Labelling

Label the User A and B monitor sources.

When selected the keyboard input screen is shown on the touch screen. You can relabel the 'USER A' and 'USER B' via the screen keyboard or a connected USB keyboard.

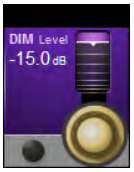
MONITORING > MON SETUP



Outputs Solo Assign

Assign individual Output Solos to Monitor A, B, and Headphones outputs.

When selected, an output assignment screen is displayed in the master touch screen area. You can select individual output buss solos for the specific monitor output. This is useful for stage monitoring applications where the solos of in-ear monitor mixes could be programmed to appear only on Monitor B for example, which could have an in-ear headphone system connected to it. In this example, conventional wedge monitor mixes could be programmed to appear only on Monitor A when soloed, where Monitor A could be connected to a wedge speaker system.



DIM Level

Adjust the monitoring DIM Level between 0 and –infinity.

The DIM function is activated if the Return Talkback function is activated via the GPIO facility or any of the front-panel Talkback activation switches are pressed..

SOLO SYSTEM BEHAVIOR



Solo management in the Monitors SETUP screen - [SETUP] button above MNTR A/B fader.

AFL, PFL and SIP

The solo system on the console comprises a non-destructive (solo bus routing only) PFL and AFL Solo capability for Inputs and Outputs, and also the option of a 'destructive' Solo-In-Place (SIP) mode (channel muting solo), for use only during sound-checks or rehearsals. Solo-In-Place mode has to be enabled from within the Monitor Setup and changes the mode of operation of the Solo System. It is automatically cancelled if the Monitor Setup screen is closed, to ensure that it cannot be left on accidentally.

PFL and AFL Soloing (SIP Mode OFF)

• Soloing a single Input locally generates a PFL or AFL Solo onto the Solo Bus, depending on the mode selected in the Monitor Setup page (default is AUTO, see next point).

• If the Input Solo Mode is set to AUTO in the Monitor Setup page, soloing more than one Input locally (by pressing and holding the first Solo/Sel switch then pressing others, or switching off AUTO CANCEL mode) will generate an AFL Solo onto the Solo Bus from all soloed Inputs. If a single output is soloed, the solo will be PFL.

• The PFL feed from Mono Inputs is taken after the channel Pan control so the signal heard on the Monitor outs will reflect the Pan setting. If the Input is a Stereo Input, the PFL feed is left channel to left Solo Bus, right to right.

• The AFL feed from Mono or stereo (paired) Inputs is stereo and follows the channel Pan.

• Soloing an Output (either locally or via a VCA Master Solo) generates an AFL or PFL Solo, depending on the mode selected in Monitor Setup (default is AFL for outputs). The post-fade Output signal is switched onto the Solo Bus.

• In all cases when SIP Mode is OFF, operation of any Solo will switch the audio onto the Solo Bus and the Monitor section will be automatically switched so that the Solo audio replaces the previous monitor source selection (if any) assuming that IN Solo and/or OUT Solo have been selected as a Monitor Source in the Monitor Setup Page.

Solo-In-Place Soloing (SIP mode ON)

• Soloing an Input generates a 'destructive' SIP Solo, actually muting or dimming all other Inputs which are not Soloed or set to Mute Safe, as if you had pressed their channel ON switches. This has the effect of soloing on the main outputs. Other channels which are subsequently soloed will be unmuted. The amount of dimming is controlled by the SOLO BLEND control on the front panel.

• Soloing an Input (or group of Inputs) by soloing a VCA Master that the input is assigned to, generates a SIP Solo on all Inputs in the VCA Group.

* The Input signal is not switched onto the Solo Bus, and the Monitor section does not switch the Solo Bus audio to override the monitor source selection (you hear the soloed signal(s) via the Master LRC busses directly.

• Soloing an Output generates a normal Output AFL or PFL Solo (according to the mode selection made in Monitor Setup), the same as if SIP Mode was OFF. The Output signal is switched onto the Solo Bus and the Monitor section switches so that the Output. Soloed audio is heard on the Monitors, replacing the previous monitor source, if any.

SOLO SYSTEM > SOLO LOGIC

Solo Operation Logic

Activating Solos



A Solo is activated when any Solo/Sel switch on the console is pressed, as long as the following is true:
• Gang Mode is not active
• VCA or Mute Group Setup page is not open
• Input or Output Fader Page Setup page is not open

Clearing Solos



• All active Solos can be cleared by pressing the momentary Solo Clear switch in the Master Section of the console. This switch illuminates when any Solos are active.
• Solos can be switched off manually.
• Solos can be cleared by pressing other Solos, if AUTO CANCEL mode is enabled (see later in this chapter).

Input Priority Mode



When Input Priority is enabled, via the INPUT PRI button in the master bay, it allows an Output Solo to remain active, whilst an Input Solo is temporarily activated 'over the top' of it. When the Input Solo is activated, its audio replaces the Output Solo audio on the Solo Bus, although the Output's Solo/Sel switch remains illuminated. When the Input Solo is deactivated, the Output Solo's audio will return to the Solo Bus.
HINT: Input Priority mode is normally used by Monitor engineers, who tend to work with an Output Solo always active, but occasionally need to solo an input to troubleshoot a problem. The Input Priority mode ensures that they automatically return to the Output Solo they were listening to, after the Input Solo is deactivated.

Autocancel Mode



In the majority of cases in live sound mixing, only one channel is soloed at any time, so it has become common practice for solos to 'autocancel' so that pressing any solo cancels the previous one, and only one solo can be ON at any time. This speeds up operation by eliminating the need to switch solos off before soloing the next channel. In some applications however, such as line checking, it is necessary to keep one channel soloed as a shout talkback feed whilst soloing others, so additive solo is required in these cases. The {Autocancel} On/off switch in the Monitor Setup page allows the behaviour to be set according to requirements. It is also possible to solo more than one channel or bus at a time even if {Autocancel} mode is ON, this is achieved by holding down the first solo whilst pressing the next.
This kind of additive solo is temporary however, and the next solo to be pressed will cancel all of the active solos.

Input priority and Autocancel mode work together if both enabled so that Output Solos cancel among themselves and input solos cancel among themselves, but Input Solos do not cancel Output solos or vice versa.

Follow Output Solo Mode



A key feature of modern digital consoles is a method of assigning the channel faders to control Aux send levels for monitor mixing. This mode is called 'Follow' mode, meaning that the Input faders will Follow the selection of the Output bus solo, which doubles as a bus select switch.
Follow mode is engaged using the distinctive, triangular Follow Output Solo [FLW] keys. There are three [FLW] keys: one fader key and two Vistonics area keys. Only one key can be active at any one time (or none can be selected).

SOLO SYSTEM > SOLO LOGIC

Selecting Follow mode for the Vistonics area means that a horizontal row of Vistonics encoders will be assigned across the whole console to the contribution levels of the currently soloed bus. There is one [FLW] key for the upper Vistonics row and one for the lower row. To use Follow mode on Vistonics area, the Vistonics screens must have all processing views deselected.

• If Follow Solo is NOT active for faders or Vistonics encoders, pressing a Group or Aux Output Solo will activate an Output Solo, and it will also display the EQ/Dyn/Misc touch screen area for the Soloed Output, on the Master section touch screen (in the space normally occupied by the Input Meter overview display).

• If Follow Solo IS active for either faders or Vistonics encoders, pressing a Group or Aux Output Solo/Sel will work as described above, and will also switch the input channel faders or encoders to be assigned to the Soloed bus's contributing sends (equivalent to 'Sends On Faders' on some other consoles). The Faderglow colour will illuminate in this case to indicate the type of bus contributions being assigned to the input faders: Orange for Aux sends and Green for Group sends (although the faders themselves have no function with Group sends - only the channel ON keys are used).

• If a Matrix Output Solo/Sel is pressed, regardless of the setting of Follow Solo modes, the Output Solo will be activated, the EQ/Dyn/Misc touch screen for the Matrix Output displayed, and the channel faders will be assigned to the contribution levels from the Outputs to the Soloed Matrix Output. The Faderglow of input faders will change to Pale Blue (Cyan) to indicate Matrix contributions are now being controlled.

Whilst an Output Solo is active, the [FLW] key can be pressed at any time to drop out of Follow mode and switch back to channel faders for global level adjustments, and then pressed again to drop back into Follow mode for adjustments to contribution mix again.

The [FLW] key for fader area is automatically switched on when you load the Monitor Default Show, as this functionality is primarily intended for monitor mixing.

Note: the [FLW] key for fader area has an additional function in allowing VCAs to control the Aux sends of input channels. See chapter 9 for details.

CONSOLE METERING

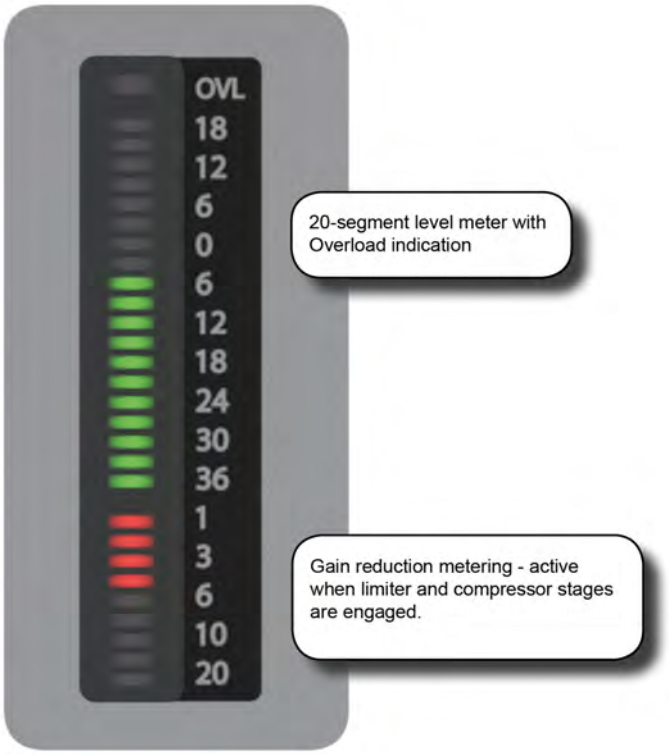


Level metering in the console goes from +18dB to -36 dB. 0dB on the meters represents 4dBu at the analogue output. Gain reduction is displayed in the Range 0 - 20dB. The console uses full floating-point calculations in its DSP, which means that the audio signal inside the mixer cannot be overloaded. If the signal level is too high at the master output meters, it is necessary only to pull down the master fader level until the correct level is obtained.

There are meters on every channel and bus master, plus metering for every console path in the default Master Bay touchscreen areas.

The touchscreen meters can be touched to assign the third (right-hand-end) input bay mapping.

CONSOLE METERING > INPUT METERING



Each input channel has an input meter and every input meter has two parts: A 20-segment level meter (top), and a 9-segment Gain Reduction Meter (GRM).

Both the measured point in the channel signal flow and the metering peak hold time can be selected in the main settings screen (via the [MENU] button), in the (Settings) tab.



Input Level Meter

Shows the input level at the selected point in the input channel signal path.

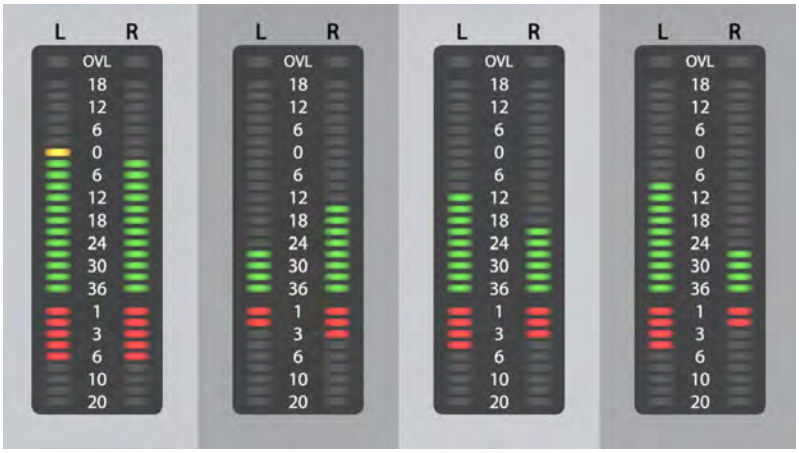
The overload (OVL) LED indicates an overloaded analogue input (mic preamp clipping).



Input Gain Reduction Meter

Shows the overall gain reduction of the limiter and compressor stages, if they are engaged.

CONSOLE METERING > BUS MASTER METERING





Bus Meters

Show the bus master signal levels.

Bus Masters can be configured as stereo channels, therefore the Bus Master Strip level meters have left and right meters (i.e. the left and right channels share a single set of controls). The GRM shows the overall gain reduction of the limiter and compressor, if they are engaged. The Dynamics from Stereo Bus Masters are always linked, therefore both GRMs show the same value.
The overload LED indicates a value that is higher than Full Scale (analogue output-stage clipping).



Mix Master Meters

The L, R and C Output Masters each have a Level Meter and a Gain Reduction Meter. The L and R masters share a stereo meter, and the C master has a mono version.

The overload LED indicates a value that is higher than Full Scale (analogue output-stage clipping).

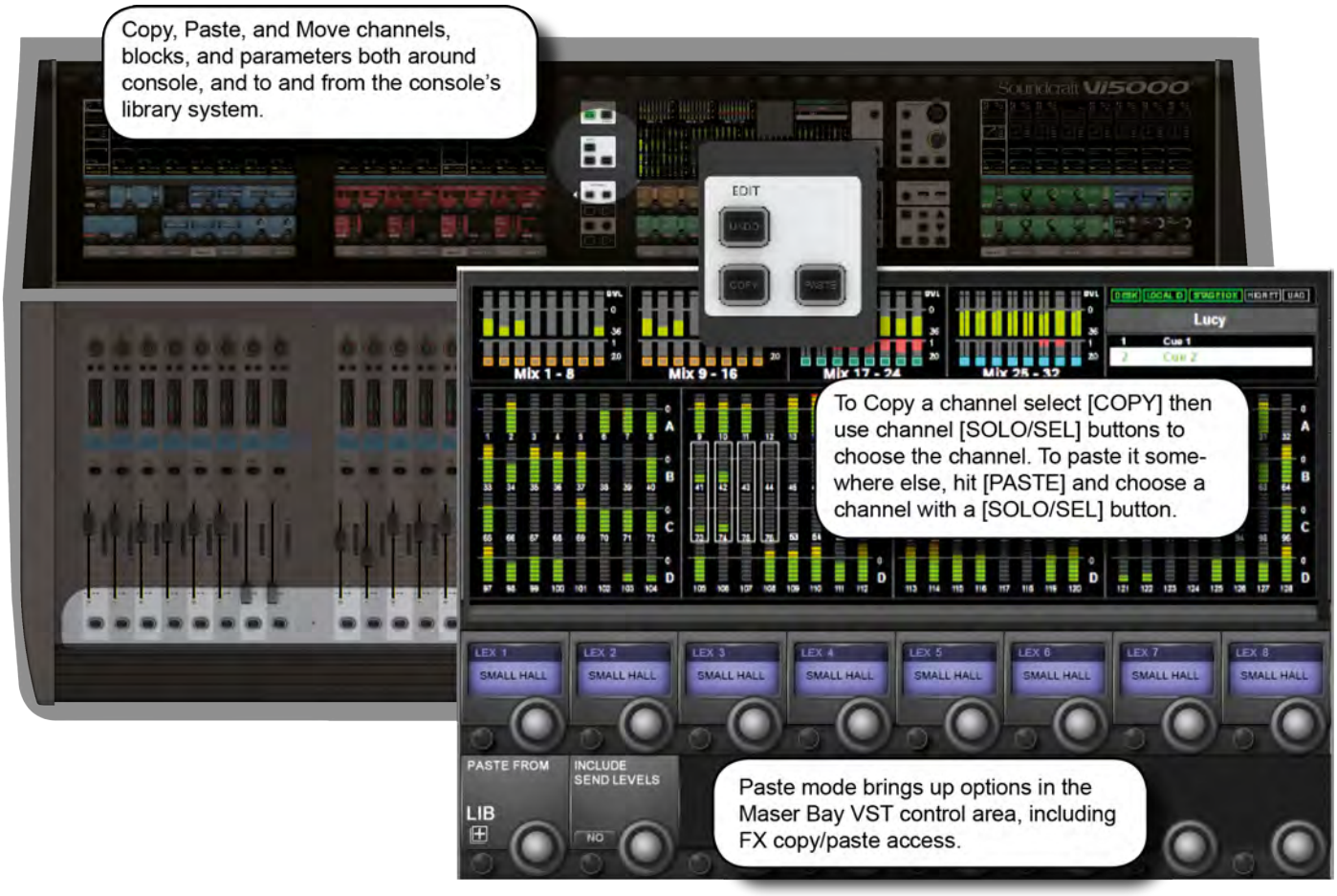


Monitor Bus Metering

The monitor section has a stereo level meter, but there is no GRM associated with the monitor.

The overload (OVL) LED indicates analogue output-stage clipping.

EDIT & LIBRARY: Copy, Paste, Undo, and Library storage



The Copy/Paste function allows the settings of any channel, bus, FX section or processing element to be copied and pasted to any number of other channels, saving set up time and helping eliminate errors.

The Copy and Paste modes also work with the console's powerful Library functionality. This allows you to select any set of parameters in use on the desk and store them in the internal library, independent of the Show Files, and export to or import from a USB storage device.

If the console is in either copy or paste mode a yellow bar appears across all the Input Screens and the console switches into 'Copy/Paste' mode, which is where the surface is used for item selection rather than control. In addition, the VST area of the Master Bay Vistonics displays the Lexicon FX blocks (for convenient selection) and the {LIB} Library access button.

In these modes control of channel strip audio from the Vistonics screens is not possible.

EDIT & LIBRARY > COPY, PASTE, UNDO

Ranges of channels or busses, individual channels or busses, processing blocks (eg EQ), or even individual parameters (eg mic gain control) within a processing block can be copied to the clipboard then pasted to another channel, block, or range. This includes the console's internal Lexicon FX units.

The last paste operation can be quickly reversed with the UNDO function.



Copy

Put the console into Copy mode.

Used for selecting channels, processing blocks, and parameters and copying them to the clipboard. In copy mode, the console [SEL] keys illuminate in blue to indicate their function as selects rather than as Solos.

Copy Options

- Put the console into Copy mode.
- Copy A Channel Or Bus Master

In copy mode, select or deselect a full input channel or output bus by pressing its [SEL] Key. Select a range by holding the [SEL] Key of the first Channel and pressing [SEL] of the last Channel. When a whole channel or range of channels are selected, the grey Channel border is replaced with a yellow border.

Some parameters or process blocks are not included with a channel or bus copy by default, but most can be added to the copy by selecting them individually (see below). The only ones that CANNOT be added are the Insert Point and Direct Output settings.

Parameters not included with a whole input channel copy but that CAN be added individually are: Input 1/2 switch, Input 1 & 2 patch, Mic Input Gain and all associated switches (48V, PAD, Phase Inv etc), Fader and Mute, VCA & Mute Group Assignment, FX settings.

Parameters not included with a whole bus copy but that CAN be added individually are the output patch and FX settings.

• Copy A Block

Select or deselect a function block by touching that block's Touch field. Indication that a whole function block is selected is shown with a wide yellow indicator in the top left corner of the field.

• Copy A Parameter

Select a single parameter by touching the corresponding Rotary encoder or by pressing the small key to the left of the encoder. To make access to parameters easy while in Copy mode, enter the 'Zoom' mode with a long press on a touch field. Alternatively you can enter the Zoom mode, before you activate Copy mode by pressing the touch field in the normal way and entering Copy mode.

A selected parameter is indicated with yellow text in the Parameter Label. Note that not all parameters can be selected individually; in some cases a set of parameters will automatically be selected when one of the set is touched.

If only a set or a single Parameter from within a block is selected then a small yellow indicator is displayed.

EDIT & LIBRARY > COPY, PASTE, UNDO

• Copy FX

To make selection of the source and destination Lexicon FX units easy, a set of eight FX-select encoders appears on the central screen in both Copy and Paste modes. The selection is made by either touching the relevant encoder or pressing the corresponding Vistonics button beside the encoder. If the FX unit is assigned to a channel or bus then the units can be selected by touching the FX block on the channel strip.

Note that only the parameters relating to the currently active FX Type are copied/pasted for each unit. For example, if LEX1 is selected, and is currently set to the Small Hall FX type, then only the Small Hall parameters will be copied.



Paste

Enter paste mode to Paste any channels, processing blocks, parameters, or Lexicon FX settings (selected in Copy mode) to a new destination.

To paste anything selected in Copy mode, press [Paste] to enter Paste mode, then select the destination channel by pressing its [SOLO/SEL] button or touching any VST field on the channel. The clipboard content is immediately copied to each selected channel.

Once you have finished, press the [PASTE] button again to exit Paste Mode and enter normal operation.

IMPORTANT! Remember to switch PASTE mode off immediately after use, otherwise accidental pasting of the clipboard will occur the next time you touch a screen or press a Solo/Sel button.



Undo Edit

Undo the last paste operation.

The Paste button must be switched ON in order to use the UNDO function.

EDIT & LIBRARY > LIBRARY SYSTEM



The Libraries functionality enables various elements of the console, eg EQ section, Dynamics section, or complete channels and busses, to be copied from the desk and stored in the Library system. Later these items can be retrieved and pasted from the Library to any destination channel or bus on the console. You can also export complete Libraries or individual entries to a USB stick, allowing you to move Libraries between consoles.

The {LIB} button appears in the bottom left corner of the VST section in the Master Bay Vistonics screen when in either Copy or Paste mode. It works in much the same way as the basic copy and paste mode. To Paste from the library simply hit the [PASTE] button, select a library item, then touch a target on the console. To Copy to the Library, put the console in Copy mode, select console blocks (see the Copy/Paste section) then hit the {LIB} button. The selection will be saved to the currently loaded Library. To save selections to a different Library, you must load that library - or create a new one - first.



Library Access

Access Library functionality while in Copy (Copy To Library) or Paste (Paste from Library) modes.

The button opens up a patch selection screen with all available matrix sources, including output busses, physical console inputs, and input channel direct outs.



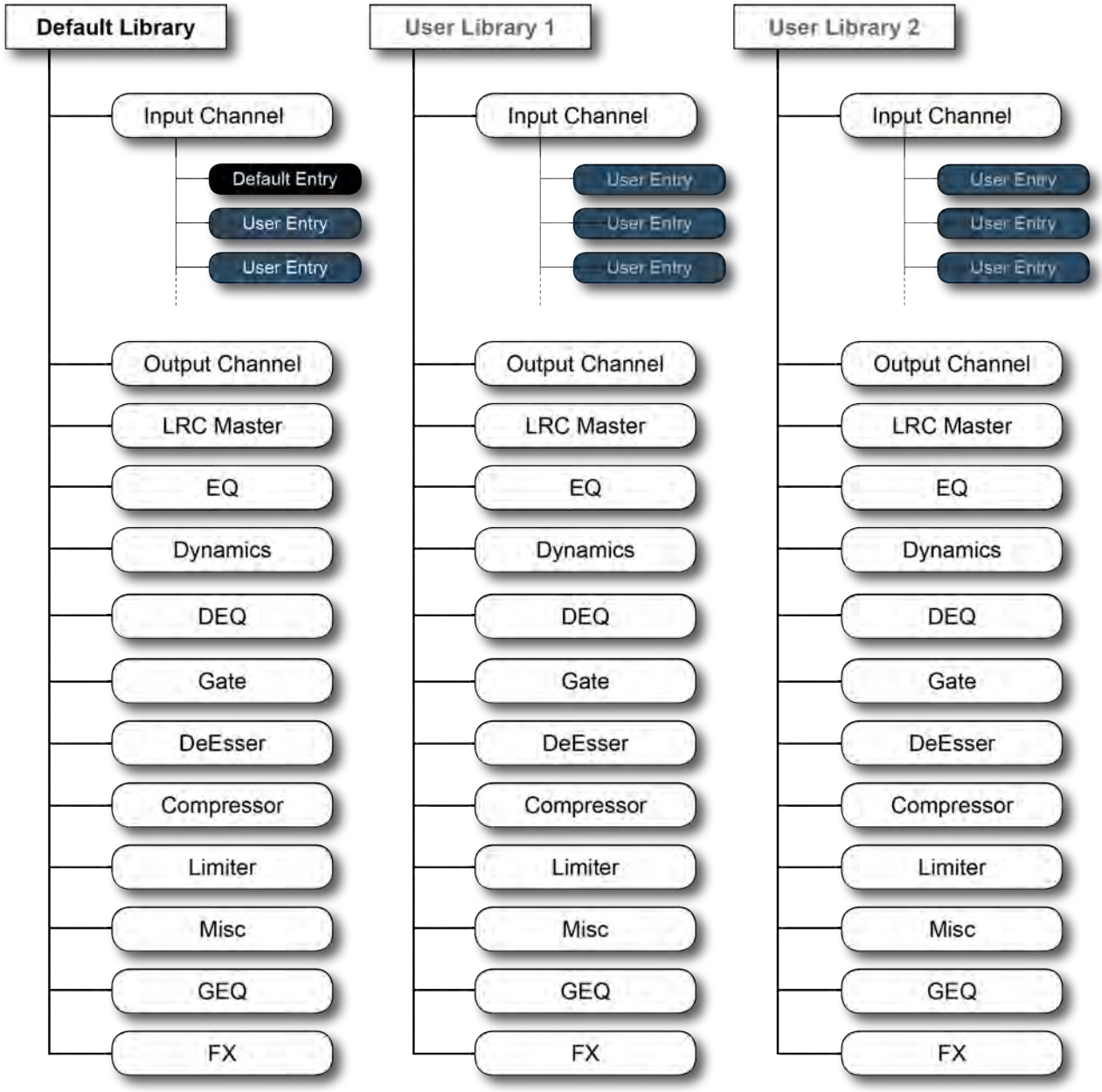
Include Send Levels

Include Send Levels in Library Copy and Paste operations.

13.2.1: LIBRARY FILE SYSTEM

EDIT & LIBRARY > LIBRARY > FILE SYSTEM

The Library file system itself comprises a system of folders that reside on the console's internal flash drive. The folder structure is preset to comprise a top-level Library folder, within which are sub-folders called Categories, corresponding to the various types of functions that can be stored in the Library (EQ, Dynamics, Channels, and so on). Within each Category, the actual Library settings are stored - these are called Entries. This structure and the names of the folders are fixed and cannot be modified.



Only elements that have a Category can be stored individually to the Library – For example, there is no Category for the Input stage (gain, PAD etc) or the Aux send parts of the channel strip, so these parameters cannot be stored to the Library other than as part of complete channel strips using the Input Channel Category.

The Misc Category is intended for storing combinations of channel or bus parameters. Currently, only a combination of EQ and Dynamics can be stored in the Misc category, but this may be extended with a future software update.

EDIT & LIBRARY >

Default Library

The console comes with one pre-installed Default Library. This Default Library contains factory presets for a variety of common applications. These factory default library entries are ‘read-only’ and cannot be deleted.

A mechanism exists for the default entries in the Default Library to be updated via a special update procedure. New, modified or additional default entries may be released from the factory from time to time in the form of a web download that can be copied to a USB stick and transferred to the console.

New Entries can be stored in the Default Library, alongside the read-only factory default Entries, or one or more new Libraries can be created which will then only contain user-stored Entries.

User Libraries

In addition to the Default Library, as many additional new Libraries can be created as required. Either a new empty Library can be created or the existing Default Library can be duplicated using the Save As function, creating a new Library that also contains the factory default Entries. When a new empty Library is created, only the folder structure is created – the Library itself contains no Entries and needs to be populated by the user.

The only limit to the number of Libraries or entries that can be created is available disk space in the console, but the file size of each Library entry is very small (typically less than 100kB for individual single channel Entries).

EDIT & LIBRARY > LIBRARY > FILE SCREEN



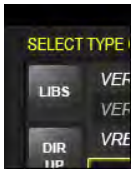
The Library file system appears in the Master Bay Vistonics screen when the {LIB} button is pressed in Copy or Paste Mode. It allows you to navigate the console library and perform import/export operations with connect USB storage devices.

Important! In Copy mode you must have your intended source library loaded before copying.



File Path

Always shows the currently selected directory, and the file path to it.



LIBS

Navigates directly to the Library directory, which is the top level of the folder structure..

EDIT & LIBRARY > LIBRARY > FILE SCREEN



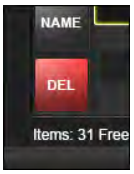
Directory UP

Navigate the file tree upwards, to the next 'parent' directory.



Name Folder/File

Name the currently highlighted Library or library item.



Delete Folder/File

Delete the currently highlighted Library or Library item.

You cannot delete a complete Library that is currently loaded.



Save As

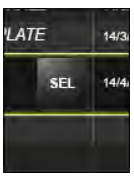
Save a copy of the library under a new name.



New Library

Create a new (empty) User Library.

The {NEW} button only appears at the Library directory level.



Select Folder/File

Select the highlighted item.

Libraries that are currently loaded will have the {SEL} button and be labelled in yellow text. Libraries that are not loaded have the {LOAD} button.

EDIT & LIBRARY > LIBRARY > FILE SCREEN



Load Library

Load the highlighted Library.



Export/Import Library/Item

Import or export a Library or Library item to or from the selected USB storage device.

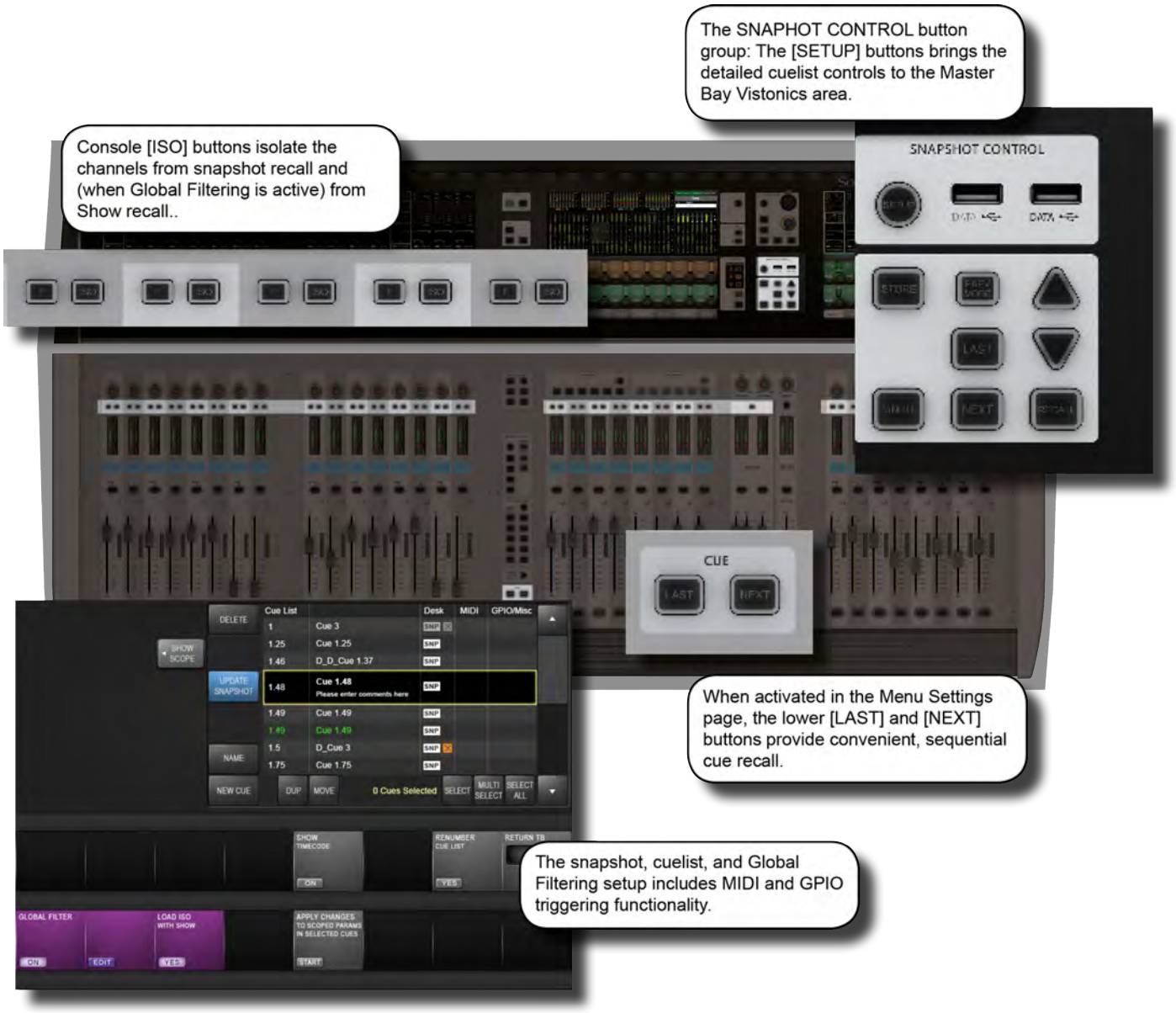
For example, with a Library item selected in the internal Library (right hand side of the file screen) and a USB storage device selected, hitting the Export arrow will copy the selection into currently selected Library on the storage device, into the correct category. If there are no libraries present yet on the external storage device, the complete library structure from the currently loaded Library will be copied to the storage device.



USB Storage Select

Select a connected USB Storage device.

SNAPSHOTS, CUELISTS, FILTERS



Snapshots are records of parameter settings across the console. Snapshots are stored in and recalled from cues in the cuelist. Cues can also trigger (or be triggered by) timecode, MIDI, and GPIO events. The ‘scope’ of a snapshot describes which parts of the console are included in the snapshot recall, and which parts are left out. Scope is set individually for every snapshot.

There is a separate snapshot filtering system called ‘global filtering’ which affects all snapshots. This uses the [ISO] buttons on the console. The global filtering system can isolate console channels, blocks, and parameters from either snapshot or show file recall.

Details on all aspects of snapshots, cue lists, and global filtering are included in this reference chapter.

NOTE: In order to store snapshots, you must first save a show. Default shows are write-protected as they are intended as basic templates. Save a show by touching the Show’s name in the Master Bay Vistonics Touchscreen (top right corner) or by navigating to the Shows tab of the main menu.

For more information on Show storage / recall see section 16.2. Recall filtering (Global Filters, ISO) is covered in this chapter, Section 14.4.

SNAPSHOTS, CUELISTS, FILTERS > CONTROLS



The main console snapshot controls are found in the 'SNAPSHOTS' group of buttons in the Master Bay. The 'ISO' (Isolate) button in the channel strips also plays an important role by allowing you to keep selected channels and sections 'protected' from snapshot (and Show) recall.



Snapshot/Cues Setup

Access snapshot and cue filtering set-up page in the Master Bay touch screen.



Data Sockets

Accept USB Storage devices for show storage and Media playback.

There are two more USB sockets on the rear of the console.



Undo Action

Undo the effect of pressing the [RECALL], [NEXT], or [LAST] buttons.



Store Snapshot

Store a snapshot of the current console state into a new cue and open up the Snapshot/Cue settings page in the Master Control screen.

The new cue will be stored and numbered between the currently selected cue and the next. For example, if you have selected cue 2 of 3 and then press the [STORE] button, the new cue will be numbered 1.5. If you have cue 3 of 3 selected, the new cue will be numbered 4.

SNAPSHOTS, CUELISTS, FILTERS > CONTROLS



Preview Mode

Make the cue list available for preview via the cue selection buttons [LAST], [NEXT], [UP}, [DOWN, and [RECALL].

When Preview Mode is deactivated, the cue list will return to the cue selected before Preview Mode was activated.



Recall Last Cue

Select AND recall the previous cue in the cue list.



Recall Next Cue

Select AND recall the next cue in the cue list.



Recall Selected Cue

Recall a cue that has been pre-selected with the [UP] and [DOWN] keys



Up / Down Cue List

Pre-Select cues by scrolling up and down the cue list. Press [RECALL] to load that cue.



Isolate From Recall

Isolate ('protect') this channel or selected parameters from the effects of snapshot recall or Show recall.

The [ISO buttons are found on the console channel strips. Isolate whole channels by pressing that channel's [ISO] button. Isolate sections, processing blocks, and so on by holding an [ISO] key down and selecting those blocks/ sections. Any isolated channel or selection will not be affected by the RECALL of a snapshot from the cue list. If you have 'LOAD ISO WITH SHOW' (Snapshot Setup) set to 'No' then your selections will also be isolated from a Show load. See the Snapshots Filtering And Isolation section below for more information. In addition to its usefulness in performance, the ISO function can also be useful for updating snapshots with new settings. For example, you could update a channel's EQ settings, isolate it, then recall cues and 'UPDATE CUE' with the new settings.

SNAPSHOTS, CUELISTS, FILTERS > CUELIST



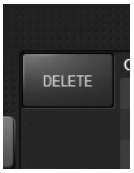
The Cue List shows all stored cues in the current Show's cuelist in several columns of information. A large cuelist and controls appears in the Master Bay touch screen area when the Snapshots Controls [SETUP] button is pressed or when a snapshot is stored. The cuelist is also displayed next to the channel meters in the Master Bay touch screen during normal operation.



Cue Selection

Select next and previous snapshots in the cue list.

Like the [UP] and [DOWN] buttons on the console, these buttons select the cue but do not recall a snapshot. Pressing [RECALL] will recall a selected snapshot. These buttons are also used to select multiple cues, together with the (SELECT) and (MULTI SELECT) keys.



Delete Cue

Delete selected cue / cues.

You will be asked "Are you sure you sure you want to delete the selected cue/s?" and given response options (YES) or (NO).

SNAPSHOTS, CUELISTS, FILTERS > CUELIST



New Cue

Store a snapshot of the current console state into a new cue.

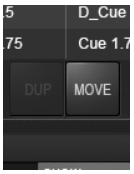
Has the same function as the [STORE] button on console surface. The new cue will be stored and numbered between the currently selected cue and the next. For example, if you have selected cue 2 of 3 and then press the [NEW CUE] button, the new cue will be numbered 1.5. If you have cue 3 of 3 selected, the new cue will be numbered 4.



Duplicate Cue

Duplicate the currently selected cue or cues.

If multiple cues are selected, the duplicates will be stored sequentially, after the final cue in the selection. Duplicate cues are renamed with a 'D_' prefix. Duplicating a Cue creates a copy of the entire Cue, including the desk snapshot and any MIDI or other events that were part of the original Cue.



Move Cue

Move the selected cue or cues to a new position in the cue list.

The (MOVE) button becomes the (DROP) button when selected.



Select Cue

Add the currently highlighted cue to a selection of multiple cues.

Used in conjunction with Move/Drop, Delete and Duplicate Cue functions. To create a multi-selection of non-contiguous cues, highlight cues with the (UP) and (DOWN) buttons and click (SELECT) to add them to the selection.



Multi-select Cues

Create a contiguous selection of multiple cues.

Used in conjunction with Move/Drop, Delete and Duplicate Cue functions. To select multiple, contiguous cues, highlight the first or last in the selection, click (MULTI SELECT), then select the range with the (UP) and (DOWN) buttons.



Select All Cues

Select all cues in the current cue list.

Used in conjunction with Move/Drop, Delete and Duplicate Cue functions.

SNAPSHOTS, CUELISTS, FILTERS > CUELIST



Update Snapshot

Update the snapshot in the currently selected cue to current console settings.



Show Timecode

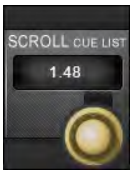
Show the recall timecode for all cues in the cue number column of the cue list.



Renumber Cuelist

Renumber all cues in the cue list as consecutive integers.

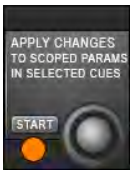
Cue names, which may incorporate the original cue numbering, remain unchanged.



Scroll The Cue List

Scroll through the cue list using the encoder.

Usually faster than the Up and Down buttons.

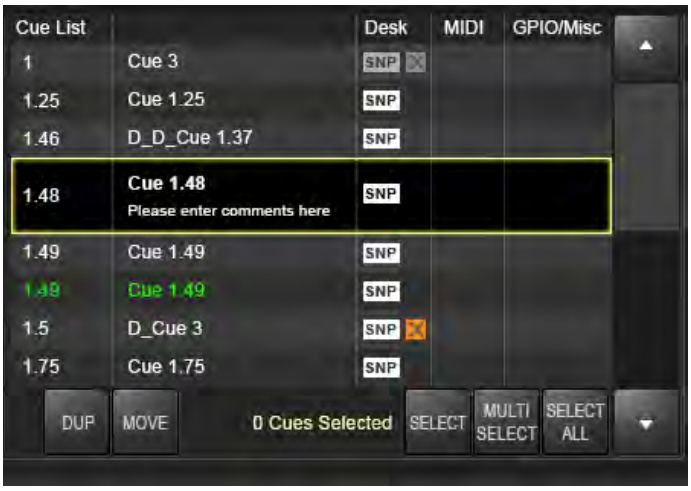


Apply Changes To Scoped Parameters In Selected Cues

Mode used to apply new settings to multiple cues.

Select {START} to enter this mode. You can then use the scope settings to filter process and channel selections, and use the cue list's cue-selection functions to choose which cues will be affected. Only 'scoped' selections will be applied to selected cues when you click the (APPLY} button. You can click the {CANCEL} button at any time to leave this mode.

SNAPSHOTS, CUELISTS, FILTERS > CUE DETAILS



Four segments / columns of each cue entry in the cuelist can be touched to enable extended functionality via the Master Bay Vistonics VST area. For example, by touching the Cue 2 entry in the cue list, in the MIDI column, that cue's MIDI functions (MIDI-based recall and a MIDI message output list) will be available in the VST area.

Also, touching the column headers in the cuelist will disable or enable that aspect of all cues. For example, touching the MIDI column header will disable MIDI functions in all cue recalls. Touching again will enable them.

CUES > NUMBER & TIMECODE



Selecting a cue number in the cue list gives access to that cue's basic settings, plus MIDI timecode recall event settings and the GO TO CUE 'local' sequencing function.



Timecode On/Off

Activate MIDI timecode-based recall for this cue.



Copy Timecode

Copy the current incoming MIDI timecode to the cue's recall TIMECODE fields.



Set Timecode

Manually set or adjust the MIDI timecode at which the cue is recalled.

Encoders adjust H (Hours), M (Minutes), S (Seconds), and F (Frames).

CUES > NUMBER & TIMECODE



Enable Cue

Enable or disable this cue.

Disabled Cues will be 'jumped over' if recalling successive Cues using the [NEXT] button. Enabled cues are shown in white in the cue list. Disabled cues are shown 'greyed out'.



Go To Cue Number Select

Set a following cue number - to be recalled a set amount of time after this cue has been recalled.

Can be very useful for firing a number of events from one cue activation. When combined with the Crossfade function, it's possible to create pseudo-dynamic Cues.



'Go To Cue' On

Activate the GO TO CUE functionality for this cue.



'Go To Cue' Time

The amount of time that will elapse before the 'GO TO CUE' cue is recalled.

Times ranging from 0 to 30 seconds can be set, in 0.5s steps. Using a time of 0s results in the chained Cues being recalled almost at the same time. This can be useful in conjunction with the Scope system to combine several Cue recalls each affecting different aspects of the console, into one recallable event.

CUES > SNAPSHOT / CUE NAME



Selecting a snapshot name in the cue list gives access to that cue's snapshot-specific settings and functions.



Enable UA Realtime Rack Snapshot Recall

Enable snapshot recall for connected Soundcraft / Universal Audio Realtime Racks.

This is enabled by default. When connected and enabled, a Soundcraft Realtime Rack sends its plug-in snapshot data to the console via an ethernet link when the console's [STORE] button is pressed and the data is automatically stored in the Cue list in a separate UA snapshot.



Snapshot Recall Enable

Enable or disable the desk snapshot in this cue.

This is enabled by default, though you can choose to disable it and therefore only trigger cue-based events such as MIDI outputs and GPO events.

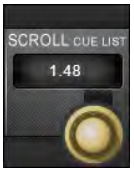


Snapshot Notes

Activate keyboard entry for snapshot notes. You can use the on-screen keyboard or an external USB keyboard.

Snapshot notes can be seen in the message area above the mini-Cue list in the Master touchscreen.

CUES > SNAPSHOT / CUE NAME



Scroll The Cue List

Scroll through the cue list using the encoder.

Usually faster than the Up and Down buttons.



XFade On

Activate snapshot-to-snapshot cross fading function.



Crossfade Time Setting

Adjust duration for snapshot crossfades.

Most variable parameters are included in the crossfade. The exceptions are: EQ and Hi/Lo Cut frequencies, and all Lexicon FX parameters. The Crossfade time is global for all parameters and all channels and busses.



Switches

Set the point in crossfade where switched parameters are switched.

In snapshot crossfades, variable parameters are incremented or decremented over the specified time, between values. Switched parameters are all switched at the same time, at the point during the crossfade specified by the Switches parameter: beginning, middle, or end.

CUES > MIDI



Selecting the MIDI column in a cue list entry gives access to that cue's MIDI input trigger and output settings. The output section defines an event list of MIDI messages that are sent when the cue is triggered. Note that some options rely on, or are governed by, global MIDI settings, accessed via the Settings menu.



MIDI Input Triggering Enable

Activate the MIDI input recall trigger for this cue.

When active, this cue will be recalled when the specified MIDI message is received from the specified MIDI device.



MIDI In Message Type

Select a message type to be the cue recall trigger.

New VST area fields will be activated depending on which MIDI message type is selected. For example, if a Note On message is selected, only the {NOTE VALUE} field will be available. If a Controller message is selected, then {CONTROLLER NUMBER} and {VALUE} will be available. When any valid message is selected, the {CHANNEL} field becomes available.

CUES > MIDI



MIDI In Message Learn

Activate Learn mode for MIDI input message type.

When LEARN mode is active, the next MIDI message received will be entered into the MIDI message fields.



MIDI Input Channel or Device ID

Select a MIDI device as the recall trigger 'carrier'.

Depending on the type of MIDI message selected, either a MIDI channel (1-16), or a Device ID (1-127) control will be presented. MIDI devices are set up in the master MIDI Settings section, though default to numbered (and available) MIDI inputs and channels. A Global device is also available, which defaults to 'OMNI' but can be customised in the master MIDI settings section.



MIDI Output Enable/Disable

Enable or Disable MIDI output for this cue.

A Cue's MIDI output consists of a message event list that is transmitted when the cue is triggered.



MIDI Output Event Number

Set the MIDI output event number to be edited.

There are up to 20 MIDI output events available for each cue. The encoder scrolls through the list of events and allows the parameters for each event to be set up using the associated Vistonics controls.



MIDI Output Event List

When selected, the MIDI event list for this cue is displayed in the 'snapshot scope' section of the snapshot settings screen.

Up to 20 output MIDI events can be set up for simultaneous recall from each Cue. Opening the event list allows the event settings to be viewed simultaneously as well as whether each of the 20 events is enabled or disabled.



MIDI Out Event Message Type

Select a message type for the selected MIDI event.

New VST area fields will be activated depending on which MIDI message type is selected. For example, if a Note On message is selected, the {NOTE VALUE} and {VELOCITY} fields will be available. If a Controller message is selected, then {CONTROLLER NUMBER} and {VALUE} will be available. When any valid message is selected, the {CHANNEL} field becomes available.

CUES > MIDI



MIDI Out Message Fire

Manually trigger the selected MIDI message for this event.

Useful for testing any MIDI message events you have set up. The FIRE button tranmits only the currently visible MIDI event parameters, not all 20 events.



MIDI Event Output Channel or Device ID

Select a MIDI device as the MIDI event 'carrier'.

Depending on the type of MIDI message selected, either a MIDI channel (Out1, ch1-16 and Out2, ch1-16), or a Device ID (1-127) control will be presented. MIDI devices are set up in the master MIDI Settings section, though default to numbered (and available) MIDI inputs and channels. A Global device is also available, which defaults to 'OMNI' but can be customised in the master MIDI settings section.

CUES > GPIO

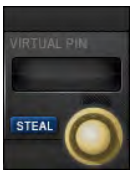


Selecting a the GPIO column in the cue list gives access to that cue's GPIO (General Purpose Input Output) settings. The VST screen is split into GPI (input) and GPO (output) sections. The inputs section sets an input to be used as a recall trigger for this cue, while the output section defines an output to be triggered when the cue is recalled. The cue GPIO system uses 'virtual pins' that are mapped to physical pins in the master GPIO settings screen.



GPI On

Enable GPI triggering for recall of this cue.



GPI Virtual Pin Select

Select a virtual GPIO pin as a recall trigger for this cue.

CUES > GPIO



GPI Pin Steal

'Steal' a virtual GPIO pin that is being used by another snapshot.



GPI Physical Pin

Displays the physical pin number that corresponds to the selected virtual pin.

The mapping from physical pins to virtual pins is set up in the GPIO page in the Main Menu.



GPO On

Enable the GPO trigger functionality of this cue.



GPO Virtual Pin

Select a virtual pin as the GPO assignment for this cue.



GPO Physical Pin

Displays the physical pin number that corresponds to the selected virtual pin.



Console Blackout

Sets a 'console blackout' for this snapshot.

The blackout turns off all console lights except for two that will deactivate the blackout. When the console is in Blackout state, pressing the [F6] key in the Master bay will cancel the blackout without changing to a different Cue. Alternatively, the [NEXT] or [LAST] buttons can be pressed to move to another Cue which does not have a blackout event set.

CUES > GPIO



HiQNet Venue Number

Select a HiQNet Venue Number

A HiQnet Venue number can be transmitted from a Cue recall and will send a message to all Harman devices in a HiQnet network reuesting them to recall a specific preset. The preset will be mapped to the Venue number within the settings of the Harman device concerned.



HiQNet On/Off

Enable HiQNet functionality for this Cue

SNAPSHOTS & SHOWS > RECALL FILTERING

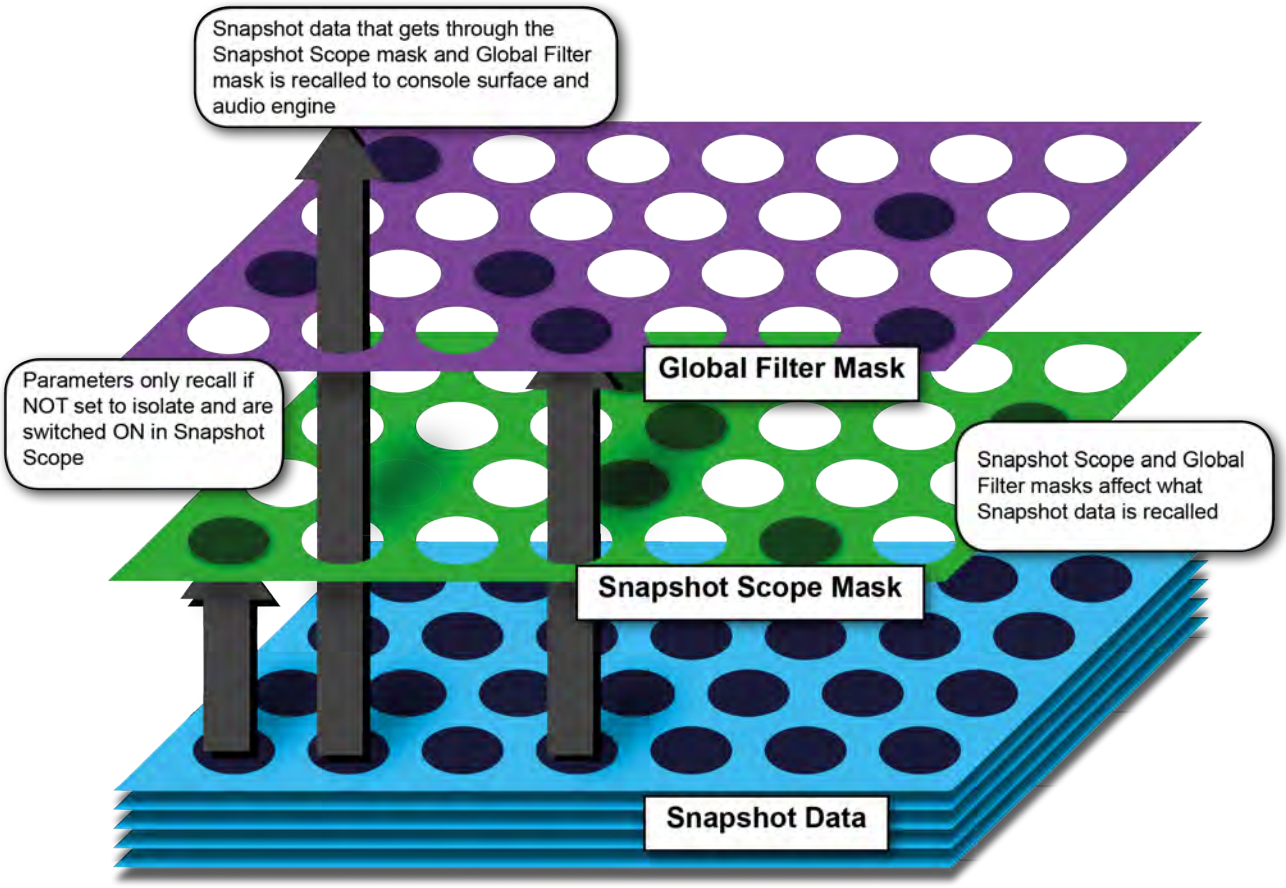


Snapshot filtering is a way of preventing certain settings on the console from changing when a Cue is recalled. There are two different reasons why this is useful, and therefore there are two different types of snapshot filtering: Global Filtering (also called Isolation), and Snapshot Scope.

Global Filtering only affects recall and is controlled with the console's [ISO] buttons or via the Global Edit mode; it stays on the desk when set and applies to ALL Cues that are recalled.

Snapshot Scope can be used to create partial snapshots; it is a specific filter that is stored with each Cue,and controls what data that Cue's desk snapshot will recall.

SNAPSHOTS & SHOWS > RECALL FILTERING



The Global Filter (or ISO) and the Snapshot Scope work together in series, so in order for a parameter to be recalled from the Snapshot memory, it must have its Scope setting set to ON, and must NOT be isolated. Scope defaults to 'all' for snapshots, so you can ignore it unless you specifically want to filter a snapshot.

RECALL FILTERING > SNAPSHOT SCOPE

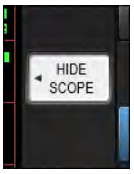


It can be very useful to make a Cue which, when it is recalled, only affects a specific area of the console - a 'Partial' snapshot. For example, you could make cues that will only set the Gains and EQ of all the channels, but leave the fader values unchanged.

There are two approaches to changing the scope of a cue. With Snapshot SCOPE active - press the (SHOW SCOPE) button - you will see selectable process/section blocks for input and output channels shown in the touch screen area, and a graphic of console channels shown in the VST area. Both can be used to set a specific scope for the cue.

Note that Scope is actually a non-destructive filter stored with the cue - the full snapshot is always stored and can be either restored or further limited by changing the cue's scope settings at any time.

Scope settings are done on a per cue basis, but whatever settings are showing will be used when the next cues is created.



Show/Hide Scope

Show or Hide the scope settings in the touch screen and VST areas.

Scope defaults to include all desk parameters, so you can hide scope and ignore it if you do not wish to use it. Scope defaults to hidden when you recall a console Default Show.



Select All/None Process blocks

Select or deselect ALL process blocks to be within the scope of this cue.

RECALL FILTERING > SNAPSHOT SCOPE



Block Select

Select or deselect individual process blocks from the scope of this cue.



Channel Scope - Select All

Select all busses and VCA masters to be within the scope of this cue.



Channel Scope - Select None

Deselect all channels/busses/VCA masters from the scope of this cue.

FILTERS > SNAPSHOT & SHOW ISOLATION (ISO)

To get manual control back while in a sequence of cues - either of a whole channel, groups of channels, or an element within a channel, you can use Global Filtering. The console [ISO] buttons are the main tools for this. Once a parameter is isolated, that parameter will remain under manual control regardless of which desk snapshot is recalled – hence the term Global filter; however, controls that are isolated can still be stored, so you can update snapshots with the manually adjusted changes at any time.

To isolate channels, busses, or parts of channels and busses you can use either the [ISO] button, or put the whole desk into Global Edit mode.

The Global edit filter (LOAD ISO WITH SHOW) can also be used to isolate aspects of the console's set-up from entire Show loads. So for example, if you have an engineer who brings a pre-configured show file to a shared console at a festival, you can isolate the console's output settings, which would normally be configured specifically for the venue, and protect them from being over-written by the new Show file.

The Global Edit button puts the whole desk into Edit mode so you don't have to hold the [ISO] button.



Isolate [ISO]

Select channels, groups of channels, and parameter blocks, and individual parameters to be part of the Global Filter.

To quickly Isolate a whole channel or bus from the snapshot recall, press its [ISO] button. The channel strip border on the Vistonics screen changes to purple to indicate the 'All Isolated' state. To quickly isolate groups of channels or busses, the Gang function can be used.
To isolate parts of a channel or buss you can use either the [ISO] button, or put the whole desk into Global Edit mode.
With an [ISO] button held, simultaneously touch the part of the channel strip that you want to isolate, in the touch screen. A purple LED appears in the top corner of the block to indicate that the block is isolated. To isolate an individual parameter, touch the Vistonics controls for the parameters you want to isolate. You can either zoom into the function block before selection, so the parameters within that block are already mapped onto the VST area, or touch and hold a screen block as well as the [ISO] button - that will zoom into the parameters for the block.



Global Filter Enable

Activate Global Filter

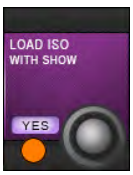
The Global Filter ON mode is automatically activated as soon as any ISO buttons are pressed on the console. With the Global Filter activated, 'ISO' selected channels and parameters will be isolated from snapshot recall. Switching the Global Filter mode to OFF is a quick way of disabling the isolation of all parameters across the desk.



Global Filter Edit Mode

Activate Global Edit mode.

All desk parameters are available for adding to the Global Filter, without having to hold down an [ISO] button.



Load ISO With Show

Enable or disable Global Filter for Show loading.

When set to YES the console ISO settings will be cleared and overwritten by loading a new Show file. Set this to NO to protect isolated sections of the console from being overwritten by a new Show file. Note that this setting always defaults to YES each time the console is switched on.

TALKBACK & OSCILLATOR



Most main oscillator and talkback controls are grouped together in the Master Bay, though there is a Talkback Assignment button [TB ASSN] in the VISTONICS button group, which allows manual assignment of the talkback group bus destinations.

Talkback is a independent signal path, normally from a microphone input to group and physical outputs, activated with push-to-talk talkback buttons on the console itself. It is most often used to allow the console operator to talk to assistants, performers, and so on. There is also a talkback return, which is an independent input path to the monitor section - most often used to allow assistants and performers to talk directly to the console operator.

The oscillator is a signal generator provided as a general-purpose calibration and checking tool.

TALKBACK & OSCILLATOR > CONTROLS





Talkback And Oscillator Setup

Open and close the Talkback and Oscillator setup screen in the master Vistonics screen.



Talkback XLR Connector

The Talkback Mic input.

This connector is duplicated on the rear panel of the surface. There is a switch for phantom power adjacent to the rear panel Talkback Mic connector if required.



Talkback / OSC Level Control

Control of Talkback or Oscillator level.

For Talkback, the analogue Mic Amp gain is variable from 46-66dB by 3 internal jumpers which are located on the PCB that is located adjacent to the rear-panel talkback mic connector.. This is set by default to a mid-range setting. If the Oscillator is active, the control adjusts the Osc level to all destinations. The level setting is stored independently for TB and Osc. The front-panel Talkback/Osc level control applies an additional digital gain trim to the talkback mic signal.



Talkback To Internal

Route the Talkback signal to pre-selected output busses.

The 'INT' routed Output busses are selected via their {TB} VST keys in the VST area of the master screen after selecting the [TB ASSN] key (located to the right of the master screen's VST area). Switch between the two sets of 16 busses with the [PAGE A] and [PAGE B] keys.

TALKBACK & OSCILLATOR > CONTROLS



Talkback to External

Route the TB signal to any available output in the console's IO system.

Change the EXT TB assignment via the Talkback/Oscillator SETUP screen.



Talkback Presets 1-3

Press-to-talk (momentary and latching), user-programmable Talkback switches.

Select the output routing for the Talkback presets via the Talkback/Oscillator SETUP screen.

OSCILLATOR > SETUP



The main oscillator controls appear in the master bay touch screen area when the Oscillator / Talkback [SETUP] button is pressed. More detailed setup functions appear in the VST area, colour-coded orange.
The oscillator can also be patched to the Input Channels by using the {OSC} key in the Input Channel VST screen.



Pink Noise

Set the oscillator waveform to pink noise.

Pink noise is generated noise with equal power per octave.



White Noise

Set the oscillator waveform to white noise.

White noise is generated noise with equal power per Hertz (Hz).



Sine Wave

Set the oscillator waveform to sine.

A sine wave is a waveform generated from y= sin x.



Oscillator To Bus

Feed the Oscillator signal to the preconfigured Busses.

OSC To Bus assignments are selected in the Bus Assign page via the Talkback/Oscillator SETUP screen.

OSCILLATOR > SETUP



Oscillator To Talkback

Route the oscillator signal to the TB Bus (replaces the TB signal).



Oscillator Frequency

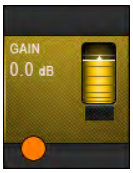
Adjust the Oscillator sine wave Frequency in the range 20Hz to 20kHz.



Oscillator Bus Assignment

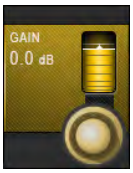
Used in conjunction with the Oscillator to Bus touch screen switch.

Opens an output assignment page where you can select any of the 32 group busses or the Mix L, R, and C outputs.



Oscillator On

Activate the console's oscillator.



Oscillator Gain

Adjust the oscillator output level



Oscillator Output

Assign the oscillator to individual physical outputs.

Opens an output assignment page with all available physical outputs available.

OSCILLATOR / TALKBACK > TALKBACK SETUP



Talkback is a communications path between the console operators and the artists, crew, and other personnel. There is a direct-plug mic input (XLR) on the console surface with the TALKBACK/OSC group of controls. This and other talkback routing and configuration options are detailed below.



Talkback Source

Select the Talkback source / input.

If (NONE) is selected the console's TB mic (front and rear panel) XLR is automatically selected. The button opens up an assignment selection page in the master touch screen area where you can select any available microphone input.



Talkback External Output

Assign the Talkback output directly to individual physical console outputs.

Opens an output assignment page with all available physical outputs. The talkback mic signal will be routed to the selected connector when the [EXT] button is pressed on the front panel.



Talkback Preset Assignment

Assign a selection of output busses to the selected Talkback preset

There are 3 Talkback Presets that can be set up. Pressing the [PRESET 1], [PRESET 2] or [PRESET 3] front panel buttons routes the Talkback mic signal to the assigned busses.

TALKBACK > TALKBACK RETURN



Talkback return allows the console operator to route inputs directly into the monitor circuit. This allows assistants within a venue to talk directly to the console operator. The talkback function must be enabled via the MONITORING setup page by selecting the TB RET touch screen button for Monitor A, B or Phones outputs.



Talkback Return ON

Enable the Talkback return.



Talkback Return Gain

Adjust the Gain of the talkback return input.



Talkback Return Assignment

Assign the Talkback Return input.

Opens an input assignment page with all available physical inputs.

CONSOLE MENU & SETTINGS



The Main Menu contains settings for general console systems and functionality, including GPIO and MIDI setup; Show loading duplicating, and export/import; Tielines (input > output patching) setup, Lexicon FX operation, and more...



Menu Button

Access the main menu.

The main menu will appear on the master bay touch screen. Navigate the menu by selecting the various tabs. Some tabs map additional controls to the VST area.

MENU > MAIN



This menu tab is displayed by default whenever the [MENU] button is pressed.

Software and Platform version information
The currently loaded software version number is displayed. This will be in the form of a number YYMMDD-V6.x.x.xxx, and will change when software updates (available from the Soundcraft website from time to time) are applied to the console.
The Platform version number denotes the operating system version which is not updatable except by replacing the console's internal SSD drive. The version number will be in the fomat 7.x.x.x.

Soundcraft service agents may request interrogation of the platform and software version numbers when performing maintenance or troubleshooting.



Brightness

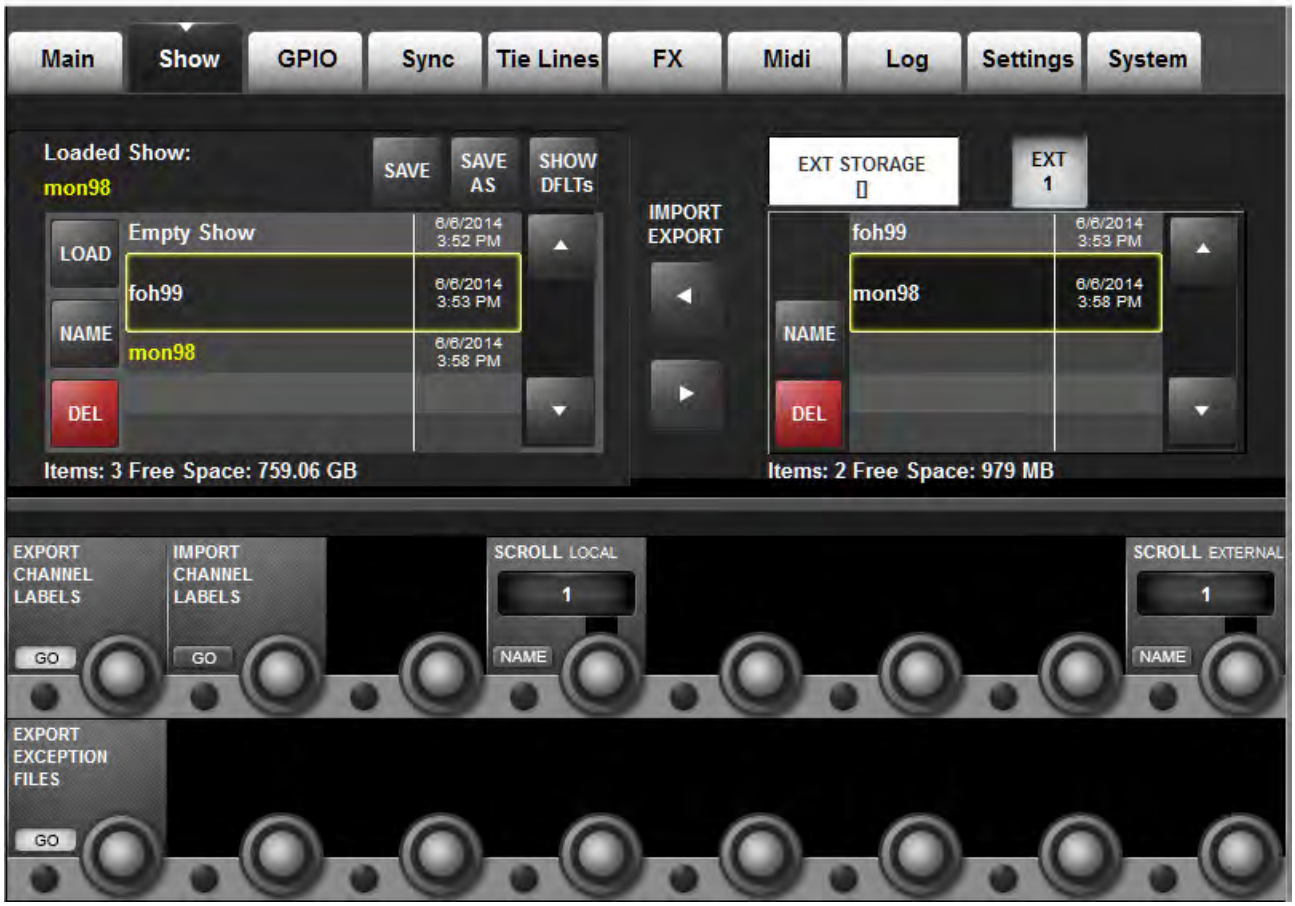
Choose the brightness level for all screens, illuminated keys, and FaderGlow.



Desk Lock

Lock all switches, encoders, and faders on the console, except for the [LOCK] key.

MENU > SHOWS



Shows are the main vehicle for storing console control settings, cue lists and snapshots, and audio set-up.

Loading Shows, copying Shows to and from USB data storage devices, and creating new Shows is done from the Show tab in the Main Menu.

The left-hand side of the page displays the shows that are present on the console's hard drive. If there are any USB storage devices connected, these will be selectable and displayed on the right hand side of the page. Normally the front panel USB Data Sockets will be used (USB1 and 2), but it is possible to connect USB data storage devices to the rear panel USB sockets (USB3 and USB4), and select the required device with the (USB1), (USB2), (USB3), and (USB4) buttons. The up and down-arrows on both Show file lists are used to scroll though the available Show titles.

The user can select a default show or another existing show and use the (SAVE) and (SAVE AS) keys as required.

IMPORTANT: If you want to base your Show on one of the defaults, you MUST save it as a new show before you start working, otherwise you will not be able to store Snapshots. This is because of the write-protection assigned to the default Shows.

MENU > SHOWS



Load Show

Load the selected show from the hard drive.

Shows can only be loaded from the internal hard drive, they must be imported from the external storage first.



Name

Rename the selected Show file.

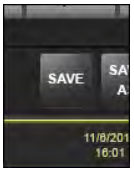
Opens up the text entry screen.



Delete Show

Delete the selected Show file.

An additional screen will appear asking you to confirm the deletion or cancel it.



Save

Update the loaded Show file.

A confirmation dialogue will appear.



Save As

Save all current show parameters as a new Show.

You will be prompted to input a new show name via the on-screen keyboard or an external USB keyboard.



Show Defaults

Display the default Shows in the Show file list.

The console comes with some factory-installed shows. These are not normally visible in the list, but they can be seen by pressing the (SHOW DFLTs) button. The default Shows are at the top of the list and are shown in italic text. They cannot be deleted with the (DEL) key, nor renamed. If you want to base your Show on one of the defaults, you MUST save it as a new show before you start working, otherwise you will not be able to store Snapshots. This is because of the write-protection assigned to the default Shows.

MENU > SHOWS



Scroll Local Shows

Scroll through the local (internal) show file list.

The file list can be sorted alphabetically or by last modified date, by toggling the NAME/DATE button.



USB Import / Export

Exporting A Show TO, or import a show FROM A USB Data Storage Device.

Pressing the right-facing arrow will export the currently-selected Show to an installed USB data storage device. Pressing the left facing arrow will import the selected USB-stored show to the console's internal storage. Note that the date and time of the latest save is shown in both lists.



Export Channel Labels

Choose to export channel labels with a Show export.

The labels are exported to a file called 'Soundcraft Vi Channel labels.csv' in the root of the external storage device, and can be imported into a spreadsheet application.



Import Channel Labels

Choose to import channel labels with a Show export.

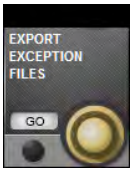
If a file named Soundcraft Vi Channel labels.csv' is found in the root of the external storage device, and it is formatted according the the file exported by the console, the channel labels in this file can be imported into the console, replacing any existing labels on the control surface. In order to make imported labels permanent, the Show must be saved after importing them.
The GO buton will only be visible if a valid file is found on the external storage.



Scroll External Files

Scroll through the external (selected USB storage) Show file list.

The file list can be sorted alphabetically or by last modified date, by toggling the NAME/DATE button.



Export Exception Files

Exports log files that are generated if the console software restarts unexpectedly. For use under instruction of Soundcraft service personnel.

The files are saved in a folder called 'Exceptions' on the USB stick, and should be zipped and emailed to Soundcraft support when requested.

MENU > SHOWS > RECORDED SHOW DATA

Menu: Shows - Recorded Data

Some settings are recorded as part of a Show, others as part of a Snapshot, and a few are not recorded at all.

Please note that in addition to the snapshots in the cue list, a Show has one ‘Show Snapshot’. This Show Snapshot is generated automatically, and it cannot be seen by the user. It holds the Cue List, the Show Settings and Audio Settings. These three sets of data contain all of the parameters that change when a Show is loaded.

The Show Snapshot’s Audio Settings contains all the settings which a standard Snapshot can store; as a result, the Show Snapshot can be thought of as the last settings before the Show was unloaded (Assuming you choose to save the existing show before loading a new one). The Show Snapshot allows the complete status of the console to be recorded with the Show, even if no Snapshots have been saved.

Settings Recorded Within A Show Snapshot’s Show Settings.

- Monitoring Settings: Monitor Level; Phones Volume; Solo Trims; Solo Blend; Monitor A/B Switch Status; Monitor On/Off Switch Status; Monitor Source Selection Status; Monitor Setup states.
- All switches in Talkback section.
- Talkback Settings: Talkback Levels; Talkback setup.
- Generator Settings: OSC Level, Type.
- Mute Safe Status (Input & Output).
- System Preferences: Current Sample Rate.
- Automation Setup States.
- VCA / MG Assign View switches status: Currently selected view.
- Bus Config States: Bus Formats; Bus Types and Bus Labels.
- Channel Pairing States: Stereo Channel pairings.
- HiQnet Setup.
- MIDI Configuration Setup: MIDI Channel names.
- ISO switch status (Input & Output).
- O/P Vistonics Lock Mtr switch status.
- O/P Vistonics Solo/On/Off/TB switches.
- All parameters in the Menu/Settings page.
- Follow Solo switches.

MENU > SHOWS > RECORDED SHOW DATA

Settings Recorded Within Audio Settings.

This applies to the Show Snapshot and to any standard Snapshots.

• All Channel audio settings on the console: Channel ON; Fader positions; Pan; Channel Parameters (EQ, Dynamics, input/output, insert).
• All Channel Bus assigns, levels, Pre/Post states and Channel Labels.
• All Channel Patch ordering.
• All I/O routing.
• All I/O controls.

Settings Not Recorded

• PFL/SOLO switch status.
• SEL Switch status.
• LRC Sel switch status.
• Metering values.
• All round Setup switch status.
• User Defined switches status.
• All switches in Snapshot Control section.
• Power On switch status.
• Copy/Paste/Undo switch status.
• Set Pre/Post modes switch status.
• Pan/Level toggle switch status.
• Gang Mode switch status.
• Solo Clear switch status.

MENU > GPIO



You can configure all available GPIO channels in the GPIO (General Purpose Input Output) tab of the main menu. However, please note that 'virtual' GPIO pins can also be assigned to cues in the cue list via the Snapshot SETUP page. Virtual GPIO Pins are assigned via the function/parameter setting in the main GPIO settings.

The GPIO card is fitted as standard to the Local Rack and Vi Stagebox and provides electrically isolated opto-coupler inputs with integrated current sink (5 to 24 VDC) and electrically isolated outputs using SPST relay contacts. 5 VDC and Gnd supply pins are provided. Inputs and outputs are on standard D-type connectors (female).

The main screen shows a list of available GPIO connection 'pins' on the selected hardware (console or stage box). The upper section shown in blue is the GPI (input) section, and the lower section shown in red is the GPO (output) section. Each entry shows its various configurations across several columns; these are set in the VST section of the master bay Vistonics.

MENU > GPIO



Console GPIO Select

Selects the GPIO in the control surface (16 GPIO channels).



Stagebox GPIO Select

Selects the GPIO in the Stage Box (8 GPIO).

(No function unless one or more Stageboxes attached)



GPIO Up / Down List

Scroll though the GPI and GPO lists

The currently-selected input and output channels are outlined in yellow.



GPI On / off

Enable and disable GPI functionality

Individual setting for each Pin.



Function Select

Select the function to be assigned to the GPI pin.

Depending on the function selection, the {Parameter} field may or may not be available. For example, the Channel Mute (CH MUTE) function requires a channel assignment - set using the {parameter} field; the NEXT CUE function does not require any additional parameter. Available functions: Channel Mute (CH MUTE); (F KEY LED); Talkback Input (TB INPUT); Dim Monitors (DIM MON); VIRTUAL GPI; LAST CUE; NEXT CUE; CUE UP; CUE DOWN; RECALL CUE; Undo Snapshot (UNDO SNAP).



Parameter Set

Used to assign additional values to functions

Certain GPIO functions require and additional parameter. Examples are channel number, virtual GPIO pin number, and so on.

MENU > GPIO



Trigger Edge Select

Choose the triggered edge for this GPI pin.

Choose either Rising, Falling, or Both. If you choose 'Both' the {Polarity} field will become available.



Trigger Polarity Select

Set the Polarity of the input.

The encoder changes the polarity between positive and negative. This field is only available if the {EDGE} field is set to BOTH.



GPI On / off

Enable and disable GPO functionality

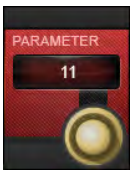
Individual setting for each Pin.



GPO Function

Select the function to be assigned to the GPO pin.

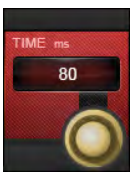
Depending on the function selection, the {Parameter} field may or may not be available. For example, the Fader Start (FDR START) function requires a channel assignment - set this using the {Parameter} field; the Talkback Output (TB OUTPUT) function does not require any additional parameter. Available functions: Channel Mute Fader Start (FDR START); F KEY; Talkback Output (TB OUTPUT); VIRTUAL GPO; Channel Mute (CH MUTE).



GPO Parameter

Used to assign additional values to functions where required.

Examples include channel number, virtual GPIO pin number, and so on.

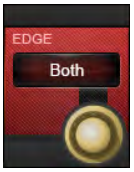


Pulse Time

Set the Relay Pulse Time.

The relay contacts will revert to their original position at the end of the pulse time. The encoder adjusts the pulse length in ms (a blank field means no pulse is generated - the relay contacts stay in their new position). It is suggested that a pulse time is set only when the EDGE field is set to 'RISING' or 'FALLING. It is also suggested that no pulse time is set when the EDGE field is set to 'BOTH'.

MENU > GPIO



Edge Select

Choose the triggered edge for this GPO pin.

Choose either Rising, Falling, or Both. If you choose 'Both' the {Polarity} field will become available.



Polarity Select

Set the Polarity of the output.

The encoder changes the polarity between positive and negative. This field is only available if the {EDGE} field is set to BOTH.

MENU > SYNC



The settings in this tab determine the digital audio clock configuration for the console



Internal Clock

Internal sync selected (not a control).

The console runs at either 48 or 96kHz. The console can be switched to run in 96kHz sampling frequency mode in the SETTINGS tab of the main menu (Section 16.9). When this is done, the available input channel count is reduced by half (to 64).



External Clock

External clock source selected (not a control)

The console switches to External Wordclock automatically if a valid wordclock is received.
A green LED adjacent to the Wordclock In BNC socket indicates a valid lock to external clock.
External wordclock frequency must be within range of the current sample rate of the console, ie the console will not lock to a 96kHz external clock if the sample rate of the console is still set at 48kHz.

MENU > TIELINES



TIE Lines are direct connections from an input connector to a output connector. They are a path through the mixer with no processing and no mixing, and there they do not use up any DSP channels. The console supports up to 128 tie lines, arranged 8 per page.



Page Select

Switch between 16 pages of 128 possible Tie Line patches.

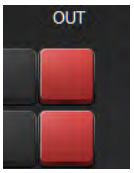
MENU > TIELINES



Tieline Input Patch

Assign a physical input to this Tie Line.

The Red INPUT buttons open an input assign page in the main touch screen area where you can choose any available console input.



Tieline Output Patch

Assign a physical output to this Tie Line.

The Red OUT buttons open an output assign page in the main touch screen area where you can choose any available console output.



Tieline Label

Define a custom label for the Tie Line.

The Tie Line Label buttons open up a text entry screen in the main touch screen. You can also use an external USB keyboard.

MENU > FX



This is where you configure the eight internal FX processors. Please see the Effects chapter (Section 17) of this reference for more specific information.



Lexicon FX Engine 'n'

Select a Lexicon FX processor to control

Algorithm selection and related parameters are mapped to the VST section of the Master Bay Vistonics screen. Most importantly, this page shows an overview of where each FX unit has been assigned, using the input/output or insert patching screens of the relevant channel/bus.

MENU > MIDI



General MIDI setup for the console. Define MIDI devices, timecode parameters, and so on.



List RX Channels

Display the MIDI input channels in the device list.

The Device List allows you to specify friendly names for each MIDI channel or Device ID, to help with identifying a source or destination device in other MIDI setup screens such as the MIDI Events list in the Cue List.



List TX Channels

Display the MIDI output channels in the device list.

The Device List allows you to specify friendly names for each MIDI channel or Device ID, to help with identifying a source or destination device in other MIDI setup screens such as the MIDI Events list in the Cue List.

MENU > MIDI



List TX Devices

Display the MIDI output Device IDs in the device list.

The Device List allows you to specify friendly names for each MIDI channel or Device ID, to help with identifying a source or destination device in other MIDI setup screens such as the MIDI Events list in the Cue List.



MIDI Device Name

MIDI Device Name display

Allows the friendly name to be entered using on-screen QWERTY keyboard.



Scroll MIDI Device List

Fast-Scroll through the device list with the encoder.



MIDI In Enable/Disable

Enable / Disable All MIDI input activity.



Global MIDI Input Channel

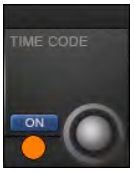
Set the Global MIDI input channel



Global MIDI Input Device ID

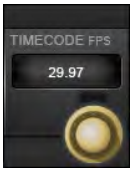
Set the Global MIDI input Device ID

MENU > MIDI



MIDI Timecode Enable

Enable MIDI Timecode Input



MIDI Timecode Frame Rate

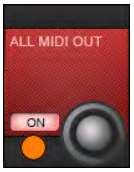
Sets desired frame rate in FPS for outgoing timecode events and incoming timecode.

When enabled, the Show name displayed in the Master Bay touch screen is replaced by the timecode display.



Timecode Display On/Off

Enable or disable the timecode display in the Master Bay touch screen area.



MIDI Out Enable/Disable

Enable / Disable All MIDI output activity.



Global MIDI Output Channel

Set the Global MIDI output channel

MENU > LOG



This page displays any errors that have occurred since the console was last powered-up. These would usually be communications errors between the various components in the system such as control surface, local rack, and stagebox.

The most recent message is also displayed at the top right of the master section's main screen, in the Error Log Display Area. The message is cleared from the main screen after the Log page has been viewed by the user. Note that the Log page can be reached as described above, or by touching the Error Log Display Area on the main screen.



Error Expand/Compact

Expand or compact the selected error message.

Expanded error messages wrap the text so the whole error message can be viewed, at the expense or the number of errors viewable at the same time.

MENU > LOG



Automatic Scrolling

Cause the most recent message to be displayed as the currently-selected one



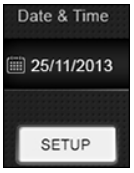
Clear Log

Clear the error log list.

MENU > SETTINGS



Miscellaneous settings for the console, including the date and time, plus input meter source point and general meter peak hold time, delay, EQ, and graphic EQ.



Set Date And Time

Send date and time settings controls to the VST section of the Master Bay Vistonics panel.



Meter Peak Hold Set

Adjust the console metering Peak Hold time from 0 to 12 Seconds.

MENU > SETTINGS



Set Meter In Point

Determine the metering point for the input channel level meters.

There are four choices for the input meter tap, from left to right: After the analogue mic gain (before digital trim and filters); after the digital trim and filters (before the Gate/EQ/Dynamics); after the Gate/EQ/Dynamics (before the fader); after the fader.



Graphic EQ Large/Small Mode

Choose between Large and Small Graphic EQ control modes.

Choose 'Large' 30-fader mode for fast access on multiple faders, at the expense of access to the input faders, or choose 'Small' (eight output faders) mode when access to input faders must be retained at all times. Frequency bands can be scrolled in banks of four or eight bands, using the Output Fader page buttons.



Set Delay Units

Change the units used in input, output, and monitoring delay adjustments.

You can choose between milliseconds, metres, or feet and inches.



EQ bandwidth units

Choose the unit for bandwidth controls in the EQ sections throughout the console

Selected either Octaves or Q-factor (defined as the ratio of frequency to bandwidth). The direction of the control is reversed between the two settings: In Q mode, clockwise narrows bandwidth, in Octaves mode, clockwise widens bandwidth.



Enable NEXT and LAST

Enable the [NEXT] and [LAST] cue list navigation buttons nearest the operator.



DSP Configuration

Set the console's DSP sample rate.

You can choose 48kHz (128 input channels) or 96kHz (64 input channels). Press {SET} to confirm a change. NOTE: Changing the sample rate will reset the console and you will lose unsaved changes. A warning/confirmation dialogue appears.

MENU > SYSTEM

System Monitoring and HiQnet/VM² Setup



System monitoring and HiQnet setup.

In normal operation, the main System Monitoring Overview Display Area is located at the top right of the Master Bay touch screen, and the Error Log Display Area is just below it. Within the System Monitoring Overview Display Area each hardware device and the HiQnet™ network state is represented with a coloured label. The label colour indicates the overall state of the system monitoring page. A green label indicates that this device is running correctly, whereas a red label indicates an error condition.

Error and warnings are displayed in the Error Log Display Area.

To access detailed System information, enter either though the main menu by pressing [MENU] then (System), or by touching the System Monitoring Overview Display Area. When the System page has been opened, there are 4 sub-pages accessed via the touch-pads on the right hand side: DESK, LOCAL I/O, STAGE BOX and HiQnet™.

MENU > SYSTEM



Desk Information

Display the current status of the desk's 4 (Vi5000) or 5 (Vi7000) bays.

The numerical data displayed for each bay gives the revision number of the firmware currently installed in each bay. This information may be needed by Soundcraft service personnel if technical or service support is requested.



StageBox Information

Display a graphical representation, and the current status, of the cards in the Stage Box.

This also shows the status of the PSU(s) and the status of the voltages for the analogue (VA) and digital (VD) power supply rails. The cooling fan status is also reported. A blue label under a card display indicates an input function, and a red label indicates an output function.

If the card configuration of the Stagebox is changed (for example when AES input or output cards are fitted in place of the analogue cards), the card labels will be automatically updated with the new card types if the 'Reconfig' button on the Stagebox front panel has been pressed after changing the cards.



VM2 / HiQnet Set-up

Enter the VM2 setup screen

VM^2 stands for 'Vistonics Microphone Monitoring'. It takes advantage of Harman's HiQnet control network to allow a level of integration between a HiQnet-enabled wireless microphone system, and a Soundcraft Vi Series console. This section is also used to set up a network IP address for use with both ViSi IPad remote control app, and the Soundcraft Realtime Rack host control for Universal Audio Plug-ins.
Please see the HiQnet/VM^2 reference chapter 18 for more detail.

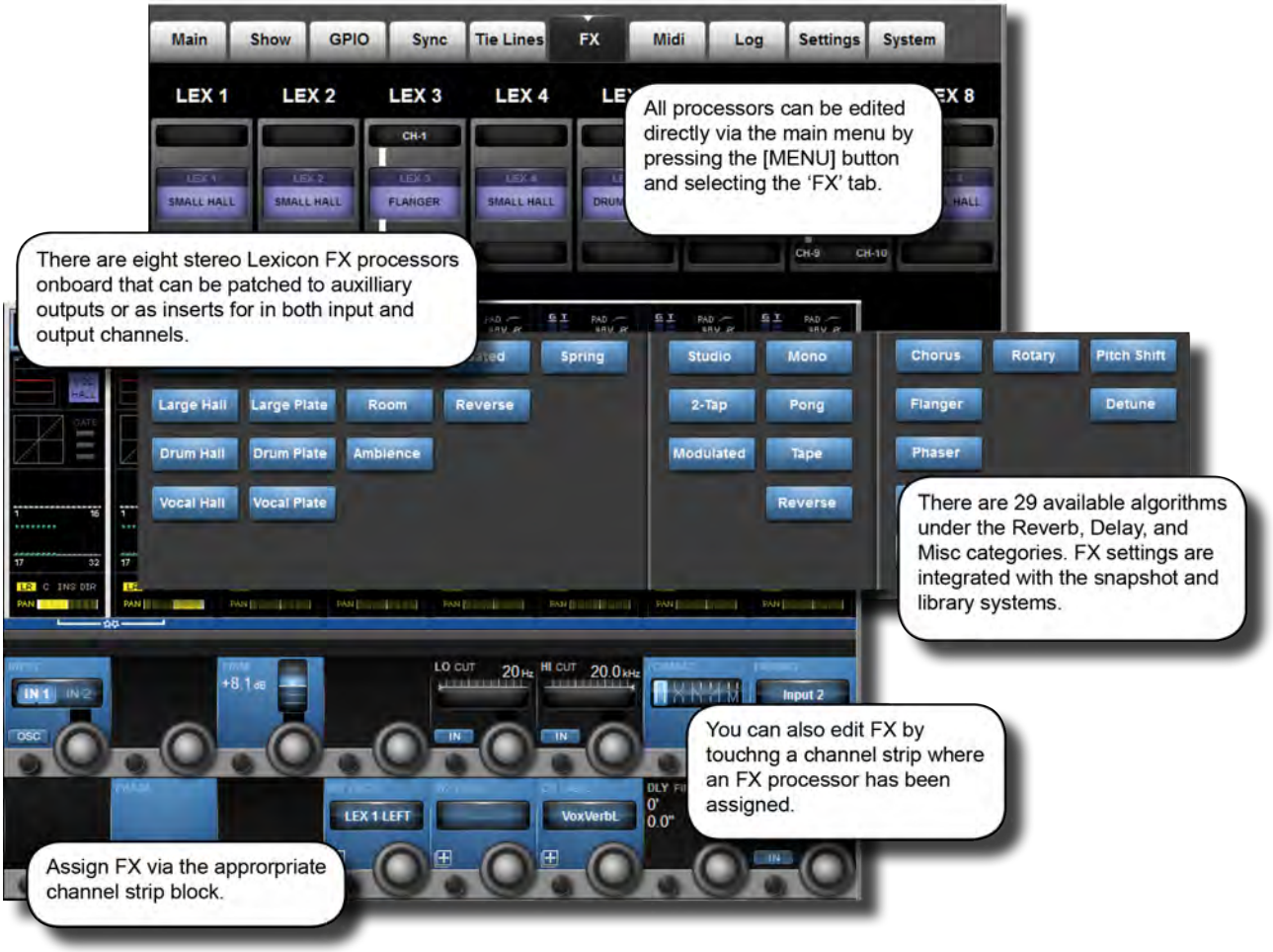
About HiQnet

HiQnet is a network communications protocol developed by Harman which enables various professional audio products within the Harman Pro-audio range to communicate with each other and provides remote control and monitoring of Harman devices such as power amplifiers, powered speakers, DSP processors and microphones. Ethernet is normally used as the transport method for HiQnet control, as it provides a robust and standardised method of connecting multiple devices in a network configuration. A Windows application called Audio Architect provides a master control and monitoring program that allows all equipment on the HiQnet network to be controlled from a single user interface.

On the Vi Series of consoles HiQnet allows error messages generated by other Harman devices to be displayed in the console's message log – for example thermal overload messages generated by Crown amplifiers. In addition the consoles are able to transmit a Venue Preset recall - the HiQnet equivalent of a MIDI Program Change – to all other devices in the network.

LEXICON FX

Lexicon FX



The console comes with eight stereo Lexicon FX processors. Each FX Unit can be inserted into any Output/Main Master bus or into any Input Channel, or it can be patched as an FX Return to an Input Channel or fed from an Aux send.

Algorithm selection and parameter adjustment is integrated into the console's Vistonics II environment - editable either from the relevant channel's bay, or from the Master Bay via the FX Overview page in the main menu.

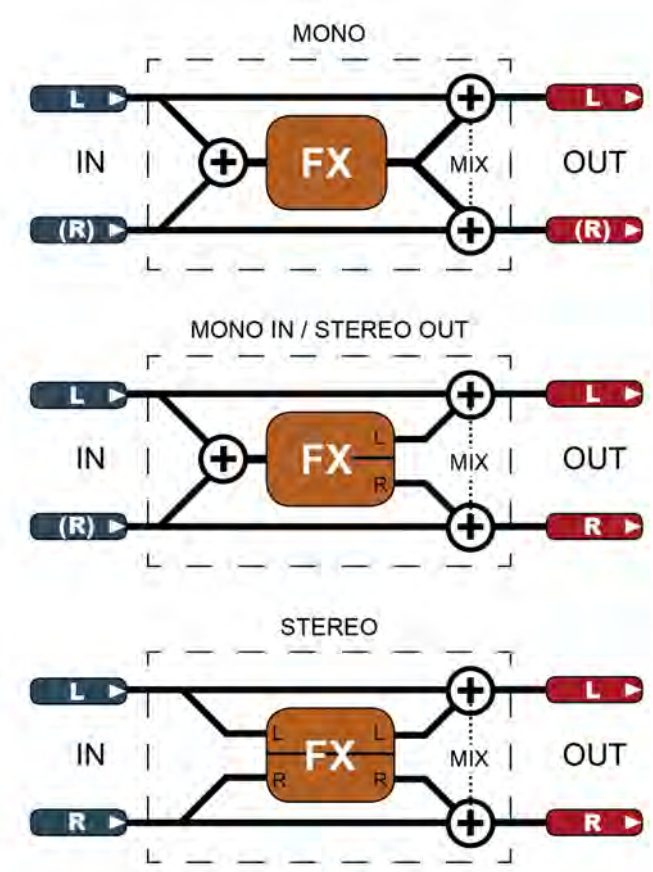
All Parameters from the four Effects Units and for all Effects Type are stored in the desk Snapshots.

LEXICON Effects Format

Depending on the selected effect type, the FX processor works internally in one of three formats: Mono, Mono In/Stereo Out, or Stereo In/Stereo Out.

The FX processor always has Stereo Inputs and Outputs. If the FX Type needs only a Mono Input, the Left and Right Input Signal are summed together. If the FX Type outputs only a Mono Signal then the Output Signal is distributed to both the Left and Right Outputs. The MIX Parameter adjusts the ratio between the original (dry) signal and the effects (wet) signal. The stereo format of the dry signal will be maintained.

LEXICON FX



The Lexicon FX processors work in three different formats.

An FX processor always has stereo inputs and outputs. For a mono-input effect, the left and right input signals are summed together. For a mono output, the single output signal is distributed to both left and right outputs.

'MIX' represents the effect's own wet/dry mix parameter.

Using The Lexicon FX Processors

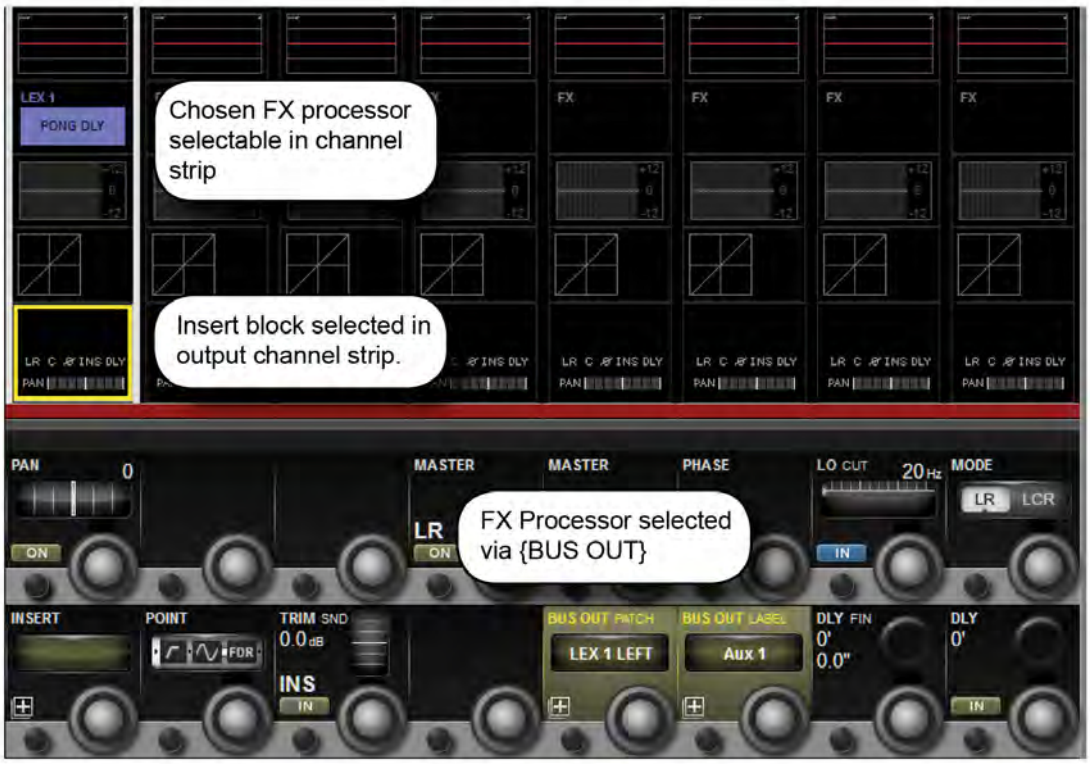
Each FX processor can be patched in three ways. Aux Masters can be patched to the inputs of FX units; input channel inputs can be patched to the outputs of FX units; and both input and output insert points can be patched to FX units.

Patching status is visible in the FX overview page, though actual patching of FX units takes place in the relevant source / destination channel strips. For example, to patch an FX unit to an input channel, go to that input channel's channel strip (input block).

To use an FX unit in a traditional aux send / return setup you would patch an aux output to the inputs of an FX unit, then patch an input channel to the outputs of the same unit.

As well as the VST {tap tempo} button, the console's F-Keys can be assigned to the tap tempo function (see below for details).

As An Aux Send



To patch an FX unit as an Aux send destination, you must first select either the output or the insert point of the chosen Aux master. A fast way to get to the aux channel strips is to press the [All BUSSES] button. See the Outputs reference chapter 7 for more options).

If you use the insert point of the aux bus you can then patch the output of that aux directly to the master LR bus, for example. Once assigned as an insert, you will be able to edit the select FX unit directly by touching the FX block in the output channel's touch screen channel strip.

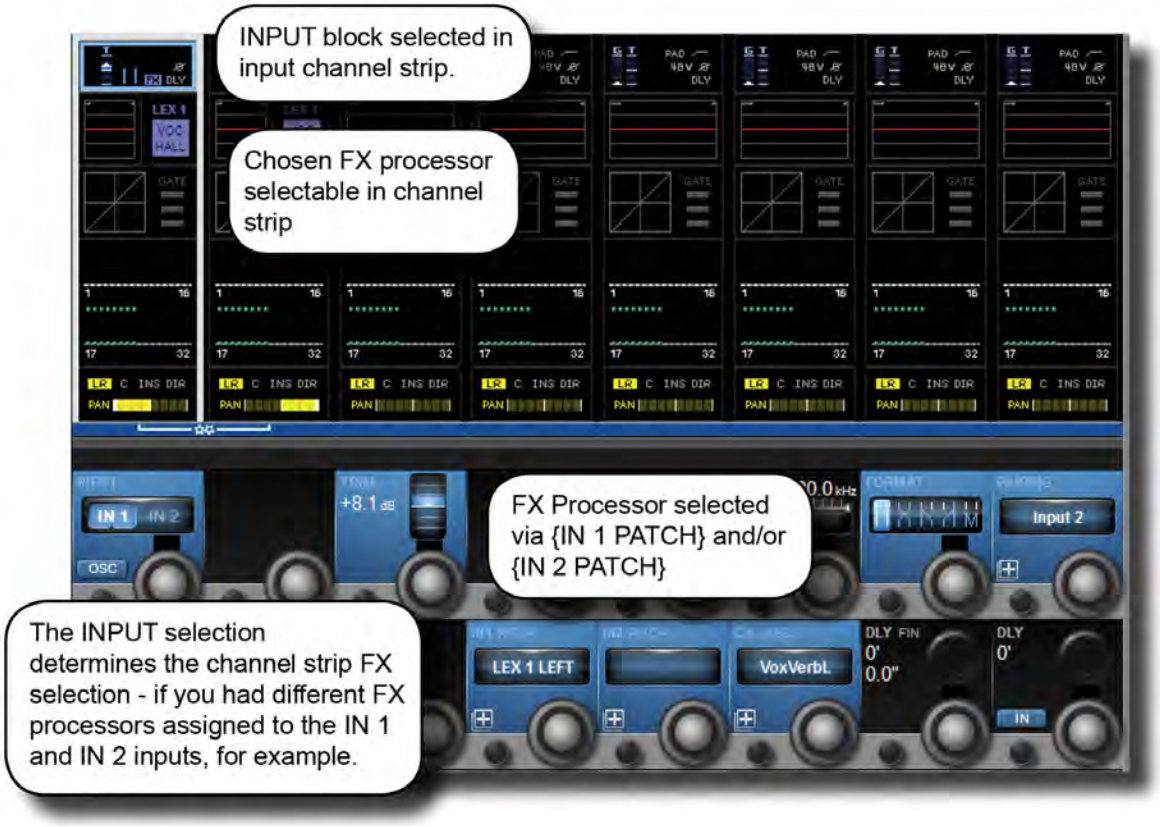
If you patch the main output of the aux bus to the input of an FX unit, you then need to select an input to patch to the output of that FX unit.

LEXICON FX > USING THE FX

Touch the Insert/output/delay block in the channel strip, then use the BUS OUT PATCH or INS PATCH options in the VST section to bring up the relevant patching options onto the touch screen. Simply select the Lexicon In page from the options on the right hand side of the touch screen and then select the required FX unit.

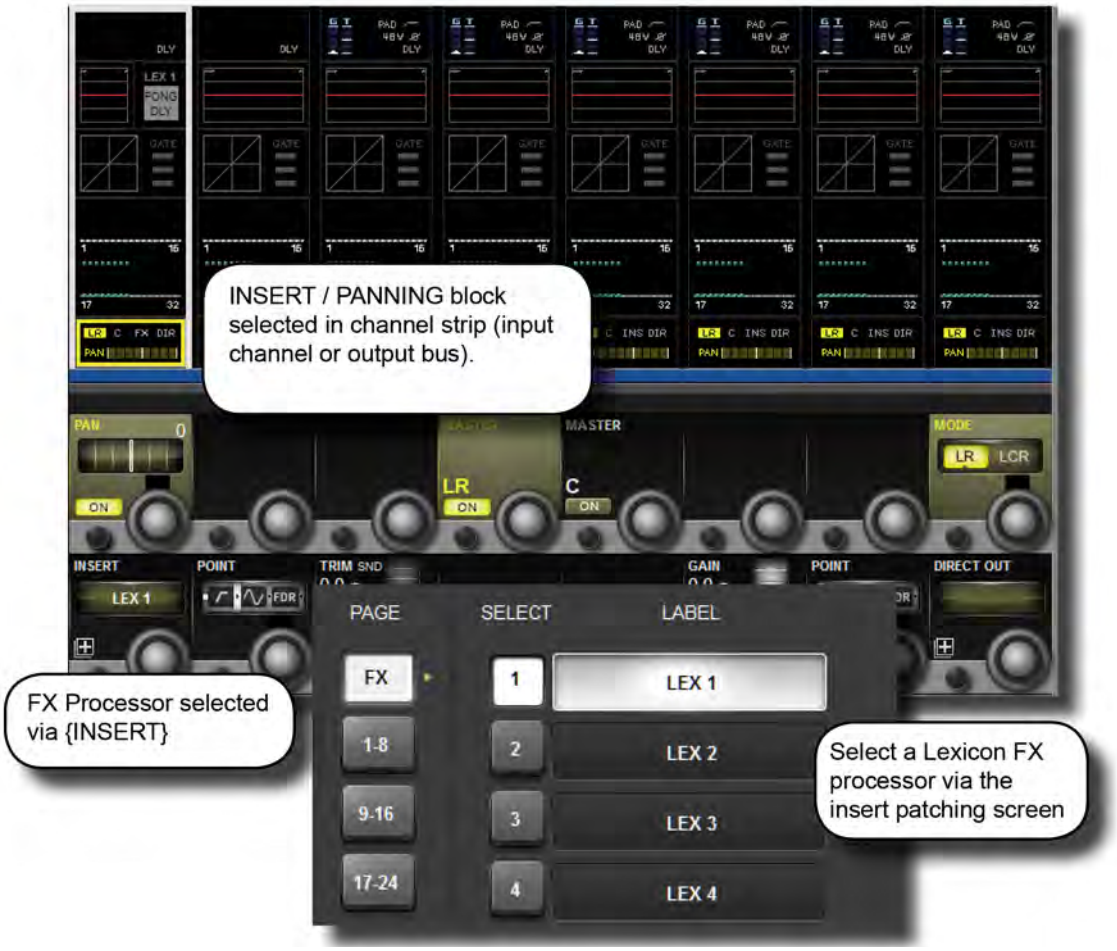
If you are patching the output of a stereo aux bus, then left and right buttons will appear in the patch screen. For normal routing you should assign each to the corresponding inputs of an FX unit.

As An FX Return



To patch an input channel as an FX return, select that input's input block from the top of the touch screen channel strip, then use either the {IN 1 PATCH} or {IN 2 PATCH} option to assign the output of an FX unit to that channel.

As An Insert



To patch an FX unit into the insert point of an input channel, select the Panning/Insert block from the bottom of the touch screen channel strip, then use the {INSERT} patch option to select an FX Unit. Touch the (FX) button on the top left of the touch screen to show the available FX selections.

Once assigned you will be able to edit the selected FX unit directly by touching the FX block in the input channel's touch screen channel strip.

LEXICON FX > TAP TEMPO

Tap Tempo Functionality



For each effect that offers TAP Tempo (Tempo synchronisation using key press), the bottom left key {TEMPO} is used as the TAP button.

In live situations it is often advantageous to be able to easily control the TAP function from a button that is permanently accessible on the console surface. The large F1-6 keys below the master screen are assigned by default on to the tap function of FX1-6, so no further setup is needed as long as shows are built from the console's internal default shows. However if you are importing a show from a Vi 4 or 6 console, only the first two of these F keys may be mapped to the tap function. The information that follows will allow any F key to be mapped to the tap function of any FX unit in the event that a Vi4 or 6 show is being used.

In order to assign the TAP functions of FX units to the F-keys, proceed as follows (refer to the screenshot above):

. Press the MENU button and select the {GPIO} menu tab.

. Ensure the {LOCAL I/O} button is selected and scroll the input and output sections down to the VGPI and VGPO Pin settings. For LEX3, select VGPI Lex Tap3 and VGPO Lex Tap3.

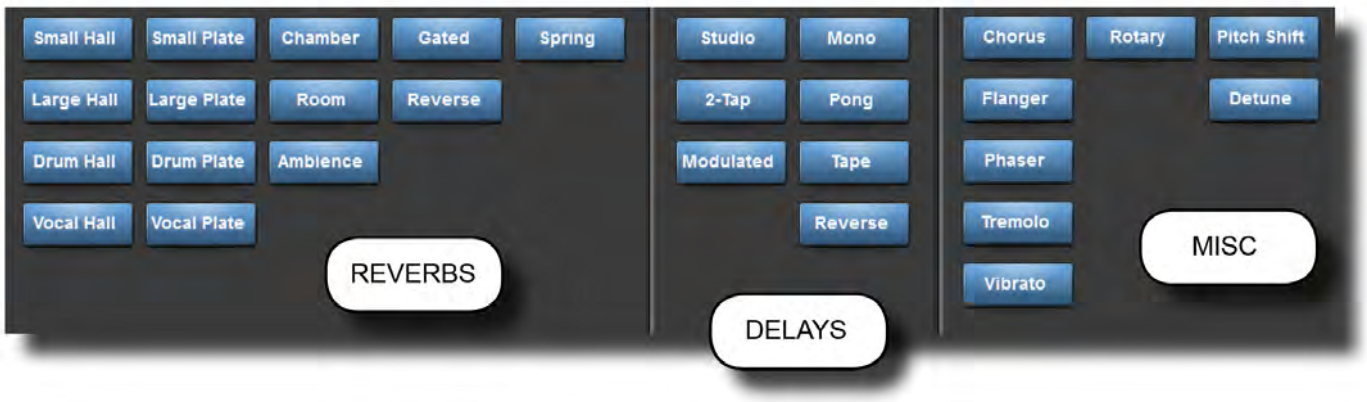
. Set the parameters for the VGPI and VGPO as shown in the above picture, and ensure the input and output are switched ON.

Use a similar procedure for other FX units.

NOTE: Normal F-Key setup functions will be over-ridden when an FX unit with Tap-Tempo function is selected in the corresponding FX unit. That is, if F1 is setup as a short-cut to a screen, the default tap-tempo VGPIO tap function for FX unit 1 will be active when FX1 has a delay selected.

LEXICON FX > ALGORITHMS

FX Algorithms and Parameters



For each of the FX processors an individual FX Type can be selected. The FX Types are grouped into the following categories: Reverb, Delay, and Miscellaneous. You can select algorithms by hitting the {TYPE} button in the VST section of any FX page.

Reverbs

Reverberation (or “reverb” for short) is the complex effect created by the way we perceive sound in an enclosed space. When sound waves encounter an object or boundary, they don’t just stop. Some of the sound is absorbed by the object, but most of the sound is reflected or is diffused. In an enclosed space, reverb is dependent on many features of that space, including the size, shape and the type of materials that line the walls. Even with closed eyes, a listener can easily tell the difference between a cupboard, a locker room and a large auditorium. Reverb is a natural component of the acoustic experience, and most people feel that something is missing without it.

Halls

SMALL HALL, LARGE HALL, DRUM HALL, VOCAL HALL - Stereo

A Hall reverb is designed to emulate the acoustics of a concert hall – a space large enough to contain an orchestra and an audience. Because of the size and characteristics,Halls are the most natural-sounding reverbs, designed to remain “behind” the direct sound – adding ambience and space, but leaving the source unchanged. This effect has a relatively low initial echo density which builds up gradually over time.

Vocal Hall and Drum Hall reverbs are specifically tailored for those uses. Vocal Hall has as lower overall diffusion which works well with program material that has softer initial transients like a voice. Drum Hall has a higher diffusion setting which is necessary to smooth out faster transient signals found in drums and percussion instruments.

In addition to general instrumental and vocal applications, the Hall program is a good choice for giving separate tracks in a mix the sense of belonging to the same performance.

Plate Reverbs

SMALL PLATE, LARGE PLATE, DRUM PLATE, VOCAL PLATE - Stereo

A Plate reverb is a large, thin sheet of metal suspended upright under tension on springs. Transducers attached to the plate transmit a signal that makes the plate vibrate, causing sounds to appear to be occurring in a large, open space. The Plates in the Vi4 FX units model the sound of metal plates with high initial diffusion and a relatively bright, coloured sound. Plate reverbs are designed to be heard as part of the music, mellowing and thickening the initial sound. Plate reverbs are often used to enhance popular music, particularly percussion.

Chamber Reverb

Stereo

Historically, recording studio chambers were oddly shaped rooms with a loudspeaker and set of microphones to collect ambience in various parts of the room. Chamber programs produce even, relatively dimensionless reverberation with little colour change as sound decays. The initial diffusion is similar to the Hall programs. However, the sense of size and space is much less obvious. This characteristic, coupled with the low colour of the decay tail, makes these programs useful on a wide range of material - especially the spoken voice, to which Chamber programs add a noticeable increase in loudness with low colour.

LEXICON FX > REVERBS

Room Reverb

Stereo

Room produces an excellent simulation of a very small room which is useful for dialogue and speech applications. Room is also practical when used judiciously for fattening up high energy signals like electric guitar amp recordings. Plate Reverbs

Ambience Reverb

Stereo

Ambience is used to simulate the effect of a small or medium sized room without noticeable decay. It is often used for voice, guitar or percussion.

Spring Reverb

Mono In/*Stereo Out*

A Spring reverb is created by a pair of piezoelectric crystals—one acting as a speaker and the other acting as a microphone—connected by a simple set of springs. The characteristic ‘boing’ of a spring is an important component of many classic rock and rockabilly guitar sounds.

Reverse Reverb

Mono In/*Stereo Out*

Reverse reverb works in the opposite fashion from normal reverb. Whereas a normal reverb has the loudest series of reflections heard first that then become quieter over time, the Reverse reverb has the softest reflections (essentially the tail of the reverb) heard first, and then grows louder over time until they abruptly cut off.

LEXICON FX > REVERBS

Reverb Parameters



Pre Delay

Adjust time delay between the source signal and the onset of reverberation.

This control is not intended to precisely mimic the time delays in natural spaces, as the build-up of reverberation is gradual, and the initial time gap is usually relatively short. For the most natural effect, the Pre Delay values should be set in the range of 10-25 milliseconds. However, if a mix is very busy or overly cluttered, increasing the Pre Delay time may help clarify it, and set each instrument apart from each other.



Reverb Time

Adjust the amount of time the reverb can be heard.

Higher settings increase reverberation times which are usually associated with larger acoustical environments, but can decrease intelligibility. Lower settings shorten reverb times and should be used when a smaller apparent space or a more subtle effect is desired.



Size

Build-up rate of diffusion after the initial period (which is controlled by Diffusion).

The Size control changes reverb sound from very large to very small. Generally, set this control to the approximate size of the acoustic space being created, before adjusting anything else. The size in meters is roughly equal to the longest dimension of the space. Audio is temporarily muted when Size is changed.



Diffusion

Initial echo density.

High settings of Diffusion result in high initial echo density, and low settings cause low initial density. In a real-world situation, irregular walls cause high diffusion, while large flat walls cause low diffusion. For drums and percussion, try using higher Diffusion settings.



Shape

Adjust the ‘contour’ of the reverberation envelope.

With Shape all the way down, reverberation builds explosively, and decays quickly. As Shape is advanced, reverberation builds up more slowly and sustains for the time set by Spread. With Shape in the middle, the build-up and sustain of the reverberation envelope emulates a large concert hall (assuming that Spread is at least halfway up, and that Size is 30 meters or larger).

LEXICON FX > REVERBS



Spread

Adjust reverb onset and sustain.

Low Spread settings result in a rapid onset of reverberation at the beginning of the envelope, with little or no sustain. Higher settings spread out both the build-up and sustain.



Reverb High Cut

Set the frequency of a 6dB/octave low-pass filter on the reverberated signal.

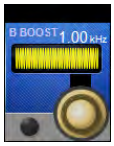
It does not attenuate the reflections. High frequencies are often rolled off with this parameter, resulting in more natural-sounding reverberation. Setting a low frequency for this parameter can actually shorten the reverb time, as it damps the audio as it recirculates.



High Cut

Adjust the amount of high frequency content in the reverberation tails.

Higher frequency settings increase high frequency response, creating brighter reverbs; lower frequency settings create darker reverbs with more bass frequency emphasis.



Bass Boost Frequency

Set the frequency at which the transition from Mid Rt to Low Rt takes place

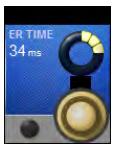
This control should be set at least two octaves higher than the low frequency you want to boost. For example, to boost a signal at 100Hz, set Bass Boost Frequency to 400Hz. (This setting works well for classical music.) Crossover works best around 400Hz for boosting low frequencies, and around 1.5 kHz for cutting low frequencies.



Bass Boost Ratio

Boost or cut frequencies below the Bass Boost Frequency.

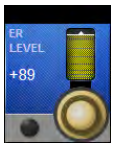
The amount of boost or cut required is highly dependent on the material being processed.



Early Reflection Time

Adjust the amount of time before reverb early reflections occur.

LEXICON FX > REVERBS



Early Reflections Level

Adjust the level of early reflections within the reverb.



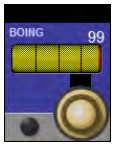
Feedback Delay

Change the resonant frequencies of Plate reverb.



Feedback Level

Adjust a Plate reverb's presence and prominence.



Boing

Increase or decrease the 'spring rattle' in a spring reverb.

This is a physical characteristic of spring tank reverbs.

LEXICON FX > DELAYS

DELAYS

Delays repeat a sound a short time after it first occurs. Delay becomes echo when the output is fed back into the input (feedback). This turns a single repeat into a series of repeats, each a little softer than the last.

Studio Delay

Stereo

The Studio Delay features up to 1 second of stereo delay and offers a built-in ducker that attenuates the delay output whenever signal is present at the input. This can be used to keep the original signal from being muddied up by delay repeats.

2-Tap Delay

Stereo

The 2-Tap Delay is probably best described as an adjustable pong delay where each tap can be individually set in relation to the delay time. The 2 taps are a calculated percentage of the actual delay time from 1-100% (for example, if the delay time is 500ms and Tap 1 is set to 50% and Tap 2 is set to 100%, Tap 1 time would be 250ms and Tap 2 time would be 500ms). Narrow spacing of the tap percentages can widen the stereo image of the delay while wider tap spacing can create rhythmic delay lines.

Modulated Delay

Stereo

The Modulated Delay is enhanced by an LFO (low frequency oscillator) that produces a chorusing effect on the delay repeats. This is a great delay for guitar and instrument passages that need that “special something.”

Mono Delay

Mono In/Stereo Out

The Mono Delay is the cleanest, most accurate of the delay programs, with up to 1 second of mono delay with panned output, and the built-in ducking feature.

Pong Delay

Mono In/Stereo Out

This delay effect pans the delay repeats from left to right, while the input signal remains at its original(centre) position.

LEXICON FX > DELAYS

Tape Delay

Mono In/Stereo Out

In the days before digital, delays were created using a special tape recorder in which the magnetic recording tape was looped, with closely-spaced recording and playback heads. The delay effect was created by the tape moving in the space between the record and playback heads – while delay time was adjusted by changing the speed of the tape loop. Although very musical-sounding, wow and flutter combined with a significant loss of high frequencies, and to some extent also low frequencies, are all elements commonly associated with tape recordings.

Reverse Delay

Mono In/Stereo Out

This delay effect emulates the old studio trick of flipping a tape over, playing it backwards through a tape delay, and recording the effect. The delays “build up” from softer to louder – creating the sensation that the delays come before the signal.

Delay Parameters



Tempo

Enter the ‘Tempo’ Time.

This time is expressed as tempo in BPM (beats per minute). Tempo works in conjunction with Delay Time to set the actual delay time that is heard.



Tap Tempo

Enter the Tempo time by tapping.

Repeatedly tap the {TAP] button to automatically set the delay tempo. The console ‘F’ keys 1-4 are assigned to the delay tap tempo.



Delay Time

Controls the length of the delay time relative to Tempo.

At the middle of its range, delay repeats are synchronous with the Tempo button; lower values create faster repeats, while higher values increase the time between repeats.



Delay Type

Choose between Tape and Digital delay types.

2-Tap Delay algorithm only.

LEXICON FX > DELAYS



Tap Ratio

Controls the Tap ratio of left and right outputs relative to the Delay time.

Specific to the Pong Delay effect, this controls the delay time offset between left and right taps, expressed as a ratio of left to right time difference. In the centre the taps have the same time (1:1).



Tap Swap

Swaps the delay time tap offset set by the Tap Ratio.

Specific to the Pong Delay effect - reverses the offset set by the Tap Ratio control.



Delay Feedback

Controls the number of delay repeats by feeding the delay output signal back into the delay input.

This creates a series of delay repeats, each slightly attenuated until they become inaudible. Higher settings create more repeats; lower settings reduce the number of repeats. When this knob is turned fully clockwise, it engages Repeat Hold – delay repeats play back in an infinite loop, but no further input signal is introduced into the delay effect. Repeat Hold is available only on Studio, Mono and Pong Delay.



Low Cut Filter Frequency

Frequencies below this level are attenuated.



High Cut Filter Frequency

Frequencies above this level are attenuated.



Smear

Ducking attenuation amount.

Available only for Tape and Reverse Delays, this parameter controls the amount of “smear,” or signal degradation and frequency loss. The higher the setting, the more each delay repeat loses intelligibility compared to the original signal.

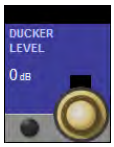
LEXICON FX > DELAYS



Ducker Threshold

Set the input signal level at which ducking occurs.

Studio, Mono and Pong delays only. Ducking causes delay repeats to be attenuated by a variable amount when an input signal is present. As the performance pauses, the delay signal level returns to its normal setting. This allows the delay to remain as an effect, but not clash with the original signal. For example whilst a vocalist is singing, the level of delay is kept down, but in the pauses the level of the repeats is brought up to provide a smooth tail to the vocal phrases. The Ducker Threshold sets the level at which the input signal has to be at for ducking to cut in – the higher the threshold, the louder the signal has to be for ducking to occur.



Ducker Level

Ducking attenuation amount.

Ducker Level sets the amount of attenuation once the signal has exceeded the threshold. 0dB is no ducking, 18dB is the maximum amount of ducking to the delayed signal.



TAP 1 & 2

Tap delay time relative to displayed delay time value.

2-Tap Delay only. Sets the percentage of the indicated delay time that is used for each tap.



Tap Level

Adjusts the output level of Tap 1 and Tap 2.

Specific to the Pong Delay effect - reverses the offset set by the Tap Ratio control.



Tap Pan

Adjust the pan positions in the stereo field of Tap 1 and Tap 2.



Modulation Depth

Control the intensity of modulation, or “depth” in the Modulated Delay.

Lower settings produce a more subtle chorus effect, while higher values give a more lush chorusing of the delay repeats.

LEXICON FX > MISC FX

Miscellaneous FX

The MISC category provides primarily modulated and pitch-varying effects.

Chorus

Stereo

Chorus creates a lush, full sound by combining two or more signals together where one is unaffected and the other signals vary in pitch very slightly over time. Chorus is commonly used to fatten up tracks and to add body to guitars without colouring the original tone. Chorus can also be used with discretion to thicken a vocal track.

Flanger

Stereo

This effect was originally created by simultaneously recording and playing back two identical programs on two tape recorders, then using hand pressure against the flange of the tape reels to slow down first one machine, then the other. The result was a series of changing phase cancellations and reinforcements, with characteristic swishing, tunnelling, and fading sounds.

Phaser

Stereo

The Phaser automatically moves frequency notches up and down the spectrum of the signal by means of a low frequency oscillator (LFO), creating an oscillating “comb filter” type effect. This effect is very useful on keyboards (especially pad presets) and guitars.

Tremolo

Stereo (Wet Only)

Tremolo/Pan creates rhythmic changes in signal amplitude. Tremolo is obtained by setting Phase to 0 degrees, and affects both channels’ amplitude simultaneously. If the Phase is set to 180 degrees, an AutoPanner effect is generated, with the amplitude of one channel being raised whilst that of the other channel is lowered. Speed settings below 1Hz are recommended in this case.

Vibrato

Stereo (Wet Only)

Vibrato is obtained by smoothly varying the pitch of the signal just sharp and flat of the original at a determined rate. Phase controls whether the pitch of both channels is modulated together, or in an opposite direction.

LEXICON FX > MISC FX

Rotary

Mono In/Stereo Out (Wet Only)

Rotary speaker cabinets were designed to provide a majestic vibrato/choir effect for electronic theatre and church organs. The most well known rotary speaker is the Leslie™ Model 122, which has two counter-rotating elements: a high-frequency horn and a low-frequency rotor with slow and fast speeds. The sound generated as the spinning elements change speed is truly magical. The swirling, spacious effect is difficult to describe – but clearly recognizable.

The Rotary effect is modelled after a Leslie-style cabinet. The input signal is split into high and low-frequency bands. The rotation effect is created by a synchronized combination of pitch shifting, tremolo, and panning. Like the physical cabinet, the high (horn) and low (rotor) frequencies are “spun” in opposite directions. Horn and rotor speeds are independent, and designed with acceleration and deceleration characteristics to simulate the inertia of the original mechanical elements. A virtual necessity for organ music, Rotary also sounds remarkable with guitar and electric piano rhythm parts. In fact, this program is a great alternative to the Chorus and Tremolo effects for any sound source.

Pitch Shift

Stereo

This effect shifts the frequency spectrum of the input signal. Altering the pitch of a sound produces a wide range effects - from subtle detunes to full interval shifts up or down a two octave range. The Pitch Shift effect is a chromatic shifter, meaning all notes of the scale are shifted by the same interval. Pitch Shift is very useful with guitar tracks, monophonic synth lines, or where special vocal effects are needed.

Detune

Stereo

Detune adds a slightly pitch-shifted version of the original source, thickening the sound. This creates a particularly effective simulation of “double-tracking.” This effect is also a great alternative to the Chorus effect, adding the richness of a chorus without the audible sweep caused by the chorus rate. It is also useful for creating a wide stereo signal from a mono source, by setting a small detune amount up on one output and down on the other, and panning the two outputs hard left and right.

Misc FX Parameters



Speed

Sets the speed at which the modulated effect cycles.



LFO Depth

Scales the intensity of the effect.

This control affects the output of the LFO only. It has no effect on the outputs of the individual waveforms.

LEXICON FX > MISC FX



Number Of Voices

Controls the number of additional Chorus voices.

Higher amounts add more richness to the Chorus effect.



Regeneration

Controls the amount of modulated signal being fed back into the input, creating feedback.

Higher amounts add more resonance to the signal.



Diffusion

Creates a time-smoothing effect similar to diffusion in reverb.

Diffusion can be a subtle effect to add a little warmth to the chorus.



Pre Delay

Determines the amount of offset between the two signals that create the flange effect.

Lower values create a tighter effect, higher values result in a more extreme “whooshing” sound.



LFO Waveform

Selects the wave pattern used by the modulated effect.

Select from sine wave, triangle wave, and random.



Phase

Controls whether amplitude or depth change occurs in both left and right outputs simultaneously or alternates between left and right outputs.

This control affects the output of the LFO only. It has no effect on the outputs of the individual waveforms.

LEXICON FX > MISC FX



Phase Stages

Selects between a 4-, 8-, or 12-state phase shifter.



Horn Min

Rotary effect only: Minimum speed of the HF rotary horn.

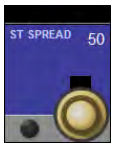
The speed limits are used when the speed parameter is changed from Slow to Fast.



Horn Max

Rotary effect only: maximum speed of the HF rotary horn

The speed limits are used when the speed parameter is changed from Slow to Fast



Stereo Spread

Increases or decreases the stereo imaging of the Rotary effect.



Overdrive Gain

Adjust overdrive gain to the preamp section of the rotary speaker effect.



Minimum Rotor Speed

Sets the minimum speed at which the LF Speaker will rotate.

Rotary effect only: The speed limits are used when the speed parameter is changed from Slow to Fast

LEXICON FX > MISC FX



Maximum Rotor Speed

Sets the maximum speed at which the LF speaker will rotate.

Rotary effect only: The speed limits are used when the speed parameter is changed from Slow to Fast



Doppler Effect

Increases or decreases the Doppler pitch effect.

The Doppler effect is a rise and fall in pitch experienced by an observer when a sound source is moving towards or away from them, due to a compressed or expanded wavefront.



Pitch Shift

Determines the amount of pitch shift or detune shift from the original signal source.

Works best with individual notes.



Shift Delay

Sets the delay time before the pitch shift or detune effect is heard in the Pitch Shift and Detune effects.

This control affects the output of the LFO only. It has no effect on the outputs of the individual waveforms.



Feedback

Adjusts how much of the shifted signal is sent back through the delay line in Pitch Shift and Detune.

For creating cascading arpeggio-type effects.



Pan

Sets the pan position in the stereo field for each tap in the 2-Tap Delay or each channel of the pitch shift/detune effects.

Allows the stereo width to be increased by panning versions of the original signal that have been shifted slightly up and down, to left and right channels respectively.

VM ² WIRELESS SYSTEM MONITORING



VM² is a feature that allows the status information for HiQnet-enabled AKG wireless microphone systems and Shure ULXD / QXLD wireless microphone systems to be displayed directly within the channel strip that they are connected to. In addition Shure receiver gain and mute can be controlled directly from the Vi channel strip. This feature takes advantage of Harman's HiQnet network control protocol (AKG) or Shure's ACN protocol and brings many benefits including streamlining of the workflow and increase in speed of problem diagnosis.

Compatible wireless systems can be connected to the console using the HiQnet socket on the back of the control surface via a standard 1GB Ethernet switch. Compatible AKG systems use the AKG Hub4000Q merge device. With Shure systems, a suitable Ethernet switch is used to connect the HiQnet port of the console to the Shure's Dante audio & control Ethernet port.

The HiQnet / Ethernet network setup is located in the main menu ([MENU]) under the (System) tab.

Available VM² status' and controls are shown in the input channel's input block in the touch screen, and in the VST section when that input block is selected. User definable receiver channel names appear in the channel label display.

VM ² WIRELESS SYSTEM MONITORING



HiQnet / 1GB Ethernet Connector

The HiQnet port is an Ethernet connection on the rear of the control surface, adjacent to the rear USB sockets.

The port is capable of driving standard Ethernet distances of up to 100m.

This port is used for connecting to:
AKG wireless receivers via Hub4000Q.
Shure ULXD/QLXD wireless receivers via the receiver's Dante port (this connection uses Shure ACN protocol, rather than HiQnet).
Wireless access point or router for connection of ViSiRemote iPad remote control app.
Soundcraft Realtime Rack for snapshot store/recall datalink.
A Harman network for transmission of Venue Preset commands and reception of HiQnet error messages.

An Ethernet switch should be used where more than one of the above connections are required

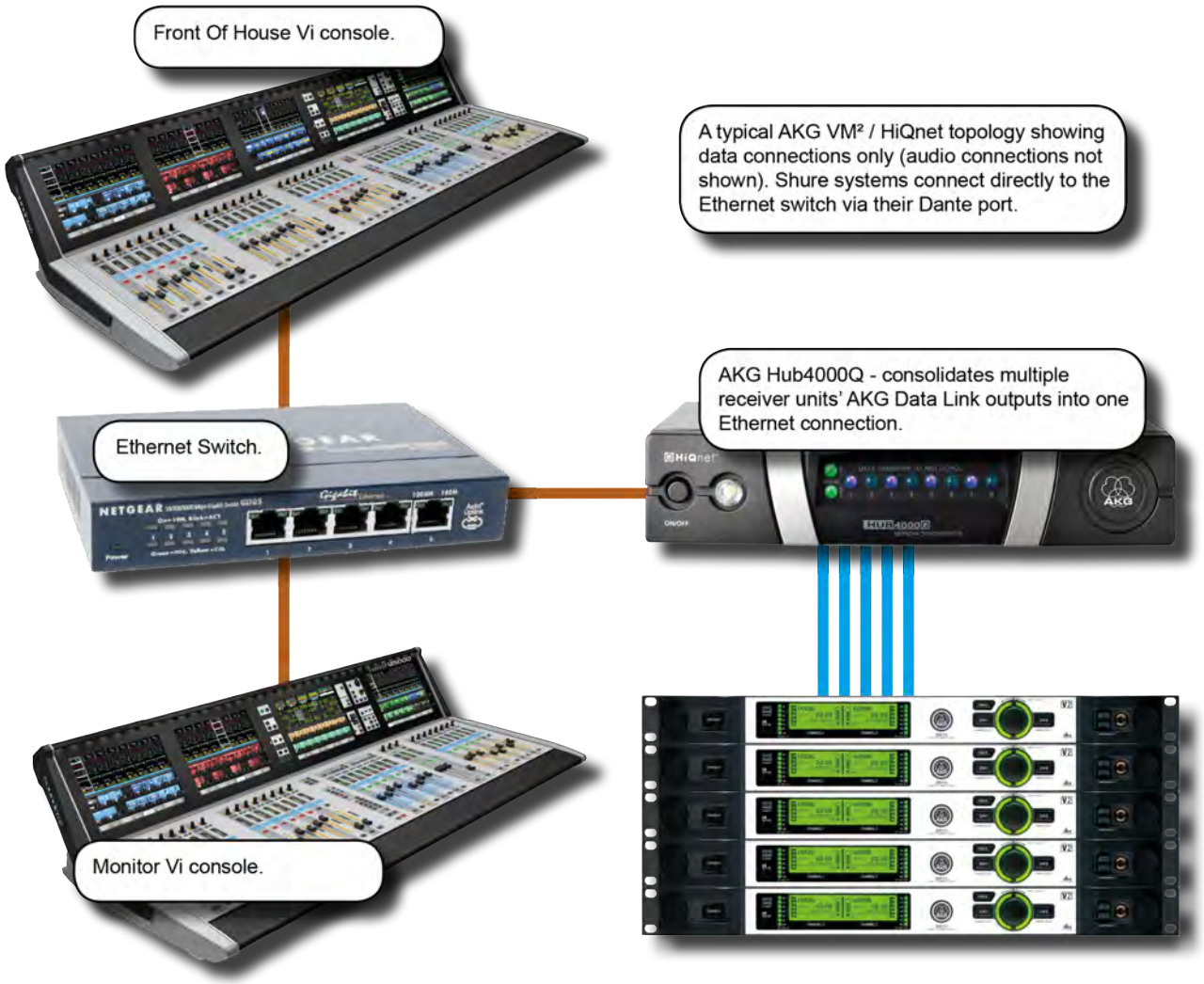


Menu Button

The [MENU] button opens the console's main menu system in the Master Bay touch screen area.

The console's HiQnet/Ethernet configuration can be accessed by pressing the main [MENU] button then the SYSTEM tab and HIQNET layer.

VM ² > HiQnet SYSTEM CONFIGURATION



The picture shows a typical AKG / HiQnet VM² set up. Shure systems can be connected directly to the Ethernet switch and both Shure and AKG systems can be connected at the same time. If only one console is being used, and there is no other HiQnet or Ethernet-connected equipment, the Hub4000Q can be directly connected to the console without using a switch, but in most cases it is better to use one. A computer running HiQnet System Architect software may also be connected to the switch, and if a wireless router is included in the network, the AKG wireless iPhone app may be used for additional monitoring of the microphone data. The same wireless router or access point can be used to connect the ViSi Remote iPad app for remote control of the console.

Setup of the console involves basic network setup (console IP address and HiQnet Node/Device ID) and assignment of detected devices to the physical inputs they are connected to.

VM ² > NETWORK CONFIGURATION



IP Configuration

Select either Dynamic (DHCP) or Manual (MAN) IP configuration.

If you are using a DHCP server to automatically configure the IP setup, wait for several seconds until an IP address appears in the IP ADDRESS Vistonics control fields. You will also see it on the far left of the HiQnet page.

After the mode has been changed, or after a manual address has been entered, a SET button will appear next to the rotary control and must be pressed to activate the change.



IP Address Entry

Set IP address for the console.

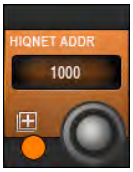
For use with Manual (MAN) IP Configuration. Use the row of VST encoders to dial-in the required IP address for the console. The available ranges of valid IP addresses are 10.0.0.0 - 10.255.255.255; 172.16.0.0 - 172.31.255.255; 192.168.0.0 - 192.168.255.255. Some IP addresses within the ranges show are not allowed due to conflicts with other parts of the Vi system – they will be greyed out and not available for selection.



Subnet Mask Entry

Set the subnet mask.

For use with Manual (MAN) IP Configuration. Use the row of VST encoders to dial-in the subnet mask.



HiQnet node address entry (console)

Set the console's HiQnet Node Address using the keyboard editor on the touch screen.

The HiQnet node address can be any number in the range 1 – 65,535 but it must be a unique number within the network in order to avoid conflicts with other HPro devices. You can use Audio Architect to find out what the node addresses of other equipment are. Note that it will take up to 10 seconds for the new address to appear after the Enter button on the keyboard is pressed, and the console will be unresponsive during this time –this is normal.



HiQnet On

Enable HiQnet functionality.

Once the IP address is set and the HiQnet node address has been confirmed as unique in the network, you can enable the HiQnet functionality by pressing the ON button in the far left Vistonics control. Enabling HiQnet will cause all the Vistonics controls in the section to change from greyed-out to orange. A list of information relating to the current HiQnet setup including a green 'OK' indicator is now displayed on the left side of the HiQnet Setup page, indicating a healthy connection.

VM ² > NETWORK CONFIGURATION > DEVICE LIST



The Device list occupies the centre of the Setup page, and will initially be completely empty. Once the network has been turned on using the WIRELESS button for Shure receivers or the HiQnet button for AKG receivers, the console will search for any attached wireless microphones and display a list of the microphones found on the network in the left-hand column of the Device List.

Note: It may take up to 30 seconds to discover AKG mics and up to 4 seconds for Shure mics.

The **DEVICE** column shows the names of all of the AKG or Shure microphones that have been discovered on the network. This column is automatically populated.

For AKG systems, the **NET** column in the list shows the HiQnet node address of the AKG Hub4000Q to which each microphone is connected. For Shure systems the NET column shows the receiver ID number.

It is possible for multiple Hub4000Qs and multiple Shure systems to be connected to the network, and in this case the number in the NET column will enable groups of mics attached to these different Hubs to be distinguished from one another.

The **SOURCE** column in the list indicates the connector to which each microphone is associated. In order for the console to know on which channel strip to display the VM² monitoring information, it is necessary for the user to manually associate each of the microphone devices with a physical connector, which will correspond to the connector that the microphone's audio output is connected to.

The LOC button activates the Locate function on the selected AKG or Shure receiver. Pressing this will cause the front panel display of the chosen receiver to flash, allowing it to be more easily identified among a rack of others.

VM ² > HiQnet CONFIGURATION > DEVICE LIST



MAP Devices (Scroll List)

Scroll through the list of available VM² devices.

Hint: To avoid opening and closing the Audio Patch page, the MAP DEVICES encoder can still be used to scroll through the Device List even whilst the Audio Patch page is still open. The name (truncated) of the currently selected Device can be seen in above the MAP DEVICES Vistonics scroll encoder.



Locate Devices

Activate the Locate function on the selected AKG receiver.

Pressing (LOC) will cause the front panel display of the chosen receiver to flash, allowing it to be more easily identified among a rack of others.



Patch Devices

Activate the VM² microphone patching screen.

The patching matrix allows all of the physical input connectors available within the Vi system to be seen and the relevant connector to be chosen to correspond to where the currently selected microphone is connected.
The chosen connector is shown in bright blue highlight, whereas connectors that are already associated with other AKG or Shure devices are shown as greyed-out. A greyed-out connector can still be chosen as an assignment for the
currently selected microphone, but a dialogue box will appear in this case to ask if you wish to reassign this connector.



Wireless

Activate wireless system detection.

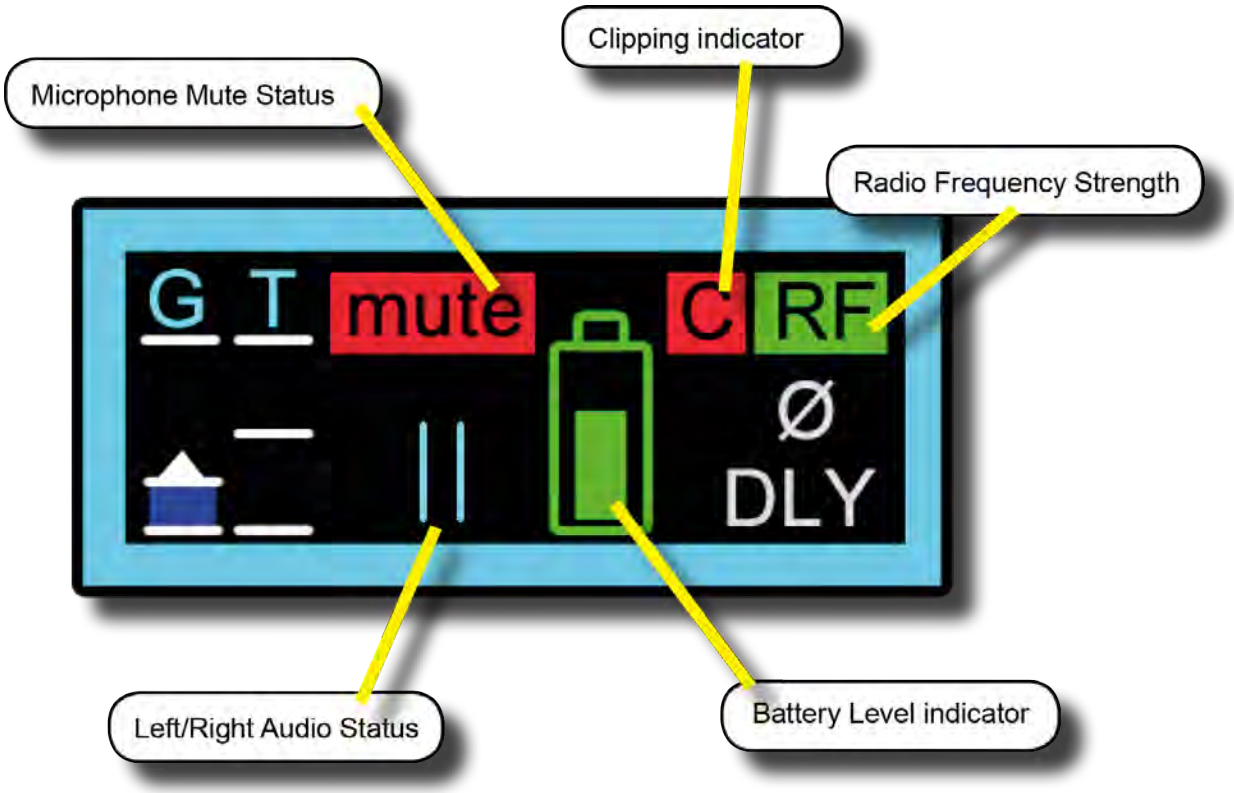
When on, the console will auto-detect Shure wireless systems. When off, detection of receivers is not possible; any previously detected units will be disabled and will be shown as greyed-out in the Device list. To discover AKG devices, use the HiQnet button.

VM ² > STATUS DISPLAYS



The console displays VM² wireless system status information in two places - the main channel strip display in the input channel touch screen area (input block), and in the VST area when the channel's input block has been selected.

VM² STATUS > TOUCH SCREEN INPUT BLOCK



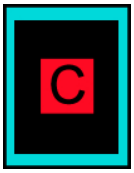
The input channel Input block graphic - shown at the top of the channel strip in the Vistonics touch screen area - provides instant feedback on the status of VM² microphones via icons for each of the monitoring elements.



Microphone Muted

Shows microphone muted state.

Shows red 'MUTE' icon if the mic is switched to muted state. When unmuted, this indicator icon disappears.



Microphone clipping

Clipping indication

A red 'C' is displayed momentarily if the audio within the wireless mic audio path reaches full scale.

VM² STATUS > TOUCH SCREEN INPUT BLOCK



RF Strength

Show the RF (Radio Frequency) status of the connected wireless microphone.

A red or green 'RF' icon is displayed to indicate the health of RF level. A green RF indicator is displayed when the signal strength at the receiver is strong enough to enable audio transmission. A red RF indicator is displayed when the signal strength is too low for audio transmission or if the transmitter if switched off. The actual value in dB of the RF signal strength is displayed in the VST section when the input block is selected.



Battery Level

Seven-segment transmitter battery level indication.

Red segments 0-1 indicate one hour or less remaining. The Amber segment (segment 2) indicates 25 percent of battery life remaining. The Green segments (segments 3-7) indicate 35-100 percent of battery life remaining. The number of hours remaining in the transmitter battery is displayed in the VTS section when the input block is selected and varies with the type of battery used.

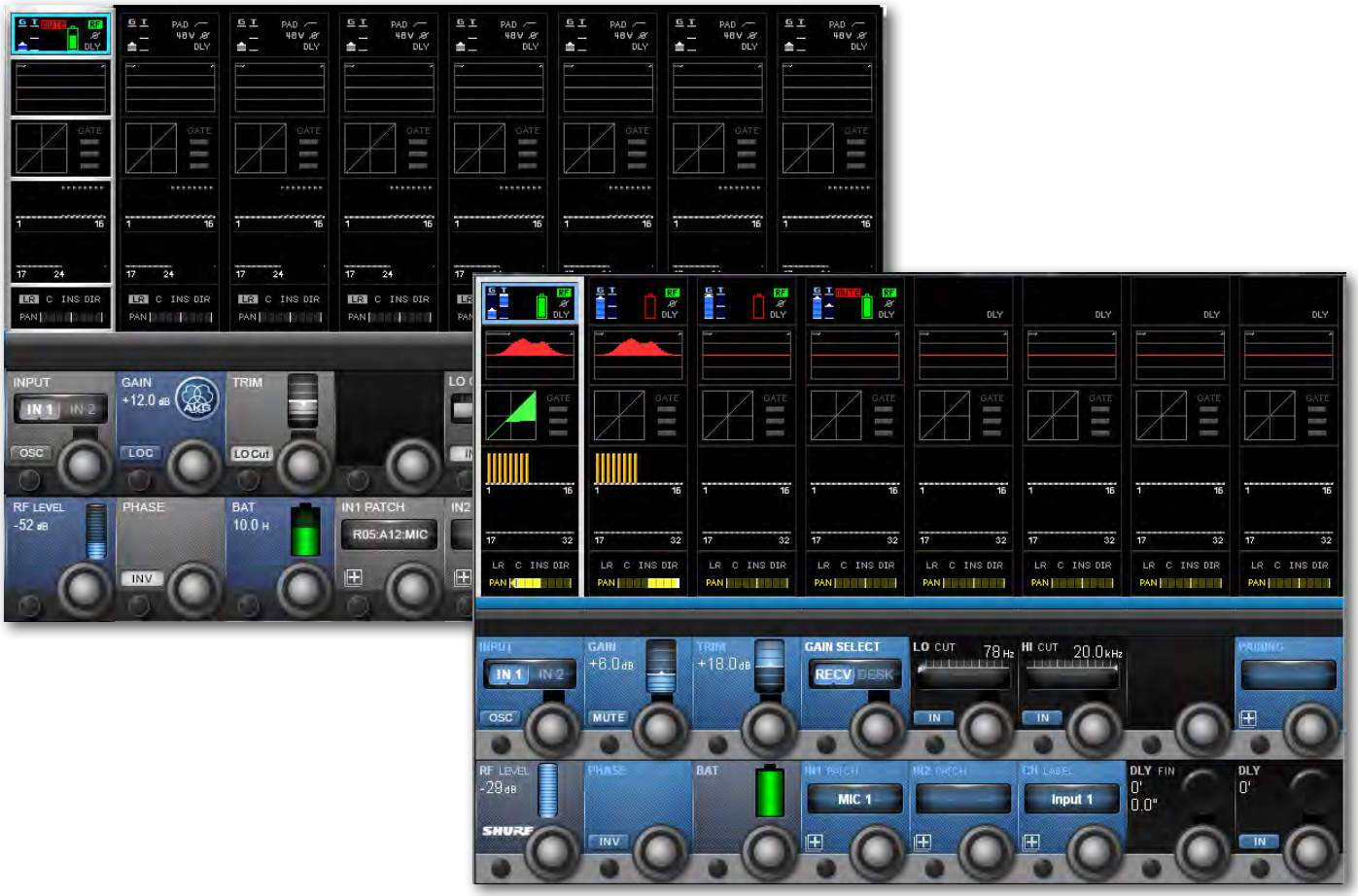


Network Error

Indicates a Network error (Grey Gain, Battery, and RF icons).

The system is able to differentiate between a network error/disconnection and out of range RF or a switched-off transmitter. In the case of a network error, the AKG or Shure elements only within the input strip overview on all channels will change to a greyed-out condition.

VM² STATUS > VST SECTION



To see more detailed information about connected microphones, the Input section of the channel strip can be zoomed by touching the Input touch field at the top of the strip. Three special Vistonics fields are added to the zoom view of channels that are patched to associated connectors, giving more detailed information of RF level (bargraph and numerical signal strength in dB), Battery level (bargraph icon plus numerical readout of remaining battery life in hours), and the internal gain of the microphone.

Note that for AKG systems this information is display-only, there is no control possible over the AKG microphones, other than with the Locate (LOC) button.

For Shure systems the Gain control operates the Receiver Gain control, and the Mute control operates the Receiver Mute function.



Gain Display (AKG)

Internal gain of the connected AKG microphone.

The display of internal Gain of the AKG microphone differs according to the type of microphone system being monitored. It cannot be controlled from the console. For WMS4500 systems the indicator displays the user-adjustable audio gain. For DMS700 systems the indicator shows the output trim level of the receiver.

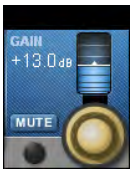
18.2.1: VM ² TOUCH SCREEN STATUS - VST ▶

VM² STATUS > VST SECTION



Mute (Shure)

Remotely mute the selected Shure wireless receiver.



Gain (Shure)

Remotely control the gain of the selected Shure wireless receiver.



Locate

Activate the Locate function on the selected AKG receiver.

Pressing this will cause the front panel display of the chosen receiver to flash, allowing it to be more easily identified among a rack of others.



Gain Select

Switch gain monitoring / control source

Choose RECV (wireless receiver) or DESK (input channel gain).



RF Strength (All Microphone types)

Radio Frequency signal strength.

Shows signal level in bargraph and dB readout.



Battery Level (All microhone types)

Remaining battery life as icon and numerical.

The remaining battery life numerical info in hours is only a guide, and depends on the type of cells used. The battery icon gives an accurate picture of the current state. Note that the remaining time for Shure receivers is only displayed if the mic has original Shure rechargeable batteries. If Alkaline batteries are used, the remaining time is not displayed.

Vi DANTE™ Card User & Setup Guide



The Soundcraft® Vi DANTE™ card is a 64 x 64 interface between a Vi series console and any Dante compatible device from Harman or other 3rd party manufacturer. The card is available in two versions: Local Rack/CSB (D21m 3U format) and Vi Stagebox (6U format). *Note that the Stagebox version is an IO card only and does not allow you to connect a Stagebox to the console using Dante instead of MADI.*

Dante provides a digital audio network that uses standard Internet Protocols over both 100Mbps and 1 Gigabit Ethernet. Dante uses audio independent, high accuracy network synchronization standards to ensure all Dante devices are synchronized at all times. Dante runs on standard computer networking hardware, and does not require dedicated network infrastructure.

The Audinate Dante Virtual Soundcard software allows connection of a PC/Mac to a Dante audio network. This system allows the computer to record from and playback to the Dante network using most common DAW packages. Dante Virtual Soundcard uses the Ethernet port on the computer to communicate with the Dante network; no special hardware is required other than installing Dante Virtual Soundcard software. One DVS license token is provided FOC with the card, this license can only be installed on one specific computer.

Additional information about the DANTE system including the latest software set-up and tools may be found on the Audinate® web site http://www.audinate.com/

Configuring The Dante Card

In a manner similar to CobraNet™, patching on the console makes a connection between console I/O slot channels and up to 64 channels on the Dante card. Connections between the Dante cards are set using a tool, in this case the Dante Controller software from Audinate®. As communication across the Dante network is essentially between named cards and channels the configuration must be correct to ensure cards talk and listen to the correct/desired channels on the Dante network.

This user guide summarises the process and provides a simple worked example; full setup details and configuration tools are downloadable from the Audinate web site www.audinate.com.

In addition it is necessary to set the clock status of the Dante card both within the network and in relation to the console hardware. This is described below.

Channel Count Setup

The Dante card uses 64 in and 64 out channels by default, but can be restricted to use less input or output channels if required. This may be necessary on Vi consoles in order to avoid exceeding the 192in/192out channel limit of the local rack (or the 64in/64out channel limit if fitted in a Vi or Compact Stagebox). For example, if only output channels are required on the Dante card, the input channels can be set to 0, to maximise the number of input channels available for other cards.

The DIP switch S1 on the Dante card main PCB is used to set the input and output channel count, in steps of 8chs from 0-32 channels. Refer to the diagram on the next page which shows the location of the channel count DIP switch S1 and the factory default setup of 64 in/64 out. To create other channel count settings that are not marked on the switch, add any two or 3 switch settings together, ie to select 48chs, the 32 and 16 switches should be moved down to the selected position. Note the maximum channel count (input or output) is 64 and minimum is zero.

Sync Setup

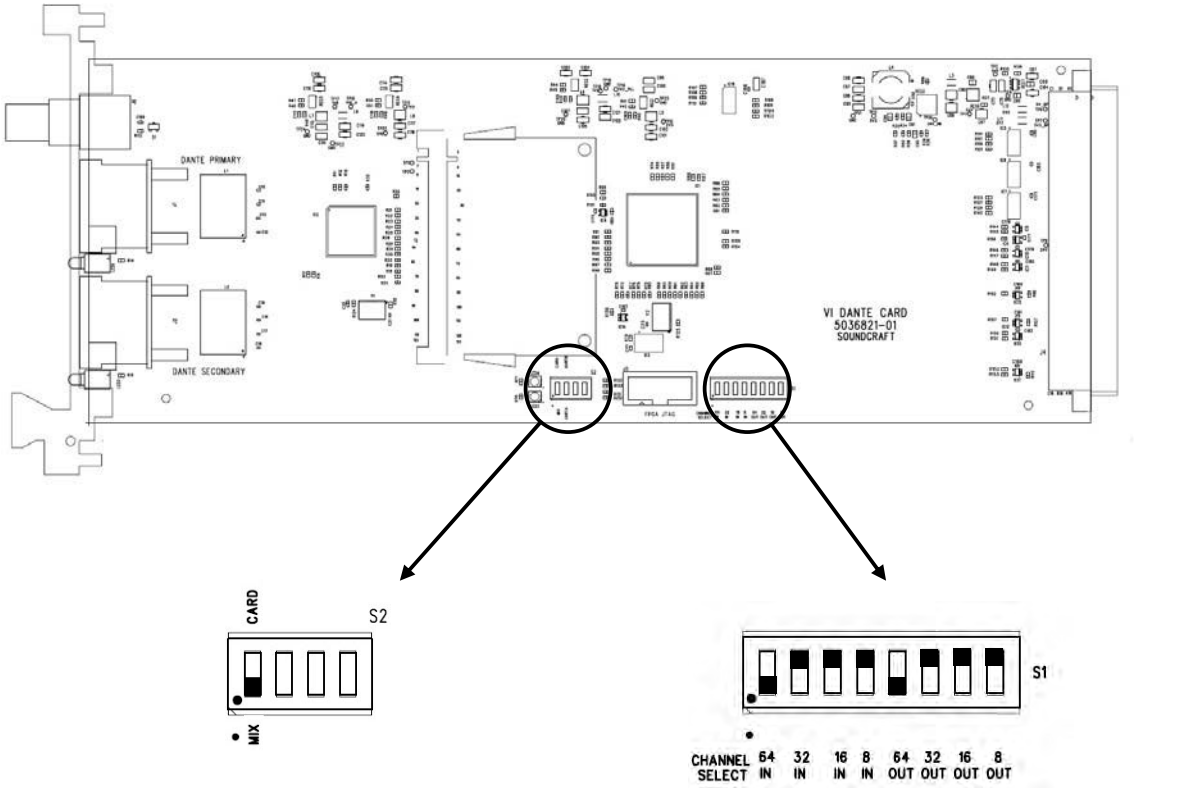
The Dante card can act as either the clock source for the Dante network or as a synchronised clock. If there is only one console on the network, this will generally be set as the clock source for the network, but in cases with more than one console, or where another device provides the clock source, it will be necessary to set the console's Dante card to be a synchronised clock. The Dante clock status is set up using Dante Controller software (see later in this guide), but in addition to the Dante clock setup, the card's clock status within the console has to be set. This is done using a DIP switch S2 on the Dante card main PCB. Refer to the diagram on the next page for the location and setting of this switch.

MIX: Select this position to make the Dante module lock to the console's internal Wordclock, only to be used when the Dante card is set as the Preferred Master for the Dante network. *This setting is the factory default.*

CARD: Select this position to make the console lock to the Dante network clock, to be used when the Dante card is set up as a synchronised clock on the network. Note: In this case, a BNC wordclock cable must be connected from the Wordclock Out BNC connector on the Dante card, to the console's Wordclock IN.

Note that it is also necessary to set up the clocking scheme of the Dante network correctly, to avoid audio clicks and dropouts, please refer to the section titled *Device Status* and *Clock Status* in the following pages.

DIP switches S1 & S2 Location Diagram



Note! Switches point **down** to select channel count – ignore the 'ON' legend on switch body!
Diagram shows the default setting of 64 in / 64 out

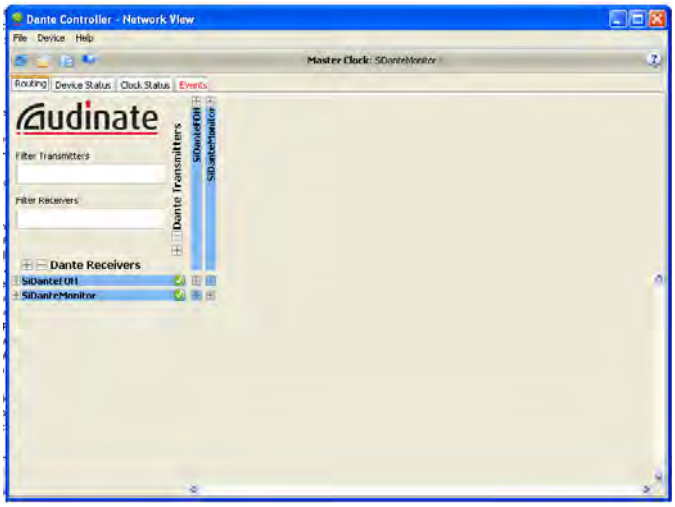
Refer to the previous page for information on the function of these switches.

Setting up the Dante card for operation on the network

As mentioned earlier, it is necessary to use a software tool connected externally to the card, in order to route audio from the card to/from another Dante-enabled device.

1. Download & install the latest software for the Vi consoles from www.soundcraft.com. (Software Version must be 4.7.5.205 or later). With earlier versions of the console software, the Dante card will not be recognised by the console and will not appear in the patch.
2. Download and install latest Dante Controller software from www.audinate.com.
3. Connect a Mac/PC to the Dante card, this may be via a switch/router or direct to either the PRIMARY or SECONDARY ports. The LINK indicator on the card will begin to flash indicating an active connection.
   NOTE: A crossover cable is not required as the network ports on the Dante card will automatically sort out the transmit and receive connections to the other device.
   NOTE: Ensure any firewall operating on the network port of the computer is switched off.

4. Start the Dante Controller application, after a few moments a screen similar to the one below will appear:

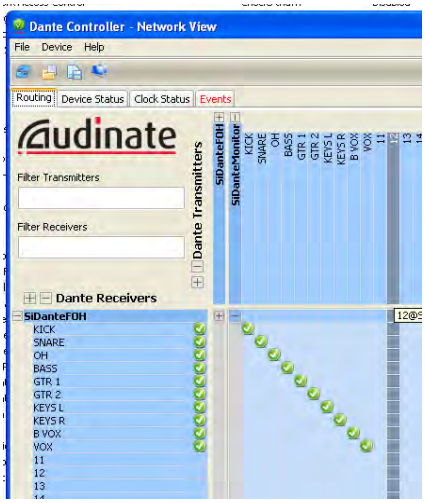


Using Dante

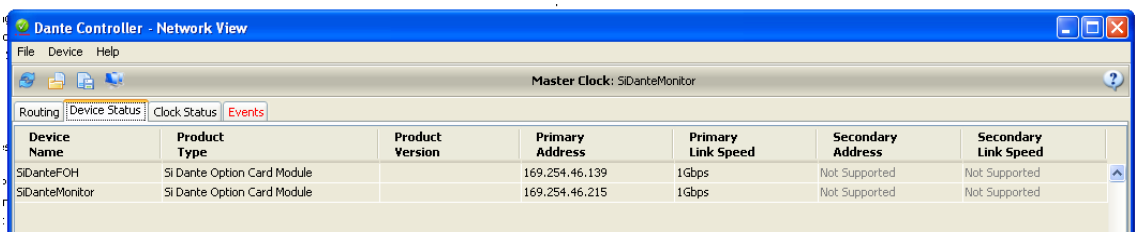
The following sections cover basic setup and use of the Vi Dante cards. For detailed information regarding Dante please refer to information on Audinate's web site www.audinate.com.

Network View

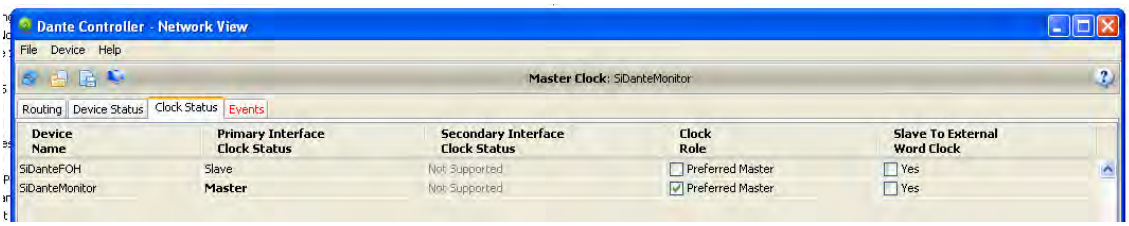
Routing – Matrix view of routing (click on a '+' symbol to expand the matrix view).



Device Status – Displays information about discovered Dante devices; double click a device name to enable its configuration.



Clock Status – View and edit Dante Network clock configuration.



Clock Role column:
Setting a single 'Preferred Master' will ensure that device is selected as the clock master, if two or more devices are set for 'Preferred Master' the master will be elected by the Dante system.
NOTE: Clock Status settings apply to the clocking of the Dante network, therefore attached devices such as consoles must be clocked to/from the Dante network as appropriate or required in any given system.
It is suggested that in most simple cases with one console connecting to peripheral equipment (eg a computer recording device) via Dante that the console is set as preferred master, which means that the item 'Sync to External Word Clock' must also be checked. If there are two or more consoles connected together then obviously only one should be set as preferred master.

'Sync to External Word Clock':
When checked the Dante card will clock from the console internal word clock, if not checked the console must be clocked to the Dante card or system word clock as appropriate or required in the given system. Normally this is achieved by connecting a short BNC jumper cable from the wordclock OUT of the Dante card, to the wordclock IN of the console.

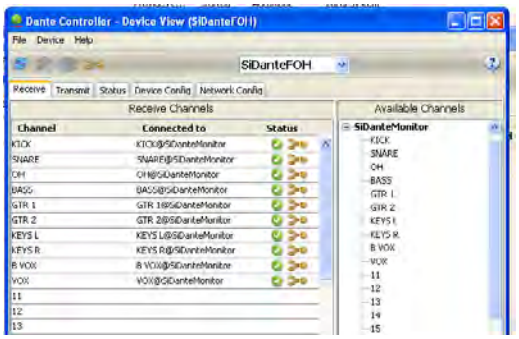
Note on Clock status when connecting the Dante card to a Vi3000 console's Dante interface

The Vi3000 console includes a built-in Dante interface, and as such has the capability for the console to automatically synchronise to the internal Dante card's clock, when the Vi3000's Dante interface status is set as a synchronised clock (ie another device is set as *Preferred Master*), and the setting '*Sync to External wordclock*' is NOT checked (ie the console is synchronised to the Dante network clock). Therefore with the Vi3000, an external BNC jumper cable is not required and the console will automatically switch to indicate 'Dante' as the clock source (viewable in the Menu-Sync page of the Vi3000).

Failure to set up the clocking scheme of the network correctly is likely to result in audio clicks during operation.

Device View

The Device View allows viewing and editing of a selected Dante device.



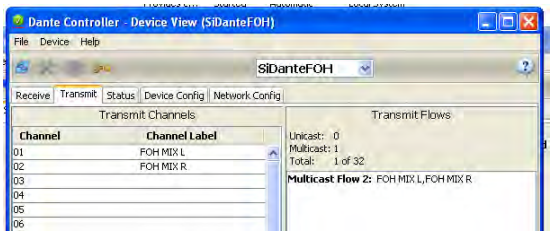
Receive – This tab presents information about all channels being received, and channels available to the current device. The tab is arranged into two areas:

Receive Channels: Lists the Rx channels in the left pane of the window. It shows whether or not they are connected and, if connected, the channel they are connected to and the status of that connection.

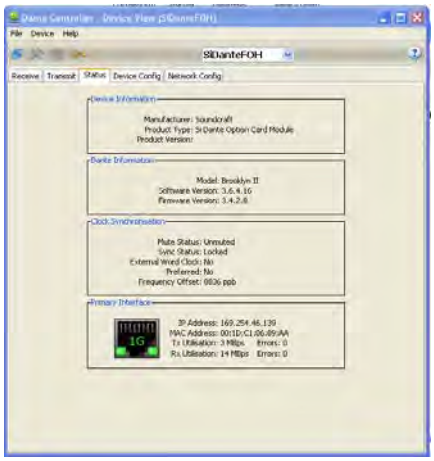
Available Channels: On the right pane of the window, lists the Tx channels from other devices that are available for subscription by the current device. You may drag and drop *Available Channels* to *Receive Channels.*

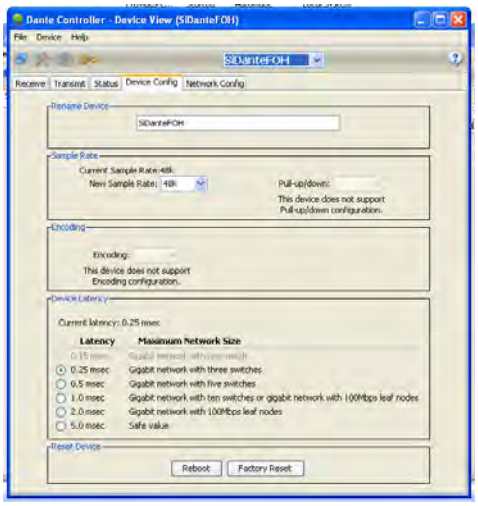
Transmit: This tab is used to view and edit the transmit configuration of a device.

Transmit Channels: The area on the left pane of the tab shows the Tx channels for the device, and any user-defined channel label. It allows the user to create labels for transmit channels. Input to the table is filtered to prevent illegal characters from being used in channel labels. To change a Tx channel label double click on the label and type in a new one. Tx channel labels must be unique on a single device.



Status: The Status tab is used to view information about a Dante device, this information may be required for trouble shooting and checking software / firmware is up-to-date.





<u>Device Config</u>: This tab on the device view allows viewing and editing of the following device specific parameters.

Rename Device: Allows renaming of the Dante device.

Sample Rate: 48K – this is fixed and cannot be edited.

Latency: Allows selection of an appropriate latency.

NOTE: The 0.15 msec setting is unavailable on Vi Dante cards as these include an internal switch.

Factory Reset: Allows remote reboot of the Dante interface and to reapply factory settings. Following a reboot it is recommended to power-cycle the console including disconnection of the AC power cord. Factory Reset wipes the following device configuration settings:

- User-defined device name
- User-defined channel labels
- Clock configuration (clock master / external clock master setting)
- Static IP addresses
- Redundancy configuration
- Sample rate setting (including pull-up/down)
- Latency setting
- Any existing audio routes

Network Config: This tab enables viewing and editing of Dante Redundancy mode.

Switched Mode - When set to Switched, the secondary Ethernet port will behave as a standard switch port, allowing daisy-chaining through the device.

Redundant Mode - When set to Redundant, the device will duplicate Dante audio traffic to both Ethernet ports, allowing the implementation of a redundant network via the secondary port. In this case care must be taken to avoid accidently making any connection *between* the two parallel networks.

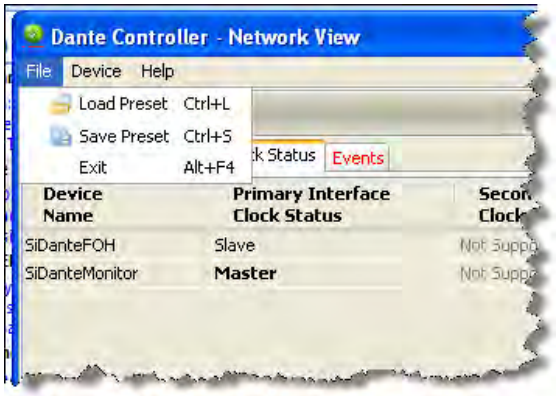
Note: All devices on a Dante network MUST be set to the same Mode (Switched or Redundant), otherwise the network will not function reliably.



Dante Presets

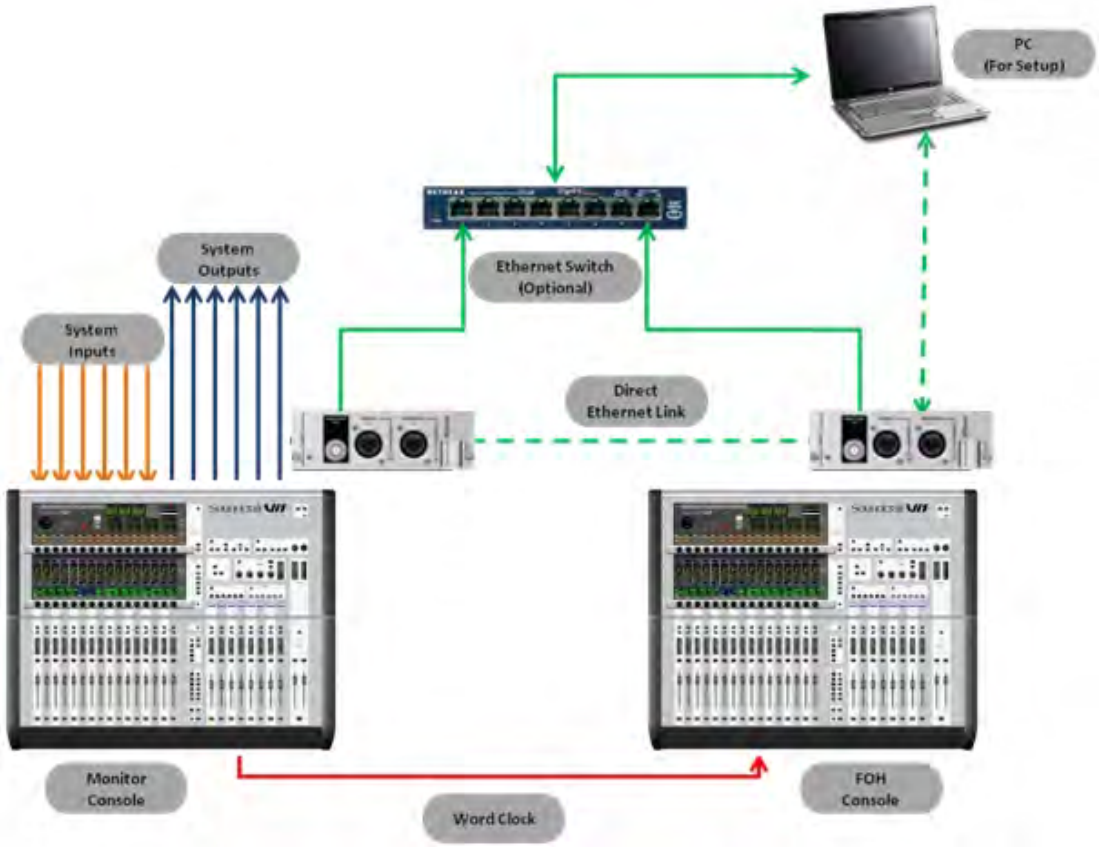
Although patching to and from the Dante card in the console is stored as part of a show or part of a cue the *configuration and routing* of the Dante cards and Dante audio streams is not.

In the event a Dante card is replaced or similar, the 'new' card will require configuring, this may be done manually or by loading a previously stored preset from the 'File' menu of the Dante Controller software



Connecting Dante Devices

The following example illustrates two Vi1 consoles 'House' and 'Monitor'.



Audio inputs are sent to the monitor console, these can be connected directly to the console or via a stagebox attached via a MADI card to the monitor console.

A direct output from each input of the monitor console feeds the FOH console which is patched to receive audio from the Dante option card.

The FOH console mix is returned to two spare channels of the monitor console and the direct outputs of those inputs are in turn patched to the system outputs; once again these can be directly on the console or attached via a MADI card to the monitor console.

The link between the Dante cards may be via a switch as shown with a solid green line or direct from card to card as shown by the dotted green line.

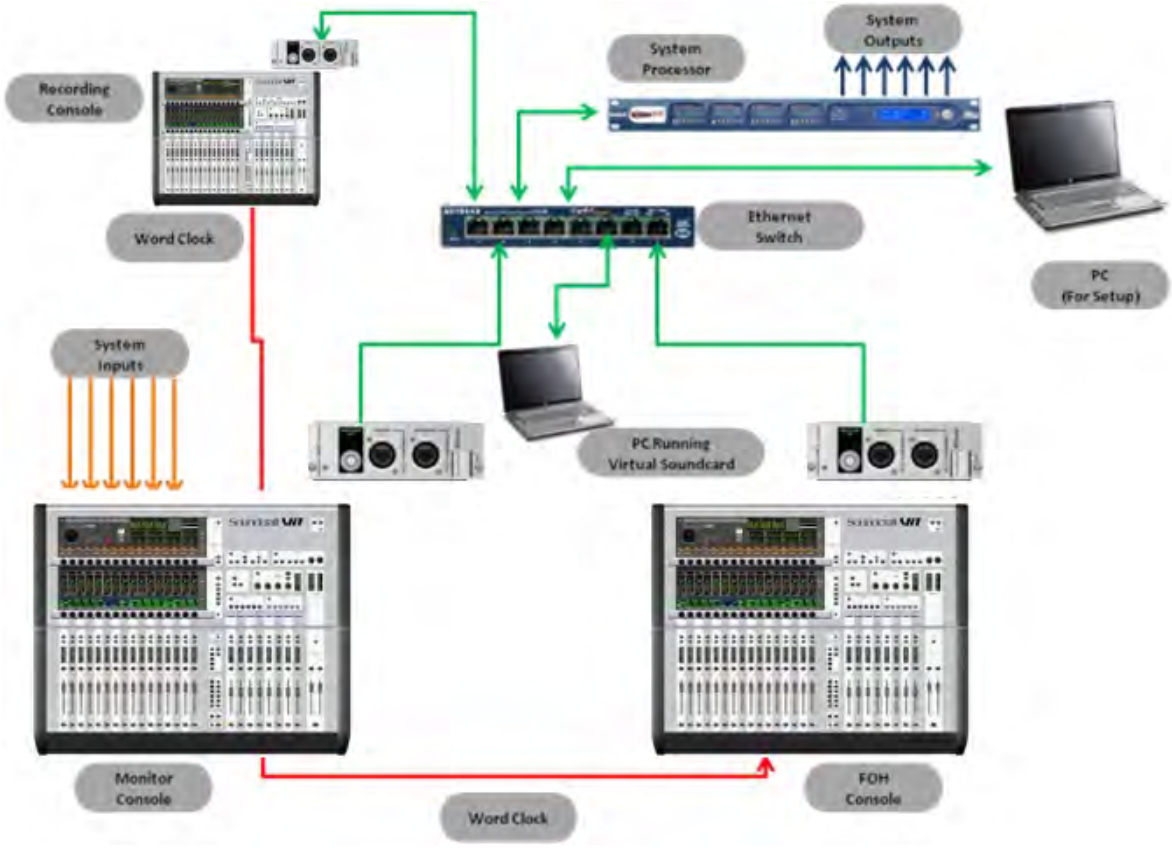
NOTE: Once configured the PC is not required.

NOTE: Dante may operate in redundant mode, this is not shown for simplicity.

Alternative Options

Once audio is in the Dante domain the possibilities become near endless, in this alternative system:

- All system outputs leave via a Dante enabled processor such as the BSS BLU 806
- An additional console does a live stereo mix-down for web streaming or similar
- A laptop running the Audinate Virtual Soundcard makes a multi-track recording



Soundcraft®

Vi Connect

Flexibility and Expansion

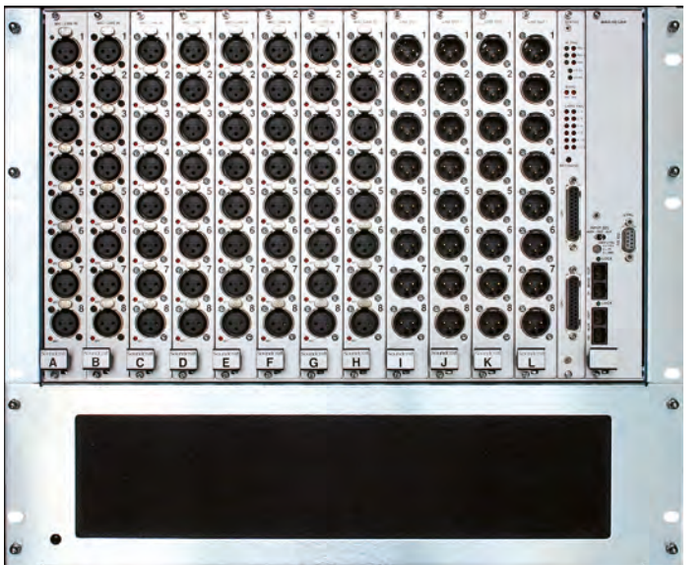
Option Cards, I/O Solutions, and Accessories for

Soundcraft Vi Series consoles

ViSi CONNECT

Stageboxes

EXPANSION OPTIONS FOR SOUNDCRAFT DIGITAL MIXING CONSOLES.



Vi STAGEBOX

The 9U Vi Stagebox houses 64 mic/line inputs and 32 analogue line outputs. Mic/line amp gain, 48 V phantom power and 100 Hz lo-cut filter before the A-D converters can all be controlled remotely from the desk.

The Stagebox is available in Standard 64/32 configuration, or can be reconfigured as part-fitted or with alternative cards as required.

Optional AES/EBU, CobraNet®, Ethersound®, Aviom A-Net® 16, Blu Link or Dante cards are available, as well as transformer balanced options for the mic input cards.

PART NUMBERS:	CAT5 VERSION:	RW5786C
	OPTICAL VERSION:	RW5786CO

COMPATIBILITY:			
Vi1	✔	Vi4	✔
Vi2	✔	Vi6	✔



COMPACT STAGEBOX (CSB)

The Soundcraft Compact Stagebox offers a high density of I/O connections in only 4U of rack space. The modular unit is configurable but is offered with a standard configuration of 32 mic/line inputs, 16 line outputs, and 2 expansion slots for standard Vi Local Rack I/O cards. These D21m option cards allow connection to most popular digital formats, including CobraNet®, AVIOM A-Net®16, Ethersound, ADAT, RockNet, Blu Link, and Dante. A MADI recording interface can also be fitted to the expansion slots. It is possible to equip the Compact Stagebox with an additional 16 mic/line input XLR module instead of the output module, providing 48 inputs. In this case, analogue or AES outputs could still be obtained on D-Type connectors via D21m cards fitted to the expansion slots. The upper section of the stagebox is reconfigurable with a choice of 16 channel I/O modules: 16 mic/line inputs; 16 line out; 8 line out + 8 AES out; 16 AES in + 16 AES out.

The Compact Stagebox is connected to the host console using either Cat-5 or Optical-fibre MADI, the same way as the larger 64 Mic/line Vi6 Stagebox is hooked up, and shares the same redundant MADI cable capability. (For Vi1 a compatible card for the console is required to be purchased separately).

The unit comes complete with twin redundant power supplies, thermostatically-controlled fan cooling and full LED status monitoring. An 8ch GPIO interface is also provided.

PART NUMBERS:	
CAT5 VERSION	E947.350000
OPTICAL VERSION	E947.351000
OPTIONAL UPPER SECTION MODULES	ALSO USED ON Vi1 CONSOLE LOCAL I/O
16 MIC/LINE IN	Vi1-ML1
16 LINE OUT	Vi1-LO1
8 LINE OUT + 8 AES OUT	Vi1-AES8
16 AES IN + 16 AES OUT	Vi1-AES16

COMPATIBILITY:			
Vi1	✔	Vi4	✔
Vi2	✔	Vi6	✔

Stageboxes

COST-EFFECTIVE EXPANSION OPTION FOR SOUNDCRAFT DIGITAL MIXING CONSOLES.



MINI STAGEBOX (MSB)

The Mini Stagebox range from Soundcraft provides a simple solution to stage connectivity offering a lower-cost option for many systems where the modularity and configurability of the larger systems are not required.

The Mini Stagebox 32 (3U) provides 32 analogue inputs and 8 analogue line outputs, with 4 pairs of AES outputs, providing a total I/O capacity of 32 inputs and 16 outputs, with the Mini Stagebox 16 (2U) model providing 16 analogue inputs and 8 line outputs. They connect to any of the Soundcraft Si Expression, Si Compact, Si Performer, Si Series and Vi Series consoles via a MADI HD card fitted to the option card slot. The mic preamps used in the Mini Stagebox are identical to those in the Compact Stagebox and Vi1 console.

The Mini Stagebox 16 is fitted with a single-ended Cat5 MADI connection (so only one cable is required), while the Mini Stagebox 32 includes dual Cat5 connections which provide signal redundancy.

The Mini Stagebox models contain an integral PSU so are completely self-contained and sit comfortably stage-side.

PART NUMBERS:	
Mini Stagebox 16 Cat5 MADI	5036292
Mini Stagebox 32 Cat5 MADI	5036290

COMPATIBILITY:			
Vi1	✔	Vi4	✔
Vi2	✔	Vi6	✔

ViSi CONNECT

Option Cards (Local Rack/Compact Stagebox)

Vi SERIES

A series of optional interface cards, designed to fit either the local rack or Compact Stagebox, and in some cases also the Vi Stagebox, enable Vi Series™ consoles to interface directly with a wide range of other systems and audio networks using industry standard protocols.

LINE IN CARD



Eight-channel line input card with 24 bit, 48kHz A/D Converter, delta-sigma conversion. Transformer-balanced inputs. 7...26 dBu input sensitivity. "Signal present" LED indicator. Inputs on standard 25-pin D-type connector (female).

PART NUMBER:
ViO-LI

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✘

LINE OUT CARD



Eight-channel, 24 bit line output card with 24 bit D/A converters with 48 kHz operation. Electronically balanced outputs. 7...26 dBu max. output level. Outputs on standard 25-pin D-type connector (female).

PART NUMBER:
ViO-LO

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✘

MIC IN CARD

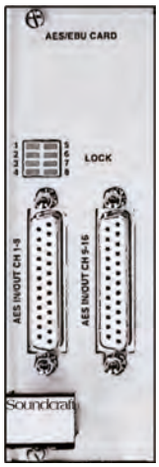


Four analogue microphone/line inputs, electronically balanced, with 24 bit, 48 kHz delta-sigma A/D converters (mic/line sensitivity, gainsetting in 1 dB steps, low-cut filter, and 48 V phantom power on/off controlled by console software). Green "signal present" and yellow "phantom power" indicators per channel. Inputs on standard 25-pin D-type connector (female).

PART NUMBER:
ViO-MI

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✘

AES/EBU CARD



AES/EBU input/output card with 16 Ch I/O. With input SFCs only. Input SFCs can be bypassed individually. Output dither is selectable for every AES/EBU output from 24 bit, 20 bit, 18 bit or 16 bit.

Settings are made with jumpers. Inputs and outputs on standard 25-pin D-type connectors (female).

PART NUMBER:
ViO-AES

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✘

Option Cards (Local Rack/Compact Stagebox)

3G SDI CARD



The 3G SDI card can de-embed up to 16 audio channels from an SDI (Serial Digital Interface) stream, and re-embed them (Embedder/DeEmbedder card only) back on to the stream for onward transmission in a broadcast environment, saving high costs of external units and, of course, weight. These channels may be patched in to any input as usual.

Two versions of the card are available: DeEmbed only, or Embed/DeEmbed

PART NUMBERS:	
DeEmbed Only	ViO-SDI
Embed/ DeEmbed	ViO-SDIO

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✘

Dolby E CARD



The Dolby E card decodes the audio channels from a Dolby E or Dolby Digital stream and allows them to be patched into separate inputs on the console for mixing. Each card provides two full decoder sections, and can dramatically save external hardware cost, space and weight.

PART NUMBER:
ViO-DOLBY

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✘

ADAT CARD



The ADAT card provides two optical eight-channel ADAT inputs and outputs, with 48 kHz operation. Optical inputs and outputs are provided on Toslink connectors and can be used to record to, for example, an Alesis HD-24 hard disk recorder or other device with ADAT inputs and outputs, and receive playback audio.

PART NUMBER:
ViO-ADAT

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✘

TDIF CARD



This card provides two 8-channel TDIF I/O interfaces with 48 kHz operation with wordclock sync outputs on BNC connectors. Inputs and outputs are provided on standard 25-pin D-type connectors (female).

PART NUMBER:
ViO-TDIF

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✘

Option Cards (Local Rack/Compact Stagebox, Vi Stagebox)

A-NET CARD



The **Soundcraft A-NET® 16V** card provides direct digital connection on a single Cat5e cable from the Aux or Group outputs into an Aviom® A-NET® Pro16™ Series Personal Mixer (compatible with A-16II and A360).

The Pro16 System enables musicians on stage to be fed with up to 16 subgroups of instruments or vocals which they can then mix to their own taste using their own on-stage controller. The card includes a switchable test tone and stereo pair linking via DIP switches.

.

PART NUMBERS:	
Local Rack/CSB	ViO-ANET
Vi Stagebox	ViSB-ANET

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✔

BLU LINK CARD



The Blu Link card provides up to 32 inputs and 32 outputs to/from a BSS Blu Link ring, and can be used to interface to BSS London networked audio systems, or to the PMC Personal Monitoring Controller.

The Blu Link redundant ring allows for up to 256 channels, so the Vi Blu Link card contains internal DIP switches that allow a block of 32 sequential channels from the 256 to be addressed by the card, separately selectable for inputs and outputs.

In addition the card can be set as clock master or synchronised clock, according to system requirements. When acting as a synchronised clock a jumper cable (included) must be connected from the card's wordclock out to the consoles wordclock input.

PART NUMBERS:	
Local Rack/CSB	ViO-BLU
Vi Stagebox	ViSB-BLU

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✔

DANTE CARD



The Dante card allows 64 channels to be sent to / received from a Dante network.

As well as providing audio exchange with other Dante-enabled equipment, the card also allows 64 channels of audio to be connected directly to a PC or Mac computer for recording or processing, by using a standard Ethernet connection*.

Routing and setup of the network is done using an external computer running Dante Controller software (available FOC from the Audinate website).

A BNC sync output connector on the card allows the console's wordclock input to be driven by the network clock, if required by the clocking setup.

**Requires Dante Virtual Soundcard software to be installed on the computer — one licence token supplied with each card.*

PART NUMBERS:	
Local Rack/CSB	ViO-DANTE
Vi Stagebox	ViSB-DANTE

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✔

CobraNet™ CARD



The **CobraNet™ card** allows up to 32 channels of input or output signals (or a combination of both) to be received by or sent from console onto a CobraNet™ network. The card must be used as the Conductor for the network. By default, the card is configured for 32 outputs.

PART NUMBERS:	
Local Rack/CSB	ViO-COB
Vi Stagebox	ViSB-COB

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✔

NOTE: Images show local Rack/CSB versions — Vi Stagebox versions have same connections but have 6U high front panel.

Option Cards (Local Rack/Compact Stagebox, Vi Stagebox)

EtherSound CARD



The Digigram EtherSound card allows up to 128 channels of audio (64 inputs/64 outputs) to be connected to the rack via a single Cat5 connection. A second Cat5 connection allows daisy chaining or full redundancy capability, ensuring the network remains uninterrupted should a cable be broken.

Available in both stagebox and local rack versions. This card is only available via Digigram's distribution network.

PART NUMBER:
Available from Digigram

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✔

MADI Cat 5 CARD



Two types of MADI card are available, both offering 64 channels of audio on dual redundant cables.The Cat 5 version offers cost-efective connections to a Soundcraft Stagebox using Cat 5 cable terminated in robust Ethercon connectors. Distances of up to 120m are possible using approved Cat 5 cables (see list below).

PART NUMBER:
ViO-C5

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✘

APPROVED CAT 5 CABLES	
TYPE	MAX. DISTANCE
NEUTRIK (CONTRIK) ETHERFLEX	80m
EVOLUTION XPC TACTICAL Cat 5E	100m
KLOTZ RC5SB	120M

Soundcraft does not guarantee reliable operation with any other type of Cat 5 cables.

MADI Optical CARD



The optical MADI card provides the same functionality as the Cat 5 version but on industry-standard SC fibre connectors. Two duplex SC connectors allow dual redundant cabling with automatic switchover. Connection distances of up to 1500m are possible using the standard multimode card, but a version with single-mode optical transceivers is also available allowing distances of up to 10km.

PART NUMBER:	
Multimode	ViO-MO
Singlemode	ViO-SO

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✘

GPIO CARD



For general-purpose switching applications requiring total electrical isolation, this card provides 16 electrically isolated opto-coupler inputs with integrated current sink (5...24 VDC) and 16 electrically isolated outputs using SPST relay contacts. 5 VDC supply pins are available. Inputs and outputs on standard 37-pin D-type connectors (female).

PART NUMBER:
ViO-GPIO

COMPATIBILITY:	
Local Rack/CSB	✔
Vi Stagebox	✘

Option Cards (Vi Stagebox only)

MIC IN (x8) CARD



The Mic In card provides 8 channels of classic Soundcraft mic preamps with 10-65dB gain range, switchable 20dB PAD, 80 Hz high-pass filter and 48V phantom power with indicator LED. A transformer balanced version is available to special order.

PART NUMBERS:	
STANDARD	ViSB-ML
TRANSFORMER BALANCED	ViSB-MLTX

COMPATIBILITY:	
Local Rack/CSB	✘
Vi Stagebox	✔

LINE OUT (x8) CARD



The Line out card provides 8 electronically balanced and protected analogue outputs. Maximum output level is +22dB

PART NUMBER:
ViSB-LO

COMPATIBILITY:	
Local Rack/CSB	✘
Vi Stagebox	✔

AES IN CARD



The AES In card provides 8 channels of AES/EBU inputs via 4 balanced XLRs. Sync LEDs indicate a valid input lock. Each input pair incorporates sample rate converters which can be bypassed, using internal jumpers, if required.

PART NUMBER:
ViSB-AESI

COMPATIBILITY:	
Local Rack/CSB	✘
Vi Stagebox	✔

AES OUT CARD



The AES Out card provides 8 channels of AES/EBU outputs via 4 balanced XLRs

PART NUMBER:
ViSB-AESO

COMPATIBILITY:	
Local Rack/CSB	✘
Vi Stagebox	✔

Soundcraft
by HARMAN

ViSi Remote

REMOTE CONTROL iPAD APP FOR SOUNDCRAFT DIGITAL MIXING CONSOLES

The Soundcraft ViSi Remote is an iPad app which provides wireless control of Soundcraft Si Expression, Si Compact, Si Performer and Vi Series consoles. ViSi Remote allows you to mix and contour your sound from anywhere in a venue without being constrained by cables.



Systems can be configured to run in several ways.

1. A single iPad can control one console's levels and graphic EQ.
2. Multiple iPads can control a single console's settings. This means that artists can each control their own monitor mix directly from their performance position on stage.
3. A single iPad could control a number of consoles in a network, which may include any of the consoles noted in the above range, so Si and Vi consoles may be mixed on a network.
4. The ViSi app may be used as an offline familiarisation tool.

Simply connect a Wireless Router* to the HiQnet port on the console and mix remotely within minutes.

** Soundcraft does not guarantee operation with **all** wireless routers.*

- Optimise the front of house mix from anywhere in the room.
- Adjust monitor levels while standing next to the artist.
- Control input, aux and output levels and graphic EQ settings.
- Use in standalone mode for familiarisation with app functions.
- Control a network of consoles from one iPad app (e.g. FOH and Monitor): Vi1, Vi2, Vi4, Vi6 (requires console software V4.6 or above).
- Use multiple iPads on the same console, so several artists can control their own monitor mixes.
- Console and audio unaffected by wireless dropouts or interference.
- Auto discovery of consoles on network — no manual address entry needed.
- Connects to consoles via Harman's HiQnet® network.

Soundcraft
by HARMAN

Soundcraft
by HARMAN

Soundcraft, Harman International Industries Ltd.,
Cranborne House, Cranborne Road, Potters Bar, Hertfordshire EN6 3JN, UK
T: +44 (0)1707 665000 F: +44 (0)1707 660742 E: soundcraft.marketing@harman.com

Soundcraft USA, 8500 Balboa Boulevard, Northridge, CA 91329, USA
T: +1-818-920-3212 F: +1-818-920-3208 E: soundcraft-usa@harman.com

www.soundcraft.com

Part No: 5042290